PE Sound Synthesiser 7

REVERBERATION AMPLIFIER
RING MODULATOR
PEAK LEVEL METER

By G.D.SHAW

THIS month the Ring Modulator, Peak Level Meter circuit and Reverberation Amplifier will be described.

THE RING MODULATOR

With the ring modulator the combination of tones follows a complex inter-relationship in which each frequency is continuously compared with and modified by the other, The resultant output provides a tone which consists of the sum and difference of the two constituent frequencies appearing at the same time and irrespective of the phase angle relationship. A typical output waveform is illustrated in Fig. 7.1.

A simplified version of a transistorised ring modulator manufactured in integrated circuit form by Silicon General, the SG3402N, is shown in Fig. 7.2 and it will be seen that the device consists essentially of a pair of cross-coupled differential pairs jointly controlled by a third. Two inputs—carrier and modulator—are required and it is important to differentiate between them since the input characteristics are dissimilar. Application of equal amplitude signals to both inputs will provide an output showing about 3dB voltage gain over either input. Removal of the modulator with the carrier still applied will result in attenuation of the output signal by about 50dB but if the input situation is reversed the output attenuation is only about 35dB.

Although designed primarily for communications work the SG3402N is capable of working satisfactorily at quite low audio frequencies by the simple expedient of increasing the value of the input and decoupling capacitors.

The frequency response of the prototype Ring Modulator is shown in Fig. 7.3 and will be seen to be effectively flat over most of the audio frequency spectrum. The theoretical circuit is shown in Fig. 7.4.

The maximum input signal to the SG3402N should not normally exceed 50mV and thus resistive attenuators are employed to raise the signal level, at the input sockets, to one more compatible with the signal level normally routed around the Synthesiser. With the value of resistors employed in the attenuators the maximum input signal at the sockets is thus 500mV.

IC2 serves to amplify the output to about 1·5V at the rated input levels and measured at an input frequency to both channels of 1kHz. The ring modulator shares a circuit board with the peak level meter and the board layout is shown in Fig. 7.5.

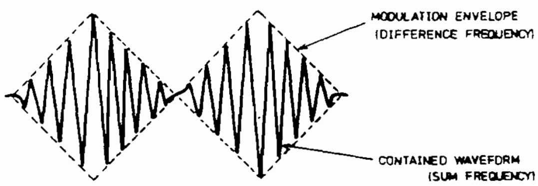


Fig. 7.1. Typical output waveform of the Ring Modulator

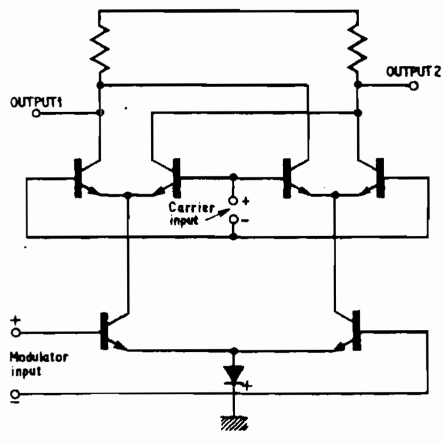


Fig. 7.2. Schematic of a transistorised Ring Modulator

RANGE OF SOUNDS PRODUCED

The type of modulation produced by the ring modulator is wholly unique and thus also is the range of sounds which can be achieved. If two pure tones are modulated together and one of them is reduced in frequency the resultant output would follow the pattern shown in the table below which relates the sum and difference output frequencies with the carrier and modulator input frequencies:

	Frequency (Hz)						
Carrier	700	600	500	400	300	200	100
Modulator	400	400	400	400	400	400	400
Sum	1100	1000	900	800	700	600	500
Difference	300	200	100	0	100	200	300

It can be seen that, whereas the resultant sum reduces in frequency at the same rate as the carrier, the frequency reduces until it reaches zero (carrier and modulator frequencies equal) and then, as the carrier continues to fall to a frequency lower than that of the modulator, the difference frequency begins to increase at a proportional rate.

When the inputs to the Ring Modulator carry harmonics in addition to the pure tones then further series of frequency relationships are established for each of the component harmonics relative to one another and to the respective fundamentals.

When the inputs to the Ring Modulator are of symmetrical triangular waveform, such as those generated by the v.c.o., an extremely complex set of frequency relationships is established due to the fact that, in common with the square wave, the triangular waveform consists of a long series of odd harmonics.

SOUNDS PRODUCED

The Ring Modulator may be used in many fascinating ways from the creation of truly "out of this world" sounds, the transposition of tones, bell-like sounds, Dalek voices and so on.

In the transposition of tones the only stipulation is that the modulating frequency should be higher than the carrier (this latter input consisting of the signal for treatment). For any range of carrier frequencies the modulator frequency has to be calculated or determined empirically, to provide the best overall effect.

An interesting experiment can be carried out by cascading two Ring Modulators. The first uses the v.c.o. output to drive carrier and modulator inputs so that the output is the octave, or second harmonic, of the v.c.o. frequency. The output of the first Ring Modulator is used to drive the modulator input of the second whilst the carrier input is derived direct from the v.c.o. Thus the difference frequency of the second Ring Modulator will follow, exactly, the performance of the v.c.o. while the sum frequency will approximate to a quarter-tone accompaniment about 1¼ octaves higher. There are very wide possibilities for further experiment in this kind of mode.

