FEATURE

A Practical Approach to Active Filters

Noodling out an active filter with available hardware.

GEORGE HYLTON

Active filters are all the rage now days. For the experimenter, however, there's a bit of a problem. The texts about them seem to come in two varieties, neither of which is very helpful.

One is full of complex maths and short on component values. The other gives component values, but for filters which never seem to be quite what you need.

Practical Case

It so happened that I needed a decent low-pass audio filter recently. I'd been working on a simple short-wave receiver. The RF front end part of the design was finished and I now needed an audio section.

Short-wave broadcast stations are packed like sardines, often only 5kHz apart. Reception is often noisy. Simple receivers of the direct conversion or synchrodyne kinds (mine is both) convert adjacent-channel signals into noise, mostly high pitched.

A good low-pass audio filter is needed to reduce this "side band splash". Ideally the filter should have a variable cutoff frequency so that it can be adjusted to suit the reception conditions of the moment. None of my books and magazines had a ready-made answer. I was stuck.

An Unusual Component

Browsing through a component store, I found an unusual component: a quad (four-gang) 50 kilohm potentiometer. Dual (two-gang) pots for stereo are common enough. Quad pots, presumably for quadraphonics, are rare.

I figured that with a quad pot I could make a four-section variable cut-off low pass −RC filter (Fig. 1). With −R variable I should get at least a ten-to-one range of cut-off frequency, more than enough for speech and music and maybe of some use for CW.

So I bought some "quad pots". They turned out to be neat little Japanese jobs. Ohmmeter tests showed that they were log law, and actually about 45k max.

Would they do the job? I assembled the filter on a plug-in breadboard, using 4n7 capacitors for *C*. Why 4n7? Well, I happened to have plenty of that value, but I did make a quick check with a nomogram which showed me that 4n7 has a reactance of 45k at about 760Hz.

The -3dB cutoff frequency of a single *RC* section falls at the point where the reactance of *C* equals *R*. With four sections it would be lower in frequency, but at least I was in the right area. With the pot set near minimum resistance the cutoff would be at least ten times higher, at 7.6kHz, which was about as much as I needed.

The next job was to hitch my audio generator to the filter input and set *R* to give a practical cutoff frequency. I chose 3kHz, which is the sort of cutoff you need when interference is bad.

The response turned out to be as shown in curve *A*. Not bad, but a bit droopy. Could it be made flatter in the pass-band and steeper beyond it?

Phase Shift Oscillator

I've always found oscillator circuits interesting, and I knew of one which can use exactly this sort of *RC* lowpass network for turning. The circuit block diagram is shown in Fig. 2. Note that the amplifier is inverting, as indicated by the minus sign in front of the gain symbol, A.

At frequencies well below cutoff the feedback through the *RC* network is negative. At DC, all the amplifier output is fed back negatively to the input and the gain is effectively one.

As the frequency is raised, the effect of *C* becomes significant. From Fig. 1, curve *A*, it's clear that *C* produces attenuation. But it also produces phase shift. This means that the feedback isn't quite so negative, so the gain isn't reduced as much as expected.

At one frequency, the phase shift is -180°. That is, the phase is inverted by the network. So there are now two phase inversions (one in the amplifier, one in the network), which means that the overall feedback becomes positive. If the gain (-*A*) *is high enough, the circuit oscillates.*

Using a double-beam oscilloscope to compare input and output signals it was easy to adjust the frequency of my audio generator to get a shift of 180° from my *RC* lowpass. I found that the output signal was then about one sixteenth of the input.

This meant that in Fig. 2 if the amplifier gains exceeds 16, the circuit will oscillate. For gains a bit short of 16 it won't, but,

a peak will appear in the response. Clearly, the peak will get sharper as the gain is raised towards the oscillation point and less sharp as it's reduced.

There seemed to be a fair chance of finding a gain at which the response is reasonably level, up to a frequency somewhere near the 180° one. Beyond it the gain must drop sharply, for two reasons. First, the attenuation of the network increases faster than the amplifier can compensate. Secondly, beyond the 180° frequency the feedback becomes less positive.

At very high frequencies each section must have a phase shift of nearly 90°, giving a total network phase shift of 360°. The feedback is then negative.

Bench Test

Theorizing is all very well, but does it work? Next step: try it and see.

The "circuit" in Fig. 2 is just an aid to understanding. It has no provision for applying input signals.

After a good deal of doodling I arrived at the practical test circuit of Fig. 3. Here, transistor TR1 is just an emitter-follower input buffer. The voltage gain comes from transistor TR2 and is about 8. TR3 is an output buffer.

Adding the input signal to the feedback is arranged for by resistors R1 and R2. At very low frequencies the gain is mainly defined by these resistances, which form a negative feedback network.

If transistor TR2 had infinite gain then the effective very-low frequency gain would be R2/R1 = 1.5. But since the actual gain of TR2 is low the real IF gain is less than 1.5. In fact, resistor R2 was selected by trial and error to set the gain as close to one as possible using E12 resistances. (It's a little over one, in fact.)

At higher frequencies, where the *RC* phase shift makes the feedback more positive the gain of TR2 has much more influence. To adjust it I used various values for resistor R4 until I found one (82k) that gave the flattest response, plotted in Fig. 1 as curve *B*. to make this comparable with *A*, the network resistances *R* were adjusted to give the same -3dB point, 3kHz. The improvement is obvious.

Having produced a useful-looking 3kHz lowpass filter, the next step was to vary *R* and confirm that the response keeps the same general shape but with different cutoff frequencies. The lowest obtainable cutoff (-3dB) proved to be 560Hz. The highest I checked was 10kHz: beyond that was of no interest to me.

