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# IIR and FIR Digital Filter: A Case Study

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**Abstract:** — this paper is about to analyze and identify the digital audio signal. Two audio signals are provided which is the original audio and the added noise audio signal. The purpose of the study is to identify and eliminate the unknown noise signals by using Finite Impulse Response and Infinite Impulse Response digital filters. The detail steps to design the both have been stated in this paper. All the results is simulated and showed in MATLAB to show the comparison between the two.

**Keywords-** FIR, IIR, MATLAB, noise, sampling frequency, passband frequency, stopband frequency.

## 1. INTRODUCTION

Filters are the most essential component used in signal processing and telecommunication system[1]. The main functions of filter are to confine a signal into a prescribed frequency, to decompose a signal into two or more sub-band signals processing and modify the frequency spectrum of a signal. There are 2 most primary types of the digital filter used in Digital Signal Processing (DSP) application which is Finite Impulse Response (FIR) and Infinite Impulse Response (IIR)[2]–[7]. There are several advantages of the digital filters compared to the analog filters which digital filters' performance does not vary with environment[8]–[12]. Digital filters also can be operated at very low frequencies and have a wide range of frequencies by mere change to the sampling frequency. In this study, both FIR and IIR filter is designed to filter out the unwanted noise from the noise audio signal. IIR filter can be used when the only important requirements are sharp cutoff and high throughput, as it requires fewer coefficients than FIR filter. However, FIR filter should be used whenever a large number of filter coefficients and phase distortion is desired. Both FIR and IIR filter design have their own pros and cons in certain application. Therefore when a digital filter is to be designed, type of filter need to be clarify either suitable in the situation or not.

## 2. METHODOLOGY

### a. Original Sound Analysis

#### i. MATLAB Programming Code

```
%get original audio file
[x,Fo] = audioread('G1o.wav');
%get time based
to=(1/Fo:1/Fo:length(x)/Fo);
 %plot the Original Waveform in Time Domain
subplot(2,1,1);
plot(to,x);
grid;
title('Original Waveform in Time Domain');
xlabel('Time');ylabel('Amplitude');
%plot the Original Waveform in Frequency Domain
subplot (2,1,2);
m=length(x)-1;
```

```
fo=0:Fo/m:Fo;
%perform FFT for original waveform
xfft=abs(fft(x));
plot(fo,xfft);
grid;
title('Original Waveform in Frequency Domain');
xlabel('Frequency');ylabel('Magnitude');
```
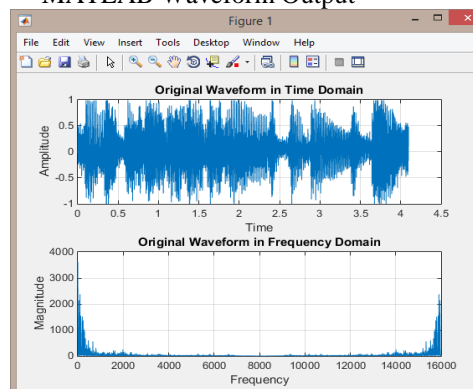
#### ii. MATLAB Waveform Output



**Figure 1:** Original Audio Signal Waveform

### A. Noise Sound Analysis

#### i. MATLAB Programming Code

```
%get noise audio file
[y,Fn] = audioread('G1n.wav');
%get time based on sampling frequency for G1n.wav
tn=(1/Fn:1/Fn:length(y)/Fn);
 %plot the Noise Waveform in Time Domain
subplot(2,1,2);
plot(tn,y);
grid;
title('Noise Waveform in Time Domain');
xlabel('Time');ylabel('Amplitude');
%plot the Noise Waveform in Frequency Domain
subplot(2,1,2);
n=length(y)-1;
fn=0:Fn/n:Fn;
%perform FFT noise waveform
yfft=abs(fft(y));
plot(fn,yfft);
grid;
title('Noise Waveform in Frequency Domain');
xlabel('Frequency');ylabel('Magnitude');
```