A true bell tone is very complex and is difficult to imitate with exactitude. A fairly close approximation may be achieved by adjusting two v.c.o.s to a mid-range frequency, say 4kHz, such that there is a slow beat between them. One v.c.o. then drives the carrier and the other v.c.o. the modulator input of the Ring Modulator. A very important characteristic of the bell-like sound lies in its envelope presentation and this will be dealt with in detail in next month's article.

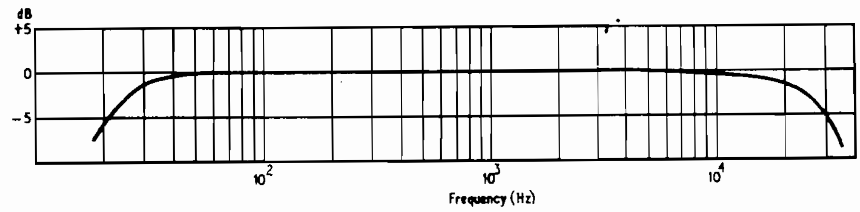


Fig. 7.3. Frequency response of prototype Ring Modulator

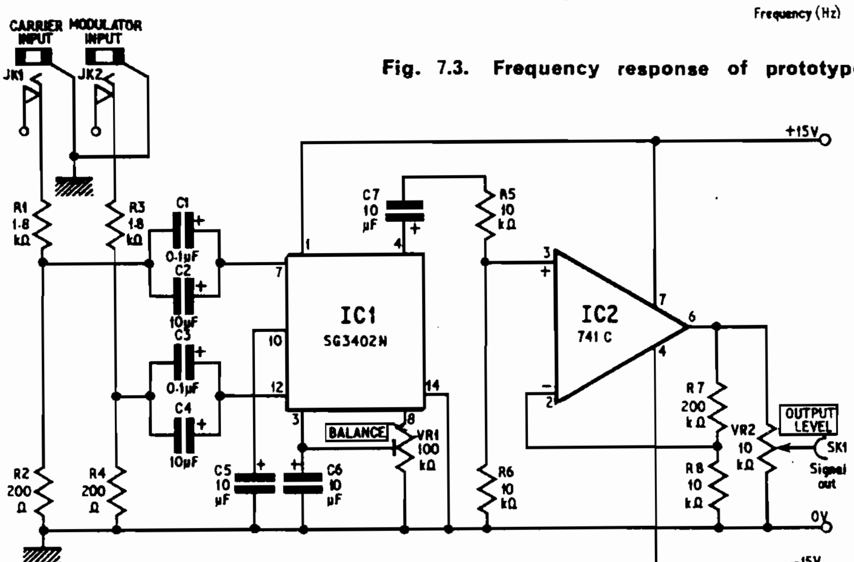


Fig. 7.4. Ring Modulator circuit

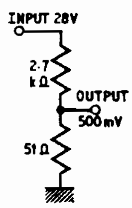


Fig. 7.5. Simple resistive attenuator for use with the Ring Modulator

PEAK LEVEL METER

RING MODULATOR

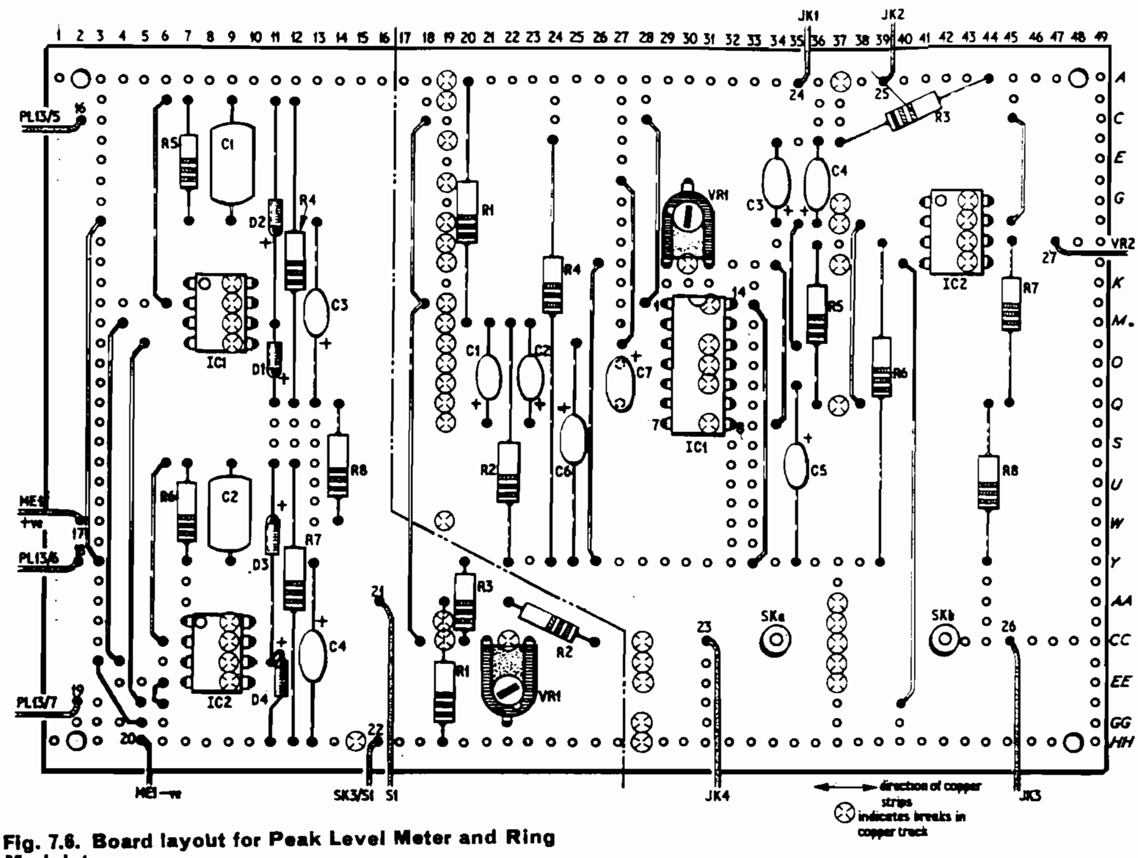


Fig. 7.6. Board layout for Peak Level Meter and Ring Modulator

COMPONENTS . . .

PEAK LEVEL METER CIRCUIT

Resistors

R1	91k Ω
R2–R3	20k Ω (2 off)
R4	91k Ω
R5–R6	3·3k Ω (2 off)
R7	91k Ω
R8	110k Ω

All 5% ½ watt carbon

Capacitors

C1–C2 1,500pF (2 off)
C3–C4 22μF 16V tantalum

Potentiometers

VR1 50k Ω carbon preset

Diodes

D1–D4 IN914 (4 off)

Integrated Circuits

IC1-IC2 741C (2 off)

Miscellaneous

ME1 MR38P SEW panel meter (G. W. Smith Ltd.)
SK3 2mm miniature socket

RING MODULATOR

Resistors

R1	1·8k Ω	R5	10k Ω
R2	200 Ω	R6	10k Ω
R3	1·8k Ω	R7	200k Ω
R4	200 Ω	R8	10k Ω

All 5% ½ watt carbon

Capacitors

C1	0·1μF 35V	C3	0·1μF 35V
C2	10μF 16V	C4–C7	10μF 16V (4 off)

All tantalum

Potentiometers

VR1 100k Ω carbon preset
VR2 10k Ω miniature moulded carbon

Integrated Circuits

IC1 SG3402N
IC2 741C

Miscellaneous

JK1, JK2 3·5mm miniature jack sockets (2 off), SK1 2mm miniature socket, Veroboard as required

Dalek voices are produced by modulating a modified speech waveform at about 15-20Hz. Speech, and certain types of music waveforms, can present a very peaky characteristic. The peaks are multiplied and added to in the Ring Modulator and thus, if the signal is remodulated several times, or if the initial frequency is high enough, the final output contains a large proportion of sound which bears a remarkable resemblance to white noise. The cure for this problem is to limit the dynamic range of the signal.

