Spectracolumn

PROJECT

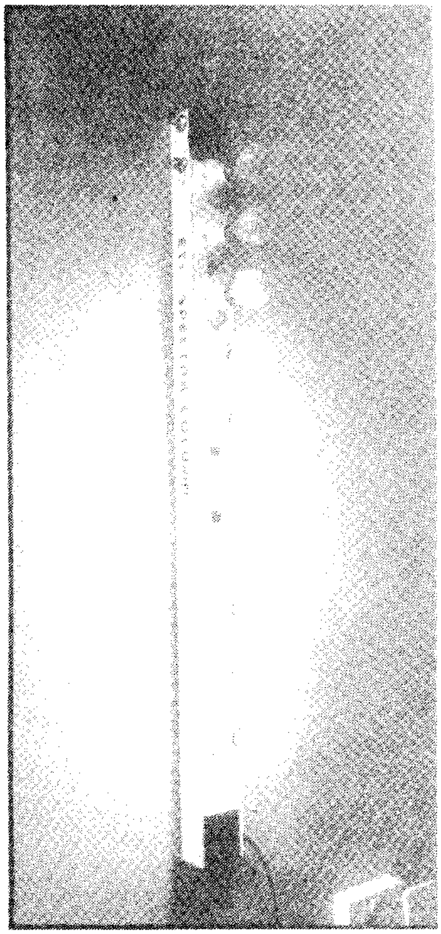
With this project we throw some light on the problem of how to jazz up your disco or party. This cost-effective, crammed-with-everything light column can be used singly or in groups to dazzle the dancefloor. Design by Roray Holmes.

THE ETI SPECTRACOLUMN is an up-market sound-to-light system; by this we mean its lighting effect is a cut above the average 'three bulb' system, although its cost is not. Ten 15 to 100 W bulbs, arranged in a column, respond to the intensity of music (or any sound signal) within a pre-selected frequency range. It works like a giant bargraph voltmeter; the more energy in the chosen frequency band, the more bulbs will illuminate, forming a column of light that rises up from the floor and follows the rhythm of the music. The display system is very versatile; it can be built with any type of bulb in any configuration, and may be expanded for large parties or discos. Multiple columns can be set to adjacent frequency bands to build into a giant spectrum analyser and display system. Imagine — a kilowatt light column devoted to each octave across the whole audio spectrum!

In designing the band-pass filter system, we have made use of the latest switched capacitor filter IC, the MF10. This device contains two second order filters whose cut-off frequencies are directly controlled by a square-wave clock input. Clock frequency control removes the constraint of having to use high tolerance filter network components and the associated difficulty of altering the filter frequency. The clock, and thus the filter frequency, can be set from a logic divider chain to provide any frequnecy in octave increments. We have configured the MF10 as a low-pass filter in cascade with a high-pass filter to allow complete control of the filter band. The upper and lower frequency limits may be set independently under logic control using rotary switches. There is no setting up or filter tuning required and the entire range of octaves is implemented with very few components.

On The Circuit

With the price of modern triacs and some economical design work from ETI, what seems to be a complex system in fact turns out to have only about $50 worth of parts (less the PCB and lightbulbs). Since the triacs don't need heatsinking, we adopted the 'let's get it all on one board' philosophy, and did exactly that. Even the small crystal mike that picks up the audio signal is mounted on the PCB to provide complete isolation between the sound equipment and the power line. Mounting a single board directly with all the bulbs in the column housing also removes the inconvenient cables that often make the dancefloor a dangerous place to negoti-



Ten white light-bulbs, hanging on a wall . . .

