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# Design and Analysis of Two Stage Model for Effective Beam forming using MATLAB and VerilogHDL

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***Abstract***

*Beam forming is a technique in signal processing that has found widespread applications in fields such as radar, wireless communication, bio-medical ultrasounds and so on. Beam forming consist of an array of antennas whose radiation pattern are adjusted to a particular direction. This paper aims in implementing a two stage design for acoustic beam forming using MATLAB and Verilog HDL. First stage is a delay and sum beam former used to obtain beams from a particular direction while the second stage is an LMS model based on Least Mean Square (LMS) algorithm. This stage improves the efficiency of delay sum by removing the unwanted signals. Different hardware parameters like logic utilization, memory usage, CPU time and delay of this design are analyzed to determine its characteristics as compared to delay sum and LMS. This study found that the two stage model gives better beam forming and noise removal than when the stages are implemented independently.*

***Keywords***- *Beam forming, Delay-Sum Beam former, FPGA, IP Core, LMS Algorithm, MATLAB, Step size*

## 1. Introduction

Beam forming is a technique in signal processing that has found widespread applications in fields such as radar, wireless communication, bio-medical ultrasounds and so on by improving the quality in communication through increasing the signal to noise ratio (SNR). The purpose of this system is to recover the source signal from the interference. The technique consists of array of antennas whose radiation pattern is adjusted to have directivity in a particular direction. Mobile communication is one of the major areas employing antenna structure extensively. It mainly consists of radiating elements used to send or receive signals from all directions. Further development to these elements gives it the characteristics of beam forming [1]. In these situations, microphone arrays are used to spatially filter the source signal from the noise [2]. The beam forming system can also utilize a Field Programmable Gate Array (FPGA) for high-throughput, real time and modular beam forming [3].

The principle behind beam forming is constructive interference which reinforces the speech signal while the concept of destructive interference is used to remove noise. There are multiple methods of implementing beam formers such as delay-sum model [4-5], LMS (Least Mean Square), MVDR(Minimum Variance Distortion-less Response), Music(Multiple Signal Classification), Root Music, ESPRIT(Estimation of Signal Parameter via Rotational Invariance Technique), Robust Constant-beam width Beam forming based on Focusing Approach *etc.*[5-7]. The objective of the work is to improve beam forming using simple and robust methods. We aim to implement a two stage design consisting of a delay sum model and LMS onto a prototype board, FPGA. This two stage approach concentrates on de-reverberation by using signal independent beam former or a

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conventional beam former in the first stage and a signal dependent algorithm or an adaptive algorithm for noise reduction in the second stage [6].

## 2. Stage (1): Delay-Sum Beam Former

The conventional form of beam forming technique is the delay and sum beam former which involves summing of outputs of array elements. The basic idea is to delay or advance the signal from microphone arrays by an amount of time so that synchronization across all sensors is obtained [4]. For a signal with normal incidence, SNR is improved whereas a signal from another direction is delayed in accordance with the signal's direction before the summation. A uniform linear array is used for this method where the distance between source and array is much greater than the length of the array [5]. The microphones considered here are in such a way that the gain does not vary. The first microphone is considered as the reference while the distance between microphones is at a fixed distance of 'd' with a particular source incidence angle. Output from each microphone is a multiplied result of gain and signal with the appropriate delay where the delay is calculated with the help of 'd' and source angle incidence. The basic principle behind this beam former is constructive/destructive interference. When a particular source signal arrive at an array of microphones kept apart equally by distance 'd' at a certain angle 'ϴ' , then the delay experienced by each input signal of the microphone is given by (1).

$$\text{Delay} = (N-1)*d*\text{Sin}(\Theta) / C \qquad (1)$$

Where Delay= time delay experienced at each microphone by the input signal
N= number of microphones in the array
C= velocity of sound in air
ϴ= direction of arrival

The delay sum beam former is so designed that it effectively passes signals coming from a particular direction *i.e.*, the delay filter is designed by keeping in consideration a particular look angle. So when these delay signals arrive at the beam former, if the source direction is same as the look angle of beam former then it results in constructive interference. Otherwise, for all other direction we get a diminished destructive signal.

