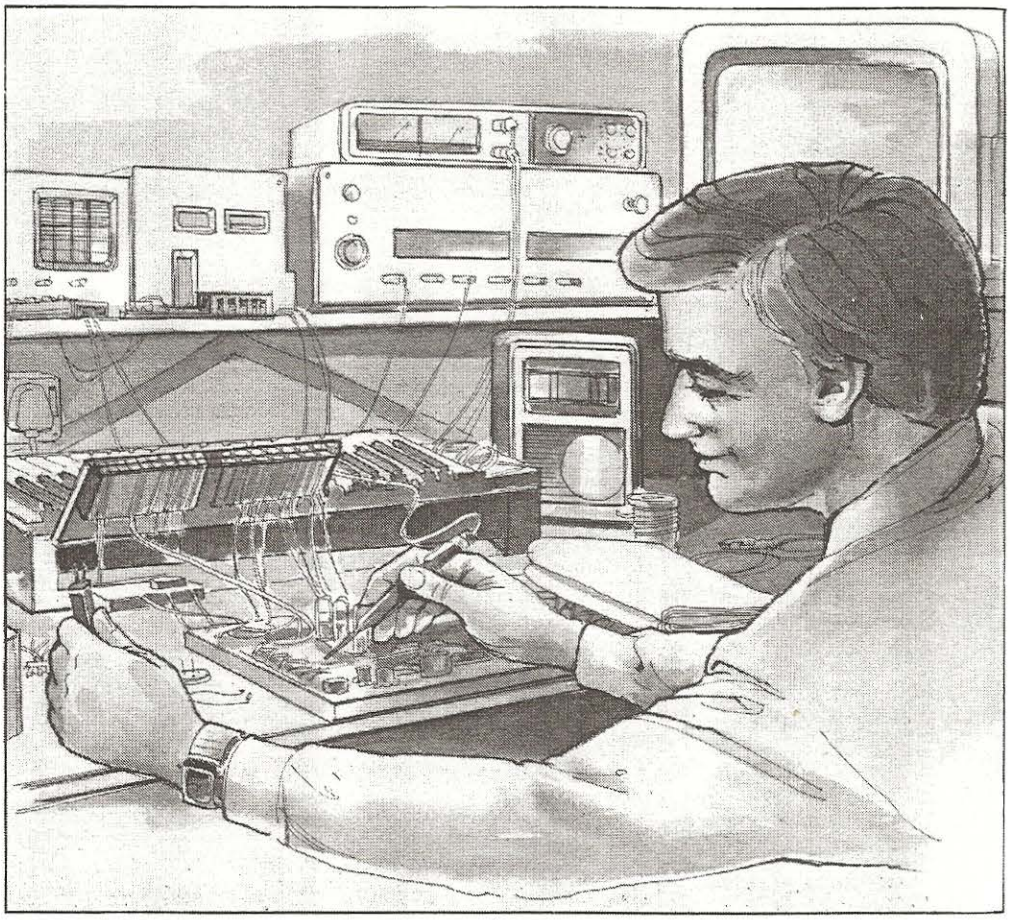
ELECTRO-MUSIC ENGINEER

Charles Blakey Digisound Ltd

Resonant Filters



Most readers will have seen pictures depicting the spectrum analyses of conventional musical instruments. Invariably these show that the amplitude of the harmonics do not fall off in a uniform manner and often the second, third and some higher harmonics are greater in amplitude than the fundamental. Thus simple analogue synthesis using a waveform with a high harmonic content, such as a sawtooth or pulse waveform, cannot faithfully reproduce the sounds of these instruments and the usual low pass filtering obviously does not help this situation.

The reinforcement of harmonics, or even the production of inharmonics, stems from the shape of traditional instruments as well as the materials used in their construction. The effect of these factors is to cause vibrations in, or within, the body of the instrument when it is played. These vibrations give rise to resonant frequencies known as "formants" and fortunately for electronic synthesis the formants remain constant irrespective of the note played. The violin provides an excellent example of the importance of formants, even though their nature is somewhat different when compared to most other acoustic instruments. It has been demonstrated that one of the features of the Stradivarius is that its resonant frequencies are about 1kHz higher than poorer quality violins and consequently the effect of this is to reinforce the higher harmonics and give a more pleasing tone.

The formant frequencies of conventional instruments are generally in the range of 600Hz for a double bass up to about 3500Hz for a modern violin. They are best synthesised by using a number of filters which may either be used to boost a comparatively narrow frequency band or which can reinforce several of the higher harmonics simultaneously. Such filters do not have to be incorporated within a synthesiser but may be a post treatment of the sound in much the same way as effects units. Furthermore the use of such filters is not confined solely to imitative synthesis of conventional instruments but can be put to good use with almost any type of sound, including vocals.

