IIR Based Digital Filter Design and Performance Analysis

Shapna Rani Sutradhar, Nazmus Sayadat,
Ashfiqur Rahman, Sirajum Munira, A.K.M. Fazlul Haque and Syed Nazmus Sakib
Department of Electronics and Telecommunication Engineering
Daffodil International University
Dhaka,Bangladesh
Email: shapna1449@diu.edu.bd

Abstract—Digital filters are mandatory for digital signal processing. This paper presents digital filter dispelling the unwanted signals or noise from the required signal and enhances the better performances of the signal. The extracted features of the digital filter have been analyzed to acquire the better output of the signal by using IIR Butterworth filter. It provides different designed parameters of IIR filter to achieve the desired result. MATLAB FDAtool is considered to find out the different responses of a digital filter. About eight parameters like- Phase response, Magnitude response, Magnitude & Phase response, Step response, Group delay, Pole/Zero Plot, Phase delay, Impulse response are used to analyze the filter responses. Some selected audio signals are used for observing the empirical response of high pass, low pass, band stop filter and band pass filter. A special tool is developed for this observation purpose.

Keywords— Digital Filters, Signal Processing, IIR Butterworth Filter, Responses, Impulse, Delay.

I.   INTRODUCTION

For any kind of communication system, signal processing is the leading part. To avoid unwanted signals, signal processing is mandatory. Analog and digital signal processing can be used to process information. More flexibility and better control of accuracy are provided by digital signal processing compared to analog signal processing. Filters are applied to process the signal. To process the raw signals, digital filters are very essential. The system which performs mathematical operations on a sampled, discrete-time signal to decrease or increase certain feature of that signal is called digital signal [1]. Different statistical methods may be introduced to observe the performance of a digital filter. Several observation methods may also be employed in designs, and often form the origin of a filter detail. Basically, one symbolizes filters by computing how they will react to a simple input such as an impulse. Another can then encompass this data to subtract the filter's response to more composite signals. In this research, different forms of filters like high-pass filter, low-pass filter, band-stop filter and band-pass filter have been used to analyze the performance of digital filter.

Some selected audio signals have been used to observe different digital filters response and also plotted and played

the output of the filters. The aims of this research are to find out the better performance of digital filters. To achieve this goal, it is needed to optimize the different parameters of digital filters. Parameters like magnitude response, phase delay, phase response, group delay, impulse response, pole/zero plots and step response have been used to optimize the result and determine the better output. To acquire the better output, IIR Butterworth filter has been used. The cooperation between attenuation and phase response is called Butterworth filter. Once in a while this is called a maximally flat filter because; it has no flow in the stop band or the pass band. The Butterworth filter acquires its flatness at the expenditure of a comparatively wide evolution section from pass band to stop band. The estimations of the mechanism of the Butterworth filter are more realistic and less vital than several filter types. Butterworth filters have practically better amplitude and transient manners [2]. Another main concern of this research is to use this designed filters in real and practical execution. All the simulation and results for the analyzing of different parameters have been analyzed and verified using FDA tool in MATLAB. A tool is developed for filtering audio signals in practical.

II.   LITERATURE REVIEW

Many works have been done on digital filter for signal processing. A.K.M. Fazlul Haque worked on enviable parameters extraction and enhanced performance of digital filters using FDATool [3]. Mohammad Saiful Islam, Shaila Shanjada and Mithun Banik worked on improved feature extraction of digital filters using FDATool [4]. Yaduvir Sing, Sweta Tripathi and Manoj Pandey performed an analysis of digital IIR filters with LabVIEW [5]. ZHANG Chengliang and WANG Aihong worked on IIR digital filter design research and simulation on MATLAB [6]. Ethan Elenberg et.al. worked on digital filter design for audio processing [7]. Er. Daljit Singh Bajwa, Er.Karamjeet Singh and Navpreet Kaur Chahal worked on survey paper of digital IIR filter design [8].

Ekta Yadav and Rupali worked on digital IIR filter design [9]. In the above research papers, all of them worked on analysis and design of IIR digital filter. Different types of design methods have been used for those analysis purposes like-window function method, impulse invariance method and some of them used LabVIEW software for their research. In this paper, diverse forms of filters like high-pass filter, low-pass filter, band-stop filter and band-pass filter have been used to examine the performance of digital filter. IIR Butterworth design method is used for this design purpose. A tool is also developed for filtering audio signal.

