

# AUDIO SIGNAL FILTERING WITH LOW-PASS AND HIGH-PASS FILTERS

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## Abstract

***The audio signal is always composed the signal with noise and these are not separated. Noise is the part of waveform with specific frequency range. In this system female speech.wav audio file is analyzed by reducing unwanted noise i.e unwanted spectrum of frequencies. The audio signal's noise can reduce by using filtering methods and in this system high-pass and low-pass filters which are designed in Matlab with the sampling frequency 8000 Hz. The high-pass filter is used to attenuate the low-frequency content in input audio signal and it can differentiate noise by choosing a cutoff frequency. The low-pass filter is established the limited bandwidth of a low frequency.***

***Keyword: audio, low-pass, high-pass, filter***

## 1. INTRODUCTION

A signal can be the form of data, voice, picture and so on and the audio is the type of waveform with human audible frequencies range. The audible range for human is approximately 20 Hz to 20 KHz. The audio signal's characteristics are bandwidth, nominal level and power level in decibels (dB) and the voltage level. The types of audio signals include speech, music and sounds. The noise is the unpleasant sound or unwanted sound that may cause in desired sound. The white noise is contained in every frequency within the range of human hearing. And pink noise can occur when the variation of the white noise and it can filter to reduce the volume. The filter is a circuit or device that passes certain frequencies and block other frequencies and it contains the pass band and stop band. The range of frequencies that are allowed to pass is referred to as pass band and stop band is stopped the range of frequencies. The low-pass filter passes the all frequencies lower than the

cutoff frequency and stops the higher frequencies. Similarity, the high-pass filter is allowed the higher frequencies to pass and removes the other lower frequencies. The cutoff frequency is the boundary in frequency response between pass band and stop band.

## 2. LITERATURE REVIEW

[1] The author analyzed the speech signal by using Discrete Wavelet Transform (DWT) to reduce unwanted higher or lower frequency components in speech signal. They assumed that high amplitude DWT coefficients represent the speech signal and low amplitude coefficients represent the noise signal. These coefficients were threshold and transformed back to time domain to get audio signal with reducing noise. [2] The audio signal may include noise while transmission therefore de-noising techniques are applied to reduce the unwanted signal from audio. The authors were used DWT and Butterworth filter, elliptical filters, Chebyshev filters with low-pass, high-pass and band-reject. [3] The authors were designed FIR filter using rectangular and hamming windowing method. The magnitude and phase response are analyzed by using Matlab simulation. The windowing method is simple to implement and require minimum amount of computational effort.

## 3. BACKGROUND THEORY

This system is analyzed the female speech audio signal by filtering low-pass and high-pass filter for reducing noise in Matlab. Moreover the low-pass and high-pass filters are designed with fdatool in Matlab. The female speech audio has the sampling frequency of 8000Hz. Firstly read the input audio signal and plot the visualization of the spectrum of frequency by using spectrogram and then estimate the signal's power

spectral density. Then normalized with Fourier transform and designed the filters to analyze the input audio signal.

### 3.1. Low-pass Filter

The cutoff frequency is the mainly associated with the low-pass filter which is the beginning and end points of frequencies. The transition band is the area between where a filter turns the corner and hit the bottom. In the transition band, the steepness of the slope is important in defining the sound. The sample of processing for low-pass filter is illustrated in Figure 1.

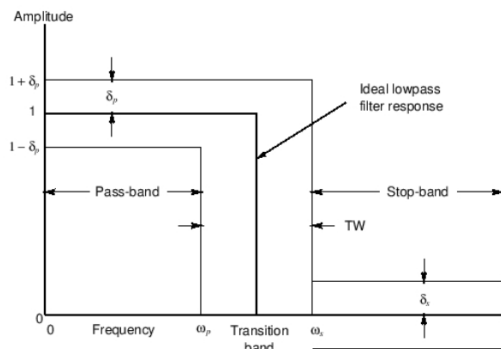


Figure 1. Observation of Low-pass Filter [4]

- $\delta_s$  = stop-band ripple
- $\delta_p$  = pass-band ripple
- $w_s$  = stop-band edge frequency
- $w_p$  = pass-band edge frequency
- TW: transition width =  $w_s - w_p$

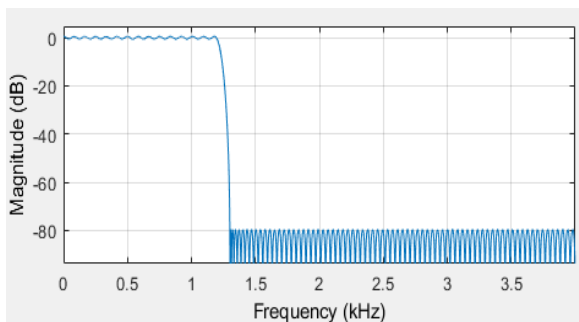


Figure 2. Design Low-pass Filter in Matlab

### 3.2. High-pass Filter

The high-pass filter is passed high frequencies above the cutoff frequency and it can be used to remove low frequency content from the audio signal. The process of high-pass filter and the design in Matlab are shown in Figure 3 and 4.

