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Performance Analysis of MUSIC Algorithm for Microphone Array Using FPGA board

Antipas T. Teologo Jr.[1], Reggie C. Gustilo[2]
[1]De La Salle University, Manila, Philippines, antipas_teologo@dlsu.edu.ph
[2]De La Salle University, Manila, Philippines, reggie.gustilo@dlsu.edu.ph

ABSTRACT

This study presents an implementation of the field programmable gate array (FPGA) and performance analysis of the multiple signal classification (MUSIC) algorithm for the direction-of-arrival (DOA) estimation of the different sound sources. DOA offers a variety of applications in the field of signal detection, communications, navigation and speech processing. MUSIC is one of the high-resolution DOA estimation techniques and the most promising technique for actual hardware implementation. A microphone array consisting of four equally-spaced microphones (sensors) is used as a basis for this study. As for wideband application, the number of sources should be less than the number of sensors thus a maximum of three sound sources are only considered and their locations are estimated. Four DOAs are used for single source, two sets of DOA for double-source and one set for triple source. Matlab simulations and test bench results are presented showing the effectiveness of the algorithm. All simulations and tests are done using five samples only because of the large logic utilization of the codes when comes to the FPGA implementation. This study utilizes an Altera Stratix II FPGA board for the realization of the MUSIC algorithm in hardware implementation and Verilog HDL serves as the programming language.

Key words: MUSIC algorithm, microphone array, audio processing, direction of arrival

1. INTRODUCTION

Nowadays, a vast choice of research topics is available in every corner of the globe. The great improvement in the fields of computer technology helps improve research and innovations. These improvements are greatly influenced by advanced researches communications engineering [1], [2], bio-medical engineering [3].[4], [5] and robotics engineering [6] ,[7], [8].

Aside from the research topics mentioned above, there is also a broad coverage of research topics for software engineering, application and development [9] and artificial intelligence using machine learning and deep learning [10].

In the field of wireless communications engineering, enhancing the capacity of the network and improving the quality of multiuser wireless communications system through the use of antenna arrays has been given enough attention in space-time signal processing techniques [11]. Array signal processing, which is covered under digital signal processing techniques, uses sensors that maybe organized in patterns, shapes or sensor arrays, to detect signals and be able to determine information about them and the most common application of this is in the detection of acoustic signals with microphones serving as the sensors. In any microphone array processing it always involves two main procedures, sound source localization and beamforming. Source localization is done to locate a sound source given signature or pattern of measurements of the sound field. Sound field can either be a sound pressure or a particle velocity. Beamforming, on the other hand, focuses the microphone array “beam” towards the source. Many studies have already been conducted for source localization using the direction-of-arrival (DOA) estimation. Knowing the exact location of a target using DOA estimation received much interest for the past decade because of the wide range of applications it offers such as in radar, sonar, communication, speech processing, oceanography and navigation [12].

There are several algorithms being implemented in order to compute for the spectral estimation of an incoming signal over a noisy channel. Among of those are the correlation, multiple signal classification (MUSIC), maximum likelihood estimator (MLE), estimation of signal parameters using rotational invariance techniques (ESPRIT) and matrix pencil. Since it is the most promising and a leading candidate for further study and for the actual hardware implementation, MUSIC algorithm was given more attention in this study. MUSIC algorithm is one of the high-resolution subspace-based methods which compute the DOA by dividing the cross-correlation matrix of the array signals into signal and noise subspaces through eigenvalue decomposition. With its wide range of applications, actual implementation in hardware of DOA is undeniably given much focus right now. But due to its computational complexity in real-time processing and high cost of implementation, it encounters some restrictions for its successful implementation.

Digital signal processing (DSP) plays an important role in this study. Almost all of the real-world applications today involve DSP ranging from high-definition TV, mobile telephony, digital audio, multimedia, digital cameras, radar, sonar detectors, biomedical imaging, global positioning, digital radio and speech recognition [13]. Much research has been conducted to develop DSP algorithms and systems for real-world applications. More recently, aside from the popular use of micro-controller chips in research [14] and mobile-based systems [15] [16][17], the field-programmable gate array (FPGA) has been proposed as a hardware technology for DSP systems as they offer the capability to develop the most suitable circuit architecture for the computational, memory and power requirements of the application. Presently, FPGAs have been gaining considerable attention in high-performance DSP applications, [18] [19] and are emerging as coprocessors for standard DSP processors that need specific accelerators. FPGAs offer a tremendous computational power by using highly parallel architectures for very high performance.