III.   METHODOLOGY

In this paper, the actions of different digital filters and the responses have been studied and analyzed based on different parameters. Also, the filter output of a selected audio signal has been examined.

MATLAB FDA (Filter design and analysis) tool has been used for analysis part and MATLAB GUIDE (Graphical user interface development environment) tool for the experiment of filtering audio signal.

The working procedure of this research has been illustrated in the following *Fig.2* and *Fig.3*.

A.   Why IIR filter instead of FIR filter:

Mainly, there are two categories of digital filters. Infinite impulse response (IIR) and Finite impulse response (FIR). Filters which usually have an impulse response containing only a finite number of values are typically called finite impulse response (FIR) filters [10]. FIR filters are involved in filtering problems where there is a condition for linear phase personality within the passband of the filter. The design of FIR filters is centered on a direct approximation of the particular magnitude response, with the frequently added obligation that the phase response be linear [11]. On the other hand, the filters which have impulse response of infinite dimension are called infinite impulse response (IIR). In this paper only IIR filters have been applied for analyzing different parameters. Because, IIR filters have more efficient filtering characteristics compared to FIR filters. IIR filters are normally used in the applications that don't need linear phase response. That means IIR filters usually have nonlinear phase response [12]. IIR filters require less memory as it needs fewer coefficients to perform similar filtering action and it is quicker than FIR filters. For this reason, IIR filters have been used.

B.   Designing digital filter with MATLAB:

1)   Creating filters in MATLAB GUI with GUIDE:

In order to create a GUI with GUIDE tool for selecting an audio signal, a digital filtering has been applied from high pass, low pass, band stop or band pass filters. Two

button groups are selected from the toolbar and renamed them to "Input" and "Filters". Then some push buttons are selected for those two groups. "Select Audio" and "Play Audio" button for the "Input" button group and "High Pass", "Low Pass", "Band Stop", "Band pass" button for the "Filter" button group. Four axes, axes1, axes2, axes 3, axes 4 are selected to plot the input and output in time domain and frequency domain. Axis1 and axis2 are used for the input and axis3 and axis 4 for plotting filter output in time and frequency domain.

When this GUI has been saved, it saves a ". Fig" and generates an ".m" file. The ".m" is the editable MATLAB file which can be modified to customize the app. In this MATLAB file, all the commands have been written to run the GUI. The work of every push buttons and the figures have also been defined to plot in the axes number.

2)   Working Process:

The block diagram shown in the *Fig.3* shows a typical working process of the designed GUI. First of all, an audio has been selected from the computer. Then the audio has been played with MATLAB and plotted the audio in time domain and frequency domain. One of the four designed digital filter is then selected to pass the audio. The output of the filter is also played and the filtered audio is also plotted in time and frequency domain.

3)   Step by step working of the GUI:

Following steps are involved while running the GUI in MATLAB.

a)   Selecting audio:

Then an audio has been selected from the computer. "Select Audio" button opens a popup window to browse and select a ".wav" file. Then our desired audio filter has been selected from a predefined directory. *Fig.5* shows the pop-up window to select the audio.
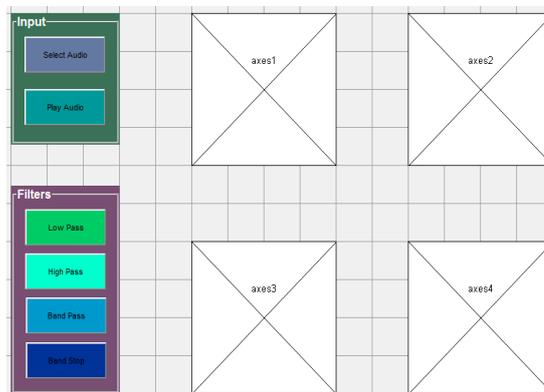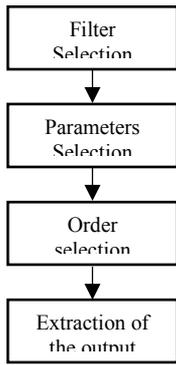

Fig. 1.   Editing GUI in MATLAB GUIDE tool

Fig. 2. Working procedure for finding the better output of filters for some selected parameters using different orders.
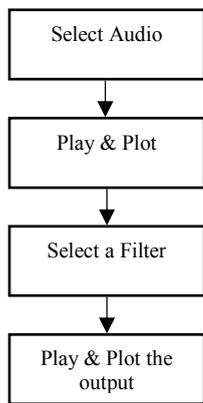

