

IIR Filter Design for De-Nosing Speech Signal using Matlab

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ABSTRACT

The design of filter has become the core issues of the signal processing. Generally speaking, filter can be divided into analog filter and digital filter. Today, the development of analog filter has been more mature. However, digital filter has many advantages, such as higher stability, higher precision. With the development of digital technology, using digital technology to realize filter function is widely used. A MATLAB-based digital filter design procedure designs the filter and applied to the voice signal. In this project, the recorded speech with simulated noises is processed. The speech signal "I Love Electronics" is taken as the input signal. AWGN is added with the input speech signal. Two types of IIR filters Butterworth and Chebyshev are designed and are applied to the noisy speech signal. The magnitude response, phase response, impulse response and order of the filters are generated.

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KEYWORDS: IIR Filter, Speech, Chebyshev, Butterworth

I. INTRODUCTION

The basic functional need for filtering is to pass a range of frequencies while rejecting others. This need for filtering has many technical uses in the digital signal processing (DSP) areas of data communications, imaging, digital video, and voice communications. Analog filters are continuous-time systems for which both the input and output are continuous-time signals. Digital filters are discrete time systems whose input and output are discrete time signals. Digital filters are implemented using electronic digital circuits that perform the operations of delay, multiplication, and addition. Analog filters are implemented using resistors, inductors, capacitors, and, possibly, amplifiers. Digital filters can be implemented using integrated circuits so that per unit cost of digital filter construction is less than a comparable analog filter.

Human voice is a commonly useful tool and it is the most important means to pass information to each other. Passing message by voice is the most important and effective way for mankind. Now with the development of the times, mankind has entered the information age, with the modern means of speech signal study, people can generate, transmit, store, access, and apply voice messaging more effectively, which has a very important significance for the promotion of social development.

Sometimes people may want to record a motivating speech, a significant lecture suddenly happen, or listen to an old

speech more clearly. However, professional audio recording devices are usually not accessible, which means mobile phones are always the only choice. Due to the undesirable noise of the environment and, poor recording quality of mobile phones and devices decades ago, noise reduction and speech enhancement is needed to get the satisfied records. In this work, the speech "I Love Electronics" with simulated noises are processed. A voice is collected and filters the noise by using IIR filters. IIR digital filters (Butterworth and Chebyshev lowpass filters) are designed and applied to the noisy signal. Then, the system is implemented in MATLAB.

II. PROPOSED SYSTEM DESCRIPTION

The speech signal is taken as the input signal. Gaussian White Noise is added with the input speech signal. Random noise is also added to the specific frequency value. Two types of IIR filters Butterworth and Chebyshev are designed and are applied to the noisy speech signal. The magnitude response, phase response, impulse response and order of the filters are generated. Figure 1 shows the overall process of the proposed method.



Figure1. System Block Diagram

III. SYSTEM STRUCTURE

The system structure will be shown in details below. Figure 2 shows the flow diagram of the proposed system.

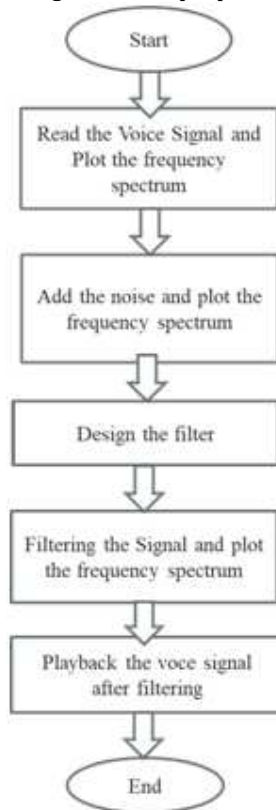


Figure2. Flow Diagram of the Proposed System

The function of system is to obtain the samples from voice card. Data analyses parts mainly have functions below:

- Collect the speech signal
- Display voice signals frequency domain figure and time domain figure.
- Put a random noise signal by using MATLAB into the original signal
- Design IIR filter to filter the noise
- Noisy signal through the filters and
- Replay the voice

The IIR filters are designed by following procedures:

- Step1.** Using the prescribed specifications, calculate minimum order of filter (n) and cutoff frequencies ω_n for an analog filter.
- Step2.** Form the normalized transfer function for n order.
- Step3.** Apply the analog-filter transformation.
- Step4.** Apply the bilinear transformation

This work considers the details of the above design procedure with the boundary frequency in band pass of this filter is 300Hz, the cutoff frequency in stop band is 500Hz, sampling frequency is 8000Hz, the maximum attenuation in band pass is 1dB, the minimum attenuation in stop band is 50dB.