## 3.Stage (2): LMS MODEL

In some applications, adaptive coefficients are required as some parameters in the processing operation will be unknown. So the adaptive filters are used in such case where feedback policy is applied to refine these filter coefficients. This process is basically a criterion for performance. These filters are important as they now routinely find application in mobile and other communication devices. There also has been development in the architecture of adaptive filter so as to attain better throughput [8].

The main idea behind an LMS adaptive algorithm is that a variable filter extracts an estimate of the desired signal. The following assumptions are made:

(a) The input signal is a combination of desired signal, d (n) and interference, i (n).

(b) The variable filter has a FIR structure where filter coefficients is equal to impulse response of the filter.

(c) Error signal is the difference between desired and estimated signal.

### 3.1 Least Mean Square Algorithm

This algorithm uses a gradient based approach on the principle of updating the filter coefficients by calculating the error in the signals. First, a filtered signal is obtained from the input signal from the FIR Transversal Filter. This signal is subtracted from the

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reference signal to obtain the error signal. The error signal is then subjected to adaptive algorithm for obtaining the updated coefficients of the transversal filter which are later updated [9]. Figure1 below shows a LMS adaptive filter.

$y(n)=\sum c(k)*x(n-k)$ (2)[10][11]

$e(n)=d(n)-y(n)$ (3)[10][11]

$c(n+1)=c(n)+\mu x(n)*e(n)$ (4)[10][11]

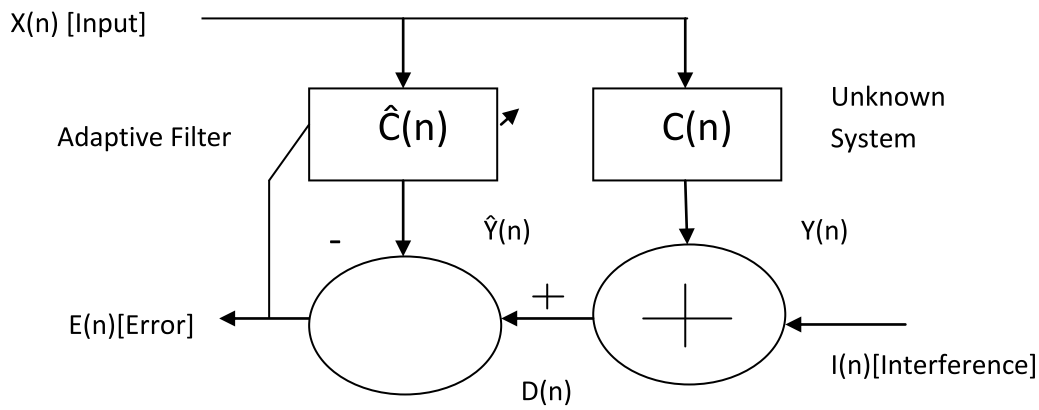


**Figure1. LMS Adaptive Filter**

The equation (2) gives the filtered output while the error signal is obtained according to the (3). The equation (4) is used by the adapting algorithm to calculate the updated coefficients [12] [13]. LMS algorithm is preferred for its simplicity as it can be easily implemented onto hardware [14]. Convergence of filter depends on the step size, $\mu$ [15]. The weight vector seem to converge and stay stable for

$$0 < \mu < 1/\lambda_{max}$$

Where $\lambda_{max=}$ largest Eigen value of correlation matrix R

If $\mu$ is small, it takes longer time to converge while for larger $\mu$, the algorithm may lead to faster convergence but lesser stability [13]. So the performance of filter depends on the step size. Another factor determining the performance is the number of weights of the filter. Therefore, better care must be given when deciding step size and weights as they are the deciding factors for the performance of an adaptive filter [13]. LMS is used in the case of fixed step size [16].

## 4. Implementation of the Design

Beam forming is a signal dispensation technique which is used in device arrays for steering signal transmission or reception [17]. Microphone array processing has the potential to relieve users from the need of having close talking microphones [18]. Implementation of our design onto a prototype board start with its simulation in MATLAB to study the functionalities of the said beamformer and algorithm and then develop a RTL code . The design of a delay sum beamformer is based on the look direction angle where it passes only the signals comng from that particular direction while all other signals are reduced as shown in Figure 3. Energy and Gain analysis are done both with and without interference and result observed. The next step involves designing an adaptive model based on LMS algorithm. Initially, filter coefficients are assigned a small value. The error is calculated after filtering done using these coefficients. This error is then used to adapt the filter coefficients so as to obtain new values according to