Although exact details for such a procedure lie outside the scope of this series a passable method is to feed the offending signal to one of the input amplifiers and, observing the output on the oscilloscope, adjust the gain of the amplifier so that a large proportion of the peaks are suitably clipped. If insufficient gain is available to allow an adequate degree of clipping the input amplifiers may be cascaded. The achievement of clipping means, of course, that the amplifier output signals are swinging between the positive and negative saturation levels and it will be necessary to attenuate the signal quite considerably. Fig. 7.5 shows a simple resistive attenuator which will give a signal of about 500mV from a 28V source.

CONSTRUCTION

Construction of the Ring Modulator is quite straightforward and the only critical requirement lies with the observation of polarity of the tantalum capacitors. Reversal of any of the capacitors will result in noisy operation and, in the case of the output capacitor, no operation at all. Tantalum capacitors have been specified in order to conserve space and there is no reason why 10V electrolytics should not be used with an alternative layout.

SETTING UP

Setting up the Ring Modulator consists only of providing a modulation balance. Set VR1 to its mid position and apply a common sine wave signal to both inputs. The output of the ring modulator will be a sine wave which is twice the frequency of the applied signal. If the modulation is out of balance alternate peaks of the output signal will be at different amplitudes. VR1 should be adjusted to bring the peaks into line at which point the modulation is balanced.

It is a wise precaution to repeat this measurement from time to time to adjust for settling down changes in the circuit.

THE PEAK LEVEL METER

In the prototype Synthesiser the meter circuit was based on a precision rectifier built around a pair of operational amplifiers arranged in such a way as to eliminate the effect of the diode forward voltage drop. Although the circuit proved to be very responsive it was found, in practice, to present a number of disadvantages. In the case of a.c. signals the meter would read only the r.m.s. value and although it was possible to determine the actual peak-to-peak value by application of a form factor for known wave shapes the determination of peak-to-peak values for complex waveforms proved to be a very hit and miss affair.

In a similar manner, when endeavouring to set up reasonably accurate programming voltage levels, the rapid response of the meter frequently made it difficult to establish the peak value with any certainty.

