CHAPTER 6

Active Filter Design Techniques

6.1 Introduction

What is a filter? According to Webster, it is “a device that passes electric signals at certain frequencies or frequency ranges while preventing the passage of others”.

Filter circuits are used in a wide variety of applications. In the field of telecommunications, bandpass filters are used in the audio frequency range (20 Hz to 20 kHz) for modems and speech processing. High-frequency bandpass filters (several hundred megahertz) are used for channel selection in telephone central offices. Data acquisition systems usually require antialiasing low-pass filters as well as low-pass noise filters in their preceding signal conditioning stages. System power supplies often use band-rejection filters to suppress the 60 Hz line-frequency and high-frequency transients.

Entire books have been written on the subject of filter design. As a young engineer at the beginning of my career, I purchased several of them from a nearby university book store. I was completely dissatisfied with them from the start: they took a mathematical/theoretical approach, and therefore were cumbersome to use when actually designing a filter. What I needed was a fast, practical design approach. After many years in this profession, I wrote my own book, one I could finally be satisfied with. I have managed to come up with manageable approaches to different types of filters, and will present them here. I have selected topologies for each type of filter that will minimize the number of op amps used.

In over 30 years as an engineer, I have never encountered a set of design requirements that said to use more op amps than are necessary, consume as much power as you deem necessary, take up extra board space with more complex circuit topologies, and decrease reliability by using more parts. Therefore, this chapter will not present topologies you will see in other books such as biquad filters, for the reason that they use three op amps when a single op amp can do the same job. I will give a slight nod to the biquad filter for flexibility and the ability to output more than one response at a time. But with the possible exception of test equipment, this advantage comes at too high a cost.

Op Amps for Everyone.

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I will first discuss different methods of filter design, and why I think they are difficult and cumbersome. After that, I will discuss my method. At the end of the chapter, as I did for gain and offset, I will give design aids and printed circuit board (PCB) layouts. Some semiconductor manufacturers have posted free filter design utilities that I will discuss in Chapter 12.

6.2 The Transfer Equation Method

The transfer equation method of designing active filters is a mathematical approach that has been championed in textbooks for decades. I certainly do not belittle it: if you are comfortable in the mathematical arena, these methods can produce a very satisfactory result. The transfer equation method is also the most general method: it will work for any filter topology and therefore is ideal for esoteric cases. This book will not use the transfer equation method for reasons I will state below.

I have condensed a single case of filter design using the transfer equation method from Thomas Kugelstadt's excellent write-up in *Op Amps for Everyone*, the ancestral work to this volume. It remains one of the most concise and easily understood treatises on the subject of transfer equation filter design ever written, and I highly recommend it for proponents of this filter design method. Without further introduction, here is an example of a transfer equation design of a two-pole, unity-gain Sallen—Key low-pass filter.

Equation 6.1 represents a general second order low-pass filter. The transfer function of a single stage is:

$$A_i(S) = \frac{A_0}{(1 + a_i S + b_i S^2)} \tag{6.1}$$

The unity-gain Sallen—Key topology in Figure 6.1 is usually applied in filter designs with high gain accuracy, unity gain, and low Q ($Q < 3$).

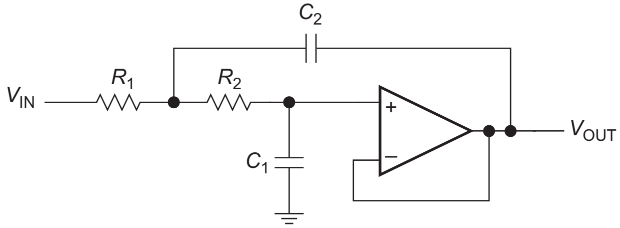

Figure 6.1
Unity-Gain Sallen—Key Low-Pass Filter

The transfer function of the circuit in Figure 6.1 is:

$$A(S) = \frac{1}{1 + \omega_c C_1 (R_1 + R_2) S + \omega_c{}^2 R_1 R_2 C_1 C_2 S^2} \tag{6.2}$$

The coefficient comparison between this transfer function and Equation 6.2 yields:

$$\begin{aligned} A_0 &= 1 \\ a_1 &= \omega_c C_1 (R_1 + R_2) \\ b_1 &= \omega_c{}^2 R_1 R_2 C_1 C_2 \end{aligned} \tag{6.3}$$

Given C_1 and C_2, the resistor values for R_1 and R_2 are calculated through:

$$R_{1,2} = \frac{a_1 C_2 \mp \sqrt{a_1{}^2 C_2{}^2 - 4 b_1 C_1 C_2}}{4\pi f_c C_1 C_2} \tag{6.4}$$

In order to obtain real values under the square root, C_2 must satisfy the following condition:

$$C_2 \geq C_1 \frac{4 b_1}{a_1{}^2} \tag{6.5}$$

Example: Second Order Unity-Gain Tschebyscheff Low-Pass Filter

The task is to design a second order unity-gain Tschebyscheff low-pass filter with a corner frequency of $F_C = 3$ kHz and a 3 dB passband ripple.

The coefficients a_1 and b_1 are obtained from Table 6.1:

$$\begin{aligned} a_1 &= 1.0650 \\ b_1 &= 1.9305 \end{aligned} \tag{6.6}$$

Specifying C_1 as 22 nF yields a C_2 of:

$$C_2 \geq C_1 \frac{4 b_1}{a_1{}^2} = 22.10^{-9}\text{nF} \cdot \frac{4 \cdot 1.9305}{1.065^2} \cong 150 \text{ nF} \tag{6.7}$$

Table 6.1: Second Order Filter Coefficients

Second Order	Bessel	Butterworth	3 dB Tschebyscheff
a_1	1.3617	1.4142	1.065
b_1	0.618	1	1.9305
Q	0.58	0.71	1.3
R_4/R_3	0.268	0.568	0.234

Inserting a_1 and b_1 into the resistor equation for $R_{1,2}$ results in:

$$R_1 = \frac{1.065 \cdot 150 \cdot 10^{-9} - \sqrt{(1.065 \cdot 150 \cdot 10^{-9})^2 - 4 \cdot 1.9305 \cdot 22 \cdot 10^{-9} \cdot 150 \cdot 10^{-9}}}{4\pi \cdot 3 \cdot 10^3 \cdot 22 \cdot 10^{-9} \cdot 150 \cdot 10^{-9}} = 1.26k\Omega \tag{6.8}$$

and

$$R_2 = \frac{1.065 \cdot 150 \cdot 10^{-9} - \sqrt{(1.065 \cdot 150 \cdot 10^{-9})^2 - 4 \cdot 1.9305 \cdot 22 \cdot 10^{-9} \cdot 150 \cdot 10^{-9}}}{4\pi \cdot 3 \cdot 10^3 \cdot 22 \cdot 10^{-9} \cdot 150 \cdot 10^{-9}} = 1.30k\Omega \tag{6.9}$$

with the final circuit shown in Figure 6.2.

So, why not just proceed to use this method for filter design?

- The transfer equation method works well for simple filters such as the one above. As filters become more complex, with more poles, and different topologies for example, the math quickly becomes unwieldy. Since filter designers are not necessarily mathematicians, forcing them to derive transfer equations for each circuit is not an efficient means of design. Transfer equation derivation for filter design easily fills an entire book, and is beyond the scope of this volume.
- If you have the transfer equation of a given filter topology, you can use Mathcad or Matlab to generate a filter design solution. Transfer equations exist for common filter topologies, and this may be preferable to you. However, transfer equations are not always available for more exotic filter topologies or multiple pole filters.
- The transfer equation method does not result in real-world component values. There are too many textbooks on the subject that give precise values of

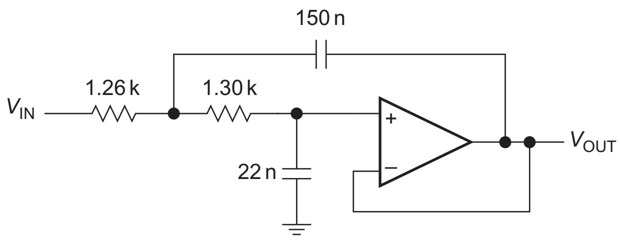

Figure 6.2
Second Order Unity-Gain Tschebyscheff Low-Pass with 3 dB Ripple

capacitors and resistors, and then leave you with no guidance about how to pick real values, in particular how to decide the scale of the components.

- The example above relied on a look-up table for filter coefficients. If you are going to use a table anyway, why not use a table for everything? There are textbooks that give filter designs in terms of R and C values on a graph of frequency and Q, and these yield fairly good designs quickly.

6.3 Fast, Practical Filter Design

The previous section introduced a rigorous theoretical approach to filter design. Many designers prefer this method because it gives the most flexibility in filter design. Years of customer support, however, have revealed that the vast majority of designers want a simple filter with the minimum of effort and development time. This section presents a pragmatic, simple design methodology that will allow you to implement all but the most complex filters rapidly, and have a reasonable expectation that they will be producible.

Some compromises have been made in order to keep this quick filter design process simple:

- A single op amp topology has been chosen for each filter type. I have picked the simplest (least number of op amps) approach. I do not cover multiple op amp topology such as biquad.
- Filter circuit responses are all Butterworth. I do not cover Tschebyscheff or Bessel responses.
- Filter gains are unity (gain of 1) except for band pass.

If you cannot work within those limitations, consult another source. I have found that these limitations are not extreme: 90% of filter design applications can be realized with these circuits. When combined with the gain/offset techniques of a previous chapter, these filter circuits provide a reasonable signal chain solution for most applications.

To design a filter, some things must be known in advance:

- the frequencies that need to be passed, and those that need to be rejected
- a transition frequency, the point at which the filter starts to work; or a center frequency around which the filter is symmetrical
- an initial capacitor value: pick one somewhere from 100 pF for high frequencies to 0.1 μF for low frequencies; if the resulting resistor values are too large or too small, pick another capacitor value.