2. STATEMENT OF THE PROBLEM

In a typical wireless environment wherein many signal sources could be expected, determining the exact location of each signal is significantly considered. In some cases wherein closely spaced signal sources are present, knowing the number of possible sources is also vital. This is most prominent in radar applications wherein the position of the moving target is being determined and most radar systems use parabolic reflector antennas for signal transmission and detection.

Because of the increasing demand for more accurate and better performance in target detection, a much newer technology called the phase array technique is being introduced. With this technique, it uses an antenna composed of many elements such as dipoles which could be implemented as either a planar array or a line array. Each element of a phased array is phase shifted to steer the beam to the desired direction [26]. This technique can provide a robust and better detection performance than the reflectors but the cost of actual implementation of this is expensive. There are a lot of studies already done abroad regarding DOA estimation in antenna array but most of these investigations only involve software simulations to show the result. Only few have implemented it in actual hardware for real-time processing because of the complexity being involved. Since DOA estimation offers very promising applications, an actual hardware implementation using FPGA will be carried out in this study.

The main objective of this study is to implement the MUSIC algorithm that will estimate the direction-of-arrival of a sound source using a linear array of microphone sensors. Software simulations will be performed to validate the effectiveness of the algorithm and to test for its veracity in real-life applications, hardware implementation will be done using FPGA.

Using microphone array in determining for the DOA of sound source has some benefits. In a video conferencing or in a long distance TV-based classroom, DOA estimates can be used to automatically steer cameras to the speaker. On the other hand, having the sound signal captured from several points, it allows for spatial filtering (also called beamforming) if being given proper processing which could result to the amplification of signal originating from the specific directions, or the beam, and attenuating signals from other directions. This seems to be promising for the hearing aids of those people with deficiency in hearing as they could perceive better the origin of a sound in their surroundings. Speech enhancement for human computer interfaces that depend on speech inputs from operators, human machine interaction for robots and acoustic surveillance systems are also among the benefits of knowing the DOA. This study aims to implement in FPGA the MUSIC algorithm that will compute for a sound source's location using uniform linear array of microphones. Once successfully carried out, this would also be a good alternative for radar system using parabolic reflectors used in military applications wherein accurate target detection is vital.

3. MATERIALS AND METHODOLOGY

The following sections will illustrate the objectives of the research project and how there are implemented. Figure 1 shows the simplified block diagram of the system.

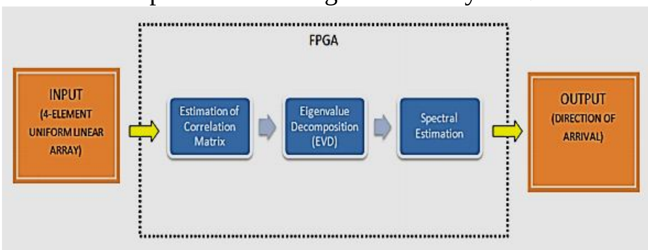

Figure 1: Simplified block diagram of the system

Figure 1 above shows the conceptual framework of the study. This provides an overview of the whole system. The four-element ULA serves as the input. After doing the necessary procedures to process the input data, the FPGA part will take into account implementing the three main steps of MUSIC algorithm. The output, which is the corresponding direction for a signal source(s), will then be displayed using the LEDs of the FPGA board. Matlab plays an important role in the input part. Input data is modelled first in Matlab and being simulated to check for the accuracy of the algorithm. The succeeding subsections will discuss in detail the methods or procedures done in each stage.

3.1 The Input Part

Basically, the input will be coming from the linear array consisting of four equally spaced sensors with distance d between each sensor as shown in figure 2.

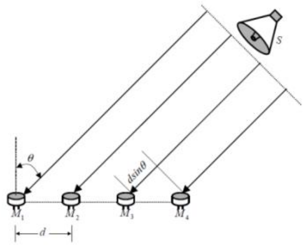

Figure 2: Uniform linear array with four elements

S, as the distance of the source from the array, is assumed to be greater than d so that we can approximate the spherical wavefront from the source as a plane wavefront. Therefore, it can be said that the sound waves arriving at each microphone are parallel to each other and with just a constant delay from each other. The direction perpendicular to the array is called the broadside direction or sometimes called the broadside of the array which will be the reference of the measurement of DOA. Angles to the right (clockwise) of the broadside direction are considered positive and those to the left are being negative. Since four elements have been used in the study, MUSIC can resolve up to three sound sources. But for the simplicity of the discussion, a single source only will be used as a reference.

