ACOUSTIC DETECTION PROTOTYPE USING GCC-PHAT ON MICRO-CONTROLLER

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Abstract: This paper presents an acoustic directional detect device's development and design based on the Generalized Cross-Correlation PHAseTransform (GCC-PHAT) technique. The topic's primary goal is to build a KIT that supports audio processing, including a 4-channel Microphone array and a popular microcontroller, STM32F103c8t6, embedded with GCC-PHAT algorithm to estimate the direction and intensity of the sound source captured. We also have tested, evaluated, and compared the performance with a sound source localization device on the market. The article focuses on data processing and optimizing computational methods to enhance performance while overcoming the limitation of the microcontroller.

Keywords: Microcontroller, STM32, sound source localization, GCC-PHAT, cross-correlation.

I. INTRODUCTION

Several studies have focused on developing humancomputer or human-mobile interfaces in intelligent environments to support user tasks and activities. Acknowledgement of voice/sound location and direction provides valuable information allowing for a better understanding of user activities and interactions in those environments, such as analyzing group activities or behaviour, deciding who is the active speaker among the participants, or determining who is talking to whom [1].

Along with a vibrant scientific and technological revolution, the sound processing engineering field has also experienced continuous development. A series of robots, AI manufactured by research institutes worldwide, has been announced with a great attraction. Their unique feature that makes the whole world pay attention is the ability to communicate and talk to people like tour-guide robot, Siri (Apple), Cortana (Microsoft), or Sophia (the first robot citizen of the world). In terms of technical aspects, to get the ability to communicate with people, the recognition of sounds and voices is indispensable in the processing process. The identification technique's quality is of utmost importance and directly affects the entire communication system's quality. Many methods and techniques developed to improve speech recognition quality, which is essential, are the sound source localization technique.

Many devices are capable of locating sound sources that have been presented using different methods to serve a variety of purposes in today's world. One of them is more specific noise source localization, such as aeroacoustic measurements in wind tunnels, flying aircraft, and engine testbeds[2]. Characterized by tests aimed at evaluating and verifying an aviation vehicle's efficiency, the methods used in this field place a heavy emphasis on accuracy and, in fact, use hefty hardware to achieve the most accurate results. Among them are deconvolution methods such as DAMAS[3], CLEAN[4], LPD[5], beamformingbased methods like GIBF[6], RAB[7] or others like SEM[8], SODIX[9], IBIA[10]. Another application of sound source locating is speech recognition and speech separation[11] for human-machine interaction applications. The most prominent example in this area might be the ASIMO robot[12] using the HARK framework[13] or the Hadaly robot[14]. Other applications, such as multirotor-UAV[15], use the MUSIC method or the IHRTF[16] algorithm for robotic heads. However, one common feature of the applications mentioned above is that they are all production prototypes using tightly linked hardware and software. On the other hand, heavy and expensive hardware is also a limitation for new researchers entering this field. Realizing this, STMicroelectronics released STEVALBCNKT01V1[17], also known as the BlueCoin kit. However, at $80, this device is not affordable for many people. Therefore, with the idea of creating a simple, affordable acoustic direction detection device, suitable for students, engineers, and researchers to assist the learning and research process. The topic content will present the device's development process and design to support the sound direction of arrival (DOA) detection with a simple microcontroller's hardware platform and localization algorithm.

This paper is composed of 5 sections. Section II summarizes the theoretical basis. The process of optimizing hardware and software is presented in section III. While section IV covers system design and evaluationg, finally concludes are presented in section V.

II. THEORETICAL BASIS

One of the most common ways to estimate DOA is to exploit the time-difference-of-arrival (TDOA) of the sound wave to a microphone array. Its popularity base on the fact that it can be modified to match the researcher's

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interest, such as one or two-dimensional localization of single or multiple sound sources. But to understand the basic principle, let's look at an example on a pair of microphones.