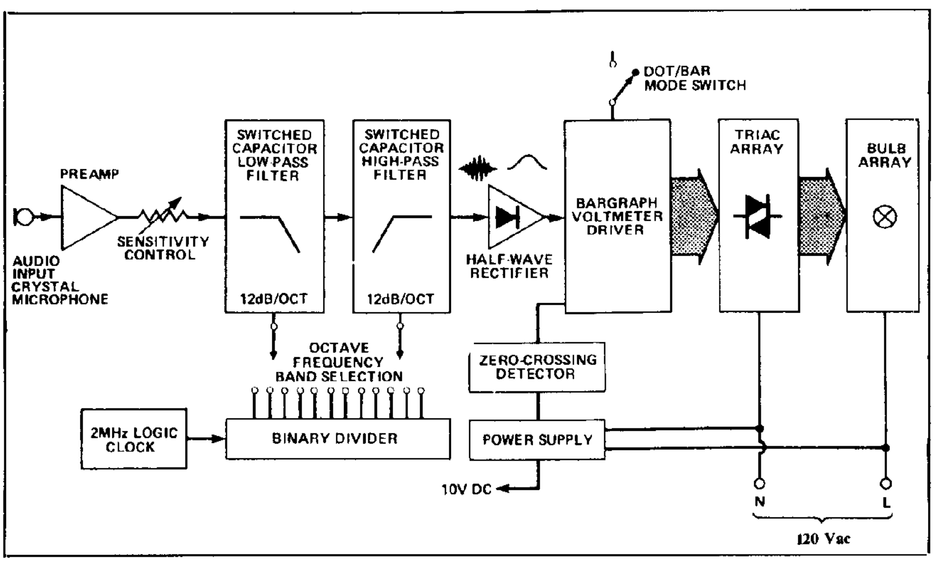


Fig. 1 Block diagram of the Spectracolumn.

FEATURES

- Drive 10 100 W bulbs in bargraph or dot display
- Zero-crossing switching give RFI elimination
- Logarithmically proportional display to correspond with music volume
- Independent high-pass and low-pass filters, 12 dB per octave
- Digitally-controlled switched capacitor filters eliminate setting up
- Pass band switchable in octave increments over 10 octaves anywhere in the audio spectrum
- Internal crystal mike gives complete isolation from sound equipment
- All parts on one PCB powered directly from the line

