

Lab 9

Noise Removal for Audio Files *

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Introduction

Like any signal, audio recordings are very susceptible to noise and interference. This noise can come from many sources. Dust in the grooves on an LP record causes the needle to jump, creating the typical ticks and pops. Even if a record is dust free, imperfections in the cutting process leave the inside of the track rough, creating the characteristic hissing sound of an old record. Professional recording studios are designed to eliminate noise. Anechoic chambers are used to stifle reflected sounds and eliminate echoes. Pop screens over microphones reduce the noise of a persons breath when they speak from close range. Real world recordings encounter even more types of noise. Fluorescent lights can cause a hum that will barely noticeable to a person, but can still be picked up by a microphone and then amplified in the final output.

No matter what the source of the noise, it is often desirable to remove the noise. In this project you will be given an audio file that has been corrupted by noise. You are to indentify the signal, as well as any noise sources that are present. Then using the DSP board, clean the file and remove the noise.

Procedure

1. Open the corrupted audio file, noise_corrupted.wav using Windows Media Player and listen to it. Can you make out who the voice belongs to and what is being said? What other noises do you hear?
2. Using MATLAB, open the file, and determine its Power Spectral Density. From the Power Spectral Density attempt to locate and identify the base signal and any noise sources that are present. Consider the range of human hearing, as well as the typical range for voices. The following MATLAB commands may be useful in doing this: wavread, fft, fftshift. Be careful when plotting the PSD that the x scale is correct. List a description of any signals you detect.
3. Using MATLAB, design filters to remove the noise. What types of filters are needed to remove each noise source? Are IIR or FIR filters more appropriate for filtering audio signals? Why? Include plots of the magnitude, phase, and impulse response of each filter that you use.
4. Using the coefficients generated by MATLAB, implement your filters on the DSP board. The audio file can be input into the board by connecting the line out port on the computer to the line in port on the DSP board, and then playing the file using Windows Media Player. Listen to your filtered output using the headphone jack. Have your filters

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removed the noise? Can you make out what the original audio signal was? If so, what is it and who is speaking? If your output still contains significant amounts of noise, redesign your filters until the signal is clear. Can you remove all of the noise? Why or why not?

Example of Power Spectral Density Plot