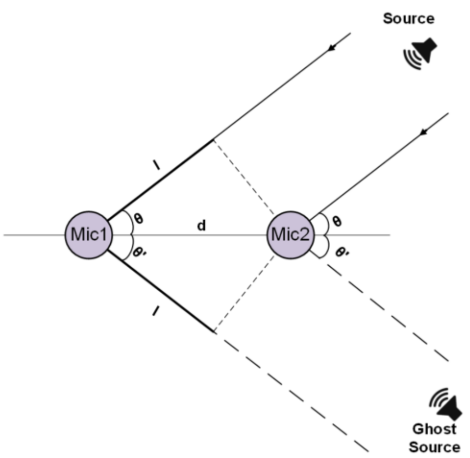

Figure 1. Utilized feature example

In a free/far-field propagation model, as shown in Fig. 1, there is a sound source which is transmitting a sound signal to two microphones 1 and 2. Consider the sound wave that approaching the microphone pair is a planar wave, which means there is an amount of time difference of arrival (TDOA) between these microphones let us call it t_{delay}. Therefore, there is a simple feature-to-location mapping between t_{delay} and the DOA of the source.

$$\theta = \arccos\left(\frac{l}{d}\right) = \arccos\left(\frac{V_{sound} \times t_{delay}}{d}\right) \tag{1}$$

where θ is the angle created between the sound source direction and the imaginary line between the microphone pair; d is the spacing of the microphone pair; l is the distance difference of arrival between Mic1 and Mic2, which is constructed from the multiplication of speed of sound V_{sound} = 343.2(m/s) and TDOA as t_{delay}.

However, with each obtained *t*delay, there is another DOA result that can be produced as θ', which mirroring the actual source. This ghost source appearance is addressed as an ambiguity problem which has been existed since the beginning of the overall binaural approach. To avoid this, in section III-A, we will provide a simple square array solution and how to extract the incoming direction from it.

TDOA processing mentioned in the article is an effective way to solve the deviation between microphones. Many algorithms, such as GCC, GCC-PHAT, and DFSE, can effectively estimate TDOA. However, we have chosen the GCC-PHAT method for this research because the PHAT method has better performance in Gaussian noise distribution and the actual noise environment. GCC-PHAT has also required less computational resources, which is essential when executing on a microcontroller.

A. Cross correlation in time domain (CC_T)

In signal processing, cross-correlation is a measure of the similarity of two signal chains or like a function of the displacement between one signal series relative to the other. It is commonly used for searching a long signal for a shorter, known feature. It has applications in pattern recognition techniques, electronic tomography, averaging techniques, decoding, and neurophysiology.

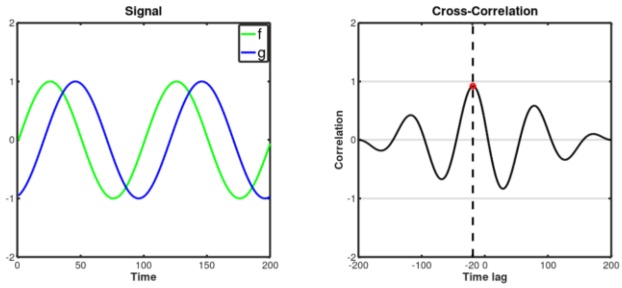

Figure 2. Cross-Correlation example

Consider two real-valued functions, f and g, differing only by an unknown shift along the x-axis. One can use the cross-correlation to find how much g must be shifted along the x-axis to make it identical to f. The formula practically slides the g function along the x-axis, calculating the integral of their product at each position. When the functions match, the value of $(f * g)$ is maximized because when peaks (positive areas) are aligned, they make an enormous contribution to the integral. Similarly, when troughs (negative areas) align, they also make a positive contribution to the integral because the product of two negative numbers is positive.

For example, with the Fig. 2, the two sine wave signals, f and g, have been deviated by 20 unit in the time domain, their cross-correlation result has peaks located at position -20 in time lag axis. This implied that if we shift g sooner by 20 unit, we will get f.

However, this cross-correlation execution has been proved to be very sensitive to reverberation and noise. In these cases, the cross-correlation results are "spread" to other TDOA regions, making it difficult to determine the right time difference. On the other hand, these resulting peaks are quite dull, causing in low certainty.

