How Active Filters Work

Here are full details on how to build filters with op-amps instead of inductors. This stable and reliable method works for practically all audio and sub-audio low-pass, bandpass, and high-pass designs

by DON LANCASTER

FOR YEARS, PEOPLE WORKING ON CIRCUIT theory have been trying to do-in the inductor. This is especially true in the audio and sub-audio regions where inductors are inherently big, expensive, difficult to adjust, and subject to fields and hum pickup. After a lot of false trys and some rather poor ways of going about this, a batch of solid, reliable methods now exist that can do the job. Almost all these methods use low cost, readily available operational amplifiers. In most of the methods, the energy storage of an inductor is simulated by taking energy from a power supply and delivering it at the right point and in the right amount in a circuit to simulate exactly the bahavior of an inductor.

Actually most methods don't work *directly* on replacing inductors. Instead, they look at the whole picture and attempt to come up with a *functionally equivalent* circuit that does exactly the same thing that the original one did, but internally does it in a wildly different way. These functionally equivalent circuits are often called *active filters,* and an active filter is simply any circuit that uses at least one operational amplifier or its equivalent to simulate exactly a circuit that normally would need at least one inductor to get the same result.

A filter itself is any frequency selective network. Three popular styles are the *low-pass* filter that passes only low frequencies and stops higher ones; the *bandpass* filter that passes only a few or a range of median frequencies; and a *high-pass* filter that allows only high frequencies to reach its output. A rumble filter on a turntable is a high-pass filter. The tuning on an AM radio is a bandpass filter, and the treble cut control on a hi-fi is a form of low-pass filter.

A comparison

Before we go into the nuts and bolts details of how to build your own active filter, let's compare a simple active low-pass filter with an equivalent low-pass passive one (see Fig. 1). And, if we wanted to, we

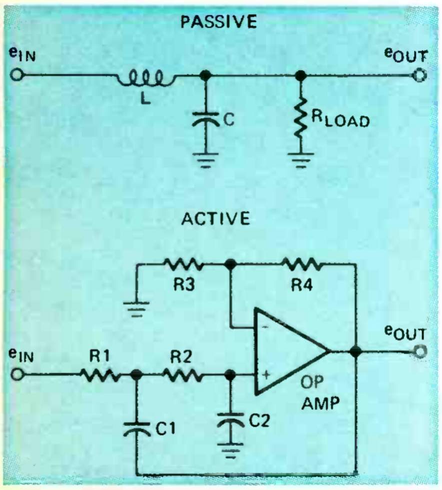


FIG. 1—LOW-PASS FILTERS. The passive L-C type is simpler, the active is more versatile.

could select the L-C ratio of the passive filter or the ratio of C1 and C2 in the active filter to get a response that looks like Fig. 2.

The way we get the response differs for the two circuits, but the result is the same. In the passive filter, the inductive reactance increases and the capacitive reactance decreases as we increase frequency, shunting more and more signal to ground. In the active filter, we essentially have two cascaded R-C sections at very high frequencies that also shunts the signal to ground. The problem is that if we left the amplifier out of the circuit, the response would droop very sloppily and very badly around the cutoff frequency.

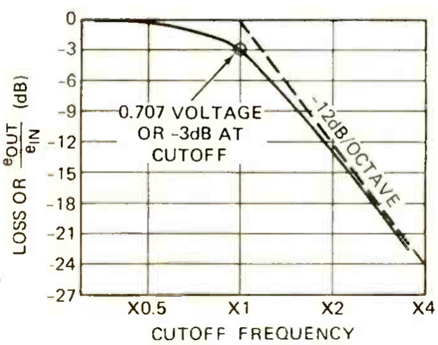


FIG. 2—RESPONSE OF LOW-PASS FILTER. Slope shown is 12 dB per octave.

What the op amp does is use circuit *feedback* via C1. It takes energy from the supply and introduces it in the middle of the R-C network to simulate exactly the same effect as energy storage in the inductor. Thus no R-C network by itself can ever hope to be as good as an L-C one, but an R-C network with some energy feedback controlled by an op amp is another story, and you can replace virtually any L-C network with a group of op amps, resistors, and capacitors. In fact, there's even things you can do actively that you can't with conventional circuits. Gain for instance.