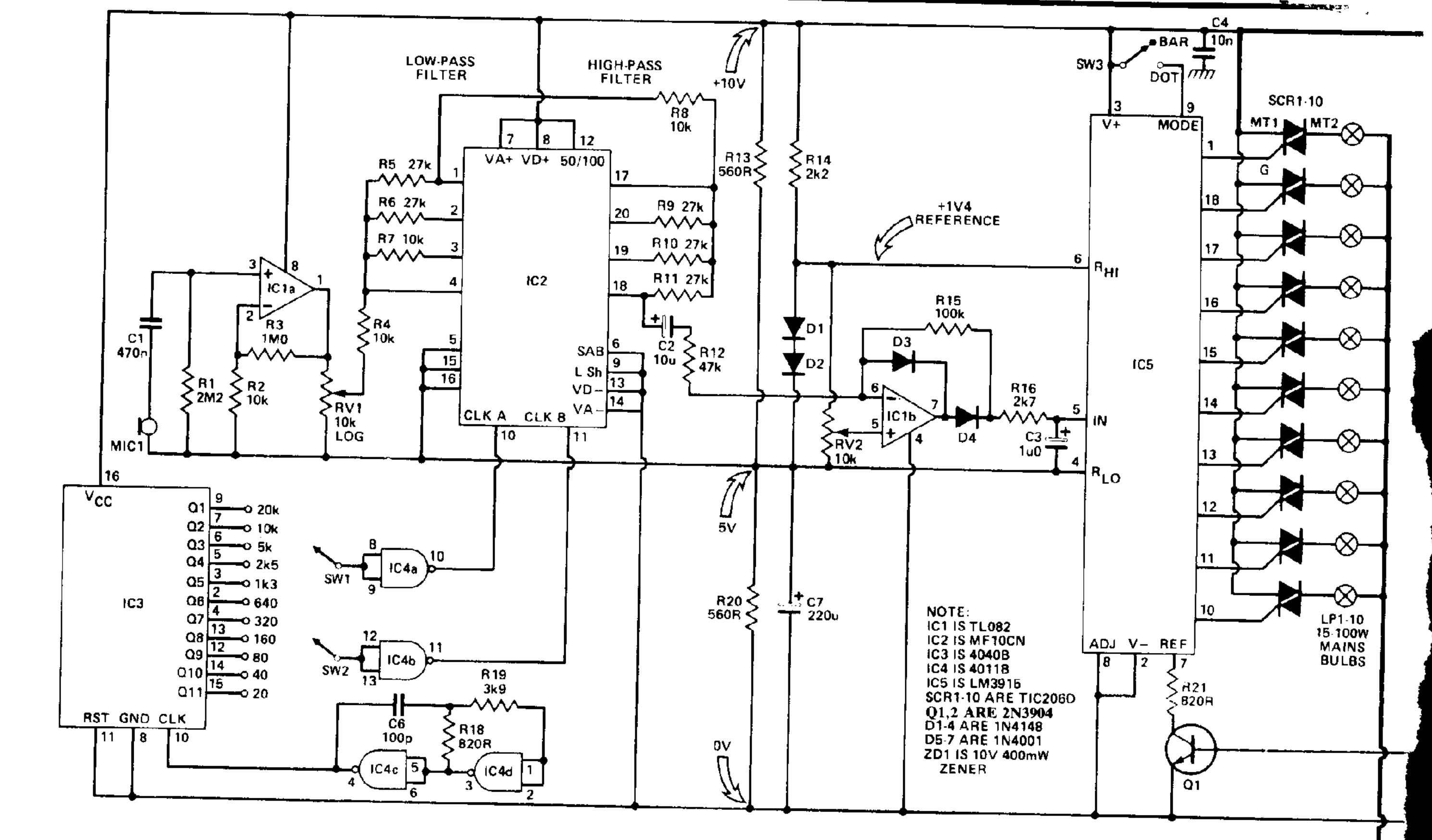


Fig. 2 Circuit diagram for the complete Spectracolumn.

TABLE 1

		FREQUENCY (Hz)	
DIVIDER OUTPUTS	DIVIDED CLOCK	RESULTING FILTER Fc	STANDARD CENTRE
$Q_1(\div 2)$	1M	20k	
$Q_2(\div 4)$	500k	10k	16k
$Q_3(\div 8)$	250k	5k	8k
$Q_4(\div 16)$	125k	2k5	4k
$Q_5(\div 32)$	62k5	1k25	2k
$Q_6(\div 64)$	31k2	625	1k
$Q_7(\div 128)$	15k6	312	500
$Q_8(\div 256)$	7k8	156	250
$Q_9(\div 512)$	3k9	78	128
$Q_{10}(\div 1024)$	1k9	39	64
$Q_{11}(\div 2048)$	980	20	32

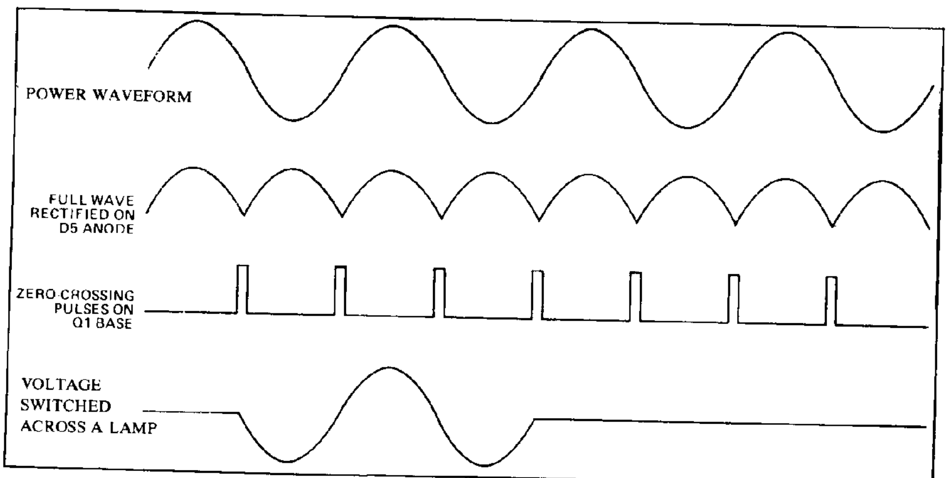


Fig. 3 Triac zero-crossing switching waveforms.

ate. Finally, the design features zero-crossing triac control, so your sound equipment won't be plagued with RFI.

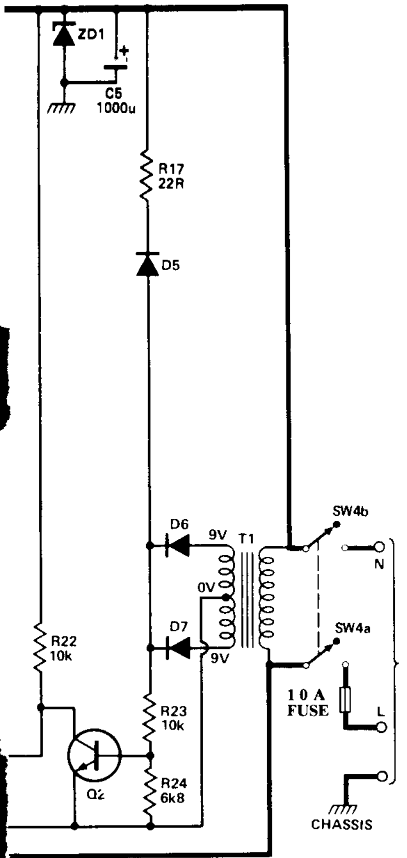
Using the system couldn't be easier; just plug it into the power and switch on! No other connections are needed, because the internal mike picks up the music signal. The sensitivity control is turned up as required for the sound level, and a 'background' control is available which moves the illumination 'baseline' up or down the column, so increasing or decreasing the amount of light. With no sound it acts as a giant dimmer control.