B. Cross correlation on frequency domain based on GCC-PHAT

The researchers have proposed cross-correlation in the frequency domain (CC_F)[18] to solve the above problem as an alternative to conventional crosscorrelation. This technique gave a different result F-1 (CC_F) with cross-correlation in the time domain CC_T, but it has been shown that the positions of the peaks match each other.

The primary purpose when doing cross-correlation over the frequency domain is that we can apply some weighting functions to the result; this technique is called general cross-correlation (GCC_F). So if this weight is equal to the magnitude of the correlation itself, we will normalize the magnitudes in GCC_F, causing the inverse Fourier transformation results to have peaks in the Dirac delta function approximate form in the high correlation TDOA. This normalization left the information about the phase intact, so it is called phase transform (PHAT).

$$GCC-PHAT_F(f) = \frac{Y_0(f) \times Y_1(f)^*}{|Y_0(f)||Y_1(f)|} \tag{2}$$

The $GCC\text{-}PHAT_F$ is a cross-correlation in the frequency domain incorporate with phase transform. To put this in perspective, we show the theoretical comparison results (Fig. 4)[19] of two similar input signals, which is considered an audio source with one is delayed intentionally. Both graphs show the same deviation peaks, but in the graph (b) we can easily see that it has a sharper peak and is more recognizable.

Considering a more interesting case as the second signal is an interfering component. Specifically, in the follow-up investigation, the signal was virtually experiencing reverberation with a known delay. The graph below (Fig. 5) shows the cross-correlation results of both methods in this situation.

In graph (a) with the conventional cross-correlation, it was proved that it failed to separate the delay between the echo and the actual signal, directly affecting the obtained deviation's quality. In contrast, with GCC-PHAT, we separated two signals and obtained two separate sharp peaks, and showed the signal's exact delay (Fig. 5(b)). GCC-PHAT distinguished them because it had sharp peaks that did not spread to neighbouring peaks like time-domain cross-correlation.

Thus, with the GCC-PHAT technique applied to the system, the noise and echo problem have been resolved, providing a more stable system and can be used in many different environments.