We picked this particular response because it has the maximum possible *flatness* in the passband. It is called a *Butterworth* filter. If we try to steepen the response without adding any more parts, we'd get a hump in the passband, and the size of the hump would decide the *initial* but not the ultimate rate of falloff. Filters with humps are called *Chebycheff* filters if the humps are in the passband and *Elliptical* filters if the humps are both in the passband and the stopband. We could also make the response less flat and more gradual. This would improve the pulse response and overshoot at the expense of tilt in the passband and a more gradual rolloff. The best of these is called a *Bessell* filter. The term that controls the shape of the filter *near the cutoff frequency,* but not at very low or very high frequencies is called the *damping* of the filter. The damping is controlled by the L/C ratio in the passive filter and the C1 to C2 ratio in the active filter, or by holding C1 and C2 constant and changing the ratio of R3 and R4.

We picked the Butterworth here because it is the most popular and the easiest to use. We'll stick with Butterworth filters all the way through this story. Other types are just as easy to build. All you have to do is move the damping and cutoff frequencies around a bit.

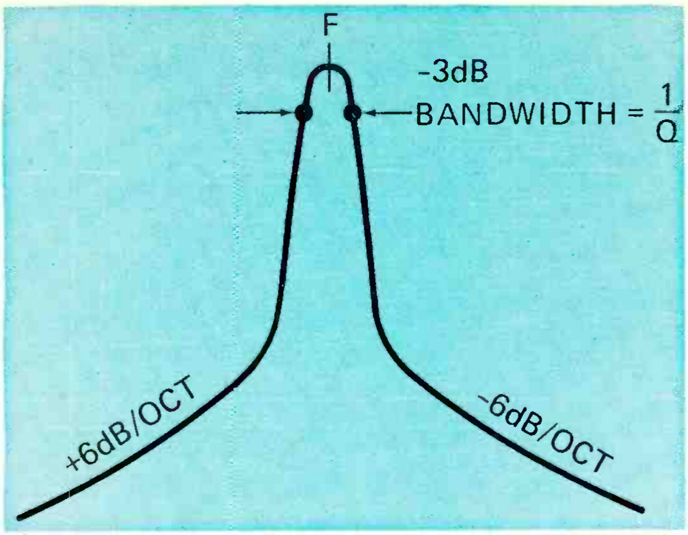
If we wanted something steeper than a 12-dB-per-octave rolloff, we'd have to add more parts. Two inductors and a capacitor would give you a 18-dB-per-octave filter, and two inductors and two capacitors would give you a 24-dB-per-octave rolloff, and so on. We call this the *order* of the filter. Second-, third-, and fourth-order filters have rolloff rates of 12, 18, and 24-dB-per-octave, and are the most popular normally used. Normally it takes one op amp for a second or third-order filter and two for a fourth.

Note that the damping of the filter controls the response near cutoff, particularly the flatness in the passband, the time delay and overshoot, and the *initial* rate of falloff. The order of the filter controls the ultimate or *asymptotic* rate of falloff for the filter.

Why go active?

The operational amplifier serves as a gain block with a very high input impedance and a very low output impedance. Its essential function is to provide for energy feedback to simulate the effect of energy storage in an inductor. Two nice benefits are the ability to drive any load and to use higher impedance (and almost always cheaper) components. So what are the benefits of an active filter? What do we gain and what do we lose when we go active?

The first and obvious thing we lose is the inductor, along with its cost, size, difficulty of adjustment, and sensitivity to hum and other magnetic fields. Note also that the passive filter has a load resistor. The value of this resistor is critical, for if you change it, the relative effects of the reactance changes of the inductor and capacitor



change and the response shape or the cutoff frequency may change. This is not true of the op amp active filter, for the op amp can drive most any reasonable load without changing the filter's response. We can vary the load from an open circuit down to anything the op-amp can reasonably drive without changing the response.

The input to the op amp is a very high impedance. This means you can use high-impedance resistors and capacitors for a given response at a given frequency. The benefits here are obvious. A high-impedance resistor costs the same as a low-ohms one, but a high-impedance capacitor is much smaller, and much cheaper.

Passive filters are inherently lossy, and the best we could expect to hope for would be slightly less than unity gain. With op amps and active filter designs, you sometimes can design for any circuit gain you want. Those we're going to show you have gains above unity.

