

Speech Enhancement and Noise Suppression using FIR Filters

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ABSTRACT: The project aims to develop a program to remove noise from a live recorded speech signal using four types of finite impulse response (FIR) using MATLAB. The project involves design and realizing all the 4 types of filters i.e., low pass, high pass, band pass, band stop filters using the window approach. The audio is recorded live and is then subjected to filtration by obtaining the coefficients of the filter with the specified pass band and stop band frequencies with sampling frequency being set at 44100 Hertz. The stop band attenuation varies based on the amount of signal that needs to be attenuated. To verify the program we have coded, we use the Fast Fourier Transform and to prove that the filter chosen reduced noise, the FFT output will second the theoretical idea.^[1]

KEYWORDS: FIR Filters, Noise Suppression, Window Technique. Attenuation, Fourier Transform

I. INTRODUCTION

Finite impulse response (FIR)^[6] filters produce finite number of impulse responses for a given signal which settles down to zero at a definite point. FIR filters are designed using various techniques. We employ window technique in our project.

FIR filters are defined by the mathematical model:

$$y[n] = \sum b_i x[n - i]N_i$$

where: $x[n-i]$ is the sampled input signal delayed by 'i' units,

$y[n]$ is the output of the filter,

N is the order of the filter,

b_i is the impulse filter coefficient.

The working of a FIR filter^[7] is the summation of the product of the delayed input signal with the impulse filter coefficients. This can be visualized using the Z-transform for the difference equation of the filter. The difference equation can be obtained by the i^{th} derivative of $x(t)$. The difference equation with Z transform for a filter is given by

$$\sum Z \{x(n)\} = \sum xk_z - kN_k = 0$$

We follow the typical design flow of determining the filter order 'N' using the given parameters, choosing the suitable window and determining the impulse response for the filter. We have modeled our program to employ three types of windows for the filter - Rectangular Window, Hamming^[8] Window and Blackmann Window. Each window provides different levels of attenuation between the peaks of the main lobe and the side lobes and there is a considerable difference between their impulse responses.

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II. LITERATURE SURVEY

Filters are an essential part in any digital signal processor^[2] (DSP). Filter design is the process of designing a signal processing filter that satisfies a set of requirements, some of which are contradictory. The purpose is to find a realization of the filter that meets each of the requirements to a sufficient degree to make it useful^[3]. Band-pass filter was designed by using MATLAB which possesses superior characteristics of devising filters in a fast and effective way when using window functions originated in it, and the filter was applied to a real time speech signal^[4].

We describe a new Toolbox for the frequency-domain analysis and design of multivariable feedback systems, to be used with PC-MATLAB. Its capabilities are then illustrated by a worked design example, which shows the use of Nyquist criteria^[5].

III. FILTER DESIGN AND IMPLEMENTATION

The software in use is MATLAB^[9] version R2018a. MATLAB offers easy coding and has a large inbuilt library for various functions and provides additional support for interfacing the system and the physical entity. The input signal is recorded at the beginning of the execution of the code for a user defined duration using MATLAB inbuilt function **audiorecorder**. This recording is later passed on to the filter. The filter parameters are determined by the user defined values of pass band frequency, stop band frequency and stop band attenuation. Since the input signal is a speech signal, the sampling frequency is fixed to 44,100Hz as per Nyquist Rate for sampling^[10]. We created a MATLAB function called **subchoice** for the nesting of CHOICE statements to optimize the code. A value alpha is determined using N which determines the point about which the filter has a symmetrical response. Impulse Response is computed with its delayed version. The window coefficients are computed. Desired Impulse Response is computed by the summation of product between the window coefficients and Impulse Response. These values are passed on to the inbuilt function **freqz** which returns the n-point frequency vector H and angular vector W. Desired Impulse Response is the filter output at various instances. At 60dB attenuation, only 0.01% of the original signal is passed. Hence, we have fixed 60dB attenuation for all test cases. Accordingly, Blackmann^[11] Window was used for the same. The implementation of the filter was used to filter out the noise of a live recording.