6.3.1 Picking the Response

For the beginner, the filter responses will be presented pictorially. The area shaded in blue represents the frequencies that will be passed, and the area in white the frequencies that will be rejected. Do not be concerned with the exact frequency yet; that will be taken care of in the following sections. Look at the responses in Figures 6.3–6.7, and pick one where the desired frequencies are in the shaded area and the rejected frequencies in the unshaded area.

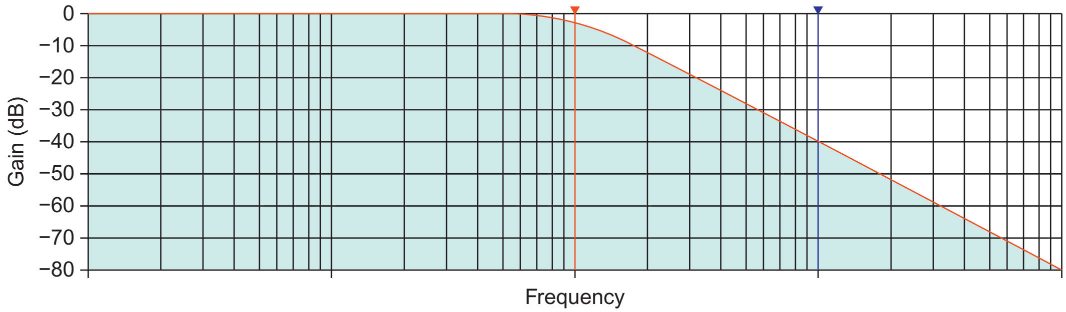

Figure 6.3
Low-Pass Response — Go to Section 6.3.2

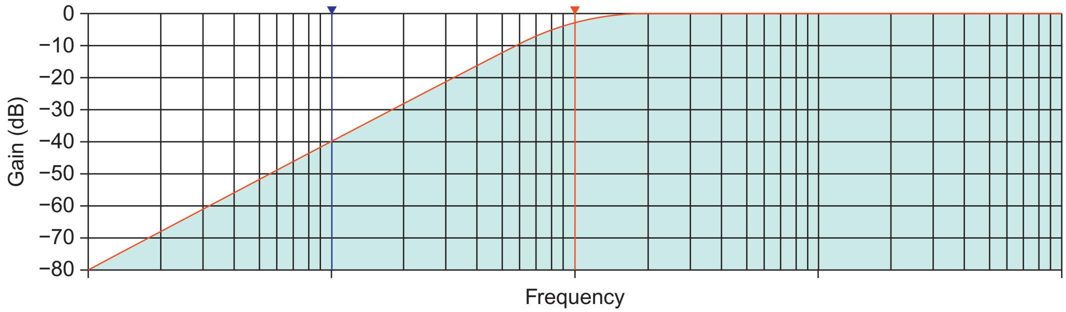

Figure 6.4
High-Pass Response — Go to Section 6.3.3

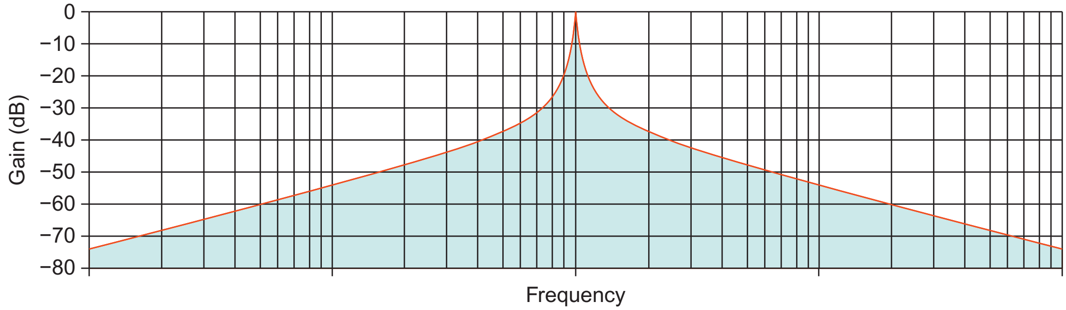

Figure 6.5
Narrow (Single-Frequency) Band Pass — Go to Section 6.3.4

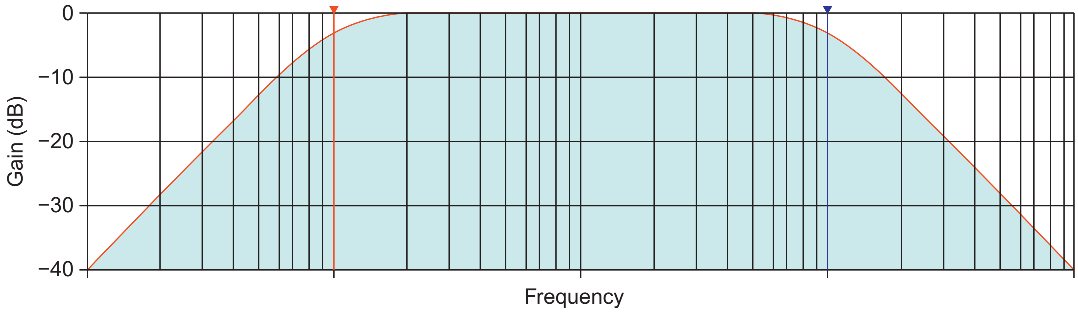

Figure 6.6
Wide Band Pass — Go to Section 6.3.6

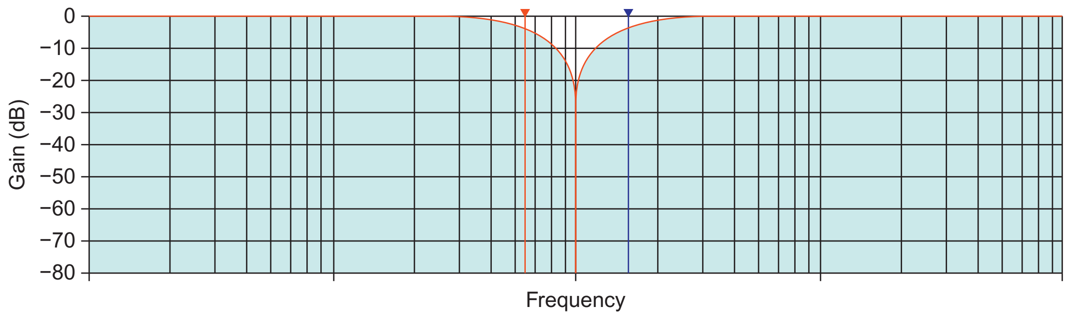

Figure 6.7
Single-Frequency Notch Filter

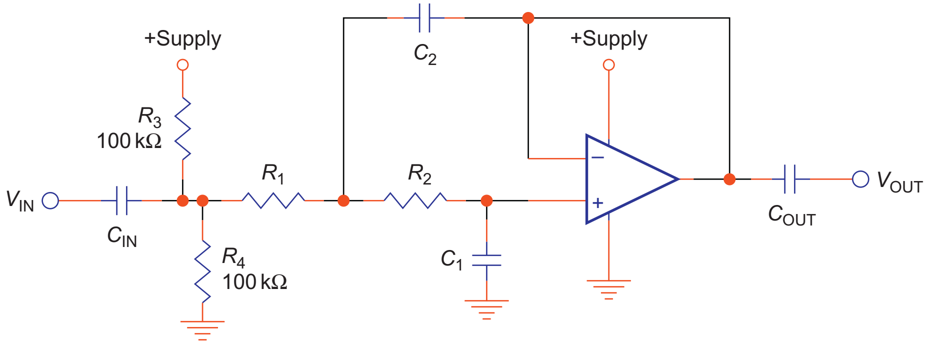

Figure 6.8
Low-Pass Filter

6.3.2 Low-Pass Filter

A low-pass filter is shown in Figure 6.8.

Design Procedure

- Pick C_1.
- Calculate $C_2 = C_1{*}2$.
- Calculate R_1 and R_2: $\dfrac{1}{2\sqrt{2}{*}\pi{*}\mathrm{C}_1{*}\mathrm{Frequency}}$.
- Calculate $C_{IN} = C_{OUT} = 100$ to 1000 times C_1 (not critical).
- DONE!

Digging Deeper

The filter selected is a unity-gain Sallen–Key Filter, with a Butterworth response characteristic. Note that with the addition of C_{IN} and C_{OUT}, the filter is no longer purely a low-pass filter. It is a wide bandpass filter, but the high-pass response characteristic can be placed well below the frequencies of interest. If a DC response is required, the circuit should be modified to operate off split supplies.

6.3.3 High-Pass Filter

A high-pass filter is shown in Figure 6.9.

Design Procedure

- Pick $C_1 = C_2$.
- Calculate R_1: $\dfrac{1}{\sqrt{2}{*}\pi{*}\mathrm{C}_1{*}\mathrm{Frequency}}$.

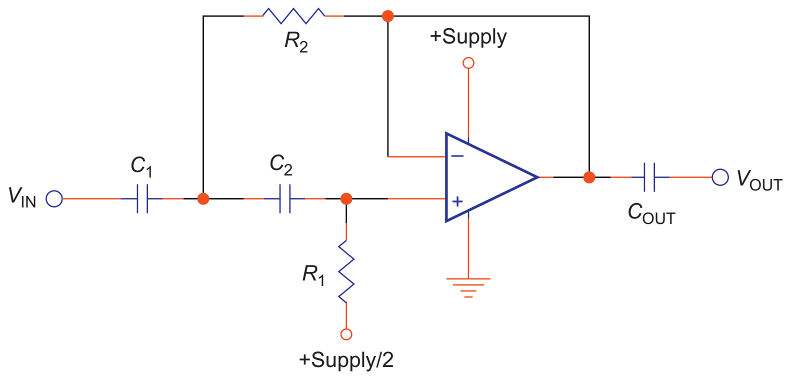

Figure 6.9
High-Pass Filter

- Calculate R_2: $\dfrac{1}{2\sqrt{2}*\pi*C_1*\text{Frequency}}$.
- Calculate C_{OUT} = 100 to 1000 times C_1 (not critical).
- DONE!