Another big benefit is tuning. Large variable capacitors are nonexistent, while large variable inductors are expensive and a pain to adjust. On the active side, we have resistors R1 and R2 and surely changing them will change the response. For this particular circuit, we have to change *both* at once to change frequency without hurting the damping and response shape. This is easy to do with a dual pot, and we can easily get at least a 10:1 range. Even for slight tuning adjustments, the resistors are easy to change to get exactly the response you need. Because of this, active filters are generally more tuneable and easier to adjust than passive ones.

A final benefit is a bit subtle, but very important when we want a fancier higher order filter with faster cutoff slopes. We can *cascade* active filter blocks without any interaction, since they are free from fields and mutual inductance and since they generally have a high input impedance and a low output impedance. Cascadeability is a very big benefit. You normally can't simply cascade identical stages, for what was a -3-dB point becomes a -6 and so on. What you do is take the math expression for the higher order filter you want and *factor* it into second-order terms, and then build each second-order term separately. Generally, the individual block responses will be less damped and appear peaked when compared to the final result.

Disadvantages and problems

If active filters are so good, why doesn't everybody use them? First and foremost, it's because very few people understand or appreciate what they are and what they can do. But, over and above this, there are some limitations and disadvantages to their use. Let's take a closer look.

Obviously, we need some supply power and the noise characteristics of the op amp can effect very low-level signals. More important is the high-frequency limitations of the operational amplifier. As you increase operating frequency, an op amps open-loop gain decreases and its phase characteristics change so that you are limited to the upper frequency you can handle with a given operational amplifier.

For amplifiers like the 741 style or its dual and quad combinations, a reasonable upper frequency limit for active filters is between 20 and 50 kHz for low-pass and high-pass versions, and between 2 and 5 kHz for bandpass designs. If you go to a higher performance internally compensated amplifier such as the *National* LM318, you can work up into the hundreds of kilohertz. Finally, if you go to really exotic op amps you can work higher, and even microwave active filter structures have been built. Thus, we have an upper audio limit for active filters built with the cheapest available op amps and a fractional megahertz limit for op amps in the \$3 to \$5 class.

Bandpass filters need more gain for resonance and thus are generally limited to lower frequencies. One way around the problem is to distribute the problem among two or more op amps so that each only has to provide some of the gain.

The low-frequency limit is another story. It's decided mostly by how much you want to pay for big capacitors and how high you're willing to let impedances get. With FET op amps, this can be a bunch, and operation down below 0.1 *Hertz* is certainly possible. Thus active filters are ideal for such sub audio work as brain wave research, seismology, geophysics, and fields like this.

One limitation, and the big one, is called the *sensitivity* problem. You have to ask how the individual components in the active filter are going to change the response if they are out of tolerance or drift with time. For instance, if a particular parameter such as a gain or a capacitance value happens to have a sensitivity of 0.5, the result is a 5% change in cutoff frequency or damping for a 10% change in component value. On the other hand, if a 1% variation makes a 50% change in something you've got problems. This is clearly ungood. When picking a way to build active filters, you have to be aware of the sensitivity problems and how to use them. The method we'll be showing you in a minute is very well behaved at fixed low gains and for lower to moderate Q bandpass designs.

A final limitation is one of method. There are about a dozen good and proven ways to design active filters. These all vary with their ease of understanding and what they can and cannot do. Some can't handle all three basic responses. Some allow single resistor tuning; others allow separate tuning of bandpass gain, center frequency, and Q. Some are well behaved at certain gains, but at others are too highly sensitive or actually unstable. You have to pick a method that works for you, is reliable, behaves well, and

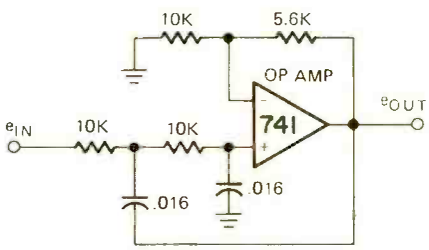


FIG. 3—ACTIVE FILTERS HAVE GAIN. Passive types are lossy circuits.

