DESIGNING & USING ACTIVE FILTERS PART 3

Concluding our detailed examination of this particular building block Tim Orr takes a good look at band pass and band reject circuits.

Band reject (notch) filters

SO FAR NOTCH FILTERS have been realised in this article by two methods; by mixing a bandpass signal with the original or by mixing the low and high pass outputs. There are of course, many other methods of obtaining a notch.

Firstly, the Twin T circuit, Fig. 1.

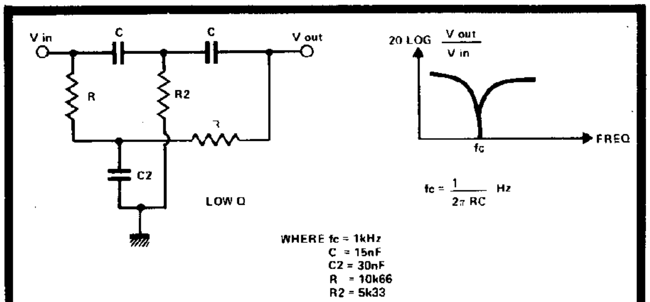


Fig. 1: Twin-T band notch filter configuration.

This is interesting, in as much as by using only resistors and capacitors, a notch response can be obtained! However, as this filter is passive, only a low Q is possible. This circuit is not used very much, because it has *six* components that determine its notch frequency. However, it is of interest to note that, when the Twin T is placed in the feedback loop of a high gain inverting amplifier, a bandpass response is obtained. Also if R is made variable it is possible to move the centre frequency, although in doing so, the Q varies. This has been the basis of many Wah Wah effects pedals, Fig. 2.

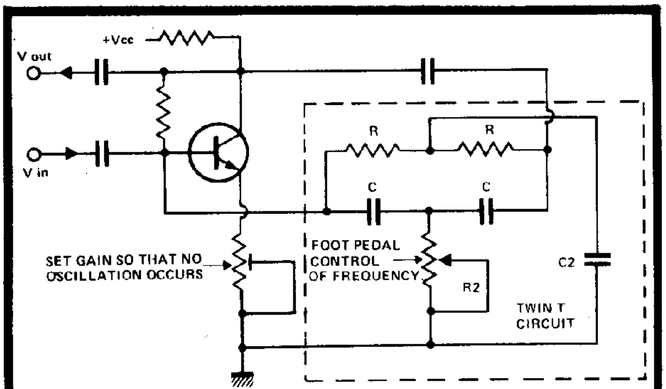


Fig. 2: Block Diagram of a typical Waa-Waa Pedal.

Another method of obtaining a notch is to use the 'Allpass' filter, Fig. 3. The frequency response shows that its output is flat! Not much of a filter I hear you saying. However, it suffers a phase shift which goes from 180 , through 90 at fc, to 0 . By cascading two of these filters, the phase shift is doubled.

