ECEN 444: Matlab Assignment 3

Due April 16, 2009

1 Overview

In this assignment, we will explore digital filtering as a way "clean-up" signals with additive periodic noise. Your job is to analyze the included wav file and design a digital filter that removes a number of sinusoids that are "jamming" the signal. Your basic tool be the discrete Fourier transform (DFT) and a digital notch filter. The idea is that you can use the DFT to estimate the frequencies of the interferers and a set of notch filters to remove them.

When working on the project, please follow the instructions and respond to each item listed. Your project grade is based on: (1) your Matlab script, (2) your report (plots, explanations, etc. as required), and (3) your final results. Several example Matlab script files have been posted on the website. The desktop version of Matlab also includes a script editor which highlights the syntax.

The project report should include all your scripts and the requested plots. It is often easier to combine these using Microsoft Word or Powerpoint. For example, you can copy/paste figures from MATLAB into these applications. You must clearly display the associated problem number and label the axes and on your plots to get full credit. Submission can be done electronically in PDF format or on paper.

MATLAB can be started by typing "matlab" at most workstations. Links to some MATLAB tutorials are located on the website and you can also look at:

http://www.mathworks.com/access/helpdesk/help/techdoc/matlab.html

You may also be able check out a copy of "The Student Edition of MATLAB User's Guide" or "The Matlab User's Guide". However, in most cases you should be able to get the explanation of the functions needed for the homework by typing "help [function name]" in the MATLAB command window. Or if you just have some idea of what you are looking for, you may type 'lookfor [keyword]'. It will list all functions related to the keyword.

2 Exercises

2.1 Finding and Filtering Tones

The best way to understand this type of problem it first create your own synthetic version of the problem and solve that. For example, will take the "speech0.wav" sound file from the previous lab, add a sinusoid, and then remove it.

(a) Use `[y,Fs,nbit] = wavread('speech0.wav')` to read the speech signal. Let $y(n)$ be the speech signal, $F_s =$`Fs` be the sampling rate, and $N =$`length(y)` be the length in samples. Play the sound with `soundsc(y,Fs)` to make sure this is working.

(b) Generate a $F = 1000$ Hz tone $x(n) = \frac{1}{2}\sin(2\pi f n)$ with $f = \frac{F}{F_s}$ for $n = 1, 2, \ldots, N$. Now, play this signal to make sure you can hear the tone.

(c) Add the two signals to get $z(n) = y(n) + x(n)$. Compute the energy P_x, P_y in each signal and the signal-to-noise ratio (SNR) $\frac{P_y}{P_x}$. Play this sound and notice that (even as this SNR) your ear is still good at identifying the speech.

(d) Plot the first 200 samples of $z(n)$ to see the interfering signal.

(e) Compute the DFT $Z(k)$ of $z(n)$ with `Z=fft(z)`. The frequency associated with $Z(k)$ is $\frac{k}{N}$ cycles/sample for $k = 0, 1, \ldots, \frac{N}{2}$ where $N =$`length(z)`. Converting to Hertz gives a frequency of $\frac{kF_s}{N}$ Hz. Plot the magnitude of the DFT versus frequency and use this plot to estimate the frequency of the interferer.

(f) Let $\omega_0 =$`w0` be the normalized angular frequency of the interferer. Design a notch filter for this frequency. From Eqn. 5.4.31, the non-recursive filter coefficients are given by `b=[1,-2*cos(w0),1]` and the recursive filter coefficients are given by `a=[1,-2*r*cos(w0),r*r]` where $r = 0.95$ is a good choice.

(g) Use `freqz(b,a)` to plot the frequency and phase response of this filter.

(h) Use `zfilt=filter(b,a,z)` to compute the filtered version of this signal. Play this sound to verify the tone has been removed.

(i) Plot the first 200 samples of this signal to see the transient response of your filter.

2.2 Unknown Signal and Interference

Use `[y,Fs,nbit] = wavread('lab3a.wav')` to read the unknown signal. This unknown signal is obscured by additive noise and sinusoidal interference. Design a notch filter to cancel out each sinusoidal interferer (up to 5). Then, use your knowledge of pop culture to identify the signal. Please notice that the sample rate has changed from the previous example.

Extra Credit

The same signal is corrupted by a new periodic interferer and resulting file is `'lab3b.wav'`. Can you cancel this interference using a digital filter that requires at most two multiply-add operations per output sample?