The display could be hung on the wall, as we did for our photograph, or stood vertically on the floor. Large sheets of acetate (available from most good art shops) may be wrapped around the entire column to provide a coloured tube, which also tones down the display. But keep the plastic well away from the light bulbs!

The alternative is to use coloured bulbs. A three column system, using red, green, and blue for the bass, middle, and treble ranges would be an ideal starting system for most disco light shows. The filters could, for example, be set at 20 Hz to 312 Hz, 312 Hz to 2.5 kHz, and 2.5 kHz to 20 kHz. As more Spectracolumns are added into the system the filter ranges can be instantly amended according to taste; but watch out for the current rating of your power sockets!

Construction

All the components except the controls are mounted on our PCB. The triacs, the transformer, and even the microphone are mounted on board, as the overlay diagram of Fig. 4 illustrates. Assembly should begin first with the links, then resistors, followed by ICs and so on. IC sockets should be used as a good precaution, but note that IC5 is an 18-pin device



and IC2 is a 20-pin DIL! Follow the overlay diagram for the orientation of all the components and solder in everything except the PCB-mounting transformer, the triacs, and the crystal mike.

The metal heatsink tab of the triacs has been used to form a screw terminal for the lamp connections (it's connected internally to the central leadout wire MT2). Hence the middle terminal lead of all the triacs must be completely cut off, which immensely simplifies board design too. The remaining two leads are inserted into the board and a nut and bolt are used to clamp the metal tab to the PCB. The bolt protrudes above the component side and a further washer and nut can be added to create a screw terminal. When all the triacs are bolted in place their leadout wires should be soldered and cropped as normal. The lamp wires will be retained on the screw terminals using solder tags.

The PCB-mounting transformer has been used simply for convenience and should be soldered in as a normal component. Other types could also be used provided they are connected to the PCB pads as per the circuit diagram. Bolts should also be fitted, in the same manner as the triacs, to make screw terminals on the pads marked for the mains connections. The photographs of our completed PCB show these terminal connections.

Our crystal microphone insert was 23 mm in diameter; it should be mounted last. The metal shielding case of the insert

How it Works

The block diagram of Fig. 1 illustrates the different parts of the system. Sound from a microphone is amplified and fed through both low-pass and high-pass filters (digitally controlled); the resulting audio signal is then rectified to produce a voltage envelope proportional to the sound intensity within the pre-defined frequency band. This envelope is displayed using a bargraph voltmeter IC to drive triac-switched power bulbs which light up in a column according to the instantaneous sound level. A simple power supply provides both the 10 V DC rail and the 120 Hz signal for zero-crossing triac control.

Figure 2 shows the complete circuit diagram for the Spectracolumn. The audio signal provided by the music or other sound is picked up by the microphone insert MIC1 and amplified by IC1a, which is configured as a straightforward non-inverting amplifier with a gain of 100. The high input impedance required by the crystal mike is set by R1 to be about 2M0.

The audio input from this gain stage is taken via the sensitivity control. RV1 (acting as a potential divider) to the input of the filter system at R4. The audio filter system is built out of an MF10 monolithic switched capacitor filter. This IC contains two identical second order (12 dB per octave) filter systems which can be configured in a number of different modes, with the filter corner frequency being determined by a single square wave clock input.

We have used the MF10 to construct both a low-pass and a high-pass filter, which are wired in cascade. The resistor values shown have been chosen to give a pass band gain of 3 and a Q of 1. The cut-off frequencies are set to be 1/50th of the applied clock signals, which can be independently varied for each filter. Using high and low-pass filters in cascade results in a band-pass type of response, where the bandwidth can be very effectively controlled using the two input clocks, and positioned in any part of the spectrum. The clock on pin 10 of the MF10 controls the low-pass filter determining the upper frequency limit, and the clock input on pin 11 determines the high-pass filter's corner frequency, thus setting the lowest frequency that will be passed.