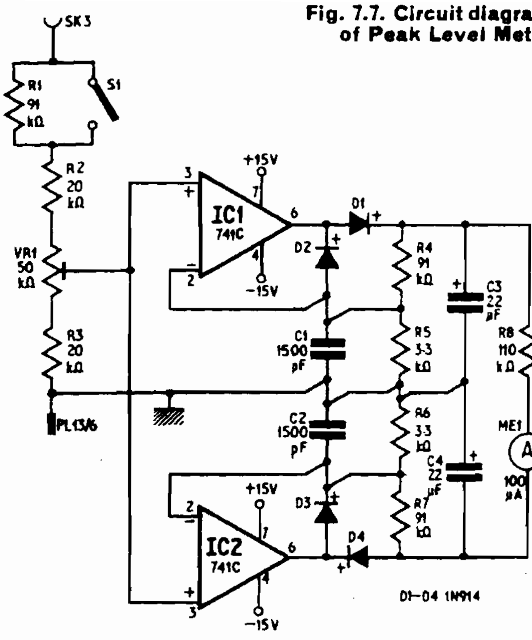


Fig. 7.7. Circuit diagram of Peak Level Meter

In consequence it was decided to redesign the meter circuit to provide a peak reading facility which would be independent of waveform configuration and which would have a reasonably long decay time to ease the establishment of transient level readings. The final circuit is shown in Fig. 7.7. IC1 and its associated circuitry is used to read the positive going peaks while IC2 deals with the negative side of the signal. The operation of the circuit is as follows.

CIRCUIT OPERATION

The input sensitivity of the circuit at the i.c. is about 200mV for full scale deflection of the meter. A positive going peak of this value appearing at the input of IC1 will swing the output positive to a level determined by the values of R4 and R5, about 6·2V with the values shown, and capacitor C3 will charge at a rate determined essentially by the effective current output of the i.c. The charging time for C3 is thus rather less than 2mS.

If, after the capacitor is charged, the 200mV peak is replaced by a lower amplitude peak the tendency would be for the i.c. to swing hard negative due to the effect of the positive voltage from the capacitor appearing at the inverting input via R4. This tendency is prevented by D2 which limits the negative excursion of the output to about 700mV.

Capacitor C3 discharges through R4 + R5 in parallel with R8 + R (meter) and with the values shown takes about 1 second. C1 serves to decouple a.c. from the feedback loop and thus effectively extends the accurate range of the meter to about 15kHz.

The negative reading side of the circuit around IC2 operates in the same way. The circuit is adjusted to give full scale deflection with inputs of 1·0V and 0·5V by means of the attenuator R1, 2, 3 and VR1.

In using the meter it should be borne in mind that the peak values recorded represent only half the total peak to peak value of the signal being measured and this applies whether the signal is symmetrical or assymmetrical about zero. When measuring low frequency programming signals of greater than 1Hz the minimum reading of the meter between peaks does not represent the lowest level of programming signal.

This particular meter circuit can be used to measure the peak level of single transients of not less than 2mS duration.

ADVANTAGES

In tape recording the peak level meter scores heavily over the more conventionally employed v.u. meter. This latter meter will record what is essentially the mean value of signal presented to the recording amplifier and if, as is generally the practice, the mean level is kept to about −3dB transient peaks are likely to be clipped or otherwise distorted. The use of a peak level meter, on the other hand, enables the peaks to be kept within the limits imposed by the recording amplifier and thus enhances the overall quality of the recording.

THE REVERBERATION AMPLIFIER

Reverberation, or re-echo, in varying degrees is a characteristic observed in the majority of large halls, public buildings, cathedrals and so on. In a properly designed and proportioned hall the inherent reverberation characteristic can provide a high degree of enhancement to the sounds occurring therein.

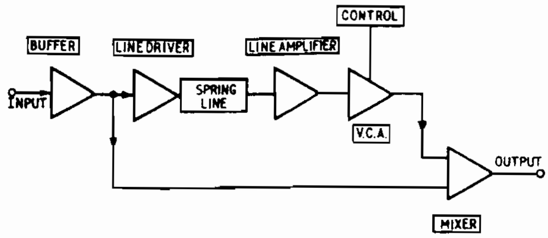


Fig. 7.8. Block diagram of Reverberation Amplifier

COMPONENTS . . .

REVERBERATION AMPLIFIER

Resistors

R1	10k Ω	R15	3·9kΩ
R2	270k Ω	R16	110k Ω
R3	39k Ω	R17	13k Ω
R4–R5	10k Ω	R18	13k Ω
R6	47k Ω	R19	0·5 Ω (3W wire-wound)
R7	10k Ω	R20	0·5 Ω (3W wire-wound)
R8	6·8k Ω	R21	56 Ω (3W wire-wound)
R9	390 Ω	R22	10k Ω
R10	10k Ω	R23	10k Ω
R11	20k Ω	R24	120k Ω
R12	10k Ω	R25	10k Ω
R13	10k Ω	R26	See text
R14	10k Ω		

All 5% ¼ watt carbon unless otherwise stated

Capacitors

C1, C2	0·1μF ceramic
C3	620pF silver mica
C4	4μF 15V elect.
C5	0·1μF ceramic
C6	0·1μF ceramic
C7, C8	10μF 16V tantalum
C9	0·003μF ceramic
C10	See text
C11, C12	100μF elect. 22V

