EE3TR4 Lab 1: Fourier Analysis

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Room: lab- ITC 157
Please work in pairs. This lab starts on Mon Jan 28, 2013. The final report (one per pair) is due Monday Feb. 25, 2013 in the EE3TR4 assignment slot in Rm ITB A110 by 4:00pm. ***Please note that each group is expected to work independently*.** Please be familiar with the guidelines on academic integrity, summarized on the course outline available on the web. The late penalty is 10% per day.

This lab is project based, consisting of two parts. The first part involves Fourier analysis. Here, you calculate the output of a filter of your choice due to an input of your choice. The second part is less intensive and involves listening to various forms of filtered speech, to give you an idea what the effect of filtering operations are on audio signals.

Part I: Filter Analysis

This is an open-ended project. Design and implement a filter of your choice, and apply an input waveform of your choice. Calculate the corresponding output waveform. Compare your calculated waveforms with the measurements you obtain in the lab. Investigate the effect of varying the values of parameters such as cutoff frequency of the filter and the frequency of the input signal.

Write a report that summarizes the results of your investigations. View your report as if you were working in industry and were submitting your design recommendations for a particular system to your boss. Your grade for this lab will be determined on the basis of the quality and quantity of your report relative to that of your peers. We expect a report that is clear, articulate and accurate.

Here is a list of various components you will have available in the lab:

Op Amps
0.001, 0.01 and 0.1 uF capacitors
Resistors: 470, 560, 680, 820, 1K, 1.2K, 1.8K, 2.0K, 2.4K, 3.3K, 3.9K, 4.7K, 5.6K, 6.2K, 8.2K, 10K.

Hints and Suggestions:

1. You can calculate the output response $y\ t$ of the filter in at least two different ways-- in the frequency domain and in the time domain. For the frequency domain, we have $Y\ nf_0\ = X\ nf_0\ H\ nf_0$, where f_0 is the fundamental frequency of the input. The spectrum $X\ nf_0$ is known. The filter frequency response $H(f)$ can be calculated

knowing the transfer function of the filter, which is available from the supplementary material for this lab. Thus, $Y(nf_0)$ can be determined. The corresponding time-domain signal $y(t)$ can be found by inverting the Fourier transform quantity $Y(f)$, as we discussed in class. Note that the transfer functions of the filters shown in the supplementary material are normalized to 1 radian/sec. To achieve a cutoff frequency of ω_c radians/sec., replace ω with the quantity $\frac{\omega}{\omega_c}$.

The output can be determined in the time domain as follows. We know that $y(t) = h(t) \circledast x(t)$, where $\circledast$ denotes convolution. The input signal $x(t)$ is known. The impulse response $h(t)$ can be determined as the inverse FFT of the frequency response $H(f)$. This can easily be done using the matlab command "ifft". There are a few tricks to be aware of in this operation, so be careful. See point 8 below.

2. Design your filter circuit using the components available before you go into the lab.

3. The spectrum of any signal can be measured with the scopes, using the FFT function. A scope manual is available as an icon on the desktop of the lab computers.

4. The Butterworth lowpass active filter configuration is probably the most straightforward. You can use a 1st-, 2nd-, 3rd- or 4th-order filter, as you see fit. See the material on active filter design that has been posted on the website.

5. Make sure your cutoff frequency is a reasonable choice, relative to the frequency of your input signal.

6. You can measure the frequency response of your filter if you wish, by applying an input sinusoid at varying frequencies, and measuring the corresponding relative amplitude and phase of the output.
7. Try varying the frequency of the input wave. Comment on its effect on the output waveform.

8. To evaluate the impulse response $h(nT_s)$ of your filter, assemble a vector in Matlab of values of $H(j2\pi f)$ according to the format of the Discrete Fourier Transform, as discussed in class. Make sure the sampling frequency f_s is large enough so that $\left|H\left(\frac{f_s}{2}\right)\right|$ is small (e.g. 0.01) in comparison to $|H(0)|$ to avoid aliasing error in frequency, and that N (the number of samples) is large enough so that $h(NT_s)$ is small compared to its largest magnitude, to avoid aliasing in time.
9. A 2nd-order or larger filter is more interesting than a first order.

Part 2: Speech Filtering

This should be a bit more fun than part I. On the computers in the lab, run the matlab program "filtDemo". It plays speech that has been passed through various forms of filter. Plots of the input and output spectra are shown on the displays. Note especially how the notch filter can suppress the sinusoidal tone. Pay attention to the difference in sound of the outputs of each type of filter.