Digging Deeper

The filter selected is a unity-gain Sallen–Key filter, with a Butterworth response characteristic. Just as was the case with the low-pass filter, there is no such thing as an active high-pass filter, but for a different reason. The gain/bandwidth product of the op amp used will ultimately produce a low-pass response characteristic, making this a wide bandpass filter. It is your responsibility to choose an op amp with a frequency limit well above the bandwidth of interest.

6.3.4 Narrow (Single-Frequency) Bandpass Filter

A narrow bandpass filter is shown in Figure 6.10.

Design Procedure

- Pick $C_1 = C_2$.
- Calculate $R_1 = R_4$: $\dfrac{1}{2*\pi*C_1*\text{Frequency}}$.
- Calculate $R_3 = 19*R_1$.
- Calculate $R_2 = \dfrac{R_1}{19}$.
- Calculate $C_{IN} = C_{OUT}$ = 100 to 1000 times C_1 (not critical).
- DONE!

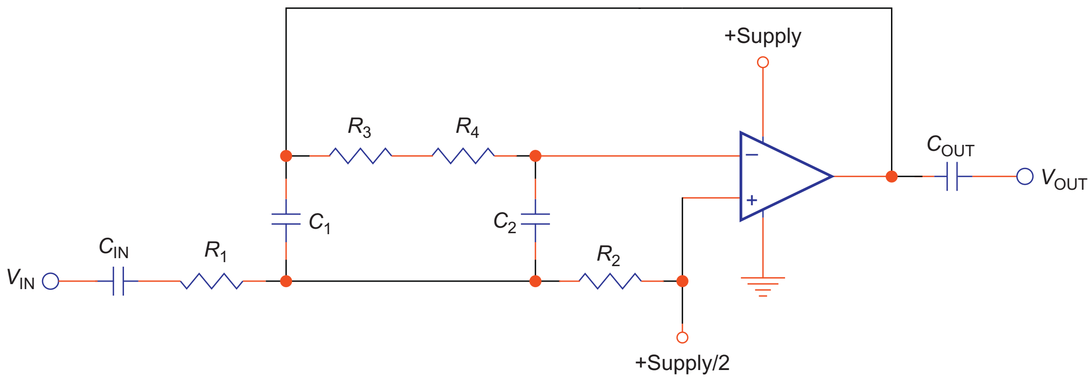

Figure 6.10
Narrow Bandpass Filter

Digging Deeper

The filter selected is a modified Deliyannis filter. A Deliyannis filter is a special case of the multiple-feedback (MFB) bandpass configuration, one that is very stable and relatively insensitive to component variation. The Q is set at 10, which also locks the gain at 10, as the two are related by the expression:

$$\frac{R_3 + R_4}{2 \cdot R_1} = Q = \text{Gain} \tag{6.10}$$

A higher Q was not selected, because the op amp gain bandwidth product can be easily reached, even with a gain of 20 dB. At least 40 dB of headroom should be allowed above the center frequency peak. The op amp slew rate should also be sufficient to allow the waveform at the center frequency to swing to the amplitude required.

6.3.5 Wide Bandpass Filter

A wide bandpass filter is shown in Figure 6.11.

Design Procedure

- Go to Section 6.3.3, and design a high-pass filter for the low end of the band.
- Go to Section 6.3.2, and design a low-pass filter for the high end of the band.
- Calculate $C_{IN} = C_{OUT} = 100$ to 1000 times C_1 in the low-pass filter section (not critical).
- DONE!

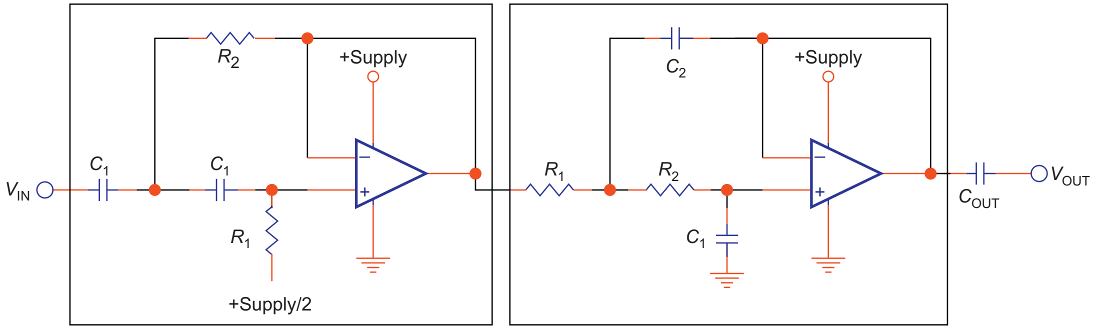

Figure 6.11
Wide Bandpass Filter

Digging Deeper

This is nothing more than cascaded Sallen—Key high-pass and low-pass filters. The high pass comes first, so noise from it will be low passed.

Digging Deeper: Narrow vs. Wide Bandpass Filter

At what point is it better to implement a bandpass filter as a narrow/single-frequency filter rather than as a wide band pass? At high Q values, the single-frequency band pass is clearly the better choice. However, as Q values decrease, the difference begins to blur. What can be a very sharp peak at resonance erodes to a single-pole roll-off on the low end, and single-pole roll-off on the high end. This results in a lot of unwanted energy in the stop bands.

For Q values of 0.1 (and below) and 0.2, the best implementation is high pass cascaded with low pass. The two implementations have almost an identical pass band response for a Q of 0.5. You are presented with a choice: use a bandpass filter (which can be implemented with a single op amp) to save money, or use a cascaded approach that has better rejection in the stop bands. As the Q becomes higher and higher, however, the responses of two separate stages begin to interact, destroying the amplitude of the signal. A good rule of thumb is that the start and ending frequencies of a wide bandpass filter should be at least a factor of five different.

6.3.6 Notch (Single-Frequency Rejection) Filter

Design Procedure

- Pick C_o.
- Calculate R_o: $\frac{1}{2*\pi*C_1*\text{Frequency}}$.

- Calculate $R_Q = 20^*R_o$.
- If you do not want to tune, replace R_{o_low} and R_{o_adj} with R_o. If tuning of the center frequency is desired, make a pot part of the value of R_o by going down one standard resistor value, and making sure the pot covers the range of center frequencies. If you do the job right, you can fine-tune the center frequency while preserving the depth of the notch. The nice thing about this topology is that you will get a very deep notch – somewhere!
- DONE!

Digging Deeper

This is the Fliege filter topology, set to a Q of 10. The Q can be adjusted independently from the center frequency by changing R_Q. Q is related to the center frequency set resistor by the following:

$$R_Q = 2^*Q^*R_o$$

The Fliege filter topology has a fixed gain of 1. It is best to implement it from split supplies, although it can be operated from a single supply. Inject a reference into RQ instead of ground to operate off split supplies. The input and output will have to be isolated by DC blocking capacitors as with other filter types.

Many designers use the "Twin-T" notch topology of Section 6.5.4 for notches. While it is a popular topology, it has many problems. The biggest is that it is not producible. Many runs of simulation with component tolerances of 1% have shown tremendous variation in notch center frequency and notch depth. The only real advantage is that it can be implemented with a single op amp. Some additional stability can be obtained from the two op amp configuration, but if two op amps are used, then why not use a different topology such as the Fliege? To successfully use the Twin-T topology, six precision components are required. The Fliege will produce a deep null at some frequency, and it is easy to tune that frequency by adjusting one of the R_o resistors; the null will remain as deep over a fairly wide range. The response plot shown in Figure 6.13 was made by varying the potentiometer on a 10 kHz Fliege filter in 5% increments.

Some key "takeaways" from the Fliege filter response: it does not disturb the frequencies around it to any significant degree. The response in Figure 6.12 shows that a high Q 10 kHz bandpass filter leaves everything under 9 kHz and above 11 kHz almost unchanged. The response in Figure 6.13 shows that a pot forming 2% of the R_o value in the position shown allows adjustment of the center frequency over about a $\pm 1\%$ range around the center frequency. The depth of the notch

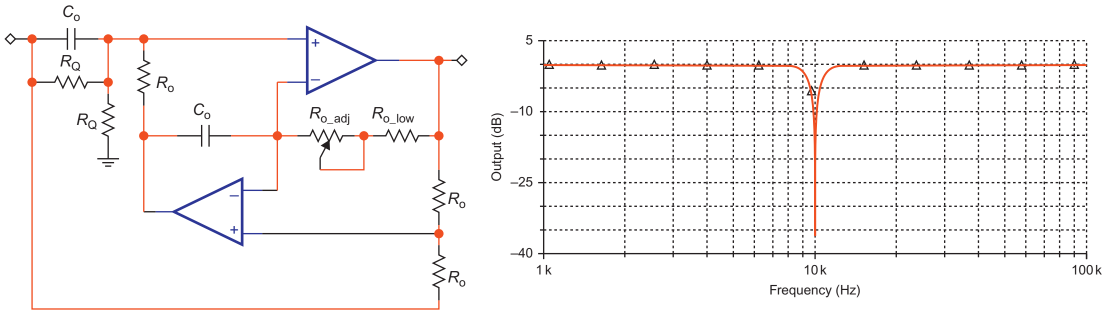

Figure 6.12
Notch Filter

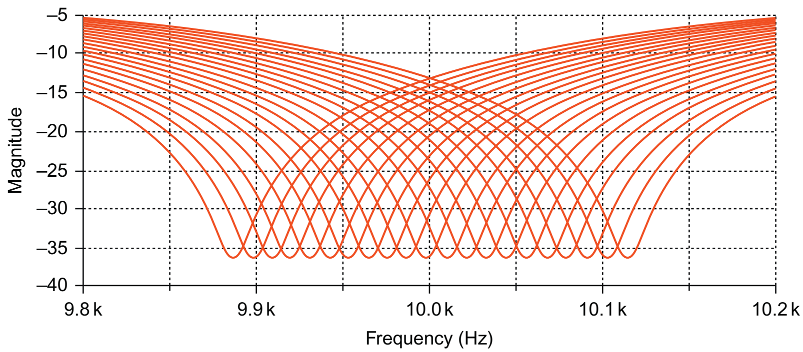

Figure 6.13
Variable-Frequency Notch Filter

remains unchanged over that range of adjustment, making it an ideal way of tuning the notch frequency.