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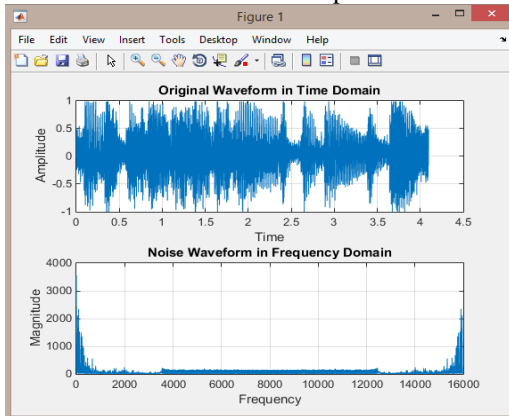*WINGS TO YOUR THOUGHTS.....*

ii.     MATLAB Waveform Output



**Figure 2:** Noise Audio Signal Waveform
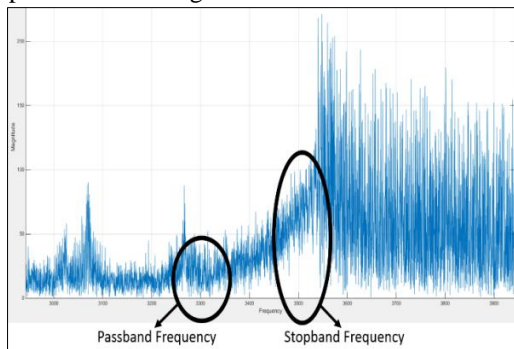
*B.   Specification Design*



**Figure 3:** Define Passband and Stopband Frequency

From observation of the original sound wave and noise sound wave, required specification for designing the digital filter as below[13-29].

*Sampling Frequency, $f_t$ = 16000Hz*
*Passband Frequency, $f_p$ = 3300Hz*
$\omega_p = 0.4125\pi$
*Stopband Frequency, $f_s$ = 3500Hz*
$\omega_n = 0.4375\pi$

Besides that, from the waveform, assumptions of several variables are defined.

*Peak Passband Deviation, $a_p$ = 1 dB*
*Stopband Deviation, $\delta_s$ = 50*
*Minimum Stopband Deviation, $a_s = |20 \log \delta_s|$ dB*
*= 34 dB*

*C.   FIR Filter Design*
    i.     Design Calculation
From the observation, the minimum stopband deviation was 34dB. Therefore, Hanning window is chosen as the value obtain was the nearest with less than 44dB.
Then the transition bandwidth is calculated.

$$\Delta f = \frac{f_s - f_p}{f_t} = \frac{3500 - 3300}{16000} = 0.0125$$

Based on the transition bandwidth, the length of Hanning window is obtained by

$$\Delta f = \frac{3.1}{M}$$
$$M = \frac{3.1}{0.0125} = 248$$

ii.     MATLAB Programming Code

```
%filter design
fs = 3500;          %stopband frequency
fp = 3300;          %passband frequency
ft = 16000;         %sampling frequency
Wn = (fs+fp)/ft;
%Window-based fir filter design
a = fir1(248,Wn,hann(Wn));
%plot the frequency response
freqz(a,1);
 figure(2);
%plot the low pass filter
plot(w/pi, 20*log10(abs(h)));
title('Low Pass Filter');
xlabel('\omega/\pi');ylabel('Gain (dB)');
grid;
```
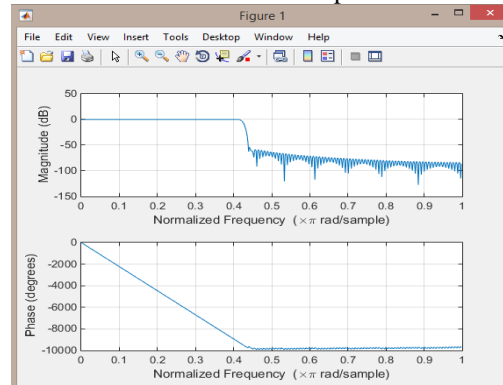
iii.     MATLAB Waveform Output



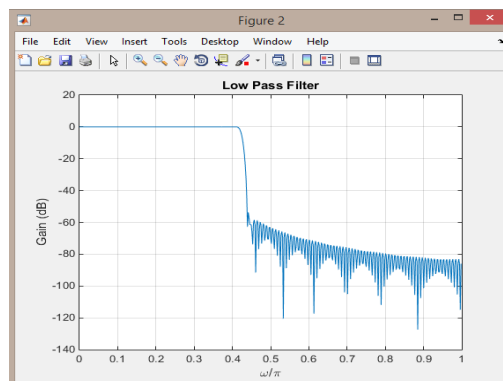**Figure 4:** Frequency Response of the FIR Filter (1)