In all cases the response was like curve *B*: fairly level in the pass band and fairly steep in the stop band. Very satisfactory, considering that I'd done no maths and, used no unusual or close tolerance component values (the 4n7 capacitors were 10 per cent).

Also, the filter has equal values of *C* and equal values of *R*. My research through the literature turned up designs where if the *Rs* were equal the *Cs* were not, and vice versa.

I was beginning to get quite smug about it when I ran a test which showed that one of my tacit assumptions was quite wrong: the response at the 180° frequency was well down. I'd assumed that the 180° frequency would lie in the passband, not outside it.

Fixed Filters

If you want to use fixed values of *R* and *C* and don't want to resort to cut-and-try you need more information. How much? The essentials seem to be *C*, *R* and -3dB frequency for one filter. From these it should be possible to estimate the values for other filters.

I set up my circuit using fixed close tolerance components: *R* = 10k, *C* = 10n. These gave a -3dB response at exactly 1kHz.

Very convenient. If either *C* or R is increased the cutoff frequency is decreased. The response, then, is inversely proportional to *C* times *R*.

My 1kHz filter has *CR* = 100, if *C* is in nF and *R* in kilohms. This suggests a simple design formula: *CR* = 100/fc, where fc is the -3dB frequency in kHz, *C* is inF and R is in kilohms.

Thus for a 4kHz filter *CR would be 25. If you happen to have plenty of one nanofarad capacitors then R* needs to be 25 kilohms. If you use 22k the bandwidth will be a bit more than 4kHz; with 27k it will be a bit less.

This is all you need to design your own "active" lowpass filter. Well, not quite. You have to make sure that the filter impedance is compatible with the circuit in which you connect it.

The network should be driven from a source whose impedance is much less than *R. It should be terminated by an impedance much greater than R.*

My circuit should work for most practical values, provided that it is driven from a source impedance small compared with resistor R1 (if not, reduce R1 to keep it, plus the actual source impedance equal to 100k approx.) Also, the load connected to the output (capacitor C2 and ground) should be at least 10k.

Any high gain audio transistors will do. ■

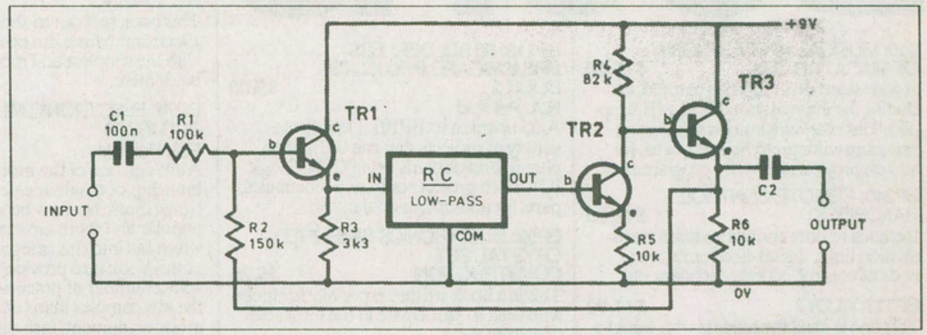

Fig.3. Circuit diagram for a practical lowpass active filter embodying a four-section RC network with equal C and equal R.

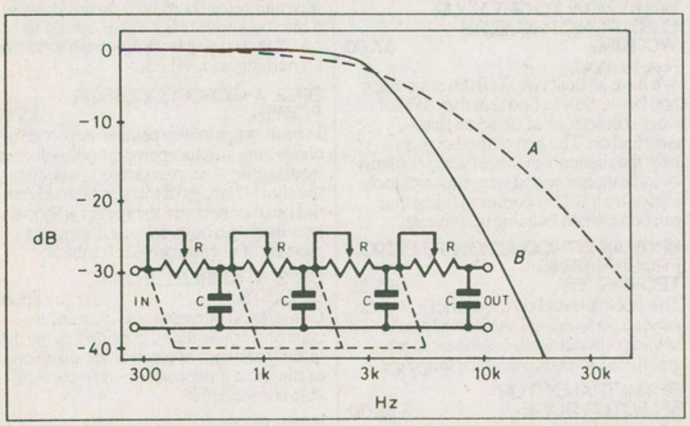

Fig. 1. Four-section RC low-pass network. Curve A shows the response of the network alone for values of R and C which produce a -3dB point at 3kHz. Curve B is for an active filter with a similar net-

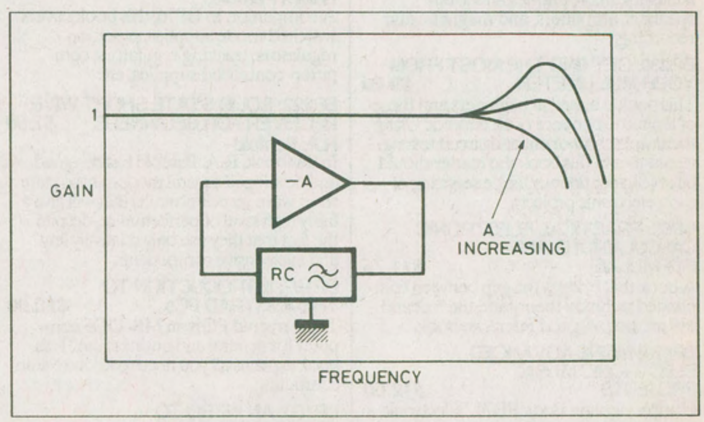

Fig. 2. When an RC lowpass with three or more section is connected as a feedback path in an inverting amplifier the frequency response becomes very dependent on the gain when the phase shift of the network is close to 180°.