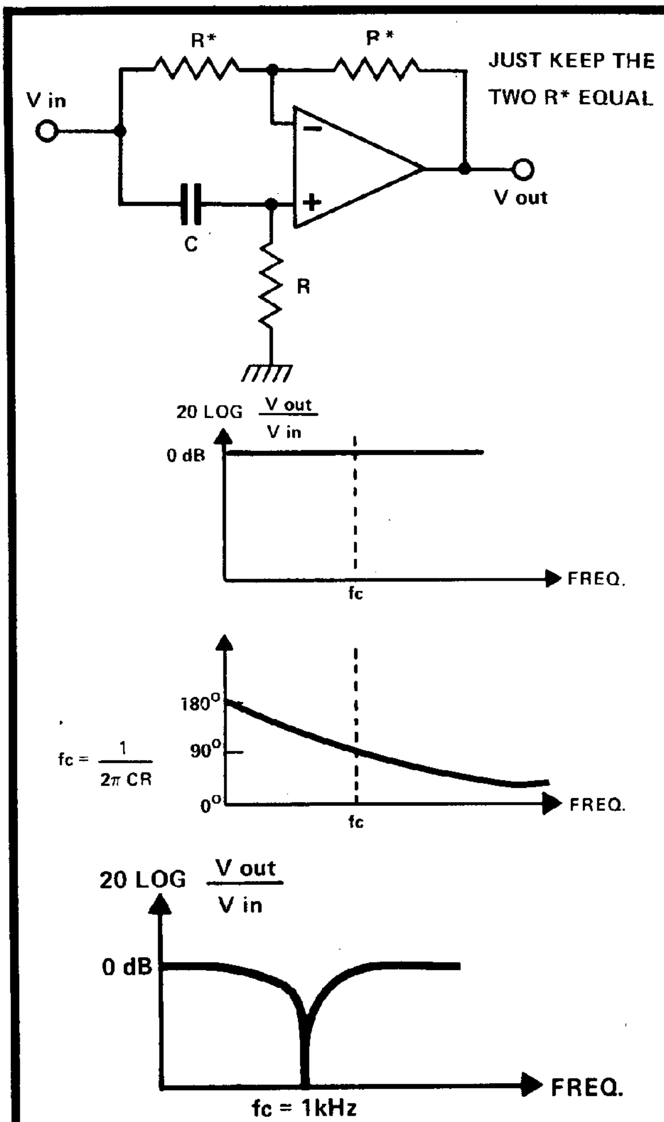


Fig. 3: All-pass filter. At the top is the circuit for such a device. Its frequency and phase responses are shown below it, with the obtainable notch at the bottom.

If we then mix the phase delayed signal with the original, a notch response is obtained. This is because at fc the two signals have the same magnitude, but the opposite phase and so they cancel each other out.

If the notch is to be made tuneable, then the RC time constants must be varied. For a small tuning range just one R can be varied, for a large tuning range then the R's must be realised with a 'stereo' pot.

All change

If lots of Allpass filters are cascaded then several notches can be produced. This type of filter is known as a comb filter. Note that it takes two Allpass filters to produce a usable 180° phase shift, and therefore every notch in the comb requires two Allpass sections. By making the R's variable then the notches can be made to move up and down in frequency. This filter forms the well known 'phasing' effect unit, widely used in the music industry to produce colouration!

Fig. 4 shows a small section of just such a unit. A CMOS chip is used to provide a MATCHED set of six MOS FETS. A common voltage is used to control the MOS FETS channel resistance. Thus as the control voltage varies then so do the six MOSFET resistors, and the three notches move in unison along the frequency axis.