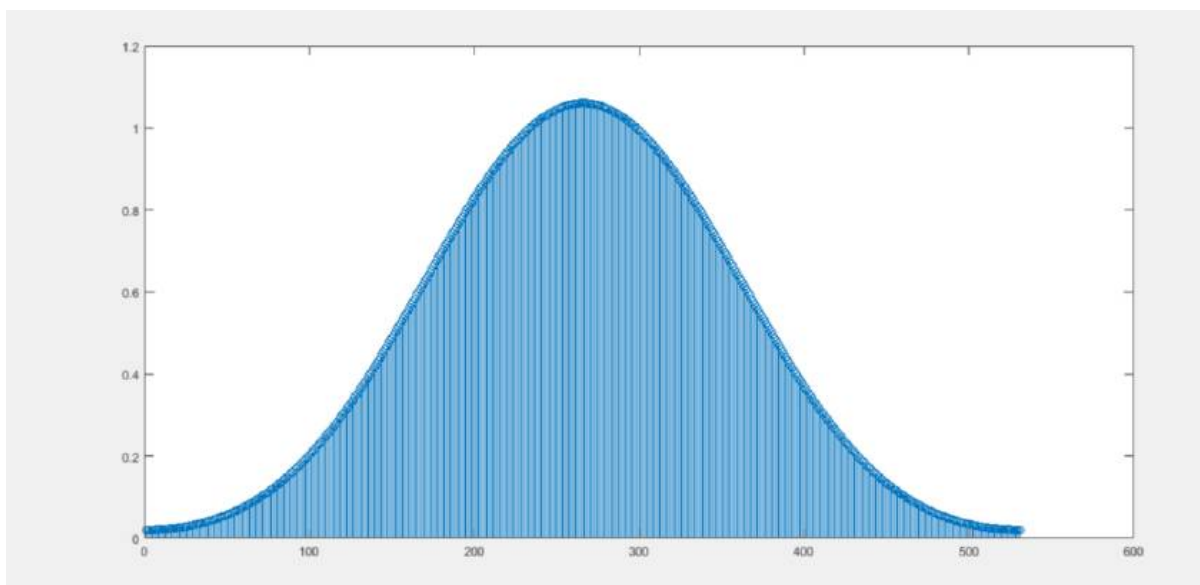


Figure 2.1 – A plot of amplitude versus number of samples for Blackmann Window which then will be multiplied with filter coefficients to get the filtered output.

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IV. RESULTS

To obtain results that are within the vicinity of the speech enhancement and noise suppression, the range of speech signal i.e., 300Hz to 4000Hz was chosen. The stop band attenuation was chosen to be 60dB as no significant attenuation was observed for stop band attenuation values greater than 60dB. All 4 types of filters - Low Pass, High Pass, Band Pass and Band Stop were studied and analyzed^[12].

a) Low pass filter - The low pass filter attenuates any signal with frequency greater than the Stop Band Frequency measured in Hertz.

The following specifications were used:

Pass band frequency: 4400Hz

Stop band frequency: 4500Hz

Stop band attenuation: 60dB

The input was a live recording and was in a noisy environment. The output observed had significant decrease in the noise and the speech was enhanced.

