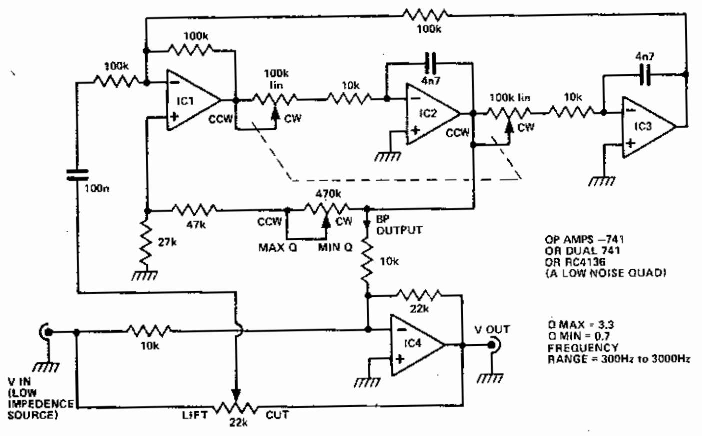
PARAMETRIC EQUALISER



This is possibly the equaliser for the amplifier system that has everything. The parametric equaliser has got three controls. It is a bandpass filter which can have variable cut or lift, so that a particular frequency band can be enhanced or rejected. The resonance can also be controlled so that area of frequency affected can be broad or narrow. Also the centre frequency of the bandpass filter can be varied so that it can be tuned to operate at a particular frequency. The circuit operation is quite simple.

Op amps IC 1, 2, 3 form a state variable filter, the Q and centre frequency of which can be varied. Op amp IC4 is a virtual earth amplifier. When the equaliser is in the lift position, the signal is fed into the state variable filter. It then comes out of the bandpass output and into IC4. In this feed forward position the equaliser has got a peak (lift) in its response. When the equaliser is in its cut position, the bandpass filter is in the feedback loop of IC4 and so there is a notch in the frequency response.

Care must be taken not to cause overloading and clipping when using high Q lifts.

Parametric equalizer improves Baxandall tone control

by Henrique Sarmento Malvar
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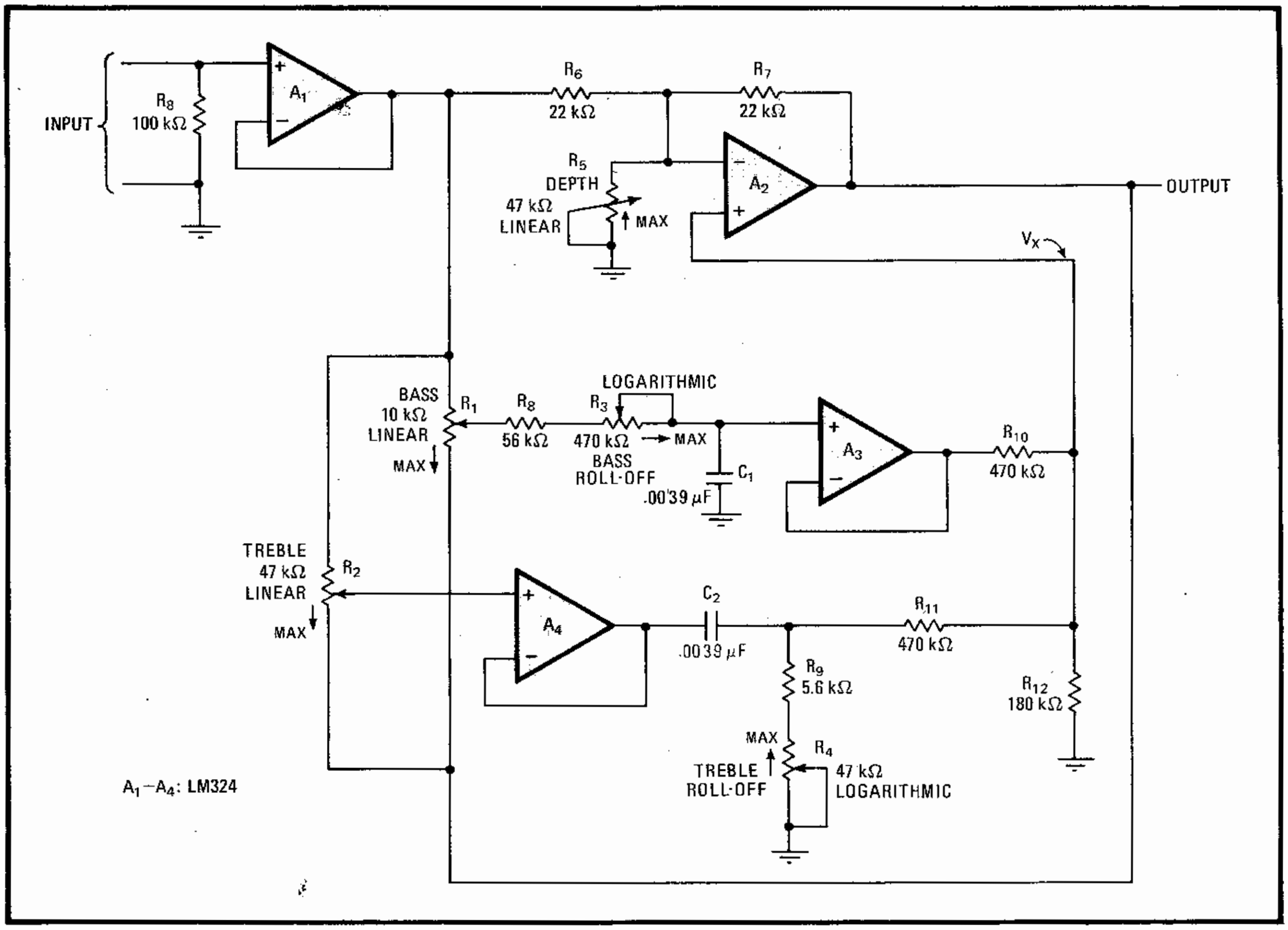
Simple active filters are used here to build a continuously adjustable parametric equalizer having the same general response as the popular Baxandall circuit, which utilizes a switch-selectable scheme for bass and treble equalization. Center frequencies for both upper and lower bands, as well as their individual roll-off characteristics, may be independently controlled, and the depth of the equalization is also adjustable.

The circuit (see Fig. 1), an adaptation of an idea proposed by Thomas,[1] utilizes positive feedback and/or feed-forward principles to achieve the type and amount of equalization required. Five potentiometers set the aforementioned parameters, with the circuit operating on all simultaneously.

The center of the low-frequency passband is set by R_1. If the wiper of the bass control is moved towards the input operational amplifier, A_1, more of the low-frequency components of the input signal will pass through low-pass filter $C_1R_3R_8$ and appear at V_x, with potentiometer R_3 determining the roll-off. Because op amp A_2 inverts the signal, partial cancelation of the low-frequency components occurs and the total bass content is reduced at the output. As R_1 is moved in the opposite direction, a positive feedback loop around op amps A_2 and A_3 is formed, and the bass gain increases. In similar



1. Trimming timbre. One-chip equalizer provides continuously variable control of bass and treble center frequencies, as well as individual roll-off characteristics. Depth of audio-band equalization is also adjustable. Unit costs little more than standard switch-selectable devices.

Tunable audio equalizer

Flexible parametric equalizer with variable Q

by Martin Thomas

MOST AUDIO EQUALIZER circuits represent a compromise between cost, facilities and ease of use, and the Baxandall tone control has been by far the most successful design. For domestic audio equipment its simplicity and ease of use outweigh its disadvantage of providing only a limited degree of equalization, although the circuit can be modified to increase its flexibility[1]. Clearly, however, the "bass" and "treble" subdivision of the audio band is insufficient for many purposes, and the graphic equalizer approach of having a larger number of frequency bands becomes necessary. The only problem with this approach is that a large number of controls must be used to cover the audio band if the individual frequency bands are narrow, so even with this circuit the number of controls is a compromise.

Unquestionably the most versatile equalizer is the parametric, or tunable, type. In its simplest form it can consist of only a single boost/cut element, but its centre frequency can be varied continuously over a wide range (possibly over the whole audio band), and the Q can also be varied so that either a broad or a narrow frequency band can be equalized. This approach allows almost any equalization requirement to be met with only a small number of such elements, and since the elements are identical, no more than are actually needed can be connected together for any particular application. A parametric equalizer may not be so straightforward to use as a graphic equalizer, but once you become accustomed to the rather different controls it's much easier than you might expect.

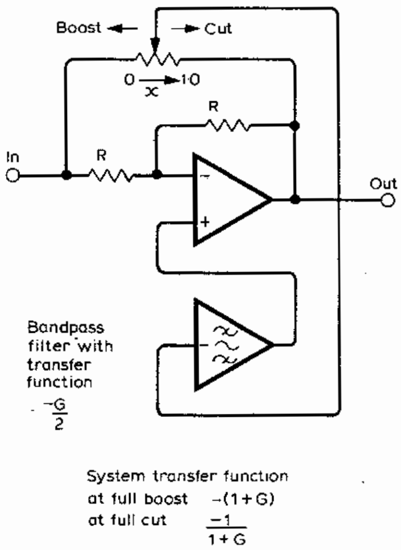


Fig. 1. *Basic equalizer design shows how to achieve either boost or cut with a single active element.*

Circuits of this type have been around for a number of years, but relatively little has been published about them. In this article I shall take the opportunity to discuss some aspects of the design theory, in addition to describing my own design. The circuit has continuously and independently variable centre frequency, boost/cut amplitude and Q, and also allows a choice of two different sets of boost/cut amplitude-frequency response curves as the control setting is varied; more about that later.

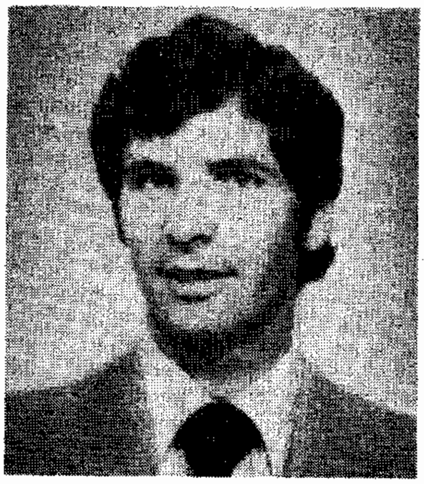
The circuit for a tunable equalizer can be broken into two sections, by which point the basic design is almost complete! The first problem is how to use a single active element either to boost or to cut a given frequency range, and this can be achieved by the circuit shown in Fig. 1. The filter used in the present design is phase-inverting, so its output is connected to the non-inverting input of the amplifier to give overall negative feedback. With this connection there is a gain of two from the filter output to the amplifier output, so the filter transfer function is specified as $-G/2$ to express the system transfer function in its simplest form. When the boost/cut potentiometer is at either end of its travel, the filter is entirely in either the forward or feedback signal path, giving transfer functions of $-(1 + G)$ and $-1/(1+G)$ respectively. An exact expression for the transfer function at other control settings will be developed later, but for now notice that when the control is at its midpoint, the forward and feedback contributions will be equal, giving a transfer function of -1 ("flat"). An extension of this circuit to include several filters and potentiometers yields the basic design for a graphic equalizer, of course, and a typical circuit is described in ref. 2.

The second problem is the design of the tunable filter. In theory this is very simple, but there are several practical difficulties. Although capacitors can be switched to change the frequency range, the variable control clearly has to be resistive, which rules out some otherwise very promising circuits such as the multi-feedback filter[3], since the Q will then also vary. The Wien-bridge configuration does meet this requirement if the resistors in the forward and feedback arms of the bridge are varied together[3], but the Q is sensitive to mismatch between these resistors. As the

Three years ago, Martin Thomas left Cambridge University, having collected the B.A. and M.A. degrees in Natural Sciences and a Ph.D. in neurophysiology, to become first a research fellow and later an assist-professor at Boston University. Now, he's returning to the UK to join the Physiology Department at Oxford University.