Incidentally, if the reader wants to construct a medium-wave heterodyne filter such as the one shown above, the component values are $R_o = 4.42$ k, $C_o = 3600$ pF, $R_{o\_low} = 4.32$ k, $R_{o\_adj} = 200\ \Omega$, and $R_Q = 88.7$ k. The op amps should be at least 100 MHz in bandwidth.

6.4 High-Speed Filter Design

At high speeds, filter design gets particularly interesting. Of course, interesting is a word and a half word. If you love challenges, high-speed filter design will test the limits of what you can do with analog filters, with the foreknowledge that things will start to get strange!

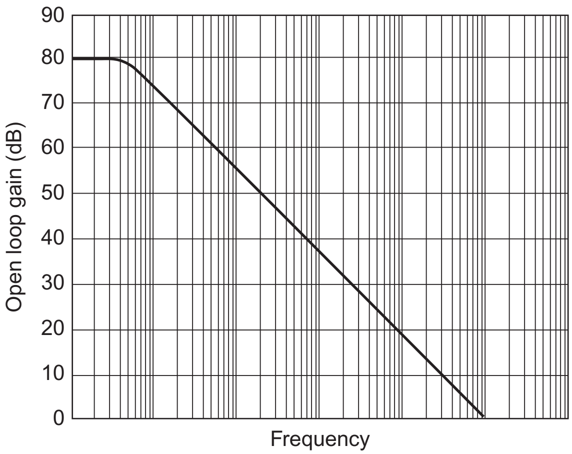

Figure 6.14
Open-loop Response

6.4.1 High-Speed Low-Pass Filters

Reviewing a figure from earlier in the book, it is obvious from Figure 6.14 that the general shape of the open-loop response plot is that of a low-pass filter. Therefore, if the low-pass breakpoint is not terribly critical, all you have to do is to select an op amp with a unity gain bandwidth at the desired −3 dB breakpoint. Imagine explaining to a manager how the low-pass filter actually is identical to a unity gain stage. The high-speed low-pass filter can also have gain, although it will lower the −3 dB breakpoint by approximately a decade for each 20 dB of gain.

6.4.2 High-Speed High-Pass Filters

I refer the reader back to a statement I made in Section 6.3.3: “There is no such thing as an active high-pass filter, but for a different reason. The gain/bandwidth product of the op amp used will ultimately produce a low-pass response characteristic, making this a wide bandpass filter. It is your responsibility to choose an op amp with a frequency limit well above the bandwidth of interest.” This is doubly so at high speeds, because you are inevitably closer to the open-loop limitation of the op amp.

6.4.3 High-Speed Bandpass Filters

This is where things get really interesting, because the physics of an op amp can and will actually force the resonant peak off frequency (lower) and erode the peak.

You may be asking: how can this be? The capacitors and resistors are supposed to define the center frequency of the filter; they will not change with frequency. This is a valid question, and it leads to the answer: the frequency shift is coming from the op amp itself. But again, why? The answer comes from the location of the open-loop response characteristic of the op amp, which is the ultimate speed limit of the op amp at any frequency. A bandpass filter is composed of both low-pass and high-pass elements, and the high-pass characteristics will tend to get chopped off as they approach the open-loop characteristic. This will appear first as an amplitude limitation, and finally as a frequency shift as the bandwidth limitation of the high-pass elements comes into play and limits the point at which they interact with the low-pass filter elements. The result is a truncated response that appears to be a frequency shift.

To illustrate this effect, bandpass filters were constructed, using the topology of Section 6.3.4. The results are shown in Figure 6.15. Three frequencies were constructed, indicated by the sets of peaks in the figure. For 10 MHz, the third set of peaks, a Q (and gain) of 1, the open-loop response of the op amp was a little over 30 dB above the peak at 10 MHz, and the filter actually worked very well. As the Q (and gain in V/V) was raised in steps of 5, however, things began to change. By the time a Q of 25 was attempted, the gain of the filter was almost back to

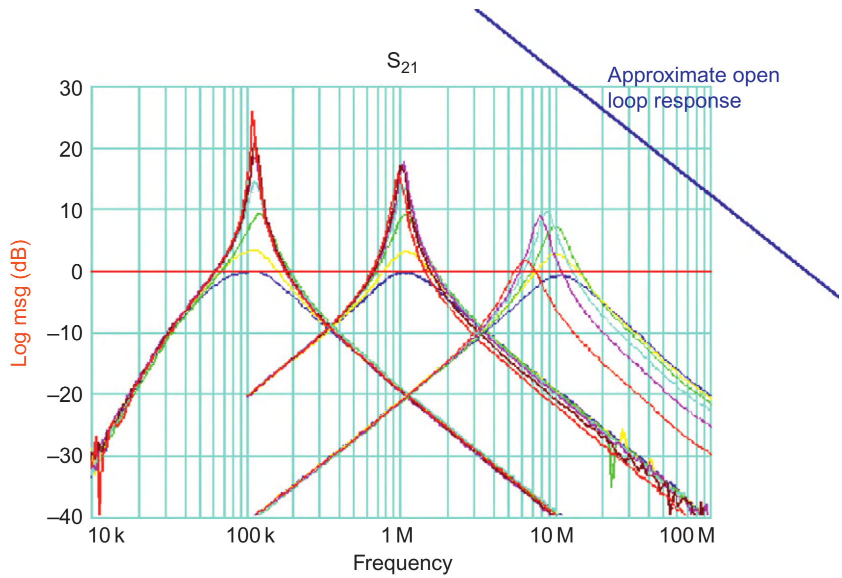

Figure 6.15
Bandpass Response

unity, and the frequency shifted to the left to about 6.5 MHz. Clearly, the proximity of the open-loop response was affecting the op amp. Even the attempts to make a 1 MHz bandpass filter – although not showing the undesirable frequency shift – still show an amplitude compression effect. Only the 100 kHz filter shows anything close to lab results matching theoretical results.

Note that the open-loop response of this particular op amp indicates that it is approximately a 1 GHz op amp – in other words, close to the state of the art in op amp design. Therefore, there is a practical limit to how fast a bandpass filter can be constructed: about 10 MHz for unity gain and Q of 1, or about 1/100 the rated bandwidth of the op amp. If a higher Q is desired, say 10, then the practical limit is about 1/1000 the rated bandwidth of the op amp. In other words:

$$\text{Center frequency (maximum)} = Aol/(100^{*}Q) \tag{6.11}$$

This limitation may also hit lower frequency bandpass filters, so be extremely careful! Even a fairly low-frequency bandpass filter may require a very fast op amp to accommodate higher values of Q. You have two choices if you encounter this limit: either select an op amp with a higher gain/bandwidth product, or lower the Q of the bandpass filter.

6.4.4 High-Speed Notch Filters

A very similar bandwidth restriction affects notch filters. Instead of eroding the amplitude of the peak, as it does bandpass filters, the bandwidth restriction erodes the depth of the notch (Figure 6.16).

Using the exact same op amp as in the bandpass section above, and the notch filter topology of Section 6.3.6, notch filters were constructed at 10 MHz, 1 MHz, and 100 kHz. No center frequency tuning was attempted. The 10 MHz results were so terrible they are not included here. Even the 1 MHz results show dramatically how bandwidth was affecting the notch. At a Q of 1, a 30 dB notch is possible at 1 MHz. However, higher Q values only erode the depth of the notch further. This erosion is even evident at 100 kHz. At this point, it is clear that even a 1 GHz op amp can only be used to construct notch filters at 1 MHz and a Q of 1. If a Q of 10 is desired, a 1 GHz op amp can only be used to construct a notch filter of 100 kHz. This amazing degree of limitation was totally unexpected, to say the least.

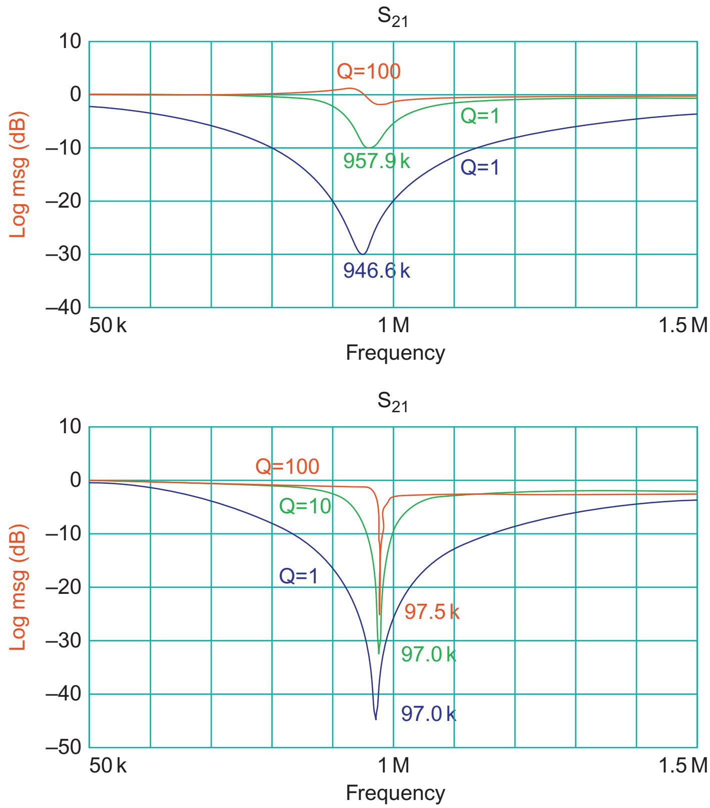

Figure 6.16
Notch Filter Response

6.5 Getting the Most Out of a Single Op Amp

As unexpected as some of the results above were, there are even stranger things that can be done with op amps. Well, some not so strange, but this chapter will not disappoint in later sections!

