

Sound Localization using VHDL

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Abstract— Sound localization based on listener's potential to address the location or source of a located sound in orientation and distance. Rather than detecting what has been spoken, the work concerns about the detection of where the sound comes from. Speech enhancement aims to improve speech quality by using various algorithms. The purpose of enhancement is to increase user-friendliness and overall excellence of degraded speech signal using audio signal filtering techniques. The better speech quality improves by sign-error LMS algorithm. FPGAs have become a competitive alternative for high performance DSP applications, previously dominated by general purpose DSP and ASIC devices. The FPGA is useful for many multimedia applications and functional systems. The FPGA can be programmed to perform any number of parallel paths.

Keywords- PSNR, CORDIC, FPGA

I. INTRODUCTION

Sound is a vibration that travels through a medium and it is an audible mechanical wave of compulsion and displacement. Sound is nothing but the adaption of such waves in air or liquid and their realization by the brain [1]. Sound can propagate through a medium such as air, water and solids as travelling waves and also as a transversal wave in solids. The sound waves are produced by a sound source through a vibrating surface of a stereo speaker. Frequency is the rate, or number of times per second, that a sound wave cycles from positive to negative to positive again [1]. Humans have a range of hearing from 20 Hz to 20,000 Hz [2]. The auditory system is the sensory system for the sense of hearing. It includes both the sensory organs and the auditory parts of the sensory system [2]. CORDIC calculation mostly used for angle detection. There is possibility of loss of information due to disturbance in .wav file. LMS filter is used for producing better result for sound signal. Speech enhancement normally refers not only to reduce the noise but also to de-reverberate and separate the independent signals [4]. The use prototyping mechanism like MATLAB Simulink and Xilinx software becomes most important because of time-to-commercial constraints. This paper presents a methodology for implementing real time DSP applications on a Zedboard platform using Xilinx system generator for Xilinx. Many algorithms have been proposed based on adaptive filters for speech enhancement. A similar approach is active noise cancellation, in which the noise signal is directly measured and then removed from the noisy speech signal [5]. Active noise cancellation requires a separate microphone to record the noise alone. It is successfully implemented to improve speech intelligibility and this method is now widely used commercially. there are many types of adaptive filters used for

different schemes to adjust the weights based on different criteria, and the sign-error LMS algorithm is the most commonly used algorithm [5]. Transfer domain LMS adaptive filters were used to improve the performance of LMS-based algorithm, and another adaptive algorithm such as RLS was used for speech enhancement [13]. This chapter will give you brief analysis of different types of noise cancellation technique and how to calculate source of angle.

II. PROPOSED SYSTEM

A. FIR

High- performance and low power DSP systems are more popular in various applications. In DSP systems FIR filter is one of the processing elements for giving high performance. FIR's frequently used in Video and Image processing. FIR filter circuit can be operate in moderate frequencies and also has low-power circuit with high throughput. Adaptive digital filters are used widely in many DSP applications like channel equalization for sound recovering, harmonics reduction in RADARs, echo cancellation etc., because of their adaptability to the changes in the signals they process. In proposed system a FIR filter takes an input signal $x[u]$, and produces an output signal $y[u]$ with finite response. This rule is generally called a difference equation, each sample of the output signal $y[u]$ as a weighted sum of samples of the input signal $x[u]$.

$$y[u] = \sum_{k=0}^M b_k x[u - k] \dots\dots(1)$$

$$= b_0x[u] + b_1x[u - 1] + b_2x[u - 2] + \dots + b_Mx[u - M] \dots\dots(2)$$

Variable size should be decided before implementation of above filter. In this proposed system variable size is five.

Windowing techniques are used for illustrating fixed weight of the signals.

