

## Eliminating unwanted signals in sound by using digital signal processing system

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### ABSTRACT

Unwanted signals or noise signals in sound files are considered one of the major challenges and issues for a thousand users. It is impossible to reduce or remove these noise signals without identifying their types and ranges. Therefore, to address one of the big problems in the digital or analogue communication, which is noise signals or unwanted signals, an adaptive selection method and noise signal removal algorithm are proposed in this research. The proposed algorithm is done through specifying the types of undesirable signals, frequency, and time range, then utilizing digital signal processing system which includes design several types of digital filters based on the types and numbers of unwanted signals. Four digital filters are used in this research to remove noise signals from the sound file by implementing the proposed algorithm using Matlab Code. Results show that our proposed algorithm was done successfully and the whole noise signals were removed without any negative consequence in the output sound signal.

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## 1. INTRODUCTION

Noise can be defined as an undesirable signal that interferes with the measurement or the communication of another signal [1]. Noise is available in several forms and results from vary and many sources, such as audio frequency acoustic noise coming from moving, vibrating, or radio-frequency electromagnetic noise that can interfere with the reception and transmission of data, voice, and image over the radio-frequency spectrum [2-3]. Unwanted signals in sound files are considered one of the major challenges and issues for a thousand users [4-5]. Therefore, many companies and researchers have tried to propose a solution differently. However, most of these studies have not proposed an adaptive solution with a low cost [6-7]. In [8], different filters are used to perform speech enhancement. In this real noisy environment, the form of Gaussian noise is taken into consideration. Authors concluded that the Butterworth filter could be used successfully to decrease the noise from the audio signals with the different ripple factor and frequency. Different results are produced from the order of different frequency. In [9-10], authors worked on developing adaptive filter. In order to design the optimal digital filters, several researchers have worked on the signal reconstruction from noisy measurements by proposing a number of numerical algorithms [11-14]. In [15], noise from the audio signal is removed by using Audio Noise Reduction System. Filters are used in this system in order to manipulate the phase response and/or amplitude of a signal based on their frequency. While authors used two types of filters: static and tunable filters in [16]. When the stopband frequencies and passband frequencies are fixed, filters are called static. In contract, filters are named as tunable filters when stopband and passband frequencies are adjustable. The change of these frequencies depends on the requirement of the applications. Tunable digital filters are broadly used in digital audio equipment, medical electronics, telecommunications, and control systems. This is the basic need for

elimination of unwanted signal from the audio signal. While in [17], authors used Audio Noise Reduction System based upon graphical user interface GUI in MATLAB to remove noise from the audio signal.

To sum up, most of the studies have not find an adaptive solution with a low cost. The cost plays an important factor in our life. Therefore, in this paper, the issue of unwanted signals will be solved by using a low-cost method that depends on designing adaptive-cheap filter-digital filters that are suitable for each noise signals. The aim of this paper is to remove different types of noise signals in sound files by using the digital signal processing system through designing several digital filters. These digital filters are used in this research to remove unwanted frequencies in the sound file. The time-domain, as well as a frequency domain, were plotted and analyzed in order to get the big picture of the characteristics of used filters. The algorithm of removing the noise signal from the original signal is performed by using the MATLAB program. The filters eliminate the specific frequency components of noise and recuperate the original speaking signal.

This study is focused on a using adaptive selection method to identify noise signals and then utilizing digital filters to remove unwanted signal in the sound file. This method reads the whole file at first and then unwanted signals are specified based on the comparison between the characteristics of the original signal and noise signals. According to the selection method, the number and types of filters are allocated. The performance of the proposed methods is evaluated by simulation results obtained using MATLAB/Simulink.

## 2. MODELING & SIMULATION

### 2.1. Overview

The Matlab program is an advanced tool, used broadly in numerous engineering projects [18]. It is used in this research to implement the noise removal algorithm that aims to remove noise signal from the original signal in the sound file. In the below section, the proposed method and description of how to implement this proposed algorithm are done. In addition to that, results are discussed in more details and the summary is presented.

