Audio Analyser

For creative audio applications

AUDIO spectrum analysers can be used for a variety of tasks. Their most valuable use is to set up a room acoustically, for live music or in conjunction with a graphic equaliser connected to your hi-fi. This will allow you to compensate for deficiencies in either your speaker system alone, or the system/living room combination. The procedure used involves feeding pink noise (more on this later) into the room via your hi-fi system and monitoring the sound with the Analyser. The Analyser points out the peaks and troughs in the audio so that you can get rid of them by adjusting the graphic equaliser controls — hopefully this will produce a flat response.

Other uses for a spectrum analyser include monitoring live programme material or (let's be honest!) as a great little gadget to impress your friends.

To Sweep Or Not To Sweep

There are two main methods of performing spectrum analysis. The first uses a single tuneable filter which can have its centre frequency swept across the band of interest. When the filter output is displayed on an oscilloscope screen it constitutes a graph of amplitude against frequency for the input signal. This gives a well-formatted and accurate display but unfortunately it has the disadvantage of not being 'real time'; if something happens at one frequency while the filter is sweeping somewhere else, it will not be recorded. Consequently, this method is normally only used where the spectral content is constant and the sweep is to be made over a small percentage of the total frequency. A typical example of this would be checking that the emissions of a CB rig were within legal limits; the rig is turned on, with no audio input, so that only the carrier wave is being transmitted; a sweep is then made either side of 27 MHz to check that there are no spurious emissions.

When the spectrum of the input is rapidly changing, as is the case with an audio signal, then we must choose a different method. For real time analysis we use several bandpass filters, with fixed centre frequencies, to chop up the frequency spectrum into several bands. The content of each band is rectified, averaged and displayed on an oscilloscope or, as in this project, on columns of LEDs. Commercial spectrum analysers are available with anything from 10 one-octave steps to 30 third-octave steps, but the cost and complexity of the filters increases dramatically as you make the bands narrower. Consequently, we have

PARTS LIST

RESISTORS (All ¼ watt 5% carbon)	
R1,2	220k
R3	2k2
R4,5,69	15k
R6-10,31-35, 72	10k
R11-15,21-25 36-40,46-50, 78	1M
R16-20,41-45	220R
R26-30,51-55 68,76	100k
R56	680k
R57	6k8
R58-67	47k
R68	100k
R69	15k
R70	430R
R71	27k
R73	4k7
R74	180k
R75	18k
R77	390k

POTENTIOMETERS	
RV1	47k log carbon
PR1-10	220k min horiz preset

CAPACITORS (All metallised Polycarbonate unless noted)	
C1,13,43,49, 50	100n
C2,3,41	10u 16V tantalum bead
C4,5,7	1u0
C6	56n
C8,10	27n
C9,11	270n
C16-20,35-39	2u2 35V tantalum bead
C12	6n8
C14	18n
C15	3n9
C21,22	39n
C23	1n5
C24	33n
C25,32,33	2n2
C26,44	820p ceramic disc
C27	12n
C28	3n3
C29	470p ceramic disc
C30	10n
C31	180p ceramic disc
C34	100p ceramic disc
C40	22n
C42	1n
C45	2n7
C46	5n6
C47,48	220u 16V axial electrolytic

SEMICONDUCTORS	
IC1	LF353 dual BIFET op-amp
IC2-6	TL064 quad lo-power op-amp
IC7	4011 CMOS quad 2-input NAND
IC8,9,10	4016 CMOS quad analogue switch
IC11	LM3915 bargraph driver
IC12	4017 CMOS decade counter/divider
IC13	4070 CMOS quad EX-OR
IC14	4006 CMOS 18-stage shift register
Q1-11	2N3904 silicon NPN transistor
D1-22	1N4148 signal diode
LED1-100	high efficiency red LED

MISCELLANEOUS	
SK1	¼" jack socket with break contacts
MIC1	electret microphone

9V battery clips (2 off); IC sockets (13 off); case; wire; solder; PCBs, etc.

