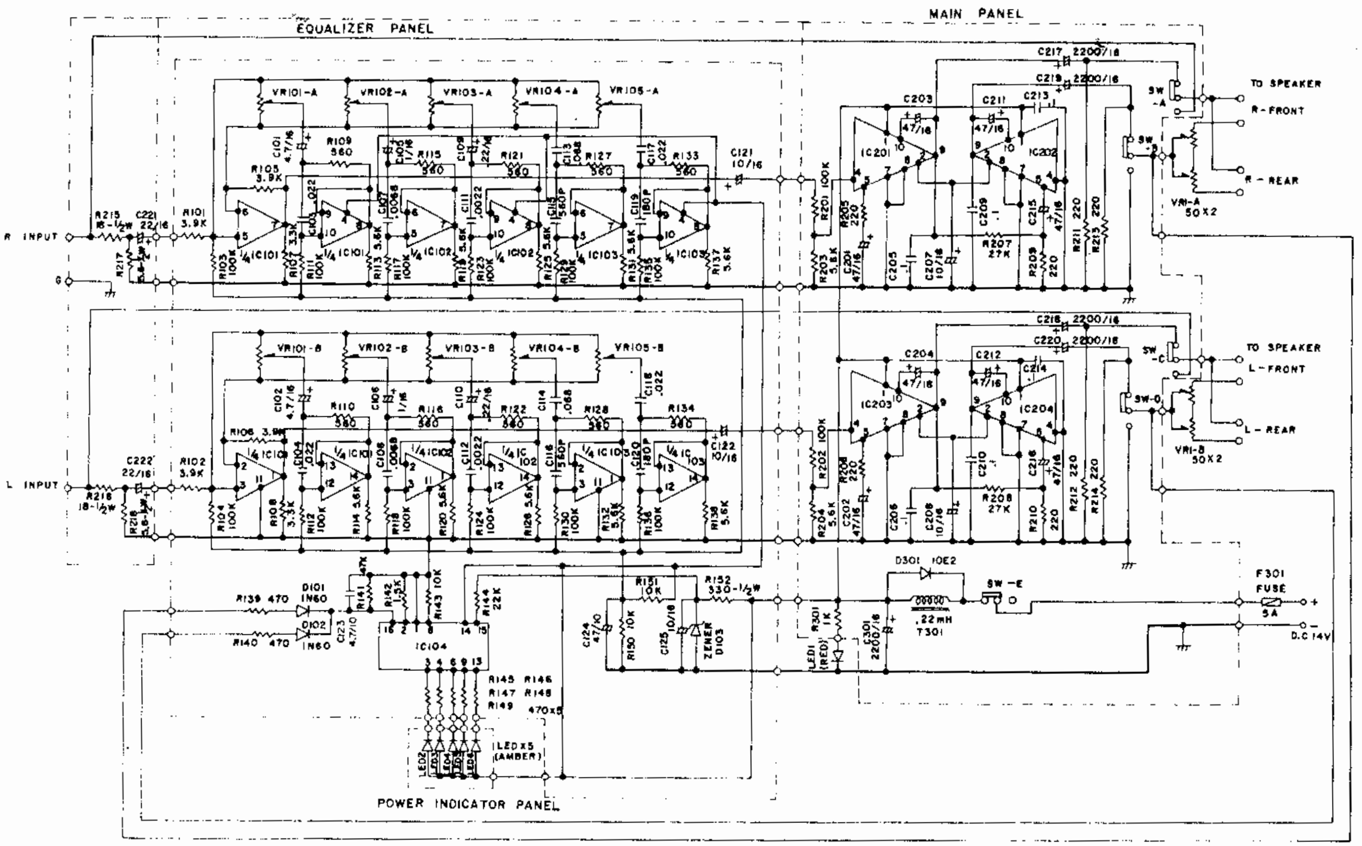

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SCHEMATIC DIAGRAM

The Optimized GRAPHIC EQUALIZER

Part 2—An integral analyzer for accurately setting up the audio equalizer

By Joe Gorin

IN THE first part of this article, we presented a new kind of equalizer circuit that offers high performance at an economical price. This month we will construct the Flatness Analyzer, an accessory used to adjust the equalizer rapidly and accurately.

Circuit Operation. Figure 6A is a block diagram of the equalizer/analyzer combination (part of which is identical to Fig. 1). The analyzer plugs directly into the equalizer. Figure 6B is a block diagram of the equalization test procedure.

Here's how the Flatness Analyzer tests one channel (the right) of the Optimized Equalizer. Pink noise is applied to the right-channel input of the equalizer. The equalized output of the right channel is then fed through an amplifier and speakers into the room. From here, the microphone picks it up.

The signal is then amplified by the microphone preamp and applied to the left-channel input of the equalizer, as well as two filters in the analyzer. The outputs of these 12 filters drive simple biased-diode detectors and a bank of 12 meters to show the deviations from flatness. If the system response is flat, all meters will have equal deflections. The output of the left channel is grounded to prevent the amplified microphone signal from passing back out through the left speaker and perturbing the measurements or causing oscillations.

To test the left channel, the interconnecting plug is reversed and offset in its socket, and the above procedure is repeated with left and right channels reversed.

Figure 7 is the schematic of the analyzer. Integrated circuits *IC2* and *IC3* constitute a digital white-noise generator. The circuits in *IC3A* and *IC3B* form a square-wave oscillator with an output frequency of about 100 kHz. This clocks 18-stage shift register *IC2*, which keeps shifting the output of *IC3D*, the exclusive-OR function of the 14th and 17th stages of the shift register. These taps (14 and 17) are chosen so that the register outputs random ones and zeroes; it only repeats after going through all but one of the 2^{17} possible states. This is called a pseudo-random sequence generator (since it repeats, it isn't truly random). Its output spectrum is very white if you pass the digital output through a low-pass filter. Integrated circuit *IC3C* and its associated components ensure that *IC2* cannot get locked up in the all-zeroes state.

Components *R29* through *R32* and *C20* through *C23* are a pinking filter. The gain vs. frequency of this network falls off at 3 dB per octave on the average, about half as fast as a single RC filter. The noise is amplified by *IC4B* and rolled off at high frequencies to compensate for the increased gain of the testing channel at high frequencies (due to the reduction in input attenuation as explained previously).

The output is ac coupled with *C25*, and its level is controlled with *R32*. The level could be controlled with the stereo's master volume control, but having a control on the analyzer is a real convenience. The signal from the level control now passes to the channel under test.

The stereo speakers convert the noise to sound, which comes back for analysis through the microphone, *MIC1*. A small electret is used here, which has typical accuracy of ±1 dB with help from the preamp, *IC1B*. This stage provides a gain of 27, and *C33* and *R44* tame an upper-midrange peak that is common to most inexpensive electret microphones.

The microphone signal is further amplified in *IC1A* and passed through *R48* and *C32* to the testing channel's filters. Resistor *R48* is provided as protection in case the input to the equalizer is not disconnected.

Besides the ten filters in the equalizer, *IC1C* and *IC1D* filter the frequencies around 40 to 100 Hz and 140 Hz to help adjust the bottom bands of the equalizer.

The filtered signals from the equalizer are ac-coupled by *C1* through *C10* (to remove the dc components) and detected by *D3* through *D12*. To minimize the errors due to the *on* voltage of these diodes, a small current is passed through *D15* and buffered by *IC4A* to offset the positive side of the meters by approximately the diode *on* voltage. As a result, the meters respond to the average value of the noise level, which is a much more accurate parameter than the peak response frequently used in such an analyzer.

The outputs of *IC1C* and *IC1D* are passed through RC filters *R18*, *R19*, *R24*, and *R25* and *C12* and

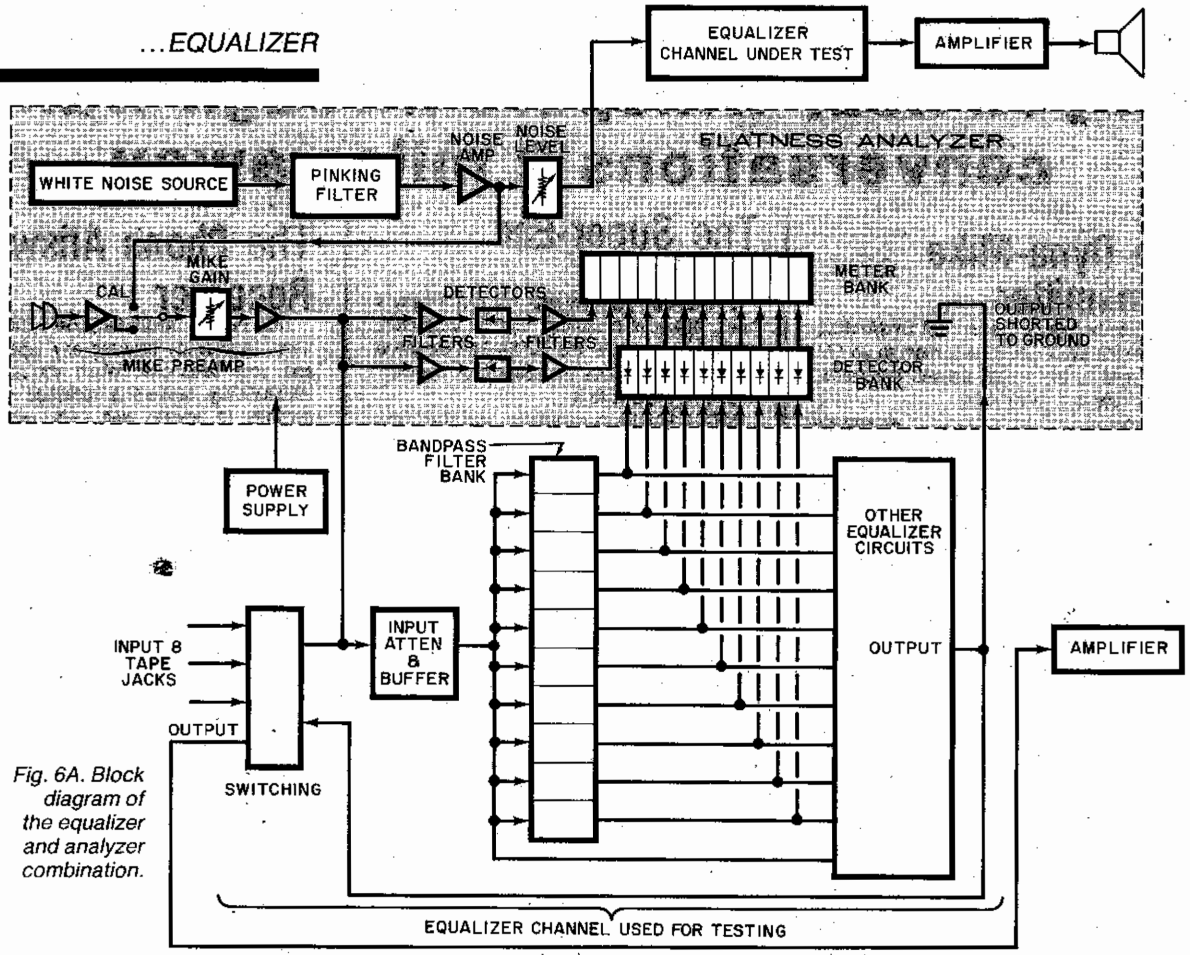


Fig. 6A. Block diagram of the equalizer and analyzer combination.

