ACTIVE BANDPASS FILTERS

SOMETIMES A NEW VERSION OF A CIRcuit comes along that seems complicated at first, but ends up solving a lot of problems in a simple way. This is certainly true of an active bandpass filter technique called *biquadratic section*. It seems to take a lot of parts —including three operational amplifiers. But you end up with stable and simple operation, high Q, easy design, and independent control of practically everything. You'll find the circuit handy for bandpass filters of all types, particularly in electronic music and percussion circuits.

A *bandpass* filter is one that favors one frequency or a narrow group of frequencies. A series R-L-C circuit such as Fig. 1 is called a *single pole* bandpass filter. It passes one frequency with theoretically zero loss, and provides progressively higher losses above or below that selected frequency.

There are two things you can control. One is the *center frequency* or *resonant frequency*. This is determined by the size of the inductance and capacitance, using the familiar formula

$$f_{res} = \frac{1}{2\pi\sqrt{LC}}$$

The curves in Fig. 1 are shown *normalized* to some center frequency F_0. Double the L-C product, and the frequency gets cut in *half*, and so on.

The other thing we can control is the *bandwidth*. This is also known as 1/Q and is the width of the response between the upper and lower −3-decibel (0.707 amplitude) points. The Q is controlled by setting the *ratio* of *inductance* to *capacitance*. Another form for Q is

$$Q = \frac{2\pi Lf}{R}$$

Bandwidth is normally referred to the center frequency with these two expressions:

Bandwith =

$$\frac{\text{Upper } F_{3DB} - \text{Lower } F_{3DB}}{\text{Center Frequency}}$$

and

Center Frequency =

$$\sqrt{(\text{Upper } F_{3DB})(\text{Lower } F_{3DB})}$$

Bandwidth and Q not only determine the width between the 3-dB points, but they also determine how steeply the curve initially falls off either side of the passband. Note that all the curves end up *eventually* falling off in either direction at a rate of 6 dB per octave (half amplitude as you double or halve the frequency of the stopbands). This ultimate levelling off is caused by the reactance of either the inductor or the capacitor becoming negligible at frequencies well above or well below resonance.

If we want more steepness than we can get with a single R-L-C section, we can cascade several identical sections, perhaps staggering them in frequency to get an overall response shape that is flat or slightly dipped instead of peaked.

There are several problems with the passive circuit. The inductor is usually big, expensive, and not adjustable over a wide range. Secondly, the circuit is load sensitive. Finally, there's a bunch of minor problems such as hum getting into the inductor field, and the difficulty of cascading sections without interaction. To get around these problems, circuit designers have come up with a number of *active filter* circuits that use resistors, capacitors, and operational amplifiers to *simulate* the performance of L-C circuits. One particular circuit of this type is called the *biquadratic bandpass section*. It is shown in Fig. 2.

Unlike many simpler bandpass active circuits, this one lets you *independently* control the circuit gain, the center frequency, and the bandwidth or Q. Gain and Q are changed by changing a single resistor. You can use either one or two resistor adjustments to change the center frequency, and do so independent of Q. The table shows the component values for various Q's and center frequencies.

To trim the center frequency, you vary R2. To change Q, you change R1, and to change gain, you vary R3. The

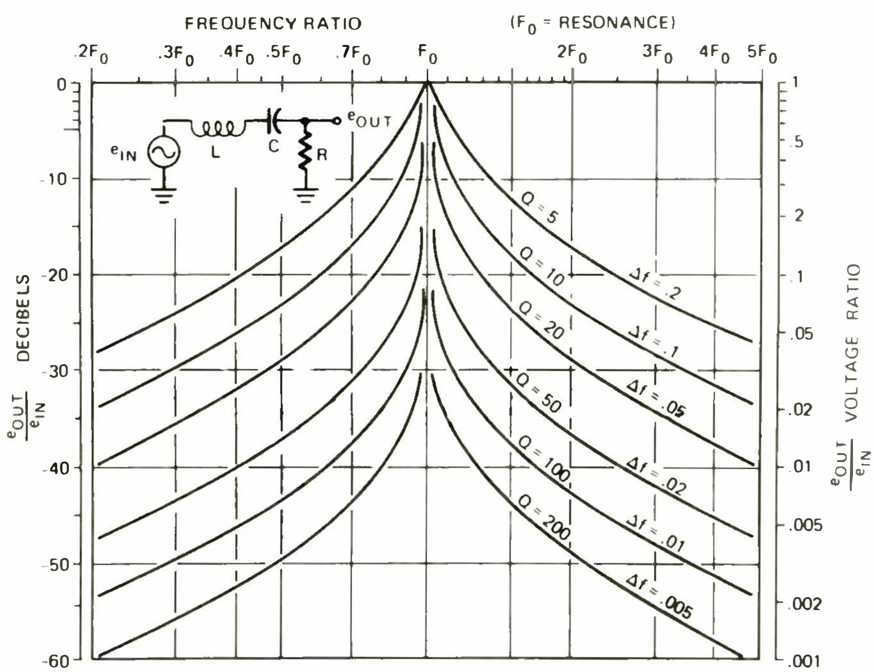

FIG. 1—SINGLE-POLE R-L-C filter and its universal response curves. The bandwidth is determined by the Q of the circuit.

An inductor-free bandpass filter that is easy to work with and control. Use it for electronic music chimes, percussion and bell effects, and audio filters.

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three controls do not interact at all, giving you a simple design and easy, flexible adjustments.

Large frequency changes are made by changing both capacitors (C) simultaneously. You get the best dynamic range and stability by keeping both capacitors and both frequency determining resistors (R2 and R2') *equal* in value. Note that a single resistor frequency adjustment is also non linear—you have to change the resistance 9:1 to get a 3:1 frequency change.