**Figure 5:** Frequency Response of the FIR Filter (2)

*D.   IIR Filter Design*
    i.     Design Calculation
    From the design specification obtained, cutoff frequency of the design filter is calculated as below.

$$f_c = \frac{f_p + f_s}{2} = \frac{3300 + 3500}{2} = 3400Hz$$
$$\omega_c = 0.425\pi$$

All the design specification in coding in MATLAB and the function of "buttord" is used to calculate the order of the filter design as well as the cutoff frequency.

ii.     MATLAB Programming Code

```
%filter design
Ws = 0.4375;          %stopband frequency
```

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```
Wp = 0.4125;            %passband frequency
Ft = 16000;
ft = Ft/2;              %sampling frequency
Rp = 1;                 %passband Ripple
Rs = 34;                %stopband Attenuation
%butterworth filter design
%Butterworth filter order and cutoff frequency
[n,Wn] = buttord(Wp,Ws,Rp,Rs);
[b,a] = butter(n,Wn);  %perform butterworth
filter
%Frequency response of digital filter
[hd wd] = freqz(b,a);
figure(1);
freqz(b,a,512,1);
%Plot the low pass filter
figure(2);
plot(wd/pi,abs(hd));
title('Low Pass Filter');
xlabel('\omega/\pi');ylabel('Gain (dB)');
grid;
```

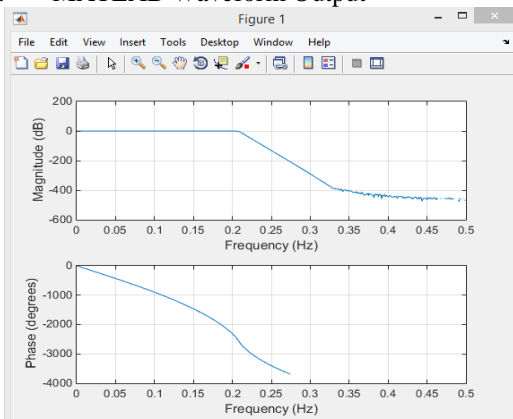### iii.      MATLAB Waveform Output
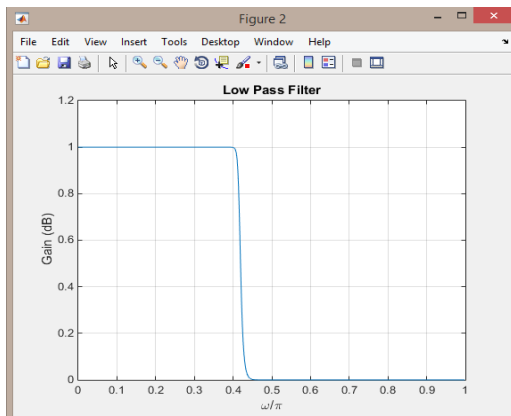


**Figure 6:** Frequency Response of the IIR Filter (1)
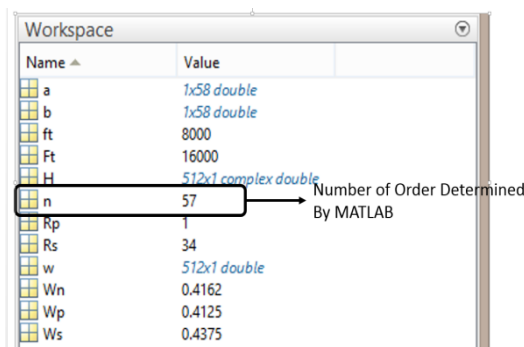


**Figure 7:** Frequency Response of the IIR Filter (2)



**Figure 8:** Number of Order for IIR

## 3.   RESULT

### A.   FIR Filter Design

#### i.      MATLAB Coding