His audio interests developed while he was at Cambridge, and he designed the prototype for the equalizer there. He says his research activities aren't directly related to his audio and electronics interests, although in practice there's a lot of overlap between them. Using ion-specific dyes to follow changes in ion concentrations inside nerve and muscle cells during excitation is basically a technical problem. "I had to build a sensitive microspectrophotometer to be able to resolve the very small changes in dye absorbance, and it involved a fair amount of electronics."

Would he ever consider moving out of research and into industry? "That depends. Most companies really aren't very interested in my type of background, and I'd probably have to set up a business of my own. But I certainly wouldn't rule out doing that at some time in the future."



sensitivity increases with Q, the circuit is suitable for use only at low Q, and the long-term reliability is questionable, once the resistor tracks start getting dirty!

The state-variable filter, which is synthesized from integrators, meets the requirements very well, and has the additional advantage that it is inherently stable even at high Q. Its only drawback is that it uses three operational amplifiers rather than one, but the number of passive components is almost the same as for other circuits, and it has been chosen for the present design. Fig. 2 shows the circuit diagram. Further information on state-variable filters is given in the appendix and in ref. 4, but the basic equations are reproduced below. Referring to the component values in Fig. 2, the transfer function is

$$\frac{V_o(s)}{V_i(s)} = -\frac{R_2(R_3+R_4)}{(R_1+R_2)R_4} \times \frac{R_6C_2s}{R_5C_1R_6C_2s^2+[R_1(R_3+R_4)/(R_1+R_2)R_4]R_6C_2s+R_3/R_4}$$

and the bandpass centre frequency is

$$\omega_o = \frac{R_3}{R_5C_1R_6C_2R_4}$$

For $R_3 = R_4$, and $\omega_o = 1/R_5C_1 = 1/R_6C_2$, the transfer function becomes

$$\frac{V_o(s)}{V_i(s)} = -\frac{2R_2}{R_1+R_2} \times \frac{s/\omega_o}{s^2w_0^2 + [2R_1/(R_1/(R_1+R_2))]\, s/\omega_o + 1}$$

$$= -\frac{2R_2}{R_1+R_2} \times \frac{\omega_o s}{s^2+[2R_1/(R_1+R_2)]\omega_o s+\omega_o^2}$$

Comparison of this last equation with the generalized second-order bandpass transfer function

$$\frac{V_o(s)}{V_i(s)} = \frac{\omega_o A_o s/Q}{s^2+\omega_o s/Q+\omega_o^2}$$

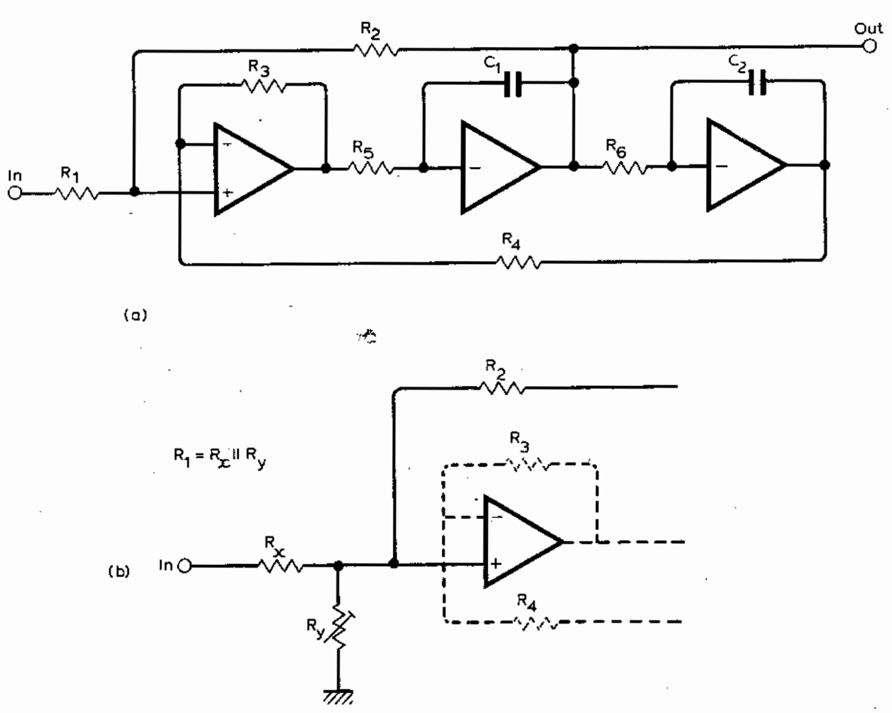
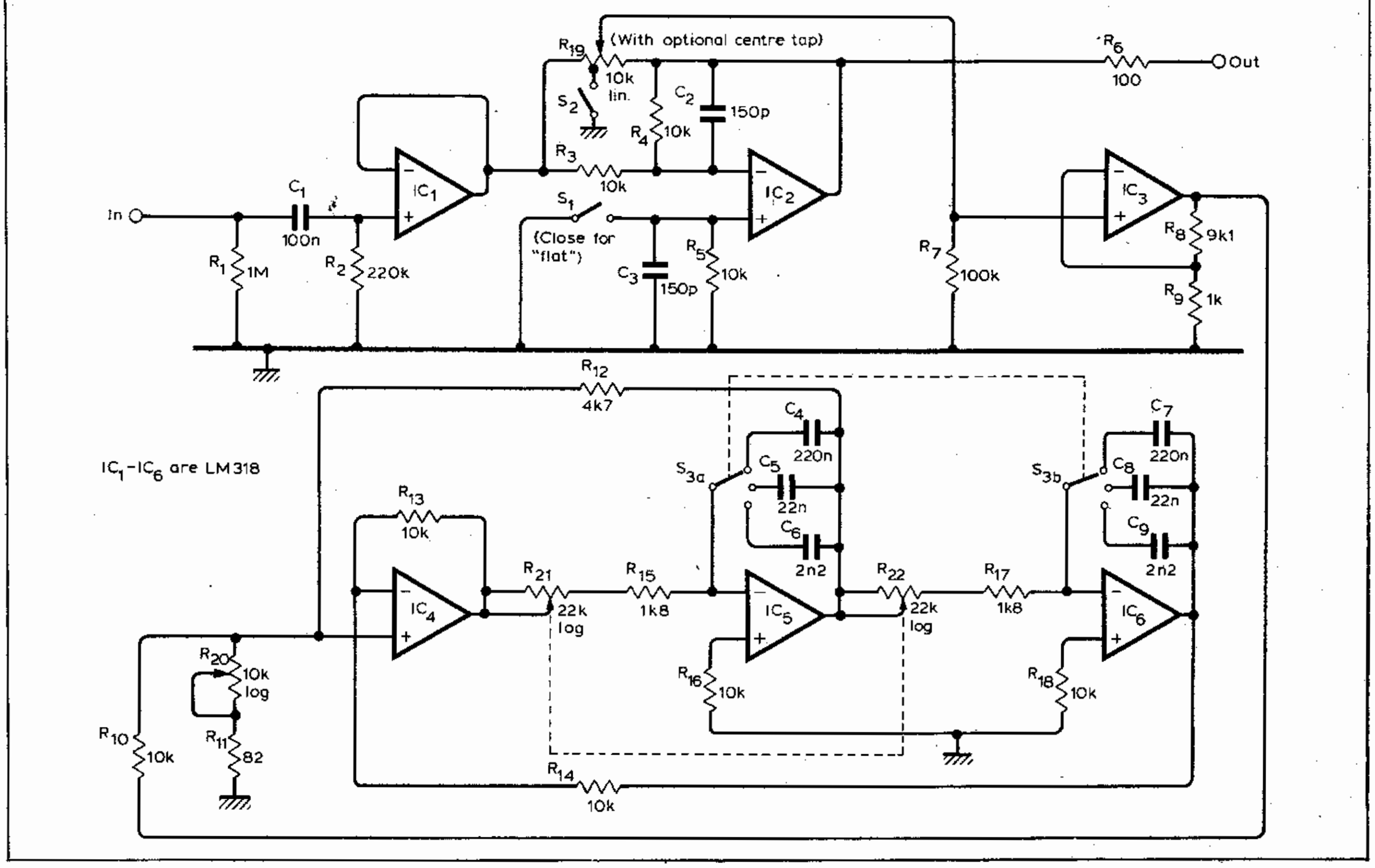


Fig. 2. *"State-variable" bandpass filter is inherently stable even at high Q, (a). Modification for constant centre-frequency amplitude, (b). Varying Ry gives independent control of Q.*

Fig. 3. *Circuit diagram for a single-section tunable equalizer. Ganged resistors R_{21}, $_{22}$ determine centre frequency, together with range switch S_3. Boost/cut control is R_{19} while Q is varied with R_{20}. Fig. 5 illustrates function of S_2.*



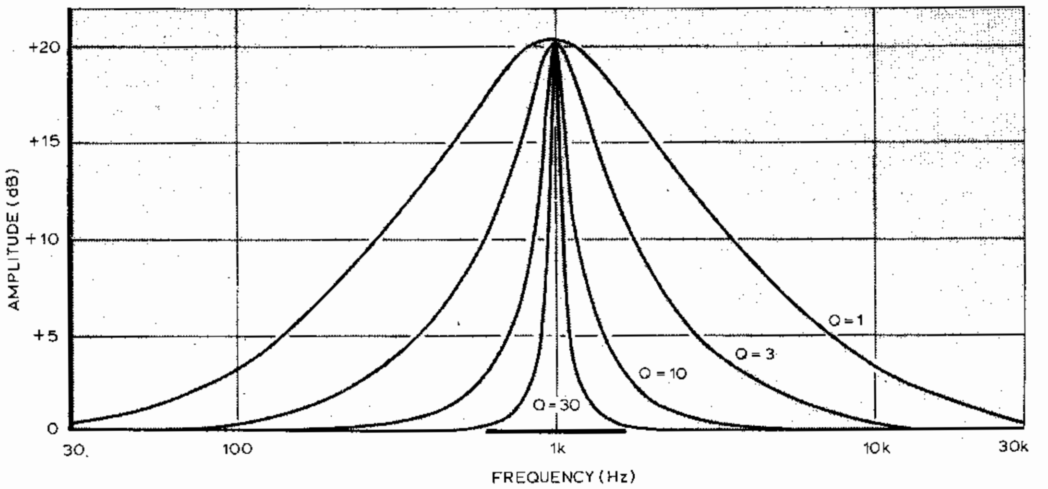


Fig. 4. *Effect of varying Q with boost/cut control at maximum boost.*

shows that Q is $(R_1+R_2)/2R_1$, and the centre-frequency gain A_o is $-R_2/R_1$. By varying R_5C_1 and R_6C_2 together, it is thus possible to vary w_o independently of Q and A_o. The Q will change if there is any mismatch between these two time constants (although A_o will remain constant), but you can see from the transfer function that the sensitivity does *not* increase with Q, and hence accurate component matching is *not* necessary.

The Q can be varied independently of ω_0 by varying R_1 or R_2, but this will also alter A_o, and the relation between A_o and Q is non-linear. Fortunately, a simple modification to the basic circuit, as shown in Fig. 2(b), overcomes these problems. Resistance R_1 is replaced by two resistors, R_x and R_y, so A_o is now — $R_2(R_x+R_y)/R_xR_y$. Resistors R_x and R_y also form an attenuator for the input signal, the gain being $R_y/(R_x+R_y)$. The overall filter gain is the product of these two terms, i.e. $-R_2/R_x$, hence by varying R_y the Q can be varied independently of the centre-frequency gain. The only disadvantage of this modification is that the overall centre-frequency gain of the filter with the component values used in the final design is only 0.5, so additional amplification in the filter path is necessary. The extra amplifier is placed before the filter, where it also provides the necessary low-impedance source.

