<u>Designing Passive Low Pass Butterworth Audio Filters</u>

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After building a portable HF station, I wanted to build a low pass audio filter when using SSB with the rig. One has basically three choices, an active filter built with op amps, a switched capacitor filter, or a passive filter. After considering all three I decided to build a passive filter as it is the simplest approach. A bit of reading convinced me that the best type of low pass (LP) filter for audio frequencies is the Butterworth design. This is because the Butterworth filter minimizes ripple in the passband, thus giving somewhat better fidelity than other types.

It is quite easy to design your own filter. As a first example, we will design a third order LP filter with a cutoff of 1000 Hz. I built such a filter for installation in a small QRP CW rig. For a three pole filter we need three components in one of two configurations, as shown in Figure 1.

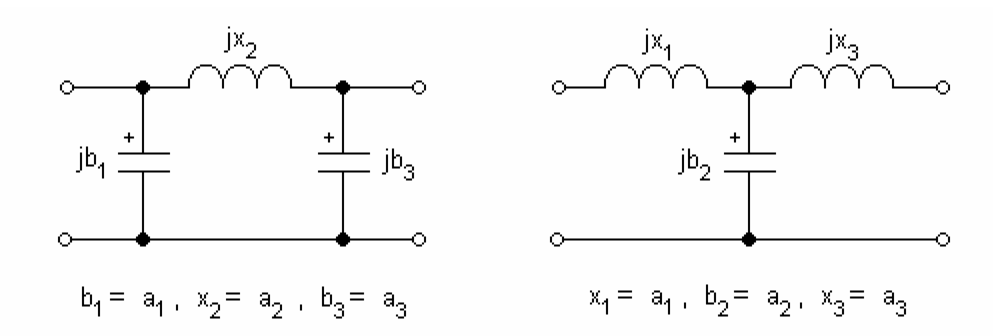


Figure 1. Two 3-pole LP filter configurations

The values a_i are from a filter table, or are calculated by the formula

$$a_i = 2\sin\left(\frac{(2i-1)p}{2n}\right)$$

where n is the total number of poles. For a three pole filter, the a_i values are simple, namely, $a_1 = 1$, $a_2 = 2$, $a_3 = 1$. The x values in the figure are called “normalized reactances” and the b values are called “normalized susceptances.” These will be converted into actual component values after we specify two more quantities, the cutoff frequency f_c and the filter (input and output) impedance Z_0.

For our first example, we will choose a cutoff frequency of 1000 Hz. This is the frequency where the output falls off 3 decibels. Since the filter is for simple insertion in an audio circuit just ahead of headphones, we will choose the impedance to be 16 ohms. The component values are now given by the formulas:

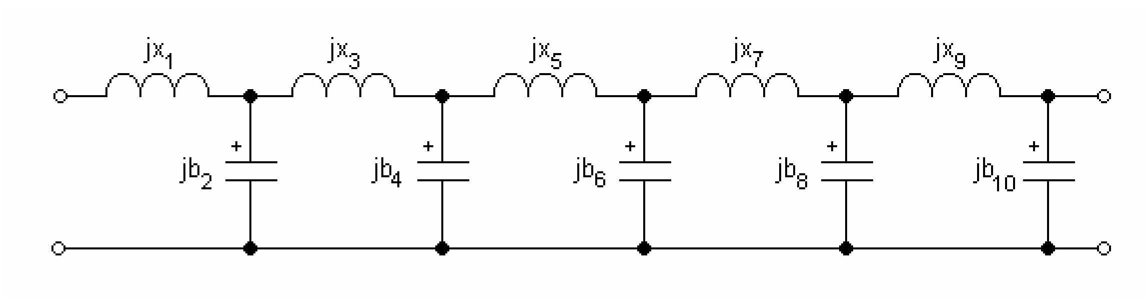
$$C_1 = \frac{a_1}{Z_0 \; 2p \;\; f_c} = \frac{1}{16 \cdot 2p \cdot 1000} \cong 10 \; m \; f$$

$$L_2 = \frac{a_2 \; Z_0}{2p \;\; f_c} = \frac{2 \cdot 16}{2p \cdot 1000} \cong 5.1 \, mH$$

$$C_3 = C_1$$

We are fortunate that the capacitor values came out very close to a standard value. The inductor value is not standard, so we will either substitute the closest standard value, or put standard values in series to better approximate the desired value. And that’s it, we are done with this filter.

A ten-pole Butterworth LP filter follows the same formulas, but there are now ten component values to calculate. Figure 2 shows the configuration I chose.



Since this filter is for SSB, we are going to pick a cutoff frequency of 3000 Hz. Also, to keep capacitor values within reason, we will pick a filter impedance of 200 ohms. This means that we will need an audio 8 ohm to 200 ohm audio transformer at each end (for example Mouser Part No. 42TL004-RC). We will use metallized polypropylene film capacitors because of their superior stability and radial enclosed RF chokes for the inductors (Xicon brand for both).

The filter table values calculated from the equation for a_i above are:

$$a_1 = a_{10} = 0.312869, \quad a_2 = a_9 = 0.907981, \quad a_3 = a_8 = 1.414214$$
$$a_4 = a_7 = 1.782013, \quad a_5 = a_6 = 1.975377$$

We use the same equations as above to convert these normalized values into actual component values. For example, the first two components are given by

$$L_1 = \frac{a_1\, Z_0}{2p\ \ f_c} = \frac{0.31287 \cdot 200}{2p \cdot 3000} \cong 3.32\ mH$$

$$C_2 = \frac{a_2}{Z_0\ 2p\ \ f_c} = \frac{0.90798}{200 \cdot 2p \cdot 3000} \cong 0.241\, m\ f$$

For the first inductor we will put 3.3 mH and 0.022 mH standard inductors in <u>series</u>. For the capacitor we will put 0.22 and 0.022 μf capacitors in <u>parallel</u>. You can calculate the remaining components using these formulas with the appropriate a_i values.

The unit I built was done “ugly style” on a single-sided piece of printed circuit board. The audio transformers were hot-glued upside down at opposite ends of the board, and the inductors and capacitors were soldered in between, all also upside down. For maximum flexibility, I installed both mono and stereo 3.5mm jacks for the input and the output. Everything was put in a cast aluminum case for ruggedness, with the PCB held to the bottom with double sided foam.

The SSB filter does seem to improve intelligibility of SSB signals, but the effect is somewhat subtle. Don’t expect a night and day difference. I think this filter will reduce fatigue during long periods of listening to 75m nets.