```
%get original audio file
[x,Fo] = audioread('Glo.wav');
%get time based on sampling frequency for G10.wav
to=(1/Fo:1/Fo:length(x)/Fo);
%get noise audio file
[y,Fn] = audioread('G1n.wav');
%get time based on sampling frequency for G1n.wav
tn=(1/Fn:1/Fn:length(y)/Fn);
figure(1);
%plot the Original Waveform in Time Domain
subplot(2,1,1);
plot(to,x);
grid;
title('Original Waveform in Time Domain');
xlabel('Time');ylabel('Amplitude');
%plot the Original Waveform in Frequency Domain
subplot (2,1,2);
m=length(x)-1;
fo=0:Fo/m:Fo;
%perform FFT for original waveform
xfft=abs(fft(x));
plot(fo,xfft);
grid;
title('Original Waveform in Frequency Domain');
xlabel('Frequency');ylabel('Magnitude');
 figure(2);
%plot the Noise Waveform in Time Domain
subplot(2,1,1);
plot(tn,y);
grid;
title('Noise Waveform in Time Domain');
xlabel('Time');ylabel('Amplitude');
%plot the Noise Waveform in Frequency Domain
subplot(2,1,2);
n=length(y)-1;
fn=0:Fn/n:Fn;
%perform FFT noise waveform
yfft=abs(fft(y));
plot(fn,yfft);
grid;
title('Noise Waveform in Frequency Domain');
xlabel('Frequency');ylabel('Magnitude');
%filter design
fs = 3500;          %stopband frequency
fp = 3300;          %passband frequency
ft = 16000;         %sampling frequency
Wn = (fs+fp)/ft;
%Window-based fir filter design
a = fir1(248,Wn);
%Frequency response of digital filter
[h,w] = freqz(a,1);
z = filter(a,1,y);
%filter the noise data, y
%write the filtered sound
audiowrite('Filtered_FIR_G1.wav',z,ft);
%get the filtered audio file
[n,Ff]=audioread('Filtered_FIR_G1.wav');
%get time based
tf=(1/Ff:1/Ff:length(n)/Ff);
figure(3);
%plot the waveform in time domain
subplot(2,1,1);
plot(tf,n);
grid;
title('Filtered Waveform in Time Domain');
xlabel('Time');ylabel('Amplitude');
subplot (2,1,2);
%plot the waveform in frequency domain
m=length(n)-1;
ff=0:Ff/m:Ff;
nfft=abs(fft(n));    %perform fft
plot(ff,nfft);
grid;
```

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```
title('Filtered Waveform in Frequency
Domain');
xlabel('Frequency');
ylabel('Magnitude');
```
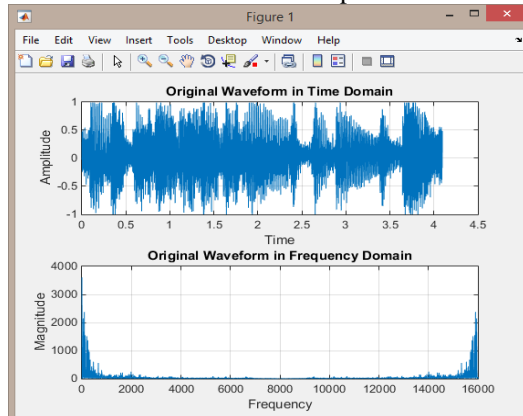
### ii.    MATLAB Waveform Output
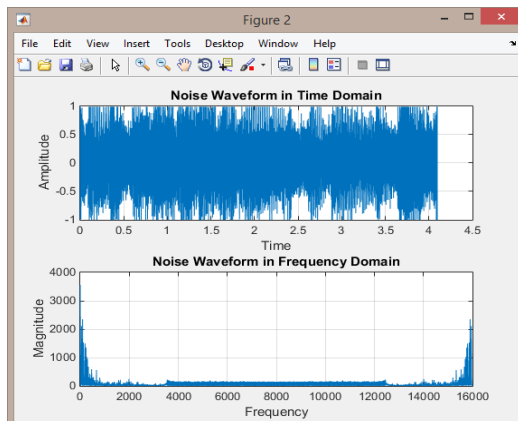
**Figure 9:** Result of Original Audio Signal Waveform

