# EE3TR4 Lab 1: Fourier Analysis

Jim Reilly

ITB A312
reillyj@.mcmaster.ca

Last modified: Jan 19, 2015
**Room:** lab- ITC 157
Please work in pairs. This lab starts on Mon Jan 25, 2015. The final report (one per pair) is due Monday Feb. 23, 2015 in the EE3TR4 assignment slot in Rm ITB A110 by 4:00pm. ***Please note that each group is expected to work independently.*** The late penalty is 10% per day.

This lab is project based, consisting of two parts. The first part involves Fourier series analysis. Here, you design a filter to convert an input periodic waveform of your choice into an output sinusoid. 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 design project, and hence details are purposely left vague. The objective is to design a filter that will convert a periodic input signal into an output sinusoid at the fundamental frequency of the input. Design your filter so that all unwanted harmonics are suppressed by at least 30 dB relative to the desired fundamental component.

Write a report that summarizes your design process in some detail. In your results section, demonstrate the successful operation of your filter and include a comparison of the actual output spectrum you obtained (using the FFT function on the scopes) vs. your calculated values.

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 might want to design your filter circuit using the components available before you go into the lab. The Butterworth lowpass active filter configuration is probably the most straightforward. See the material on active filter design that has been posted on the website.

2. Please do NOT use a $4^{th}$ –order lowpass Butterworth response with a cutoff of 1 KHz. The reason is that this is precisely the example given in the active filter design material.

3. Don’t worry about any DC component in the output signal. It can easily be blocked using a series capacitor.

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

5. Make sure the frequencies you use are within the capabilities of the op amps.

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.

**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. No write up is required for this portion.