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Reconfigurable Hardware Implementation of Adaptive LMS algorithm for Noise Cancellation on Real-time Audio signals

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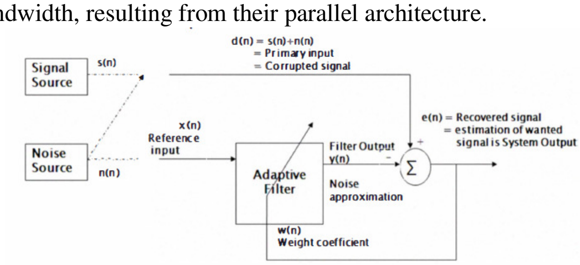
Abstract— The voice signal is first processed in MATLAB and the spectrogram is obtained for the same. Gaussian noise is deliberately added to the signal and the two text files 'with noise' and 'without noise' files are obtained. These two files are first modified to bring it in matrix form and used as the header files in design of adaptive LMS algorithm in embedded C on Xilinx Platform Studio and implemented on FPGA for processing and recovered text file is obtained. The recovered signal is plotted and played back on MATLAB and verified with the original signal and was found to be matching.

Keywords-Adaptive LMS, Noise Cancellation, Spectrogram

I. INTRODUCTION

The digital signal processing applications impose considerable constrains on area, power dissipation, speed and cost. Thus the design tool should be carefully chosen. The most common tools for the design of such application are ASIC, DSP and FPGA .[1]The DSP used for extremely complex math-intensive tasks but can't process high sampling rate applications due to its serial architecture. Whereas ASIC faces lack of flexibility and require long design cycle. The FPGA (Field programmable Gate Array) can make up disadvantages of ASIC and DSP. Hence FPGA has become the best choice for the design of signal processing system due to their greater flexibility and higher bandwidth, resulting from their parallel architecture.



This paper investigates the applicability of a FPGA system for real time audio processing systems. In recent years , acoustic noises become more evident due to wide spread use of industrial equipments. An Active(also called as Adaptive) noise cancellation (ANC) is a technique that effectively attenuates low frequencies unwanted noise where as passive methods are either ineffective or tends to be very expensive or bulky.[3] An ANC system is based on a destructive interference of an anti-noise, which have equal amplitude and opposite phase replica of primary unwanted noise. Following the superposition principle, the result is noise free original sound.

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II.ADAPTIVE LEAST MEAN SQUARE(LMS)

Least mean square(LMS) is one of the widely used algorithm used for noise cancellation in speech signal because it is highly stable.

Consider the transversal filter with input x(n) i.e. vector of the M(filter length) most recent input samples at sampling point n.

x(n) = [x(n), x(n -1), ... x(n - M +1)] (1) and w(n) i.e. vector of filter coefficients as w(n) = [w 0 (n), w 1 (n), ... w M -1 (n)](2)

At some discrete time n, the filter produces an output yen) which is linear convolution sum given by

$$y(n) = \sum_{k=0}^{M-1} w(n)x(n-k) \quad (3)$$

Also can be represent in vector form as

$$y(n) = w(n)^T u(n) \quad (3)$$

The error signal is difference of this output with the primary signal den) given by ,

e(n) = den) - yen) (4)

To optimize the filter design, we choose to minimize the mean-square value of e(n). Thus the cost function is defined as the MSE denoted by J

$$J = E[e(n)e^*(n)]$$
$$=E[|e(n)|^2] \quad (5)$$

Where E denotes the statistical expectation operator. To find an optimal solution w_o (initial guess) that satisfies the condition

$$J(w_o) \leq J(w) \text{ for all } w \quad (6)$$

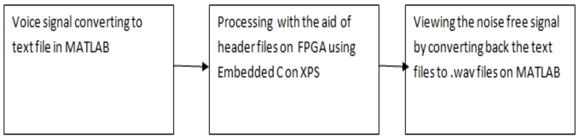
Which is a mathematical statement of unconstrained optimization.