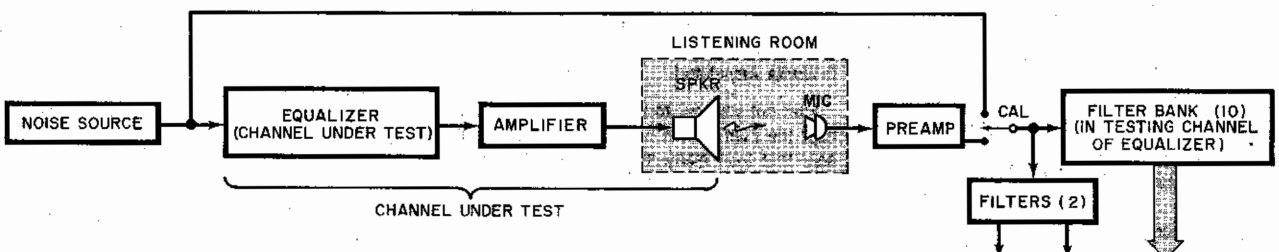


Fig. 6B. Block diagram of the equalization test procedure with the analyzer and equalizer connected.

C16 to reduce the fluctuations of the bottom band meters and to reduce the gain, in order to make up for the effect of the attenuator at the input of the testing channel on the ten other bands.

Resistor *R17* and diode *D16* provide a +9-V supply for the microphone and white-noise generator, and also supply bias for *IC1*.

Switch *S1* allows the response of the analyzer to be observed without the speaker-microphone link, to see how flat it is. This calibration permits adjustments to be made that will provide compensation for component tolerance errors, especially in the meter sensitivities (±1 dB) and pinking-filter components.

Construction. Figure 8 is the foil pattern for the analyzer pc board, and Fig. 9 is the foil pattern for the interconnection pc board. A component-placement diagram for the analyzer is given in Fig. 10.

Solder all components to the board, except the slide potentiometers. Don't forget the two jumpers. Carefully orient the ICs, diodes, and electrolytic capacitors according to pin number or polarity. Integrated circuits *IC2* and *IC3* are CMOS, and thus static-sensitive; so don't remove them from their conductive packaging until you are ready to install them. Then discharge yourself, your soldering iron, and the pc traces to ground.

Connect the microphone element, *MIC1,* to the shielded pair cord and solder the cord to the appropriate pc board holes—red wire for positive, white for signal, and shield for ground. Connect a stiff piece of wire over the shield and solder to the two holes right behind it to act as a strain relief.

The connection to the equalizer is through a DIP plug. Cut a standard DIP-plug to DIP-plug 16-wire cable in half and solder the unterminated wires to the appropriate pads of the DIP pattern on your board (the wires will alternate sides). Or just install a whole DIP-plug right in the pattern. Pass the wires across *R35*'s position, and then mount *R35*

PARTS LIST

C1 through C10,C12,C13,C16,C17, C24,C30,C32—10-μF, 25-V aluminum electrolytic
C11,C28,C29—0.1-μF, 50-V ceramic disc capacitor
C14—0.0047-μF, 5% polyester capacitor
C15,C18,C19,C25—0.1-μF, 5% polyester capacitor
C20—0.022-μF, 5% polyester capacitor
C21—0.0068-μF, 5% polyester capacitor
C22,C26,C34—0.0022-μF, 5% polyester capacitor
C23—0.001-μF, 5% polyester capacitor
C27—24-pF, 5% capacitor
C31—Not used
C33—390-pF ceramic disc capacitor
D1-D15—1N4148
D16—9.1-V zener (1N5239 or 1N960)
IC1—RC4136 quad op amp
IC2—CD4006 18-stage shift register
IC3—CD4070 quad ex-OR gate
IC4—LM358 dual op amp
M1-M12—200-μA 1-kilohm edgewise meter
MIC1—Electret microphone element
P1—16-pin DIP plug
The following are ¼-W, 5% carbon-film resistors unless otherwise noted:
R1 through R10,R40—470 ohms
R11,R39,R46,R49—1.5 megohms
R12 through R16—Not used
R17,R20,R26,R28,R48—2.2 kilohms
R18,R19,R50—8.2 kilohms
R21—300 kilohms
R22,R34,R43—3.9 kilohms
R23—39 kilohms
R24,R25—11 kilohms
R27—62 kilohms
R29—270 kilohms
R30,R37,R38—150 kilohms
R31,R41—47 kilohms
R32,R33,R36,R44,R47—15 kilohms
R35,R45—50-kilohm potentiometer
R42—100 kilohms
S1—Spst slide switch
Misc.—Pc board for analyzer, press-on rubber feet (4), 16-wire ribbon cable, jumper wires, etc.

Note: The following are available from Symmetric Sound Systems, 856 Lynn Rose Ct., Santa Rosa, CA 95404 (707-546-3895): complete Optimized Equalizer kit (EQ-4) with unfinished walnut end panels at $100; complete Analyzer kit (AN-1) at $60. Also available separately: horizontal and vertical pc boards for Equalizer (EQ-4PC) at $17; analyzer and interconnect pc boards (AN-1PC) at $13; slide potentiometers (#EQ-4SP) at $.95 each; quad op amp IC #4136 at $1.75 each; set of ICs for analyzer (#AN-1IC) at $6.00. Wall-plug transformer (#EQ-4PT) at $7.50. Minimum order $10.00. All prices include shipping on prepaid orders in the U.S. Canadians add $4.00 shipping and handling. California residents, add sales tax.

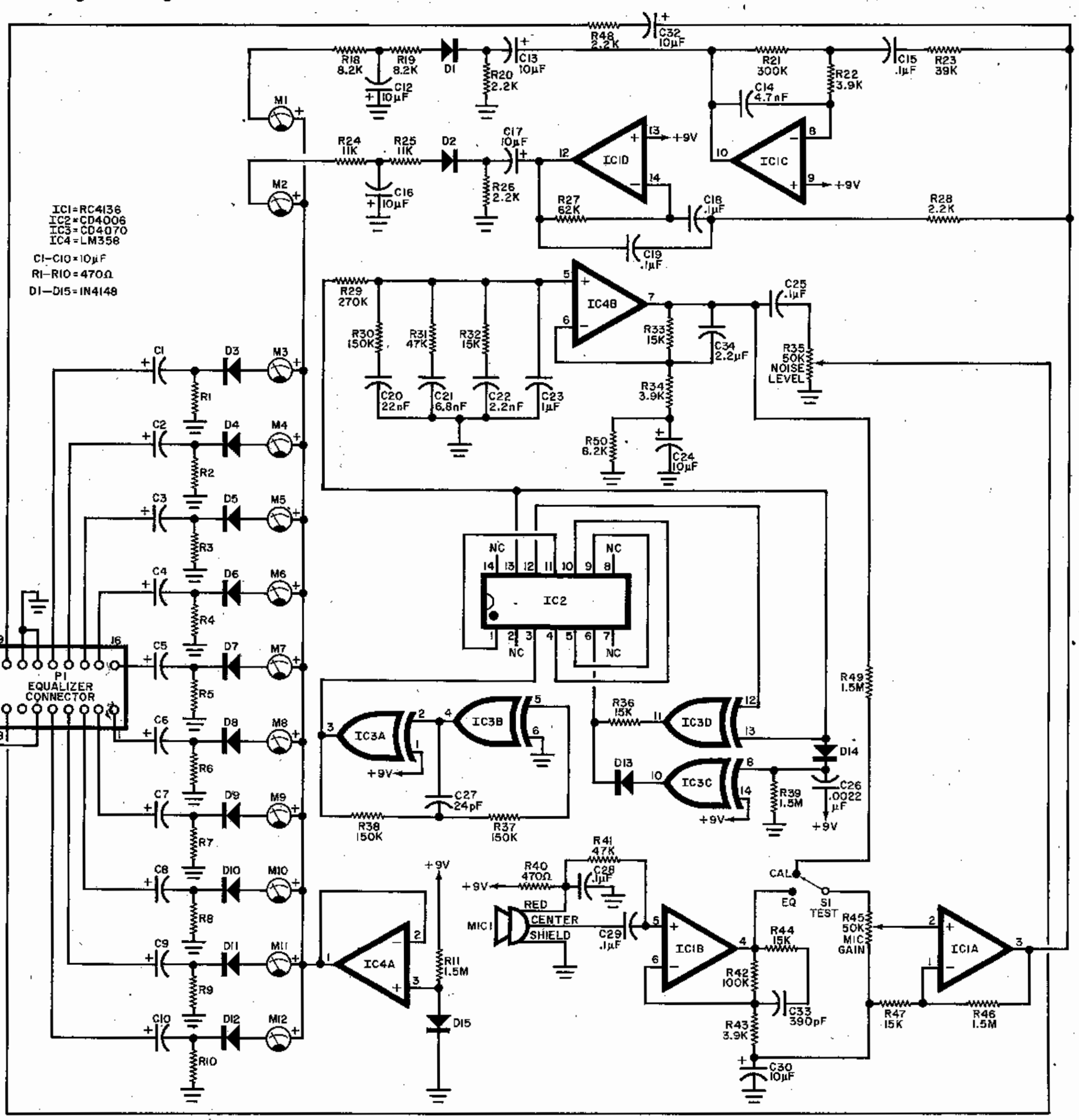


Fig. 7. Schematic of the circuit in the analyzer.

over them as a strain relief. Also mount the other slide potentiometer in its proper location.

In the prototype, the bases of the edgewise meters were glued to the pc board and wired with short jumpers. It is a good idea to use stick-on rubber feet to prevent shorting to the chassis of the equalizer or scratching it during use.

Since the analyzer is a sophisticated accessory and not for display, to save effort and expense, you need not put it in a fancy chassis.

Adjustment and Use. Using the Optimized Equalizer and the Analyzer combination is easy because all the information you need is right in front of you at all times.

With the power off, connect the equalizer outputs to your stereo. Do not connect the equalizer inputs to anything. Connect the analyzer to the equalizer and turn the slide pots to OFF. Set the TEST switch to EQ and the EQUALIZER switch to IN. Set all the equalizer controls to 0 dB. Place the measuring microphone at your favorite listening location. Apply power to the equalizer/analyzer and your stereo.

Adjust the mike gain upwards until there is significant deflection of some of the meters. This point shows how large the room noise is. Back down on the gain until there is no more than 10% deflection on any meter. Now slowly advance your noise-level control and stereo-volume control until you are getting an average of over 70% of full deflection on your meters. Depending on the ambient levels in your room, this is likely to be relatively loud.

Adjust the bands of the channel

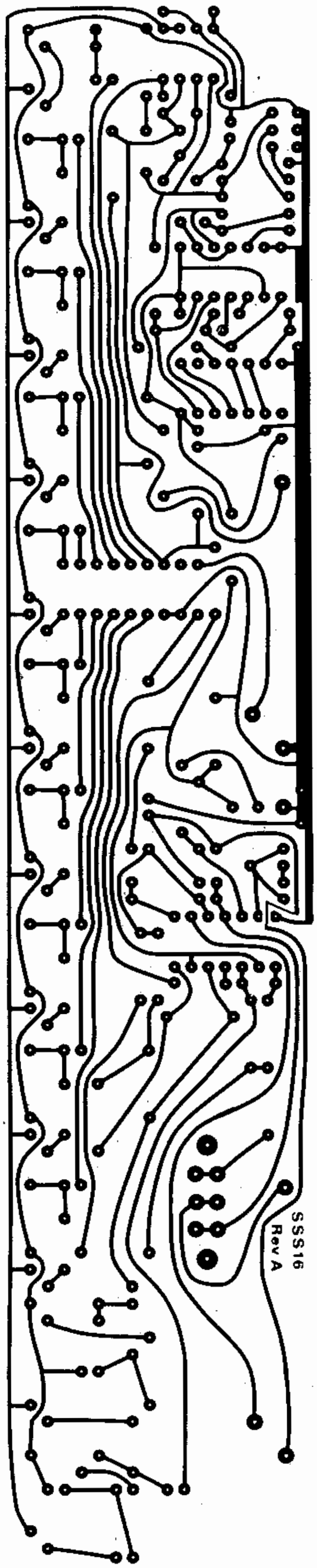
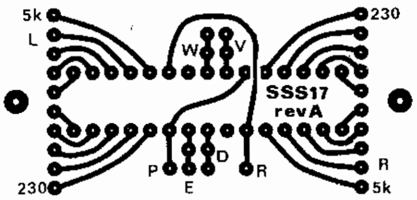


Fig. 8. Foil pattern for the analyzer pc board.