IV. FILTER DESIGN SIMULATION PARAMETERS

Digital IIR filters have been derived from their analog counterparts. Classical prototype analog filters are Butterworth, Chebyshev I, Chebyshev II and Elliptic. Parameters required designing IIR filters are sampling frequency, Pass band edge frequency, Stop band edge frequency, Pass band ripples and Stop band ripples. Using

these parameters, magnitude response, phase response, impulse response and order of the filter can be generated.

In this work, the IIR Filter is designed that satisfy the following specifications:

- frequency at the start of the pass band in normalized frequency units - 300Hz (0.075π)
- frequency at the end of the stop band in normalized frequency units - 500Hz (0.125π)
- Passband ripple in decibels - 1
- Stopband attenuation in decibels - 50
- Sampling Frequency in Hz - 8000 Hz

V. FILTER DESIGN SIMULATION RESULTS

In this work, IIR lowpass filter Butterworth and Chebyshev are designed by using the design parameters in the above section.

A. Butterworth Filter

The transfer function for Butterworth filter is as follow:

$$z^{13} - 10.89 z^{12} + 54.9 z^{11} - 169.6 z^{10} + 357.9 z^9 - 545.3 z^8 + 616.8 z^7 - 524.5 z^6 + 335.2 z^5 - 159 z^4 + 54.41 z^3 - 12.72 z^2 + 1.821z - 0.1205$$

$$8.379 \times 10^{-13} z^{13} + 1.09 \times 10^{-11} z^{12} + 6.531 \times 10^{-11} z^{11} + 2.398 \times 10^{-10} z^{10} + 5.986 \times 10^{-10} z^9 + 1.079 \times 10^{-09} z^8 + 1.436 \times 10^{-09} z^7 + 1.439 \times 10^{-09} z^6 + 1.078 \times 10^{-09} z^5 + 5.994 \times 10^{-10} z^4 + 2.395 \times 10^{-10} z^3 + 6.539 \times 10^{-11} z^2 + 1.089 \times 10^{-11} z + 8.383 \times 10^{-13}$$

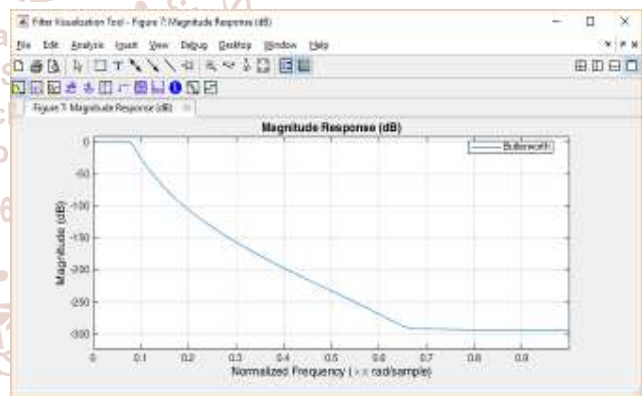


Figure3. Magnitude Response of Butterworth Lowpass Filter

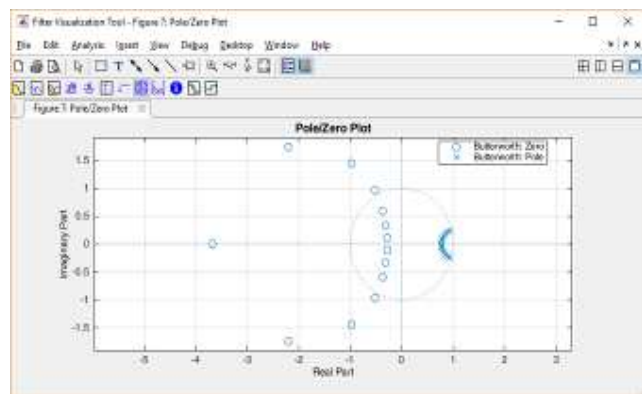


Figure4. Pole/Zero Plot of Butterworth Lowpass Filter

Figure 3 is Butterworth lowpass filter; it can be seen that the cutoff frequency is 300 Hz (0.075π), which means the performance of low pass filter is good. It has a perfect cutoff frequency that can filter all high components.

Figure 4 shows the pole/zero plot of the proposed designed Butterworth lowpass filter. All poles are in the unit circle and all zeros are on the unit circle. Therefore the filter is stable. The simulation shows that the designed filter is stable.