Starting with w(O), generate a sequence of weight vector w(1), w(2), ... ,such that the cost function J(w) is reduced at each iteration of the algorithm. therefore

$$J(w(n+1)) < J(w(n)) \quad (7)$$

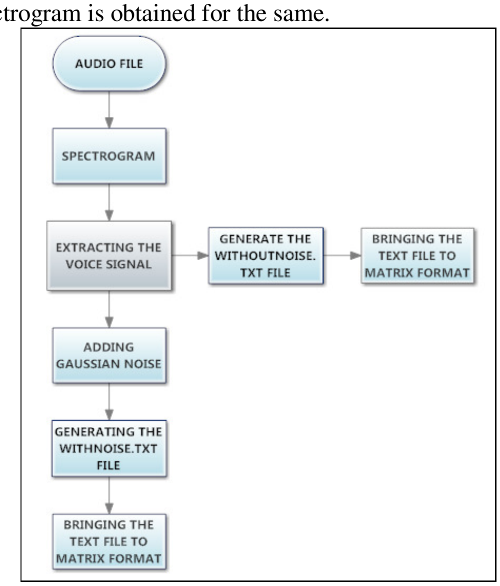
Where w(n) is the old value of the weight vector and w(n+ 1) is its updated value

III. IMPLEMENTATION

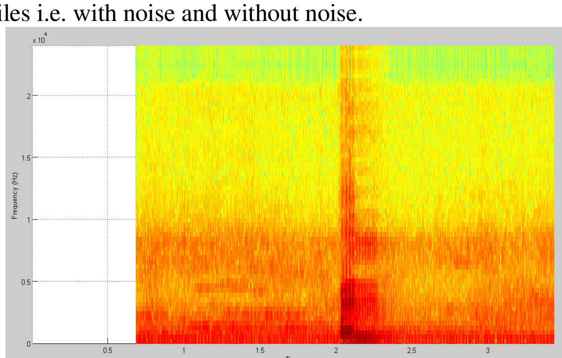


A. MATLAB front end

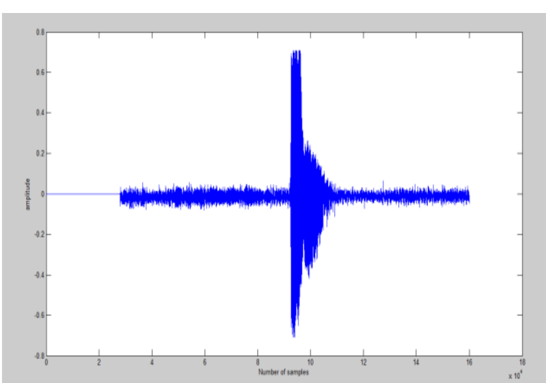
The voice signal is first processed in MATLAB and the spectrogram is obtained for the same.



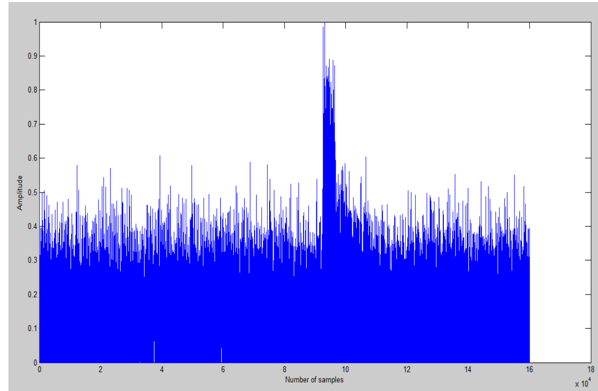
Flowchart depicting the conversion of voice signal to text files i.e. with noise and without noise.



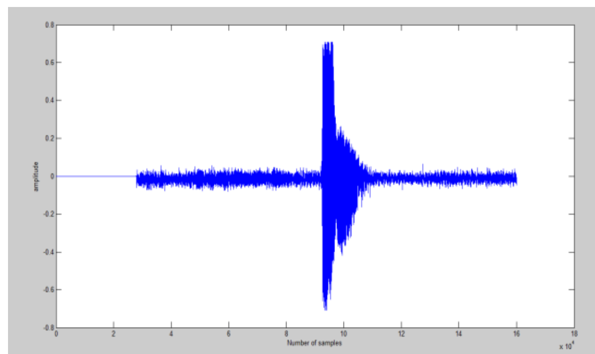
Spectrogram indicating the spectrum of frequencies in the sound



The voice signal is obtained from the spectrogram as shown. The Gaussian noise is deliberately added to the signal and with noise file is obtained and plotted as shown below.



Now the LMS algorithm is applied in embedded C on Xilinx Platform studio. The recovered signal is plotted and played back on MATLAB and verified with the original signal and was found to be matching



IV. RESULTS AND DISCUSSION

The recovered signal obtained as text file is plotted and played back on MATLAB and verified with the original signal and was found to be matching.

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