Fig. 3. Working block diagram of GUI for filtering audio signal.
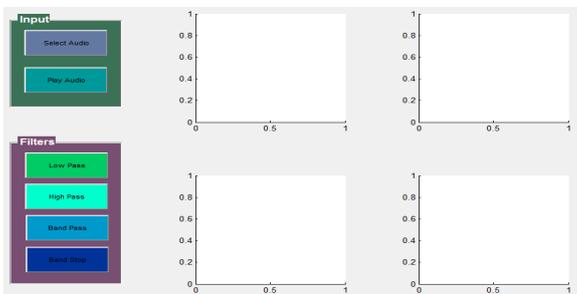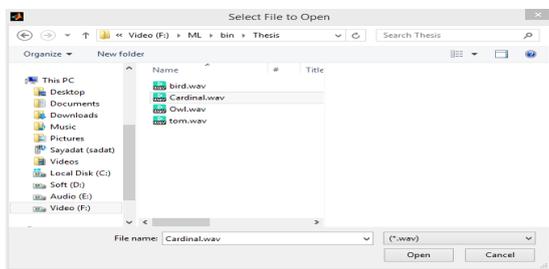

Fig. 4. Running the GUI.

Fig. 5. Selecting audio file from computer.

b) Play and plot the audio:

Pressing the "Play Audio" button plays back the selected audio and plot the audio in frequency domain and time domain. For simplification, .wav file is only used. To plot the audio, a time vector and a frequency vector have been created from the selected audio. The selected length of the fft was 2^{16}, which is also the length of the frequency of the audio. *Fig.6* shows the time and frequency domain representation of that selected audio.

IV. SIMULATION & ANALYSIS:

A. Input data table:

The input values that have been used to analyze the performance of different filters is shown in TABLE I. Different orders like- 400, 600, 800, 1000 and 1500 are randomly selected to find out the better output for different filters.
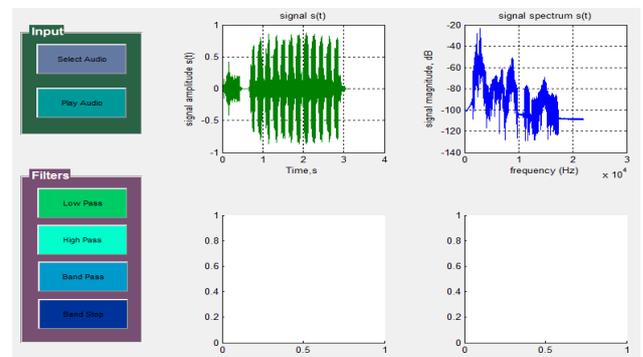

Fig. 6. Play & plot the selected audio

TABLE I: Input Data

Filter Type	Frequency Specifications(Hz)		Magnitude Specification
	F_s	F_c	
Low Pass	48000	10800	3 dB
High Pass	48000	10800	3 dB
Band Pass	48000	4800 (F_{c1})	13200 (F_{c2})	3 dB
Band Stop	48000	4800 (F_{c1})	13200 (F_{c2})	3 dB

B. Performance analysis of different filters:

1) Low pass filter:

a) Magnitude Response:

From the magnitude response, IIR Butterworth can easily find out the basic properties of filters. The *Fig.7* shows that, the filter passes only low frequencies while reject higher frequency. IIR Butterworth used different orders like 400, 600, 800, 1000 and 1500 to find out which gives the better output. Then it has been found out that, order 1500 gives more sharp output. It passes lowest frequency below cutoff frequency.

b) Phase Response:

Phase response is one of the most important parameters of digital filters. It shows the relationships between phases of the input and output signal. The phase response of low pass filters is incorporated with some applications such as time delay simulation, cascaded filter stages and particularly for process-control loops. IIR Butterworth low pass filter used different orders such as 400, 600, 800, 1000 and 1500 to evaluate the output which is shown in *Fig.8*.
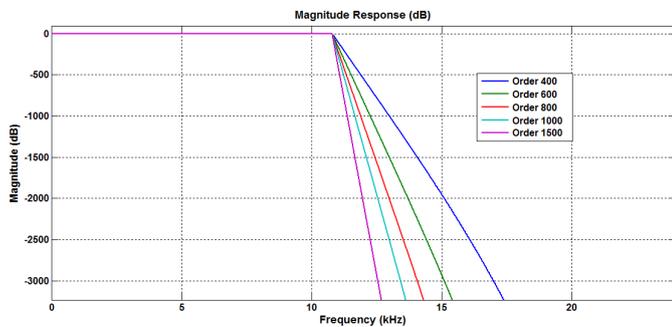
Fig. 7.   Magnitude Response of low pass filter for different orders
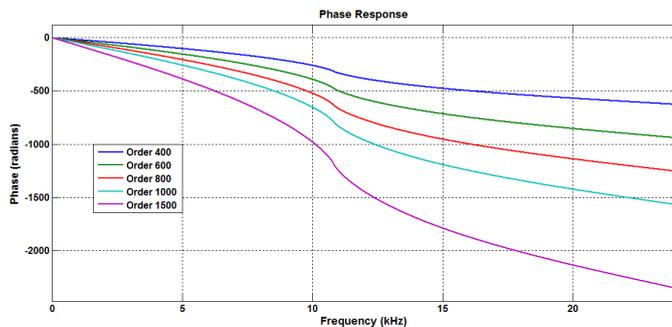
Fig. 8.   Phase Response of low pass filter for different orders

2) High pass filter:

a) Magnitude Response:

This is based on high pass filter and high pass filter can pass high frequency signal. Now IIR Butterworth is discussing about high pass filter in magnitude response and trying to know in what situation high pass filter can gives better result. Here the *Fig.9* shows the magnitude response of high pass filters. Butterworth takes various orders such as, 400, 600, 800, 1000, 1500 and it gives different result. In FDAtool, IIR Butterworth high pass filters have been used and found that, 1500 magnitude response's quality is better than others.

b) Phase Response:

When Phase response has been discussed, it can be said that, it is a part of a signal processing. If IIR Butterworth filter wants efficient phase response in high pass filter, then it measures some orders like- 400, 600, 800, 1000 and 1500 shows in the *Fig.10* which contains lower properties of high pass filter and the response of order 1500 is stronger.

3) Band Pass filters:

a) Magnitude Response:

Magnitude response in IIR Butterworth band pass filter is used to find out the basic characteristics of digital filter. The best magnitude response in band pass filter will be found out. For this, different filter orders like- 400, 600, 800, 1000 and 1500 have been used which showed in the *Fig.11* and it is definitely high filter order delivered the better output. So order 1500 is sharper and magnitude response takes the best output.
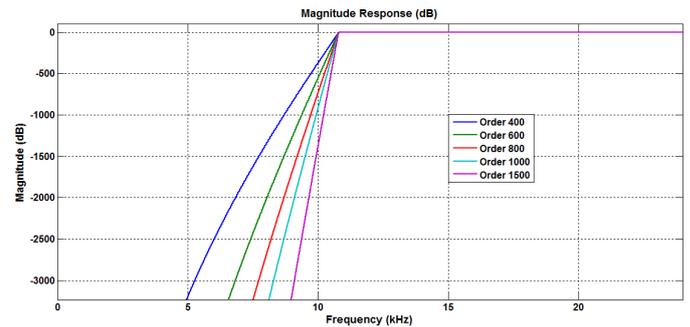
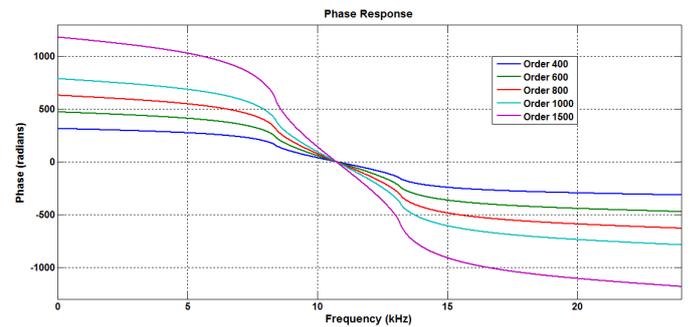Fig. 9.   Magnitude Response of high pass filter for different orders

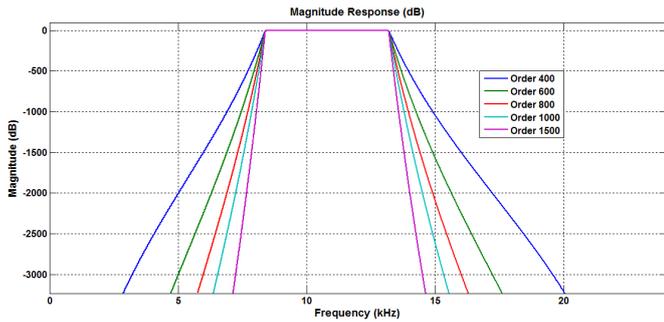Fig. 10.   Phase Response of high pass filter for different orders

Fig. 11. Magnitude Response of band pass filter for different orders

b) *Phase response:*

In IIR Butterworth band pass filter, the connection between the phase of a sinusoidal input and the output signal passing through any expedient is known as phase response. Here, in the *Fig.12*, IIR Butterworth uses three filter orders to get the best response where 1500 order gives the beneficial result.