6.5.1 Three-Pole Low-Pass Filters

Section 6.3.2 showed how to implement low-pass filters easily and quickly. However, why implement only a two-pole filter, when a single op amp can just as easily implement a three-pole filter (Figure 6.17)?

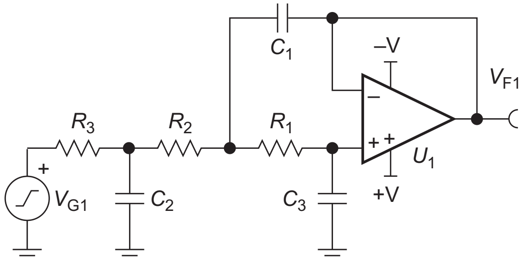

Figure 6.17
Three-Pole Low-Pass Filter

The response, in this case, will roll off 60 dB per decade instead of 40 dB per decade as seen in the two-pole filter. This topology also solves a different problem associated with the Sallen–Key architecture, that of feedthrough. In a two-pole Sallen–Key filter, high frequencies will leak through the filter, especially when the amplifier is turned off. This three-pole topology adds an RC low-pass filter to the input of a two-pole Sallen–Key architecture, thus absolutely guaranteeing at least a 20 dB per decade roll-off of high frequencies no matter what happens to the amplifier.

To design this three-pole low-pass filter:

- Pick a value of resistance $R = R_1 = R_2 = R_3$.
- Calculate a base value fsf = 2*π*R*Frequency.
- Calculate C_1 = 3.546/fsf.
- Calculate C_2 = 1.392/fsf.
- Calculate C_3 = 0.2024/fsf.
- Pick standard value capacitors closest to the calculated ones above.
- DONE!

6.5.2 Three-Pole High-Pass Filters

Just as it is easy to implement three-pole low-pass filters, it is also easy to implement three-pole high-pass filters (Figure 6.18):

To design this three-pole high-pass filter:

- Pick a value of capacitance $C = C_1 = C_2 = C_3$.
- Calculate a base value fsf = 2*π*C*Frequency.
- Calculate R_1 = 3.546/fsf.
- Calculate R_2 = 1.392/fsf.
- Calculate R_3 = 0.2024/fsf.

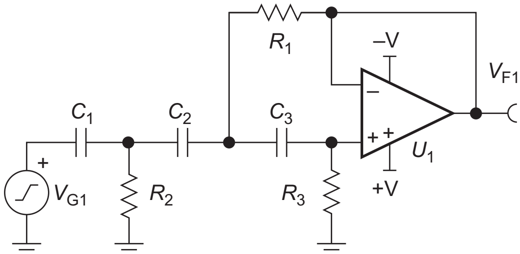

Figure 6.18
Three-Pole High-Pass Filter

- Pick standard value resistors closest to the calculated ones above.
- DONE!

6.5.3 Stagger-Tuned and Multiple-Peak Bandpass Filters

This book has purposely not covered an interesting topology that is very popular: the Twin-T topology. Part of the reason why this has not been done is that for bandpass filters, it is not a true bandpass topology. It is more of a resonator at the center frequency, and also has theoretically infinite gain at its resonance (for ideal components). Therefore, in the real world, it is difficult to control the gain at the center frequency, and the ultimate stop band rejection is 0 dB – therefore unity gain. Thus, it is not very useful in rejecting out-of-band signals.

The Twin-T topology for bandpass filters and its response are shown in Figure 6.19. This topology is very difficult to work with. It requires the user to obtain three resistors, one of which is exactly half the other, and three capacitors, one of which

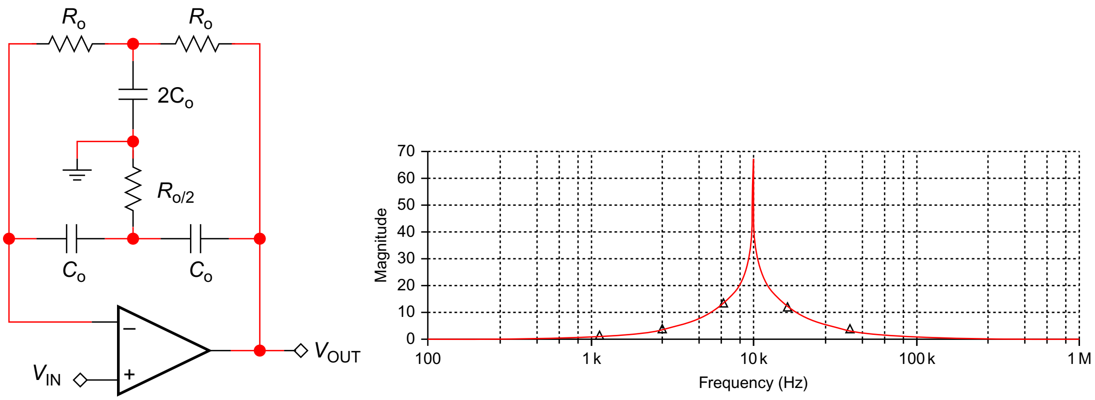

Figure 6.19
Twin-T Bandpass Filter Response

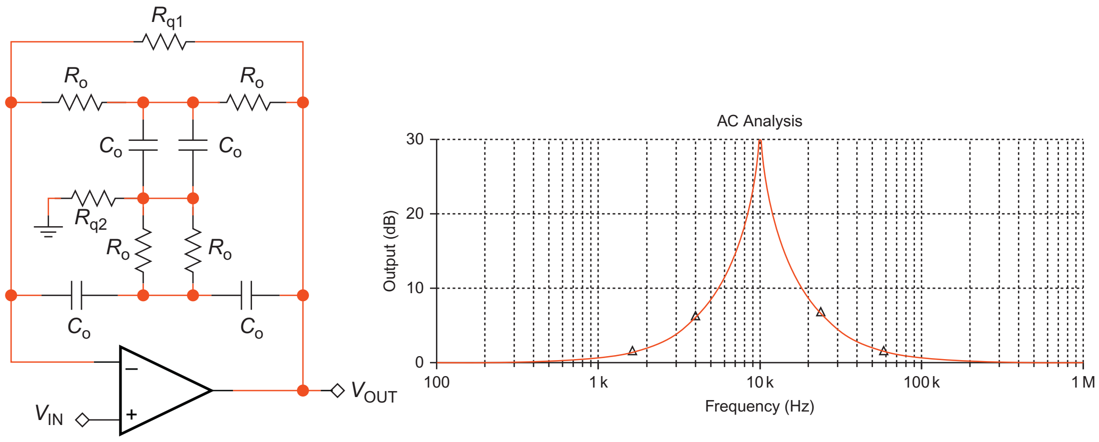

Figure 6.20
Modified Twin-T Topology

is exactly twice the other. Even if this is possible, the chances are that they would not match exactly, or track with temperature. Furthermore, the peak is so sharp that real-world components might erode the peak or miss it entirely. To overcome these shortcomings, the modifications in Figure 6.20 are made.

This configuration takes advantage of parallel resistance and capacitance characteristics to make your job easier. An additional R_o and C_o have been added, but this means that you now only have to find four identical values of resistance and four identical values of capacitance — no more 2 C_o and ½ R_o required. Because components manufactured in the same lot often have virtually identical characteristics, this should be easy.

A more subtle and unfamiliar change to the familiar Twin-T topology is the addition of R_{q1} and R_{q2}. This takes advantage of the only two places where the circuit's Q can be adjusted. As long as $R_{q1} >> R_o$, it has minimal effect on the center frequency, but it acts to make the two C_os that are in series appear to have more leakage, therefore eroding the amplitude of the peak. Similarly, if $R_{q2} << R_o$, it makes the two parallel C_os appear to have higher equivalent series resistance (ESR), also eroding the peak. Acting together, it is possible to have a measure of control over the amplitude and peak, although not as precise as you might like. Comparing the response of the modified and unmodified circuits, you can see that the tendency of the amplitude to be unbounded at resonance has been mitigated, and the response is more reasonable. The Q has also been modified, although it is hard to see on the 4 decade log frequency scale.

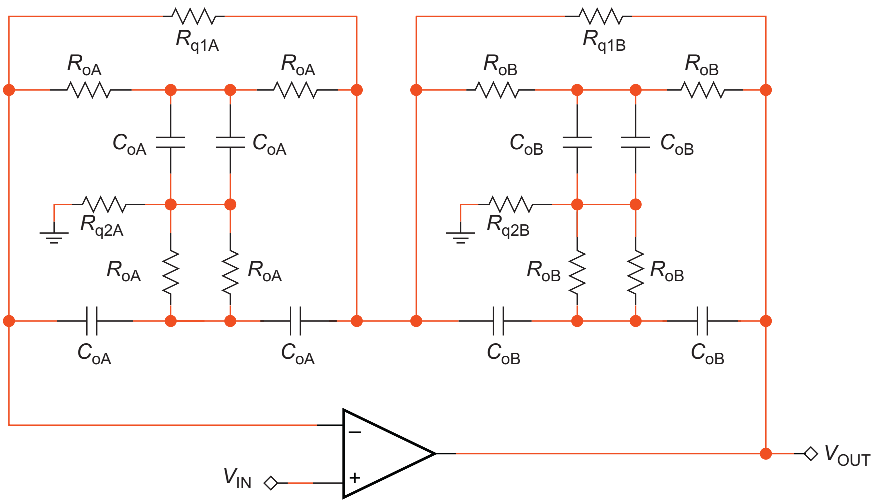

Figure 6.21
Two Twin-T Networks Inside the Feedback Loop

Another technique that has been used to adjust the *Q* of a Twin-T circuit is to unbalance R_o and C_o in the legs of the circuit that are in series. This, however, requires you to find multiple precision component values, all of which could potentially track differently over temperature.