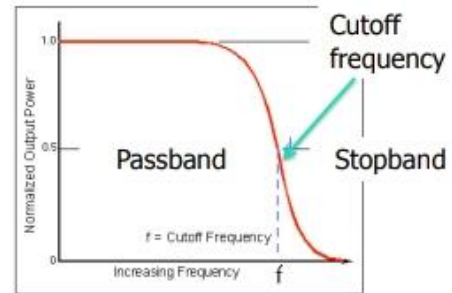


Figure 3. Observation of High-pass Filter [4]

The high-pass filter is used in studio recording and sound reinforcement to eliminate the extraneous low frequency noise. The use of high-pass filter on speech or vocals, it can improve the clarity of the voice. The main in high-pass filter is how apart from the cutoff frequency is slope.

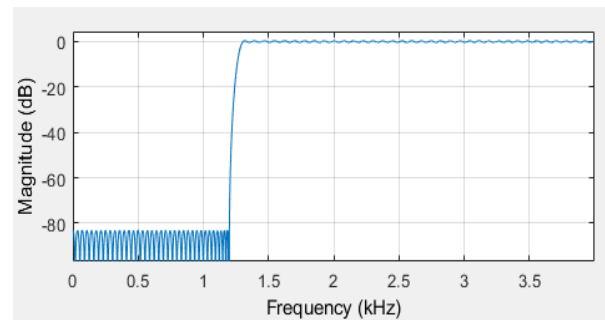


Figure 4. Design High-pass Filter in Matlab

### 3.3. Fourier Transform

The fourier transform convert a function from time domain to frequency domain with no loss of information. The signal sample over a period of time and divided into its frequency components. The equation of fourier transform is

$$F(w) = \frac{1}{\sqrt{2\pi}} \int_{-\infty}^{\infty} f(t) e^{-iwt} dt$$

Where w is frequency spectrum and t is the time.

[5] The process of fourier transform has mainly two parameters, the first one is to find the sampling rate or sampling frequency  $f_s$ , which is the average number of samples in one second. The last one is to select the number of samples and it is always an integer (eg. 1024 samples). Moreover the bandwidth  $f_n$ , Nyquist frequency indicates the theoretical maximum frequency. The viewing of signal in time domain and frequency domain by using fourier is shown in Figure 5.

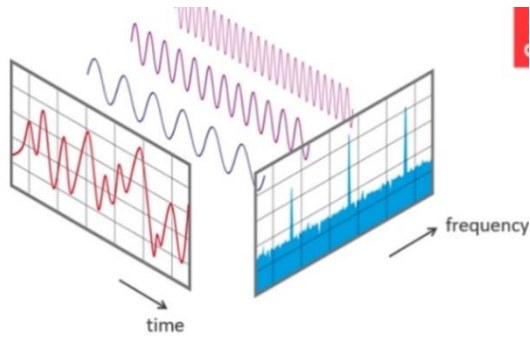


Figure 5. Signals in Time and Frequency domain [5]

#### 4.RESULT

This system is analyzed the audio signal by using low pass and high pass filters and then it is implemented by using Matlab Programming. The audio "Female Speech.wav" is used as a sample for analysis with the sampling rate of signal is 8000 Hz. The visual representation of the spectrum of frequencies of a signal is referred to as spectrogram and it is shown for the input audio signal in the following Figure 6.

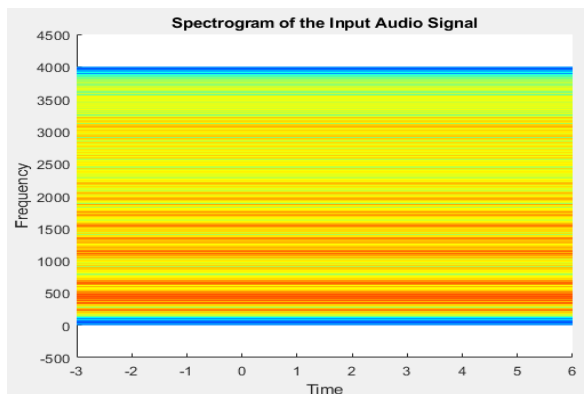


Figure 6. Spectrogram of Audio Signal

The periodogram is used to estimate the signal's power spectral density. It can examine between the amplitude and frequency characteristics of FIR filters and window functions. [wiki] The power density spectral of the audio signal is described in Figure 7.