The clock signals are generated and selected using a separate block of CMOS logic circuitry. IC4c and d are configured as a standard CMOS astable to provide the master clock of 2 MHz. This clock is fed directly to the counter divider chip IC3 (a 4040). The Q outputs progressively divide the clock frequency by two to give those frequencies shown in Table 1. As music lovers will know, dividing the frequency thus will give us equal octave increments; the entire audio bandwidth is thus catered for using the 11 outputs of the 4040. The two remaining gates of IC4 take their inputs from the common pole of each 10-way rotary switch, SW1 and SW2, buffering the outputs from the divider chip and providing selectable clock frequencies to program the high and low-pass filters.

The band-pass filtered audio signal is coupled via C2 to a precision half-wave rectifier, built around IC1b. A positive-going audio envelope thus appears across C3. R16 determines the attack time constant and R15 the decay time constant. Potential divider RV2 supplies an offset voltage derived from the 1V4 reference to the non-inverting input terminal of the op amp IC1b. This allows a 'background' voltage level to be superimposed on the envelope voltage, giving an independent control of the light column's illumination. The 1V4 reference is created by the forward voltage drop across D1 and D2 which are biased by resistor R14; this reference is also used to feed the internal resistor chain of the LM 3915 at pin 6 of IC5. The LM3915 converts the envelope voltage applied at the pin 5 signal input to an array of 10 switched outputs. Pin 4 is the ground reference for the signal and resistor chain voltages; it is tied to the 5 V 'pseudo ground' rail. This half supply-volts rail is derived from the low-impedance potential divider R13,20.

Direct drive from IC5 to the triacs is achieved by connecting the neutral to the positive rail on IC5 and the common MT1 terminals of all the triacs. The switched outputs of IC5, which provide constant current, are taken directly to the gates of the triacs and the bulbs are placed in series with the triacs in the returning live lead. Now, resistor R21 is normally used for setting the output drive current of the LM3915, going from the pin 7 reference to ground.

In our arrangement, however, it is switched to ground using Q1. Thus when Q1 is off, the constant current sources that drive the gates of triacs SCR1 to 10 will all be disabled, and the triacs cannot turn on. Q1 is driven by brief pulses derived from the zero-crossings of the power cycle; in other words, when the AC cycle reaches 0 V (which occurs 120 times per second), transistor Q1 turns on and allows the triacs to be triggered on only at this moment. The triacs automatically turn off again as the line current falls away to zero, assuming there is no further drive signal. For the triac to turn on then, the corresponding output from IC5 must be 'active' due to the sound level, and at the same time as a zero-crossing pulse occurs. By turning on the triacs and thus the lamp current flow only when the line voltage is close to zero, the problems of radio frequency interference are effectively avoided.

The circuitry is powered from a 10 V supply rail, regulated by the 10 V zener diode ZD1, and decoupled by C5. The centre-tapped 9-0-9 V transformer is full-wave rectified by D6 and D7; Q2 is driven by the 120 Hz signal at the junction of D6 and 7 to detect the zero crossing points. As the voltage cycle falls down to zero the voltage on the base of Q2 also falls. When it goes below 0V6, Q2 will turn off (the zero crossing point), thus allowing Q1 to turn on. D5 and R17 isolate the full wave rectified DC from the 10 V power rail.

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is connected internally to one of its terminals. This shield terminal should be identified (use an ohm-meter) and wired to the mid-rail reference as shown on the overlay; ensure that the wire used is very thin and flexible. A piece of sponge foam about the size of the mike should be stuck to the PCB and the mike may then be glued on top of this to provide a resilient mounting, free from direct vibration pickup.

An electret condensor type of mike insert could also be used and would probably give better quality sound pickup. They usually come with their own internal FET preamplifier, which requires a 1V5 power supply. Luckily, the 1V4 reference terminal indicated on the overlay is ideal for this job, and may be wired directly to the insert.

When the board is completely assembled, the two control pots and the mode switch can then be wired up as indicate Veropins should be inserted as terminals at the appropriate points. The two rotary switches for the frequency selection should also be wired up using ribbon cable as shown in the diagram. Note that the rotary switches are both set to select one out of 10 corner frequency outputs from the PCB and the rotary switches are offset by one frequency band relative to each other i.e., the upper limit switch ranges from 40 Hz to 20 kHz while the lower limit ranges from 20 Hz to 10 kHz.