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LMS algorithm. Step size is fixed and it is observed that change in stepsize is related to the rate of filter convergence. If stepsize is increased, the rate increases with decrease in stability while decrease in stepsize causes decease in rate. Main application of this algorithm is to remove noise. We provide input with interference for analyzing this algorithm. Further analysis is done using original voice signal with noise as input. Mean Square Error is calculated to analyze the convergence rate of the said algorithm. Next, we implemented our design onto the protype board. Inputs from MATLAB through a .coe file are taken into Xilinx and stored into the memory IP core so that they can be called into the design. The simulated result thus obtained from adaptive design is shown as in Figure 2.

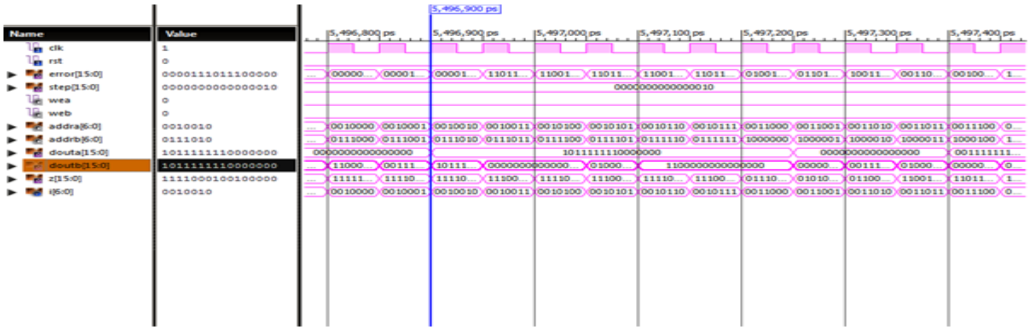

**Figure 2. Simulated Result of the Adaptive Design using IP Core**

Design of the above models necessitates awareness about fractional delay filter which enable us to design RTL [19].

## 5.Analysis and Results

MATLAB R2010a. and Xilinx ver 12.4 are used for the analysis. It is divided into various parts and worked upon individually. The results thus obtained are analyzed for improving the design algorithms for better efficiency and performance. The main components are.

(a) Conventional Beamformer
(b) Adaptive Model using LMS
(c) Mean Square Error using Adaptive LMS
(d) RTL developed for FPGA Implementation

Figure 3 and Figure 4 show the variation of energy and gain with respect to look direction in a delay sum beamformer while Figure 5 shows the polar plot. This graph enable us to analyze how attuenation or gain varies with the direction. Given result shows the look direction of 60 to have maximum gain as we have designed the beamformer for direction 60 degree while all others have attuenated power or attuenated gain.

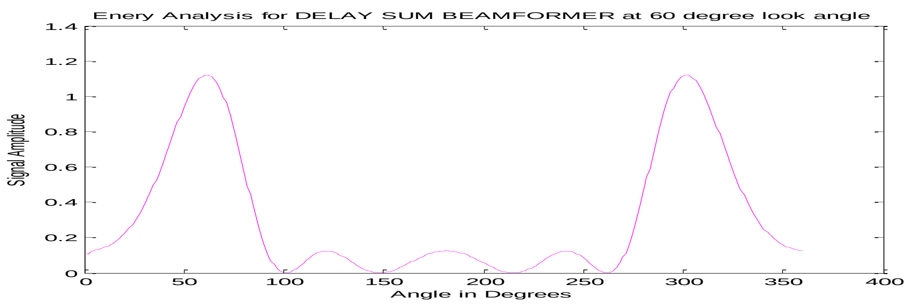


**Figure 3. Energy-Angle Analysis for Delay Sum Beamformer**

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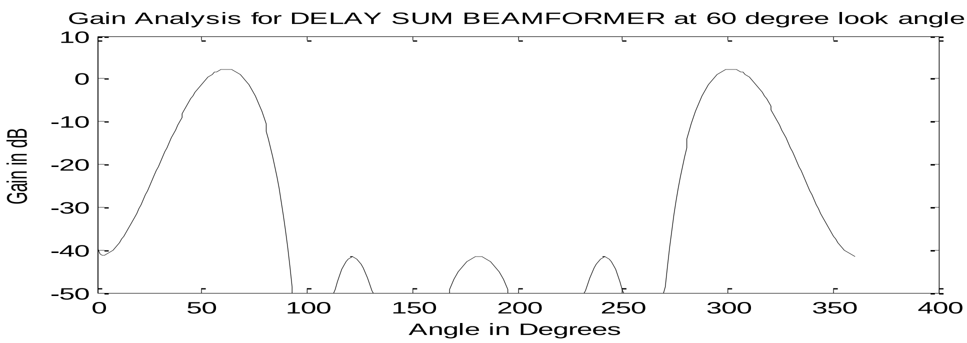