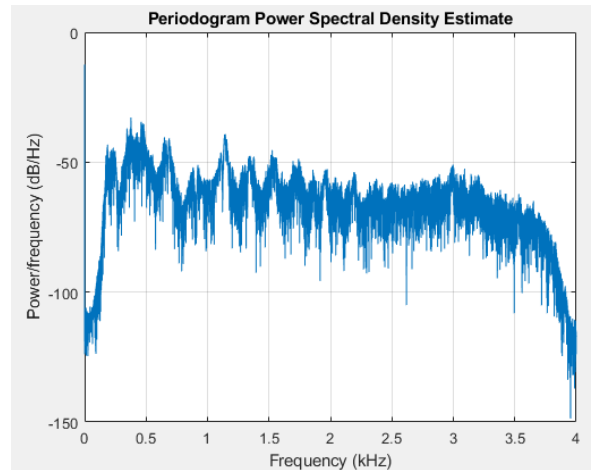


Figure 7. Periodogram Power Spectral Density

The audio signal is composed of multiple frequencies and the Fourier transform is the mathematical concept to decompose a signal according to their frequencies. It can convert the audio signal with time domain into frequency domain and the its result of input audio signal is shown in Figure 8.

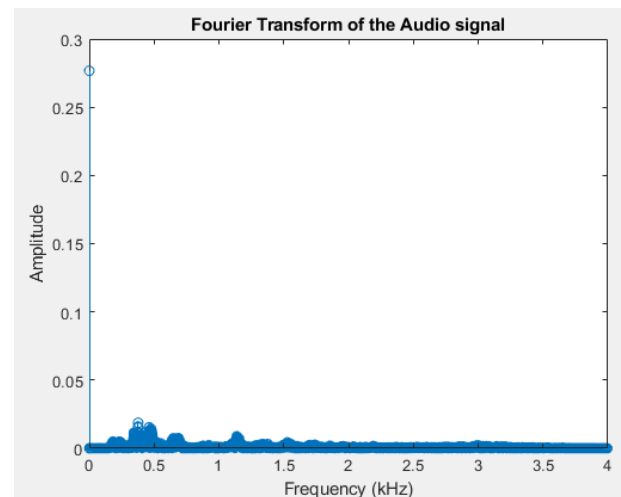


Figure 8. Decomposition of Audio Signal using Fourier transform

The bandwidth representation, the amount of information contained, of the input audio signal is described in the following Figure 9.

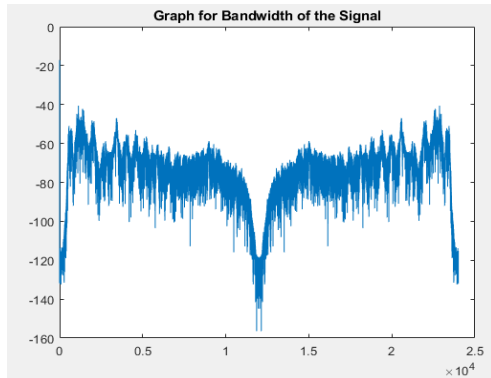


Figure 9. Representation Bandwidth of Audio Signal

The filtered audio signal by using the Low-pass filter is described in below Figure.

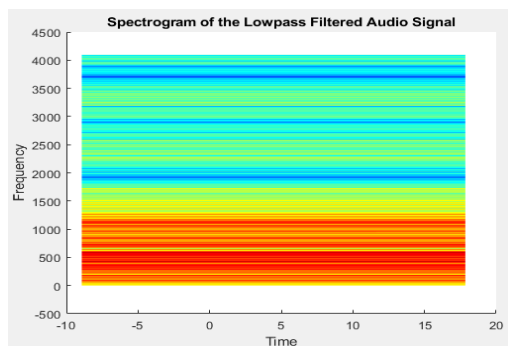


Figure 10. Filtered Audio Signal with Low-pass Filter

The lower frequencies are removed from input audio signal by using high-pass filter and then the results is shown with spectrogram in Figure 11.

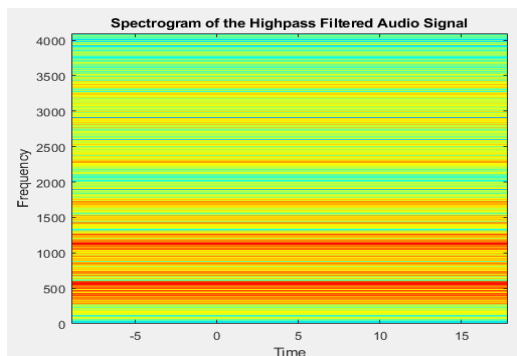


Figure 11. Filtered Audio Signal with High-pass Filter

## 5.CONCLUSION

The presece of background noise is the greatest challenge for hearing in audio and it becomes the source of dissatisfaction. The noise reduction can support the signal detection and classification system because it can test to determine the desirable signal or noise. If the lowest amplitude from audio is assumed to be noise and then each frequency band is calculated to separate and also to minimize the identified lowest amplitude noise. In this system, the unwanted background noise from female speech.wav audio file is reduced by using two filters high-pass and low-pass filter. Moreover, the audio signal classification can also extend from this research.

## REFERENCES

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