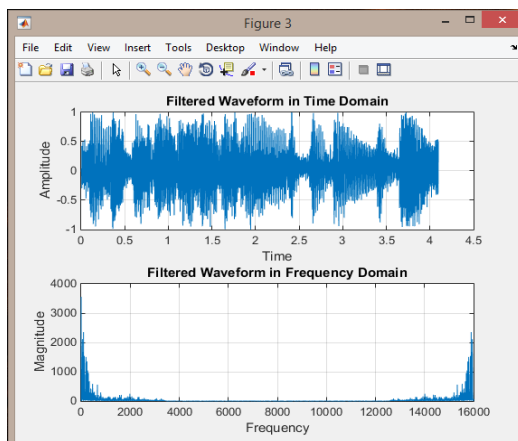**Figure 10:** Result of Noise Audio Signal Waveform

**Figure 11:** Result of Filtered Waveform by FIR Filter

### B.   IIR Filter Design
### i.    MATLAB Coding

```
%get original audio file
[x,Fo] = audioread('G1o.wav');
%get time based on sampling frequency for
G10.wav
to=(1/Fo:1/Fo:length(x)/Fo);
%get noise audio file
[y,Fn] = audioread('G1n.wav');
%get time based on sampling frequency for
G1n.wav
tn=(1/Fn:1/Fn:length(y)/Fn);
figure(1);
%plot the Original Waveform in Time Domain
subplot(2,1,1);
plot(to,x);
grid;
title('Original Waveform in Time Domain');
xlabel('Time');ylabel('Amplitude');
%plot the Original Waveform in Frequency Domain
subplot (2,1,2);
m=length(x)-1;
fo=0:Fo/m:Fo;
%perform FFT for original waveform
xfft=abs(fft(x));
plot(fo,xfft);
grid;
title('Original Waveform in Frequency Domain');
xlabel('Frequency');ylabel('Magnitude');
figure(2);
%plot the Noise Waveform in Time Domain
subplot(2,1,1);
plot(tn,y);
grid;
title('Noise Waveform in Time Domain');
xlabel('Time');ylabel('Amplitude');
%plot the Noise Waveform in Frequency Domain
subplot(2,1,2);
n=length(y)-1;
fn=0:Fn/n:Fn;
%perform FFT noise waveform
yfft=abs(fft(y));
plot(fn,yfft);
grid;
title('Noise Waveform in Frequency Domain');
xlabel('Frequency');ylabel('Magnitude');
%filter design
Ws = 0.4375;        %stopband frequency
Wp = 0.4125;        %passband frequency
Ft = 16000;
ft = Ft/2;          %sampling frequency
Rp = 1;             %passband Ripple
Rs = 34;            %stopband Attenuation
%butterworth filter design
%Butterworth filter order and cutoff frequency
[n,Wn] = buttord(Wp,Ws,Rp,Rs);
[b,a] = butter(n,Wn); %perform butterworth filter
%Frequency response of digital filter
[H,w] = freqz(b,a,512,1);
z = filter(b,a,y);     %filter with noise sound
%Frequency response of digital filter
[hd wd] = freqz(b,a);
%write the new filtered sound
audiowrite('Filtered_IIR_G1.wav',z,Ft);
%get filtered audio file
[n Ff] = audioread('Filtered_IIR_G1.wav');
%get time based on sampling frequency for G10.wav
tf=(1/Ff:1/Ff:length(x)/Ff);
figure(3);
%plot the waveform in time domain
subplot(2,1,1);
plot(tf,n);
grid;
title('Filtered Waveform in Time Domain');
xlabel('Time');ylabel('Amplitude');
subplot (2,1,2);
%plot the waveform in frequency domain
m=length(n)-1;
ff=0:Ff/m:Ff;
nfft=abs(fft(n));
%perform fft
plot(ff,nfft);
grid;
title('Filtered Waveform in Frequency Domain');
xlabel('Frequency');
ylabel('Magnitude');
```