Transistors and Diodes

TR1	2N2905	TR3	BC209C
TR2	2N2219	D1, D2	IN914

Potentiometers

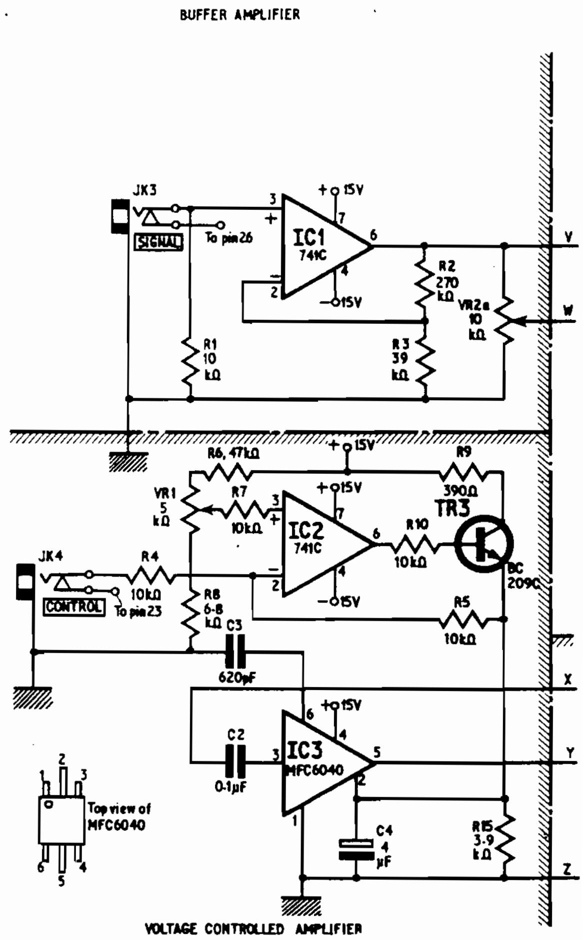
VR1	5k Ω lin.
VR2a/b	10k Ω lin. (two ganged)
VR3	10k Ω lin.

Integrated Circuits

IC1, IC2	741C
IC3	MFC6040
IC4, IC5, IC6	741C

Miscellaneous

Spring line type HR42 (Henry's Radio), JK3, JK4 3·5mm jack sockets (2 off), SK2 2mm miniature socket



There are a number of ways in which a reverberation characteristic may be simulated and for the Synthesiser the spring line has been adopted. The spring line consists essentially of a coiled wire, usually steel, which is supported at each end in a compliant mounting. At the supported ends of the wire are fitted electro-magnetic transducers. The line driving transducer is excited by an electrical signal and the varying field produced causes mechanical wave motion to be set up in the spring line. When the wave motion reaches the far end of the line it sets up an electrical disturbance in the line output transducer which is, in turn, amplified and added to the original signal.

Part of the original mechanical wave motion is reflected back down the spring line where it serves to modify further on-coming waves.

Because a mechanical wave motion travels much more slowly than its electrical counterpart the signals received by the line output transducer are delayed in relation to their source, such delay being a function of the length of wire used in the spring line. Thus the mixing of the mechanically routed signal with the source signal constitutes the addition of an echo. However, since the wave motion, once initiated, travels back and forth along the line until its amplitude becomes negligible, multiple echoes are received and added to the original signal.

The spring provides a further useful feature having its origin in the fundamental resonance of the system. When the driving signal passes through the frequency at which the system resonates the output is characterised by a sudden increase in amplitude which can be as much as three times the value of the normal mean signal. Similarly when the input signal passes through any of the harmonics of the resonant frequency there is an increase in output signal amplitude, and this despite the fact that the useful range of the HR42 spring line, specified for this project, is limited at its upper end to about 4kHz. In the prototype unit quite high resonant peaks were occurring at up to 25kHz.

The combination of multiple echoes and varying amplitude imparts a very useful "singing" quality to an otherwise uninteresting sound.