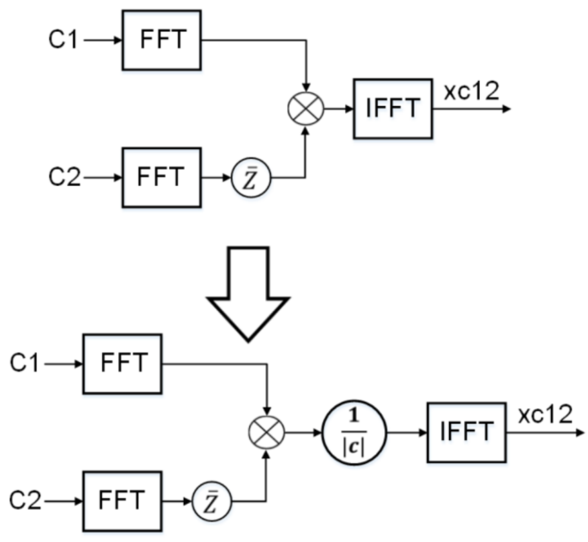

Figure 3. GCC-PHAT transformation

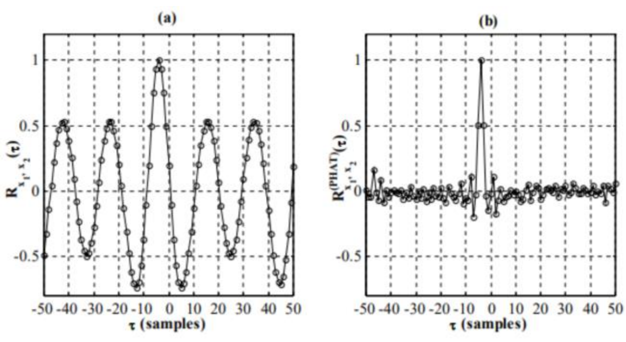

Figure 4. Result difference between two method

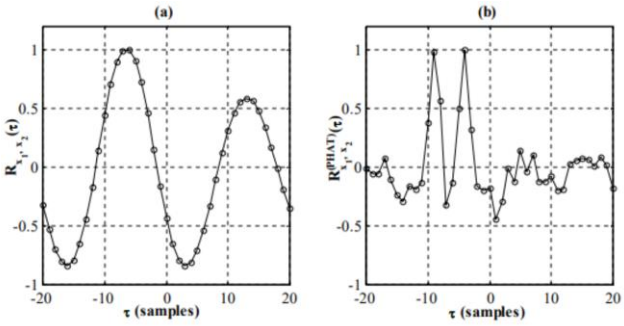

Figure 5. Result difference between two method in reverberation environment

III. PROPOSED DESIGN

A. TDOA to DOA

As mentioned in section II, if we only use two microphones or a microphone array lined up in a straight line, we will be challenging to determine whether the sound is coming from the front or the back of the array(ambiguity). Therefore, to define TDOA on a plane (specifically in this topic is the azimuth plane), we need to have a microphone array made up of multiple microphone pairs to form a parallel plane to the detecting one.

To detect the sound source in a 1-dimensional plane, we consider a set of 4 microphones placed at the four angles of a square. Fig. 6 shows the layout of the Microphones on the device. The distance between the microphone pairs is a constant D, assuming that the sound source is the only source, no noise. With a free/far-field propagation model, the sound wave can be considered planar, its direction creat with the sides of the square two angles: α and β. The value l_1, l_2, indicates the distance difference between two microphone pairs to the sound source. Obtaining this difference, we can calculate the angles α and β or the DOA of the source.

$$
\begin{aligned}
\alpha &= \arccos\left(\frac{l_1}{D}\right) \\
&= \arccos\left(\frac{V_{sound} \times \tau_{12}}{f_{sample} \times D}\right) = \arccos\left(\frac{V_{sound} \times \tau_{34}}{f_{sample} \times D}\right) \\
\beta &= \arccos\left(\frac{l_2}{D}\right) \\
&= \arccos\left(\frac{V_{sound} \times \tau_{23}}{f_{sample} \times D}\right) = \arccos\left(\frac{V_{sound} \times \tau_{14}}{f_{sample} \times D}\right)
\end{aligned}
\quad (3)
$$

However, if we only use the angle α or β, the ambiguity problem mentioned in section II begins to emerge. So using the microphone array as above will allow us to combine the resulting angle obtained from each microphone pair to produce a final result as μ, with

$$\tan(\mu) = \frac{\sin(\mu)}{\cos(\mu)} = \frac{\cos(\beta)}{\cos(\alpha)}$$
$$\mu = \arctan\left(\frac{\cos(\beta)}{\cos(\alpha)}\right) \tag{4}$$

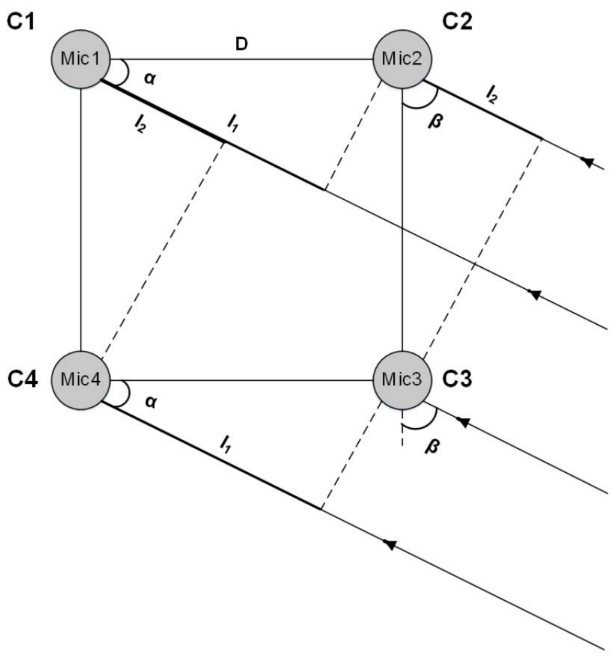

Figure 6. Array layout of the proposed device

B. Overall algorithm

So, after calculating the cross-correlation between two signals as mentioned in section II-B, the maximum value of the cross-correlation function will indicate when the signals are best aligned, or the time delay between the two signals is determined by the maximum value or the maximum angle of the cross-correlation.