The essential features of resonant filters are the ability to manually adjust their frequency within the range given above, a variable Q (quality factor) and also control of gain. Perhaps it will help some readers if Q is briefly explained. Figure 1 shows the output from two band pass filters and the first peak has a Q of about 4 and the second peak a Q of about 1. It is evident that the higher the Q the narrower the band of frequencies that are passed by the filter. Q is easily determined by first measuring the amplitude of the peak and the frequency at which it occurs and then measuring the frequencies on either side of the peak where the amplitude is 0.707 (-3dB) of the peak amplitude. If the peak frequency is now divided by the difference of the other frequency measurements then the answer is the Q value.

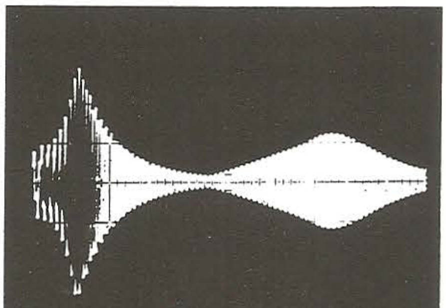


Figure 1. Band pass filtered signal.

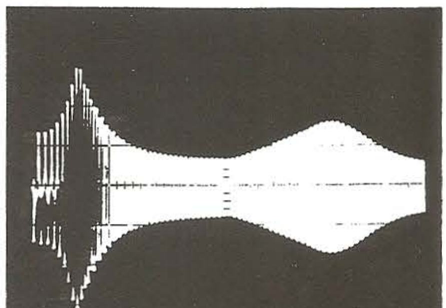


Figure 2. Band pass filtered signal plus a proportion of the original signal.

A resonant filter is essentially a band pass filter but if the type of filters shown in Figure 1 were used then much of the original signal will have been filtered out whereas the intention, as described earlier, is to enhance certain frequencies and add them to the original signal. This is achieved by mixing a proportion of the original signal with the band pass filtered signal and the result of this is shown in Figure 2. The other control mentioned was gain since we may only wish to slightly enhance a narrow band of frequencies (high Q) or alternatively strongly enhance a wide band of frequencies (low Q). Often this is achieved by increasing the overall gain of the filter but the problem here is that one has to constantly be aware of the Q factor since it is easy to overload the output causing distortion or clipping. The usual approach, however, is to limit the gain such that the signal will not distort at highest Q but this inevitably means a restricted boost at low Q settings. In the design shown in Figure 3 the approach is quite different since the Q may be varied from about 0.5 to 10 and over the frequency range 25Hz to 3,300Hz the amplitude of the output remains substantially constant, that is, within ±1dB. Thus the output will not overload irrespective of Q and the "gain" is varied by the attenuating potentiometers RV3 and RV6.

The design is based on a new filter IC from Curtis Electromusic Specialities (CES). The part number is CEM 3350 and its correct description is a "Dual Voltage Controlled State Variable Filter". Since it is a dual filter then two resonant filters may be constructed