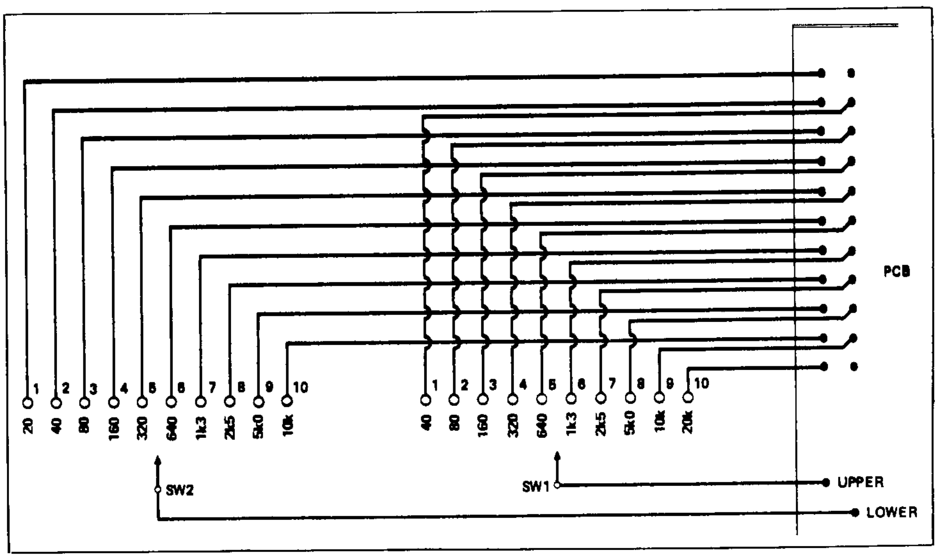


Fig. 4 This diagram shows how to wire up SW1 and SW2.

Testing And Setting Up

After wiring up the controls, some initial tests can be made before completing the assembly. Initially, do not connect any light bulbs and do not plug in any ICs; but do remember that all parts of the circuit are effectively live. Connect the power as shown via a double pole toggle switch and a 10 amp fuse, and then switch on. Using a voltmeter check that there is about 10 V across C5 and 5 V across C7. 10 V should also appear across pins 8 and 4 of IC1, pins 8 and 13 of IC2, pins 16 and 8 of IC3, and pins 14 and 7 of IC4 and pins 3 and 2 of IC5. If all is well, unplug from the power and insert all the ICs. One light bulb can now be wired onto the SCR5 terminal, its other lead returning to the line. Set the upper limit switch to 5 kHz, and the lower limit to 640 Hz; this gives a fairly broad frequency band for vocal testing. The unit should be turned on again with SW3 set in bar mode. Altering the background control RV2 should cause the bulb to switch on and off at some point. As the bulb switches off continue to turn RV2 in the same direction to the end of its travel. The background illumination control is then at its zero setting. Now, depending on the sensitivity setting, a loud noise should re-illuminate the bulb. Increasing the sensitivity control should eventually allow the bulb to come on with normal speech volume. If this test works satisfactorily, then all the bulbs can be wired up to their corresponding terminal posts and the entire display can be tested.

Turning the background control up should result in the successive illumination of bulbs; now turn it down to zero, when all the bulbs should be off. Increasing the sensitivity control will now allow sound to illuminate all the bulbs. Having established a good sensitivity setting, different types of music from a record deck or radio can be used to check the different frequency bands available on the rotary switches. The display can be switched to dot mode at any time, which provides an interesting effect with constant light level.