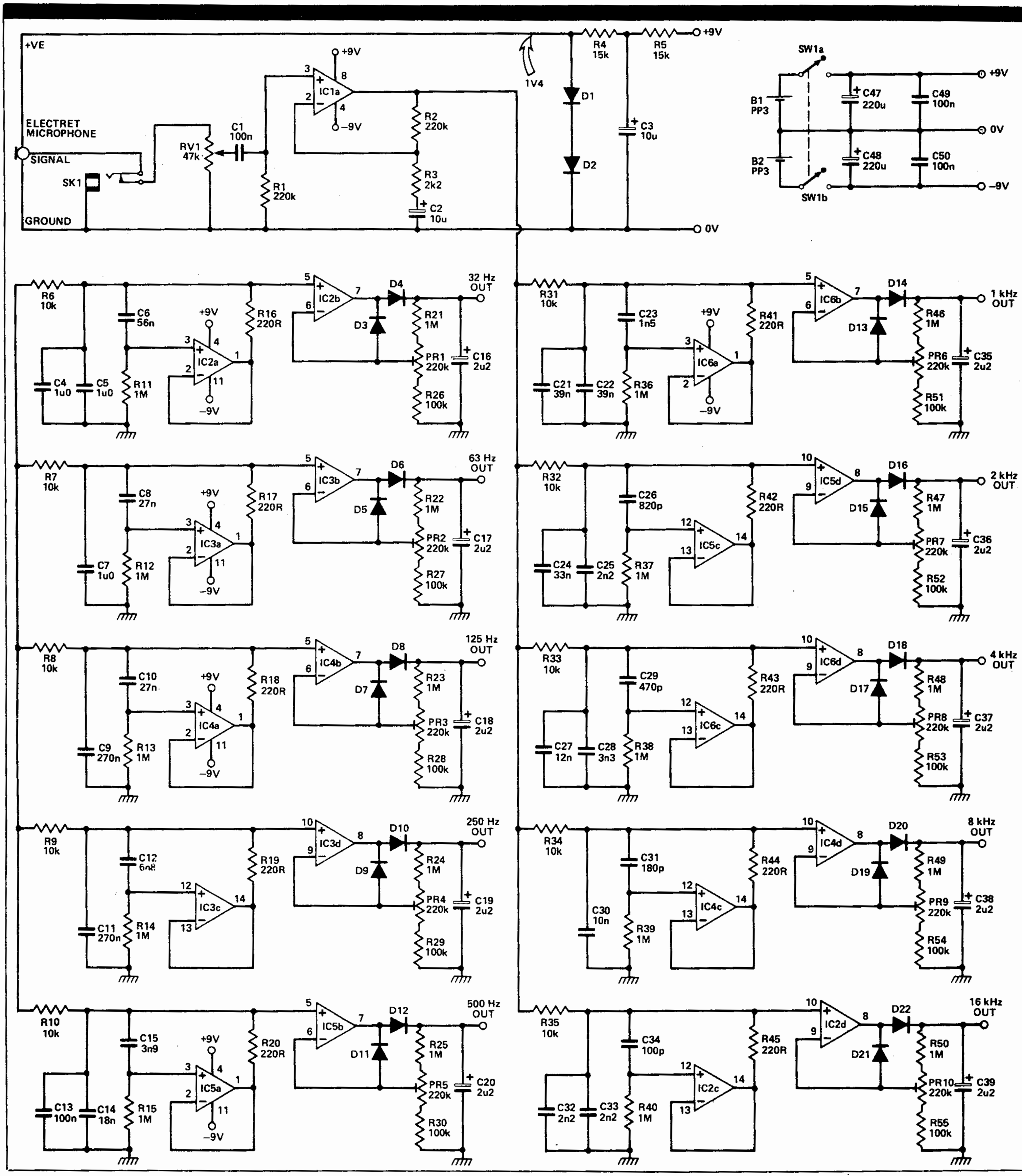


Fig. 1 The input and filter stages of the circuit.

opted for a 10 channel version, the filters' outputs being 12 dB down one octave from the centre frequency. The centre frequencies of the filters follow the standard scale; measured in Hertz they are 32, 63, 125, 250, 500, 1k, 2k, 4k, 8k, and 16k. The amplitude scale has 3 dB steps.

Admittedly, the fact that this type of analyser breaks up the frequency spectrum into octave chunks means that it isn't capable of picking out individual harmonics in the way that the sweep analyser can. Nevertheless, it does allow you to instantaneously determine the average spectrum of a sound, which is all we require.

The Circuit

The input to the circuit (Figure 1) is either from the built-in microphone or via the external input socket. The jack socket automatically disconnects the mic if a plug is inserted. The microphone requires a reasonably flat frequency response but must be relatively inexpensive, so we chose an electret condenser type which meets these requirements. However, electret mics require a 1V5 power supply, normally provided by an AA cell. Ours has a built-in regulated supply built around D1-D2-R4-R5-C3. Zener diodes with a value of 1V5 aren't available but, by using two ordinary diodes in series, we can get an output voltage of about 1V2-1V4 (each diode has a forward voltage drop of about 0V6-0V7).

The input sensitivity can be adjusted with level control RV1, while IC1 a boosts the signal to a suitable level to drive the filter bank. The gain of IC1a is set at 101, ie, (R2 + R3)/R3. Each of the ten filter-rectifier blocks is identical in structure. To obtain a bandpass response with the required roll-off, the simplest solution is to use a parallel LC network with a series resistor. Unfortunately, large value inductors are both bulky and expensive, which rules out their use. We can overcome this easily, however, since the only electrical difference between an inductor and a capacitor is the phase relationship between the current and the voltage. By using an op-amp to reverse the phase relationship of a capacitor we can make it look like an inductor — this type of circuit (Figure 3) is known as a gyrator. The value of the equivalent inductance is given by:

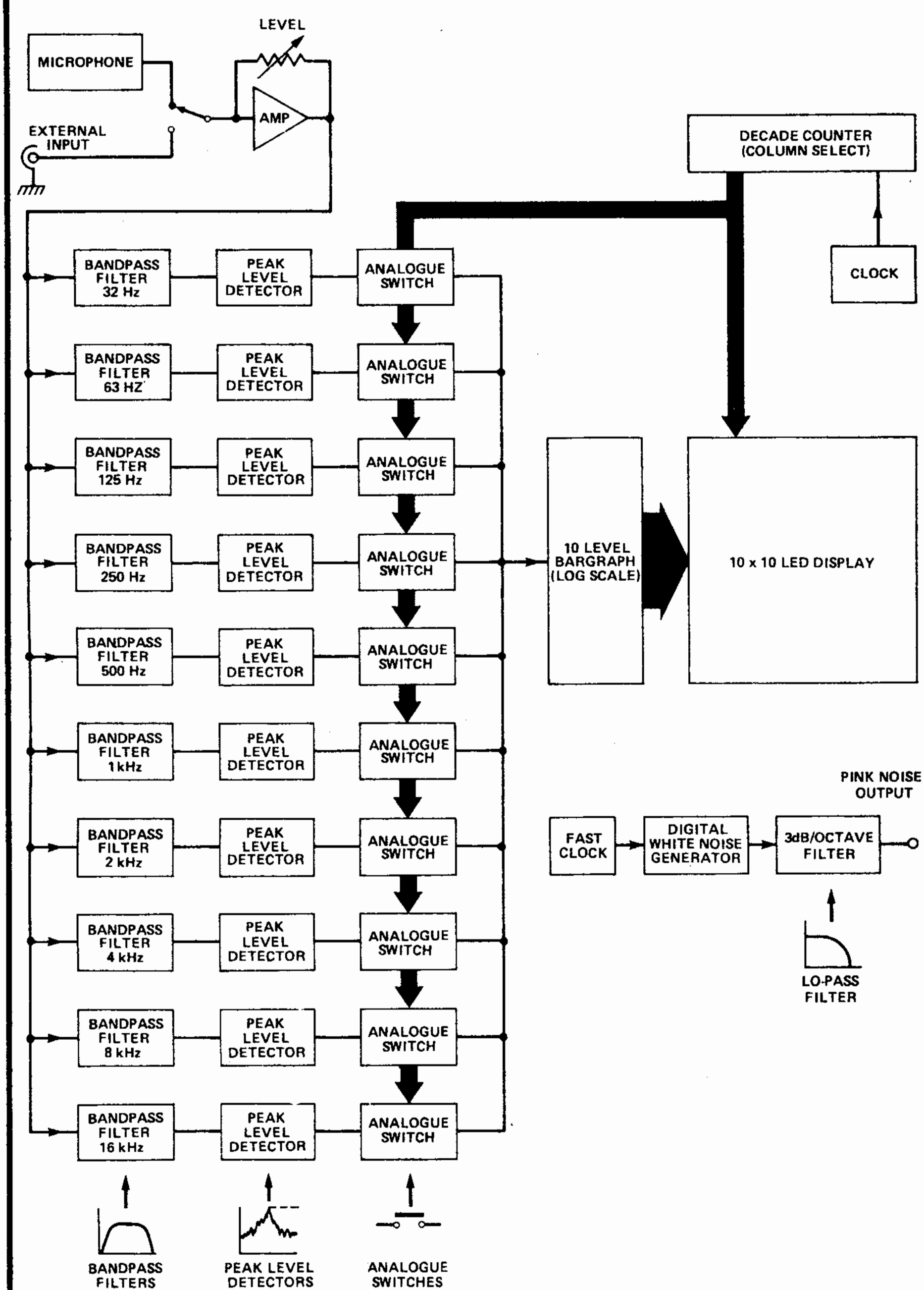
$$L1 = C1 \times R1 \times R2 \text{ Henries}$$

where C is in Farads, R in ohms. Just like a real inductor, we also have a series resistance (winding resistance) which is R2, and a parallel resistance, R1 (in a real coil this is due to winding capacitance). Hence we can tune our filters to the required frequencies by altering the capacitor values in each one, using parallel pairs in some cases, to get the correct values.

The rectifier section is a half-wave type, with a gain variable from about four to 12, using the presets. When the output of the op-amp swings positive, capacitor C1 charges rapidly via the diode; D1; when the output falls, the capacitor can only discharge slowly via the resistor chain. The second diode D2, from the op-amp output back to the inverting input, keeps the op-amp in the linear region on the negative half-cycle.

The outputs of the ten rectifiers are multiplexed to reduce the compo-

HOW IT WORKS



The audio signal to be analysed is taken from the microphone or external input socket to the level control/preamplifier section. This amplifies the signal to a suitable level to drive the circuitry that follows. The signal is fed to 10 bandpass filters spaced one octave apart, each of which will only allow through a small section of the signal around the centre frequency. Each filter is followed by a peak level detector which averages out the signal, responding quickly to peaks but decaying slowly so that the display is easy to read. The outputs of the 10 peak detectors are connected one at a time (by the CMOS analogue switches) to the input of a 10-level LED bargraph driver. A logarithmic driver is used to give 3 dB steps. To reduce current consumption the bargraph operates in dot mode, so that the height of the illuminated LED up the column represents the peak level. The decade counter which controls the analogue switches also switches on the correct column of LEDs for each passband. All the columns are blanked for a short period, as the switches change over, to prevent garbage being displayed.