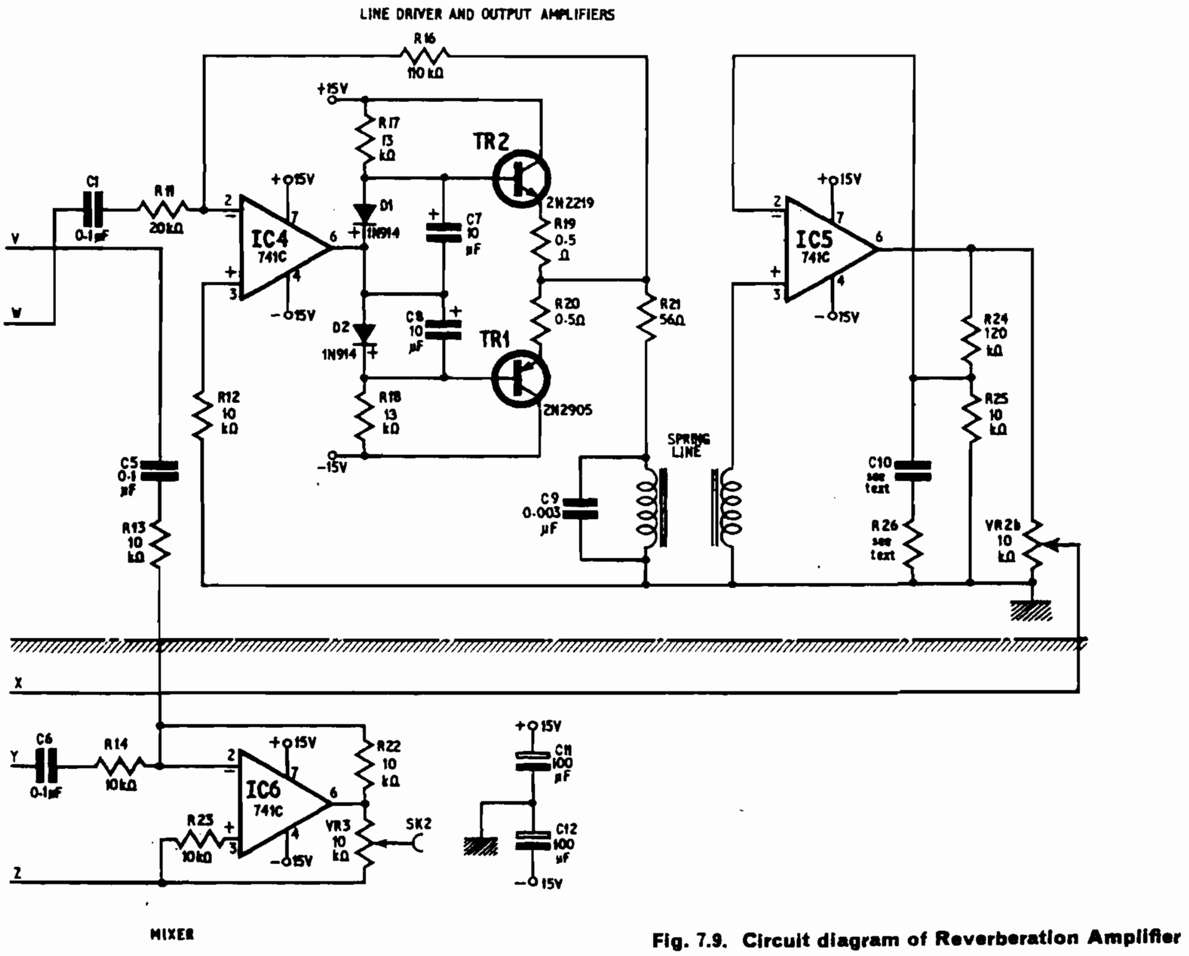
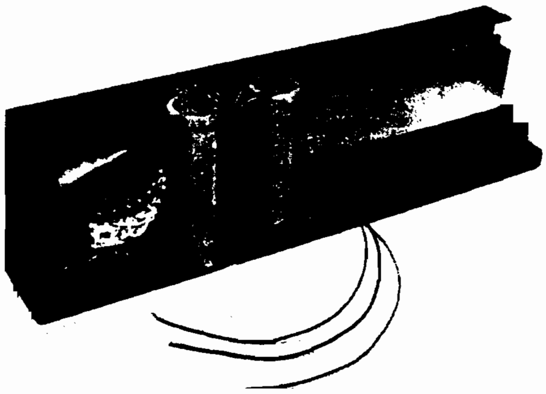


Fig. 7.9. Circuit diagram of Reverberation Amplifier



The spring line unit is attached to the p.s.u. sub-frame

DESIGN CONSIDERATIONS

In the prototype the line driving amplifier employed a single transistor operating in what was effectively Class A. The current consumption was thus quite high even in the quiescent state and small variations in the power supply rails gave rise to noise in the system which was apparent when the line was not being driven hard. For the modular version of the Synthesiser therefore the Reverberation Amplifier was redesigned to reduce current consumption, reduce hum and noise to negligible proportions and to enable a complete divorcing of the voltage controlled part of the system so that the amplifier may be built as a separate unit outside the Synthesiser project altogether.

A fortuitous advantage of the re-design provides sufficient power capability to drive two HR42 or one HR42 and one HR162 spring lines in series. It is also theoretically possible to drive up to four of the above spring lines in any combination although this latter method has not been tested.

The advantage in using more than one spring line in the system lies in the fact that it is rare for two units to have identical resonances and delays and thus two or more units can only improve the overall reverberation characteristic.

CIRCUIT ACTION

The Reverberation Amplifier is shown in block form in Fig. 7.8 and the circuit diagram in Fig. 7.9. The input signal is led to a buffer stage, which has a gain of about six, and the output is divided to drive the line amplifier and output mixer. The line driving amplifier consists of a pre-amplifier built around a 741 and having a gain of about five which, in turn, provides drive to a complementary pair of output transistors having a current gain of about a hundred and arranged in what may be described as a modified form of Class B. The output from this latter stage provides drive direct to the spring line through a current limiting resistor.