Fig. 9. Foil pattern for connector board to equalizer.



WHAT'S WRONG WITH THE FLATNESS ANALYZER?

According to traditional thinking, there is quite a bit wrong with the analyzer. First, its output devices are meters. Unlike bar-graph LEDs, meters cannot be easily read from far away. They are also slow and cannot show the dynamics of music well, due to mechanical inertia. But we are not building a music analyzer; we are building a flatness analyzer. It is designed to be placed next to the equalizer so that the controls can be adjusted while watching the meters. Only the microphone needs to be usable from a distance, and it comes with a long cord.

The slowness of the meters is in fact desirable because it evens out the fluctutations in the noise levels. Actually the meters act as filters without extra components to do that filtering (except in the lowest bands, where the fluctuations are slow enough that additional filtering is desirable). However, the most important reason for using meters is that they give better resolution and "feel" for that signal level. Their fluctuations can be averaged visually much faster and more accurately than LEDs, especially in designs with 2.5 dB/step LED resolution.

Next, the Flatness Analyzer will not analyze music. Since the signal levels in the testing channel must be adjusted to drive the meters appropriately, this channel cannot be used to process music. This precludes the fascinating light-shows of some analyzers, but it is necessary for the economy of reusing the equalizer's filters. We're out for performance here, not a show.

Finally, the Flatness Analyzer does not have a top-end meter to help adjust the equalizer's 10-kHz control. One is easily added, but it is not worthwhile for a number of reasons. First, a microphone that has even marginally predictable response in the top octave will cost more than the entire equalizer/analyzer combination; using it would produce the worst kind of diminishing return on your investment. Secondly, recorded music in the top octave is notoriously variable in relative level due to varying microphone techniques and engineer's tastes. Finally, all speakers, microphones, musical instruments and ears are extremely directional at high frequencies. Unlike the situation at lower frequencies where most of the signal you equalize has been reflected from room boundaries; at high frequencies, you would be equalizing the direct signal from the loudspeakers. The desired ratio of this signal level to the reverberantly measured levels at other frequencies is not well controlled.

Thus, no one equalizes for a flat high end. Rather, they try to accomplish some smooth roll-off. The author strongly recommends setting this band by ear and resetting it (and perhaps the top two or three narrow bands slightly) according to the particular piece of music being played. ◇

under test by reducing the level of the band corresponding to the meter with the highest deflection. After you have adjusted a few bands this way, continue by moving the bands either up or down to come as close as possible to uniform deflection of all bands. Adjust the noise level as necessary to keep the average deflection at about 70%.

The noise source, being pseudorandom, audibly repeats every 1.5 seconds, and the meters will show this periodicity. When fine tuning, visually average the motion during this interval. When the result is close to flat, switch the TEST switch to CAL, adjust the MIKE GAIN for 70% average deflection, and observe the errors of the test system. Then switch back to EQ and fine-tune the equalizer to match the CAL response, which will be slightly different than truly flat. Then turn everything off, switch the connection from the analyzer to the equalizer, and repeat for the other channel. Then remove the analyzer and connect the equalizer normally.

Hints on Equalizing. Over the long term, the sound from your system will be exceptionally smooth and accurate. But be wary of short-term reactions. After listening so long to the errors that your system and room make, your mind gets accustomed to these distortions of reality and expects them. Thus, any change toward either more or less realistic sound is initially perceived as unnatural. Also, the equalization technique given will reduce the overall level somewhat. Unless you compensate by increasing the volume control setting, you are likely to initially consider the sound to be poorer when equalized.

But give yourself about 15 minutes with your de-resonated stereo and then switch to unequalized. You will notice a hollow, boxy sound that you missed before because you were so used to it. Now simply switch back to equalized sound and you will find some really fine listening. ◇

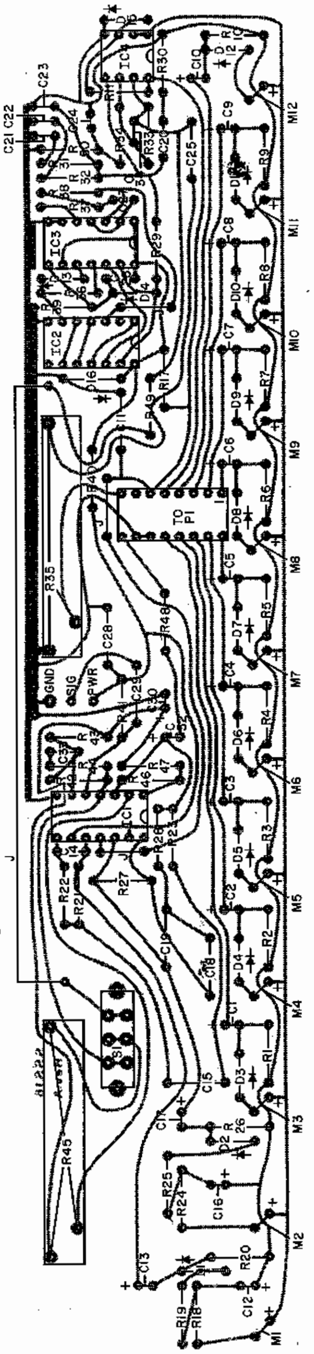
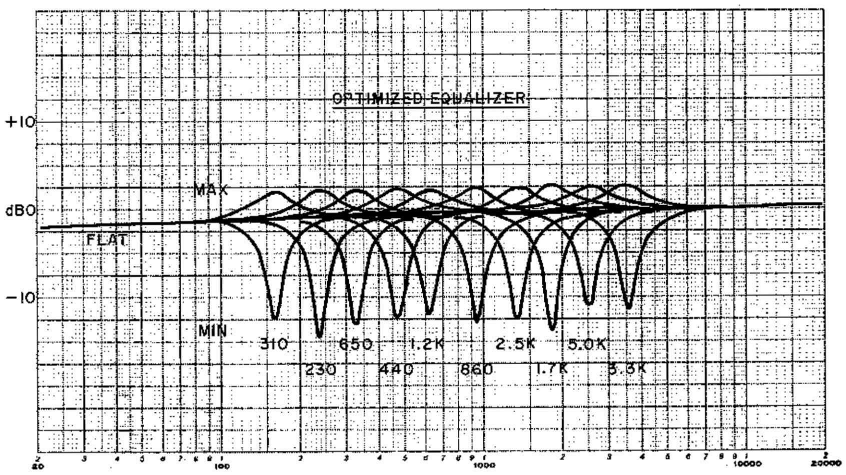


Fig. 10. Component layout on analyzer pc board.

TESTING THE EQUALIZER/ANALYZER



Boost and attenuation effects of the ten filter points.

The concept of having the analyzer use some of the circuits in the equalizer is an interesting one and makes for economy in achieving both analysis and equalization. In addition, the recognition that a limited amount of boost and much more "cut," are required for room/speaker equalization is something we have not seen discussed before. It differs sharply from conventional practice, which provides symmetrical (more or less) boost.

The measured characteristics of the various filters in the analyzer and equalizer confirm the statements made in the article. It is interesting to note that using only the extreme controls (40 Hz and 10 kHz) one can simulate quite well the effect of a conventional tone control system. The distortion of the equalizer was negligible and well within the stated limits. The noise (which was below our measurement limit) appeared to meet the claimed performance comfortably.

Following the instructions, we used the system to equalize a stereo music system. It would be helpful if the meters could be marked to match the corresponding slider controls; we had to use some "cut and try" methods in doing the qualization, but the end result seemed to be reasonable. According to a spectrum analysis of the "pink noise" from the system, it is not quiet pink. However, since one uses the meters to read the noise spectrum as well as the qualized acoustic spectrum, this error is of no importance. —*Julian Hirsch*

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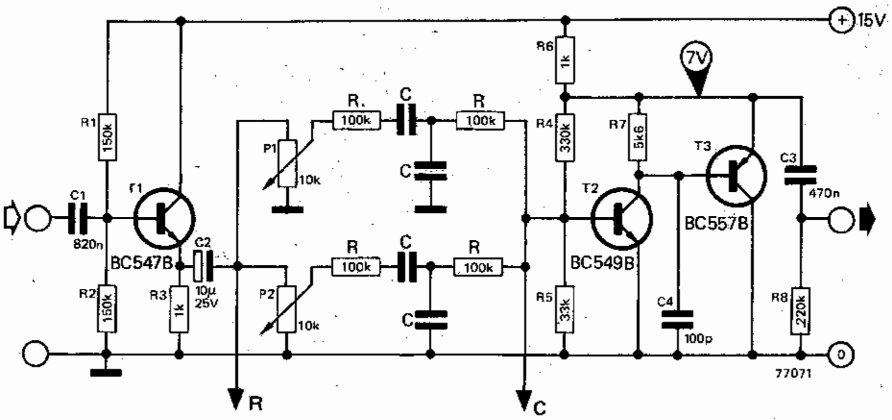
π-qualiser

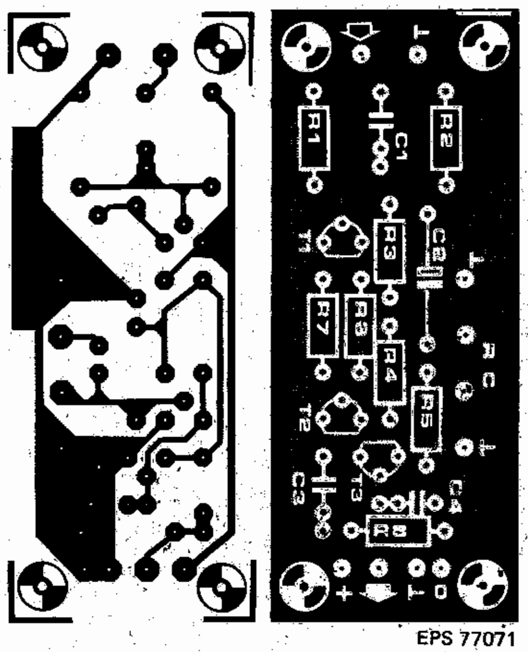


This simple equaliser is intended mainly for tailoring of room acoustics in such applications as disco and p.a. work, and for sound effects in electronic music. It is not intended for hi-fi use as more sophisticated circuits are needed to give satisfactory results in this application.

The circuit comprises an emitter follower T1 feeding a number of Wien networks, each of which passes a band of frequencies about its centre frequency. P1, P2 . . . etc. vary the proportion of the total signal that is fed to each Wien network and hence the proportion of each selected frequency band that appears at the output. By using a number of Wien networks with centre frequencies spaced at suitable intervals throughout the audio spectrum it is possible to boost or cut selected bands of frequencies, and thus adjust the response of the equaliser to compensate for room acoustics etc.

The output of each Wien network is fed to a summing amplifier consisting of T2 and T3, which has a gain of three to overcome the attenuation of three introduced by the Wien networks at their centre frequencies.

