ECE 5630: Digital Signal and Image Processing - The Fast Fourier Transform (FFT)

Due: December 5, 2014

Name (Print):

Objectives

This programming assignment has two objectives:

1. Give you practice in using a program to perform digital signal processing.
2. Help to solidify your understanding of methods for the fast computation of the DFT.

Instructions

1. The program should be written in C or C++.
2. Use the `.wav` file `galway11_mono_45sec.wav` to test your program. This file is available on Canvas and is sampled at 11.025 kHz.
3. Attach your work, answers, plots, and code to this page with a single staple in the top left-hand corner.

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1. Create the signal flow-graph for the butterfly for a decimation-in-time **radix-6** fast Fourier transform (FFT). (Only one stage.)
2. In C or C++, write a function that performs the decimation-in-time **radix-6** fast Fourier transform (FFT).
   (a) What are the number of multiplies and adds required to perform a 1296-point DFT? What about a radix-6 FFT?
   (b) Verify that your FFT works as expected by computing the FFT of 1296 points of a signal created by adding together sinusoids of frequencies at $f = 17.01$ Hz, $f = 297.74$ Hz, $f = 425.35$ Hz, and $f = 2637$ Hz. Use a sample rate of 11.025 kHz to create the test sinusoids.
   (c) Plot the magnitude of the FFT output. Which bins have values larger than the others? (Remember that there may be some computation noise in each bin.)
3. Use the Matlab function `wavread()` to generate the samples of the file `galway11_mono_45sec.wav`. Use your FFT from 1. above, and your frequency-domain fast convolution program and filter from Programming Assignment 2, to filter the sound file. Use a FFT length of 1296 points. The results should be the same as for the last programming assignment. Does the filter remove the high frequency components? Does the processed file sound as you expected? Write out the final results in a .wav file for the instructor to listen to.