AUDIO OUT

By Jake Rothman

Variable audio filters – Part 2: voltage-controlled filters

Last month, we looked at *manually* controlled variable corner frequency filter circuits that used potentiometers to change a filter's parameters. This month, we'll look at *voltage*-controlled filters (VCFs), which are the most important processor circuits in sound synthesisers. In VCFs a voltage rather than a change in a physical resistance controls the corner frequency.

VCFs use the same technology as voltage-controlled amplifiers (VCAs) which are normally used to control gain or volume. The simplest versions use an LED shining on light-dependent resistors, FETs and diodes, which effectively give voltage-controlled resistance. The problem with this technique is that there are large individual variations in resistance and control curve with such devices. The tracking between the elements is worse than that between potentiometer sections. Better, but more complex control devices use the variation of transconductance with current exhibited by bipolar transistors in fully balanced or differential configurations.

Breakthrough

Accurate balancing is important to prevent the control voltage itself from appearing on the output. This is called 'control voltage breakthrough' and can manifest itself as thumps when the control voltage is rapidly changed. Since the control voltage is normally in phase with the signal it has to be cancelled by a differential amplifier. Transistor matching of V_{be} and H_{fe} is important to ensure balance in these circuits. This is where chips and dual transistors have an advantage because the transistors are fabricated on the same piece of silicon and hence accurately share semiconductor characteristics.

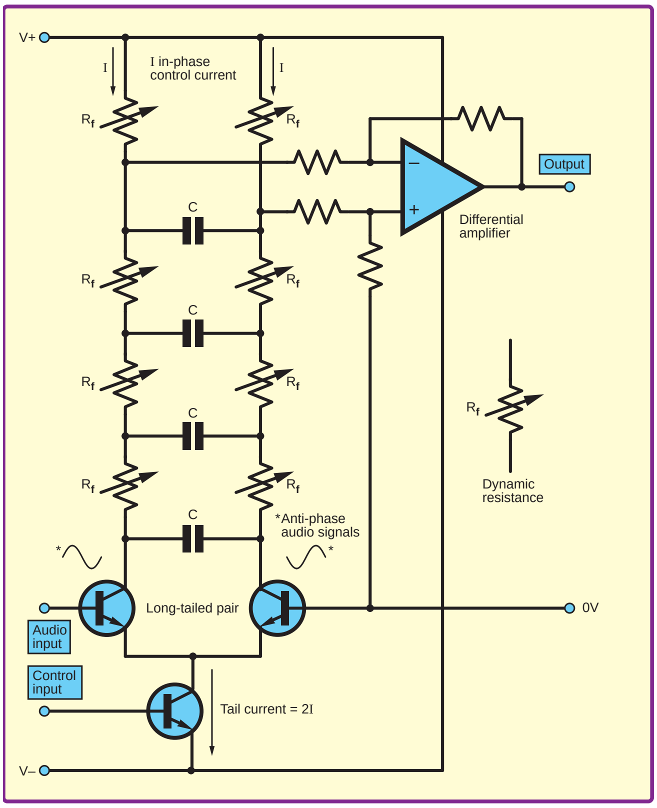

Fig.18. Structure of a ladder filter.

Bob Moog filter

The Moog synthesiser filter must be the most famous filter in music history. It has a unique circuit design and sound, which is possibly why it was the only part of Moog's Minimoog synthesiser that was patented (in 1966). (A little piece of *Audio Out* trivia – 'Moog' is pronounced 'mohg' with an 'oh' sound, not a bovine 'moo'. I discovered this from the horse's mouth when my friends and I were corrected by the late Robert Moog at a Theremin convention in 1996.)

Moog's circuit evolved from Blumlein's long-tailed pair differential amplifier. The two collector-load resistors are replaced with a resistor/capacitor ladder shown in Fig.18. The output is derived from a differential amplifier placed at the top of the ladder.

To vary the frequency, the value of all the ladder resistors has to be varied simultaneously. This is achieved using a circuit element to replace the resistors whose resistance varies with current. By modulating the current though the 'tail' (the bottom common-emitter resistor of the transistors) the current through both sides of the ladder is controlled, thus varying the cut-off frequency. The simplest component for this job is a diode, whose non-linear forward resistance is dependent on the current flow – see Fig.19. This resistance is the forward voltage (V_f) divided by the forward current I_d, often called 'dynamic resistance'. This is not very consistent between individual diodes, so instead, Bob Moog used individually biased transistors.