3.2 Spectral Estimation

This is the last phase in the MUSIC algorithm. To perform this part, it is very important that the number of signal sources is known for Qn depends on the number of sources. Referring to equation (3-5), finding the possible DOA involves scanning the whole spectrum. Take note of the steering vector (θ). The scanning of the angle is set by one-degree interval, as shown in figure 3, starting from -90 degrees to + 90 degrees. In every angle, a steering vector is used in the formula. Using a one-degree interval means $P(\theta)$ will be computed 181 times and those highest values (peaks) will correspond to the estimated DOA of the signal.

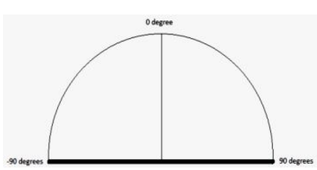

Figure 3: Spectral estimation through scanning of angles

3.3 Program Flowchart

The flowchart below shows how the programs works in an FPGA environment.

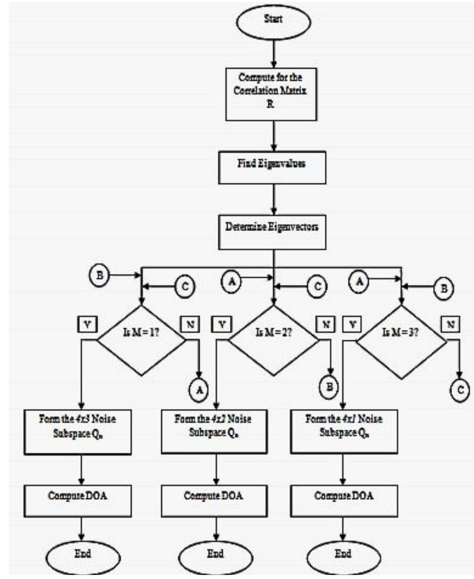

Figure 4: Simplified Flowchart of the Study

Figure 4 above shows the simplified flowchart of the whole study. The program will start after it receives the complete set of inputs. After getting all those values in the matrix X, it will now compute for the correlation matrix R. The program will wait until it receives a signal that matrix R is now complete. After that, determination of the eigenvalues follows next and their corresponding eigenvectors will then be computed. Before forming the noise subspace, knowing the number of sound sources (M) should be determined first. Having the knowledge now of how many sources are there, performing the spectral estimation to find the DOA will be the next step.

3.4 Hardware Implementation

The actual hardware implementation is shown in the figure below

Figure 5: Actual hardware set-up

Figure 5 shows the FPGA setup used in the research project. This FPGA module is interfaced into a laptop using the Quartus software. The input signal to the FPGA board is taken from the microphone array setup that is presented below.

3.5 Data Processing from Microphones

To realize its implementation for real-time processing, actual generation of data coming from the microphones were considered. The actual hardware set-up is composed of four microphones (served as sensors) spaced 5 centimeters from each other. This is shown in figure 6 below.

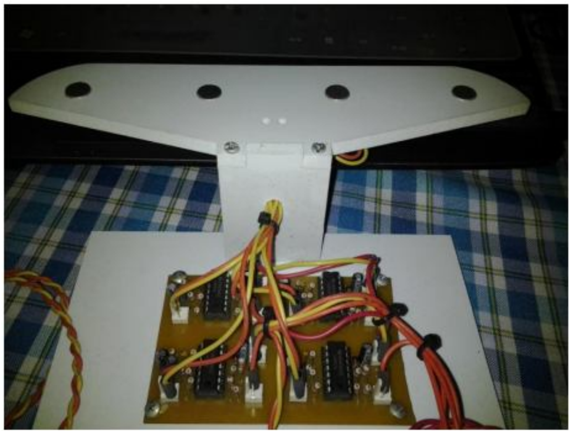

Figure 6: The actual hardware set-up for the four microphones.

The device is interfaced in the laptop thru the hypertext terminal to display the captured data. By pressing the ON button of the device, the microphones start to sample data and process the sampled analog signal into a 16-bit PCM output signals. The first line in the display represents the captured signal by the first microphone during the first sampling. Since there are four microphones, it is expected that during the first sample there are four outputs coming from the four microphones. So, the first four lines of data displayed, correspond to the captured signal during the first sampling for microphones 1, 2, 3 and 4, respectively. Considering that there are five samples, a total of 20 lines of data should reflect on the display.

