

EE3TR4 Lab 1

Fourier Analysis

Please work in pairs. ***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.

This lab is project based, consisting of two parts. The first part involves passing a square wave through a low pass filter and comparing your calculations of the output signal waveform with what you actually observe in the lab. The second part involves listening to various forms of filtered speech. The idea is to try to figure out why the filtered speech sounds as it does.

Part I: Filter Analysis

The objective of this lab is to compare theoretical predictions and experimental measurements of the output of a filter, driven by a function generator outputting a square wave at 500 Hz, as shown in Figure 1. This comparison is in both the time and frequency domains. The filter is a second-order low-pass Butterworth filter with a cutoff frequency of 1.591 KHz. Its circuit diagram is shown in Figure 2.

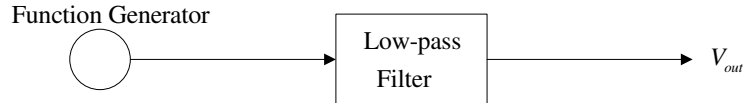


Figure 1 Configuration of filtering operation

The transfer function $H(\omega)$ for the second-order Butterworth low-pass filter is given as

$$H(s) = \frac{1}{\left(\frac{s}{\omega_0}\right)^2 + 1.414 \left(\frac{s}{\omega_0}\right) + 1} \quad (1)$$

where $s = j\omega$, f_0 is the cutoff frequency in Hertz, equal to 1.591 KHz for the circuit of Figure 2, and $\omega_0 = 2\pi f_0$.

1a. Procedure for Analyzing the Output Waveform in the Time Domain

Here we use Fourier analysis as we discussed in class. Given that the output spectrum $Y(\omega) = H(\omega)X(\omega)$, where $Y(\omega)$ is the output spectrum, $H(\omega)$ is the frequency response of the filter (given by Eq. (1)), and $X(\omega)$ is the spectrum of the input (a discrete sinc function in frequency), calculate the output waveform in MATLAB using the following description:

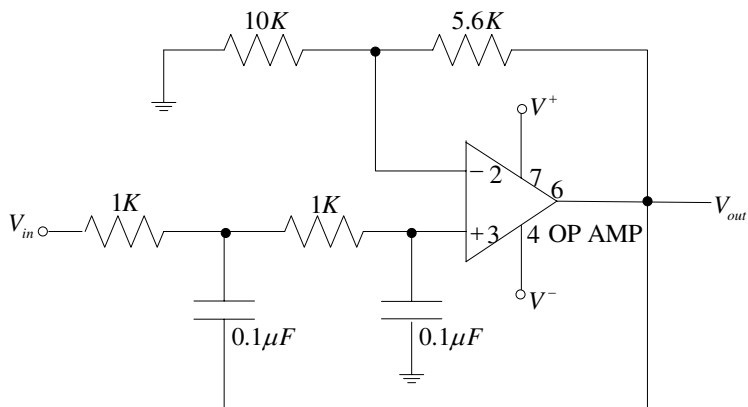


Figure 2 2nd-order low-pass Butterworth filter, cutoff=1.591 KHz, $\omega_0 = 2\pi f_0$

1. For each frequency component at $\omega_k = 2\pi f_k$, $k = 1, \dots, K$ contained in the input signal,
 - (a) Evaluate the complex frequency response $H(\omega_k)$ of the filter using Eq. (1).
 - (b) Evaluate the corresponding frequency component $Y(\omega_k)$ of the output.
2. Given the set of Fourier coefficients of the output obtained above, compute the output waveform in time using the appropriate Fourier series formula. Plot two or three periods of the corresponding output waveform $y(t)$.
3. Measure and record the actual output waveform in the lab.
4. Compare the calculated output waveform with that measured in the lab.

Hint: Choose K so that $|Y(\omega_K)|$ is very small in magnitude compared to the values at low frequencies.

1b. Procedure for Analyzing the Output Waveform in the Frequency Domain

The scopes available in the lab have a frequency-domain (FFT) display, which shows only the magnitude spectrum of the signal applied to the input of the instrument. The file “fft_operations.pdf” (on the website) explains how to use this display. A few tips on the settings to use are included in the appendix of this document. Note that with these scopes, you can dump the screen to a PC-compatible floppy disk. **Therefore, you should bring a formatted floppy with you to the lab.** Information on saving the screen to a floppy is given in the file “saving_to_floppy.pdf” (also available on the website).

The idea of this section is to compare the calculated spectrum of the output signal (i.e., the values of the magnitudes of the Fourier coefficients obtained above) with the values

obtained by FFT measurement on the scope. Plot the calculated magnitudes vs. frequency, and compare with the plot obtained from the scope. As an optional exercise, try varying the input frequency of the square wave. Try to predict what happens at the output. Also, consider how you would measure the frequency response of the filter by replacing the square wave input signal with a sinusoid with a variable frequency.

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. Why do the outputs sound different from the original input speech?

REPORT:

For your report, present a brief writeup that shows you understand the underlying theory. Explain the use of any formulas you may use. Include graphs showing the time domain and the frequency domain representations from parts 1a and 1b. Make sure the axes of the graphs are properly labelled. Give an explanation of any discrepancies between the theoretical and experimental versions. Include your MATLAB program and give a very brief description of how it works.

For part 2, explain what causes the characteristic sound of the speech from each type of filter. Give a brief explanation of how the filtering process that generates the output speech was performed.

APPENDIX – Tips on Using the FFT facility on the Scopes

Set the “window” soft key to “Hanning”.

The frequency scan need not be greater than 10 KHz. This will show the first 20 harmonics of the signal.

The frequency domain display will appear as a sequence of spikes. Measure their frequency and their corresponding amplitude values (note the amplitude scale is in dB).

When measuring the signal at the output of the low-pass filter, note that the higher-frequency components of the signal are significantly suppressed at the output, relative to the input, by the low-pass filter. This is what causes the output waveform to be significantly different from that at the input.