The white noise is generated digitally by cycling a scrambled sequence of 1s and 0s through a shift register. The white noise is passed through a filter with a slope of 3 dB/octave to produce pink noise.

nent count and cost; if we drive each column of LEDs separately we'd need ten LM3915s, which is a bit expensive! Multiplexing means that each rectifier output is switched to the input of the LM3915 (IC11) one after another, by the analogue switches IC8, 9 and 10. The switches are controlled by a 4017 decade counter (IC12) with ten decoded outputs, each of which is high for one clock period only. These outputs also switch on one of the transistors Q1-10, connecting the required column of LEDs to the positive supply rail. Meanwhile the LM3915 has turned on one of its outputs corresponding to the voltage on its input (remember, it's wired in dot mode). Hence a current path between the supply rails exists for only one LED of the 100 in the display, so at any moment only one LED is turned on. By clocking the 4017 at a fairly slow speed (about 500 Hz) the display cycles through all ten columns 50 times a second and the eye sees ten LEDs 'continuously' lit.

To generate an adequate light level, a red LED requires at least 4 or 5 mA continuous current. Since each

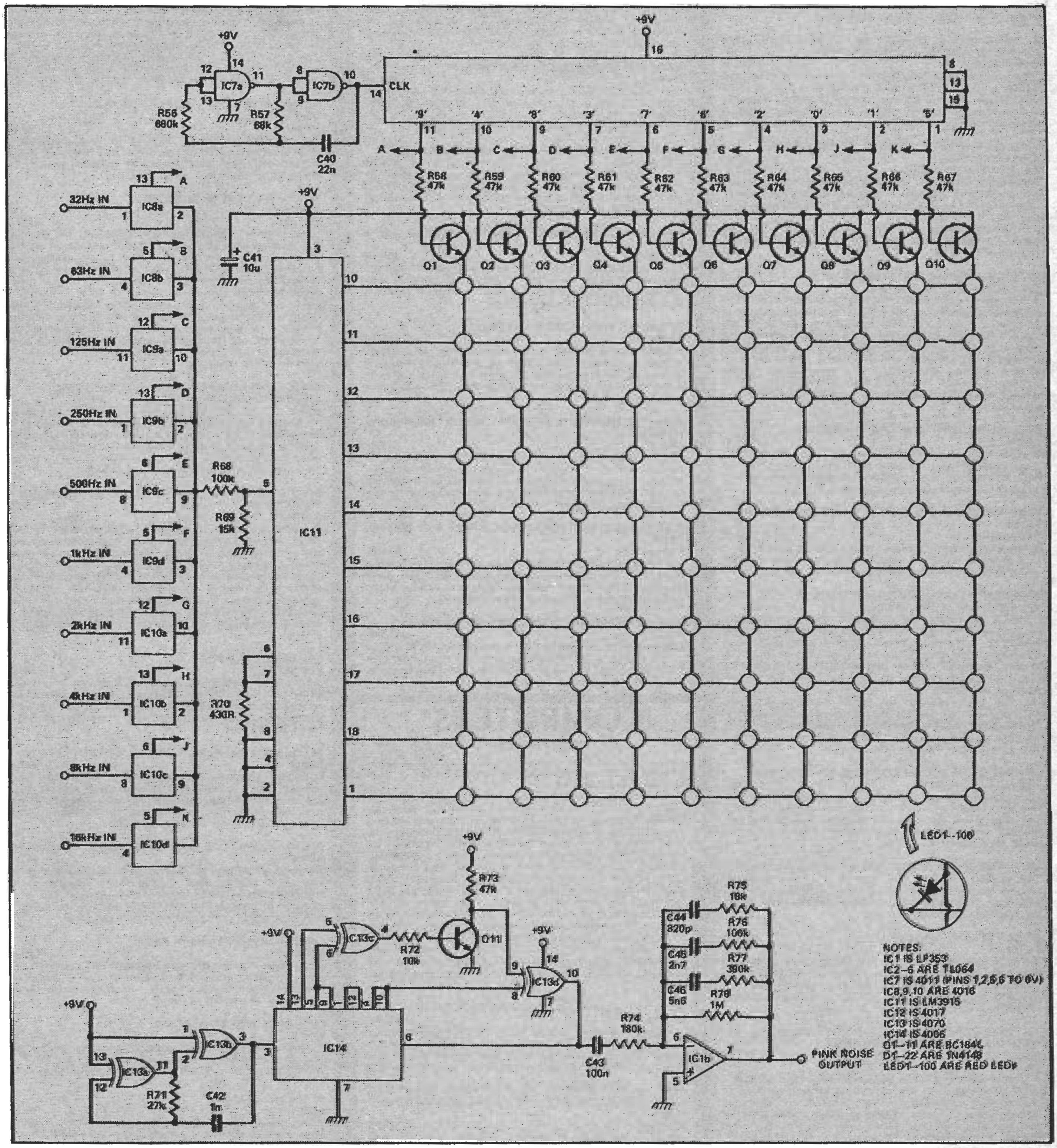


Fig. 2 The display generation circuitry.