B. Chebyshev Filter

The transfer function for Chebyshev filter is as follow:
 $z^7 - 6.688 z^6 + 19.26 z^5 - 30.98 z^4 + 30.03 z^3 - 17.55 z^2 + 5.727z - 0.8045$

$$8.746 \times 10^{-09} z^7 + 6.122 \times 10^{-08} z^6 + 1.837 \times 10^{-07} z^5 + 3.061 \times 10^{-07} z^4 + 3.061 \times 10^{-07} z^3 + 1.837 \times 10^{-07} z^2 + 6.122 \times 10^{-08} z + 8.746 \times 10^{-09}$$

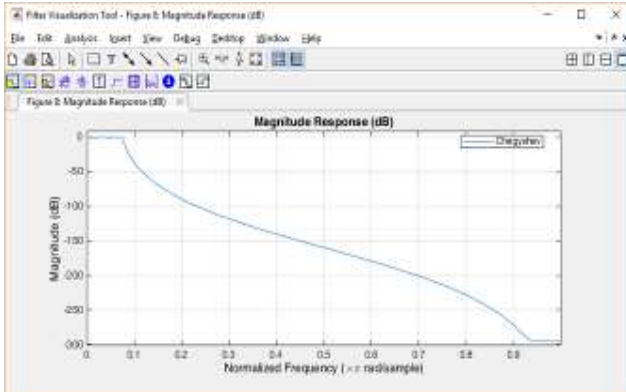


Figure5. Magnitude Response of Chebyshev Lowpass Filter

Figure 5 is Chebyshev lowpass filter; it can be seen that the cutoff frequency is 300 Hz (0.075*pi). It has a perfect cutoff frequency that can filter all high components. Figure 6 shows the pole/zero plot of the proposed designed Butterworth lowpass filter. All poles are in the unit circle and all zeros are on the unit circle. Therefore the filter is stable.

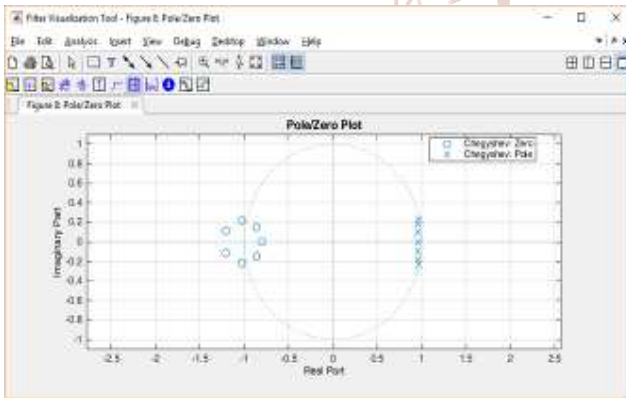


Figure6. Pole/Zero Plot of Chebyshev Lowpass Filter

VI. STEP BY STEP TESTING RESULTS

The simulation results of the proposed methods are shown in the below figures.

A. Collection of Voice Signal

Collect the digital signals by using the sound card on pc and WINDOWS operating system. Microphone is already built in laptop. So it isn't needed to add the microphone for recording the voice. For recording the voice the "recording.m" program is run and said "I love electronics" and then stop recording. The length of the recording voice is 3 sec in this work. After recording the voice, the speech we have made is replayed.

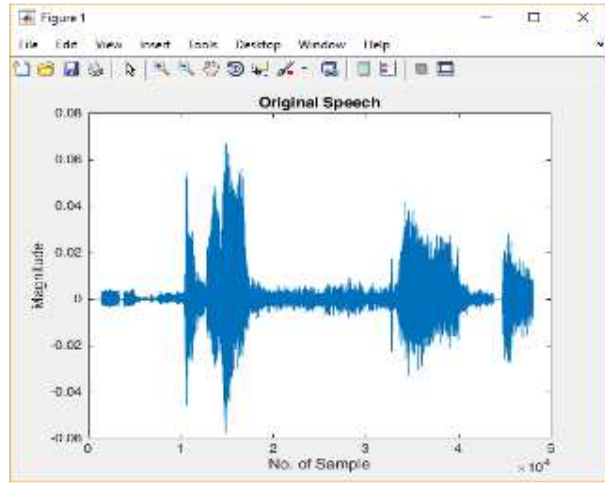


Figure7. Original Speech in Time Domain

The Figure 7 shows the original signal representation of speech signal which was spoken by a woman. The voice is recorded and it is stored as a wave file for further usage in MATLAB.