The complete circuit is shown in Fig. 3. A low-impedance source is provided by IC_1 for the boost/cut control and its associated amplifier IC_2. The input amplifier for the filter is IC_3, and IC_4-IC_6 comprise the filter itself. Although the circuit may appear elaborate, the number of passive components is relatively small, and the size and component count can be reduced further if dual or quad ICs are used. It should be borne in mind, however, that the circuit was designed around the LM318, which is a high bandwidth device. Use of other ICs will result in inferior performance at high frequencies, although it may still be satisfactory for many applications, and possible substitutions will be discussed later on.

The present design is a general-purpose one, but the range of centre frequencies, boost/cut and Q can easily be modified if required. The centre frequency is determined by the ganged variable resistors R_{21}, R_{22} and by capacitors C_4 to C_9 (see Fig. 3). The variable frequency range is just over tenfold, and capacitor switching gives a total range of three decades, the nominal frequency ranges being 30 to 300, 300 to 3,000 and 3,000 to 30,000Hz. The range switch S_3 should be a make-before-break (shorting) type, so that the capacitative feedback paths around IC_5 and IC_6 are not interrupted during the instant of switching, otherwise the circuit could oscillate. The sliders of R_{21} & R_{22} should be connected to the clockwise end of their track, so that there is a d.c. path through the control even if the slider loses contact with the track, as the control has to provide a path for the input bias currents of IC_5 and IC_6. A logarithmic control has been specified, although it will have to be turned anticlockwise to increase the frequency, whereas a clockwise law would be preferable. A clockwise law can be obtained by using a dual antilog control (in which case the sliders should be connected to the anticlockwise ends of the tracks), but this component may not be readily available from most suppliers.

The boost/cut range is determined by the gain of IC_3 and the attenuator at the input of IC_4. There is a gain of two from the output of the filter network (at IC_5) to the output of the circuit (at IC_2), and a two-fold attenuation at the centre frequency through the filter itself, so the overall gain through the filter path at the centre frequency is given by the gain of IC_3, which is 10 with the component values shown in Fig. 3. Reference to Fig. 1 shows that the maximum boost and cut are $-(1+10)$ and $-1/(1+10)$, i.e. ± just over 20dB, but these values could easily be modified by changing the gain of IC_3, which is of course $(R_8+R_9)/R_9$. Input bias current for IC_3 is provided by the boost/cut control R_{19} and by R_7, which provides an independent path in case of poor slider contact in R_{19}.

The method of Q variation is as already described with reference to Fig. 2(b), and R_y is represented by R_{20} and its series resistor R_{11} in Fig. 3. The Q range of the circuit is 1 to 30 with the component values shown, and although the upper limit is unnecessarily high for many applications, the value of 30 was chosen simply because it can be achieved over the entire audio range when LM318s are used. It can be reduced by increasing R_{11}. Once again we have a law problem with the control; if R_{20} is a logarithmic control, the Q will increase as the control is turned anticlockwise. An antilog control can again be used to obtain a clockwise law (slider now connected to the anticlockwise end of the track), or a range of fixed, switch-selected Q values can be obtained by replacing R_{20} and R_{11} by a series of fixed resistors, Q being

$$\frac{R_{12}}{R_{10}//(R_{20}+R_{11})}$$

Obviously it is useful to be able to switch out the filtering, and this is achieved by S_1, which simply shorts the non-inverting input of IC_2. If the d.c. output of the IC_5 is not exactly zero, the d.c. output of IC_2 will shift when S_1 is closed, but the shift on the prototypes was only a few millivolts and did not cause any audible effects. In fact, the circuit is d.c. coupled throughout (apart from the input to IC_1), and it may be advisable to add a coupling capacitor at the output of IC_2 if the possibility of output offset cannot be tolerated.

The output resistor R_6 is not for protection, but rather to isolate any capacitative load from IC_2 to ensure stability. Capacitors C_2 and C_3 also help to ensure stability by rolling off the amplitude response above 100 kHz. The only other stability precaution — but which is perhaps the most important — is to decouple the ±15V supplies with

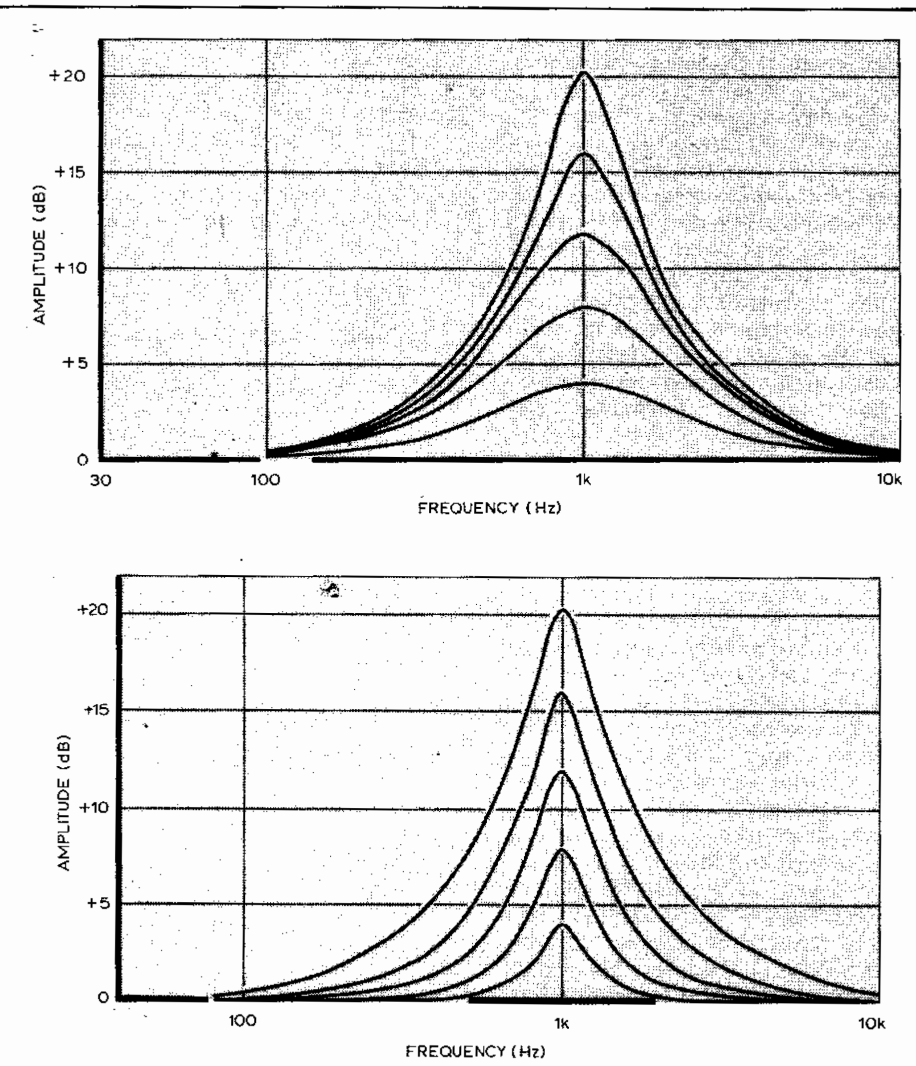


Fig. 5. *Two sets of amplitude-frequency response curves can be generated by the circuit with S_2 open circuit (a) and with S_2 closed (b). Corresponding cut curves are symmetrical about log. frequency axis.*

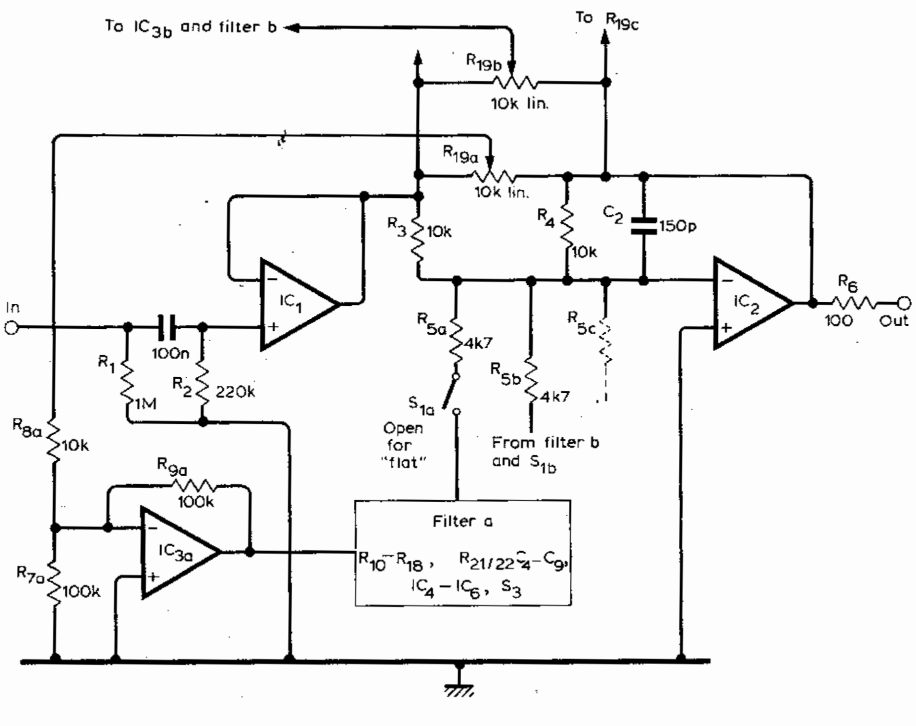


Fig. 6. *Modified circuit suitable for use with any number of filter sections.*

0.1μF capacitors. Such decoupling is very important with high bandwidth devices like the LM318, and if any instability problems are experienced, they can almost certainly be traced to this cause. On the prototypes it was found helpful to connect a capacitor directly between the supply lines in addition to the normal practice of decoupling between each supply line and ground, and the capacitors should of course be sited as close as possible to the ICs. In spite of the impression which may have been gained from these remarks, I was pleasantly surprised by the high stability of the prototypes, and there is no reason to suppose that such results cannot be obtained consistently if standard layout practices are followed.

Performance details of the circuit are given in Figs 4 and 5. These graphs have all been obtained for a centre frequency $\Omega/2\pi$ of 1kHz, but the performance for any other frequency in the audio band can be obtained by appropriately shifting the log. frequency axis. Only the boost curves are shown; the corresponding cut curves are symmetrical about the log. frequency axis. Fig. 4 shows the effect of varying the Q control when the boost/cut control is at maximum boost, and gives an idea of the very wide range of equalization curves which can be generated by the circuit. The centre-frequency gain remains independent of the Q control setting as the boost/cut control is varied, but the effect of the boost/cut control on the frequency response is not as straightforward as might be imagined. Fig. 5(a) shows the effect of the boost/cut control when R_{20} is set for a Q of 3, from which it can be seen that the Q is reduced as the control is rotated toward its centre ("flat") position. By a simple modification to the circuit, however, it is possible to generate the curves shown in Fig. 5(b), where the shape of the response remains relatively constant as the boost/cut control is varied. The reason why these two families of curves can be obtained may not be intuitively obvious, but it can be explained by the following analysis.

It has already been shown that the system transfer functions at full boost and full cut are given by $-(1+G)$ and $-1/(1+G)$. At intermediate positions of the boost/cut control R_{19}, both forward (boost) and feedback (cut) signals will pass through the filter. Let the fractional rotation of R_{19} be represented by x, such that at full boost $x=0$ and at full cut $x=1$ (see Fig. 1). Resistor R_{19} will act as a potential divider for the two signals, so the forward signal contribution to the transfer function will be $-(1+(1-x)G)$, and the feedback contribution will be $-1/(1+xG)$, which yields the system transfer function $-(1+(1-x)G)/(1+xG)$. This reduces to the forms previously given for $x=0$ and $x=1$, and to -1 (flat) when $x=0.5$. Gain G can be written as AN/D, where N and D are the numerator and denominator terms of G, and A is the centre-frequency gain through the entire filter pathway (including the A_0 term, defined previously), which is equal to 10 in the present circuit. The system transfer function now becomes

$$-(D+(1-x)AN)/(D+xAN).$$

Setting ω to 1 for convenience, we have $N = s/Q$ and $D = s^2 + s/Q + 1$.