**Figure 4. Gain-Angle Analysis for Delay Sum Beamformer**

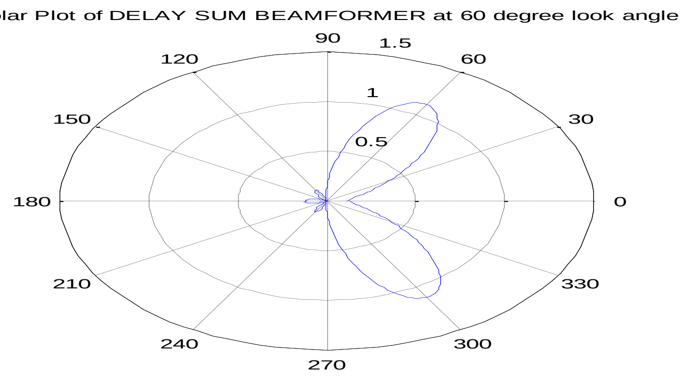


**Figure 5. Polar Plot for Delay Sum Beamformer**

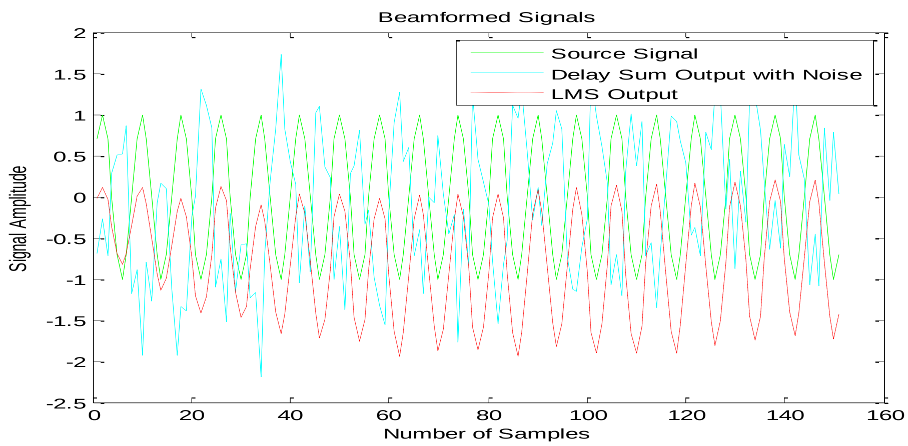


**Figure 6. Simulated Result from the Two Stage Design**

Figure 6 depicts the constructive output obtained after performing LMS adaptive algorithm. From the figure, we can observe how LMS algorithm helps in reducing the noise.

The Mean Square Error (MSE) plot for different step sizes using the LMS algorithm is shown in Figure7. This helps in analysing the convergence rate of different step sizes when using LMS algorithm.

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**Figure 7. Mean Square Error**

Figure 8 shows the simulated result where the input sine wave is replaced by an original voice signal. The figure is divided into three parts with source, signal with noise and the resultant output of LMS.