### 2.2. The Proposed Method

The proposed solution is focusing on how to design different digital filters that are suitable to remove noise signals. Our algorithm is started by reading and analysis the original signals that are needed to be treated. From the initial analysis of the original signals, the number of required design filters will be estimated. Then, the type of filter will be determined based on the nature of the noise signal. For example, if the noise occurs in the first range of the noise signal frequency, it is better to design low pass filter while design the high pass filter will be suitable for the noise that occurs on the last range. In this research, authors will focus only on the digital filters since they are cheap and easy to implement. Digital filters include two main types: Infinite Impulse Response (IIR) and Finite Impulse Response (FIR) [19]. FIR filters are always stable, so the mathematical equations of FIR are presented and explained in more details [20].

$$y(n) = b_0 x(n) + b_1 x(n-1) + b_2 x(n-2) \dots + b_M x(n-M)$$

$$-a_1 y(n-1) - a_2 y(n-2) - \dots - a_N y(n-N)$$

$$y(n) = \sum_{k=0}^M b_k x(n-k) - \sum_{l=1}^N a_l y(n-l)$$

Where  $x$ : the input signal,  $y$ : the output signal,

The constant  $b_k, k=0,1,2,3,4,5 \dots M, a_l, l=1,2,3,4,5 \dots N$  are called the coefficients

The filter will be design based on three main factors: identifying filter type, the values of cut off frequencies, and the number of poles and zeros.

The proposed algorithm of our scheme procedures are illustrated in Figure 1.

The mechanism of the proposed scheme depends on using Digital Signal Processing system by utilizing several types of digital filters (adaptive digital filters). After analyzing the sound file (audio signals), the characteristics and the number of unwanted or noise signals will be determined. Then,  $n$  filters with the same type or  $L$  filters with  $s$  types are designed. The first one is designed when all unwanted signals contain almost the same characteristics which are rarely happening while the second types are designed when the audio file contains different types of unwanted signals, so there will be  $L$  number of filters with  $s$  types. For example, if the sound file conations on three noise or unwanted signals- two of them have the same characteristics and the third on has not-, in this case, three filters with two types will be used in order to remove undesirable signals. Based on the characteristic of noise signals, several types of filters can be used in order to remove these signals, such as low pass filter (LPF), high pass filter (HPF), bandpass filter (BPF),

stopband filter (SPF), and notch filter. The purpose of using these types of filter is to remove noise signals and recover the original signals.