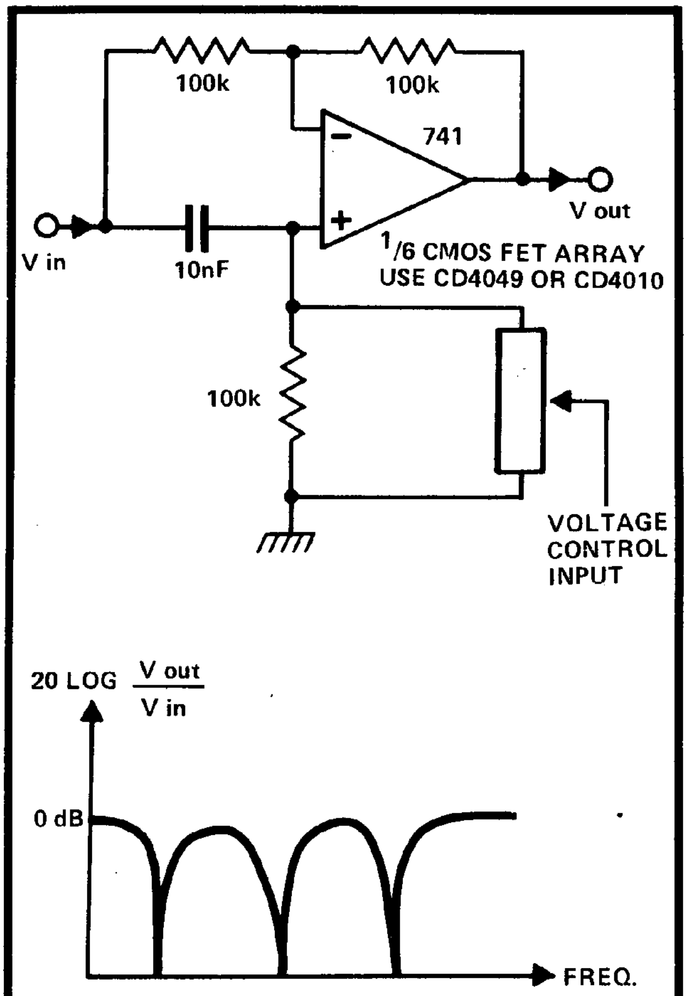


Fig4: One section of a comb filter. The response produced by the full (six times above) circuit is shown below the circuit.

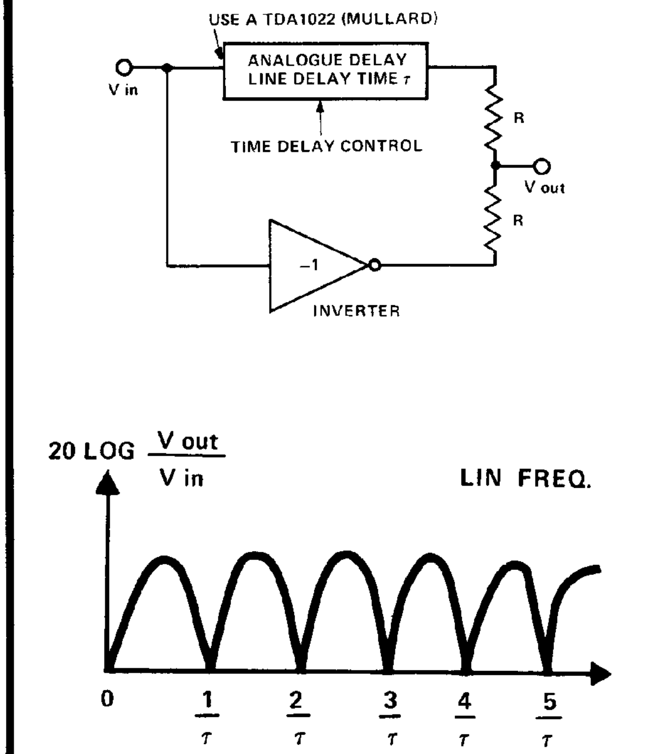


Fig. 5: Alternative method of producing a comb filter using a Mullard delay line.

Another form of comb filter is shown in Fig. 5.

Instead of a phase delay line, a time delay line is used. This produces a large number of notches which are linearly spread along the frequency axis. Their spacing being determined by the delay time.

A bucket brigade delay line can be used to implement the time delay and this can be made variable. This type of filter is known as a Flanger, which is a superior type of phasing unit, and is used to generate high quality phasing effects. An even more impressive sound can be produced by adding some feedback around the delay line. A multi peak, high Q filter is formed which makes very interesting musical sounds when swept.

Variable Tuning

Very often a variable centre or cut off frequency is wanted. This causes problems in filters of order greater than two, simply because getting ganged potentiometers with more than two sections is difficult. One well known manufacturer uses four presets mounted on a common spindle to produce a fourth order Rumble and scratch filter. For manually controlled filters, the resistors are made variable by using ganged potentiometers or switched resistor networks.