For most purposes five Wien networks should be sufficient with centre frequencies of 40 Hz (C = 39 n), 155 Hz (10 n), 625 Hz (2n2 in parallel with 330 p), 2.5 kHz (680 p), 10 kHz (160 p).



EPS 77071

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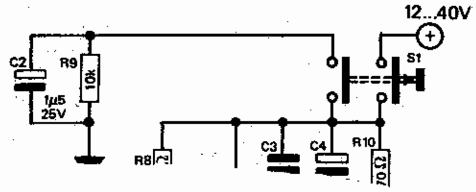
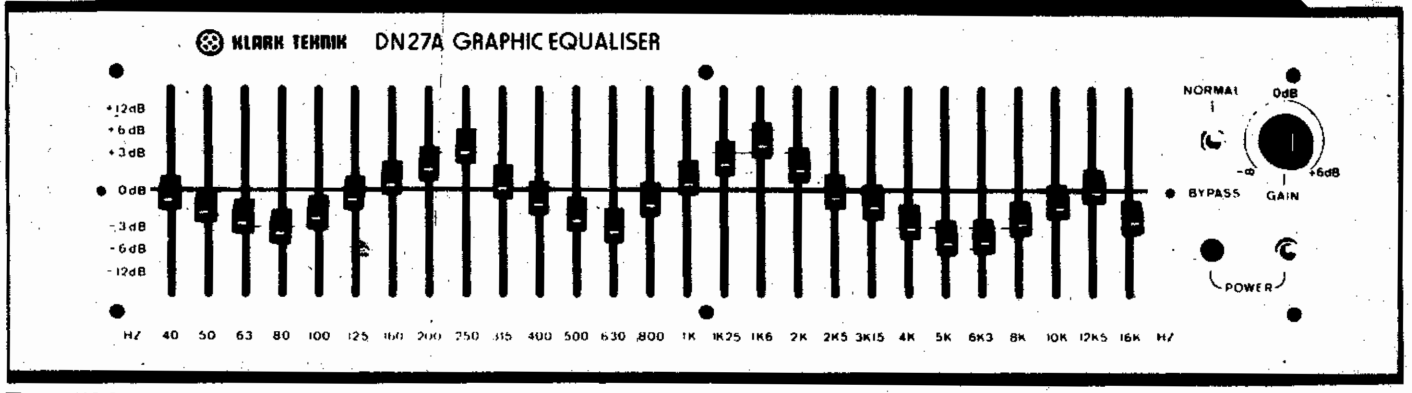
dog!

This 'inaudible' car horn is intended for warning four-footed pedestrians of imminent danger, without needlessly scaring the (hopefully) better-behaved two-footed variety.

Basically, the unit is a power multivibrator that oscillates at a frequency above the



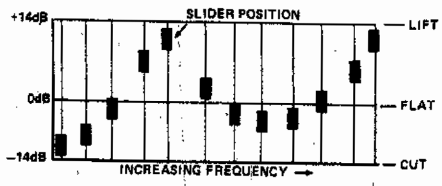
EQUALISER CIRCUIT DESIGN



Everything in life has its little ups and downs, and the frequency spectrum of an audio signal is no exception. Tim Orr gives you a whole bunch of circuits for ironing out the bumps.

AN EQUALISER IS a signal process device used to modify the frequency spectrum of an audio signal. A graphic equaliser is an equaliser that graphically displays the frequency response curve being imposed upon the audio signal, as shown in Fig. 1 and the photograph. The frequency spectrum is split up into bands, the gain of each band being controlled by a slider pot. The normal control range is about ± 14 dB. Each band or channel is, in fact, a filter which can be controlled so as to give a continuously variable response from a peaky band-pass to a notch (Fig. 2). Note that the Q factor of the filter reaches maximum at maximum lift and cut, the response being flat in the central position. If a bank of these filter networks is employed to process an audio signal, then their individual responses may be concatenated to define an overall frequency response. The control sliders will graphically dictate the signal gain at their respective frequencies.

There are of course problems involved in using such a method. The precision with which the sliders can define the frequency response will depend upon number of sliders used; more sliders will give a better resolution and vice versa. Also, the band-pass response of the individual channel is not ideal. Perhaps a rectangular response would be best, but this would be impossible to construct and would suffer time domain ringing effects.



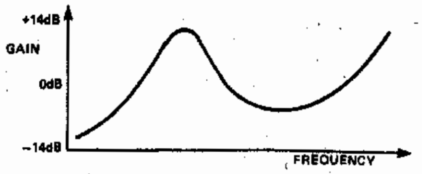


Fig. 1 Operation of a graphic equaliser. The slider positions correspond to the overall frequency response.

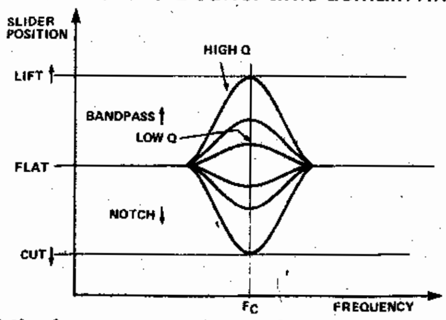
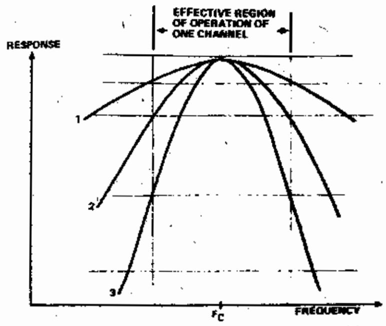


Fig. 2 the frequency response of a single channel. Note that the q factor increases as the amount of cut of lift increases.



SPACING IN OCTAVES	2	1	1/2	1/3
TYPICAL Q	1 TO 1.1	2.7 TO 4	6	7 TO 8

Fig. 3 choosing Q factors. Curve 2 shows how optimum Q results in the best compromise between channel interaction and inter-channel ripple. The chart gives typical Q factors that are used in graphic equaliser design.

Q Dips

There are two interlinked problems associated with using the band-pass response. To reduce interaction between adjacent channels the filter response must be relatively sharp (high Q), but a high Q response will cause large dips (ripple) to occur between the filter peaks when all the sliders are set to maximum or minimum positions (Fig. 3). The chart in Fig. 3 shows the best compromise for Q factor versus the frequency spacing of the filter channels. This assumes a control range of ± 14 dB. It is relatively easy to change the design so that the control range is ± 40 dB for individual channels taken in isolation, but the whole system would have severe interaction between channels, thus destroying the 'graphic' feature, and would also suffer from a large amount of ripple. Figure 4 shows the frequency responses for several Q factors.

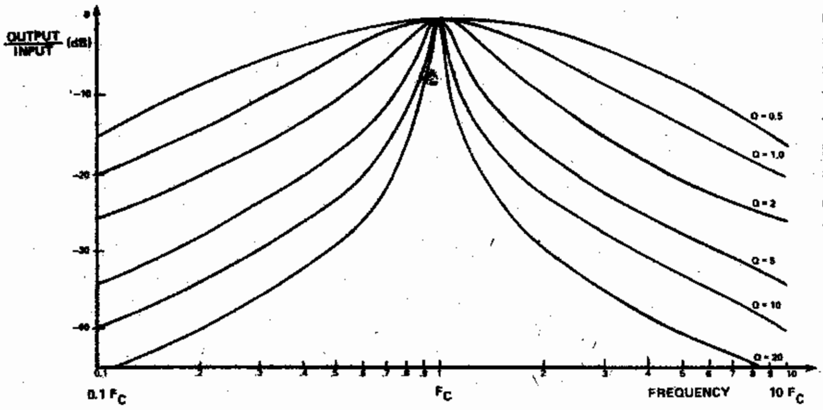


Fig. 4 Transfer characteristics for a band-pass filter. The responses for various Q factors have been normalised.

A Passing Phase

Figure 5 shows the classic equaliser circuit. The band-pass resonator is a series resonant circuit having minimum impedance at the resonant frequency F_C and also zero phase shift (Fig. 6). The amount of cut and lift can be calculated as follows. Let the resonant impedance of the filter be R_F. The phase shift at resonance is zero and so this impedance may be treated as a resistance. With the slider in the cut position, the input signal is attenuated (shunted to ground) by the resonator. The attenuation is 20. log $(R_F/(R_F + R_A))$ dB.

At frequencies other than resonance the attenuation will be defined by R_A and the complex impedance of the resonator; the attenuation will thus follow the frequency response, eventually ending up at zero. In the cut mode the circuit behaves as a voltage follower; whatever signal is seen at the non-inverting terminal of the op-amp appears at its output. With the slider in the lift position the feedback signal is attenuated, with the result that the output signal must be proportionally larger. The gain at resonance is also 20. log $(R_F/(R_F + R_A))$ dB. With the slider in the central position there is an equal attenuation of both input and feedback signals and so the overall response is flat (Fig. 2).

Shifting Shifts

Graphic equaliser designs suffer from all the usual circuit design problems plus a few that are unique to themselves. Stability is a problem. If several high Q resonators are introduced into the feedback loop of an op-amp then their accumulated phase shift may push the network into instability. This is usually overcome by splitting up odd and even resonators into two separate networks so that the phase shifts of adjacent channels do not add up in the feedback loop of one op-amp. Noise is also a problem. The more treatments that operate upon a signal, the worse the signal-to-noise ratio becomes. A graphic equaliser with lots of channels introduces more noise than one with fewer. Noise problems may be minimised by using low noise op-amps and by operating at as high a signal level as possible.

Once a design has been selected for Q factor, frequency spacing and absolute centre frequencies, the component values for the resonators are calculated, and guess what; not a single one of them is a preferred value! It is usual to find that the capacitors are constructed from two components in parallel, and the resistors are precision types. The component accuracy is dependent upon the channel spacing. For a one-octave spacing design, a tolerance of better than 5% is recommended. Component tolerance errors will manifest themselves in two ways; as a spread in Q factors and centre frequencies. Both of these will cause the overall frequency response to be arbitrary and lumpy. Yet another problem that affects the overall response is band-width limiting in the op-amps. This causes active resonators to go flat in frequency at high frequencies.

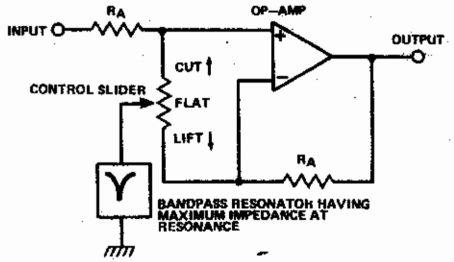


Fig. 5 A typical graphic equaliser section.

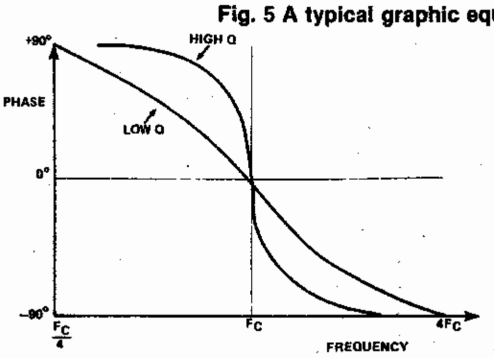


Fig. 6 The phase response of a simple band-pass resonator.