Control breakthrough is eliminated by cancellation in the differential output stage. The filtered signal is amplified rather than rejected because the audio outputs of the ladder are out of phase.

In music synthesis fourth-order filters are normally used to get a more dramatic effect, which means four capacitors are needed, along with eight transistors. Fig.20 shows a Moog ladder filter and a prototype built on Veroboard (Fig.21). Note the use of a CA3046 transistor array to ensure the lower and upper long-tailed pair transistors are matched.

The Gen X-1 I recently designed (Fig.22a) uses the cheaper relation of the Moog ladder filter, the diode ladder filter. The famous 1974 *Practical Electronics* Minisonic and Gakken SX-150 synthesisers also use this circuit. In the Gen X-1 filter, the diodes are transistors wired as diodes, which exhibit a more consistent forward voltage. The top of the diode ladder must be isolated from power supply

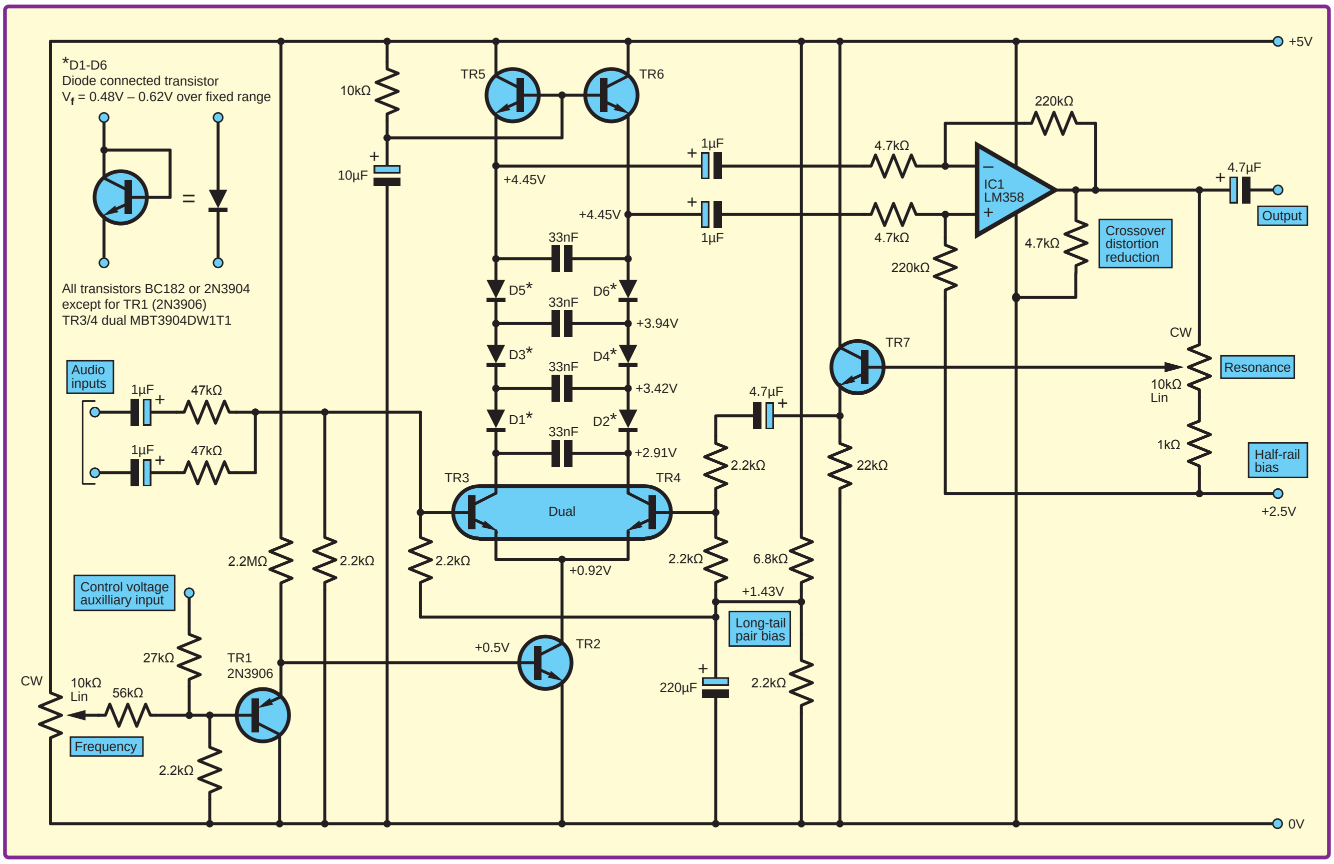

Fig.19. Diode ladder filter used in the Gen X-1. This actually uses transistors wired as diodes, since the matching of the forward-biased resistance characteristic is better.

noise, hence the use of a pair of emitter followers built around TR5 and 6 with decoupled biasing. The construction is surface-mount, where the devices cost pennies, and is shown Fig.22b. The lower long-tailed-pair transistor is a dual version of the 2N3904, which costs around 10p and ensures the ladder currents are equal. The moral of this story is that it's now cheaper to build analogue circuitry with discrete surface mount technology than using single-sourced chips.

Positive feedback

For musical synthesis, it is useful to be able to put plenty of positive feedback around the filter to make it highly resonant, often to the point of oscillation. This gives synths their characteristic 'wah' sound as the filter opens up. Ladder filter circuits do this very well because the gain is reduced proportionally as the Q increases. The circuits also soft-clip and there is intermodulation between the input signal and the filter resonant frequency in a unique manner that is recognised as the 'Moog sound'.

Transconductance op amp filters