B. Adaptive LMS

The demand for high-speed data services has been increasing as in wireless devices. Traditional least mean square (LMS) is one of the most popular algorithms for adaptive system identification. LMS-based adaptive channel estimation can be easily implemented due to its low computational complexity. Adaptive filters have huge range of advantages over conventional filter. Adaptive filter widely used in system identification, harmonic reduction in sound wave and echo cancellation.

$$W_{n+1} = W_n + \mu * e(n) * x(n) \dots (3)$$

Where,

W_n = Current Co-efficient

W_{n+1} = Future co-efficient

μ = controls speed of the coefficient

e = error signal

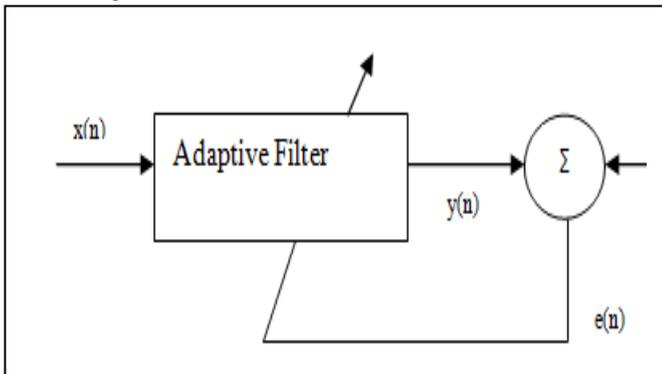


Figure1. Block diagram Adaptive Filter

It gives advantages in two situation where,

1. Convergence time constant is small for μ values.
2. MSE is value is minimum for small values μ .

C. Signed LMS algorithm

Sign-error LMS simplify the computational requirements of LMS filter by reducing number of multipliers. SELMS algorithm should reduce complexity by quantization operation. Implementing this method in account of achieve high throughput with less degradation of convergence characteristics compared to LMS algorithm.

$$\left. \begin{aligned} \text{Sgn}(a) &= 1; a > 0 \\ \text{Sgn}(a) &= 0; a = 0 \\ \text{Sgn}(a) &= -1; a < 0 \end{aligned} \right\} \dots (4)$$

Signed LMS algorithm obtained accurate values of PSNR values. In this method error signal will compare with 1, 0,-1 respectively. Hardware implementation of LMS filter also necessary for obtaining real time values of proposed system

$$e(n) = d(n) - y(n) \dots (5)$$

$e(n)$ is error value derived from subtracting desired signal from filtered signal.

PSNR is calculated by taking ratio between the maximum power value of the signal and the power of distorting noise that affects the quality of its representation. PSNR is usually expressed in terms of the logarithmic decibel scale. The sign algorithm derived as LMS algorithm for minimizing the mean absolute error criterion.

D. CORDIC

The trigonometric CORDIC algorithms were originally developed as a digital solution for real-time navigation problems. Most of the algorithm which are used in DSP and matrix require elementary functions such as trigonometric, inverse trigonometric, logarithm, exponential, multiplication and division and for all these calculations one algorithm is used called CORDIC. CORDIC calculates all arithmetic operations and elementary functions. CORDIC works by rotating the coordinate system through constant angles until angle reduces to zero.

Nowadays most of processors are designed using CORDIC technique because of its hardware-efficient solutions and this is a iterative solutions for trigonometric algorithm is called CORDIC.

$$x' = k(Y \cos \lambda - X \sin \lambda) \dots (6)$$

$$y' = k(X \cos \lambda + Y \sin \lambda) \dots (7)$$

Taking $\cos \lambda$ out equation, 5, and 5 can be rewritten as:

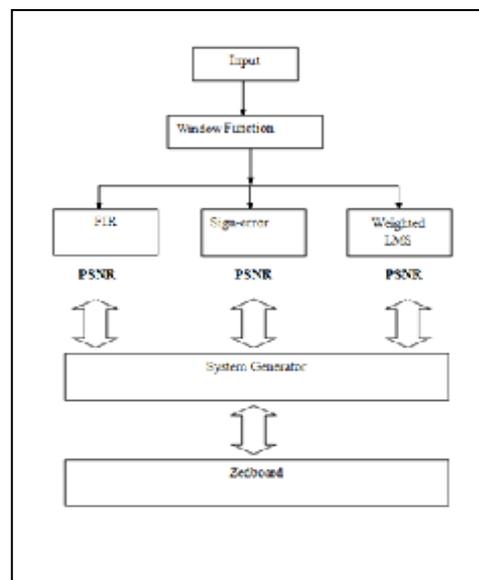
$$x' = k \cos \lambda (Y - X \tan \lambda) \dots (8)$$

$$y' = k \cos \lambda (X + Y \tan \lambda) \dots (9)$$

Where, K is a constant.