With this brief introduction to Twin-T bandpass filters complete, now the real fun can begin! There is no reason that two Twin-T networks cannot be placed inside the feedback loop of an op amp (Figure 6.21). The two Twin-T sections are effectively in series inside the feedback loop of the op amp. They can be tuned entirely independently, resulting in the response shown in Figure 6.22.

Such a strategy is often employed when it is necessary to have flat phase (and group delay) in a narrow region around the center frequency. The trade-off is that the passband of the filter is widened to accommodate the flat phase response in the middle. You are also reminded to select a wide band op amp that will be able to handle not only the gain at the center frequency, but also the gain of the upper peak. The open-loop response should be at least 40 dB above the upper peak. In this case, an op amp with 1.5 GHz bandwidth was used to simulate a stagger tuned 10 kHz filter, in other words a bandwidth more than five orders of magnitude greater than the center frequency.

If the two Twin-T networks are separated enough in frequency, this topology will produce a bandpass filter that has more than one resonant peak, as shown in Figure 6.23. Notice that the flat phase response has been sacrificed between the

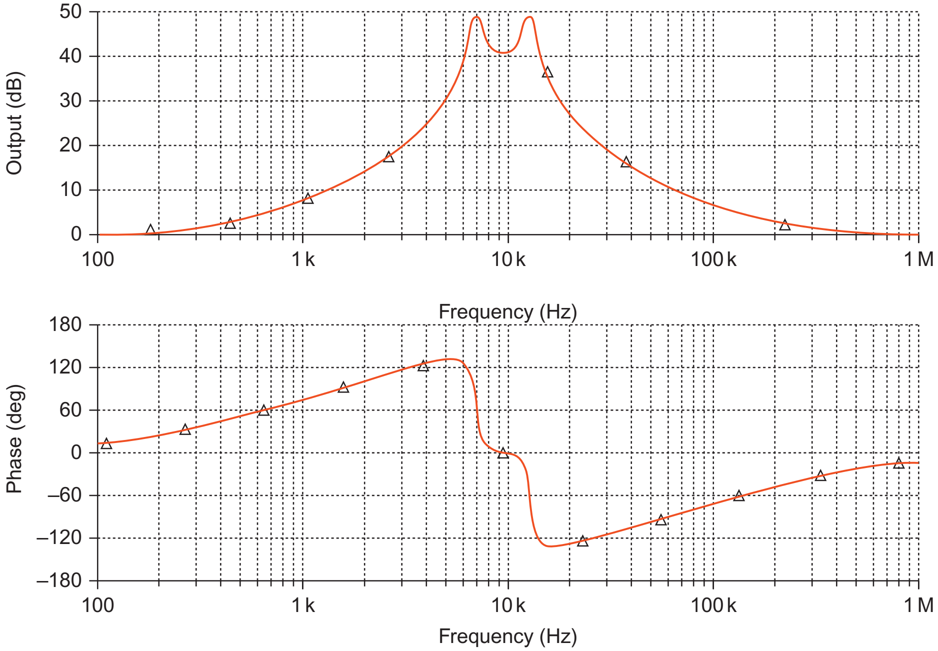

Figure 6.22
Stagger-Tuned Filter Response

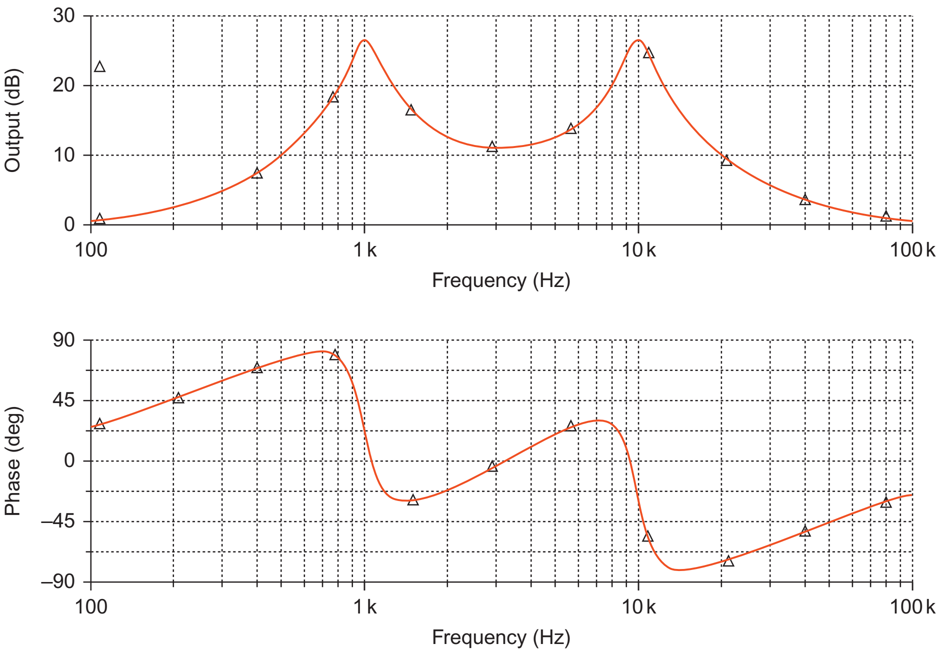

Figure 6.23
Multiple-Peak Bandpass Filter

peaks, although the phase still crosses through 0 degrees. But a flat phase between the peaks is no longer the design goal; rather, having two peaks is. The "valley" between the two peaks is not that low; it would be lower if the peaks were further separated in frequency or the *Q* was higher. But for quick detection of multiple tone frequencies, this method is the most economical possible.

You are cautioned again about op amp bandwidth, but in this case the open-loop gain at the upper peak may be less of a constraint because the overall gain is lower.

6.5.4 Single-Amplifier Notch and Multiple-Notch Filters

The Twin-T topology can also be used to create a single amplifier notch filter (Figure 6.24). As was the case in the Twin-T bandpass filter, two resistors were added that allow some control over the *Q*, also affecting notch depth. This is very useful, because the Twin-T notch is difficult to tune for center frequency. If the *Q* is lowered, the chances are better of actually placing a notch where it is needed to reject an unwanted frequency.

Unfortunately, the Twin-T topology also has the drawback that the notch disturbs the amplitude in the decades above and below the center frequency. Contrast the response above with the response in Figure 6.24. Where the Fliege topology leaves the surrounding frequencies almost untouched, the Twin-T has a very significant effect almost a decade above and below the center frequency, making it a poor choice for a notch filter. So, why use it at all? Besides the advantage of being a single amplifier notch, the other advantage is, like the Twin-T bandpass topology,