With this information, it is possible to exploit and calculate the incoming sound direction with a more extensive microphone array system, as mentioned in section III-A. From there, we created a block chart representing the calculation process of the system as follows.

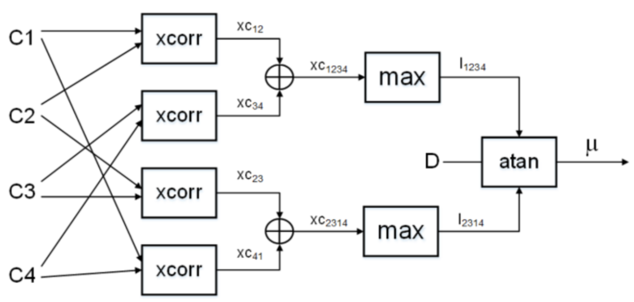

Figure 7. Overall algorithm of the device

Fig. 7 is the block diagram of processing audio signals captured from the microphones. In turn, each microphone pair will be cross-correlated and determine the deviations in time between them. The xcorr blocks in the figure are cross-correlations; through the array layout in fig. 6, we can see the results of the correlations xc_{12} ~ xc_{34} and xc_{23} ~ xc_{14}. Therefore, we add these two cross-correlations together to get xc_{1234} and xc_{2314}; the goal is to reduce later calculation and, at the same time, reduce the error of the cross-correlation. The Max blocks are the blocks to find the highest peaks to obtain the corresponding difference of l_{1234} and l_{2314}. From there, by some trigonometric formulas related to the microphone array spacing D, we can calculate the incoming angle of the sound μ.

C. Methods to enhance processing accuracy

Since the entire algorithm will be executed on a microcontroller, we will encounter some limitations that affect the overall system's efficiency. For that reason, in the research and development process, we have performed some investigations, tests to find out about these limitations and how to solve them, eventually improve the performance.

1) DC Offset Removal

The DC Offset Removal technique is a technique to remove the mean amplitude deviation from the coordinate axis. In other words, the central horizontal axis of audio signals we receive does not match with the horizontal axis of the coordinate (Fig. 8).

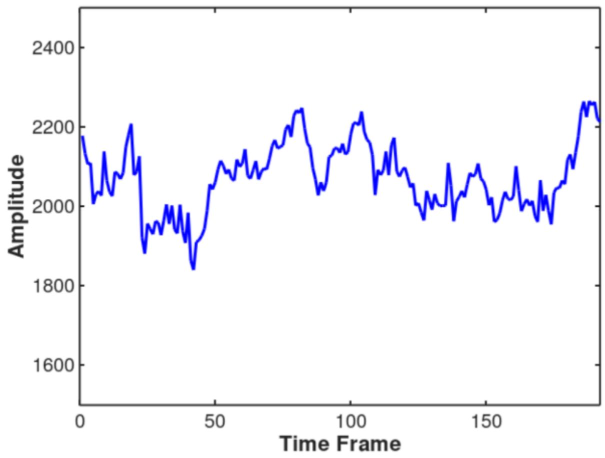

Figure 8. The DC Offset problem

In particular, this error occurs because there is always an electric compensation fixed somewhere in the sound signal before it is converted from analog to digital. These offsets are usually so small that it is not noticeable, but it can become large enough to be a problem on low-quality hardware.

There are many ways to solve this problem; the simplest one is that the signals after ADC will be subtracted with its average value using the Moving Average method. As a result, the signal will align with the horizontal coordinate axis.

$$S_t = \begin{cases} Y_1, & t = 1 \\ \alpha \cdot Y_t + (1-\alpha) \cdot S_{t-1}, & t > 1 \end{cases} \tag{5}$$

2) Signal interpolation

Another limitation worth mentioning when using lightweight hardware is that the analyzed data will be discrete. Therefore, the resolution of the obtained result will depend significantly on the number of samples taken. However, with limited hardware, this number must be carefully controlled to ensure the device's performance. Therefore, to increase the resolution of the results without affecting the device, we decided to use the interpolation method.