4) *Band stop filter:*

a) *Magnitude Response:*

The magnitude response of a band stop filter is a filter that passes most frequency response is unchanged. A general ideal band stop filter can display both positive and negative angular frequency. In IIR Butterworth band stop filter in the *Fig.13* takes different filter order, order 400, order 600, order 800, order 1000 where higher order 1000 gets more priority and gives accurate output.

b) *Phase Response:*

In IIR Butterworth band stop filter, phase response is the stage of the output with the input as reference. In *Fig.14*, IIR Butterworth method gets different filter order for taking favorable output. Higher filter order deployed much better output.
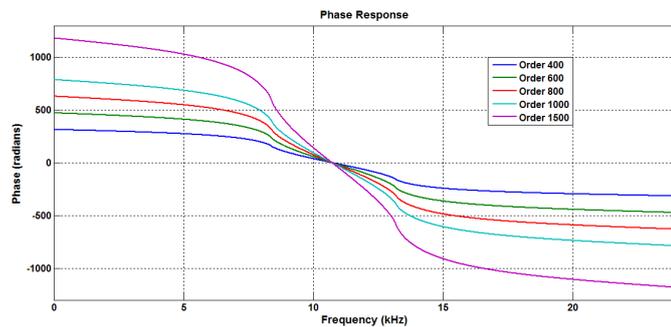

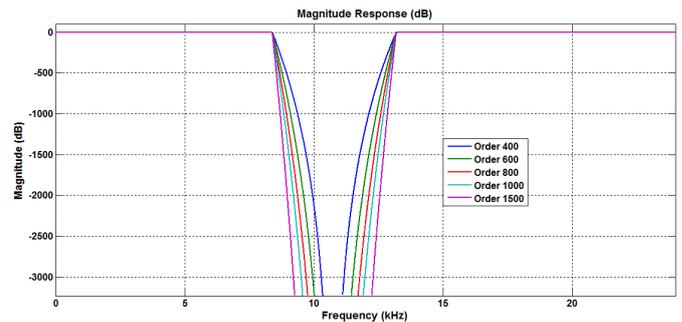Fig. 12. Phase Response of band pass filter for different orders

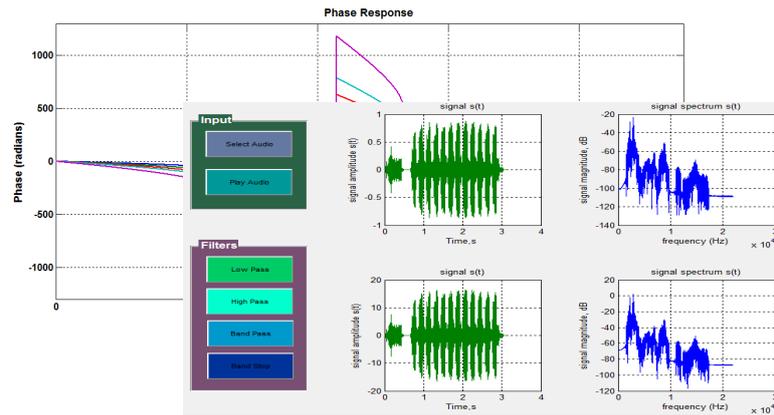Fig. 13. Magnitude Response of band stop filter for different orders

Fig 14. Phase Response of band stop filter for different orders

All Parameter's values have been taken accordingly.

C. *Filtering audio signal:*

1) *Low pass filtering:*

When "Low Pass" button has been selected, it executes a callback function in background, which generates a low pass filter output and plays back the filtered audio. To design a low pass filter, some parameters like, sampling frequency 44100 Hz, cutoff frequency 10000 Hz have been used. After filtering the audio signal, the change in the time and frequency domain could be visualized. *Fig.15* shows the output of the low pass filter. The change in the audio sound quality is also noticeable. It has been found out that, for a high frequency audio signal with noise, applying the low pass filter improves the sound quality by cancelling high frequency noise.

