CAD Laboratory Report

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Abstract:

The purpose of this laboratory is to design a filter with specific attenuation at designated frequencies. Based on what I have learnt from the CAD lessons, I had a deep study about configurations and functions of many filters by myself, and designed a second-order active Butterworth low-pass filter to achieve all requirements in this laboratory successfully.

Introduction:

As I was supposed to design a filter which can let pass the signal with a useful bandwidth between 0Hz and 1 Hz but attenuate the undesired 10kHz harmonic by at least 40 dB. I studied a huge variety of filters to find out the most appropriate filter design — Butterworth low-pass filter. With cutoff frequency, attenuation, tolerance and ripple parameter obtained from the designing requirement, I found the order, the poles, the zeros, and the transfer function, and selected the component values by using the Sallen-Key equal R, equal C method and applying magnitude scaling and frequency scaling. Once all the component values are selected, I implemented my filter using corresponding circuits, and did both simulations in LTspice and demonstrations on breadboard to verify the design successfully.

Methodology:

Overall, my Butterworth LPF design procedure can be broken down to several steps as below.

1. Analyzed and obtained basic data from the lab problem:

$f_p = 1kHz$ $\quad f_s = 10kHz$

$\omega_p = 2\pi f_p = 2\pi krad/s$ $\quad \omega_s = 2\pi f_s = 20\pi k\ rad/s$

$G_p = \frac{1}{\sqrt{2}}$ $\quad G_s = 0.01$

$A_p = -20log_{10}G_p = 3dB$ $\quad A_s = 40dB$

$\delta_p = 1 - \frac{1}{\sqrt{2}}$ $\quad \delta_s = 0.01$

2. Changed cutoff frequencies from Hertz to Radians:

$\omega_p = 2\pi f_p = 2\pi k\ rad/s$ $\quad \omega_s = 2\pi f_s = 20\pi k\ rad/s$

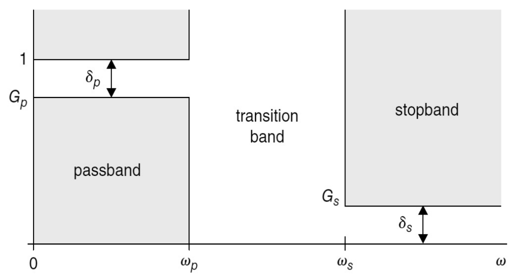
3. Found the backward transformation:

$\Omega_p = 1\ rad/s$ $\quad \Omega_s = \frac{\omega_s}{\omega_p} = 10\ rad/s$

4. Calculated and set the order of the filter:

$\omega_p = 2\pi f_p$ = passband cutoff frequency in rad/s.
$\omega_s = 2\pi f_s$ = stopband cutoff frequency in rad/s.
G_p = minimum gain in the passband. Maximum gain in the passband is 1.
G_s = maximum gain in the stopband. Minimum gain in the stopband is 0.
δ_p = passband tolerance.
δ_s = stopband tolerance.
A_p = maximum attenuation in decibels in the passband.
A_s = minimum attenuation in decibels in the stopband.

Figure1. Specification of Low-pass Filter in Gain



$$n = \text{ceil}\left[\frac{log\left(\frac{10^{0.1A_s}-1}{10^{0.1A_p}-1}\right)}{2log\left(\frac{\Omega_s}{\Omega_p}\right)}\right] = 2$$

Since n is even, and $\frac{\text{n}}{2} = 1$, which means the order and the number of the filter should be second and one, respectively.

5. Found low-pass poles and zeros: The pole locations are -0.7071±j0.7071

6. Found frequency transformation: $\omega_0 = \Omega_\text{p}\omega_0 = 2\pi\text{kHz}$

7. Found transfer function: For the second-order filter, $\text{H(S)} = \frac{1}{\text{S}^2+\frac{1}{\text{Q}}\text{S}+1} = \frac{1}{\text{S}^2+0.707\text{S}+1}$

8. Normalized component values:

$R_2 = 1\Omega$ $R_3 = 1\Omega$ $R_4 = 0.586\Omega$ $R_1 = 1.586\Omega$ $R_5 = 2.707\Omega$

$C_1 = 1F$ $C_2 = 1F$

9. Implemented using the Sallen-Key equal R, equal C method.

10. Scaled component values: Since $k_f = \omega_0 = \Omega_\text{p}\omega_0 = 2\pi\text{kHz}$, and assume that $k_m = 1000$.

$R_1 = 1000\Omega$ $R_3 = 1000\Omega$ $R_4 = 586\Omega$ $R_1 = 1586\Omega$ $R_5 = 2707\Omega$

$C_1 = \frac{1}{k_m k_f} = \frac{1}{2\pi\times10^6} = 0.159154943\text{F}$ $C_2 = \frac{1}{k_m k_f} = \frac{1}{2\pi\times10^6} = 0.159154943\text{F}$

11. Simulated in LTspice and experimented on breadboard.

The simulated and experimental results will be shown and analyzed later.

Results and Discussion:

I did both simulations in LTspice and demonstrations on breadboard.