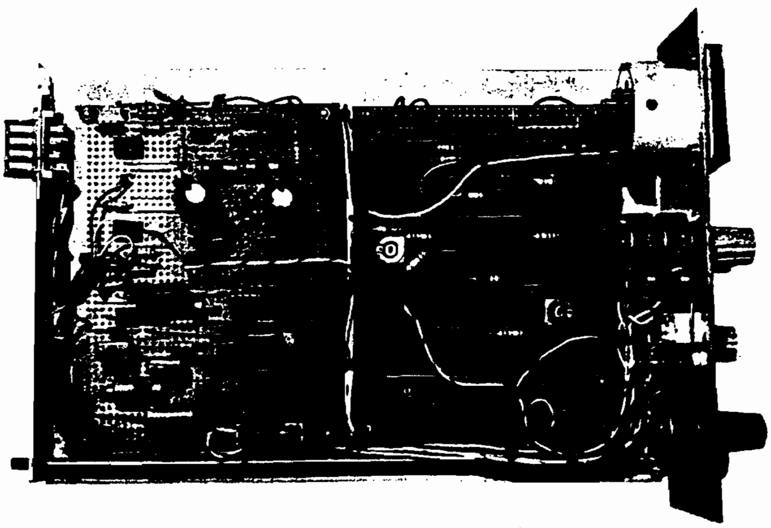
The output from the spring line is amplified by another 741 having a gain of about nine and then led to the input of the voltage controlled amplifier based on the Motorola MFC6040. This latter device has a maximum gain of 13dB and a maximum attenuation of about 77dB relative to the input signal which should not normally exceed 500mV r.m.s. The overall gain of the spring line route is thus arranged so that when the line is being driven hard at a non-resonant frequency, and with the v.c.a. at maximum gain, the output of the v.c.a. is equal to the output of the buffer stage and thus the mixer is receiving equal components of reverberated and non-reverberated signal.

The choice of component values for C10 and R26 may be arrived at by experiment on the basis of the measured response of individual spring lines. To limit the gain to 6dB a value of 2·5 kilohms for R26 will suffice.

The value of C10 is calculated on the basis of the frequency at which the 6dB gain is required. For a frequency of 15kHz the value of C10 is 1nF.

The v.c.a. is controlled by a separate 741 arranged in the differential mode. The non-inverting

Photograph of the assembled module



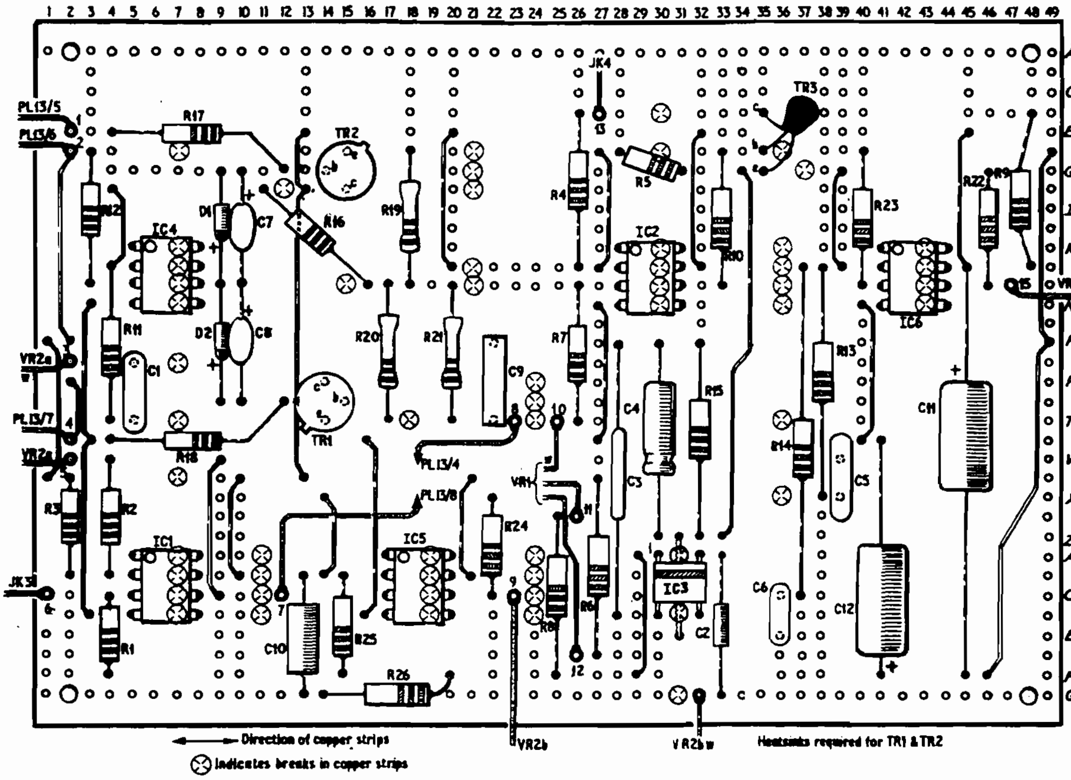


Fig. 7.10. Board assembly of Reverberation Amplifier

input is driven by a positive voltage derived from the divider R6, R8, and VR1. The high and low ends of VR1 are thus at 3·0V and 1·75V respectively. The inverting input of the 741 is driven by a control voltage which should have a swing of 2·5V maximum.

With VR1 at its minimum setting a control voltage swinging from zero to −2·5V will have the effect of attenuating the output of the MFC6040. With VR1 at its maximum setting a control voltage swinging from zero to +2·5V will have the effect of amplifying the output of the MFC6040 from −77dB to +13dB relative to its input signal. The inverting input of the 741 acting as control amplifier is prewired to a ramp generator which will, of course, provide the first mode of v.c.a. operation described due to its negative going output.

If external automatic control of reverberation is not required it is essential that a grounded jack plug be inserted into the control socket otherwise the output of the control amplifier will be insufficient to swing the MFC6040 through its full range.

