MATLAB Simulation and Modeling for Acoustic Noise Reduction Using Adaptive Filter

Nitish Nagpal¹, Ms. Nidhi Bajaj², Ms Nisha Sharma³,

¹Ece, Student , PPIMT, HISAR India
²Ece, Asst. Prof., PPIMT,HISAR India
³EE, Asst. Prof., SAITM, India

Abstract— In this paper, least mean square algorithm is used to subtract noise from input signal with the help of Simulink using MATLAB 11a software. In this thesis acoustic noise cancellation model is used to suppress acoustic noise. This model consists Acoustic Environment subsystem and adaptive filter to remove the noise from the signal output adaptive filter to remove the noise from the signal output. Reference signal is used in LMS filter and the desired signal, to automatically match the filter response. As it used to the correct filter model, the filtered noise is subtracted and the error signal should contain only the original signal. Various blocks are combined with each other to design the model. After filtration Block LMS filter produces output as rectified voice which is very similar to original signal and the noise signal is removed.

By using this technique, the acoustic noise is suppressed to a much larger extent from the original speech signal and helps in better communication. The experimental results show that it provides a greater SNR value.

Using Simulink various experimental results are obtained. We have done the analysis of SNR value for different type of noises and collect result values in the form of tables which shows how SNR value changes with time. Results are also shown on a new six dimensional model.

Index Terms— Acoustic Noise; Simulink; Filter; Noise Suppression, Digital Signal Processing

I. INTRODUCTION

In environments with ambient noise such as air-conditioning systems and engines, exposure to high decibels of sound for an extended period of time can prove distracting during communication and damaging to humans both physically and psychologically. Active noise cancellation (ANC) is a method used to reduce undesired noise. This thesis details the programming of digital signal processing (DSP) techniques used to filter out unwanted sound. The purpose of the explanation and simulation of the Filtered-X LMS FIR adaptive filter in MATLAB is to demonstrate active noise cancellation in software. This background research of algorithms and code development will be used for the eventual integration with a DSP processor or microcontroller that allows for the augmentation of a standard pair of Sennheiser PX100-II headphones, that do not provide noise isolation or cancellation, into headphones that can cancel unwanted noise. The headphones will then be used with an Amateur “Ham Radio” in an environment with unwanted, high decibel sound. This thesis supports the proof of concept through software simulation and a report of different methods of hardware integration. The research and development of digital signal processing techniques and hardware integration will lead to the augmentation of a pair of headphones used for communication in a noisy environment.
Active noise control is still awaiting a major breakthrough. The general increase in wellbeing and the desire for luxury will some day meet the decreasing price of practical active noise control systems in many areas. In some applications, like in active headsets, the breakthrough has already happened. The price of headsets with an active noise control feature is not much higher than the price of traditional headsets, meaning the decision to make an additional investment in a better listening experience is easily made. In some other potential application areas, like in the automotive industry, the challenges for active noise control are greater and there are not many practical applications of ANC. The broad interest in lightweight structures will surely generate noise related challenges and make active noise control more attractive in the automotive industry as well. Noise cancellation technology is used to reduce undesired noise or ambient sound. Two methods are used to achieve noise cancellation in systems. The first method is passive noise cancellation. Passive noise cancellation focuses on preventing sound waves from reaching the receiver, the eardrum in most cases, and is achieved through different noise isolation techniques. Passive noise reduction does not require power and is very cheap to implement. The ear is insulated from external noise through the use of an ear cup fitting snugly around the ears to block out ambient noise. This method can offer up to seventy percent noise reduction if implemented correctly. The second method of noise cancellation is active noise cancelation or control. It is achieved by introducing a canceling waveform through secondary sources. These secondary sources are interconnected through an electronic system using a specific digital signal-processing algorithm for the particular cancellation scheme. This method yields better results than passive noise cancellation, but it requires power to achieve and is more costly. Some headsets provide both active and passive noise cancellation techniques to yield the best results in reducing ambient noise from a system.

II. PROPOSED METHOD

The LMS adaptive filter uses the reference signal on the Input port and the desired signal on the Desired port to automatically match the filter response. As it converges to the correct filter model, the filtered noise is subtracted and the error signal should contain only the original signal.

In the model, the signal output at the upper port of the Acoustic Environment subsystem is white noise. The signal output at the lower port is composed of colored noise and a signal from a .wav file. This example model uses an adaptive filter to remove the noise from the signal output at the lower port. When you run the simulation, you hear both noise and a person playing the drums.

