Mode Selective Simulation for Noise Reduction in Digital Hearing Aid

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Abstract— Hearing aids are gadgets utilized by hearing hindered persons to balance the hearing loss. They can't totally beat the perceptual bends brought on by a hearing loss however support the user to decode speech. The focus of this paper is to develop simple Digital Hearing Aid (DHA) noise reduction model using MATLAB simulation and programming language that works in real time mode or offline mode. Implementation of the structured DHA system includes the DC blocking filter using MATLAB programming and adaptive noise reduction filter.

Key words: DHA, MATLAB, Digital filter

I. INTRODUCTION

Today, a major part of human population suffering by hearing loss. The main complaint of person with hearing loss is low ability to deduce speech in noisy environment. So, by using the Digital Signal Processing (DSP) enhance the possibility of performing signal-to-noise ratio.

The sound is processed by the hearing aid and reaching to ear. Essentially it is made of three parts, Microphone, Processor unit, receiver module. Electrical Impulse is converted by use of Microphone. Not it is converted in suitable form from electrical impulse by Processor unit; receiver converts the impulses in sound using a decoder. The audio frequency range is generally between 20 Hz to 20 kHz which capable to hear. The human ear is only sensible to hear the frequency range between 1 kHz to 4 kHz[1]. So below 1 kHz, ear will not respond and above the 4 kHz, it may damage the hearing capability.

This paper shows design of simulink block model for digital hearing aid system with mode selecting technique. This simulink block model is works in both real time mode and offline mode. The simulation result of this system shows the noise removing output signal from the noisy input signal. This is possible by using the signal processing block sets through the simulink library.

II. SIMULINK BLOCK MODEL FOR NOISE CANCELLATION IN DHA

In this section mode selective simulink that is work in both real time and offline mode is shown (in figure 1). This simulink generally divided in to the 5 blocks: Input signal block, Dc blocking digital filter block, Noise source block, NLMS filter block, Output signal block.

The simulink block model includes three switches and working of this is shown in table 1. In this switching mode switch no.1 (S1) is used for selecting real time mode and offline mode for input the signal, while switch no.2 (S2) is used for adding zero vector or random noise signal and switch no.3 (S3) is used for the NLMS algorithm output signal or desired signals.

Selection	If select 0	If select 1
Switch 1	Real time signal	Signal from audio file
Switch 2	Zero vector	Noise signal
Switch 3	NLMS output signal	Desired signal

Table 1: Switching Modes

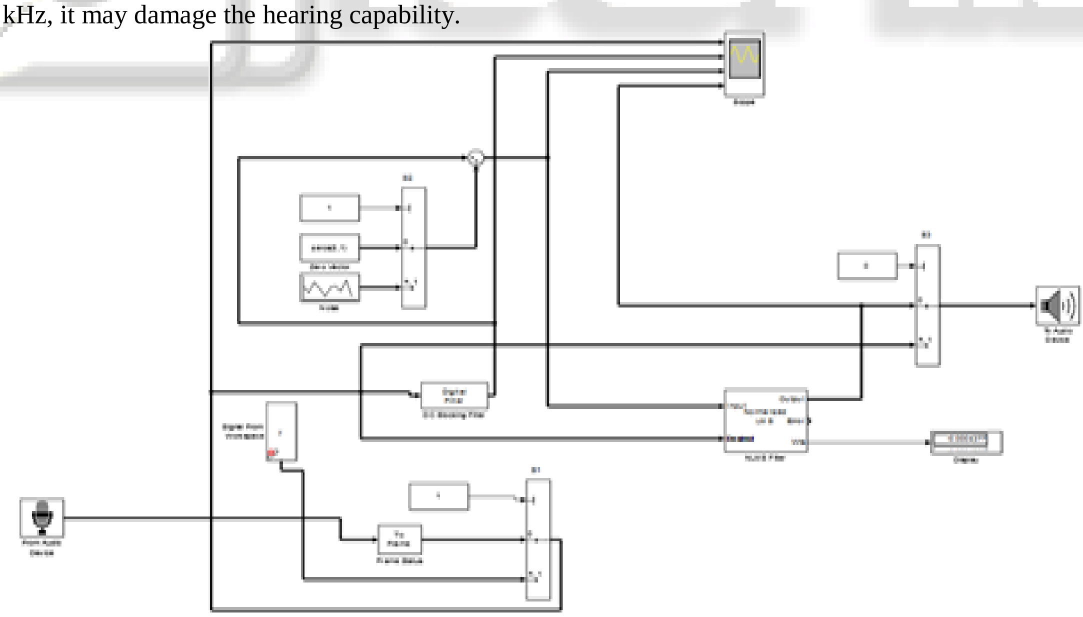

Fig. 1: Simulink block model for Noise cancellation in DHA

A. Input Signal Block:

This input signal block is divided in to the two sub blocks. One is from audio device block that is used to reads speech signal from an audio device in real time mode. This block is connected to the frame status function block for the frame conversion to set the sampling mode of the output signal. Source block parameter of from audio device block is shown in table 2. Second sub-block is signal from work space block that is used to take the audio file in offline mode from the MATLAB workspace at successive sample times. Both the sub blocks are connected to the multiport switch 1 function block to select the real time or offline mode.