TABLE
Component values versus Q and Frequency

Frequency	C	Q	R1
10 Hz	15 μF	0.5	5 K
20 Hz	7.5 μF	1	10 K
50 Hz	3.3 μF	2	20 K
100 Hz	1.5 μF	5	50 K
200 Hz	.75 μF	10	100 K
500 Hz	.33 μF	20	200 K
1 kHz	.0159 μF	50	510 K*
2 kHz	7500 pF	100	1.2 meg*
5 kHz	3300 pF	200	3.3 meg*
10 kHz	1500 pF	500	10 meg*
(values approximate—rounded to stock sizes)		(values approximate—*influenced by frequency and supply voltage)	

If R3 = 10 K, circuit gain = Q
If R3 = 100 K, circuit gain = Q /10
If R3 = 1 K, circuit gain = 10Q etc . . .

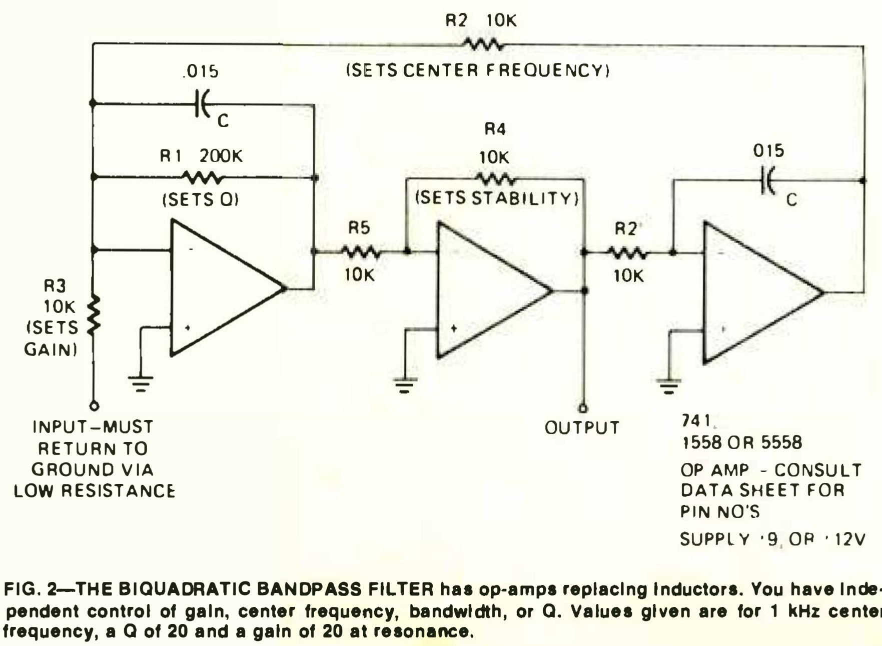

FIG. 2—THE BIQUADRATIC BANDPASS FILTER has op-amps replacing inductors. You have independent control of gain, center frequency, bandwidth, or Q. Values given are for 1 kHz center frequency, a Q of 20 and a gain of 20 at resonance.

With the low-cost 741 and 1558 and 5558 dual op-amps, operation is good to several kHz, even at very high Q's. At low frequencies, stable Q's of 50, 100, and even 200 are easy to get. For higher frequencies and higher Q's, you have to go to a premium operational amplifier such as the LM318.

As long as you use high supply voltages, the circuit performance depends only on the resistors and capacitors, and not on the op-amp or the supply voltage. At very low supply voltages, the circuit can break into sustained oscillation when R1 is very large—this is caused by changing gain and phase shifts inside the op-amps. The effects disappear with 8 volts or more across the op-amp.

Analog computer people will instantly recognize the circuit as an analog of a pendulum. R1 adds "rust" to the hinge and provides damping. For circuit theory people, the transfer function of the circuit is given by the expression

$$\frac{E_{out}}{E_{in}} = \frac{-\frac{1}{R3\times C}\,S}{S^2 + \frac{1}{R1\times C}\,S + \frac{1}{(R2\times C)^2}},$$

$$S = -j\omega = -j2\pi f$$

and is valid at any frequency where you can neglect op-amp high-frequency performance limitations and where R4=R5.

Using it—the biquad

Electronic music is an obviously good place to use this circuit. There are three ways the circuit can be used:

As a *filter*, you can selectively pass certain portions of the audio band, do format filtering, or pick out selected harmonics of a complex waveform. Or you can use it to shape noise into a desired frequency distribution or to emphasize a portion of an audio spectrum.

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As a *"rusty pendulum"*, you can use it as an electronic chime, bell, drum, or bongo. Here, you put a pulse or a very low-frequency squarewave into the circuit, shock exciting the tank, and letting it ring, giving you an exponentially decaying sinewave. For instance, if you drive the Fig. 2 circuit from a 1 hertz squarewave, you ring the chime twice each second, once on the leading and once on the trailing edge of the square wave. If you key your chime, always be sure to eliminate any switch noise and bounce to keep you from getting multiple hits.

As an *oscillator*. If you remove the Q-determining resistor, the circuit will usually be on the verge of oscillation. You can Raise R4 slightly or change the supply to a lower value to get a low-distortion sinewave oscillator. As with any oscillator of this type you or some negative feedback circuit has to ride the gain to hold low distortion and constant amplitude.

Highpass and lowpass functions can also be obtained by adding a resistor or two. but there are no real advantages in the Biquad over simpler active highpass and lowpass circuits. Only in the bandpass case does it really perform.

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