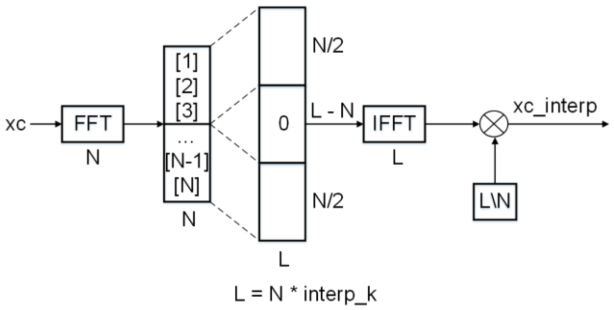

Figure 9. Interpolation process

After the cross-correlation is completed, we interpolate immediately to optimize computing FFT and IFFT. Moreover, with the above transform, we can cancel the inverted Fourier transforms of the cross-correlation block with the interpolation block's Fourier transforms.

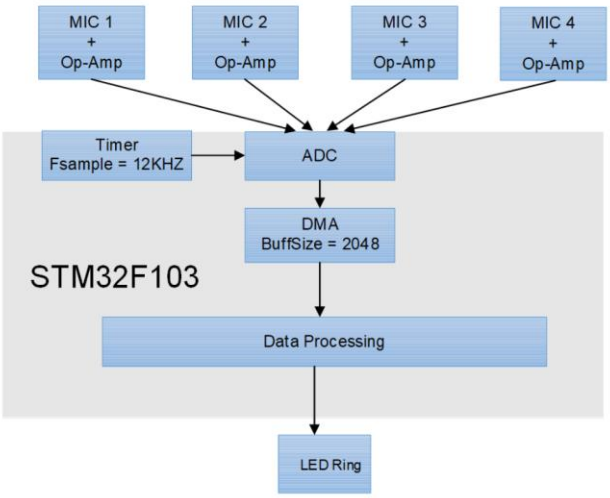

Figure 10. System design

IV. SYSTEM DESIGN AND EVALUATION

A. System design

The diagram in Fig. 10 describes the audio directional detection device general model built on the STM32F103c8t6 microcontroller. The audio signal is sampled by the STM32F103c8t6 microcontroller from four microphones that already incorporated an op-amp. The microcontroller's ADC samples the signal with a sampling frequency of 12KHz to obtain a series of digital audio signals. These are stored directly into BuffSize with a capacity of 2048 samples right after the ADC conversion is completed. DMA technique helps to reduce the information storage process, allowing the data to be calculated immediately. The signal through the processing unit using the algorithms described above will result in the incoming angle and intensity. These two values will be displayed to the user via the LED Ring module. Figure 11 is a picture of the audio direction detection device.