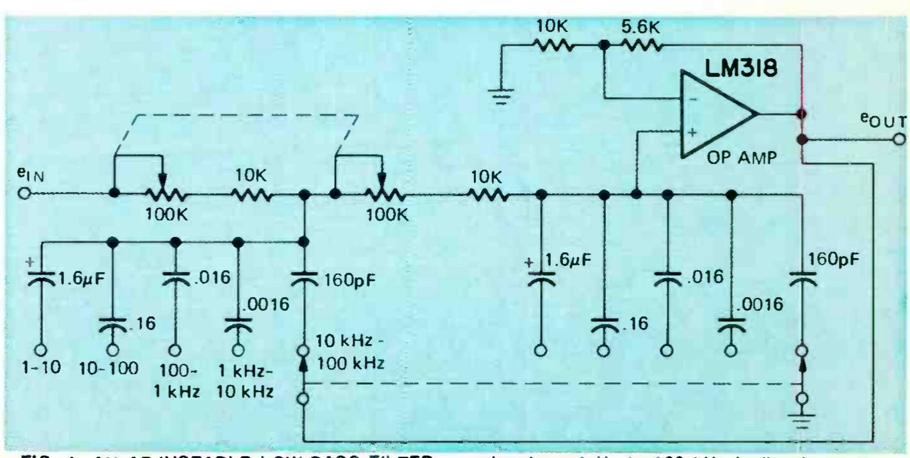


FIG. 4—AN ADJUSTABLE LOW-PASS FILTER covering from 1 Hz to 100 kHz in five frequency ranges. Switched capacitors select the bands, ganged potentiometers do the variable tuning.

does what you want it to. The one we'll be showing you is very easy to understand, stable and forgiving of component variations for fixed low gains, and useable in the bandpass case for low to moderate Q's. It usually takes two resistors simultaneously adjusted to tune, and in the bandpass case, you cannot separately set the Q, gain, and center frequency without a major change in components.

The method is called the *Sallen-Key* or Voltage Controlled Voltage Source (VCVS) method, and first appeared in the March 1955 *IRE Transactions on Circuit Theory*. Other popular filter methods are called the *Integrator Lag*, the Biquadratic *Section*, the *Multiple Feedback*, and the *State Variable*. Another type of active filter uses the *gyrator* or *impedance converter* but these generally take a bunch of parts and have a high-impedance output.

Building your own

So, now we should know *why* we'd want to use active filters, and *where* to go to get complete design details, let's concentrate on *how* to actually build one. Here's a second-order Butterworth low-pass filter with a cutoff frequency of 1 kHz and a gain of 1.6 (see Fig. 3).

The response is identical to the curve in Fig. 2 with f = 1 kHz, 2f = 2 kHz, and so on. As with any low-pass active filter, there must be a low dc impedance to ground at the source. Thus your source has to be less than 10,000 ohms and must provide a route to ground for the op amp's bias current. Again, the response is Butterworth, giving us the flattest possible passband, and an attenuation of −3 dB or 0.707 amplitude at the cutoff frequency, and smoothly falls off at −12 decibels per octave. This means that in the stopband as you double frequency, you get only one quarter the amplitude, and so on.

The above circuit looks deceptively simple and it is except, that a "magic" gain of 1.6 has been used that lets you use equal resistors, equal capacitors, and still have the desired shape. Change anything from the above, and the mathematics behave wildly. The circuit is forgiving of component variations and 5% components should be more than adequate for practically all uses.

To change frequency (in Fig. 1) you change R1 and R2 to identical values, or you simultaneously change C1 and C2 to new values. Raising R lowers the operating frequency. Raising C lowers the operating frequency. Thus, a 5000-ohm value instead of 10,000 ohms puts you at a 2 kHz cutoff frequency, and so on. A 0.032 μF capacitor value puts you at 500 hertz and so on. If you change one capacitor, you *must* change the other. Similarly if you change one resistor, you *must* change the other, or the response shape will also change.

It's easy to see how we can use a dual pot to tune 10:1 and switch capacitors to get decade ranges. Fig. 4 shows a circuit that covers any cutoff frequency you want from 1 hertz to 100 kHz.

The pot rotation will generally be nonlinear since the frequency varies inversely with pot rotation and resistance value. One way to linearize the pot is to use a dual audio log pot, with a normal taper if the dial is on the pot shaft and a reverse taper if the dial is on the panel.