4. RESULTS AND DISCUSSION

This part discusses all the results generated in this study and provides some analysis on the performance of the algorithm. Basically, there are three phases involved in this study. First, is the simulation in Matlab to check the reliability of the algorithm. Second, the algorithm is being coded using Verilog HDL and simulation results in testbench are provided. Third, to test the veracity of the Verilog codes, actual hardware implementation is being carried out. Angles used are -50°, -20°, 40° and 60° for single-source. For double-source, two sets are considered: (1) -30° and 50° and (2) -10° and 20°. Only one set is used for triple-source setup which is -40°, 10° and 70°. Only five (5) samples are taken on this study, that is K = 5.

5.1 Matlab Simulations

Multiple simulation scenarios are performed to analysis the performance of the MUSIC algorithm using microphone array as the input and FPGA board as the control system. A single-source simulation was done with a DOA of -50 degrees. The simulation result is shown in figure 7. Similarly, a single-source simulation is conducted with a DOA of -20 degrees as is shown in figure 8. The comparison between the simulated results and the actual DOA is shown in table 1.

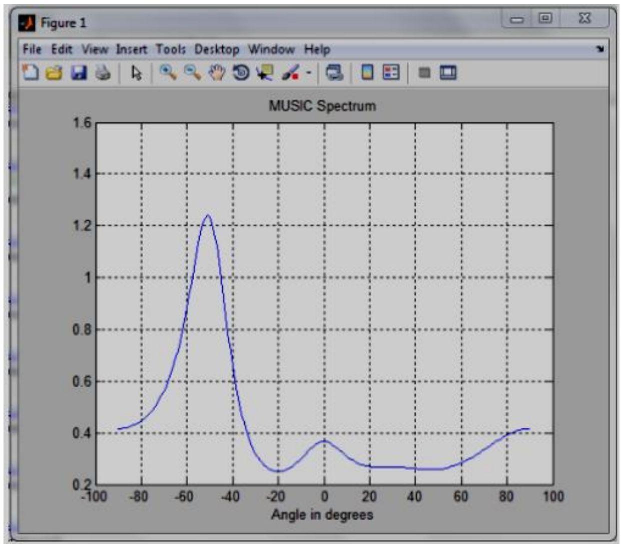

Figure 7: Matlab simulation for DOA of -50° at K=5

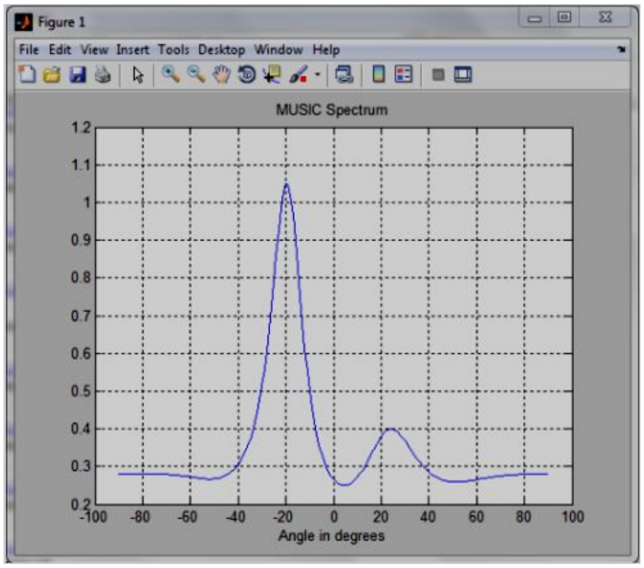

Figure 8: Matlab simulation for DOA of -20° at K=5

Table 1: Actual DOA vs. Matlab Result for Single-Source

Actual DOA (degrees)	Simulated DOA (degrees)
-50	-51
-20	-20
40	40
60	60

5.2 Test-bench Results

Implementation in FPGA is the main goal of this study. To achieve this, Matlab codes starting from the computation of the correlation matrix R is converted into Verilog codes. The different sets of input matrix X generated in Matlab are also being used as inputs in Verilog programming. Using the same sets of input, Matlab results will then be compared to testbench results Presented below are the results generated in testbench implementing the MUSIC algorithm. Table 2 and table 3 shows the results of matlab simulation and test-bench results compared to that of the actual or controlled DOA.