III. IMPLEMENTATION
Over time, the adaptive filter in the model filters out the noise so you only hear the drums.

- **Acoustic Noise Cancellation**: filtered noise is subtracted and the error signal should contain only the original signal

![Figure 1 Acoustic Noise Canceller a block diagram](image)

**Figure 2 Acoustic Noise Cancellation procedure**

- **LMS Filter**: The Block LMS Filter block implements an adaptive least mean-square (LMS) filter, where the adaptation of filter weights occurs once for every block of samples

![Figure 3 LMS Filter](image)

The block estimates the filter weights, or coefficients, needed to minimize the error, e(n), between the output signal, y(n), and the desired signal, d(n). Connect the signal you want to filter to the Input port. The input signal can be a scalar or a column vector. Connect the signal you want to model to the Desired port. The desired signal must have the same data type, complexity, and dimensions as the input signal. The Output port outputs the filtered input signal. The Error port outputs the result of subtracting the output signal from the desired signal.

- **Audio Device Writer**: The Audio Device Writer block writes audio samples to an audio output device.
Figure 4 Audio Device Writer
See System Interaction of Audio Device Writer Block for a visualization of how the Audio Device Writer block plays audio data.

- Waterfall

Figure 5 Waterfall

The Waterfall block displays multiple vectors of data at one time. These vectors represent the input data at consecutive sample times. The input to the block can be real or complex-valued data vectors of any data type including fixed-point data types. However, the input is converted to double-precision before the block processes the data. The Waterfall block displays only real-valued, double-precision vectors of data. The range of X, Y, and Z, or the current setting of the axes XLim, YLim, and ZLim properties, determines the range of the axes (also set by axis). The range of C, or the current setting of the axes CLim property, determines the color scaling (also set by axis). The CData property for the patch graphics objects specifies the color at every point along the edge of the patch, which determines the color of the lines. The waterfall plot looks like a mesh surface; however, it is a patch graphics object. To create a surface plot similar to waterfall, use the meshz function and set the Mesh Style property of the surface to 'Row'.

IV. RESULT

By running this model, we can listen to the audio signal in real time (while running the simulation). The stop time is set to infinity. This allows us to interact with the model while it is running.

Notice the colors of the blocks in the model. These are sample time colors that indicate how fast a block executes. Here, the fastest discrete sample time (e.g., the 8 kHz audio signal processing portion) is red, and the second fastest discrete sample time is green.
Figure 6: Different output values corresponding to output and error

- **Waterfall Scope**
  The Waterfall window displays the behavior of the adaptive filter's filter coefficients. It displays multiple vectors of data at one time. These vectors represent the values of the filter's coefficients of a normalized LMS adaptive filter, and are the input data at consecutive sample times. The data is displayed in a three-dimensional axis in the Waterfall window. By default, the x-axis represents amplitude, the y-axis represents samples, and the z-axis represents time.

Figure 7: Waterfall window Analysis of result

So the result obtained shows the following data:

<table>
<thead>
<tr>
<th>S.No</th>
<th>Noise (D)</th>
<th>Original signal</th>
<th>Desired (D)</th>
<th>Error(desired-output)(E)</th>
<th>Output</th>
<th>SNR</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>-0.911</td>
<td>0.0083</td>
<td>0.764</td>
<td>0.7391</td>
<td>0.02451</td>
<td>96.7%</td>
</tr>
<tr>
<td>2</td>
<td>-0.033</td>
<td>0.0085</td>
<td>0.526</td>
<td>0.5083</td>
<td>0.1992</td>
<td>96.2%</td>
</tr>
<tr>
<td>3</td>
<td>0.951</td>
<td>-0.0199</td>
<td>0.256</td>
<td>0.226</td>
<td>-0.0185</td>
<td>88.2%</td>
</tr>
<tr>
<td>4</td>
<td>2.033</td>
<td>0.02582</td>
<td>0.188</td>
<td>0.1586</td>
<td>0.02938</td>
<td>84.0%</td>
</tr>
<tr>
<td>5</td>
<td>0.0373</td>
<td>0.03012</td>
<td>0.259</td>
<td>0.236</td>
<td>0.02249</td>
<td>91.1%</td>
</tr>
</tbody>
</table>

Table 1: Final result Analysis
The desired signal must have the same data type, complexity, and dimensions as the input signal. The Output port outputs the filtered input signal, which is the estimate of the desired signal. The Error port outputs the result of subtracting the output signal from the desired signal.

V. CONCLUSION AND FUTURE SCOPE
This paper analyzed that the model designed using MATLAB 11a Simulink is very effective. The acoustic noise is suppressed greatly. The rectified voice is almost similar to the original signal. The experimental results show that the SNR value is very high and the noise value in the rectified signal is very low. In future we can use other filters like Fast block filter, NLMS and other filters in this model and SNR value can be further improved.

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