If you just want one frequency differ-

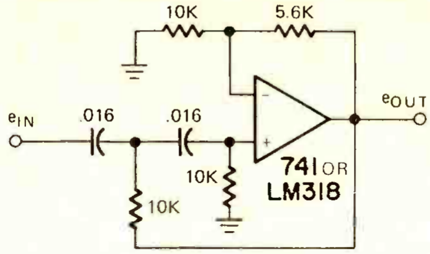


FIG. 5—IN HIGH-PASS FILTERS, the R and C shunt and series elements are transposed.

ent from 1 kHz, just calculate the capacitor value you need and change the capacitors, or change the resistors. It's simply the ratio of the capacitors equals the ratio of the frequencies and vice versa for the resistors.

High-pass designs

The high-pass filter is a snap—you inside the circuit out and by a network principle called *duality* you're done. Fig. 5 is a 1-kHz Butterworth, second-order high-pass circuit.

We can now see another big advantage to the "magic" gain value of 1.6—this circuit lets us switch from highpass to lowpass with a 4pdt switch without any change of component values.

Steeper skirts

We can cascade two low-pass second-order sections to get a fourth-order Butterworth with a 24-dB-per-octave cutoff. We can't use identical sections, but we can make everything identical except for R4 (the feedback resistor) on each section. Finding the right R4 takes a lot of math, but here's the final circuit (see Fig. 6). It has an overall gain of 2.5.

The response is twice as good as before on a decibel scale. The passband is twice as flat and still drops only to −3 dB at the cutoff frequency of 1 kHz. The attenuation drops at 24-dB-per-octave, meaning that every doubling of frequency gives you only one-sixteenth the power and so on.

The higher performance circuit is somewhat harder to tune, since you simultaneously have to change four capacitors or

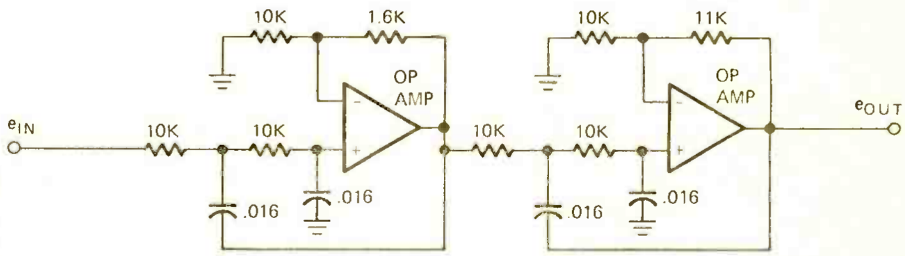


FIG. 6—TWO LOW-PASS SECTIONS IN CASCADE produce a fourth-order Butterworth filter with a rolloff slope of 24 dB per octave. The circuit's overall gain is about 8 dB.

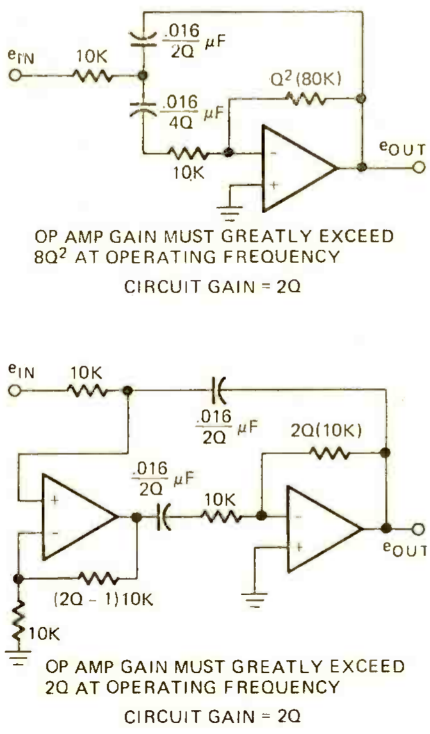


FIG. 7—ACTIVE BANDPASS CIRCUITS require high-gain operational amplifiers.

You simply interchange the resistors and capacitors on the input and you now have a highpass circuit. Again, if you change frequency, change both resistors or both capacitors to identical new values, or else the response shape will also change.