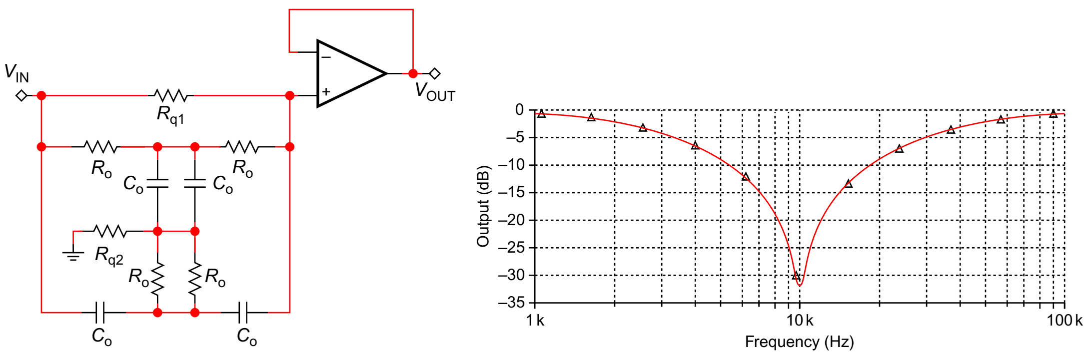

Figure 6.24
Single-Amplifier Twin-T Notch Filter

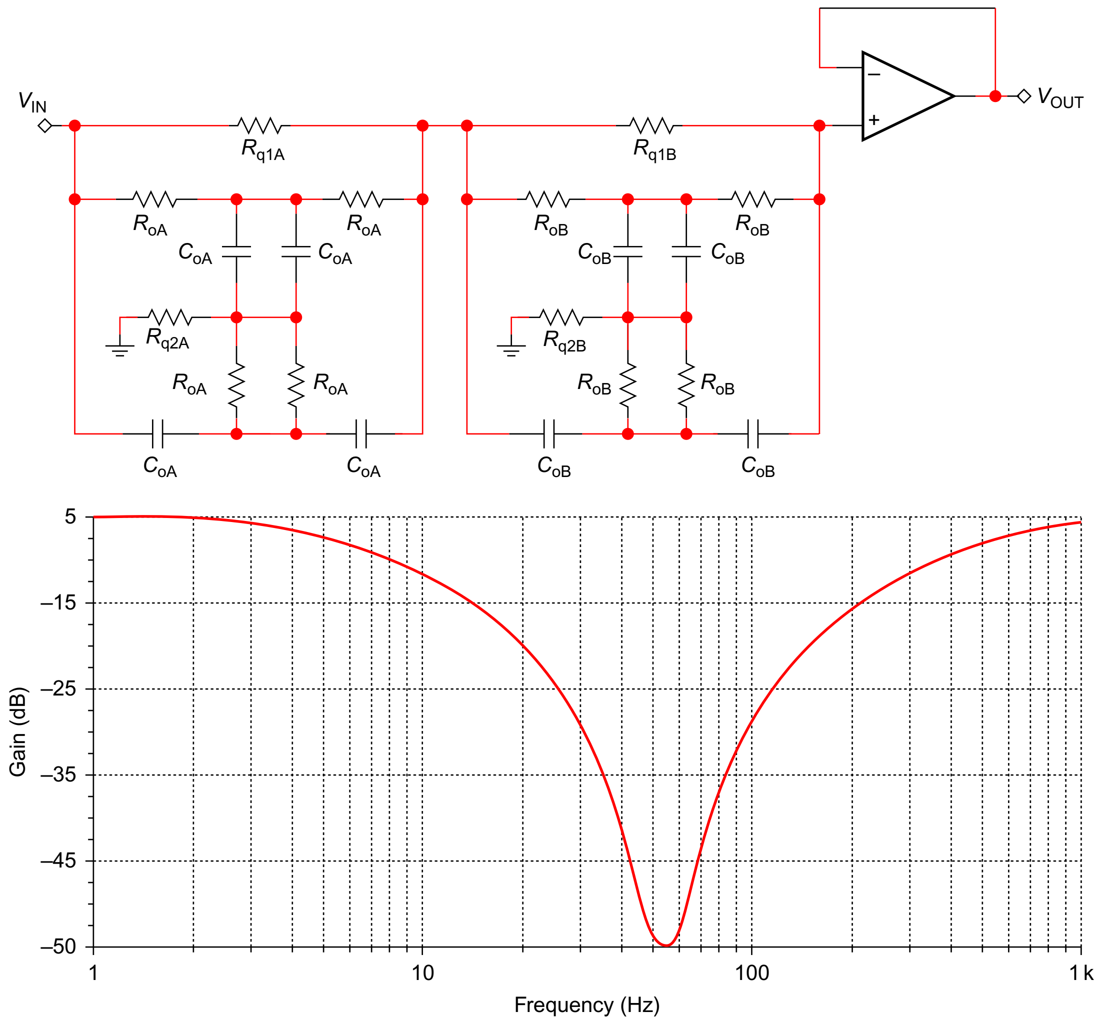

Figure 6.25
Twin-T Band Reject Filter

that it is possible to produce two notches from a single amplifier. This can be a way either to produce two individual notches at widely separated frequencies, or to produce a band rejection filter if the notch frequencies are closely spaced (Figure 6.25).

This scheme dramatically affects the audio above 10 Hz and below 300 Hz. It may, however, be suitable for severe cases of hum on speech.

6.5.5 Combination Bandpass and Notch Filters

There is no reason why a Twin-T notch and a bandpass filter cannot be combined in a single circuit (Figure 6.26).

Of course, more than one peak and/or notch is also possible. In practice, however, you should probably count on no more than two or three sections of Twin-T networks, because the number of passive components crowding the op amp will become excessive, leading to parasitics on the board and other problems.

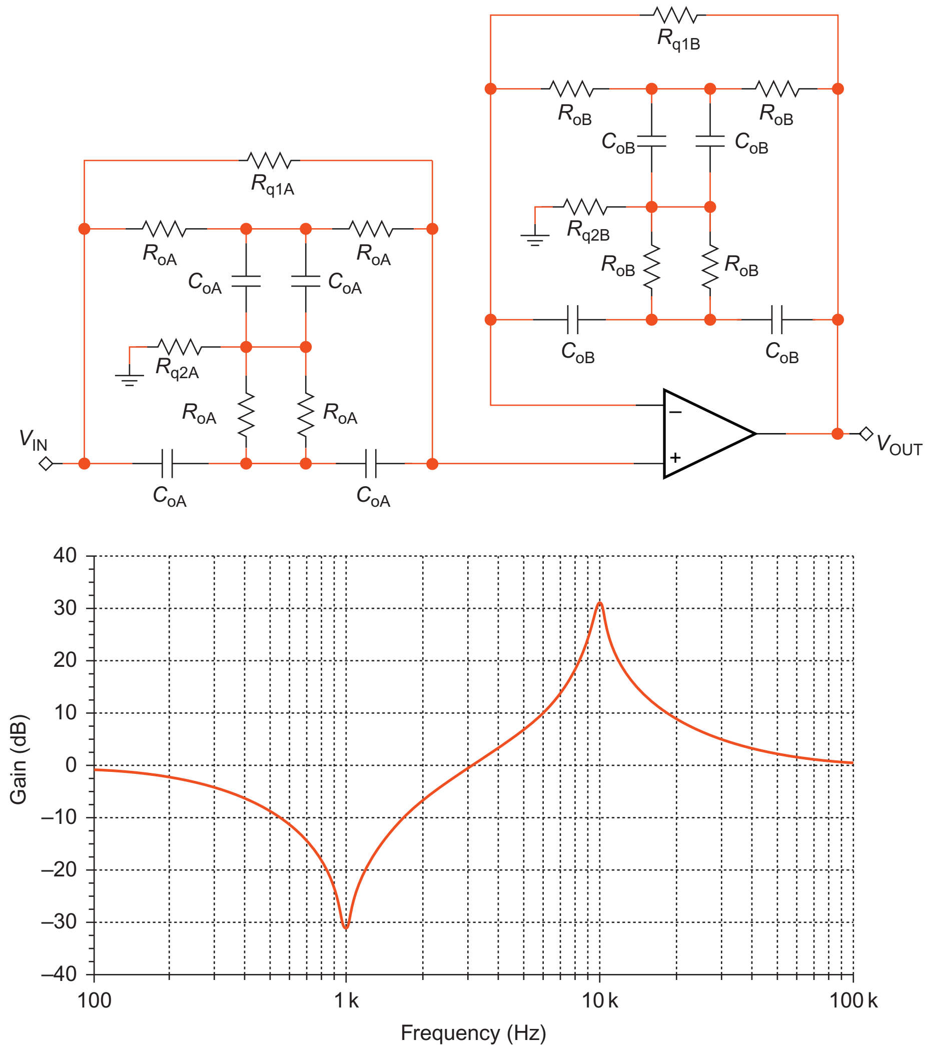

Figure 6.26
Single-Amplifier Twin-T Notch and Bandpass Filter

6.6 Biquad Filters

No three- and four-amplifier biquad circuits are presented here. The overwhelming majority of applications require you to use as few op amps as possible to save power, board space, and cost; reduce noise; and improve reliability. These advantages usually far outweigh any advantages a multiple op amp configuration might provide, such as providing more than one output function at one time, or allowing separate, independent adjustments of each filter parameter. If you need such a filter, there are many volumes out there covering multiple amplifier biquads. I will offer only one small piece of advice here: consider using a fully differential amplifier in biquad configurations. Often the third amplifier in a biquad circuit serves only to invert the signal; fully differential op amps already include that inverted output.

6.7 Design Aids

Just as I did for op amp gain and offset, I have written design aids for filter design.

6.7.1 Low-Pass, High-Pass, and Bandpass Filter Design Aids

This chapter has presented you with a bewildering variety of filter circuits to ponder. It is impossible (well, it would be very difficult) to produce a single circuit that could implement all filter types. I will leave you instead with a single schematic of a universal filter circuit (well, low pass, high pass, and band pass) (Figure 6.27).

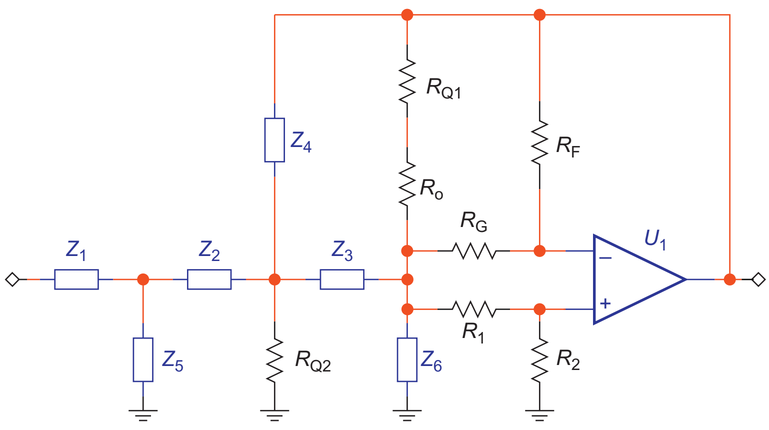

Figure 6.27
Universal Filter Schematic

The general-purpose impedances, or *Z*, can be either resistors or capacitors, depending on the filter topology selected. Not all components will be installed for every filter. Some may be 0Ω resistors, whereas others may be "open", not installed.

This single circuit can implement low-pass, high-pass, and bandpass stages. Although two-pole Sallen–Key filters can be implemented, why bother when you can have three poles just as easily?

As was the case for gain and offset, I have written a design aid for filters, which is also available on the companion website. Figure 6.28 shows a screen shot. Some explanation is in order. When the calculator first comes up, all of the impedances and units will be blank, unlike the screen shot in the figure. To use the calculator, first you need to know whether you are designing a low-pass, high-pass, or bandpass filter. These three types are available in a drop-down menu in the "Filter Type" box. Then, there are a series of boxes that depend on the type of filter you are making: For (type of filter), Enter (value) series of lines. For example, all filter types, low-pass, high-pass, and bandpass, require a frequency. For low and high pass, it will be the −3 dB cut-off frequency. For band pass, it will be the center frequency. So, for ALL filters, Enter Frequency. Because this is a low-pass filter, the only remaining decisions you need to make

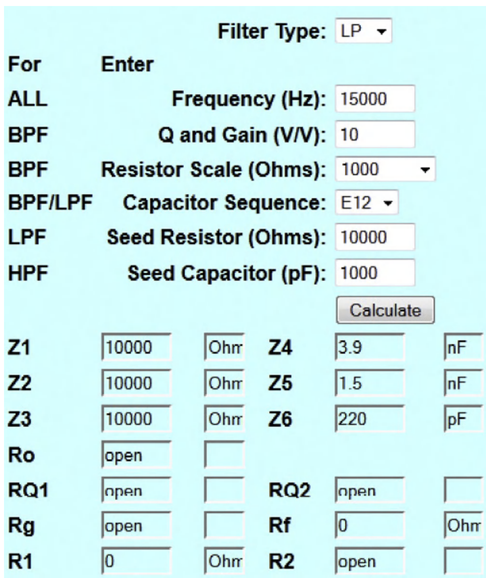

Figure 6.28
Universal Filter Calculator

are the capacitor sequence (E6, E12, or E24) and the Seed Resistor value, which will scale the capacitors. Click the "Calculate" button and the component values appear.