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LED is only on for one-tenth of the total time, it requires ten times the current to give the same apparent brightness. The maximum current capability of the LM3915 is only 30 mA, so high efficiency LEDs must be used. The 4017 is even worse at supplying current, hence the use of the drive transistors.

The clock generator for IC12 is a standard configuration built round IC7a,b.

White noise is an audio signal which contains all frequencies and has equal energy per unit bandwidth. However, what we require here is

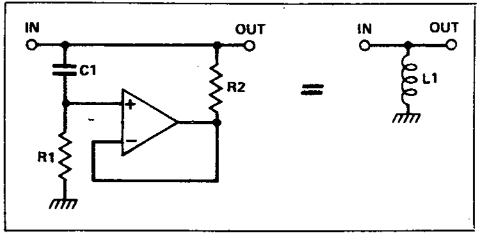

Fig. 3 A gyrator circuit 'looks like' an inductor.

equal energy per percentage bandwidth i.e. equal energy per octave). This is known as pink noise and it is obtained by passing white noise through a filter (IC1b) with a slope of 3 dB/octave. The white noise is generated digitally rather than by a Zener noise diode, which can be temperamental. IC14 is an 18-stage shift register clocked by the 30 kHz oscillator IC13a,d. Two EX-OR gates and an inverter (IC13c,d and Q11) are used to feed various outputs of IC14 back to the input (pin 6) so that a complex sequence of 1s and 0s flows through the register, repeating once every few seconds. This produces an apparently random jumble of fundamental frequencies with a vast number of harmonics, i.e. noise.

To be continued next month.

ETI

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Construction

Construction of this project is not for the faint-hearted! To make this project handheld has meant using two fairly dense PCBs. So you can put away your gas-fired soldering irons and pliers; delicacy, finesse and a steady hand are required. Use an iron with a small bit and make sure you don't leave huge blobs of solder that bridge the PCB tracks. IC sockets are not just recommended; we insist upon them. Fit the components to the boards in the usual way, filter board first, taking great care to observe component polarity where this is important. Don't miss out any of the wire links; and don't try to finish it all in one evening, or two. Time 'saved' during construction will be wasted on fault-finding, later!

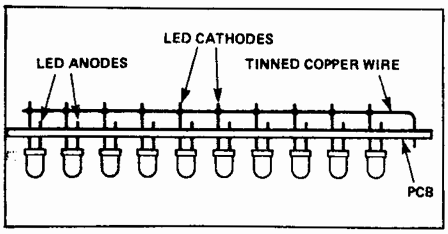


Fig. 1 Wiring the LED matrix; it is best to check all the LEDs before soldering them in, because replacing one will be very difficult once the boards are assembled.

The nice men who designed the 4017 didn't put its sequential outputs 'sequentially' on the IC pins and this often leads to tricky PCB layouts when using the IC. Fortunately we can cheat, because all we need in this project is to look at each of the 10 channels separately — so long as they appear on the right display columns it doesn't matter what the actual order is. This is why the filters don't run in sequence down the PCB — it's purely for convenience.

To avoid the use of a double-sided PCB for the display matrix, a rather cunning technique has been adopted. Solder in one row of LEDs only and cut off one pin only — the ones whose solder pads are linked by copper tracks. Now solder a length of tinned copper wire to the pad indicated, bend it over so it touches the other LED pins about ¼" away from the board, then solder all the pins to it and trim them off. Then do the next row, and so on. Mistakes made here will be almost impossible to correct later, so check that every LED in every row is the right way round before you solder it. The most certain method is to use Figure 2; flats, dots and 'one leg is shorter than the other' can all lead you astray.

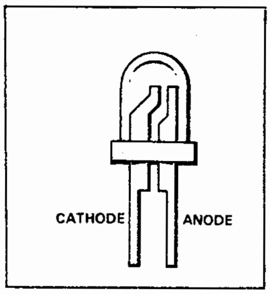


Fig. 2 Pin connections of a LED.

The two PCBs are bolted together with stand-off pillars and have a number of wire links between them. The easiest way to do this is to solder lengths of tinned copper wire to the relevant pads of the display board and cut them down slightly to different lengths, gradually increasing as you move down the board (Figure 3). Then you can insert the longest link onto the filter board, move the two closer together, insert the second and so on, until they're all through.

Assembly starts by test-fitting the display board and making a hold for the display. It's a good idea to glue a piece of red plastic or polarising filter behind it to improve the contrast of the LEDs. Fix the pillars to the display board and screw into the case — the transistors and tantalum capacitors will probably have to be bent over to give clearance. Now slide the main board over the wires, as described above, solder all the links and trim them. Now the wiring for the off-board components can be completed.

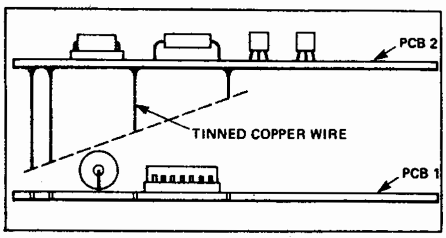
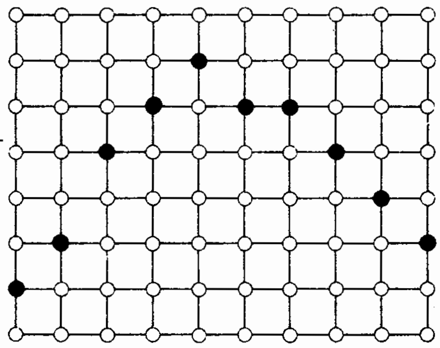


Fig. 3 Fitting the links between the two boards.