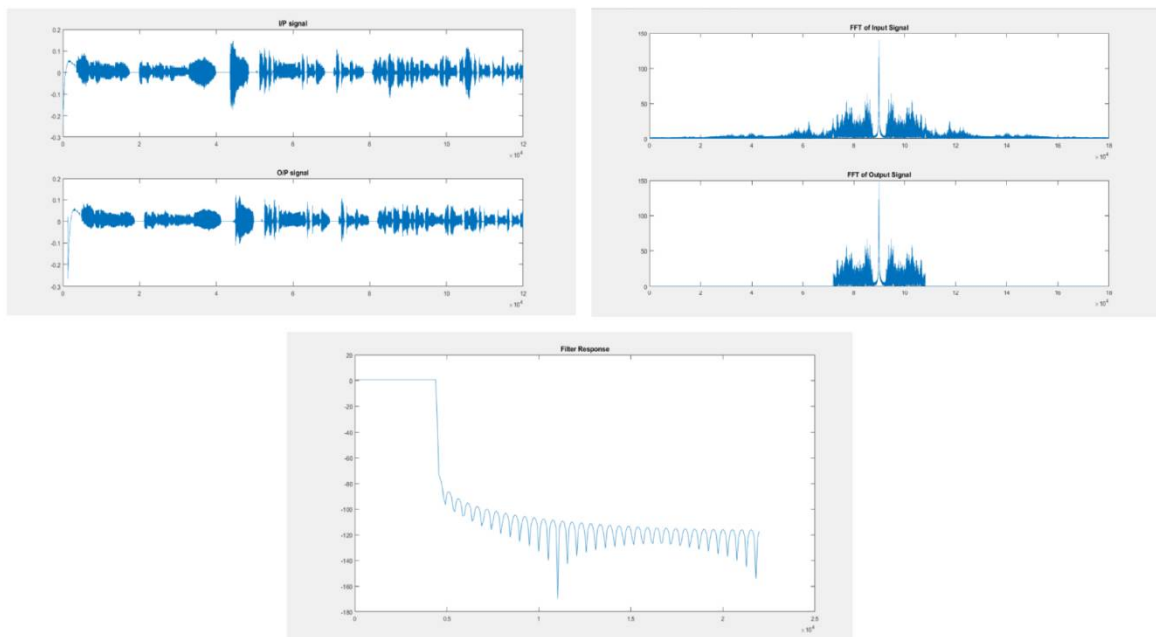


Figure3.1 - Input and output signal (before and after filtering out noise), FFT of the input and output signal for noise elimination and filter response of low pass filter.

b) High passfilter - It was observed that noise which has frequency less than the stop band frequency. The filter order was found to be 2647; hence, the maximum value for the input impulse was at 1324. There was some loss in the signal as not much information is stored at higher frequencies.

The following specifications were used:

Pass band frequency: 5000Hz

Stop band frequency: 4900Hz

Stop band attenuation: 60dB

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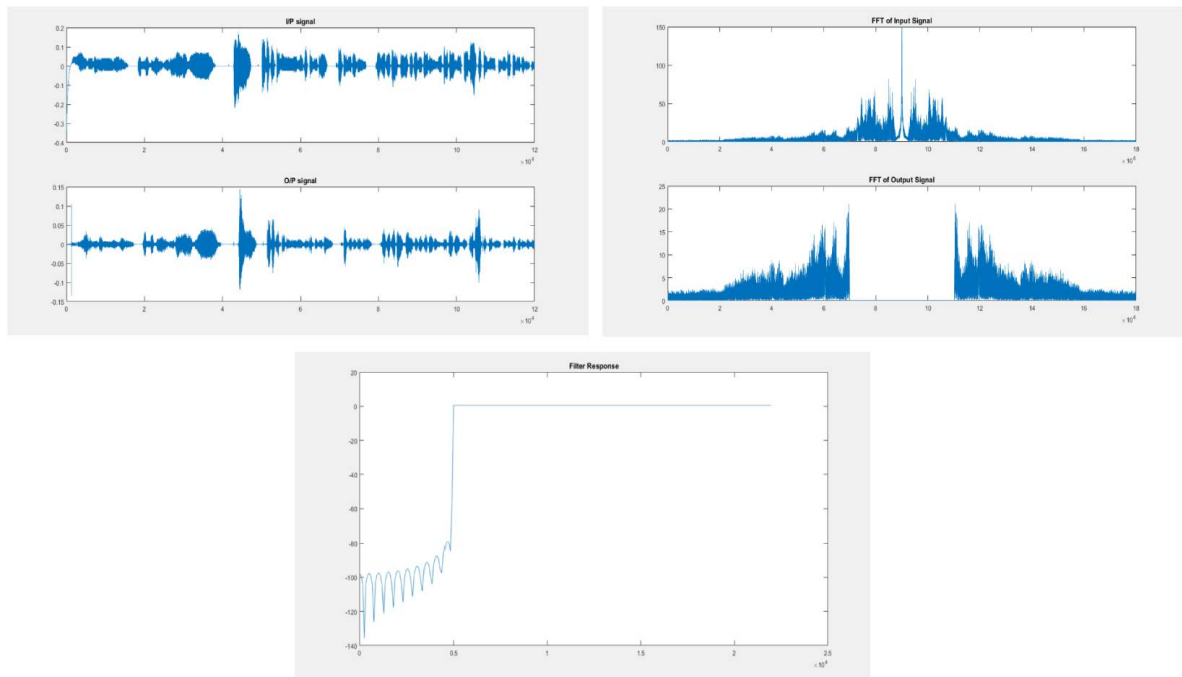


Figure 3.2 - Input and output signal (before and after filtering out noise), FFT of the input and output signal for noise elimination and filter response of high pass filter

c) Bandpass filter - The band pass filter was used to determine the range of frequencies the filter would allow in a speech signal. There was a great reduction in the noise and enhancement in the speech signal. The FFT of the input and output signals show that the noise was reduced to a large extent.