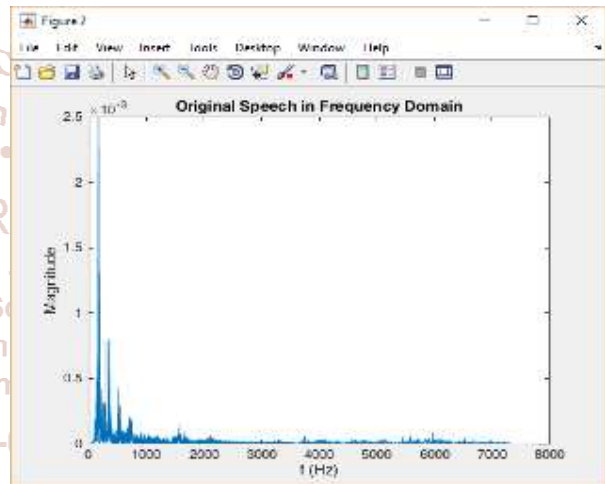


Figure8. Original Speech in Frequency Domain

B. Noising Signal

There are some common kinds of noise, such as periodic noise, pulse noise, wideband noise, lombard effect and etc. Due to the complexity of possible noise, the comparably simple one, white noise, is selected to simulate the noise in speech. The Figure 9 shows the noisy input signal representation which is treated as the original signal. Noisy speech signal is the AWGN added speech signal.

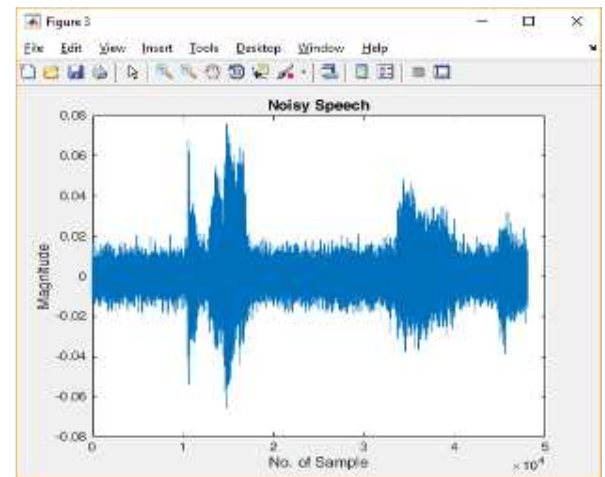


Figure9. Noisy speech in Time Domain, SNR=5 dB

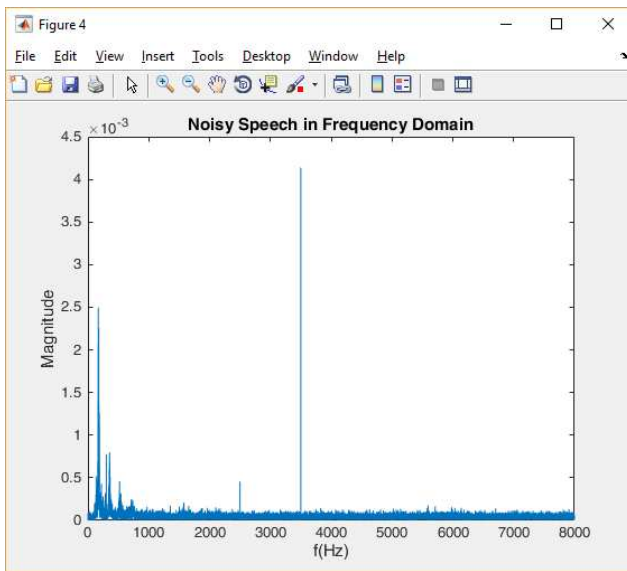


Figure10. Noisy speech in Frequency Domain, SNR=5 Db

C. Filtering Speech Signal

In this work, the two filter design Butterworth and Chebyshev are implemented. Figure 11 and 12 shows the filtering results with Butterworth filter.

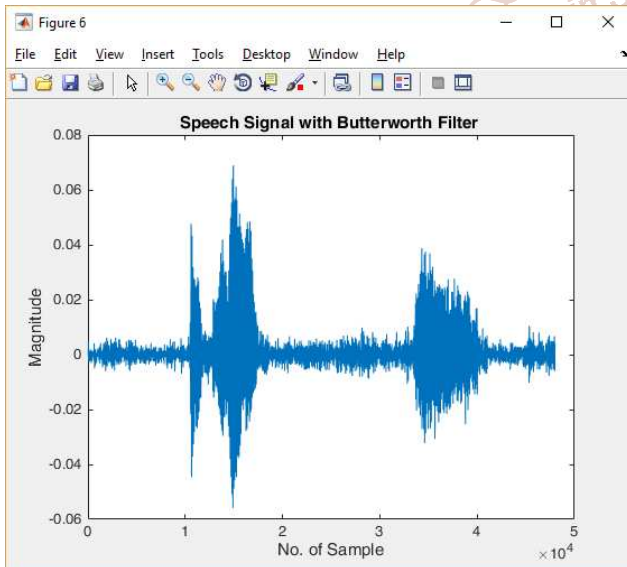


Figure11. Filtering Speech in Time Domain (Butterworth)