2) *High pass filtering:*

For filtering audio signal with high pass filter, a callback function is considered with high pass filter coefficient, which can be executed by pressing the "High Pass" button. For high pass filter, the sampling frequency and cutoff frequencies are the same as low pass filter 44100 Hz

and 10000 Hz. In this research finding, high pass filter was not a better choice for audio signal. Because, it cancels out all low frequencies while allows passing high frequencies with noise, though it could be useful for some specific applications. *Fig.16* shows the high pass filter output of the audio signal. The change in the amplitude level can be seen in this figure.

3) Band pass filtering:

For band pass filtering the sampling frequency is same as for low and high pass filtering. The pass band frequency is set from 1000 Hz to 2000 Hz.

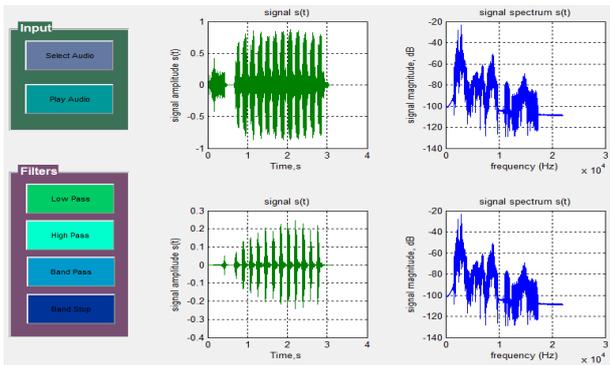Fig.15.   Audio signal after passing high pass filter.

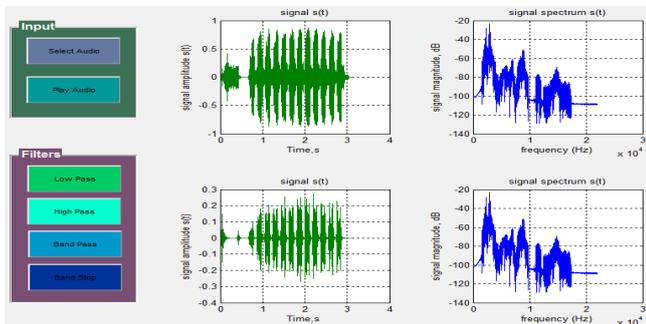Fig. 14.   Audio signal after passing band pass filter.

Fig. 15.   Audio signal after passing band stop filter.

4) Band stop filtering:

A call back function "Band Stop" button is also created to perform band stop filtering. The stop band frequency is set from 2000 Hz fast cutoff to 4000 Hz second cutoff for band stop filtering. Selecting the "Band Stop" button the output is generated as *Fig.18* and an audio is also played of that selected audio which is the band stop filer output.
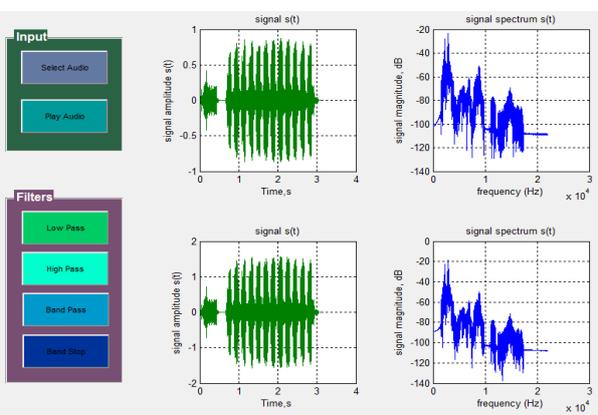
V.   CONCLUSION:

In this paper, different design parameters have been considered for IIR Butterworth filter to analyze the filter performance. Four categories of filters like high pass, low pass, band stop and band pass filters have been used for IIR Butterworth filter. To analyze the performance, MATLAB FDA tool has been used. To simulate the result in FDA tool, different orders like order 400, 600, order 800, order 1000 and order 1500 have been considered. From the performance of different order, it has been analyzed that the better output can be found from higher order. That means, better result has been found from order 1500 comparing order 1000, 800, 600 and order 400. The selected audio signal which passed through different filters can give better output in low pass filter than the other filters. MATLAB GUIDE tool is used to analyze the performance of digital filters in terms of audio signal.

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