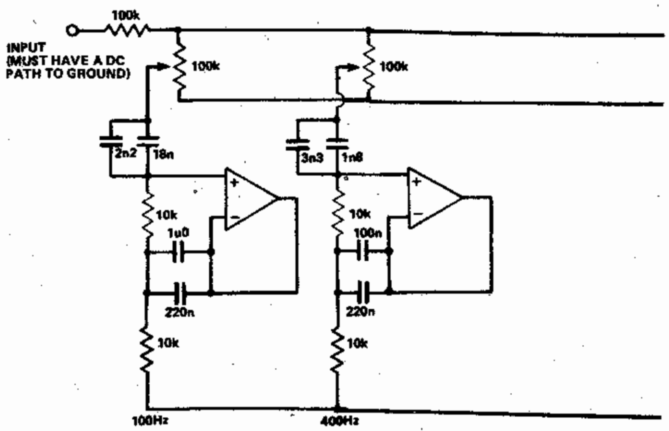
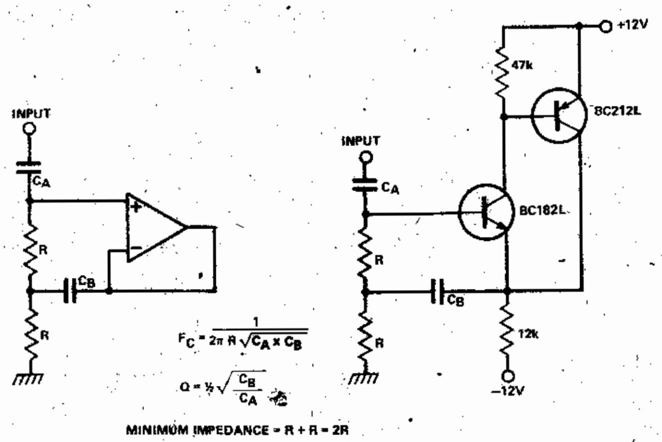


Fig. 7 A multiple feedback band-pass resonator using an op-amp (left) and discrete components (right). As a rule of thumb the op-amp should have a gain bandwidth product in excess of 20 = Q^2 = F_C.



Active Designing

A simple multiple feedback filter can be made to simulate a series resonant LCR network (Fig. 7). This is known as a gyrator, an active simulation of an inductance. Note that the two capacitors define both the Q factor and the centre frequency. This circuit makes very heavy demands upon the bandwidth of the op-amp. A resonator with a Q of 4, operating at 12.8 kHz, needs a bandwidth in excess of 20 × 4 × 4 × 12.8 × 1000 = 4.096 MHz. Even with this bandwidth its performance would not be perfect. Figure 8 shows the complete circuit for a five-section, two-octave equaliser. The design procedure is as follows. The centre frequencies and spacing are arbitrarily selected. For two-octave spacing the Q is 1.1 (Fig. 3). Therefore the component values may be determined from the equations in Fig. 7. For low noise and wide bandwidth operation, RC4558 op-amps were used.

Another equaliser is shown in Fig. 9. this is the same circuit as before, although the channel spacing is now one octave and so the Q factor is higher, having a value of 4. Note that the odd and even channels have been split up into two sections so that filter interaction may be minimised and stability maintained. The actual centre frequencies were measured and compared against the calculated ones (Fig. 10). The graph shows a random distribution caused by component tolerances plus a strong underlying trend caused by the bandwidth limitation effect of the op-amps.

Yet another active resonator is shown in Fig. 11. Again this network looks like a series resonant LCR filter, and is often found in active graphic equalisers. It has the same problems and advantages as the previous design. Active resonators possess several advantages. They are cheap, small, non-mechanical, they work well at low frequencies, and can be implemented using small capacitor values. However, they don't work well at high frequencies.

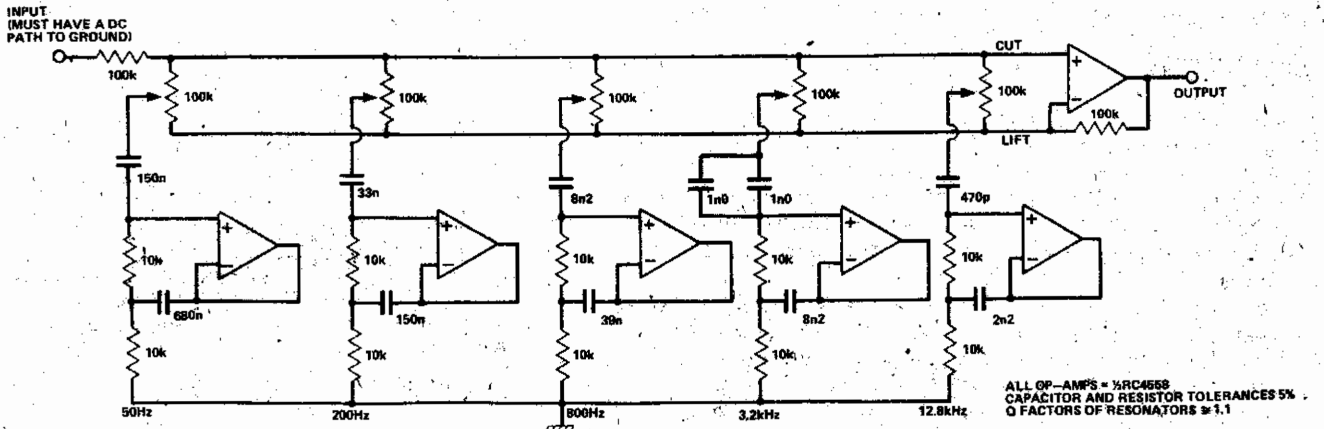
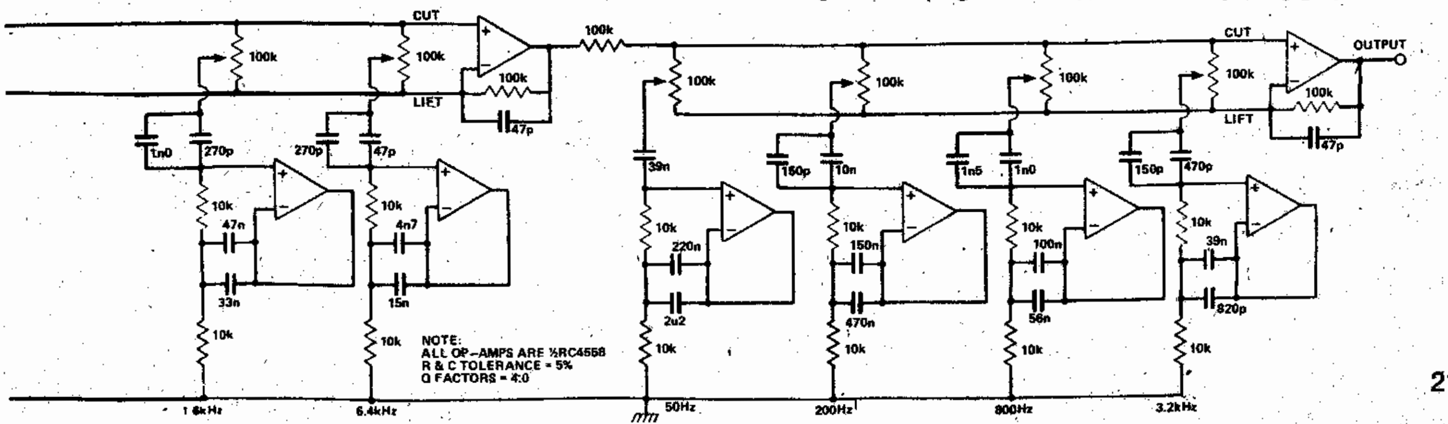


Fig. 8 Circuit diagram of a five-section, two-octave graphic equaliser (Courtesy of Powertran Electronics).

Fig. 9 Circuit diagram of an eight-channel, one-octave spacing graphic equaliser.



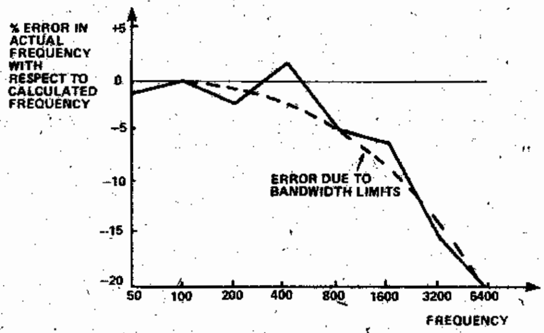
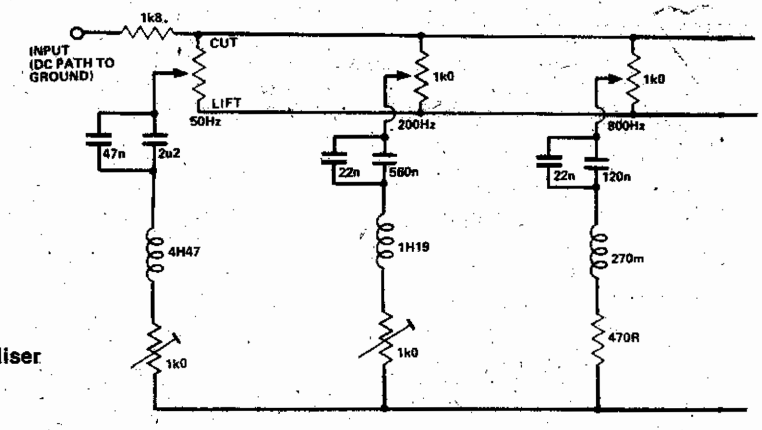
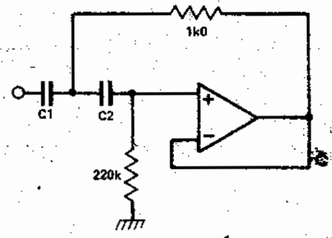


Fig. 10 The frequency error performance of the eight-channel equaliser shown in Fig. 9.





$$C1 = \frac{1}{2\pi \times F \times 3 \times 0.001}\ \text{uF}$$

$$C2 = \frac{3}{2\pi \times F \times 0.22}\ \text{uF}$$

Fig. 11 Another active resonator designed for one-octave spacing and a Q of 3; equations are given for the required capacitor values. If C1 = 390nF and C2 = 18nF, the resonant frequency is 125 Hz. Doubling the values of C1 and C2 halves the resonant frequency and vice versa.

Passive Parts

Figure 12 shows a passive design for an equaliser channel. A passive design works well at high frequencies because the resonators do not suffer from bandwidth problems, but at low frequencies the sizes of the inductors become rather large, several Henries in some cases. The inductors will have to be specifically wound; no-one supplies a range of precision high value inductors. The inductors are also very large, heavy, expensive and sensitive to AC hum pick-up. However, they work very well at high frequencies, they are relatively insensitive to signal levels and are low noise. Also, if the inductance has a tuning slug, then the resonator may be frequency tuned. Because of these advantages, passive resonators are often used in professional studio equalisers.