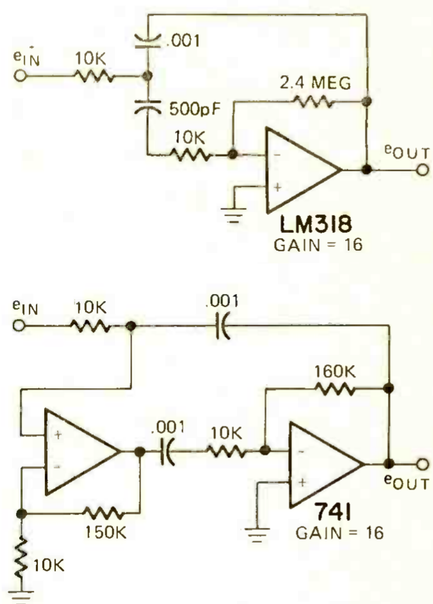


FIG. 8—CIRCUIT CONSTANTS for bandpass circuit where Q is 8 and frequency is 1 kHz.

four pots. Quadriphonic audio pots are a neat way to handle the tuning and they are reasonably available. Highpass to lowpass switching can be handled by a 8-pole-double-throw switch or two ganged 4-pole-double-throw pushbuttons, arranged so one is up when the other is down and vice versa.

Other orders and shapes of active filter

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SERVICING RECORD CHANGERS

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wheel. Replace the retaining C clip and any trim you might have had to remove to gain access to the clip.

18. Refer to Fig. 2 and apply a single drop of lubricant to the over-arm shaft assembly. Work the arm up and down to insure proper lubrication.

OPTIONAL STEPS

19. Place 4 dowels, one under each corner to support the changer slightly off the bench. Connect the universal power cord to the changer and plug into an ac outlet. Place a strobe disc on the turntable and test for correct speed. The pattern should appear stationary.

20. Thoroughly clean the changer using a liquid spray cleaner for metal surfaces, and spray wax for the wooden surfaces.

21. Replace the changer in the unit and performance test. If any problems are noted consult trouble chart (Table II).

The complete changer just described requires about 45-minutes. While these simple overhaul techniques will solve about 80% of your record-changer problems, Table II gives you additional hints for servicing problems. **R-E**

ACTIVE FILTERS

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are just as easy to do.

Bandpass designs

Bandpass filters are generally much harder to design and more subtle to use. About all we have room for here is to show you two circuits that will do the job (see Fig. 7). They're shown for any Q at a center frequency of 1 kHz. And here in Fig. 8 are the same circuits for a Q=8: (1 kHz)

The two-amplifier job requires far less stable gain and works better for higher Q's and higher frequencies. Either circuit gives you the equivalent of a single series "pole" or tuned RLC circuit. This circuit, like its passive counterpart has a nasty feature that you must allow for. Its response starts falling off very steeply either side of resonance, but for very low or very high frequencies, it falls off at a more gradual rate of six decibels per octave. The response shape looks like the diagram in Fig. 9.

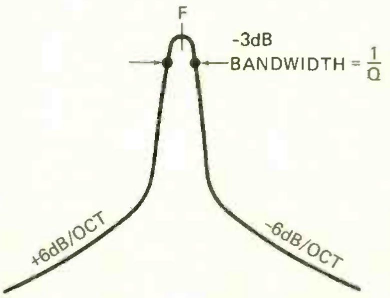

FIG. 9—BANDPASS RESPONSE CURVE measured at 3-dB point depends on circuit Q.

Normally, you cascade several poles to get the desired bandpass response. If we put the poles on top of one another, we get a very sharp response that is not very flat in the passband. We can control the response shape by staggering the poles in frequency and by altering their Q. Spreading the poles flattens out the passband, until finally you get a dip in the middle if you go too far. Another more formal way to design is to build a lowpass filter that does the job you want and then use a math process called *transformation* to get the desired bandpass shape. When you only need two poles, the simplest thing is to sit down with a breadboard and experiment with the Q and staggering for the response you need (the circuit moves around just like the lowpass and bandpass ones do by simultaneously changing capacitors or resistors); this is also a trivial problem for any computer that speaks BASIC, but the math is a bear otherwise. That's about all the details on bandpass design we have room for here. If enough readers are interested, we can put together another story with complete design curves for the two-pole bandpass designs in some other issue. **R-E**

BUILD IN ACTIVE FILTER

Next month in Radio-Electronics Don Lancaster presents complete details on how to build an active filter to meet your own needs. Don't miss it.