We are now in a position to explain the curves in Fig. 5(a). When R_{19} is close to the full boost setting, x is close to 0. As x increases (R_{19} rotated away from full boost), the numerator of the transfer function is reduced, but since A is large this reduction will be small compared to the increase in the denominator. Thus when x is close to 0 the transfer function can be approximated by $-AN/(D+xAN)$, which is to say that as R_{19} is rotated away from the full boost position, the change in frequency response can be accounted for primarily by a change in the pole positions. The denominator of the transfer function is $s^2 + (1+xA)s/Q + 1$, so the effect of increasing x is to reduce the Q to a new value Q', equal to $Q/(1+xA)$, which explains the curves in Fig. 5(a). An analogous argument can of course be developed to explain the symmetrical form of the corresponding cut curves when x is close to 1.

Whether or not the behaviour in Fig. 5(a) is desirable is a debatable point, but fortunately one can have it both ways! Suppose the feedback end of R_{19} is grounded instead of being connected to the output of IC_2. The circuit will now only boost, and the transfer function will be $-(D+(1-x)AN)/D$. Since A is large, the transfer function can be approximated by $-(1-x)AN/D$ except when x is close to 1, so the major effect of changing x is now to change the centre-frequency gain without affecting the Q. The response curves obtained under these conditions are shown in Fig. 5(b).

There are several ways of modifying the circuit to obtain these curves, and the method used is to some extent a matter of personal choice, but here are three! First, changeover switches could be used to ground one or other end of R_{19} to obtain either the boost or cut curves. Second, the gain of IC_3 could be made variable, when the curves in Fig. 5(b) would be obtained with R_{19} at maximum boost. The third possibility is my personal favourite, and I have indicated it on the circuit diagram (Fig. 3). This is to use a centre-tapped control for R_{19} (I really must apologise for continually recommending obscure potentiometers!) and to ground the tap via S_2 to obtain the Fig. 1(b) curves. The advantage of this method is that the boost/cut setting is determined only by the control setting, just as before, although the control law will be changed. As will be appreciated from the change in the form of the transfer function, the centre-frequency gain will approach 0dB less rapidly as the control is rotated towards its midpoint when the centre tap is grounded. Well, you can't have everything!

This effect can be reduced, however, by connecting a 1kΩ resistor between the slider of R_{19} and ground when the centre tap is grounded, which will mean using a double-pole switch for S_2. The compensation is not exact, but it reduces the worst-case centre-frequency mismatch to below 3dB. The ultimate solution would be to replace R_{19} by two parallel chains of resistors, one of which is grounded at the centre, and to select a point along one or other chain by a multiway switch. The resistor values would be chosen to obtain equal dB steps between the switch points, and it would probably be much quicker to determine the correct values

State-variable filters

Although the present circuit uses the state-variable approach to provide only a bandpass filter, highpass (HP) bandpass (BP) and lowpass (LP) outputs are available simultaneously, as indicated in the accompanying derivation. Note that the basic form of the transfer function is quite simple, and the final expression is relatively cumbersome only because of the form of the a_1 and a_2 coefficients. The derivation also shows more clearly how it is possible to change the Q independently of the centre-frequency gain. Since $a_2 = 1/Q$, we merely have to vary a_1 and a_2 together, which is achieved in the present circuit by a variable resistor (R_{20}) to ground from the a_1 and a_2 summing point. This obviously requires that the two signals go to the same amplifier input, and since the a_2 coefficient must be positive, a_1 has to be as well. If this facility is not required, a_1 could of course be either positive or negative.

A further advantage of the state-variable approach is that it can provide *any* second-order funetion, although this has not been exploited in the present circuit. The HP, LP and BP outputs are summed by a further amplifier (see ref. 4 for the system transfer function), which allows the corresponding reject functions to be synthesised. By making the appropriate coefficients variable, it would be possible to generate a continuous range of bandpass and band reject functions within the filter itself, rather than by changing the position of the filter within an amplifier feedback loop as in the present circuit. There may not be much to choose between the two methods, but I preferred the feedback loop method because it can be used with any kind of filter, and any number of filters can be placed within a single feedback loop as shown in Fig. 6. It also allows the choice of two sets of frequency response curves (see Fig. 5), which may not be so easy to arrange by the other method.

$$BP = HP \times -1/R_5C_1S, \text{ where } S = j\omega$$

$$LP = BP \times -1/R_6C_2S = HP \times 1/R_5C_1R_6C_2S^2$$

$$HP = a_1 \times input + a_2 \times BP - a_3 \times LP$$

$$a_1 \times input = HP - a_2 \times BP + a_3 \times LP =$$

$$HP\left(1+\frac{a_2}{R_5C_1S}+\frac{a_3}{R_5C_1R_6C_2S^2}\right)$$

$$HP\ input = \frac{a_1}{1+\frac{a_2}{R_5C_1S}+\frac{a_3}{R_5C_1R_6C_2S^2}}$$

$$= \frac{a_1R_5C_1R_6C_2S^2}{R_5C_1R_6C_2S^2 + a_2R_6C_2S + a_3}$$

Referring to Fig. 2, the a coefficients are

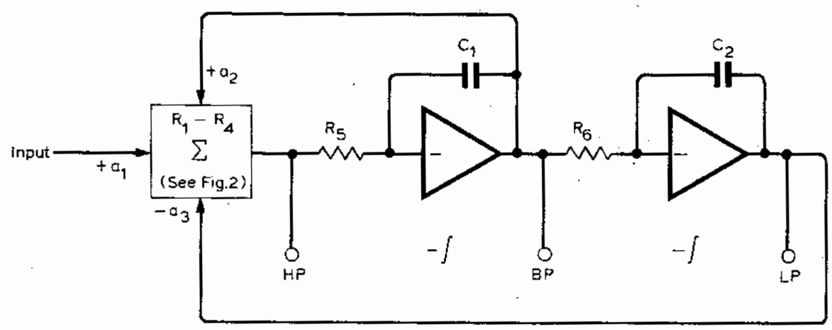
$$a_1 = \frac{R_2(R_3+R_4)}{(R_1+R_2)R_4};\ a_2 = \frac{R_1(R_3+R_4)}{(R_1+R_2)R_4};\ a_3 = \frac{R_3}{R_4}.$$

Thus the complete highpass transfer function is

$$\frac{HP}{input} = \frac{R_2(R_3+R_4)}{(R_1+R_2)R_4} \times \frac{R_5C_1R_6C_2S^2}{R_5C_1R_6C_2S^2 + \frac{R_1(R_3+R_4)}{(R_1+R_2)R_4}R_6C_2S + \frac{R_3}{R_4}}$$

The bandpass and lowpass transfer functions are obtained by multiplying the high-pass transfer function by $-1/R_5C_1S$ and $1/R_5C_1R_6C_2S^2$.

When $R_5C_1 = R_6C_2$, $a_2 = 1/Q$.



by measurement than by calculation!

We can now consider the remaining aspects of circuit performance. When LM318 devices are used, the distortion is extremely low, and it was difficult to make any reliable measurement at midfrequencies. For a +20dBm (22V peak-to-peak) output signal at 20kHz, however, I managed to obtain a value of 0.015%, but this fell rapidly as the signal level was reduced. In general, the control settings affected the distortion only insofar as they changed the output signal level. This excellent performance is a result of the very high bandwidth (15MHz) and slew rate (70V/μs) of the LM318, but there is sufficient latitude to allow the use of other devices for many applications.

The best alternative devices are the various families of f.e.t. input high bandwidth operational amplifiers, and the circuit performance was also evaluated with one of these, namely the Fairchild μAF356, which has a 5MHz bandwidth and 15V/μs slew rate. Using this device throughout, the distortion for a +20dBm output at 20kHz rose to 0.05%, and when the Q was increased at high centre frequencies, the centre-frequency gain also increased slightly – an effect not observed with the LM318. Device substitution showed that the effect, which occurred only at high Q, originated at IC_4, and most of the extra distortion was generated by IC_3. Both the LM318 and the μAF356 have an input voltage noise of around 15nV $\sqrt{Hz}$ at midfrequencies, but the μAF356 may be slightly quieter since its input current noise is lower. I have not given any noise specification for the circuit, since the amplitude and frequency content of the noise will be greatly affected by the control settings, and to quote one or two blanket values could be misleading. However, I have tried to keep circuit impedances below 10kΩ wherever possible in order to keep the noise down to a level where it should be dominated by that of the ICs.

As mentioned previously, the circuit could be made more compact by the use of dual or quad i.cs. A possible i.c. is the Texas TL074 series, but the bandwidth is only 3MHz, which will limit the performance at high frequencies. By the time this article appears, however, a wider range of quad devices may have become available.

Many applications will call for the use of more than one equaliser section, and the sections can be combined in two ways. The easier method is to connect them in series, and if the connection is permanent the buffer stage IC_1 can be omitted from the subsequent sections. This approach is best suited for a modular design, as it allows each section to be used independently. The other method is for the filter sections to be connected in parallel (as for a graphic equalizer), where IC_3-IC_6 and all the controls are duplicated, but share the same IC_1 and IC_2. The circuit configuration must be changed, however, to allow the filter outputs to be combined, and the modified circuit is shown in Fig. 6. Circuit IC_2 is now a virtual-earth mixer, which can sum any number of filter outputs without interaction, but to achieve this the outputs have to be sent to the inverting instead of the noninverting input of IC_2, so we need to make a compensating phase reversal in the filter path. In theory, this could be done by moving the filter input connection from the noninverting to the inverting input of IC_4, but we would then lose the interaction which allows the Q to be varied independently of the centre-frequency gain (see appendix). The solution adopted is to rewire IC_3 as an inverting amplifier, which has the minor disadvantage that the gain of this stage will interact slightly with the setting of R_{19}, but the effect will make no difference in practice. Each filter section can be switched out independently as shown in Fig. 6, or they can be switched out together by a single switch between the common ends of the R_5 resistors and the inverting input of IC_2.

How does the circuit sound? My advice is to build it and find out! At low Q, the response can be corrected over a large frequency range by as few as two stagger-tuned sections, and in this mode the circuit is a very useful "shelf" filter. As the Q is increased, the circuit becomes more like a graphic equalizer, and ultimately resembles a musical instrument! A wide variety of special effects can be created by tuning one or two high-Q sections up and down the audio band, and at high Q the circuit also becomes a useful notch filter. Obviously this design is too complex for it to pose a significant threat to the popularity of the Baxandall tone control, even though it is a lot more versatile. But if you really *prefer* the mode of action of the Baxandall circuit, don't worry – this design will give quite a reasonable approximation to it if you tune one section to each end of the audio band and set them both to minimum Q. Now all you have to do is to label one control bass and the other one treble. Well, I told you it was versatile!