We found the cheapest way of getting an electret mic element was to buy a cheap electret cassette recorder mic (about $6) and cut the end off it. Do this very carefully as you musn't damage the insulation on the internal wiring; you'll need it to link the mic to the PCB. Glue the mic into a suitable-sized hole cut in the end of your case and connect it to the PCB as shown on the overlay; the existing battery terminals of the mic will tell you which are the positive and negative supplies, while the third wire is the signal connection (probably screened by the ground wire). If you don't feel your constructional abilities are up to this, just plug a mic into the external jack socket; not as compact but much easier.

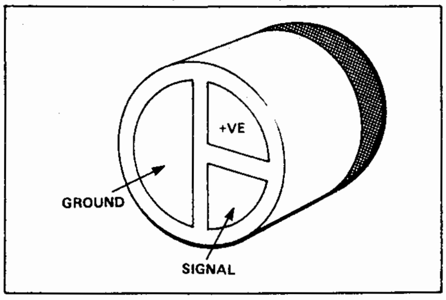


Fig. 4 Connections to an electret microphone.

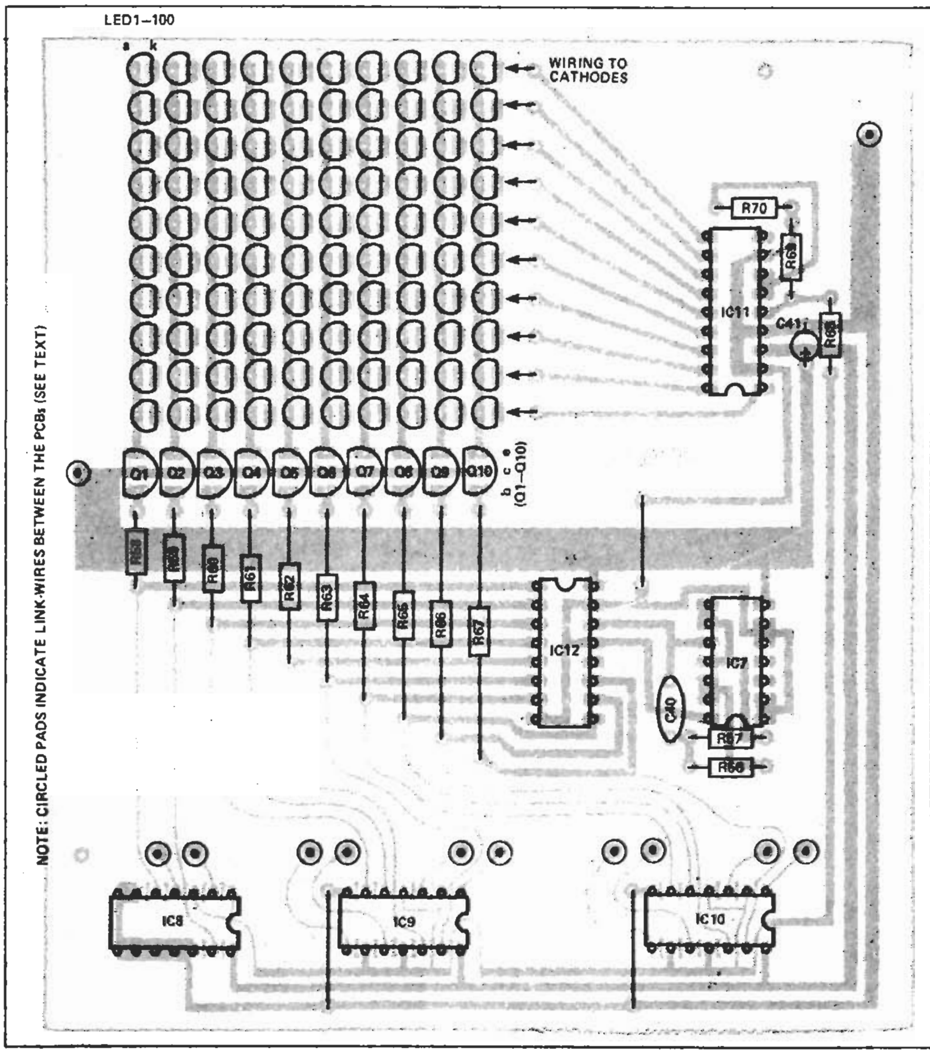
If choosing an alternative case, bear in mind that you'll want easy access to the batteries; the current consumption of the unit is quite high and you'll either have to replace your alkalines regularly or, if you've been sensible, recharge the Nicards.

Anyone who attempts to improve on our PCB design is on his own. Anyone who attempts to build the circuit on Veroboard will be recommended for committal to a mental institution.