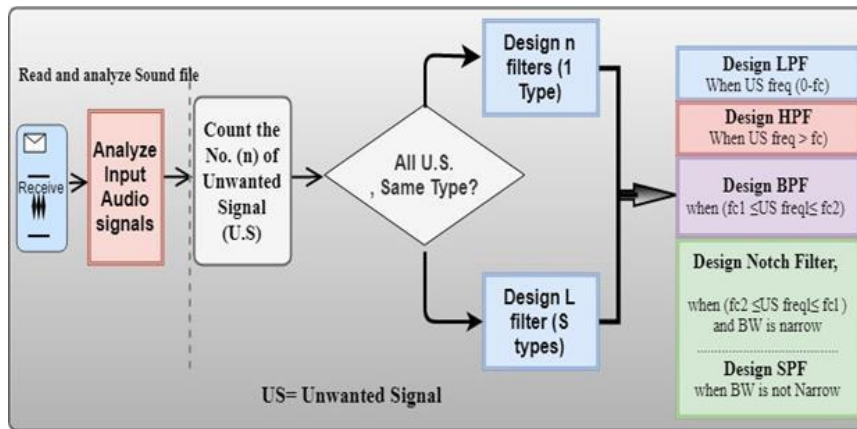


Figure 1. The proposed scheme of eliminating unwanted signals

#Pseudo code of Proposed Scheme#

- 1: Start
- 2: Read the sound file (Sf)
- 3: Count the number (n) of unwanted signals (U.s)
- 4: If U.s [1:n] =Sa? //Sa: Same type of unwanted signal
- 5: Design n filters
- 6: Else {
- 7: If Usn=0? //Usn=0, if the signal between 0 and fc
- 6: Design LPF //LPF: Low Pass filter
- 7: Else { if Usn=1? //Usn=1, if the signal between greater than fc
- 8: Design HPF filters // HPF: High Pass filter
- 9: Else { if Usn=2? //Usn=2, if the signal between fc1 and fc2
- 10: Design BPF filters // BPF: Band Pass filter
- 11: Else {if Usn=3? //Usn=3, if the  $fc2 \leq \text{signal} \leq fc1$
- 12: if Usn=NBW? //NBW: narrow bandwidth
- 13: Design Notch Filter
- 14: Else { Design SPF // SPF: Stop Pass filter
- 15: } } }
- 16: End

### 3. SIMULATION RESULTS

In order to test our proposed scheme, several metrics have been collected in this paper, such as the output spectrum for each stage, the frequency response for the cascade filter and for each filter.

The form of the original signal with noise is shown in Figure 2. According to the mentioned figure, there were four unwanted signals which are in red colours. All four noise signals start after the 7th second in time, and at different values of frequency. After extracting the original signal form: four unwanted frequencies were found. Therefore, four filters should be designed to remove those unwanted frequencies. We specify the unwanted frequency value through the drawing (0, 1575, 3150, 4725 Hz).

To remove the first unwanted frequency ( $f=0$ ), nulling digital filter is used through this code  $h1=[1 \ A(1) \ 1]$  where  $A=-2\cos(\omega d)$  to remove certain frequency. The result of removing the first unwanted signal is shown in Figure 3. In order to remove the second unwanted signals, two further step needs to be done besides removing the first frequency: first, used  $h2=[1 \ A(2) \ 1]$  to remove the second frequency, second, conduct the convolution between filter1 and 2, the result of above steps. Same steps were done to remove the third and fourth noise signals by using  $h3=[1 \ A(3) \ 1]$  and  $h4=[1 \ A(4) \ 1]$  as illustrated in Figure 3. Figure 4 shows the sound file spectrum after removing the third and fourth unwanted signals, respectively.

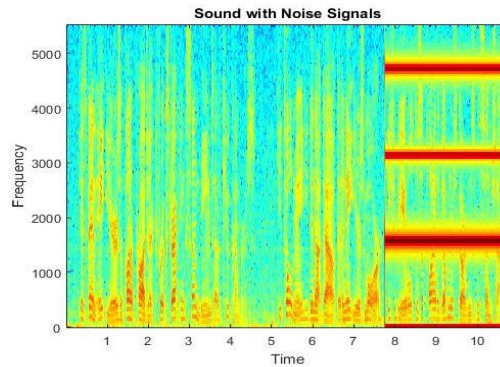


Figure 2. Sound with noise signals before implementing the proposed algorithm

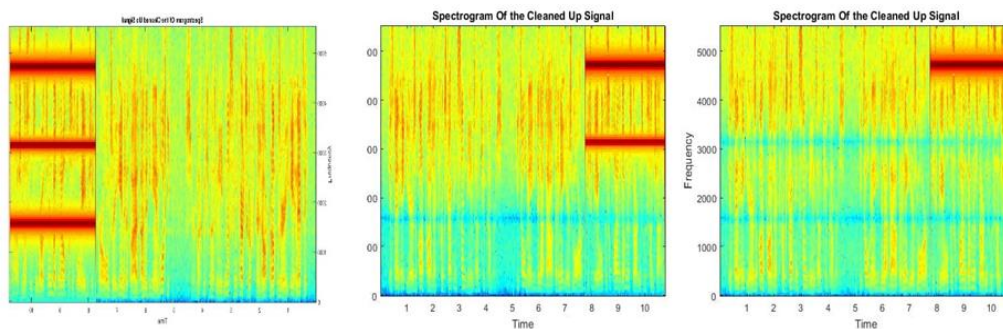


Figure 3. The Output spectrum of the signal after each step (1st, 2nd, 3rd) filters