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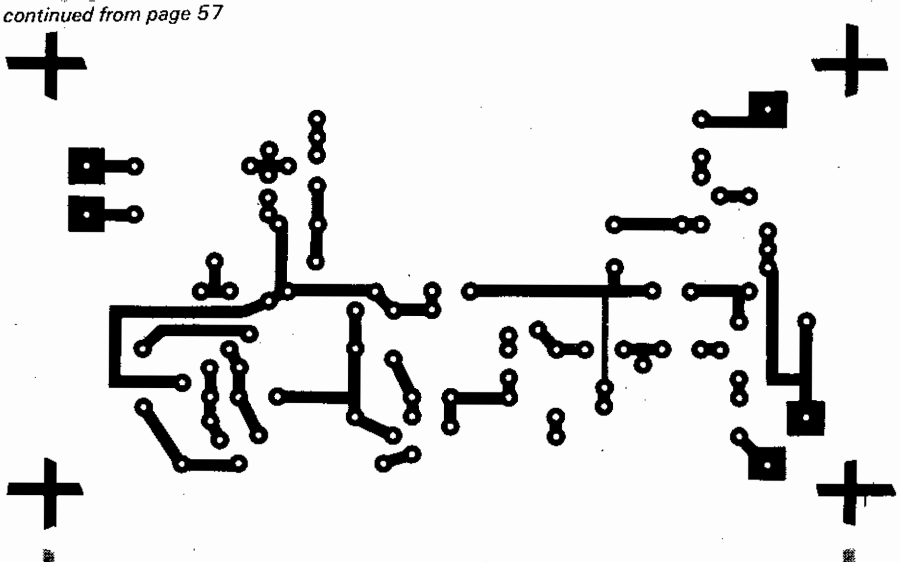
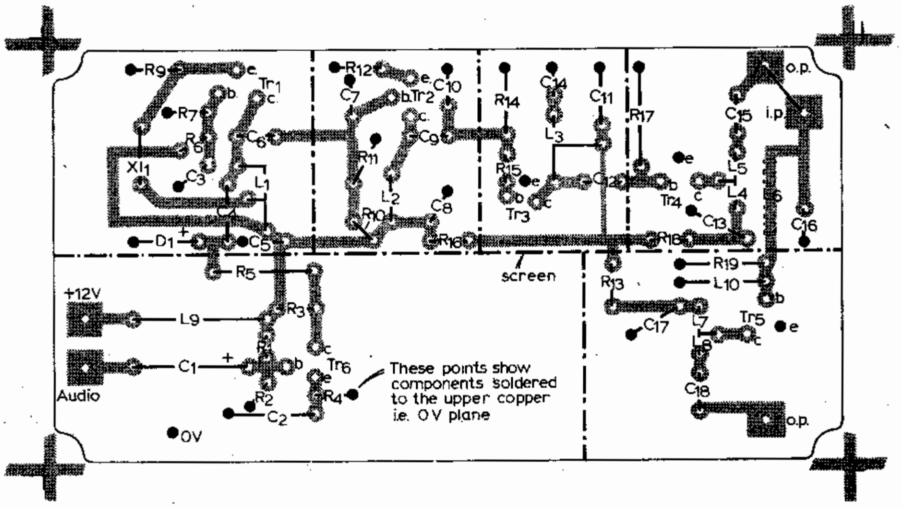




BY JOHN H. ROBERTS

TAILOR THE SOUND OF YOUR AUDIO SYSTEM WITH THIS STEREO PARAMETRIC EQUALIZER

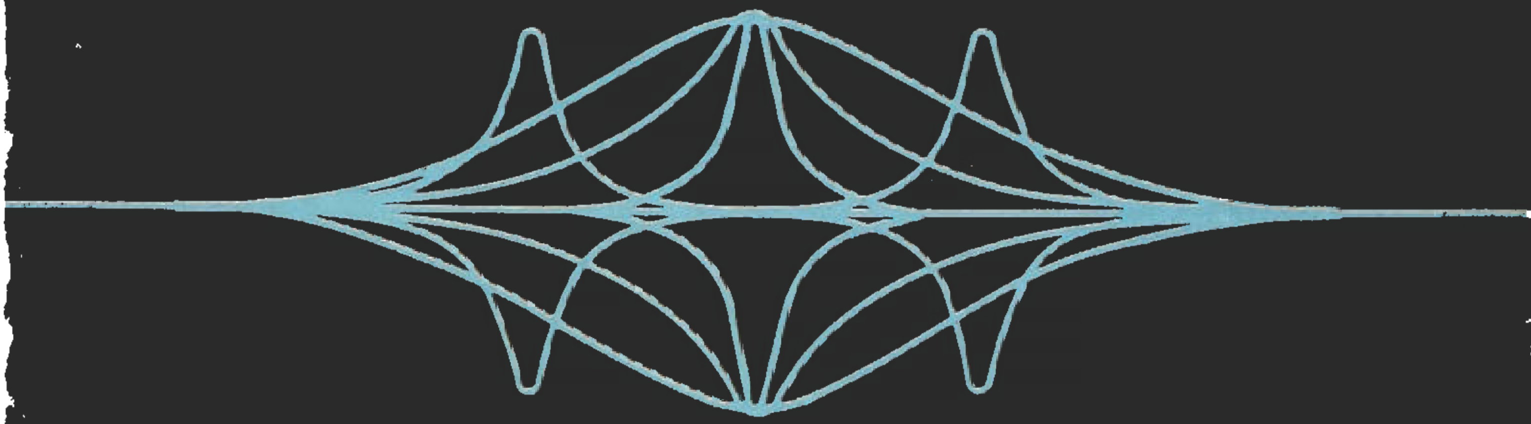
Low-cost, high-performance component employs BIFET operational amplifiers, can be powered by dc or ac sources.

AS THE state of the audio art has matured, whole new families of sophisticated components generically known as *signal processors* have become available for use in sound systems. Among the most popular category of signal processors is the equalizer. And the subcategory that has generated the most excitement among serious audio enthusiasts and sound professionals is the parametric equalizer.

As its name implies, each of the parametric equalizer's key parameters—its center frequency, filter bandwidth or Q, and amount of boost or cut introduced—can be independently adjusted. This provides extraordinary flexibility, allowing the user to tailor equalization to the precise needs for a particular program or room/system combination.

Presented here is a two-band parametric stereo equalizer with several features that commend it to the audiophile. It has been designed so that the Q and BOOST/CUT controls interact to compensate for the perceived change in loudness as filter bandwith increases or decreases. Furthermore, the circuit employs high-performance BIFET op amps, which combine the best of both junction-field-effect and bipolar-junction transistors in each amplifier. It can be powered by either the ac line or a 12-to-30-volt dc supply, making it equally "at home" in fixed, mobile, or portable applications. Finally, the Parametric Equalizer is relatively inexpensive—a line-powered stereo kit costs $99.00.

A Short Course in Equalization. Although last month's POPULAR ELECTRONICS contained a comprehensive



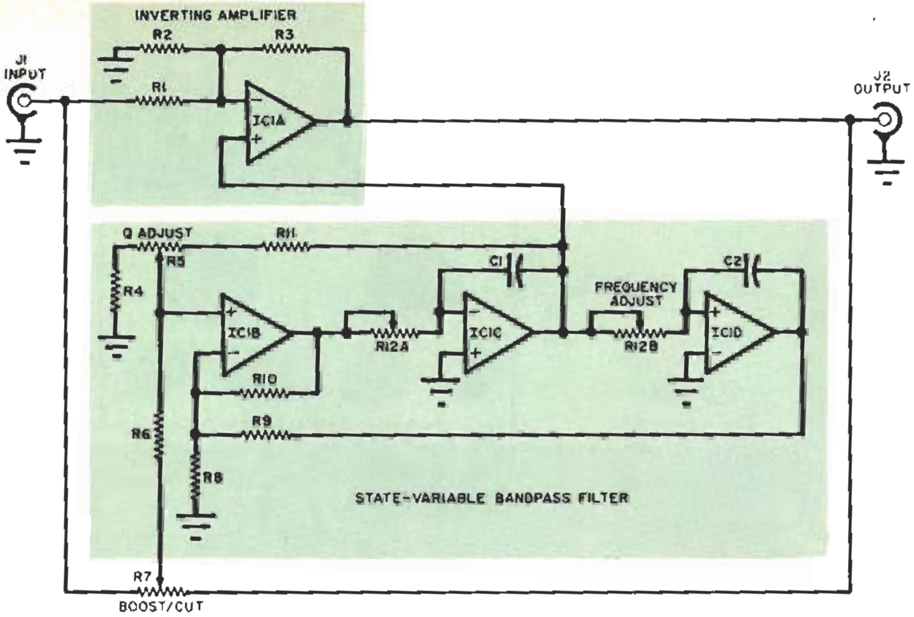
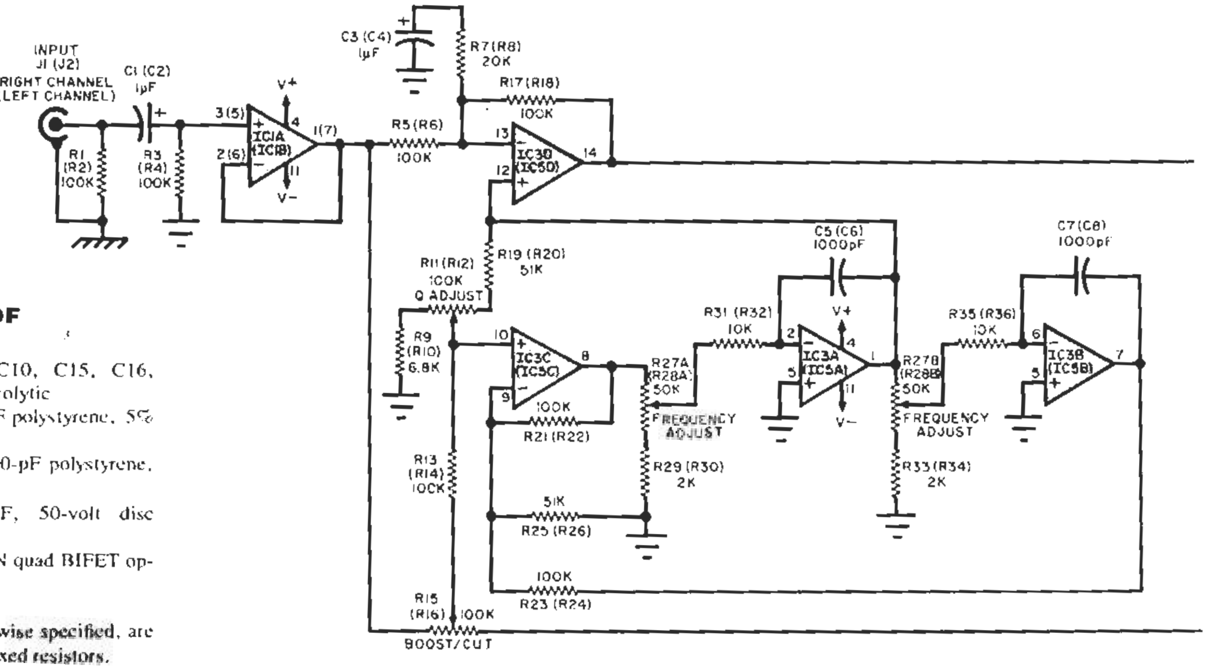


Fig. 1. Simplified schematic of one channel of equalizer shows that an inverting amplifier is interconnected with a modified state-variable active bandpass filter.

article about equalization ("The Art of Equalization" by Ethan Winer), here's a brief overview of the subject. The category of signal processors known as equalizer can be broken down into three subcategories: tone control or shelving types; graphic or peaking equalizers; and parametrics. All three are capable of boosting or cutting signal levels, but differ in the manner in which they generate the boost or cut, in the shapes of the frequency-response curves they produce, and in the size of the band of frequencies which they affect.