There are a lot of components in a ladder filter, which makes them difficult to assemble by hand, especially on stripboard. The solution is to use an IC, which will reduce the parts count considerably. The most popular devices for filters are operational transconductance amplifiers or OTAs. They are basically op amps with a gain control pin. The output is a current output (hence the current source in the symbol) as is the gain control, which is a current input. The output current is the product of the input voltage times the control current, thus giving gain control. OTAs are normally used open loop; that is, with no negative feedback. If negative feedback were used it would act to even out the desired gain variation. The resulting high gain of open-loop operation means only about 50mV peak-to-peak can be applied to the input before clipping occurs. This means an attenuator has to be inserted on the input.

The most popular OTA chip in the heyday of the analogue synth, the 1980s, was the CA3080 and the famous Oberheim synthesiser used two of them for its filter. Unfortunately, transconductance amplifiers such as the CA3080, CA3060 and the higher quality CA3082 are now expensive because they are no longer made and eBay sellers are taking advantage of demand and scarcity. The chips used the now obsolete 7µm-fabrication process. It seems there will be no new plant commissioned, a disaster for synth builders. However, for the moment, you can buy the LM13600/700, which is still available from National Semiconductor and JRC. This chip consists of a couple of 3080 masks combined with a pair of Darlington transistors available for use as buffers. This enables a whole filter to be built with one chip. There are also linearising diodes on the inputs that reduce the distortion. These work by generating a distortion opposite to that produced by the chip's input transistors. This cancellation only works up to a certain point, at which point the distortion suddenly returns.

An LM13600 OTA filter circuit is given in Fig.23. It is basically a form of state variable with band-pass and low-pass outputs. The output current can feed directly into a capacitor, forming an integrator. A buffer stage is then required after the capacitor. This block can then be put into the state-variable filter topology. With the CA series chips, the buffer can be an op amp wired as a voltage follower or a JFET source follower. The Darlington buffers in the LM13600 have their operating current linked to the control pin current to maintain a higher input impedance. In theory, this should be an advantage for filters since the drain on the integrating capacitor should be less at low control currents. The LM13700 chip does