If you choose a high-pass filter instead, you will notice that capacitors and resistors have swapped left to right. Of course, you choose a seed capacitor, instead of a seed resistor, for high-pass filters. The seven resistor values on the bottom, R_o–R_2, are the same for low-pass and high-pass filters, either 0Ω or open. These route connections configure the universal schematic and PCB to be a three-pole Sallen–Key filter topology. This changes dramatically when you choose "band pass" instead. Z_1 becomes 0Ω, Z_5 and Z_6 become open, and some of the resistor values on the bottom are populated, while others are used to configure the circuit for the modified Deliyannis configuration.

In addition, the schematic and circuit notes will appear to the right of the calculator. These change according to topology. It will take some experience on your part to properly scale resistors and capacitors. All capacitors should be 1% NPO/COG dielectric if at all possible, especially if the filter is to encounter different temperatures. Although higher value NPO/COG capacitors are becoming more common, 10,000 pF is generally the limit, and anything above 1000 pF is usually large, expensive, and hard to obtain. Resist at all costs the temptation to use 5% capacitors, especially in the bandpass filter. Choose 1% tolerance, and make that decision stick with procurement and manufacturing departments.

To further facilitate your designs, Figure 6.29 shows a board layout of the universal filter schematic above. It is designed to accommodate SO-8 single op amps and 1206 surface-mount resistors. It is implemented in a single PCB layer. The gerbers are also available from the companion website.

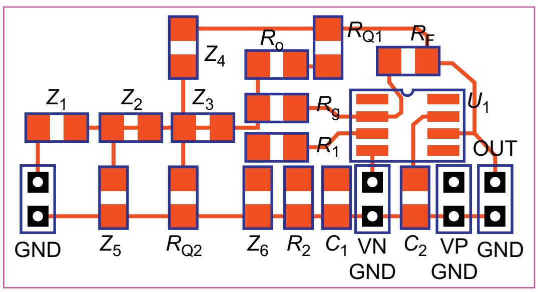

Figure 6.29
Universal Filter Board

You can use this approach to implement a universal arrangement on a PCB. This might be done when multiple filter stages are to be used, and it is unclear whether a given digital signal processor (DSP) algorithm really needs a high-pass function or a low-pass function, or whether the application can benefit from two stages of one type, for example.

6.7.2 Notch Filter Design Aids

But wait, there is more! Remember I said it would be difficult to implement a universal filter board that also contained a notch? I have not left you to your own devices. I also have written a notch filter design aid. A screen capture is shown in Figure 6.30.

Like the other filter calculator, this one also shows the schematic. This calculator only requires a center frequency; the user can adjust the Q, capacitor sequence, and resistor scale. As the notes indicate, however, you should not deviate too much from a Q of 10: this is a good compromise value that will give plenty of rejection, yet leave other frequencies untouched. The calculator also includes provisions for tuning the center frequency. If you do not want to do that, just substitute another R_o for R_{o_low} and R_{o_adj}.

I have also generated a PCB layout for the notch filter (Figure 6.31). This one had to be two layers, mainly so I could route the power. If this is a hardship, just route the power with wires on the back of a single-layer board.

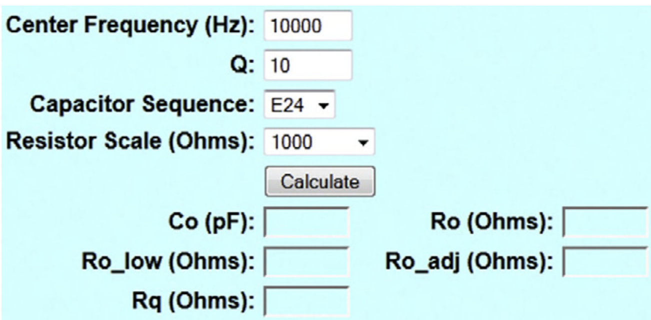

Figure 6.30
Notch Filter Calculator

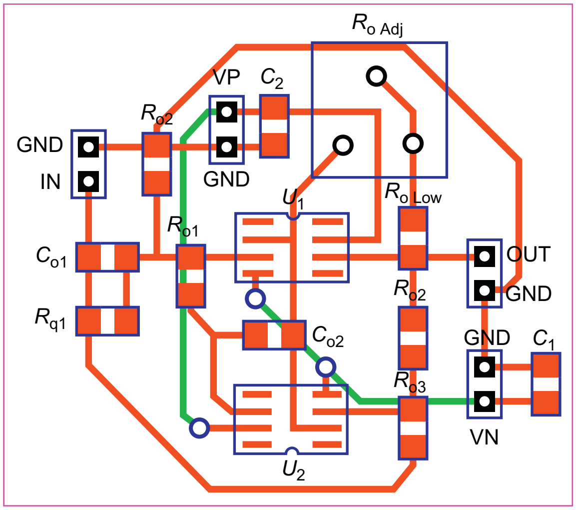

Figure 6.31
Notch Filter PCB

6.7.3 Twin-T Design Aids

I have written separate design aids for Twin-T bandpass and Twin-T notch filters. The screen capture for both of them looks the same (Figure 6.32).

There is little to decide: enter the center frequency, select the resistor and capacitor sequence, and click "Calculate". There is a resistor scale control. Leave the Q alone; it will allow about 30 dB gain for the band pass and 30 dB of rejection for the notch. Adjustment is possible, but over a relatively narrow range. Remember that Twin-T filters cannot be tuned easily for center frequency, and the response will not be ideal.

I have also generated a PCB layout for Twin-T filters (Figure 6.33). This one had to be two layers, mainly so I could route the power. Two of the Q adjustment resistors are also on the back of the board. If the Q resistors are not used, jumpers can be added to ground from the center of the R_o C_o pattern, and the Q resistors on the top of the board can be left unused. Remember this will make a very narrow bandpass/ notch with uncontrolled amplitude and center frequency.

The board is designed to support one section of notch and one section of band pass. If notch is desired, place a 0Ω resistor across the R_{q3} pads. If band pass is desired, place a 0Ω resistor across the R_{q1} pads. This board will also support the notch/ bandpass configuration of Section 6.5.5.

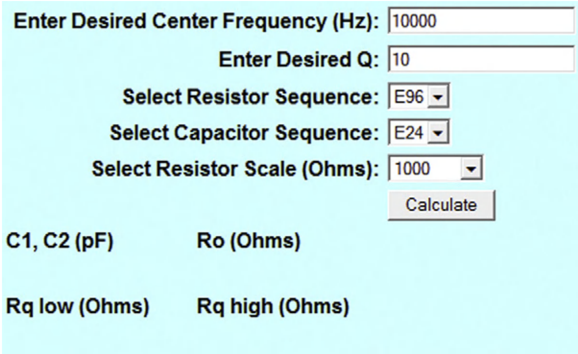

Figure 6.32
Twin-T Filter Calculator

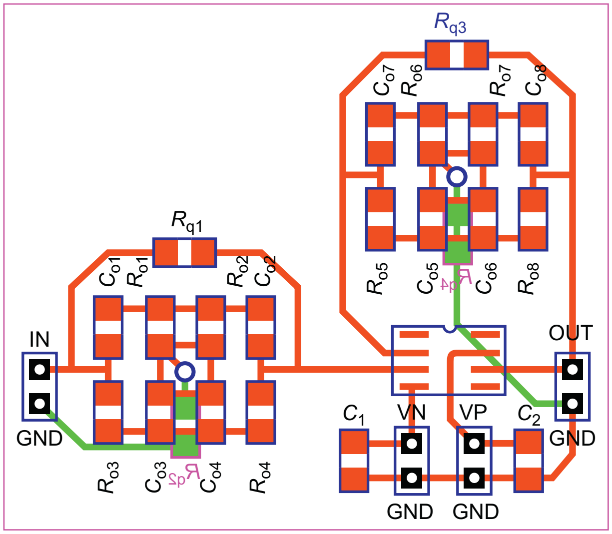

Figure 6.33
Twin-T PCB

6.7.4 Final Comments on Filter Design Aids

You can quickly implement an entire signal chain using multiples of these filter boards and the universal gain and offset board of Chapter 3. Care must be taken, of course, to get the DC operating point of the circuit right, so you might want to implement filters first, then gain and offset.

6.8 Summary

Filter design is a subject plagued by misinformation, and endless textbooks with mathematical derivations, tables, and graphs. Many hours have gone into demystifying the topic for you in this book. Gone are the derivations and math exercises, and in their place are working filter circuits with real-world component values. Many hours have also gone into providing filters that have been tested extensively in the lab, and are the simplest possible implementation available.

Filter design in general requires that you:

- know which frequencies you want to pass, and which frequencies you want to reject
- have the ability to choose a filter topology that accomplishes the passing and rejection of those frequencies
- have the ability to calculate the component values to accomplish that filtering function — it is hoped that the design aids will greatly facilitate this.

When this procedure is followed, good results follow. As you design filters successfully, you will gain confidence and experience in the complex subject of filter design.