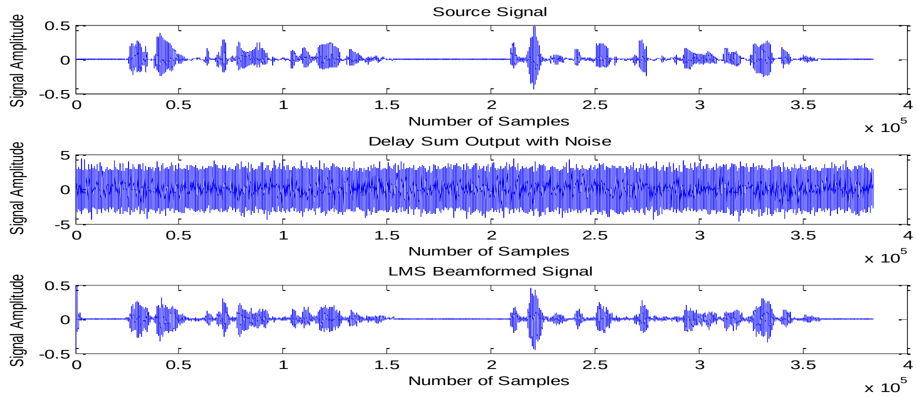


**Figure 8. Simulated Result using Original Voice**

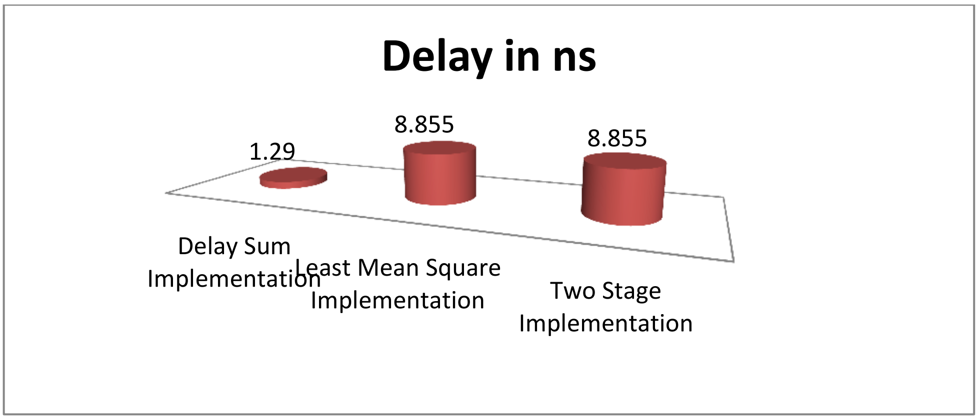


**Figure 9. Comparative Analysis of Delay for the Designs**

Comparison of delay in all the designs reveals that delay is the same for both LMS and two stage design while it is minimum for delay sum as shown in Figure 9.

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Memory usage analysis for all the designs enable us to conclude that two stage implementation is better than the other two designs. Since all the designs have comparable memory usage (Figure 10), we opt for the two stage implementation in terms of noise reduction and beamforming.