Figure 13 shows a design for a nine-channel equaliser using passive components. Note that the capacitors are made by paralleling up standard values, but the inductors are wound to the exact value. Also note that the series resistor is a preset on the four largest inductors (due to the significant resistance of the windings) and that the resistance values are very low, 1k8 compared with the 100k of the active design. The component values are designed as follows. The channel frequencies and spacing are arbitrarily selected. A Q factor of 3 is selected from the chart in Fig. 3. Therefore using the equations shown in Fig. 12 we have

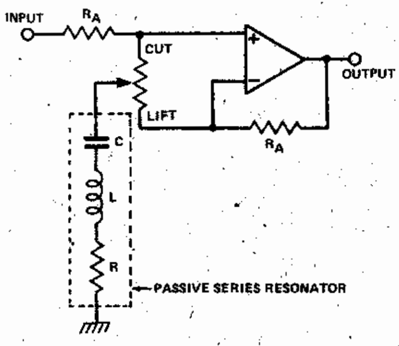
$$Q = \frac{1}{R}\sqrt{\frac{L}{C}}$$

which is held constant for each resonator at a value of 3. Select a reasonable value for R, 470R in this case. Rearranging,

$$\frac{L}{C} = (QR)^2 = (3 \times 470)^2 = 1{,}988{,}100$$

$$\text{Therefore } L = 1.988 \times 10^6 \times C$$

$$\text{But } F = \frac{1}{2\pi\sqrt{LC}}$$



$$F_C = \frac{1}{2\pi\sqrt{LC}}$$

$$Q = \frac{1}{R}\sqrt{\frac{L}{C}}$$

$$\text{CUT/LIFT} = 20\log\left(\frac{R}{R_A + R}\right)\ \text{dB}$$

WHERE F_C = RESONANT FREQUENCY IN HERTZ
C = SERIES CAPACITOR IN FARADS
L = SERIES INDUCTANCE IN HENRYS
R = TOTAL SERIES RESISTANCE INCLUDING THE RESISTANCE OF THE INDUCTANCE IN OHMS

Fig. 12 Circuit diagram and design equations for one stage of a passive equaliser.

Substituting for L and rearranging,

$$C = \frac{1}{2\pi \times 10^3 \times F \times \sqrt{1.988}}$$

Calculate the values of C for all frequency values. Then calculate the values of L from the equation

$$L = 1.988 \times 10^6 \times C$$

Now that all the component values have been calculated, all we need is a source of the inductors. The 'Winding Inductors' box gives details for calculating the number of turns necessary to produce a particular inductance. It is important that the ferrite you use is suitable for the selected operating frequency; the manufacturer's data will tell you this. Select a ferrite core and calculate the required number of turns for its particular A_L value.

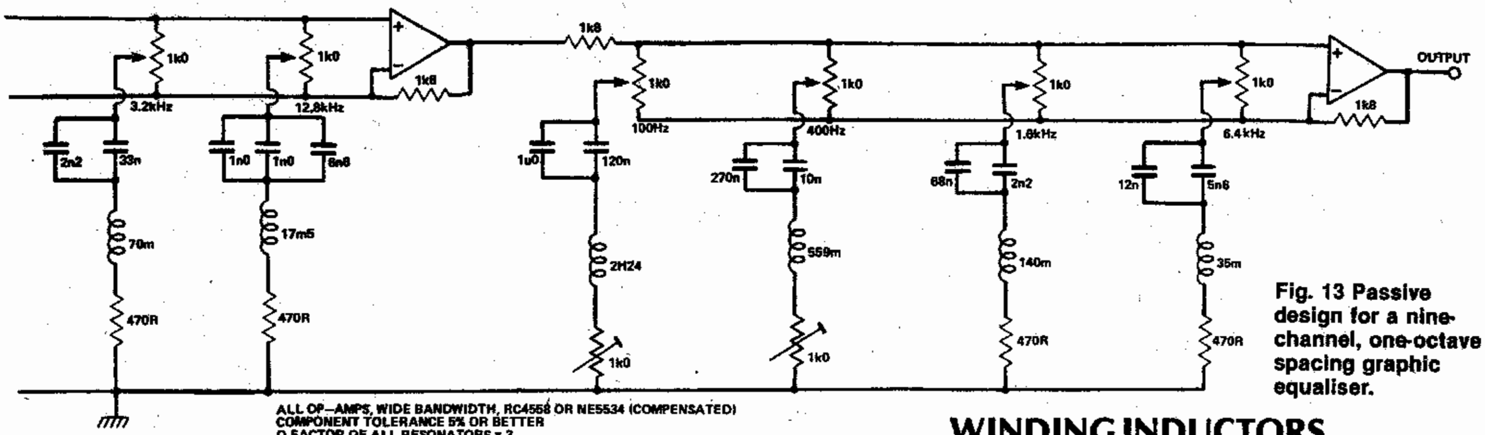


Fig. 13 Passive design for a nine-channel, one-octave spacing graphic equaliser.

ALL OP-AMPS, WIDE BANDWIDTH, RC4558 OR NE5534 (COMPENSATED)
COMPONENT TOLERANCE 5% OR BETTER
Q FACTOR OF ALL RESONATORS = 2
USE THE FOUR PRESETS TO ADJUST THE CUT AND LIFT OF THE BOTTOM FOUR FREQUENCIES

Next, decide which wire gauge will fit the core size; again manufacturer's data. Wind the inductor, and measure its inductance and DC series resistance. Subtract this resistance from the calculated external series resistor, to give you the value of the external component.

WINDING INDUCTORS

$$L = N^2 \times A_L$$

where

L = inductance in nanohenries (ie 10^{-9}H).
A_L = inductance factor. This is usually printed on the side of the ferrite core, and is a constant.
N = the number of turns of wire on the inductor.

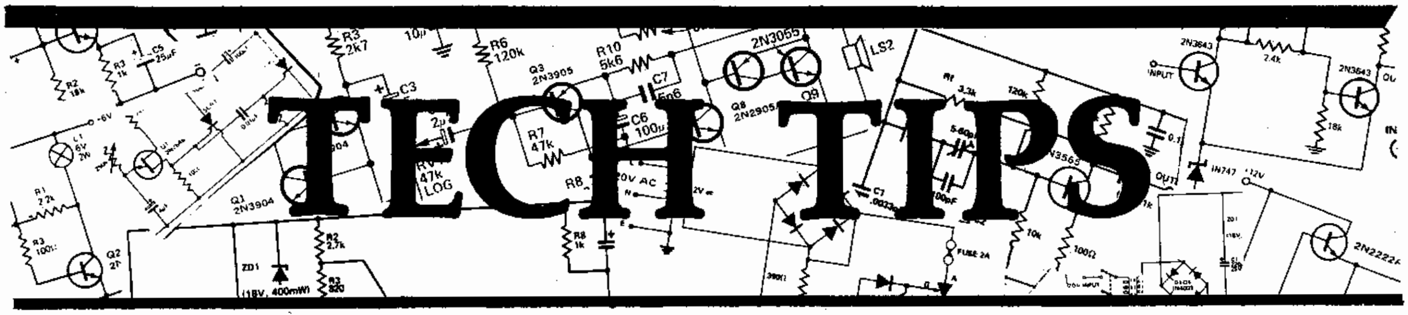
Rearranging the equation,

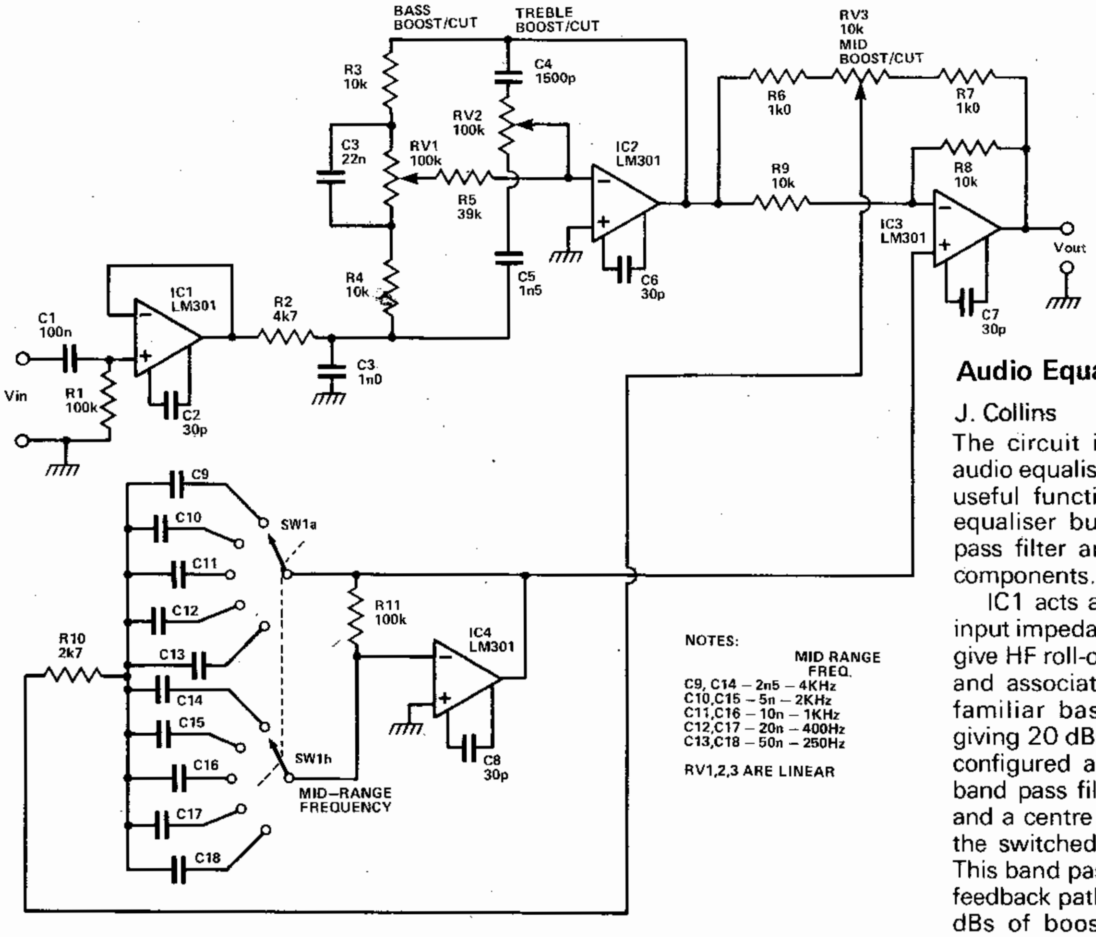
$$N = \sqrt{\frac{L}{A_L}}$$

Example: construct a 5mH inductance to work at 8 kHz. Pick a suitable ferrite core, let's try one with $A_L = 250$. Therefore

$$N = \sqrt{\frac{5 \times 10^6}{250}} = \sqrt{2 \times 10^4} = 141.4 \text{ turns}$$

Pick a suitable wire gauge that will fit the core size. An RM6 ferrite core, for example, can take 160 turns of 34 swg wire and has a recommended operating frequency range of 5.5 kHz to 800 kHz.





Audio Equalizer

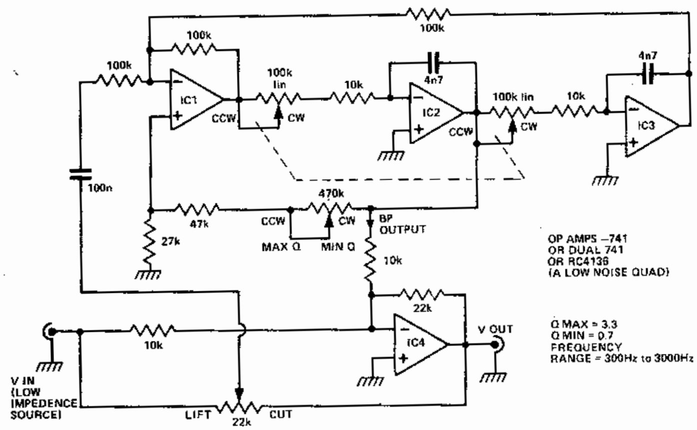
J. Collins

The circuit is a versatile line level audio equaliser providing many of the useful functions of a multi-channel equaliser but using only one band pass filter and, therefore, far fewer components.

IC1 acts as a buffer, providing an input impedance of 100 k. R2 and C3 give HF roll-off at around 30 kHz. IC2 and associated components form a familiar bass/treble tone control giving 20 dBs of boost and cut. IC4 is configured as a multi-feedback type band pass filter with a Q factor of 3 and a centre frequency selectable by the switched capacitors C9 to C18. This band pass filter is connected in a feedback path of IC3, giving up to 20 dBs of boost or cut at the centre frequency by varying RV3.