A Case In Point

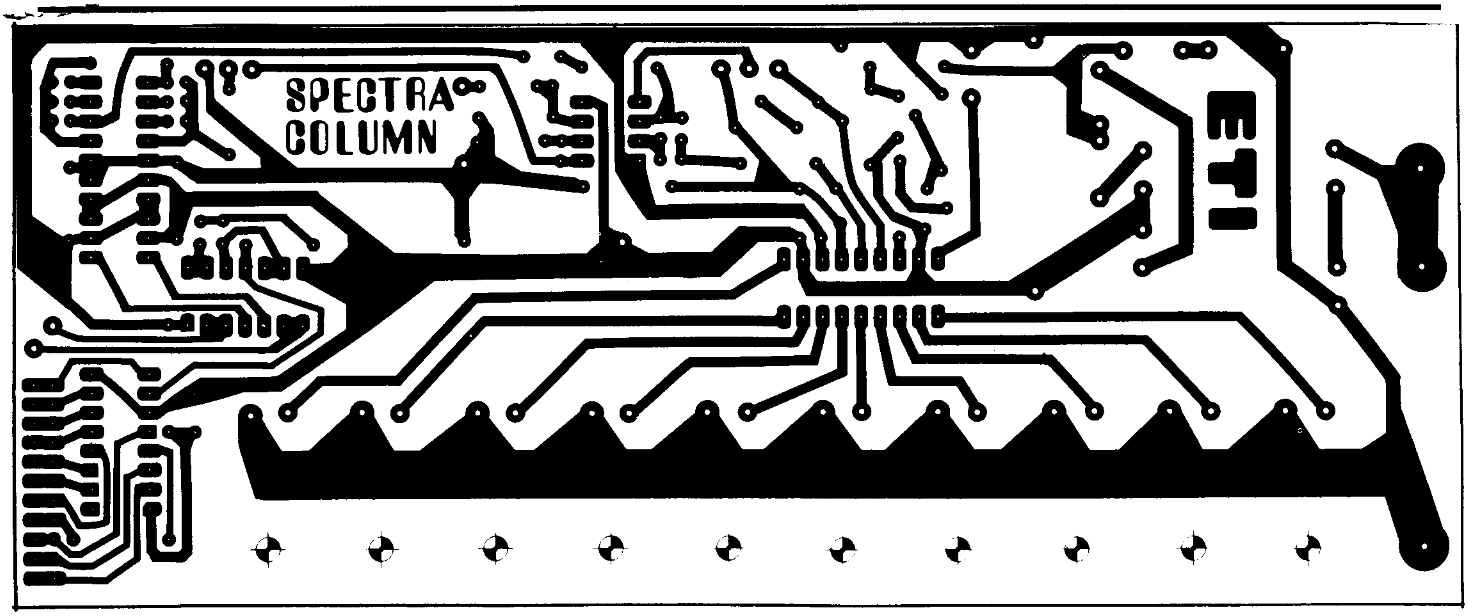
The actual hardware construction of the light bulb arrangement is very much a matter of personal choice. We used large white plastic bulb holders, and mounted the entire column and PCB in a fluorescent light case that was at hand. The case was grounded and provided a nice self-contained unit. Batten-mounting bulb holders could equally well be screwed down to a long strip of wood and the electronics mounted in a spearate diecast box. The photographs illustrate the construction method we used.

A number of important points should be noted with the final assembly. Owing to the circuitry used, the positive rail is directly connected to the neutral; therefore all parts of the circuit should be treated as being effectively *live* since somebody could easily swap the line and neutral leads by accident at the plug end. Consequently we suggest:

- The PCB should be mounted in a metal case on insulating pillars or blocks.
- The case should be grounded, but there should be no other connection between the PCB and the case.
- The mode switch and on-off switch should both be power rated and have a current rating sufficient for the total of the bulbs used.
- The pots and rotary switches should all have plastic spindles and plastic knobs. Ideally the metal pot cases should be insulated from the chassis, or they could be soldered directly to the PCB terminals such that only the plastic spindles pass through the chassis.
- For the reasons of isolation, the microphone must stay inside the case; and on no account try to connect up the mike input to a direct audio signal from your sound equipment (this could be done *only* with an audio isolating transformer).

Notes On Modifications

For those with the urge to experiment, here are some notes on modifying circuit values: R3 decreases the mike preamplifier gain; decreasing R4 and R8 increases the filter gain; increasing R6 and R10 will increase the Q of the filters; R18 alters the frequency of the master clock, currently set at 2 MHz; R21 determines the drive current to the triacs; increase C3 or R16 to increase the attack/decay display time constant; R16 could be a 22k variable pot.