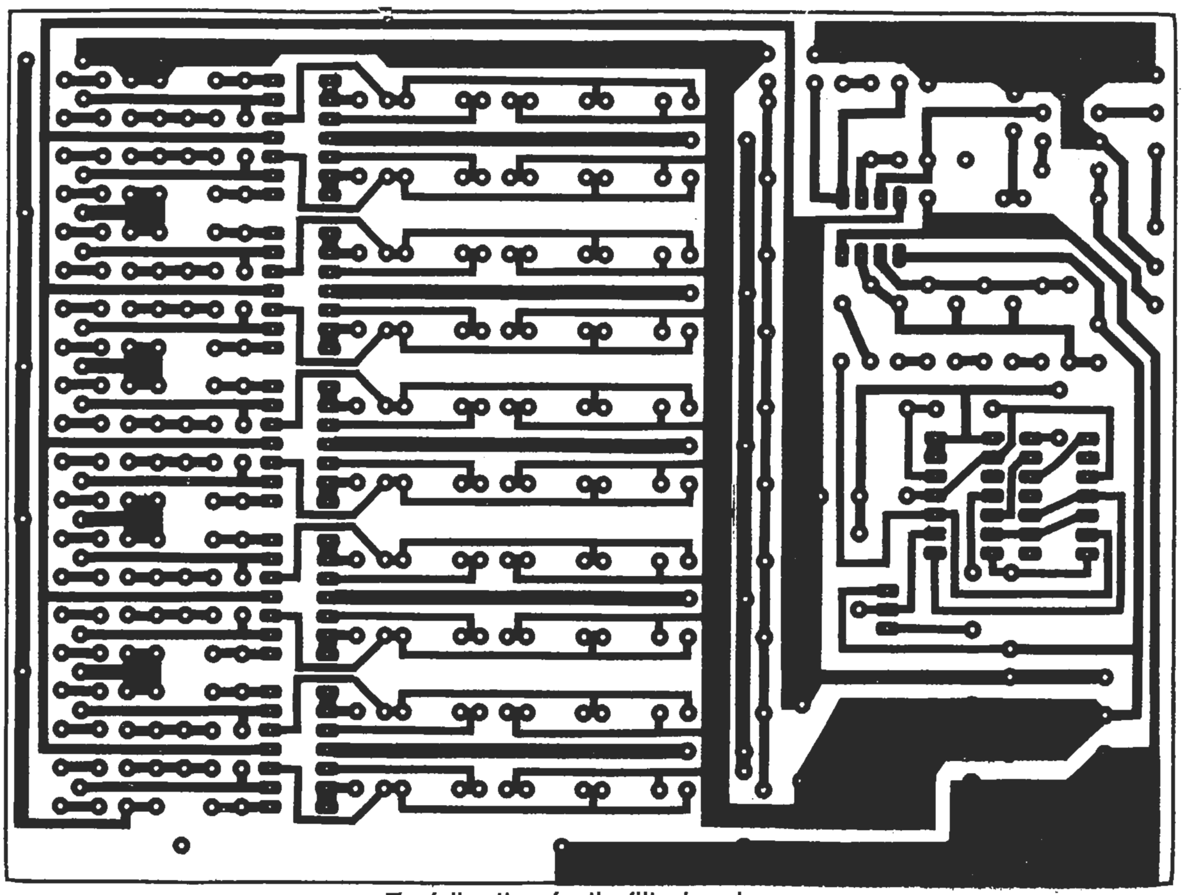
Setting up and Use

The unit can be set up using either the built-in pink noise generator or, better still, with a sine wave oscillator. Adjust PR10 to about 75% of its travel (wiper towards the clockwise direction). With the unit switched on and the sine wave oscillator connected to the external input, by sweeping the oscillator frequency, each column should come up in sequence. Adjust the sine wave frequency and the analyser level control until the 16 kHz column is peaking at a column height of about eight LEDs.

Now, using the same amplitude and without touching the level control, adjust the signal generator frequency until the 8 kHz column peaks



Component overlay for the display board.



The foil pattern for the filter board.