The switches can be mechanically operated or electronically controlled, Fig. 6. An alternative method of switching control is to use mark/space modulation, Fig. 7. This has the advantage of being a continuously variable control with a useable sweep range of 100 to 1. Also, lots of sections can be used, and they will *all* track. Therefore, if two CD4016 packs were used, then an eighth order 4 transmission gates per pack), variable frequency filter could be constructed. There are of course, problems;

ACTIVE FILTERS

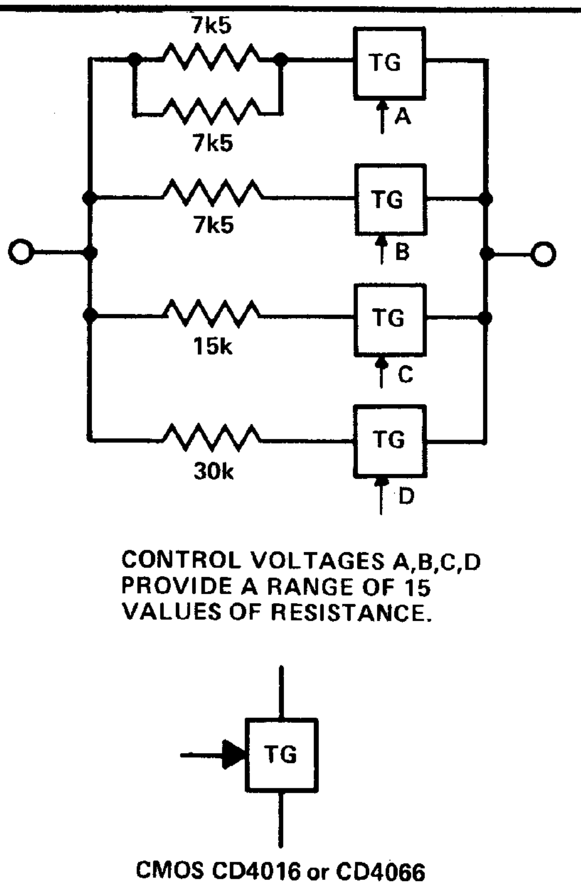


Fig. 6: Varying the tuning of an active filter by use of CD4016 transmission gates (TG) to switch in different resistor values.

1. The switching waveform must be several times higher in frequency than the highest frequency to be filtered.
2. More circuitry, to generate the switching waveform is required.
3. Switching noise is generated.

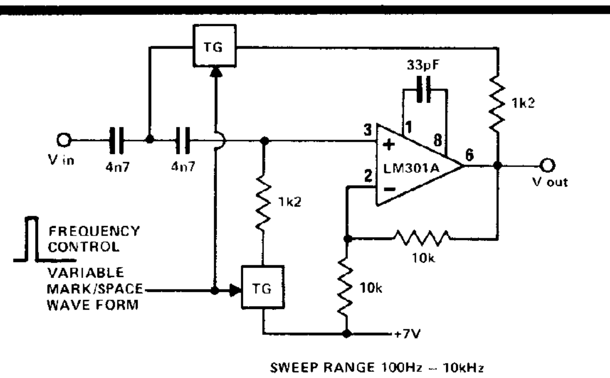


Fig. 7: Another method of varying the notch frequency, mark/space ratio modulation, and has the advantage of possessing a wide range.

Multiplying FETS

Voltage controlled resistors can be used. These take the form of junction or MOS FETS, where the gate voltage controls the channel resistance, Rds. The problems with this method are that the characteristics from FET to FET vary considerably and also the RDS does not have a predictable relationship to the gate voltage. Also, to avoid distortion, low signal levels must be used. Nevertheless, FETS are used in many variable filters such as phasing units.

A set of six MOS FETS having matched characteristics can be obtained from a CD4049 or a CD4010 pack. Alternatively LED photo conductor arrays can be used. The LED produces light which controls the photo conductor's resistance, the two devices being housed in a lightproof box. Large signals can be handled with very low distortion and low noise generation.

Again there are drawbacks. The units are quite expensive, the relationship between LED current and photo conductor resistance is rather unpredictable and the photo conducter's characteristics drift. Another method of varying a filter frequency is to use electronic multipliers. A two quadrant multiplier function can be used to vary the gain of a stage and so produce frequency scaling.