III. ARCHITECTURE

According to proposed model, sound first need to select for calculating fixed variable. Sound file should be in .wav for processing in



MATLAB. It will return the sample rate (F_s) in Hertz used to encode the data in the file. Random noise signal is added to the signal in terms of removing it in filtering process. There are different window functions for

different application oriented purpose. Windowing function is applied to order of the system for generating weight function.

Figure2. Architecture of proposed system

Fixed weight is directly used by FIR filter for calculating PSNR values. Higher values of PSNR will give you better results. System generator is used as handshaking peripheral between MATLAB and Xilinx. Initialization is necessary for running MATLAB code processing. Zedboard is used as FPGA platform for implementing the proposed system. Zedboard is connected to the PC through the USB cable.

According to results, MATLAB gives accurate results rather than on Zedboard platform. There are some limitations of Zedboard in terms of obtaining accurate result. If we look close enough there is difference between PSNR values of same window with different filtering window. Zedboard has limited resources and limited bits for processing the required VHDL file.

There is different PSNR value for different window with different filtering techniques. In terms of PSNR values it can be seen sign-error LMS filtering is most useful technique to reduce noise signal from audio signal.

VI. Implementation Technique:

All filters are implemented using MATLAB 2012 version. Implementation process carried out on Zedboard which is FPGA supportive board. All three filter comparisons mode on basis of PSNR value.

LMS filter is adaptive and all weights are calculated from processing signal. Due to low computational complexity LMS adaptive filter is widely used everywhere. Error based LMS filter comparison takes place between 1, 0,-1. In MATLAB 'int1.m' should be run first for evaluating all weights and filter values.

In FIR filter only weights are directly calculated in 'int1.m' file and system generator uses all evolved values of weights and directly given it to FIR filter. Sound localization is process of recognizing direction of sound signal. CORDIC is basically used for calculating angle of specific signal. It uses iteration process for calculating accurate angle. Zedboard has limitation with directly coding sound signal, Henceforth in research not exceeded after sound enhancement technique.

Zedboard is connected to laptop through USB cable. One can check board is connected or not by using iMPACT. Once connection is establish it is easy to operate Zedboard through MATLAB. Zedboard has blue LED which glows when code is appropriately dumps on it.

System generator is used for maintaining communication between MATLAB and Xilinx. It helps to generate VHDL code from MATLAB code. Simulink model use graphical implementation for filter design which is easily readable on Zedboard through system generator Simulink model give access to different blocks like gateway input, gateway output. PSNR value can be seen on LCD.

A. Result

TABLE I. TABLE TYPE STYLES

Sr.no.	Filter	Type of Window	PSNR MATLAB o/p	PSNR Xilinx o/p
1	LMS	BOXCAR	63.6321	62.45
		BARLETT	63.4033	62.75
		KAISER	63.4901	62.15
		HANNING	63.0891	62.02
		HAMMING	63.6839	62.28
2	FIR	BOXCAR	41.6901	41.59
		BARLETT	53.6074	53.13
		KAISER	42.0207	41.92
		HANNING	47.6888	47.48
		HAMMING	51.7911	51.43
3	Sign-error	BOXCAR	67.3285	64.02
		BARLETT	67.1343	63.83
		KAISER	67.1924	64.03
		HANNING	66.8161	63.81
		HAMMING	67.3604	63.98

Above table shows comparison between LMS adaptive filter, FIR and Sign-Error LMS filter. This comparison actually based on PSNR values of different filter with different window. Filter quality depends upon PSNR value, if PSNR is high that filter produces better signal quality. According results it clearly shows that Sign-Error method is impressive in terms obtaining PSNR value.

IV. CONCLUSION

Speech enhancement aims to improve speech quality by using various algorithms. The objective of enhancement is increasing precision of degraded speech signal using different audio signal filtering techniques. The better speech quality improves by sign-error LMS algorithm. System generator is used for linking MATAB and Xilinx software. It gives user more comfort and understanding to run MATLAB design on FPGA platform. Unlike to FPGA, MATLAB gives all require resources for generating accurate results. There are various applications in which speech enhancement is the ultimate aim, like SONAR, Seismology, echo cancellation and noise reduction in sound system. It has tremendous advantages in field of sound recognition system. According to the results, sign-error LMS method is very accurate and straightforward method to produce noise free desired output signal. In sign-error LMS algorithm maximum PSNR value is obtained.

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