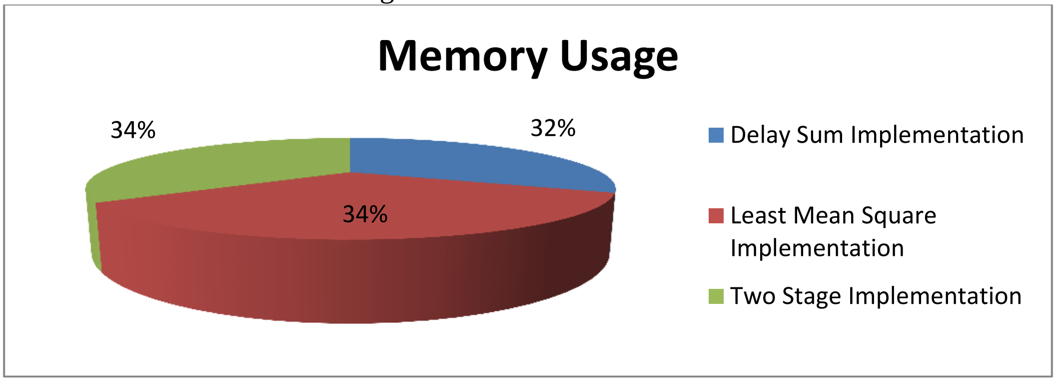


**Figure 10. Comparative Analysis of Memory Usage for the Designs**

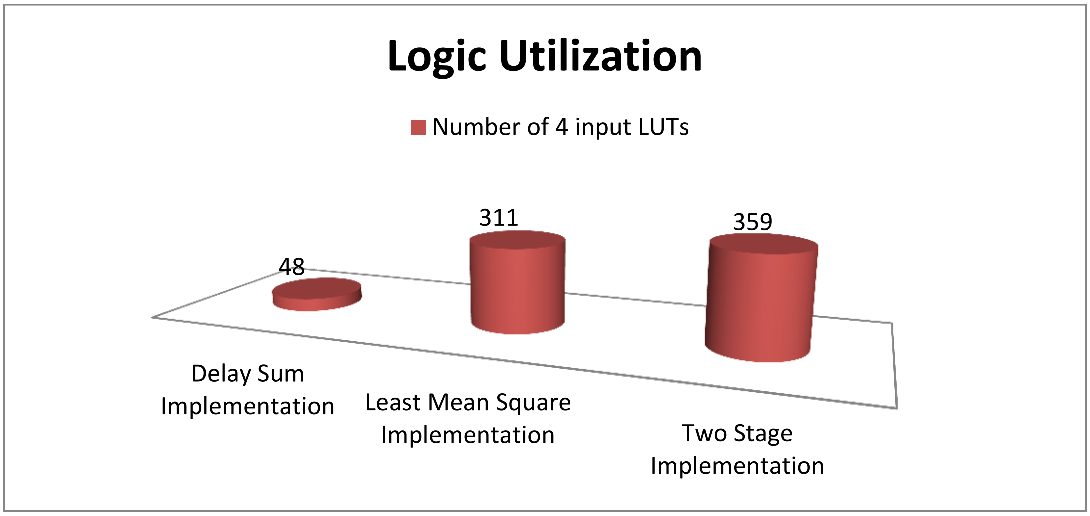


**Figure 11. Comparative analysis of Logic Utilization for the Designs**

The main parameter for hardware implementation is the utilization of logic. As shown in Figure 11, even though the logic utilisation is higher in the case of two stage than the other two, the difference between this design and LMS is narrow. Similarly,though delay sum has a much smaller hardware utilization, the model is inefficient to remove large amounts of noise.This enable us to prefer the two stage design to LMS and delay Sum. Similarly, the analysis of the different designs like delay sum model, LMS and two stage design for the CPU time required for completion when implemented onto a prototype board is shown in Figure12. We find delay sum to have minimum execution time as compared to the other two techniques. As we know, delay sum is better for filtering out signals coming from a particular direction while LMS results in maximum removal of noise. As CPU time for LMS and two stage is comparable, it is highly advisable to use both stage together for better efficiency.

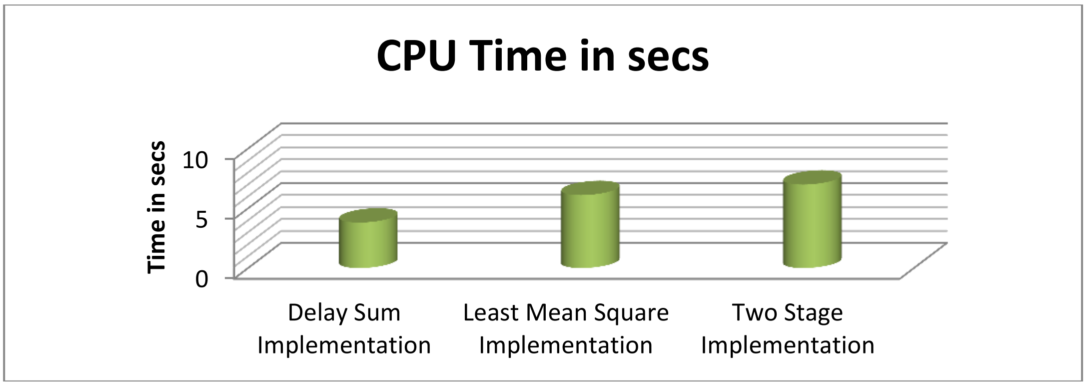


**Figure 12. Comparative Analysis of CPU Time for the Designs**

## 6. Conclusion and Future Scope

The study has proven that the two stage design is efficient in directivity and noise removal as compared to the single stage delay sum model and LMS algorithm. The results of various analyses have given us the supporting evidence in this regard. The work in acoustic beam forming gives us an idea regarding how signal processing results in producing high performance signals which have good quality SNR for long distance communications.

Future scope for this work involves development of improved algorithm to obtain better efficiency as compared to generic algorithms. Depending upon our need especially SNR, we propose the implementation of the algorithm and its analysis on a prototype board, FPGA. We also aim to study the different algorithmic techniques and compare them for their different advantages and disadvantages.

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