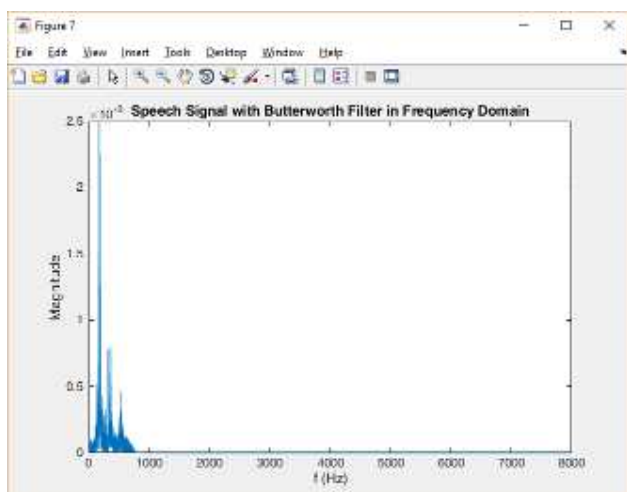


Figure12. Filtering Speech in Frequency Domain (Butterworth)

Figure 13 and 14 are described the filtering results with Chebyshev filter. We applied this filter to the noisy signal; it can be seen that the high component has been removed.

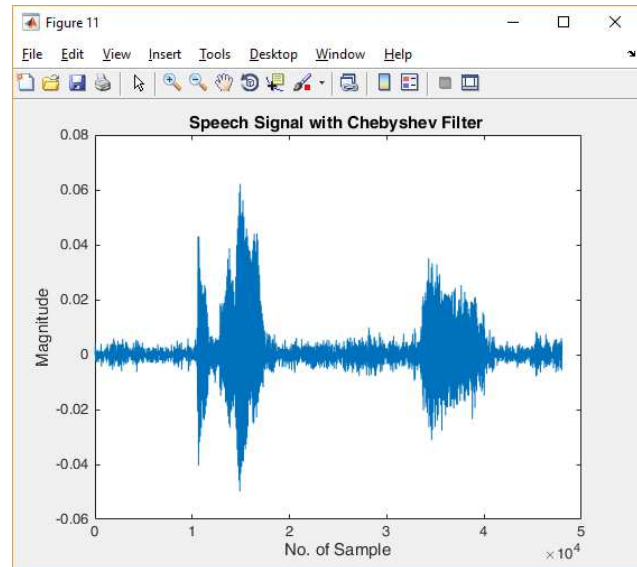


Figure13. Filtering Speech in Time Domain (Chebyshev)

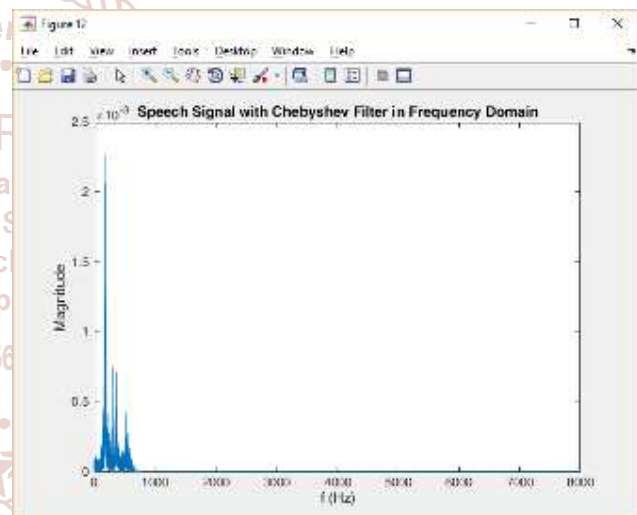


Figure14. Filtering Speech in Frequency Domain (Chebyshev)

And the waveforms width is reduced because of the removed noise. Then when playback the sound it is heard the noise sound is lowering than before.

VII. CONCLUSION

In this work, two types of IIR filter are applied for noise removal from speech signal. Noisy signal with Gaussian white noise and random noise was prepared by adding noise with clean signal using MATLAB software at 5dB SNR levels. The proposed digital IIR filter for speech signals has been designed. In this project, DSP tools from class like time domain analysis and Fast Fourier Transform are used in speech analysis. "Fdatool" are used to analyze the filter design. Two different filters are applied to original speech to reduce noise.

In this work, voice card and microphone that built in laptop are used for voice signal recording. The voice signal analysis and process can be done well using the strong data processing function in MATLAB.

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