Tone controls are characterized by a gradual transition between the non-boosted and fully boosted (or unattenuated and maximally attenuated) frequency bands, levelling off to a fixed amount of boost or cut. The resulting frequency-response curve takes on the appearance of a shelf, giving rise to the name *shelving equalizer.*

Graphic equalizers divide the audio spectrum into a given number of bands with individual boost/cut controls for each band. The transition between the unaffected and fully affected regions is determined by the number of bands in the graphic equalizer. An inexpensive five-band or two-octave (so called because each band is two octaves wide) has a lower filter Q and therefore more effect over frequencies somewhat removed from the band of interest than a sophisticated professional equalizer which breaks the audio spectrum down into 30, one-third-octave-wide bands. In most consumer graphic equalizers, the center frequency of each band is fixed, although some more sophisticated units (and most professional graphics) allow the user some leeway in setting the center frequencies. The family of frequency-response curves generated by a graphic



MAIN PARTS LIST (TWO CHANNELS OF EQUALIZATION)

C1, C2, C3, C4, C9, C10, C15, C16, C20—1-μF, 25-volt electrolytic
C5, C6, C7, C8—1000-pF polystyrene, 5% tolerance
C11, C12, C13, C14—8200-pF polystyrene, 5% tolerance
C17**,C18**,C19*—0.1-μF, 50-volt disc ceramic
IC1 through IC5—TL074CN quad BIFET operational amplifier
J1, J2, J3, J4—Phono jack
The following, unless otherwise specified, are ¼-watt, 5% carbon-film fixed resistors.
R1 through R6,R13,R14,R17,R18,R21,R22, R23,R24, R37,R38,R45,R46,R49,R50, R53, R54, R55, R56, R74, R75—100,000 ohms
R7, R8, R39, R40, R63, R64, R67, R68—20,000 ohms
R9,R10,R41,R42—6800 ohms
R11,R12,R15,R16,R43,R44,R47,R48—100,000-ohm, linear-taper potentiometer
R19,R20,R25,R26,R51,R52,R57,R58—51,000 ohms
R27,R28,R59,R60—dual 50,000-ohm linear-taper potentiometer
R29,R30,R33,R34,R61,R62,R65,R66—2000 ohms
R31, R32, R35, R36—10,000 ohms
R69,R70—100 ohms
R71**,R72**,R73*—10 ohms
Misc.—Printed circuit board, pc standoffs, IC sockets or Molex Soldercons, hookup wire, shielded cable, solder, machine hardware, control knobs, suitable enclosure, etc.
*—Dc version only
**—Ac version only

equalizer resembles a series of peaks and valleys. That's why some audiophiles refer to graphic equalizers as "peaking" types.

The parametric equalizer is a variation on the graphic equalizer theme. In addition to an individual boost/cut control, each band of a parametric equalizer also has center-frequency and bandwidth or filter Q controls. This means that the amount of boost or cut introduced, the center frequency of the band of equalization, and the bandwidth within which the equalization is applied (as well as the transition between the frequencies that are unaffected and those which are boosted or cut the most) are all independently variable. The parametric equalizer thus gives its user the ultimate in control over the sound recorded on tape or reproduced by his speakers.

About the Circuit. A simplified schematic of the Parametric Equalizer is shown in Fig. 1. Only one equalizer section of one channel's circuit is shown, and input buffering and output decoupling details are omitted. Similarly, power supply connections are not shown. It can be seen that the simplified schematic is that of an inverting amplifier (*IC1A*, *R1*, *R2*, and *R3*) interconnected with a modified "state variable" active bandpass filter. Such a filter is composed of two active integrators connected in cascade (*IC1C*, *IC1D*, and associated passive components) and a differential amplifier (*IC1B* and associated passive components).

PERFORMANCE SPECIFICATIONS (Supplied by the Author)

Center frequency range: 40 to 16,000 Hz in two bands—40 to 960 Hz, 500 to 16,000 Hz

Frequency response: 3 to 100,000 Hz, +0 dB, −1 dB with all controls at their flat settings

Input impedance: 50,000 ohms

Input/output gain: 0 dB

Intermodulation distortion (SMPTE): Less than 0.007%

Maximum output: 8 volts rms into a 10,000-ohm load when powered by ±15-volt supply

Maximum boost/cut: ±20 dB at 0.16-octave bandwidth

Output impedance: 100 ohms

Output noise: −70 dBm unweighted, −89 dBm "A" weighted

Range of Q adjustment: 0.16 to 2 octaves (−3-dB bandwidth)

Total harmonic distortion plus noise: below 0.04% from 20 to 20,000 Hz

This circuit was chosen for use in the Parametric Equalizer because its center frequency and Q can be varied independently of each other. The filter's center frequency is selected by adjusting dual potentiometer *R12*. Filter bandwidth and Q are dependent upon the values of *R4* and *R11* and the setting of potentiometer *R5*. For the component values employed in this project, filter bandwidth and Q can be adjusted over a range of 0.16 to 2 octaves at the −3-dB points. (The relationship between bandwidth at the −3-dB points and filter Q is given by the simple equation $BW_{-3\,dB} = 1/Q$.)

To convert a state variable active bandpass filter into the desired all-pass circuit with adjustable boost and cut, a potentiometer (*R7*) is connected between the inverting input and the output of unity-gain amplifier *IC1A*. The wiper of this potentiometer is connected to the input of differential amplifier *IC1B*. Signals appearing at the output of integrator *IC1C*, which are inverted with respect to those appearing at its input, are applied to the noninverting input of *IC1A*.

When the wiper of *R7* is at the *J1* extreme of its travel, the bandpassed signal adds to the input signal, boosting the amplitude of signals within the filter's passband. When the wiper is at the *J2* extreme of its travel, the bandpassed

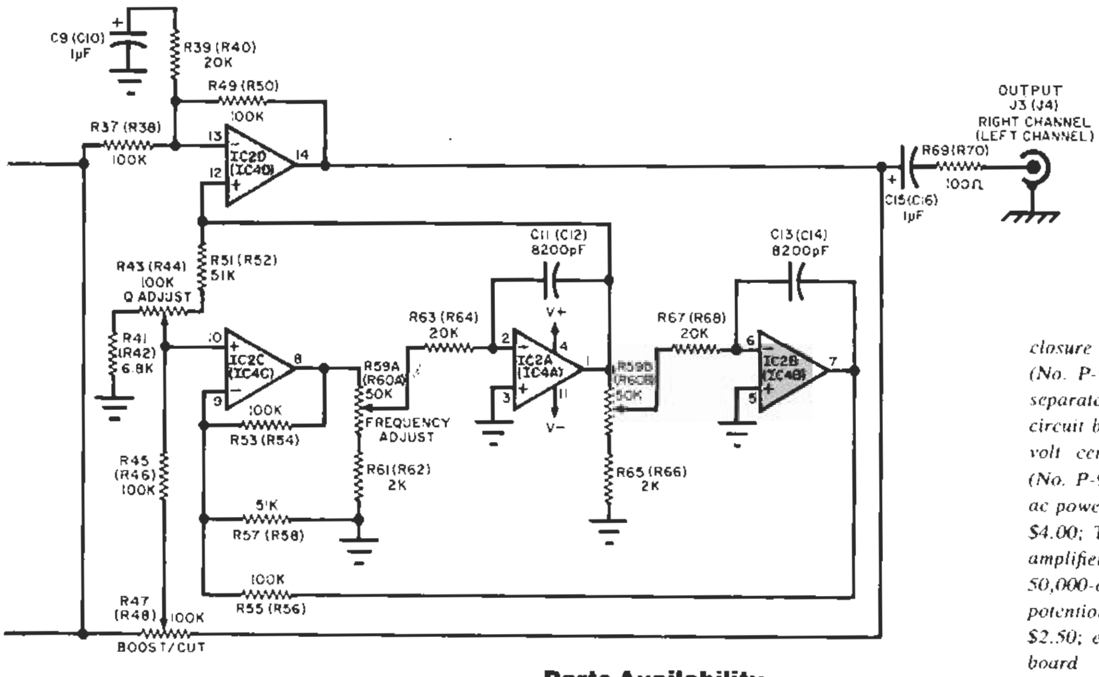


Fig. 2. The complete circuit for a two-channel equalizer. Part numbers not in parentheses are for right channel of a stereo system, others are for left channel. For components with asterisks, see Figs. 3 and 4.

Parts Availability

Note—The following are available from Phoenix Systems, 375 Springhill Road, Monroe, CT 06468 (203-261-4904): Complete kit of parts including enclosure for ac-powered stereo equalizer (No. P-94-S) for $99.00; Complete kit of parts including enclosure for dc-powered stereo equalizer (No. P-94-SC) for $89.00. Also available separately: etched and drilled main printed circuit board (No. P-94-AB) for $8.00; 20-volt center-tapped stepdown transformer (No. P-94-T) for $6.50; etched and drilled ac power supply board (No. P-04-PSB) for $4.00; TL074CN quad BIFET operational amplifier IC (No. P-94-C) for $2.50; dual 50,000-ohm, linear-taper, closely tracking potentiometer (No. P-94-2X50KB) for $2.50; etched and drilled dc power supply board (No. P-94-PSBC) for $2.00; 100,000-ohm, linear-taper potentiometer (No. P-94-100KB) for $1.00; p.c.-mount, push-on/push-off power switch (No. P-94-SL) for $1.00. Add $1.00 handling charge for orders less than $10.00. Add $1.00 for COD orders. Canadians add $2.50 postage. Connecticut residents add state tax.

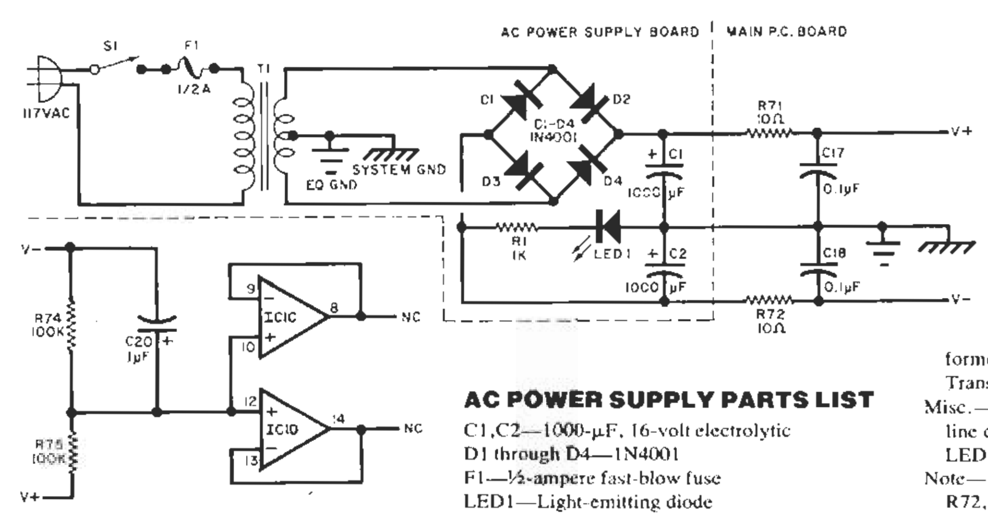


Fig. 3. Schematic of power supply to use with an ac source. It is a conventional full-wave circuit giving plus and minus 15 volts to ground.

AC POWER SUPPLY PARTS LIST

C1,C2—1000-μF, 16-volt electrolytic
D1 through D4—1N4001
F1—½-ampere fast-blow fuse
LED1—Light-emitting diode
R1—1000-ohm, ¼-watt, 5% resistor
S1—Spst switch
T1—20-volt, center-tapped stepdown transformer, secondary rating 100 mA (Signal Transformer No. ST-4-20 or equivalent)
Misc.—Printed circuit board, pc standoffs, line cord, strain relief, hookup wire, solder, LED mounting collar, hardware, etc.
Note— Components C17, C18, C20, IC1, R72, R74 and R75 are mounted on the project's main printed circuit board and are included in the Main Parts List. See Fig. 1 for Parts Availability.

signal subtracts from the input signal, attenuating input signals within the passband of the active filter. Finally, when the wiper of *R7* is at the midpoint of its travel, the output of *IC1A* cancels out that portion of the input signal appearing at the wiper because the two signals are 180° out-of-phase. This means that no signals are routed to the bandpass filter, the filter generates no output, and has no effect on *IC1A*. The result is that inverting amplifier *IC1A* exhibits a flat frequency response.

There are two equalizer sections for each signal channel. (Only one section is shown in Fig. 1.) The center frequency of the low-band equalizer can be adjusted from 40 to 960 Hz, and that of the high-band equalizer from 500 to 16,000 Hz. Both the setting of the BOOST/CUT potentiometer and the value of filter Q determine the amount of boost or cut introduced by each equalizer section. The maximum boost or cut is ±20 dB at a filter bandwidth of 0.16 octave, and ±12 dB at a bandwidth of 2 octaves. This interaction makes the Q control more convenient to use because parametric designs not incorporating it often require readjustment of equalizer gain after the filter Q has been changed.