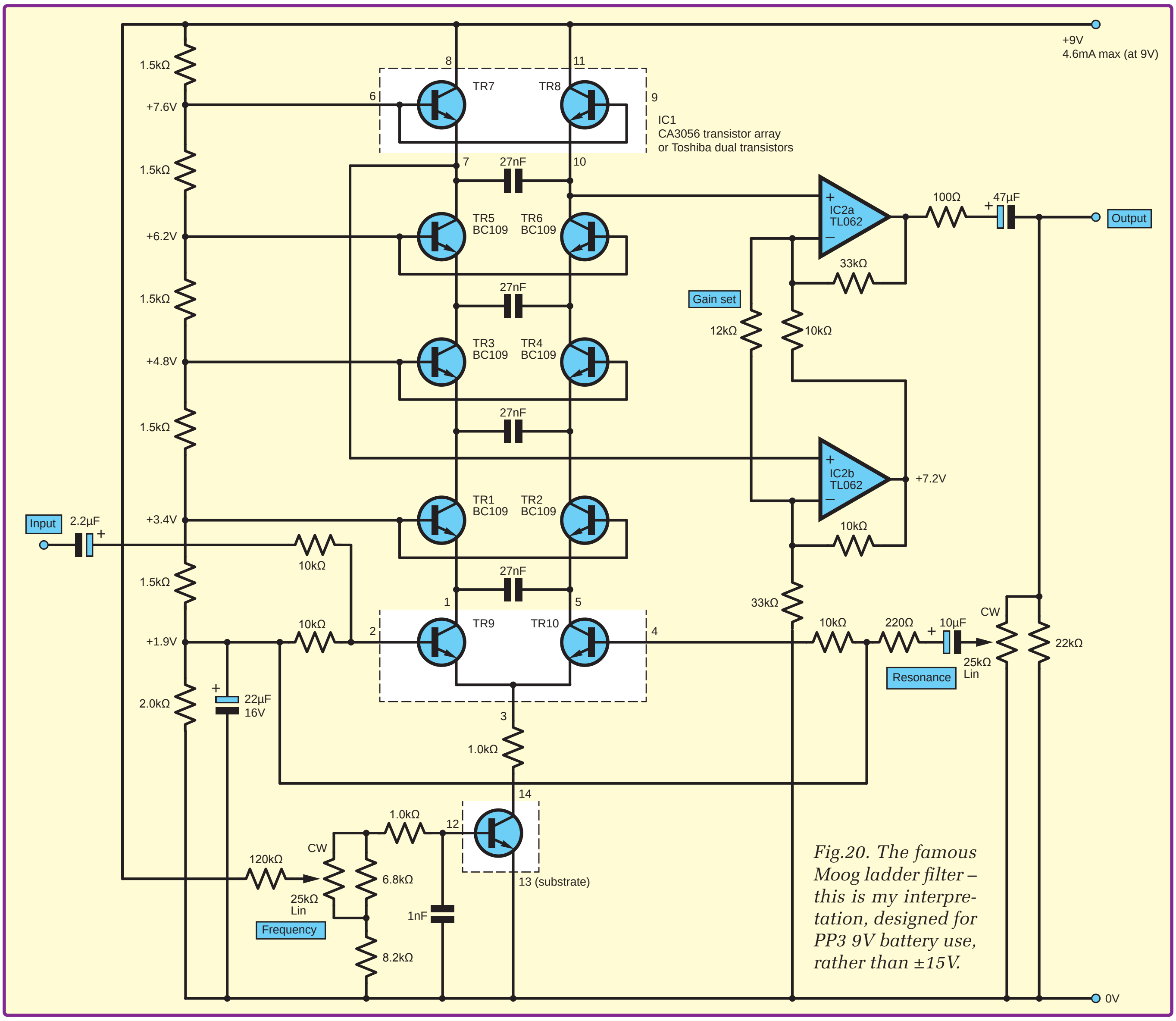

Fig.20. The famous Moog ladder filter – this is my interpretation, designed for PP3 9V battery use, rather than ±15V.