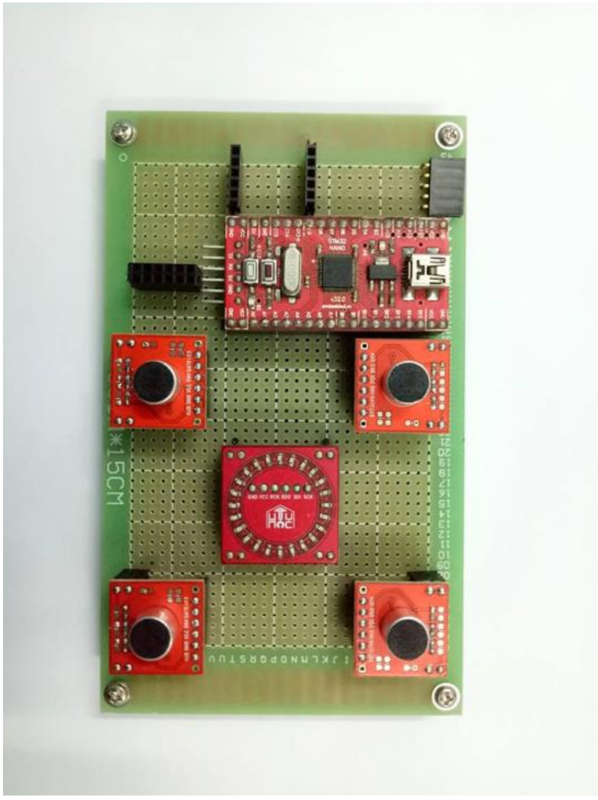

Figure 11. Acoustic detector prototype

B. Experiment and evaluation

To examine the device's performance, we operate a test with the sound source placed at ten random angles (Fig.12), with each angle, the source play at different distances, respectively. Table I contains the results obtained from the device at each run. When analyzing the experimental results, we can see that the device performs relatively uniformly in all the angular cases on the azimuth plane. With sound sources play at a longer distance, the return error is more significant, which is expected. At a distance below 40cm, the device produces very accurate results at 1.47^o to 2.40^o of average error. From the distance of 80cm-100cm, the accuracy of the device is still very high with an average error of about 3^o.