The master schematic of the main Parametric Equalizer circuit is shown in Fig. 2. The most likely application for this project is in a stereo sound system, so the schematic describes a two-channel equalizer. All components pertaining to the right signal channel have part numbers not shown in parentheses. Those for the left channel, however, have part numbers which are shown in parentheses. The rest of this discussion will refer only to the right signal channel but is equally applicable to the left.

Input signals are applied to jack *J1*, where *R1* and *R3* (which are effectively in parallel) provide a high-impedance load. Capacitor *C1* blocks any dc level that might be accompanying the input signal. Buffering is accomplished by voltage follower *IC1A* which isolates the input from the rest of the circuit. Output signals from the voltage follower are then applied to two cascaded equalizer sections, each of which employs a TL074CN quad BIFET operational amplifier IC.

Each section closely resembles the simplified schematic shown in Fig. 1. That employing *IC3* is the high-band equalizer circuit. Its center frequency is adjustable by means of dual potentiometer *R27* over a range of 500 to 16,000 Hz. Potentiometer *R11* is the filter's Q ADJUST control and potentiometer *R15* (along with the Q of the filter) determine the amount of boost or cut introduced.

The second equalizer circuit (the one

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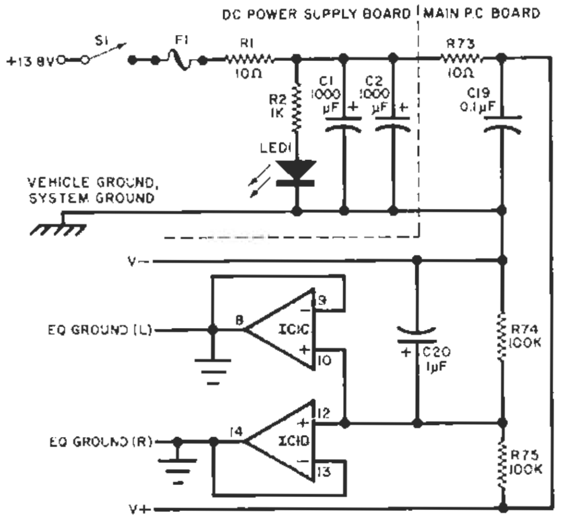

Fig. 4. Use this circuit if a dc supply is to be employed. The IC voltage followers derive an artificial equalizer ground.

DC POWER SUPPLY PARTS LIST

C1,C2—1000-μF, 16-volt electrolytic
F1—½-ampere fast-blow fuse
LED1—Light-emitting diode
R1—10-ohm, ¼-W, 5% resistor
R2—1000-ohm, ¼-W, 5% resistor
S1—spst switch
Misc.—Printed circuit board, pc standoffs, machine hardware, etc.
Note—Components C19, C20, IC1, R73, R74, and R75 are mounted on the project's main printed circuit board and are included in the Main Parts List. See Fig. 1 for Parts Availability.

(Continued from page 50)

employing *IC2*) is the low-band unit. Dual potentiometer *R59* allows adjustment of its center frequency over a range of 40 to 960 Hz. The filter's Q is adjusted by varying the setting of potentiometer *R43*. Signals within the filter passband can be boosted or cut by means of potentiometer *R47*.

Output signals from *IC2D* are coupled to output jack *J3* via *C15* and *R69*. The electrolytic capacitor blocks any dc offset appearing at the output of the operational amplifier and the resistor provides decoupling. Signals can be routed from the output jack back to the tape monitor loop of a preamplifier or receiver, if that is where drive signals were taken, or to the input of the power amplifier if drive is obtained from the preamplifier output.

Power supply details are omitted from the main schematic for simplicity's sake, but each IC's power supply pins are denoted. The Parametric Equalizer can be powered by either the ac line or a 13.8-volt dc automotive electrical system. Schematic diagrams of the ac and dc supplies are shown in Figs. 3 and 4, respectively. The ac supply is a conventional full-wave circuit employing a 20-volt, center-tapped transformer. Diodes *D1* through *D4* rectify the low-voltage ac into bipolar, pulsating dc which is filtered by *C1* and *C2*. Light-emitting diode *LED1* functions as a pilot light. All components except for decoupling resistors and capacitors *R71*, *R72*, *C17* and *C18* are mounted on a separate power supply circuit board. The output of the supply is ±15 volts dc.

The dc supply employs voltage divider *R74R75* and voltage followers *IC1C* and *IC1D* to derive an artificial equalizer ground at one-half the full voltage delivered by the electrical system powering the circuit. Note, however, that the voltage divider should be connected to the noninverting inputs of the voltage followers even if the ac supply is used to power the circuit. This is done to prevent unwanted oscillation. The outputs of the followers are left uncommitted when the ac power supply is employed.

Light-emitting diode *LED1* acts as a pilot light, and electrolytic capacitors *C1* and *C2* filter any noise present on the dc line. Note that decoupling components *R73* and *C19* as well as the "equalizer ground" deriving circuit are located on the main printed circuit board.

In the dc-powered equalizer, the negative supply voltage pins of the quad operational amplifier IC's are connected to the vehicle and sound system ground (shown in the schematics as "earth

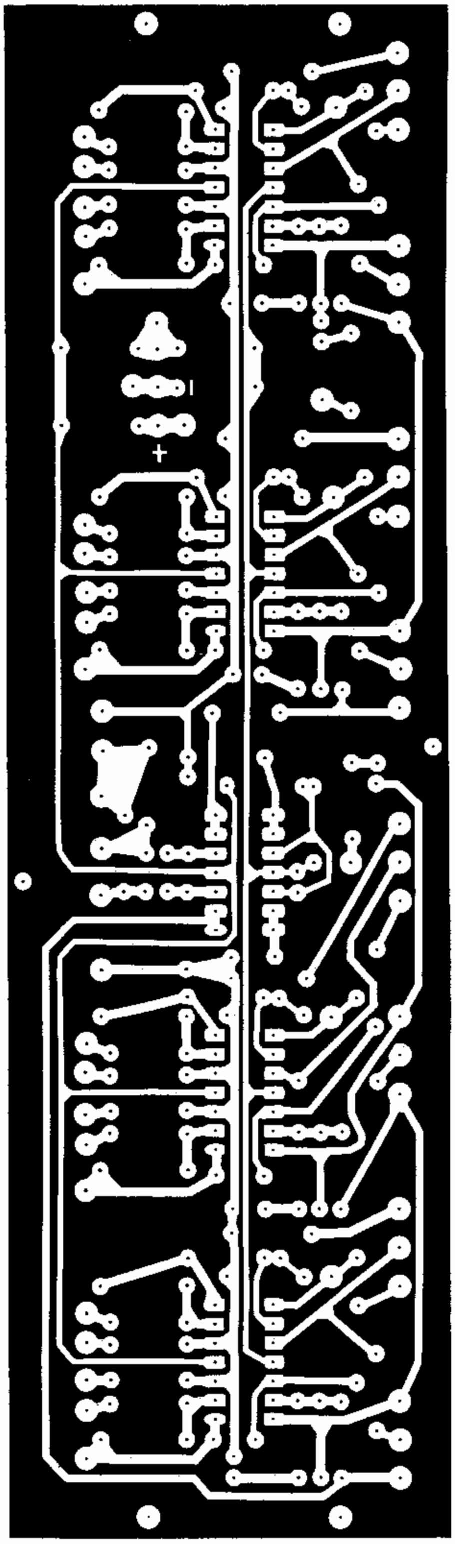

Fig. 5. Actual-size etching and drilling guide for the main printed circuit board.

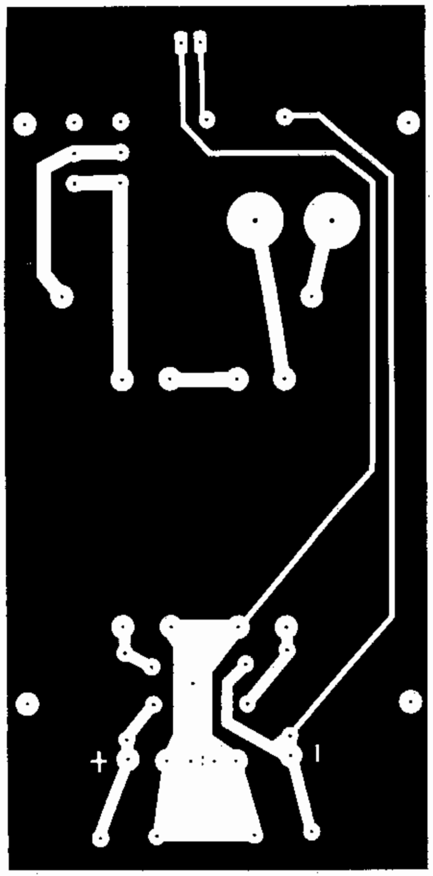

Fig. 6. Use this board for an ac power supply.

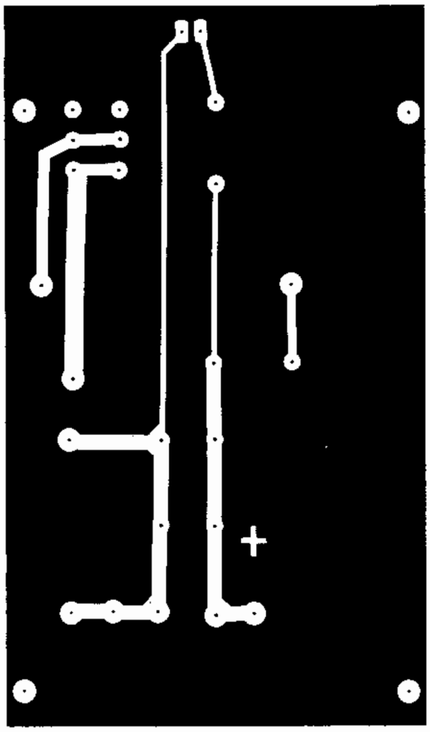

Fig. 7. If a dc supply is available, use this board.

ground" symbols). The artificial grounds derived by *IC1C* and *IC1D* are shown as conventional "chassis ground" symbols. Note that the grounds within the equalizer sections (for example, the noninverting inputs of the op amp integrators) are artificial grounds above vehicle and system ground.

Capacitive coupling between the input jack and the op amp input buffer and between the output of the high-band equalizer and output jack prevents dc offsets both internal and external to the equalizer from having a deleterious effect on the performance of the entire system. It is because of the dc offsets present in the dc-powered equalizer that the "hot" sides of the input and output jacks are returned to system ground but the signal

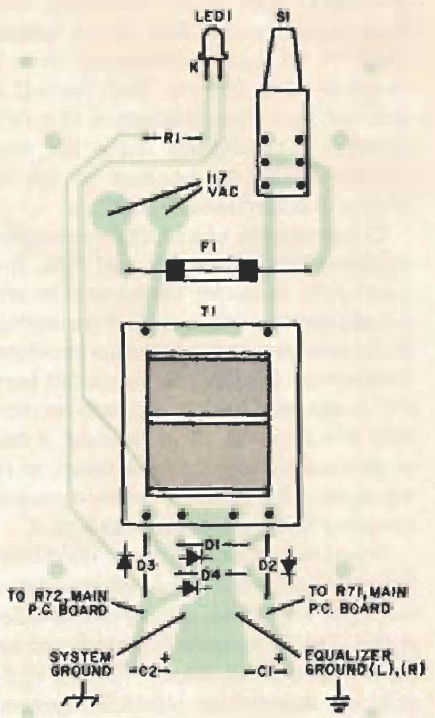

Fig. 9. Component placement for the ac power supply.