Some Audio Circuits

Active filters have found great use in equalising audio signals, from tone controls on a domestic Hi-Fi to Parametric equalisers in recording studio. Fig. 8

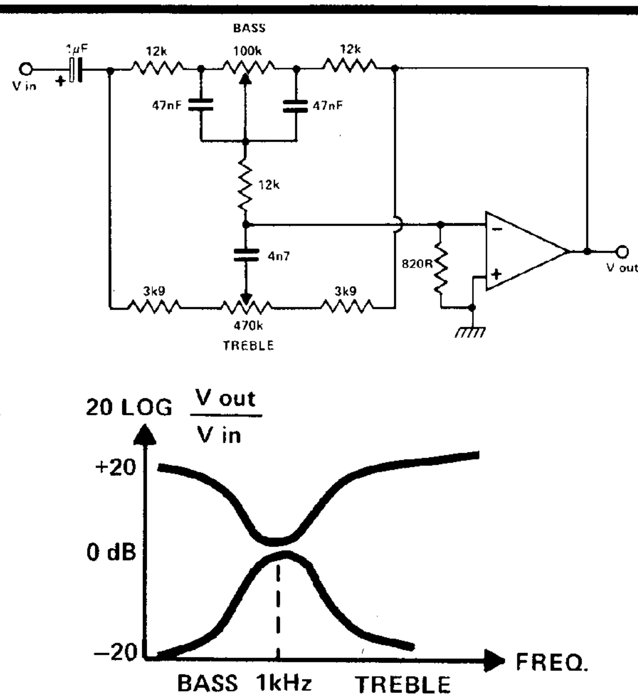
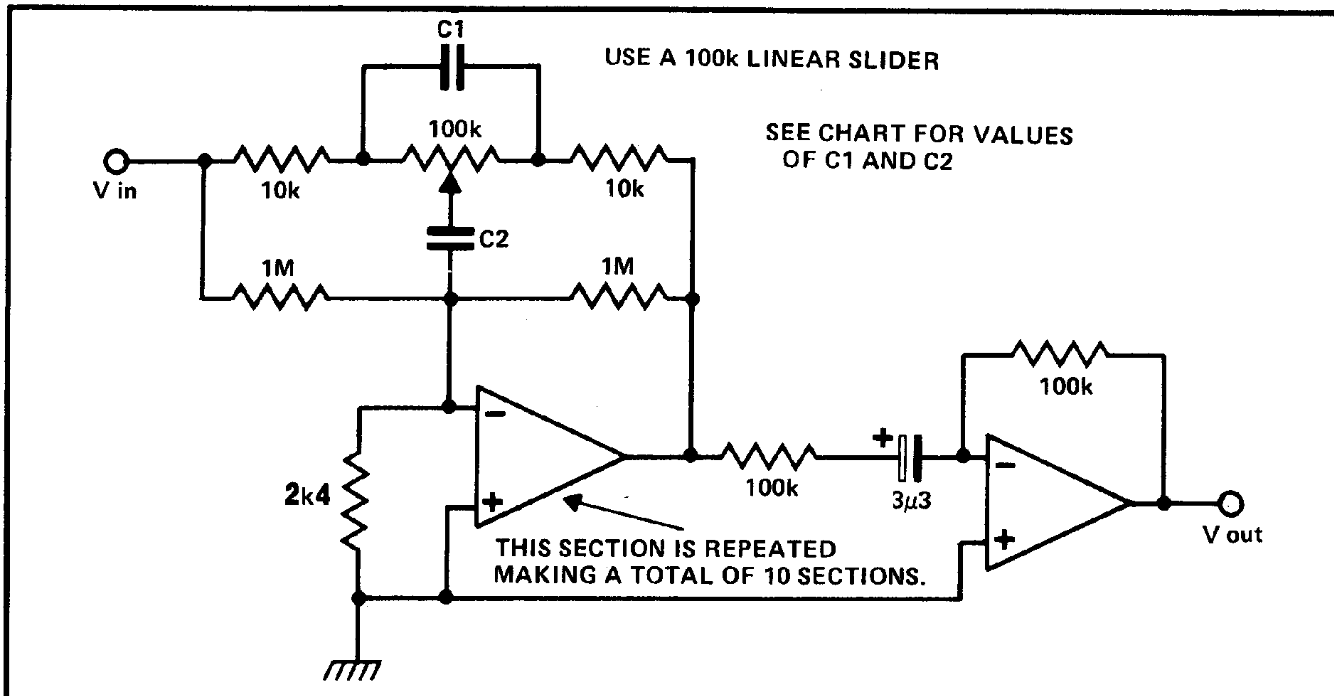
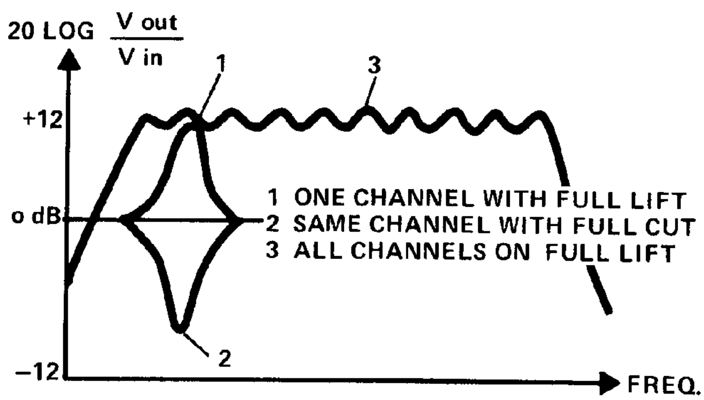