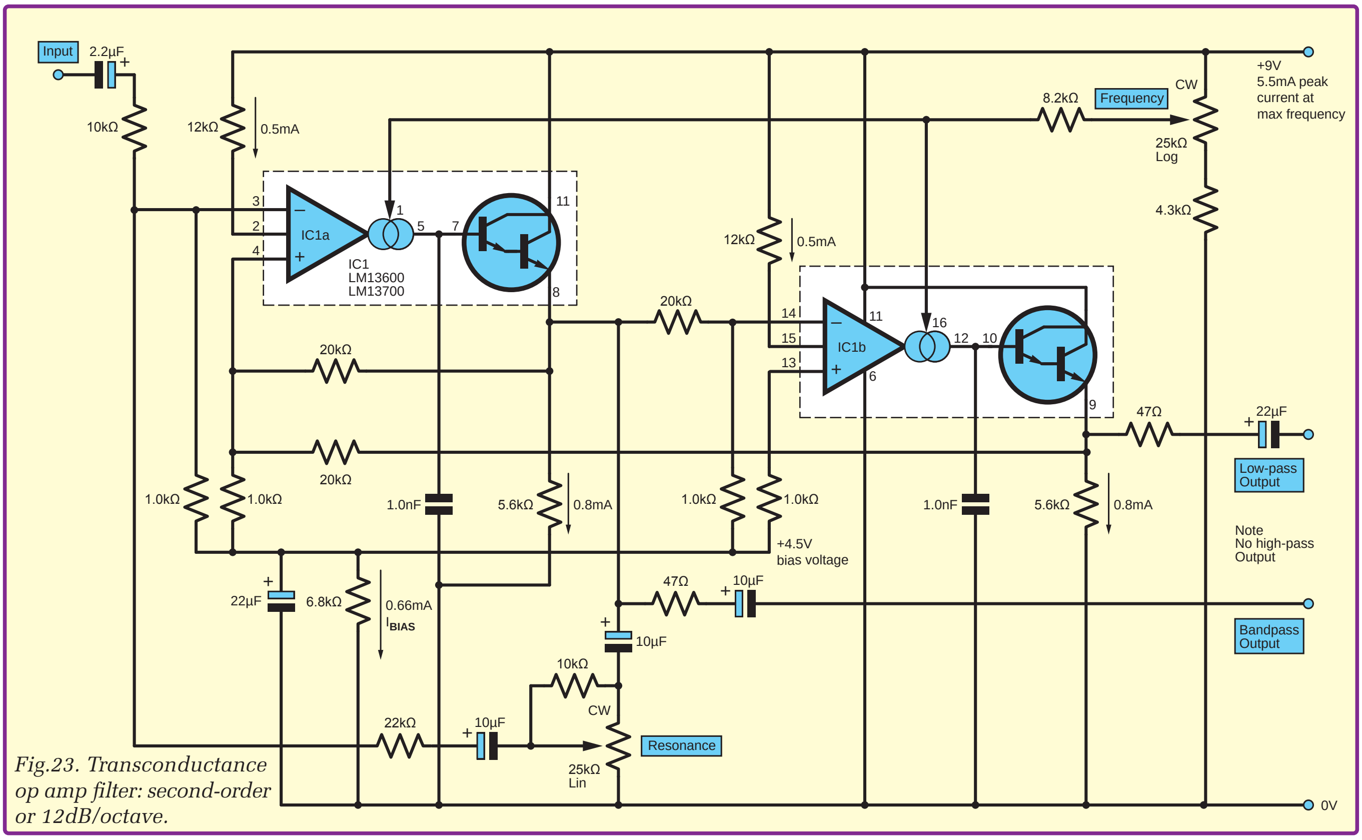

Fig.23. Transconductance op amp filter: second-order or 12dB/octave.

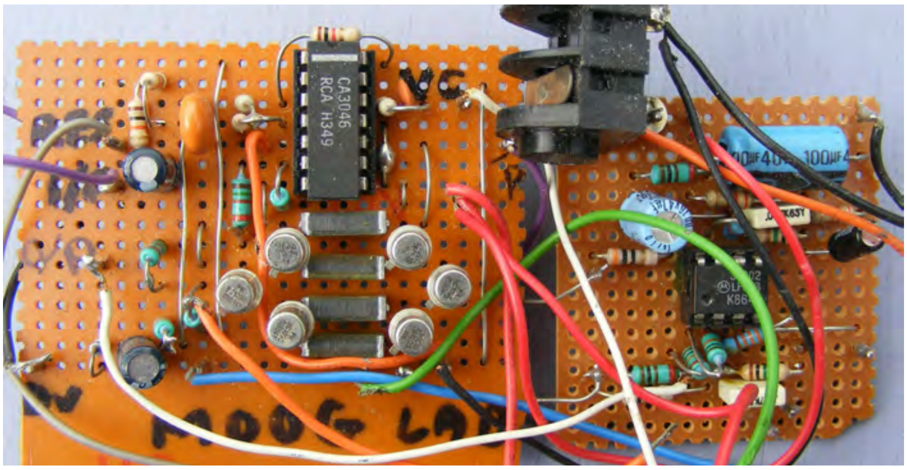

Fig.21. The Moog ladder filter built on Veroboard.