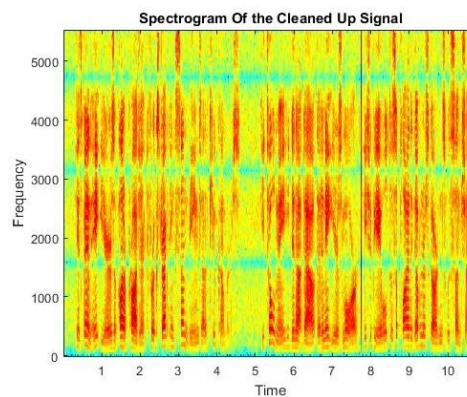


Figure 4. Sound file spectrum after removing unwanted signals

**The frequency response of each filter was plotted by using below codes**

```
plot(w,abs(h11));title('Frequency Response of Filter 1');
h22=freqz(h2,1,w);subplot(222);%plot the freq reponse of filter(2)%
plot(w,abs(h22));title('Frequency Response of Filter 2');
h33=freqz(h3,1,w);subplot(223);%plot the freq reponse of filter(3)%
plot(w,abs(h33));title('Frequency Response of Filter 3');
h44=freqz(h4,1,w);subplot(224);%plot the freq reponse of filter(4)%
plot(w,abs(h44));title('Frequency Response of Filter 4');
```

Figure 5 and 6 show the frequency response of each filter and for cascaded filter to combine those four filters into one filter by using convolution for 4 filters. Figure 6 represents the first frequency response that was used to remove the first unwanted signal. It is obvious that filter is used to remove certain noise signal.

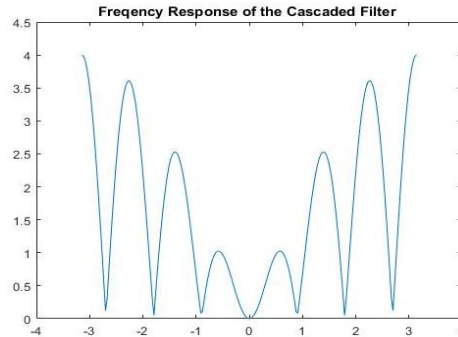


Figure 5. The frequency response for cascaded filter

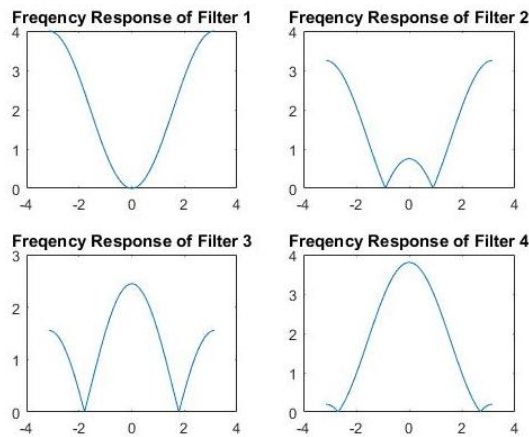


Figure 6. The frequency response of each filter

The implementation of Matlab program is done in order to remove the noise signal from the original signal in the sound file. Therefore, after removing the unwanted frequencies of the original signal and work convolution between the frequencies, so we get the sound file without noise. The project is successfully done and whole noise signals were removed. Therefore, we can hear the sound file in an obvious way without any noise.

**4. CONCLUSION**

In this article, authors are worked to address one of the big problems in the digital or analogue communication which is noise signals or unwanted signals. Noise signals in sound files are considered one of the main challenges and issues for a thousand users. The proposed algorithm based on designing several digital filters was done in this project by using Matlab code in order to solve this issue and to remove those noise signals. In this project, the authors used four digital filters to remove noise signals from the sound file. The algorithm of our project procedures is illustrated in this paper carefully. The output of each stage was plotted and explained in details. Frequency response for each filter and the resulted filter were plotted in order to give a big picture about what we have done in this project. Results show that the proposed algorithm was done successfully, and the whole noise signals were removed without any negative result in the output sound signal. An adaptive filter whose coefficients are dynamically changing is suggested to use and designed beside the FIR filter in order to get better results with a short time so the adaptive filters will cancel noise in corrupted signals.

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