All three potentiometers give no boost or cut at their centre (midway) positions and give a smooth increase in boost or cut on rotation to the right or left respectively.

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.

PINK NOISE/GRAPHIC EQUALIZER

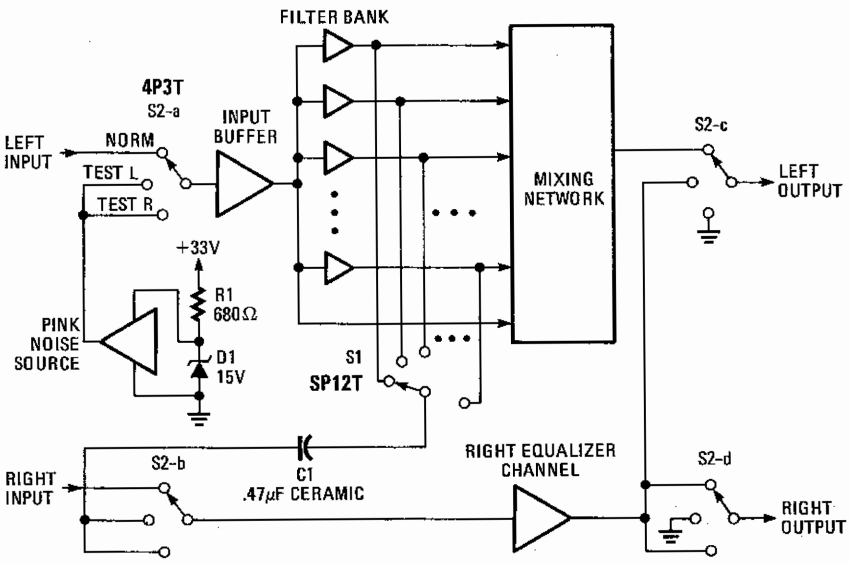
The pink-noise tester described in the January 1978 issue ("Pink Noise Generator Tests Your Hi-Fi," page 43) will not test the equalizer described in my article, "Graphic Equalizer For Your Stereo System," in the May 1978 issue, page 37. However, it *is* an excellent testing tool.

The pink-noise article suggests shorting out the filter section; this produces a phase-shift oscillator.

Furthermore, with the given topology on the equalizer described in the May 1978 issue, turning off the filter (e.g., installing a switch in series with R_{in}) will cause that band to be flat, not attenuated.

Other errors I have observed:

1. Figure 1-c—the switch arm is misdrawn.
2. Figure 2—the values of +12 dB and −12 dB are reversed.
3. The equalizers of Fig. 1 are not true graphic equalizers—they have ripples in their frequency response of typically 3-dB



continued on page 22

LETTERS

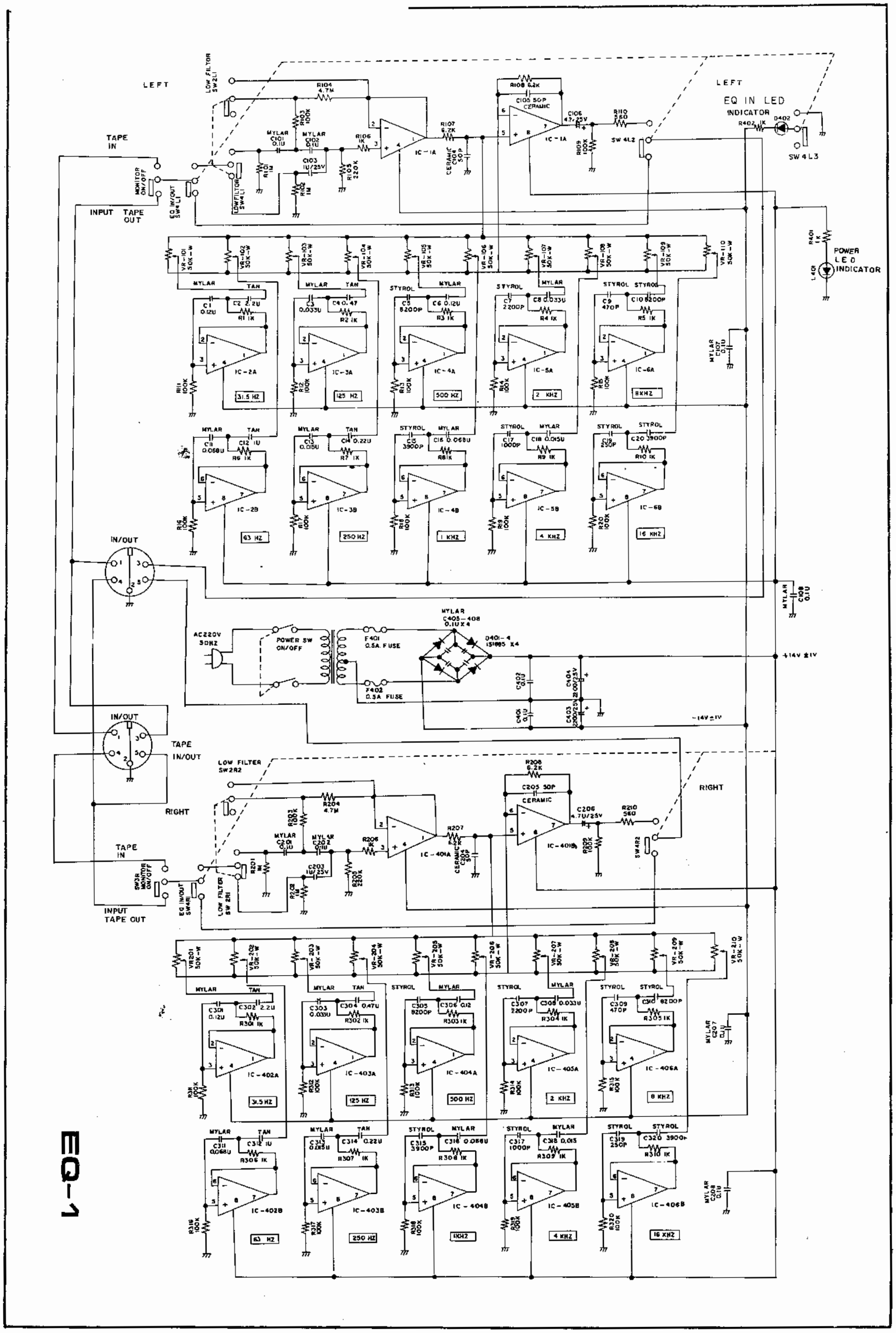
continued from page 16

P-P at a flat setting. To my knowledge there are no consumer versions of this type equalizer available. Furthermore, the circuit of Fig. 1-b seems to be copied from the *National Audio Handbook* and that circuit also oscillates.

The diagram shows a graphic equalizer/noise tester system. In this diagram, switch S1 can be replaced with a test probe, capacitor C1 only prevents clicks. Resistor R1 and diode D1 can be replaced by a battery, and switch S2 can be replaced by reconnecting patch cords. The pink-noise generator is available from West Side Electronics, Box 636, Chatsworth, CA 91311, for $9.95 postpaid; the graphic equalizer is available from Synergistic Sound Systems, 1608 S. Douglas Avenue, Loveland, CO 80537, for $90 postpaid.

To adjust the equalizer, turn switch S2 to "Test Left." Adjust the *right*-hand channel controls until the output of your system is independent of the position of S1 (adjust the bottom right-hand band with S1 connected to the bottom left-hand band, etc.). Adjust the left-hand channel slide potentiometers to be the same as the right-hand channel. Change switch S2 to "Test Right." Repeat the adjustments to the right-hand channel. Simple! There's plenty of room inside and on the rear panel of the equalizer to mount the pink-noise generator and switches.

JOE GORIN



LEFT
LOW FILTOR SW2L1
TAPE IN
MONITOR ON/OFF
EQ IN/OUT SW4L1
INPUT TAPE OUT
LOWFILTOR SW4L1
SW 4L2
EQ IN LED INDICATOR
SW 4L3
POWER LED INDICATOR
31.5 HZ
125 HZ
500 HZ
2 KHZ
8KHZ
63 HZ
250 HZ
1 KHZ
4 KHZ
16 KHZ
IN/OUT
AC220V 50HZ
POWER SW ON/OFF
F401 0.5A FUSE
F402 0.5A FUSE
D401-4 1S1885 X4
+14V ±1V
-14V ±1V
TAPE IN/OUT
RIGHT
LOW FILTER SW2R2
TAPE IN
INPUT TAPE OUT
SW4R2
EQ-1

STATE OF SOLID STATE

An audio preamp and graphic equalizer.

ROBERT F. SCOTT, SEMICONDUCTOR EDITOR

WE'D PLANNED TO OPEN THIS COLUMN WITH a discussion of some pertinent characteristics of IC operational amplifiers (op-amps), and to show how you can use those characteristics to select the best device for your needs and then design a circuit for optimum performance. But, after looking over Joe Carr's excellent article "Designing Circuits with Op-Amps" in the March 1982 issue of **Radio-Electronics,** we decided to hold off for a couple of months to give you time to digest what you learned from him. So, instead let's get on with the "circuit of the month"—which, coincidentally, is built around an op-amp.

Figure 1 shows how the 5532 dual low-noise high-performance op-amp from either Signetics or Exar can be used in a phono preamp and graphic equalizer. The circuit itself is not very new, but its performance is greatly improved by the 5532 because it develops less noise, has improved output-drive capability, and has greater small-signal and power bandwidths than most common op-amps. The characteristics of the 5532 op-amp are listed in Table 1. The device is internally compensated for unity gain. If you use it where very low noise is of prime importance, you'll want to specify the 5532A version with its guaranteed noise specifications.

TABLE 1

- Small-signal bandwidth: 10 MHz
- Output drive capability: 600Ω, 10V (rms)
- Input noise voltage: 5nV/√Hz
- DC voltage gain: 50000
- AC voltage gain: 2200 at 10 kHz
- Power bandwidth: 140 kHz
- Slew rate: 9V/µs
- Large supply voltage range: ±3V to ±20V

Returning to the schematic, the phono-preamp stage, IC1-a, provides standard RIAA equalization for a magnetic cartridge. The necessary bass-boosted response is provided by the R-C network in the op-amp's feedback loop. The cartridge loading-resistor, R1, has a value of 47K to match the impedance of the average magnetic cartridge. Adjust R1's value as required if your cartridge works best into a different load resistance.