Fig. 8: Simple tone control circuit, with the lift and cut responses shown beneath it.



CHANNEL CENTRE FREQ. IN Hz	C1	C2
32	180n	18n
64	100n	10n
125	47n	4n7
250	22n	2n2
500	12n	1n2
1000	5n6	560p
2000	2n7	270p
4000	1n5	150p
8000	680p	68p
16000	360p	36p

Fig. 9: A design for a graphic equaliser, using active filters. The above circuit is repeated however many times you wish. Use the table on the left to calculate values.



shows a common tone control with just bass and treble functions. Again cut and lift ranges are 20dBs. If a more flexible control of the spectrum is needed then a ten band graphic equaliser (Fig. 9) could come in handy.

Testing Designs

Once the process of designing active filters has been reduced to a simple procedure, testing them should also be made as easy as possible. The most basic is to use a swept sinewave oscillator (Fig. 10).

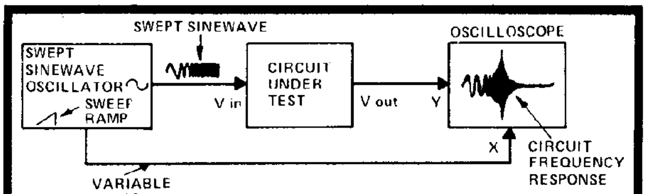


Fig. 10: Testing active filters using a sweep oscillation.

An XY oscilloscope is used to display frequency log) against amplitude linear). The ideal display would be log. amplitude, but this is not so easy to obtain. The beauty of this method of testing is that the display is real time and so any changes made to the filter, like varying one of the capacitors, appear instantly on the oscilloscope. If high Q's or rapid roll offs at low frequencies are involved, then the sweep time will have to be reduced, otherwise the effects of Ringing, will 'Time smear' the display. The harmonic distortion of the sinewave can be quite large, 0.5 to 2.0% without causing *too* much of a display problem for most filter designs. ●

TIM ORR, THE AUTHOR OF THIS SERIES, IS EMPLOYED BY EMS LTD IN THE DESIGN OF AUDIO EFFECT CIRCUITS.