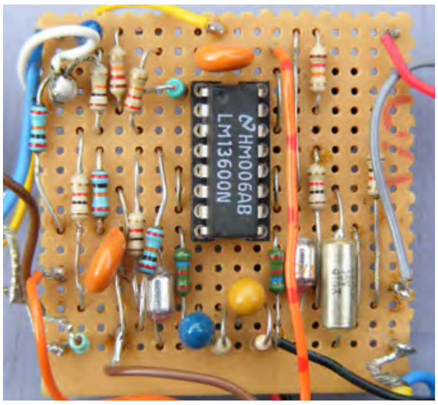

Fig.24. Veroboard construction of the transconductance op amp filter in Fig.23.

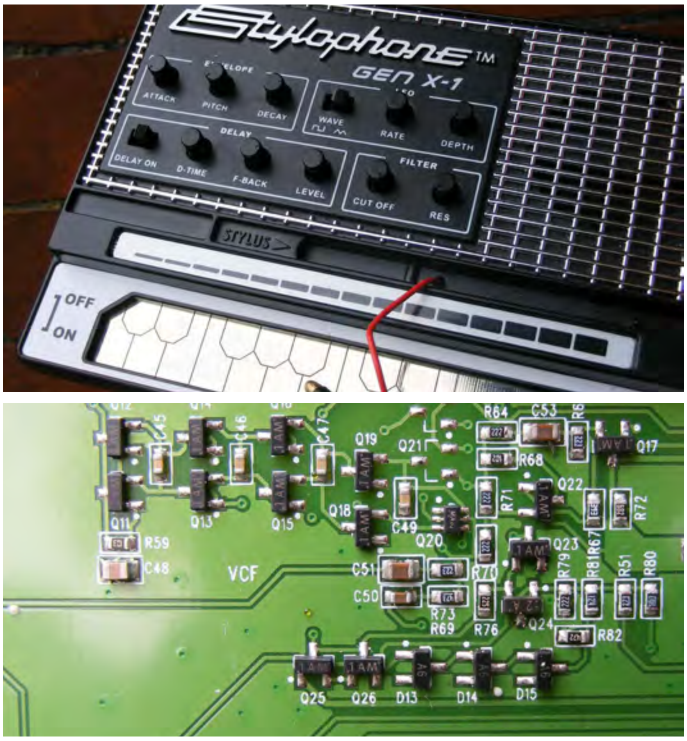

Fig.22. a) Gen X-1 synthesiser; b) Inside the synthesiser: the diode ladder filter made from discrete transistors can be seen. The total cost of the entire filter is less than an LM13600 and can run on 5V.

not have this feature, but I noticed no difference when used with this filter. The chip has the advantage of being available in surface mount.

Unusually, this circuit runs on a PP3 9V battery – the standard choice for guitarists and students, who find lugging mains dual-rail PSUs around a pain. In this circuit the current from the linearising diodes also generates the centre bias voltage across R20. The chips are quite noisy and not really 'Hi-Fi', but are effective in synthesisers or as a musical effect. Although the distortion is high at around 0.6%, it does not sound objectionable. It is more a form of soft clip. Although this design is only second-order, it is truly so, giving a 12dB/octave roll-off (the ladder filters giving a few dB less than their theoretical 24dB/octave).

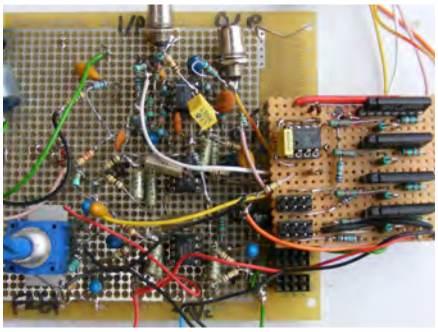

Fig.27. A prototype of the fourth-order filter. Note the plug-in board with four SIL (single in line) VCAs. The type used were the dbx 2180 (now made by That Corporation) available from Profusion in the UK. This was designed to evaluate the performance of different control elements.