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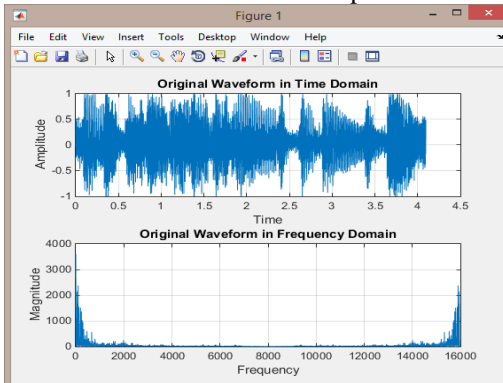*WINGS TO YOUR THOUGHTS.....*

ii.   MATLAB Waveform Output



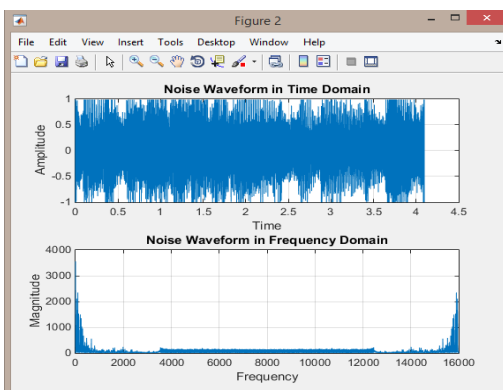**Figure 12:** Result of Original Audio Signal Waveform



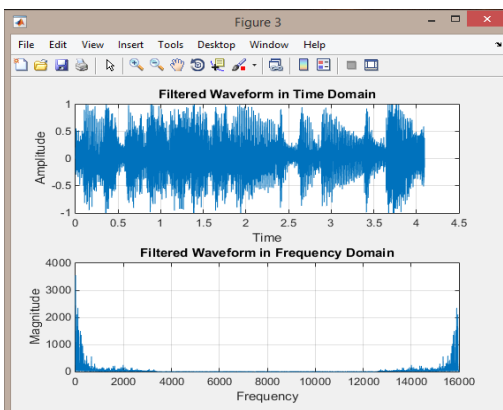**Figure 13:** Result of Noise Audio Signal Waveform



**Figure 14:** Result of the Filter Waveform by IIR Filter

## 4.   DISCUSSION

The audio signal is read into MATLAB and the waveform is observed. It is difficult to figure out the unwanted noise signal in the time domain waveform. Therefore, it is required to perform Fast Fourier Transform (FFT) to the signal into the frequency domain. From the waveform obtained after perform FFT, it is observed that the waveform is in "mirroring" at the frequency of 8000Hz. This is due to the effect of Nyquist Sampling theorem after perform the FFT. From the sampling frequency that obtained which is 16000Hz, then the bandwidth should be $B \leq f_s/2$. Hence, the Nyquist frequency should be obtained by using sampling frequency and divide it by 2 and get 8000Hz.

Therefore, the frequency domain waveform is "folding", "mirroring" or "symmetry" at the point of 8000Hz. After identifying the effect of Nyquist Sampling theorem after perform FFT, and then the signal can be observed as the unwanted noise is appeared at the high frequency band which is above 3500Hz as shown in the Figure 2. Therefore, it is decide to design the low pass filter. Then it is important to figure out the passband frequency and the stopband frequency when designing the filter which can be observed and decided from the noise signal as shown in Figure 2. After that, the passband deviation and stopband deviation also can be determined and decided from the waveform. In the FIR filter design, Hanning Window is chosen as to design our FIR filter. This is because the minimum stopband attenuation is obtained, 34dB is the most nearest to the Hanning Window which is less than 44dB. Then the transition bandwidth is calculated to obtain the length of the filter. The MATLAB function of "*fir1*" is used to create window-based finite impulse response filter design. In the IIR filter design, Butterworth filter is chosen as the design provided monotone in passband and stopband which is no ripple with the maximally flat in passband.

The MATLAB function of "*buttord*" is used to calculate the minimum order of a digital Butterworth filter and the cutoff frequency to meet the filter design specifications while "*butter*" is to create the IIR Butterworth filter. After designing both of the filters, the frequency response is plotted to ensure the filter design obtained is correctly. The filtered audio signal is then written into a new "*.wav*" file. As the result, the unwanted noise signal is being filtered as shown in the Figure 11 and Figure 14. The quality of filtered audio signal is not as good as the original audio signal due to some signal still existing within the range of the noise signal. Both FIR and IIR filter also has the same problem in term of the quality are being reduced, however, in general the noise signal is being filtered out and the objective is being achieved.

## 5.   CONCLUSION

At the end of this study, the digital audio signals could be recognized in both time domain and frequency domain waveform. By comparing the frequency domain waveform, the unwanted noise signal can be recognized. The entire unwanted noise signal has been filtered by using FIR and IIR filter.

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