The following specifications were used:

First stop band frequency: 1950Hz

First pass band frequency: 2000Hz

Second pass band frequency: 3950Hz

Second stop band frequency: 4000Hz

Attenuation at the stop bands was selected to be 60dB. The order of the filter was 5293.

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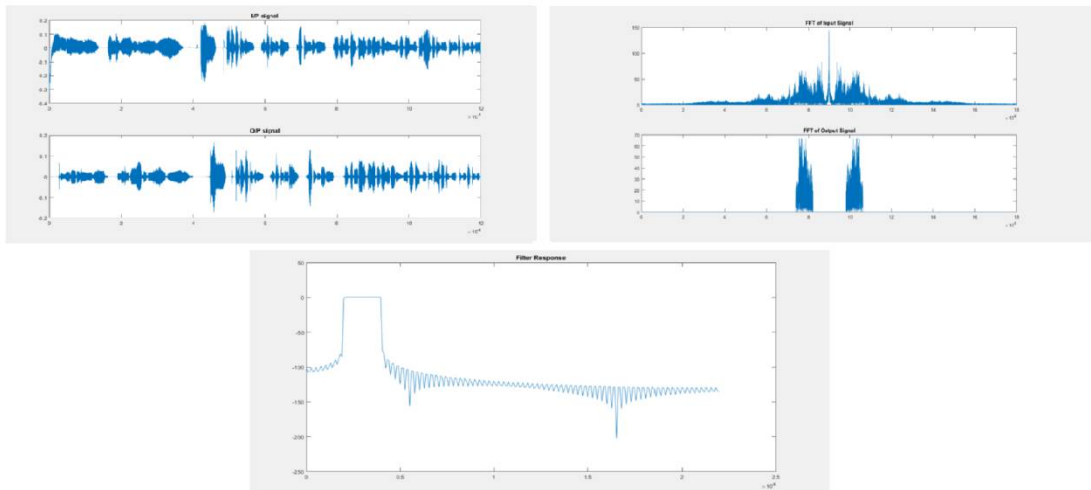


Figure 3.3- Input and output signal (before and after filtering out noise), FFT of the input and output signal for noise elimination and filter response of bandpass filter

d) Bandstop filter - It is a filter that passes most of the signals unaltered but attenuates the range of specified signals.

The following specifications were used:

First pass band frequency: 2000Hz

First stop band frequency: 2100Hz

Second stop band frequency: 3900Hz

Second pass band frequency: 4000Hz

The order of the filter was found to be 2647. The speech signal was greatly attenuated with not much suppression in the noise.