Table 2: Matlab Result vs. Test-bench Output for Single-Source

Actual DOA (degrees)	**Matlab Simulated Result (degrees)**	**Test-Bench Result (degrees)**
-50	-51	-49
-20	-20	-19
40	40	44
60	60	62

Table 3: Matlab Result vs. Test-bench Output for Double-Source

Actual DOA	Matlab Result	Test-bench Result
First Set		
-30	-36	-37
50	58	51
Second Set		
-10	-7	-7
20	20	26

Comparing the results in Matlab and in testbench, it is also evident that implementation in FPGA of the algorithm is also feasible. Outputs from the testbench do not exactly match in the results from Matlab with some little variations except in the triple-source wherein Matlab results are identical to the results produced in testbench. The matrix R generated by Matlab and the one produced in testbench is almost equal. When comes to the computation of the eigenvalues and eigenvectors, there happens some deviations in the results of Matlab and testbench. Below are the generated eigenvalues and eigenvectors using the DOA of -20°.

Table 4: Eigenvalues Computed From Matlab and Testbench

Eigenvalue x	Results frtom Matlab	Results from Test-bench
X1	1.900985	1.901072
X2	1.427373	1.426951
X3	0.290068	0.291049
X4	0.145089	0.144396

Considering table 4, results from Matlab and testbench are almost equal with two-point decimal accuracy. That could be expected since in generating for these eigenvalues in testbench, CORDIC algorithm has been utilized which basically uses iterations to compute for the trigonometric function. Due to some approximations in these values, there are some discrepancies in the output. The CORDIC employed in this study uses 16-bit format. To obtain higher accuracy, the 16-bit format can be increased to a higher one but it means more logic elements and more FPGA resources can be used up.

5. CONCLUSIONS AND RECOMMENDATIONS

Based on the data and results, there is a very high opportunity for the MUSIC algorithm to be fully implemented in FPGA. This study uses only five samples of data but still the algorithm has able to resolve multiple sources at the expense of little degradation in the output. Increasing the number of samples would mean more accurate results since the sampled data will resemble the true input but this would also mean of an increase in the LEs to be utilized. Increasing the number of microphones or sensors can also help improve the performance of the algorithm but can also affect the utilization of LEs.

It can also be concluded that MUSIC works well in dealing with single source only and its performance is slightly being affected as the number of sound sources increases. But determining the exact number of sources impinging in the array in practical applications is a very challenging task. Aside from the signal source itself there are also other forms of signal that can severely affect the algorithm’s performance just like noise and reverberation.

In this study, the number of sources is already determined before the algorithm process the data so the exact dimension of noise subspace has been used. However, in many practical applications, that kind of set up will not work. There should be some mechanism in determining how many true sources are really present. Out of the three major steps in MUSIC algorithm, EVD is the most computationally extensive. It would be very helpful if there could be some alternatives in finding for the eigenvalues and eigenvectors without undergoing a tremendous computation to save large part of FPGA logic elements.

Also, in using CORDIC to determine the angle in a trigonometric function, its accuracy can be improved if you increase the length bits for its angle since you have more iterations. However, although this means more accurate results but the utilization of LEs is also at stake. In the part of the spectral estimation, more accurate results can be expected if we make the angle interval smaller. In this study, a one-degree interval has been used but for better results most especially in real-life situations wherein the signals can be coming from anywhere, an interval with decimal point can also be considered (e.g. 10.1°, 12.4°, -40.2°, etc.). This means more precise in giving the results but as a consequence, this implies more angles to be scanned that can affect the speed of processing as well as the utilization of the LEs.

Determination of the major peaks in spectral estimation plays a significant role in this study. It is where the possible DOAs are being estimated. In this study, the first four highest values of PMUSIC are chosen to represent the possible directions but this will just be true only if the situation is noiseless wherein very sharp peaks could be expected. This is done by just arranging the values from highest to lowest. But in actual, wherein the signal could be corrupted by noise and will not give very sharp peaks, it will not give the correct estimate for the directions most especially if there are more than one source already.

There must be an algorithm that can possibly detect the different peaks in the whole spectrum. One of the main problems encountered in this study when comes to the actual hardware implementation is the board's capacity. Since the size of the code needed to implement the whole algorithm is somewhat bigger compared to the board's capability, code optimization has played a very significant role. The 32-bit floating point representation that is being used in this study can be reduced into something lower that can lessen the utilization of the LEs but with an anticipation of a decrease in the accuracy of the results. Lastly, for real-time processing, it poses several constraints and challenges. Captured data from the four microphones which are converted into 16-bit binary format are just representations of the voltage level of the signal. Before using it as an input to the FPGA, it should be manipulated first as to include the phase shit of the signal. This is very essential because it is where the information of the direction of the signal can be obtained.

Moreover, the number of possible sound sources could not be pre-determined in real-time processing so choosing the correct size or dimension for the noise subspace is also a big problem.

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