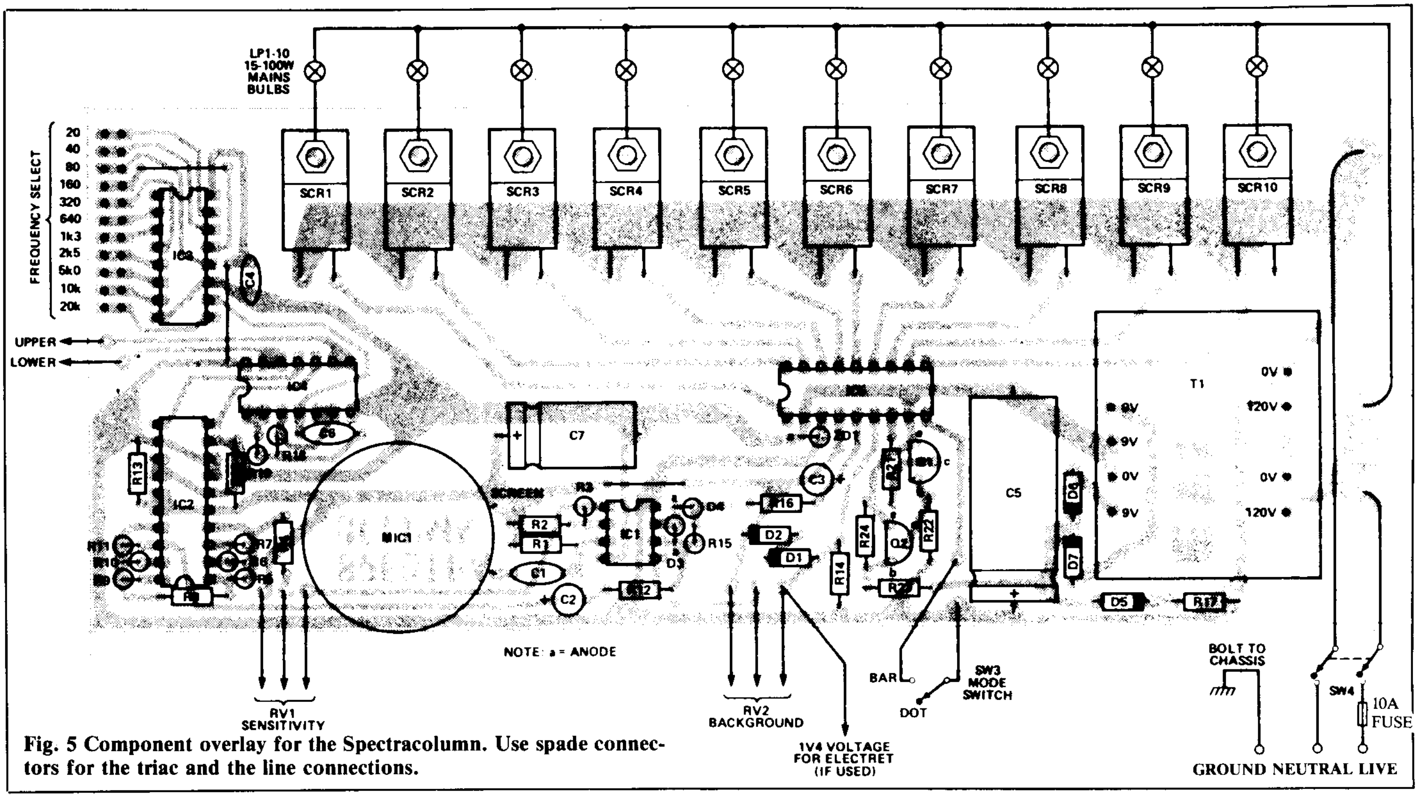


Fig. 5 Component overlay for the Spectracolumn. Use spade connectors for the triac and the line connections.

Parts List

Resistors (all ¼W, 5%)

R1	2M2
R2,4,8,22,23	10k
R3	1M0
R5-7,9-11	27k
R12	47k
R13,20	560R
R14	2k2
R15	100k
R16	2k7
R17	22R
R18,21	820R
R19	3k9
R24	6k8

Potentiometers

RV1,2	10k log or linear

Capacitors

C1	270n polycarbonate
C2	10u 16 V tantalum
C3	1u0 35 V tantalum
C4	10n ceramic
C5	1000u 25 V axial electrolytic
C6	100p polystyrene
C7	220u 16 V axial electrolytic

Semiconductors

IC1	TL082
IC2	MF10CN
IC3	4040B
IC4	4011B
IC5	LM3915
Q1,2	2N3904
SCR1-10	TIC206D
D1-4	1N4148
D5-7	1N4001
ZD1	10 V 400 mW zener

Miscellaneous

SW1,2	1-pole 12-way rotary switch
SW3	SPST toggle switch
SW4	DPST power switch
MIC1	crystal mike insert
T1	9-0-9 3 VA transformer
FS1	10 A fuse and fuseholder

PCB; 10 bulbs, 15-100 W, and holders; fluorescent lamp fitting or other suitable case.

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