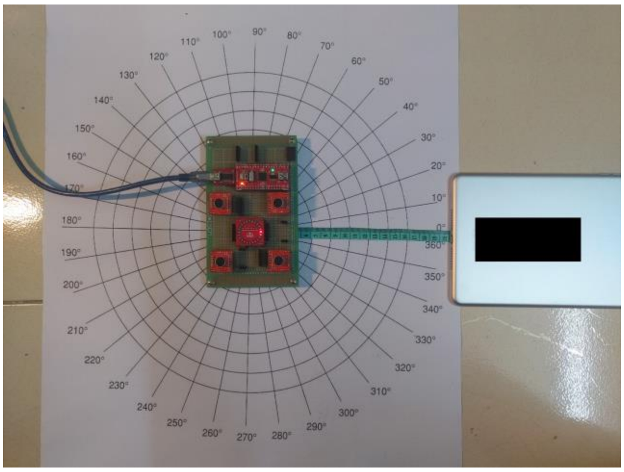

Figure 12. Experiment setup

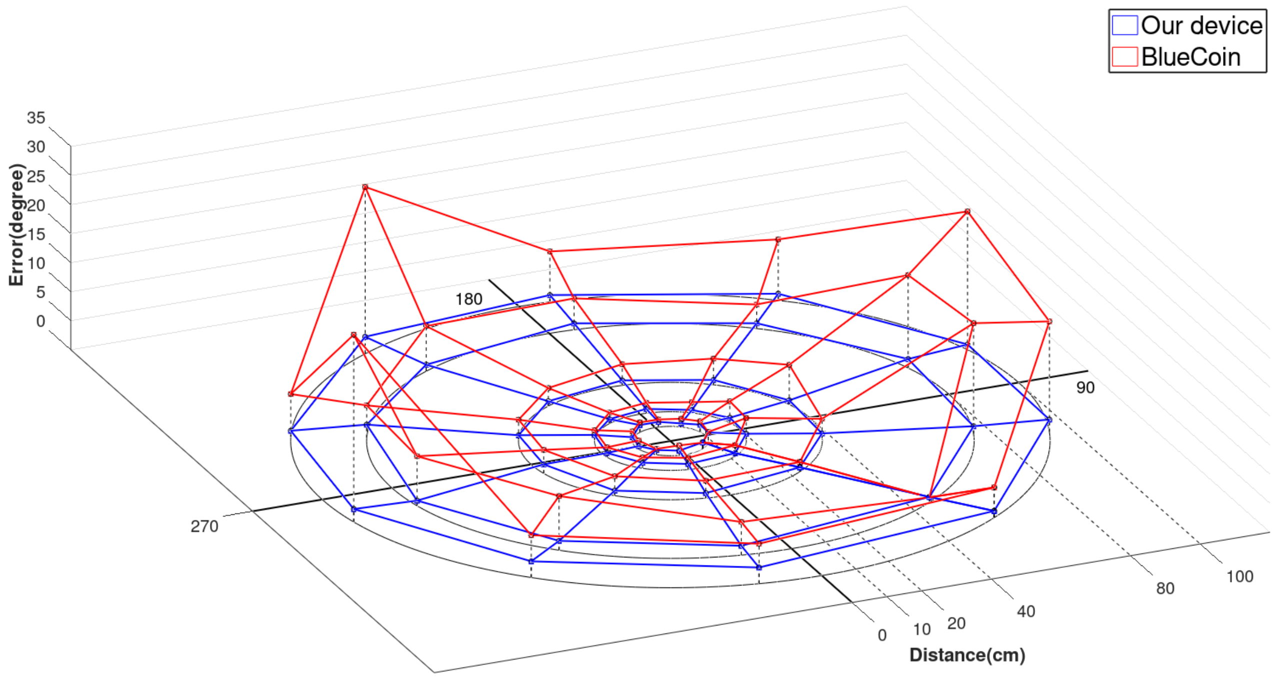

Figure 13.Performance in comparison

To put this in perspective, we compare it with an audio-source localization device, ST-BlueCoin, which is developed and competing in the market also by STMicroelectronics. The results showed the difference in performance between the two in terms of accuracy (Fig. 13).

At a distance of 10cm, both devices work well and both give accurate results when the two graph lines showing errors are low and almost identical. The difference begins to appear at a distance of 20-40cm when the BlueCoin device begins to show discrepancies in calculating the direction of the sound. At a distance of 80-100cm, the BlueCoin device is no longer stable while the device we developed still retains high accuracy.

Table 1. The results of the experiment

	Distances (cm)				
Angle	**10**	**20**	**40**	**80**	**100**
35	34.502	35.554	36.558	38.398	38.072
70	67.490	65.127	69.175	70.112	75.311
105	104.32	106.61	108.64	110.15	111.05
140	139.38	139.24	137.67	138.09	138.73
175	173.59	175.05	175.48	173.85	171.36
210	209.39	213.98	212.34	213.87	210.27
245	245.36	247.43	247.80	248.72	248.73
280	284.64	286.62	284.53	288.04	288.86
315	313.39	313.43	315.22	317.69	316.94
350	351.75	348.49	351.16	350.26	346.7

V. CONCLUSIONS

We discussed the solution to detect the sound source direction in 2-dimensional space using the 4-Microphones square array model via the STM32F103c8t6 microcontroller. We used the proposed GCC-PHAT method to improve the quality of the latency estimation algorithm. The GCC-PHAT technique was chosen because it performs well in environments with high noise density. Signal filtering and interpolation have also been added to improve accuracy. Survey results have shown the relative accuracy of the device and its quality compared to a previously developed device. The device is just designed and presented as a prototype, so this opens up a lot of new research directions for further development. For example, we can extend the effective working range of the device in terms of spatial dimension and distance. On the other hand, we can also make more optimal use of the hardware's working capabilities, eliminate redundant details, and reduce the physical size of the hardware and the complexity of the program. Furthermore, the device can act as a module in a higher purpose system such as audio source tracking, separation and recognition.

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PHÁT HIỆN HƯỚNG ÂM THANH SỬ DỤNG GCC-PHAT TRÊN VI ĐIỀU KHIỂN

Tóm tắt: Bài báo này trình bày quá trình nghiên cứu và thiết kế một thiết bị phát hiện hướng âm thanh dựa trên kỹ thuật tương quan chéo kết hợp biến đổi pha (GCC-PHAT). Mục tiêu chính của nhóm nghiên cứu là xây dựng một bộ KIT hỗ trợ xử lý âm thanh, bao gồm mảng Micro 4 kênh và bộ vi điều khiển STM32F103c8t6, được nhúng thuật toán GCC-PHAT để tính được hướng và cường độ của nguồn âm thanh thu được. Chúng tôi cũng đã thử nghiệm, đánh giá và so sánh hiệu suất với thiết bị phát hiện hướng âm thanh đang có trên thị trường. Bài báo này tập trung vào việc xử lý dữ liệu và tối ưu hóa các phương pháp tính toán nhằm nâng cao hiệu suất đồng thời khắc phục được những hạn chế của vi điều khiển.

Từ khóa: Vi điều khiển, STM32, phát hiện hướng âm thanh, GCC-PHAT, tương quan chéo.

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