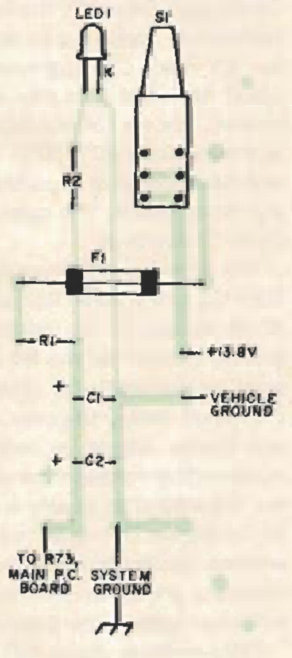

Fig. 10. Component placement for the dc power supply.

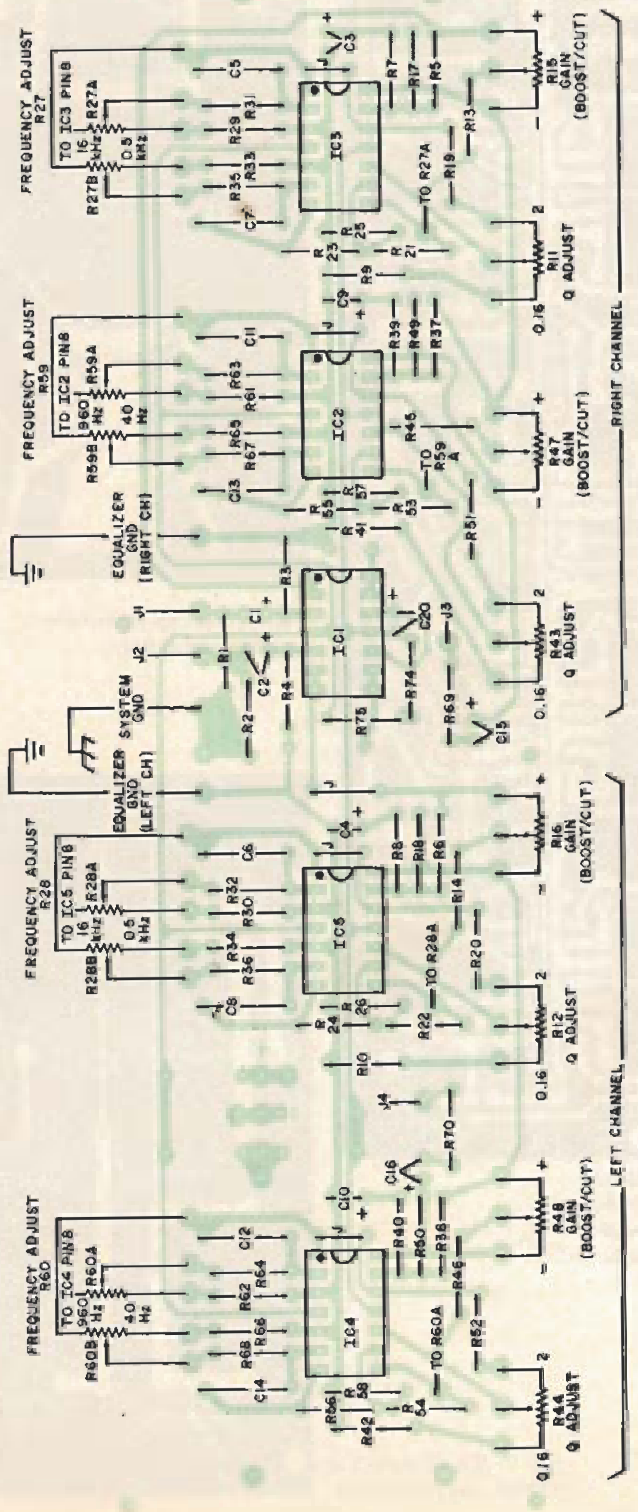

Fig. 8. Component placement for the main pc board for the equalizer. Note vacant pads near upper left to make connections to power supplies.

paths within each equalizer circuit are referenced to the artificial grounds. In the ac-powered equalizer, however, the bipolar dc voltages furnished by the power supply obviate the need for separate system and equalizer grounds. The two are shown connected together in the schematic of Fig. 3.

Results of tests on the prototype performed by the author at his own lab are shown in the box. You will note that all performance specifications but one are identical for both the dc and ac versions of the Parametric Equalizer. The one area in which the two differ is in the maximum voltage swing that can be generated at the output jack. The reason for this is that in the ac-powered equalizer the potential difference between the V+ and V− supply rails is 30 volts, but the potential difference between the supply rails in the dc-powered equalizer is less than half of this value if the dc power source delivers 13.8 volts. However, even in this situation there exists substantial headroom—most (if not all!) autosound power amplifiers require far less drive than 13.8 volts peak-to-peak to develop their maximum levels of output power. Greater output voltage swings can be obtained by increasing the voltage provided by the dc source. The circuit as shown can be used with supplies from +12 to +30 volts.

Construction. The use of printed circuit assembly techniques is recommended. Full-size etching and drilling guides for the main, ac power supply, and dc power supply circuit boards are shown in Figs. 5, 6, and 7, respectively. The corresponding parts placement guides are shown in Figs. 8, 9 and 10.

Mount all components on the circuit boards as shown in the parts placement guides. Begin by installing the jumpers on the main pc board. Then install the fixed resistors and nonpolarized capacitors. Taking care to observe polarities and pin basings, mount the electrolytic capacitors and semiconductors. The use of IC sockets or Molex Soldercons will facilitate replacement of ICs should that become necessary. Interconnection between the main board and the phono jacks and potentiometers can be made using flexible hookup wire. If desired, signal paths between the board and the jacks can be made with shielded cable.

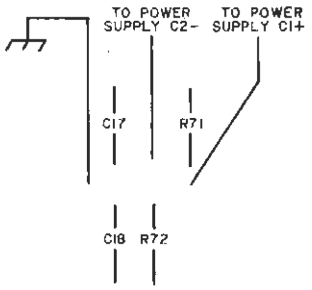

Fig. 11. Special wiring of the main pc board for use with an ac power supply.

This will not be necessary, however, if the project is housed in a grounded metallic enclosure. Special wiring of the main board for ac-powered operation is shown in Fig. 11. Wiring details for dc operation are shown in Fig. 12.

Assemble either the dc or ac power supply to fit the intended application of your Parametric Equalizer. Observe the polarities of electrolytic capacitors and diodes, including the LED pilot light. Fuse *F1* mounts directly on the board and should be soldered to it using pigtail leads. The author designed the power supply boards to accommodate a special push-on/push-off power switch, but any panel-mount switch can be used.

When assembling the circuit boards, be sure to use the minimum amount of heat and solder consistent with the formation of good solder connections. Scrutinize your work after the boards have been completed, paying close attention to polarities, pin basings, power supply wiring and interconnection between the two circuit boards. Make sure that no solder bridges have been created inadvertently.

When all wiring has been completed, mount the circuit boards, jacks and controls in a shielded enclosure. A photograph of the author's ac-powered prototype is shown in Fig. 13. Route power leads out of the enclosure using a protective strain relief. Connect the power leads to a suitable source. Using shielded patch cords, route line-level signals from the tape monitor output of your preamplifier or receiver (or from the preamplifier output) to input jacks *J1* and *J2*. Similarly, patch signals from output jacks *J3* and *J4* back to the tape monitor loop or to the input of the power amplifier. The project is now ready for use.

Using the Parametric Equalizer. Because this project is so flexible, there is no one "correct" way to use it. Its variable Q and center frequency allow the user to boost or attenuate a select group of frequencies. A high Q restricts the boost or cut introduced to a narrow part of the spectrum (less than one octave). A low Q causes broader changes to be introduced.

Adding some sharp boost at the very low and high ends of the audio spectrum allows the user to compensate for speaker rolloff. A broad dip inserted at the midband makes possible the simulation of a loudness contour to enhance low-level listening. The Parametric Equalizer is also adept at compensating for unwanted room resonances. A high-Q cut can reduce audio output at the resonant frequency with little effect on nearby frequencies.

The usual technique for coping with room resonances is as follows. Drive the system with a wideband audio signal

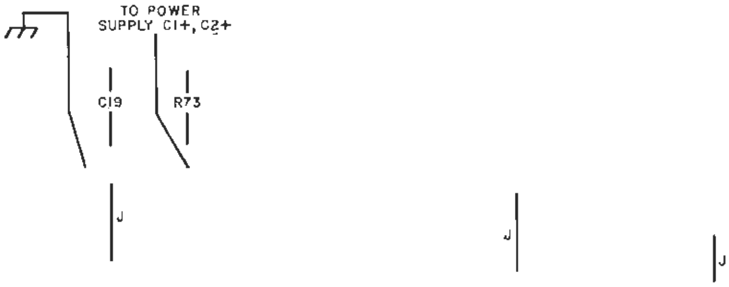

Fig. 12. Special wiring of the main pc board for use with a dc power supply. Note two jumpers on IC1 at right.

and boost the bass region using the Parametric. Using a high Q setting, vary the center frequency of the low-band equalizer until you discover the room's fundamental resonant frequency. (That's the one at which the walls start shaking and the furniture moves around the floor.) Now reduce the setting of the BOOST/CUT control for more even-sounding bass. The high-band equalizer can be used to brighten up a room that is too "dead" acoustically or to attenuate treble response in a room that is too "alive."

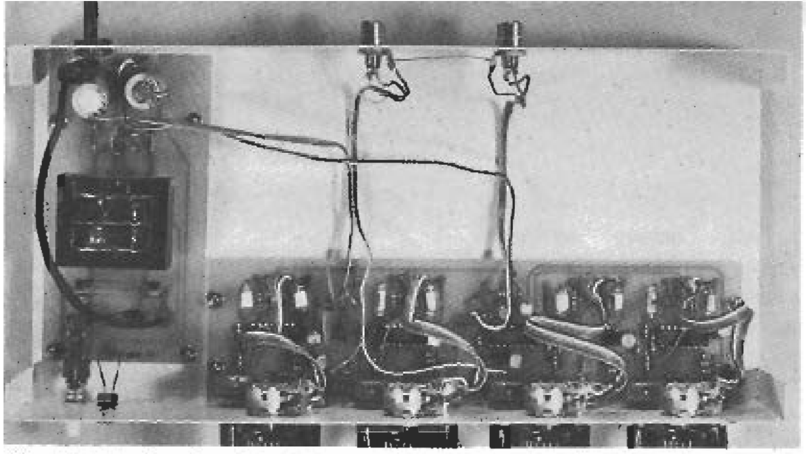

Fig. 13. Interior view of prototype using ac power supply.

You will undoubtedly find other uses for this versatile project. Those who listen to music analytically will appreciate the ability to zero in on one particular instrumental (or human) voice. Amateur recording engineers can employ the Parametric to tailor the sounds of a mix. And, of course, anyone whose speakers have response irregularities will be able to smooth them out.

One word of caution—don't blindly apply large amounts of deep bass and extreme treble boost in an attempt to flatten the response of your system at the upper and lower limits of the audible spectrum. Experience has shown that room/system combinations are best equalized by first employing acoustic methods, followed by electronic equalization. For example, you should first try repositioning the loudspeakers, modifying the absorption coefficients of the room, and adjusting the speakers' crossover level controls (if any).

Most often, a lack of deep bass and extreme highs is due to the limitations of dynamic drivers. Don't try to force flat response out of your speakers by cranking up the BOOST/CUT controls. The results of such attempts frequently include overloaded amplifiers, excessive distortion, and blown voice coils. Remember—equalization should be introduced *intelligently*.

In Conclusion. We have presented a stereo Parametric Equalizer project that is well suited for home, mobile, and portable applications. It provides a high level of performance and the flexibility of control inherent in the parametric design, enough flexibility for most readers. Those who require more bands of equalization per channel can reproduce two or more complete equalizers and connect them in cascade for even greater control over the sounds they record or reproduce. ◇

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