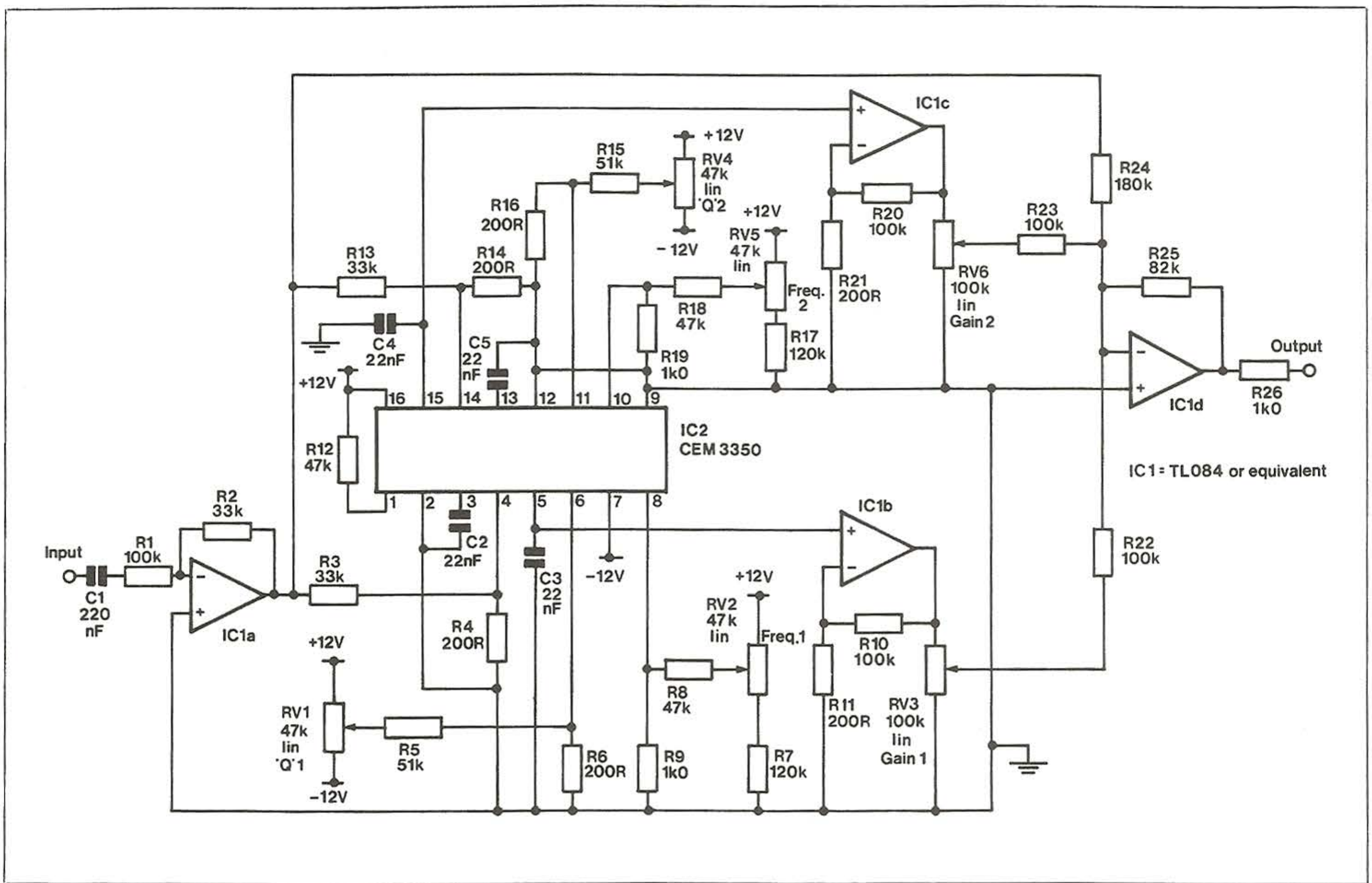


Figure 3. Two resonant filters using the CEM 3350 IC.

from a single IC. The CEM 3350 will not be described in detail at this time but its versatility is such that it is certain to gain favour for many novel filter applications as well as for conventional voltage controlled filters. One important point to note, however, is that its power requirements vary from the other CES ICs which have been described in earlier issues of E&MM. The IC is not guaranteed to withstand operation from dual power supplies whose total voltage is in excess of 26V. On the other hand it will operate with supplies down to about ±3V and so is well suited to battery powered projects. For the design example ±12V supplies are used and zeners or voltage regulators may be used with higher voltage supplies.

The dual resonant filter illustrated in Figure 3 is very compact since it requires only two DIL packages, the CEM 3350 and a quad op-amp such as the TL084, and a handful of other components. Another feature is that no trimmers are required which in turn means that no test equipment is needed for setting up the filters. The input signal is applied to IC1a and thence split three ways: to each of the resonant filters and to IC1d where the outputs of the filters are summed with the original signal. The resistors R22 to R25 are chosen such that the resonant frequency bands may be boosted by +15dB over the original signal while the overall peak to peak gain is unity when a filter is at maximum boost. In many applications the input stage, IC1a, would not be required but since the summing amplifier, IC1d, causes a polarity inversion then IC1a corrects this and prevents signal cancellation effects. A bonus from using IC1a and IC1d is that it becomes a simple matter to modify the circuit for various signal levels without having to alter components connected to IC2. For example, the design is configured for a 10V p-p signal and to alter it for, say, 5V p-p then R2 would be changed to 68k and R25 to 39k. The signal input to the filters is further attenuated by R3/4 and R13/14 and applied to pins 4 and 14 of the CEM 3350, which are termed the "Variable Gain Inputs". There are also "Fixed Gain Inputs" at pins 2 and 12, which are grounded in this application, and using these latter inputs or a combination of the two types of inputs allows user flexibility in achieving various characteristics.

Pins 3 and 13 of IC2 are the low pass outputs from the filter while pins 5 and 15 are the band pass outputs. The latter are used for the present application and since the outputs are of high impedance they are buffered by IC1b and IC1c which are configured as non-inverting amplifiers. The gain of the latter are set to restore the signal to its original level.

The Q controls for the CEM 3350 (pins 6 and 11) have an exponential response and thus give a more "natural" control response although this feature is not of special importance for the resonant filters. RV1 and RV4 and associated attenuating resistors control the Q of the filters, to the limits stated earlier, and an increasing negative voltage increases the Q factor.

The frequency controls for the filters also have an exponential response scaled at a nominal 18mV per octave. The frequency is adjusted by RV2 and RV5 and associated resistor networks, with R7 and R17 used to restrict the upper frequency limit. The reason for limiting frequency is twofold. Firstly, higher frequencies will result in wider amplitude variations to that specified above of +1dB. Secondly, since resonant filters only require a limited frequency range it enables specific frequencies to be more accurately obtained with a single potentiometer. Note that these potentiometers should also be "reverse" wired since the frequency decreases with increasing positive voltage.

The last component is R12 which is connected from the positive supply pin to the current reference pin (pin 1). For normal applications a nominal reference current of 400uA is required for the four exponential generators in the CEM 3350 and this is produced by R12.

As stated earlier, resonant filters are valuable for both imitative and creative synthesis and it is hoped that this simple and effective design will encourage synthesists to explore their applications. **E&MM**