Device	Default
No. of channels	1
Sample rate(Hz)	44100
Queue duration(sec)	1
Frame size sample	8
Output data type	signal

Table 2: Parameter of from audio device block

B. Digital Filter:

This filter block is used to remove the unwanted DC offset from sound signal caused by trivial microphone input or by processing of sound signal which adds DC signal to the sound signal. While processing on sound file, it is more suitable that if the point of waveform is centered about zero point detection, else the sound file may be get more clipped, especially if there is more sounds are mixed in a sound file. Besides it cannot hear at 0 Hz by us, it is not require recording such a file. So we actually want a signal (in this case Sine Wave) which has no partially or fully (like harmonics) mixture of DC Components.

The sound file is getting rid of 0 Hz frequencies, it should be filtered must, the subtraction is possible at a constant amount in each sample file of sound, but a more generalized solution which will better work in more situation and automatically work is to filter out the frequency which is low[2].

The simplest and most effective way of filtering the low frequency is with the Infinite Impulse Response (IIR) filter. IIR uses a recursive estimate based on a narrow, lowpass elliptic filter. This algorithm typically uses less money than FIR and is more efficient. The information regarding filter is given in table 3. And Filter Co-efficient is as shown in table 4

Filter structure	Direct from second transposed
Numerator Length	4
Denominator Length	4
Stable	YES
Linear Phase	NO

Table 3: Discrete Time IIR filter

Numerator	Denominator
0.95396504143840244	1
-2.8618951243152075	-2.9057609235003685
2.8618951243152075	2.8159101078761664
-0.95396504143840244	-0.91004930013068419

Table 4: Filter Co-efficients

The response of DC blocking filter is shown in figure 2. Figure shows when denominator value a is close to 1.0, then the filter will only filter out very low frequency. As it goes towards 0.0, more and more frequency are noticeable affected by the filter

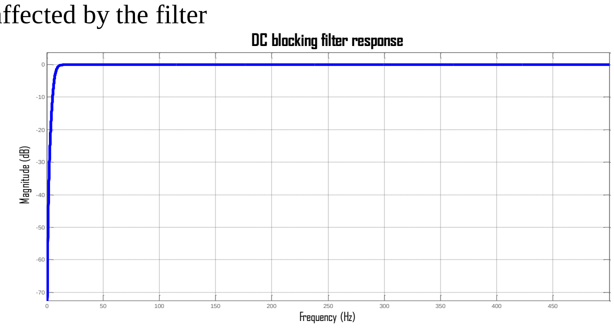

Fig. 2: DC blocking filter response

C. Noise Source Block:

Noise is the form of unwanted signal which may affect the desired signal, in the context of speech while process of that signal, speech is the signal of primary interest and there are three types of noise that effect directly to the speech intelligibility[3];

- Random noise with Power-Density Spectrum similar to that of speech.
- Competing speaker(s) noise.
- Room Reverberation.

For this system the input speech signal is approximately clean speech signal so to simulate the real situation need to add some noise. This noise is generated by using the random source block by selecting the source type to uniform Gaussian distribution then use the method that sum the 5 number of uniform values, mean value is 0 and variance is 0.01.

D. Adaptive Noise Cancellation Block:

Usually the background noise is not constant at every time, it different at every instance so noise reduction or cancellation must be adaptive process; let say it has ability to work according different condition and able to be adjust with different environments.

In this algorithm the input noise signal x(n) is passing through the linear FIR filter for digital signal processing. The output of the FIR filter Y(n) and desired signal D(n) is then compared for the purpose of error detection and minimization. As soon as E(n) signal is detected, then adaptive algorithm triggered for require adjustment and automatic update to the filter coefficients of the next iterated input signal via a feedback mechanism. The adaptation process completes and stop when the E(n) signal is minimized completely. This process is shown in figure 3[4].

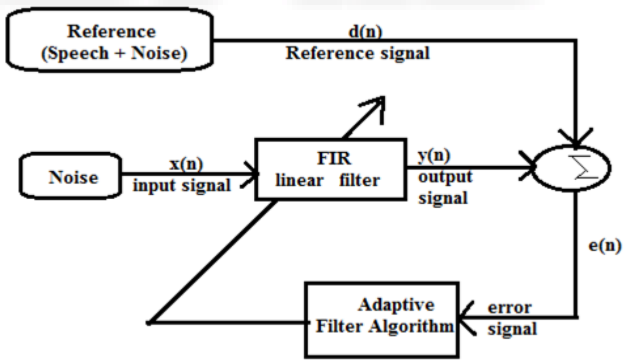

Fig. 3: Structure of Adaptive Noise Cancellation

The two adaptive algorithms are mostly in use in which one is Least Mean Square (LMS) and other is Recursive least mean square (RLS). RLS algorithm has complexity and need many memory banks for implementation. While LMS algorithm is the most popular and common algorithm due to follow reason like, implementation of it is easy while running in software as well as hardware and with computational simplicity and efficient use of memory. But the limitation of LMS (Least Mean Square) algorithm is its scaling sensitivity to its input and the time of convergence is slow[7]. This problem is overcome by using the NLMS (Normalized Least Mean square) algorithm. This algorithm is a variation of LMS algorithm that solves the problems by utilizing the standardization of input power. Each Iteration of NLMS algorithm performs the following steps:

1) *The Filtering Operation:*

$$y[n] = \sum_{n=0}^{N-1} x[n]w[n]$$

2) *The Error Calculation:*

$$e[n] = d[n] - y[n]$$

3) *The Coefficient Correction:*

$$w(n+1) = w(n) + \mu \frac{x(n)}{c + \|x(n)\|^2} e(n)x(n)$$

Where x(n)^2 is the squared Euclidean norm of the input. μ is the adaptation constant, which optimizes the convergence rate of the algorithm and the range of μ is the 0 < μ< 2. c is the constant term for normalization and its always c < 1.

E. *Output Signal Block:*

In this block there is two sub blocks, one is the To audio device block sink block which is used as speaker that send sound data to the computer's audio device and the other one is the Time scope sink block to display the time domain signals. The sequence of scope signals are shown in figure 4.

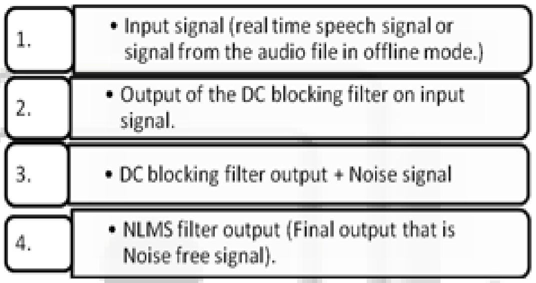

Fig. 4: sequence of scope signals

III. SIMULATION RESULTS AND DISCUSSION

The above explained mode selective simulink design and perform on MATLAB 2013 and the software simulation results are shown in figure 5 and 6. Figure 5 shows the result of real time mode in this the input signal is real time speech signal. Figure 6 shows the result of offline mode and the input signal is dialog audio file. Figures are shows that in both modes the first signal is the input signal, the second signal is the output of the DC blocking filter that shows the DC offset is remove from the input signal. Third signal is the mixing of noise signal and input signal and finally the fourth signal is the output of the NLMS algorithm that clearly shows that speech is retained successfully by removing the noise signal.

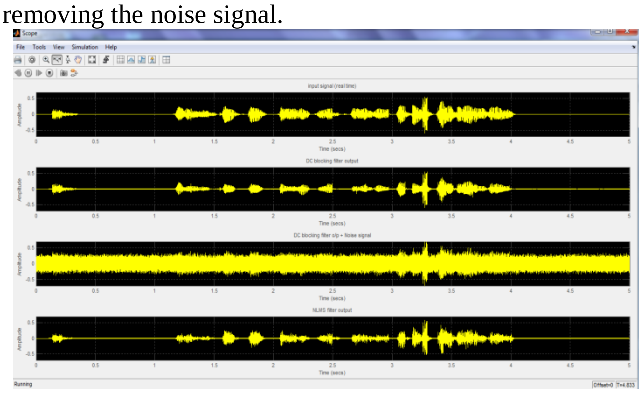

Fig. 5: Output of the real time speech signal

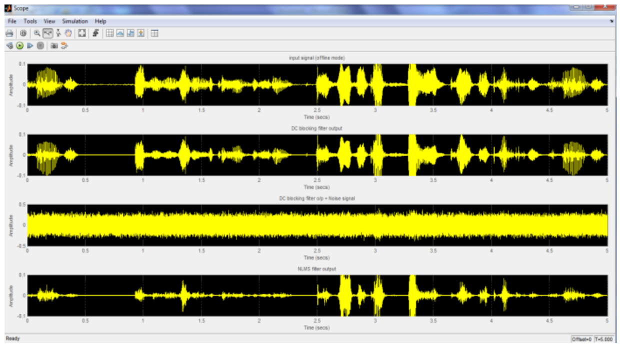

Fig. 6: Output of the offline mode dialog audio file

IV. CONCLUSION AND FUTURE WORK

This paper presents the adaptive noise cancellation algorithm to remove the noise. NLMS adaptive filter algorithm is used for this purpose. The results of simulation show that successfully achievement for the cancellation of noise in both real time and offline mode.

Functions such as frequency shaper to correct the hearing loss at certain frequency and amplitude compression function to control the overall gain of a speech amplification system are used in future design work. After the addition of these functions in to the developed simulink design this configuration of Digital Hearing System is used to implement on hardware DSP kit.

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