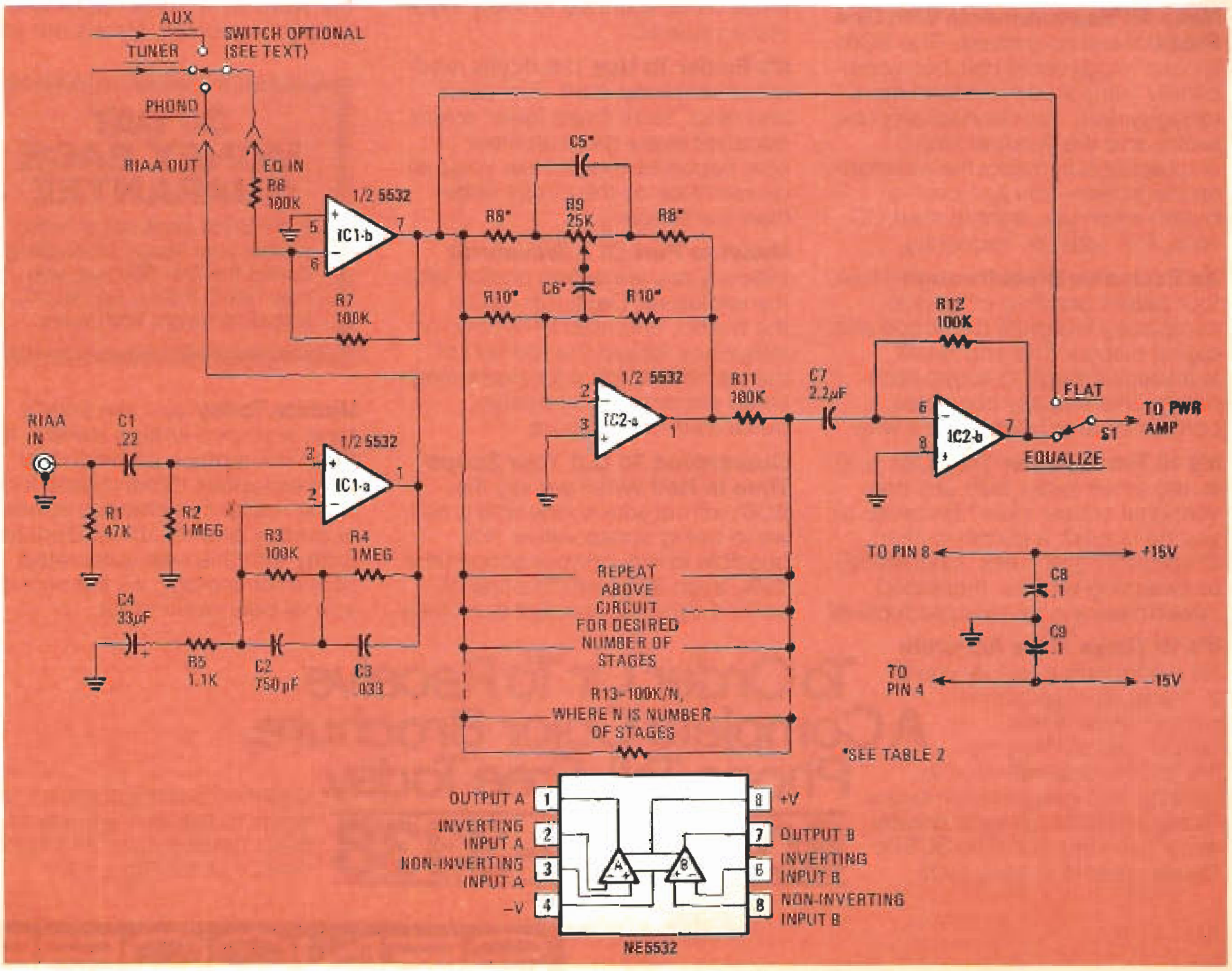


FIG. 1

The graphic-equalizer section consists of an input buffer (IC1-b), variable-boost/cut active filter (IC2-a), and output summing-amplifier (IC2-b). The IC1-b circuit is designed for unity gain and is used mainly for impedance-matching between the preamp and the equalizer (active filter) input. The filter is a variable-bandpass or notching device, depending on the setting of control R9.

Any number of equalizer filter-stages can be used within the range of about 20 Hz to 20 kHz. However, the more stages you have, the easier it is to boost or cut a particular frequency without affecting the response at adjacent frequencies. All the filter stages use the same R-C feedback-network configuration shown in Fig. 1, to provide a maximum of about 15-dB of boost or cut at f_o, the center frequency. The only differences in each stage are in the values of C5 and C6, which set the values of f_o. Table 2 lists the values for C5 and C6 for 22 center frequencies in the audio spectrum. Note that C5 is ten times as large as C6 and that the values for R8 and R10 are both related to the value of R9 by about a factor of 10. The center frequencies have been adjusted so that C5 and C6 are standard, off-the-shelf, values. We recommend using linear slide-potentiometers for R9.

The value of R13 depends on the number of filter stages used. It insures that the gain across the equalizer is unity when all con-

TABLE 2

R8 = 2.4K R9 = 25k R10 = 240k

f_o	C5	C6
23 Hz	1μF	.1μF
50 Hz	.47μF	.047μF
72 Hz	.33μF	.033μF
108 Hz	.22μF	.022μF
158 Hz	.15μF	.015μF
238 Hz	.1μF	.01μF
290 Hz	.082μF	.0082μF
350 Hz	.068μF	.0068μF
425 Hz	.056μF	.0056μF
506 Hz	.047μF	.0047μF
721 Hz	.033μF	.0033μF
1082 Hz	.022μF	.0022μF
1588 Hz	.015μF	.0015μF
2382 Hz	.01μF	.001μF
2904 Hz	.0082μF	820pF
3502 Hz	.0068μF	680pF
4253 Hz	.0056μF	560pF
5068 Hz	.0047μF	470pF
7218 Hz	.0033μF	330pF
10827 Hz	.0022μF	220pF
15880 Hz	.0015μF	150pF
23820 Hz	.001μF	100pF

trols (R9's) are in the FLAT or 0 dB position. The value of R13 is 100K divided by N, where N is the number of stages used. Note that only one audio channel is shown in the circuit in Fig. 1. You'll need two of those circuits for stereo.

This circuit was taken from the NE5532 data sheet and from the applications note "Signetics Low-Noise Operational Amplifiers."

DAC technical data sheets

D/A Converter Products offers the designer a wide and versatile range of eighteen DAC's (Digital-Analog Converters) to choose from. The series consists of the 8-bit industry-standard MC1408 and DAC-08 converters and microprocessor-compatible DAC subsystems. Also included is complete data on the recently announced microprocessor-compatible 10-bit NE5020 DAC. The 67-page data book opens with a converter cross-reference guide to aid the designer in selecting the device best suited to his requirements.—**Signetics Analog Marketing,** Data Converter Division, PO Box 409, Sunnydale, CA 94086

RF transistor catalog

Short Form Catalog No. 503A is a 10-page listing of the pertinent characteristics of approximately 150 RF transistors by TRW. Devices are cataloged under such headings as *small-signal/low-noise, linear SSB, mobile radio* (20 to 870 MHz), and *broadband microwave* (0.6 to 6.0 GHz). Package outlines and dimensions are shown in detail.—**TRW RF Semiconductors,** 14520 Aviation Blvd., Lawndale, CA 90260 **R-E**

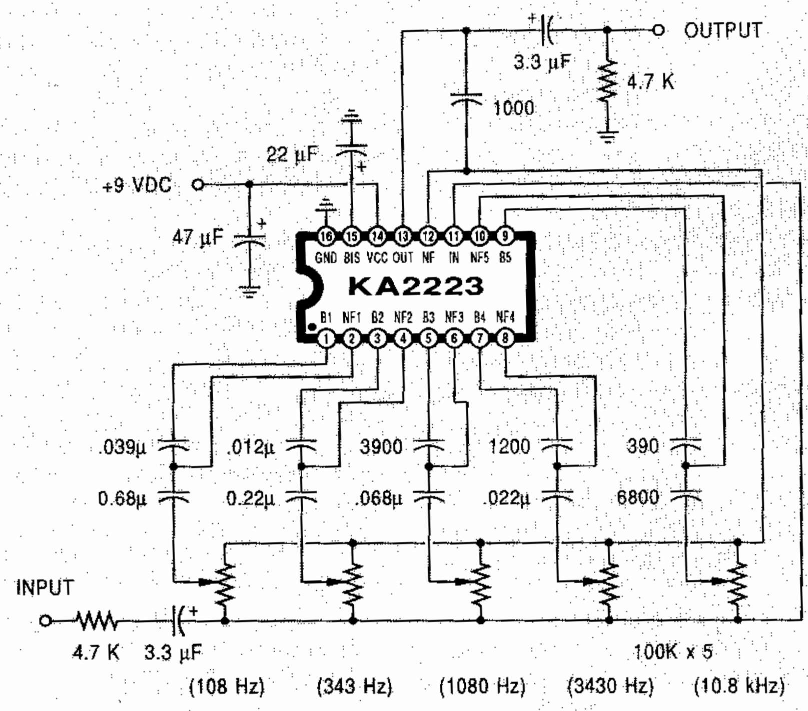


FIG. 3—A 5-BAND GRAPHIC EQUALIZER that uses a single Samsung integrated circuit. A pair of chips can be used for ten bands.

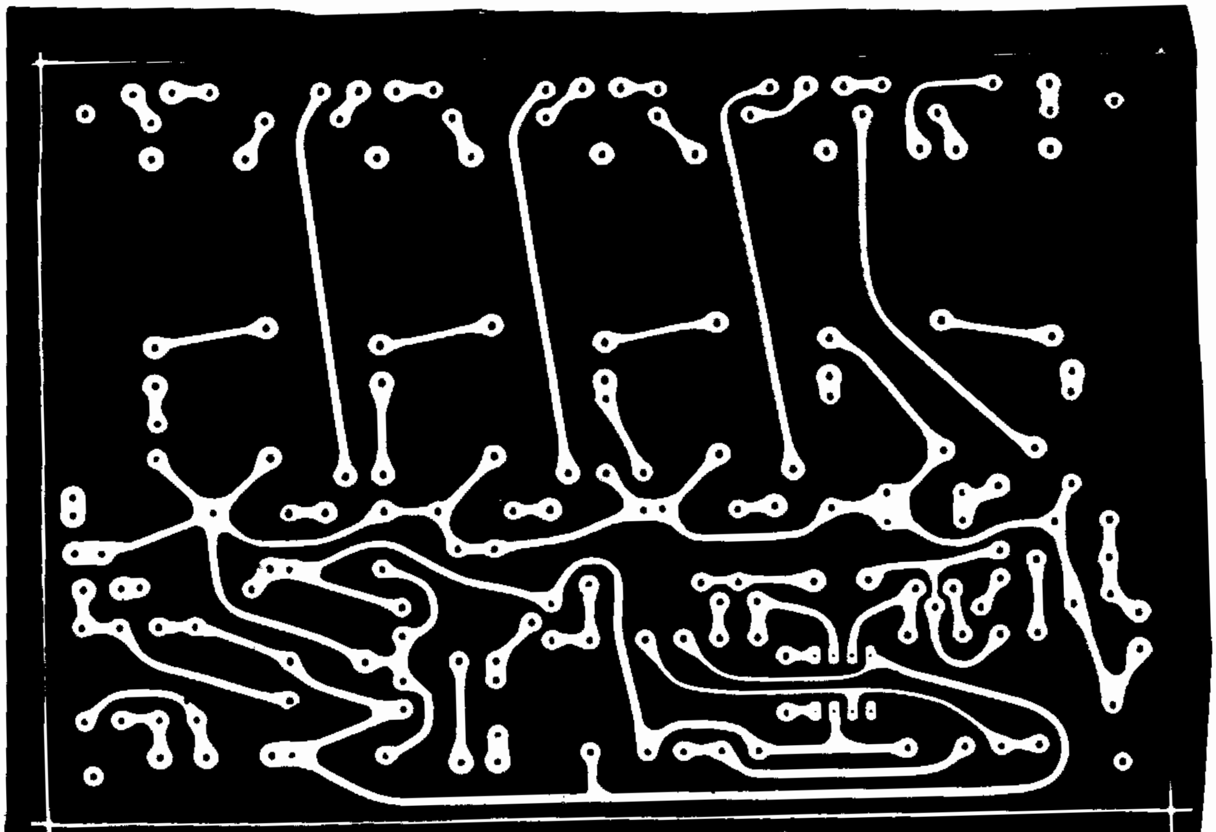


Fig. 4. Printed circuit board for the equalizer. Full size 152 x 103 mm.