1.Simulated Results: simulations in LTspice and captured corresponding screenshot as below.

Figure 2. Simulated Circuit Schematic

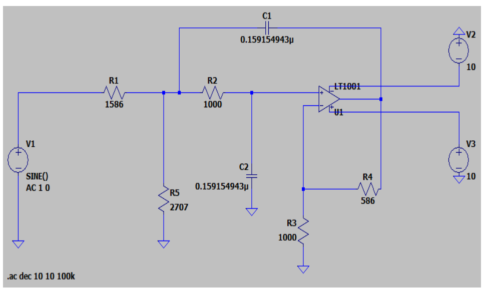
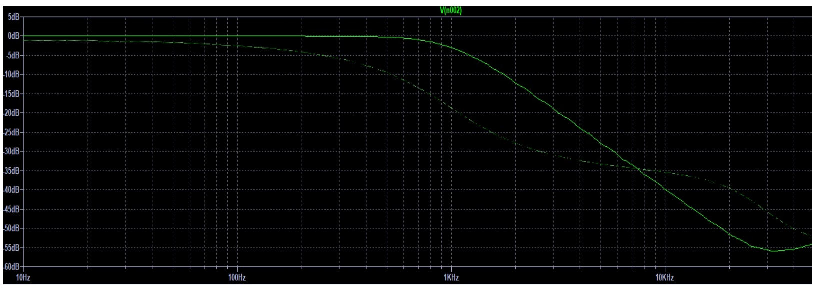


Figure 3. Simulated Result in LTspice



The key points can be observed and analyzed from this simulation:

①The values of Gain in decibel from 0Hz to 1kHz are very close to 0dB, which proves that my filter let pass the useful bandwidth(0Hz~1kHz) of the signal successfully;

②The value of Gain in decibel at 1kHz is -3dB, it looks like the outcome does not meet the criteria of success, but that is not the case. In fact, using Laplace transformation and Fourier transform can derive that the filter let pass without attenuating is quoted in terms of the -3dB from the maximum gain at its passband cutoff frequency. In this laboratory, since the filter let pass without attenuating between 0 and 1kHz, it can be obtained that the frequency at 1kHz the gain will be -3dB, which is also demonstrated in the simulation Figure 3);

③The values of Gain in decibel between 1kHz and 10kHz keep decreasing because this undesired bandwidth is supposed to be attenuated by my filter;

④The value of Gain in decibel at 10kHz is -40dB, which means my filter achieve attenuating the 10kHz harmonic by at least 40 dB required in task.

2.Experimental Result: I did experiment on breadboard and recorded typical data to plot as below:

Figure 4. Experimental Circuit on Breadboard

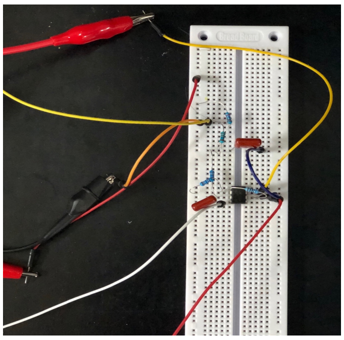
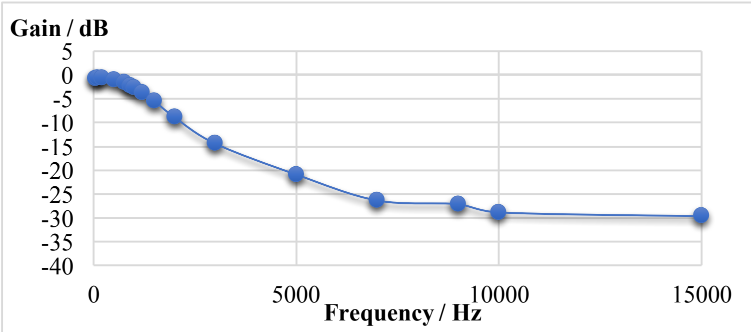


Figure 5. Experimental Result Plot



Because of some instrumental errors and personal errors in practical experiment, for example, I had to choose capacities which I could get to substitute the capacities theoretical values. But it does not influence me to observe and analyze the overall trend and some key data from the plot. It true that the values of Gain in decibel from 0Hz to 1kHz are very close to 0dB and then decrease with frequency becoming larger. And the value of Gain in decibel at 10kHz is about -30dB, notwithstanding there exists a difference between it and theoretical value(-40dB), the difference is acceptably and relatively small. Combining with the simulated result, it can be found that the experimental result I obtained fit the theory and simulation.

Conclusions:

Following the Butterworth LPF design procedure, including finding the transformation, order, normalized low-pass poles, transfer function, component values, and magnitude frequency scaling, I implemented a second-order active Butterworth low-pass filter. With simulations in LTspice and demonstrations on breadboard, the filter I designed let pass the signal with a useful bandwidth between 0Hz and 1 Hz but attenuate the undesired 10kHz harmonic by at least 40 dB successfully.

Reference:

James S.Kang, Electric Circuits, California State Polytechnic University, Pomona, 2018