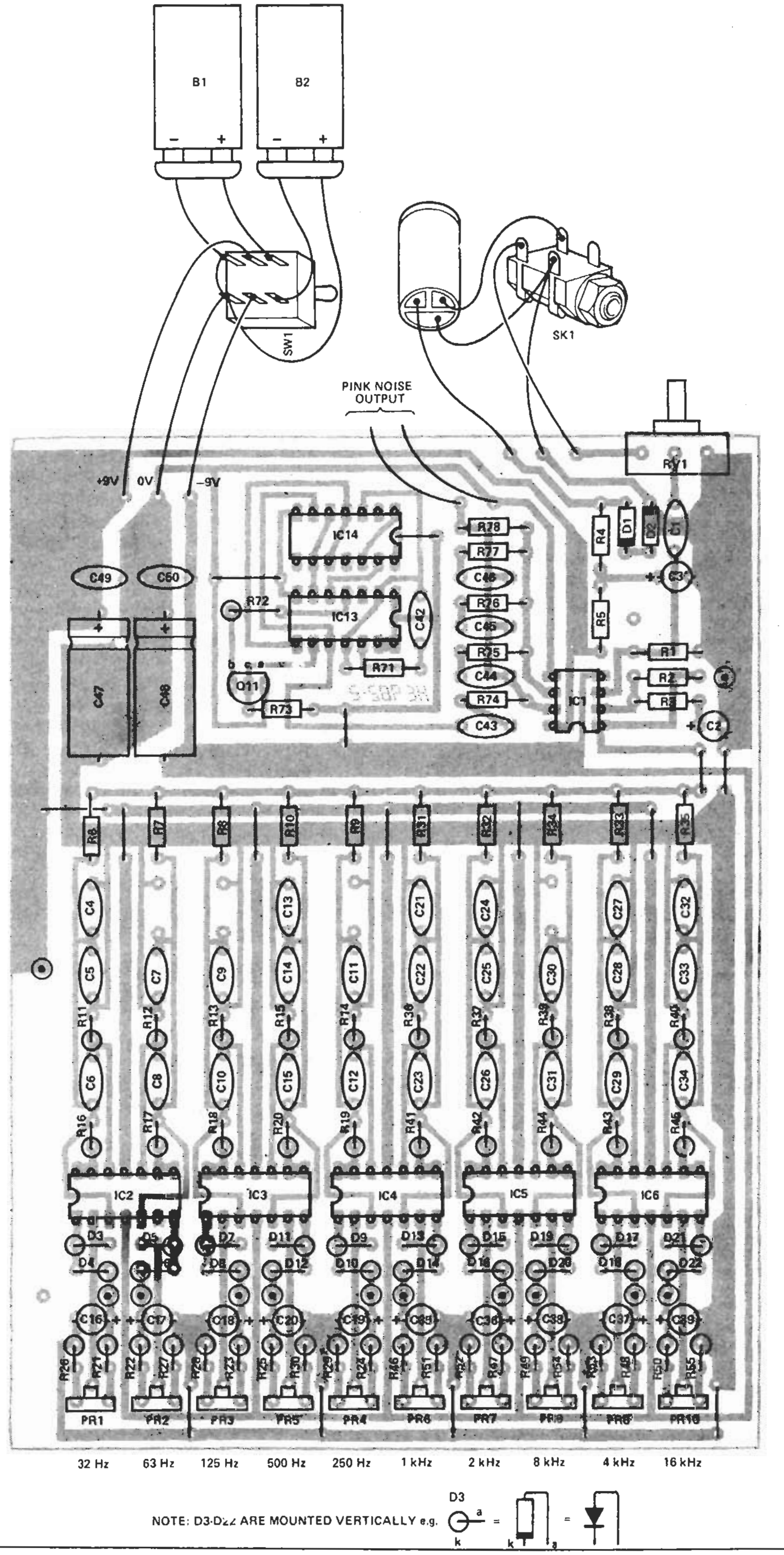


Fig. 7 The filter board component overlay.

and adjust PR9 to give the same height. Repeat this adjustment for each of the filters. Due to component tolerances the actual peak of a filter may not correspond exactly to its nominal centre frequency. The 16 kHz filter has the greatest loss which is the reason for starting with it near its maximum gain.

If a sine wave oscillator isn't available, connect the pink noise output to the external input and adjust the presets to give an even response across the 10 channels. Each column should be approximately the same height; due to the nature of noise, the top of the columns may jump up and down slightly and this should be averaged out by eye. If one of the columns appears dimmer than the rest, replace the transistor that drives that column; if only a single LED appears dim then it must be replaced by as

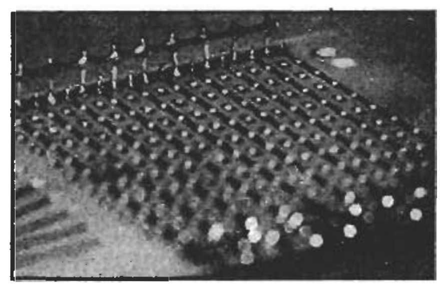


Fig. 5 View of the LED display from the foil side.

we've pointed out, the method of construction make this a bit tricky. It's a good idea to either buy good quality LEDs or test them individually for duds before commencing construction.

To measure a room set-up, feed the pink noise into the hi-fi or PA system via a cable from the listening position and adjust the graphic equaliser controls until a flat response is indicated.

A final point; the microphone us-

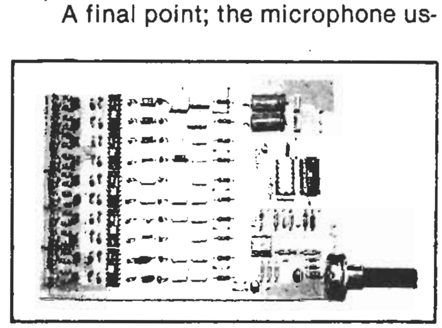


Fig. 6 The completed display board.

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ed must be fairly flat or its frequency response will affect the measurements you're making. If you use one with a limited bandwidth it's possible to use the presets to compensate; however, to do this properly you'll need to play the pink noise into the mike via a sound system/location you already know to be flat. ETI