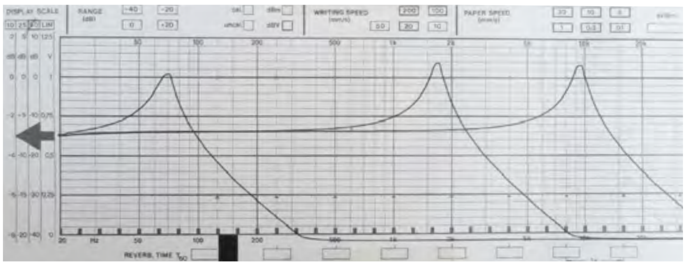

Fig.25. Peaky synthesiser response obtained by boosting the resonance.

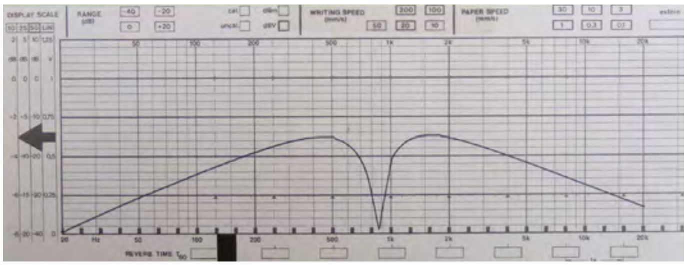

Fig.28. The notched band-pass filter output.

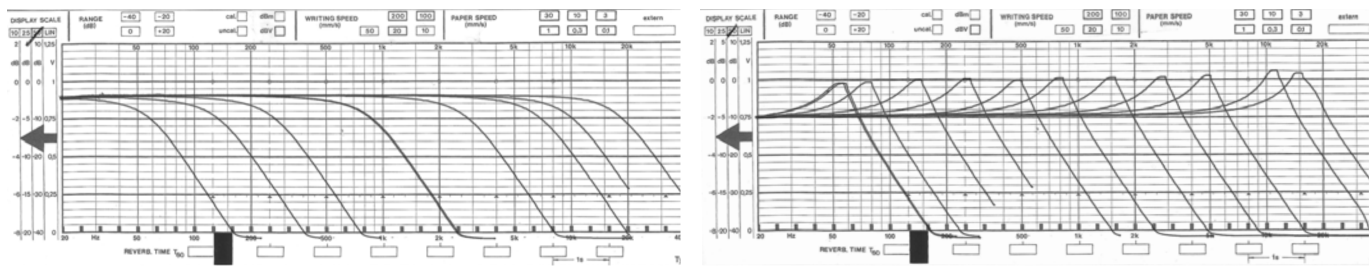

Fig.29. a) Low-pass response of the Blackmer VCA filter. Note consistency of curve as control voltage is varied. This would not be possible with a normal multi-gang potentiometer. Since this is a fourth-order filter the scale has been changed to 50dB or 2dB per a division; b) VCA filter at max Q. The slight droop in peak height towards the bass is due to the AC coupling