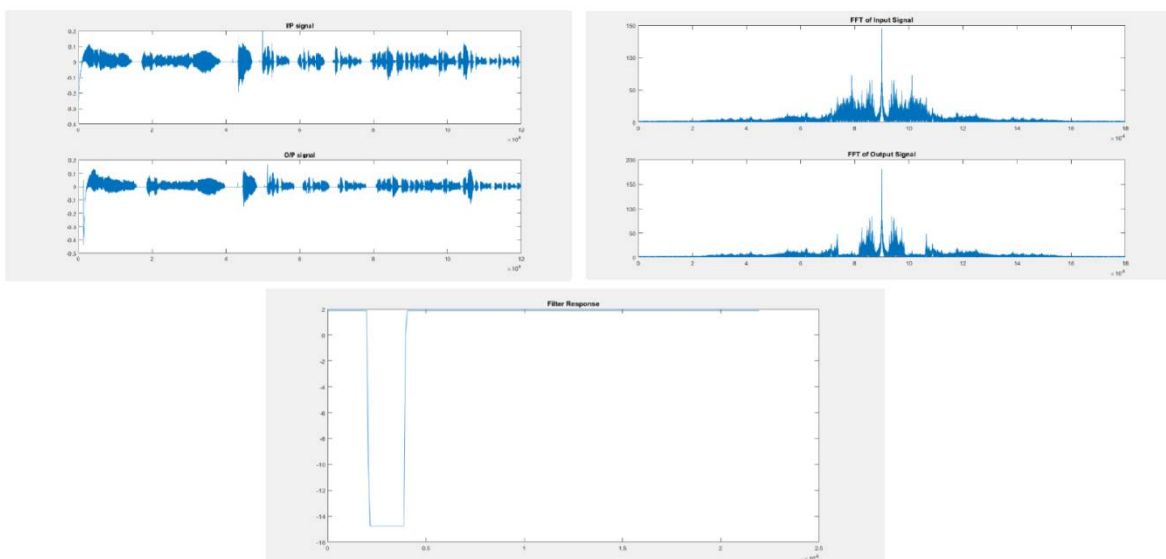


Figure 3.4 -Input and output signal (before and after filtering out noise), FFT of the input and output signal for noise elimination and filter response of bandstop filter.

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V. CONCLUSION

The filter designed and implemented using MATLAB was tested using a live recording in a noisy environment to observe the noise suppression and speech enhancement. The four filters were the conventional Low Pass, High Pass, Band Pass and Band Stop filters. The window technique was used to realize the filter. The results showed a great improvement in the speech and a drastic suppression of noise. Although noise cannot be eliminated completely, it was significantly reduced. The voice output of the speaker was heard clearly through the speakers as opposed to the input signal which was induced with noise. This paper aimed to analyze the noise suppression in the speech signal using the Finite Impulse Response Filter.

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REFERENCES

- [1] Simon Haykin and Michael Moher, "Communication Systems", International Student Edition, Fifth edition, ISBN: 978-81-265-2151-7, Wiley India Pvt. Ltd., 4435-36/7, Ansari Road, Daryaganj, New Delhi 110 002
- [2] E. Abdel-Raheem, A. Tawfik, "Proceedings of Canadian Conference on Electrical and Computer Engineering", pp 680-683, 1993
- [3] Pooja Yadav, Pooja Kumari, "REVIEW PAPER ON FIR FILTER DESIGN", IJIRT, Volume 1 Issue 12 | ISSN: 2349-6002, pp. 593-596, 2015
- [4] Zhenhe Sun, Wei Yu, Chen He, "FIR digital filter design and MATLAB simulation", Proceedings of 2012 International Conference on Measurement, Information and Control, 2012
- [5] JM Boyle, MP Ford, JM Maciejowski, "A Multivariable toolbox for use with MATLAB", 1988 American Control Conference, March 10, 2009
- [6] John G. Proakis and Dimitris G. Manolakis, "Digital Signal Processing", 4th edition, ISBN: 13:978-0131873742, 10:0131873741, Pearson Education Limited, Edinburgh Gate, Essex CM20 2JE
- [7] Ku. Damini C. Dandade, "Review on Design of Digital FIR Filters", International Research Journal of Engineering and Technology, Volume: 04 Issue: 02, pp. 358-362, 2017
- [8] Mohammed Mynuddin, Md. Tanjimuddin, Md. Masud Rana, Abdullah. "Designing a Low-Pass FIR Digital Filter by Using Hamming Window and Blackmann Window", Science Journal of Circuits, Systems and Signal Processing, pp. 9-13, 2015
- [9] Andre Quinquis, "Digital Signal Processing Using MATLAB", First Edition, ISBN: 978-18-482-1011-0, Wiley Publications Pvt. Ltd, ISTE Ltd.
- [10] Moshe Mishali, Student Member, IEEE, and Yonina C. Eldar, Senior Member, IEEE, Oleg Dounaevsky and Eli Shoshan, "Xampling: Analog to Digital at Sub-Nyquist Rates", CCIT Report No. 751, EE Pub No. 1708, Department of Electrical Engineering Technion - Israel Institute of Technology, December 10, 2009
- [11] Gao Xindo, "FIR and IIR Filters: 4.2. FIR Filter Design", School of Electronic Engineering, Xidian University, <http://see.xidian.edu.cn/teach/matlabdsp/>
- [12] Marcelo Basilio Joaquim, Carlos A. S. Lucietto, "A nearly optimum linear-phase digital FIR filters design", Digital Signal Processing, pp. 690-693, 2011