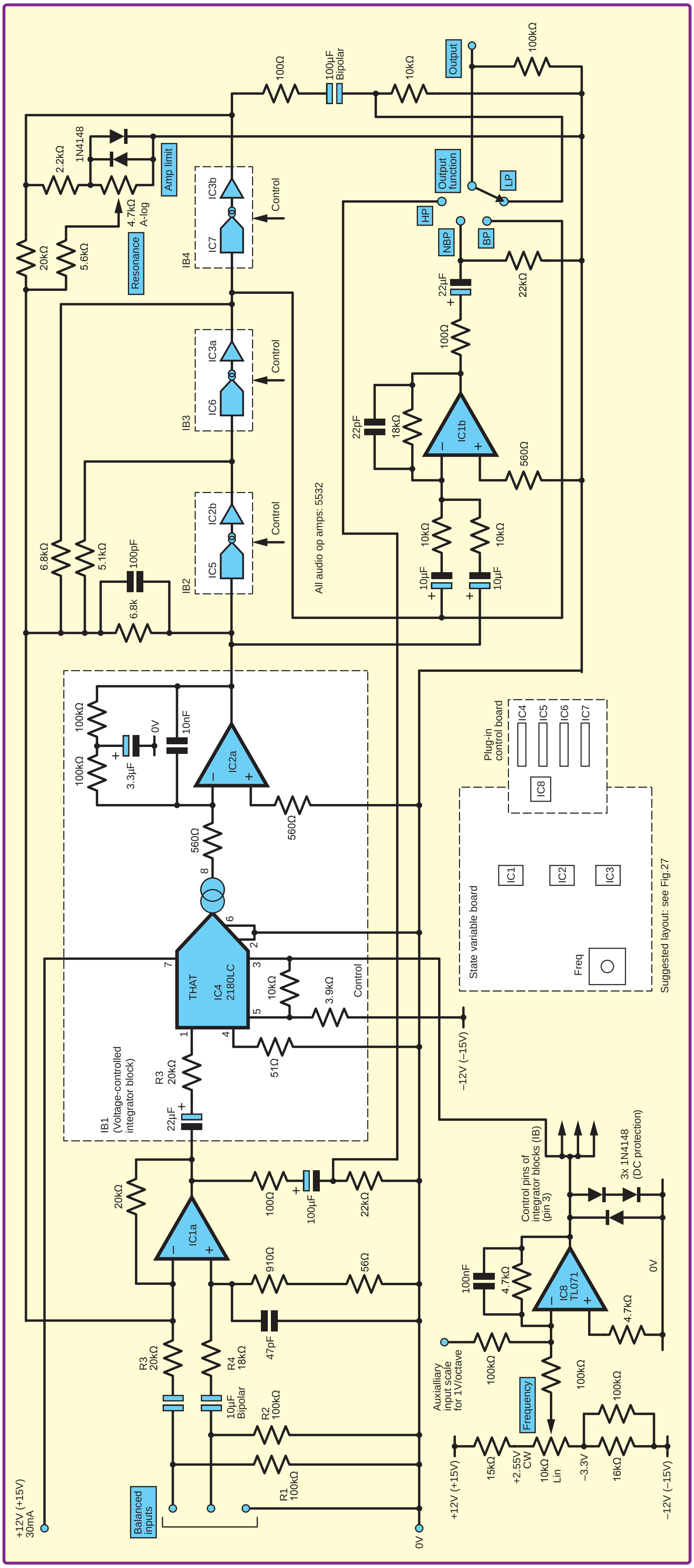

Fig.26. Fourth-order filter using 2180 Blackmer VCA chips from That Corporation as the control elements. Complex, but capable of full Hi-Fi performance. It avoids having to obtain an 8-gang precision anti-logarithmic potentiometer for stereo use.

A linear potentiometer loaded by a low-value resistor provides a logarithmic frequency control.

A photo of the simpler Veroboard construction is shown in Fig.24. Fig.25 shows the effect of turning up the resonance. It is possible to cascade two LM13600 filters to get a fourth-order filter, but the resonance control becomes difficult because of the higher phase shift. A dual-gang potentiometer could possibly be used with one for each second-order section, but this will impair reliability. In the fourth-order Oscar synthesiser filter, the resonance function was made voltage-controlled by adding an additional LM13600, allowing use of a single-gang potentiometer. I plan to try making a third-order filter with the spare section for a VCA. This would simplify the construction of a small synthesiser for *EPE* one day.

Mega-posh filter

My favourite filter is shown in Fig.26; it has proper Hi-Fi performance. It is based on the fourth-order state-variable configuration shown last month, but with Blackmer VCAs for frequency control rather than a potentiometer. As a result, the tracking of the integrator sections is much better, resulting in a more consistent response curve. A single potentiometer can be used to control all eight integrator sections in a stereo filter. The cumulative distortion of the VCAs is very low, in the order of 0.15% and subjectively benign. A full orchestra can be put through it and still sound clean and smooth. Blackmer VCAs are inverting, so the feedback paths from the first and third integrator stages have to be fed into the inverting pin of the summation stage IC1a, unlike the original fourth-order state variable. If this isn't observed the result will be an oscillator. The prototype circuit is shown in Fig.27. It has an additional output which is the two band-pass outputs summed together which results in a band-pass with a notch in the middle shown in Fig.28. This provides a unique phaser type sound when the frequency is swept up and down. Fig.29a shows the curve across the audio range and Fig.29b shows a degree of peaking applied.

When I get a PCB done, I'll do this circuit as a proper project. DJs and live electronic musicians will love it. It makes a fantastic filter swell pedal.