The current sink at the control input of the MFC6040 is specified as being 2mA but on several

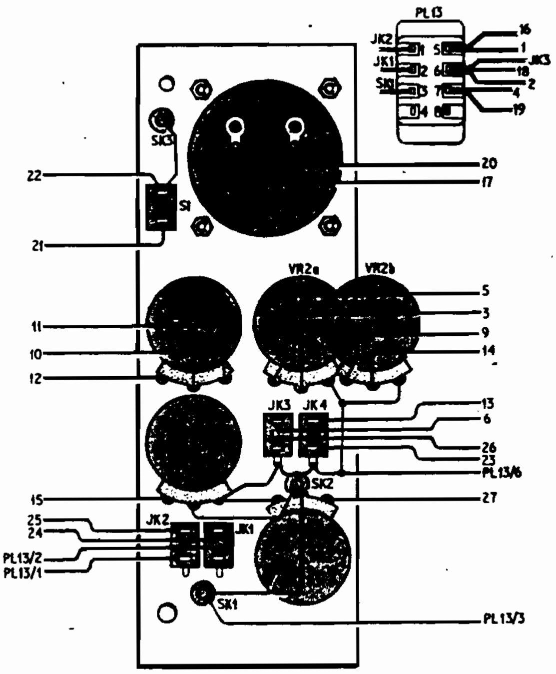


Fig. 7.11. Front Panel Wiring

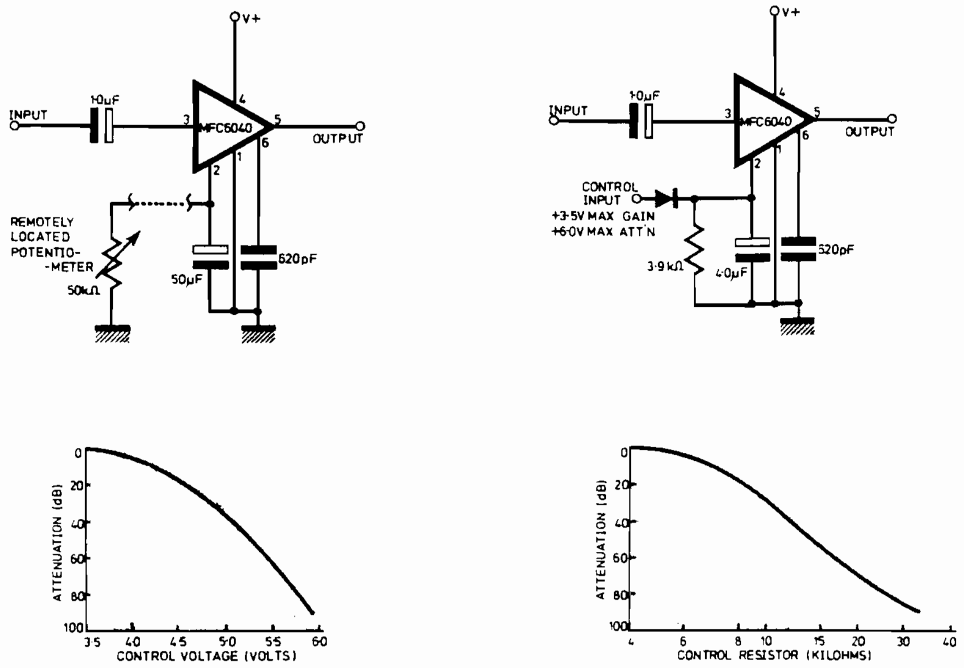


Fig.7.12. Two methods of providing external control using a potentiometer and voltage control input. The relevant response curves are located below each circuit. Here the 0dB reference equals a 13dB gain

tested in this mode of operation quite a wide variation in current sink was noted, the highest being 25mA. Consequently it is prudent to provide a series transistor on the output of the control amplifier, with overall feedback, in order that the 741 is not overloaded. The effect of overload will not necessarily damage the 741 but it could result in a reduction of the output voltage swing which would, in turn, affect the operation of the 6040.

OMITTING THE V.C.A.

For some possible applications the use of voltage control will not be required and, in these instances, the MFC6040 and associated control amplifier may be omitted from the circuit entirely. In these circumstances the gain of the line output amplifier, IC5, will have to be increased by a factor of 0·33 if equal reverberated and non-reverberated components are required at the mixer. The output of IC5 is, of course, led direct to C6 on the mixer in these latter circumstances.

CONSTRUCTION

Fig. 7.10 illustrates the recommended circuit board layout. Construction is quite straightforward and the only setting up required lies in checking the signal levels at the outputs of the buffer, line driver and line output amplifiers to ensure that equal signal components from both sources are presented at the mixer when the line is being driven hard at a suitable non-resonant frequency.

Adjustment of the gain of the line output amplifier may be necessary and is dependent upon the mechanical attenuation of the line which may differ unit to unit.

Overall construction of the module should generally follow the pattern previously described and the wiring of the components on the front panel and McMurdo plug are shown in Fig. 7.11.

In this module the McMurdo plug has insufficient ways to carry all the necessary signals and two extra leads are required to carry the control and audio signals to the reverberation amplifier. Reference to the block diagram in the first part of the series will show that the control signal is derived from RG2 while the audio signal is derived from the right channel of the output amplifiers yet to be described. Suitable leads should be run from the respective McMurdo sockets on these latter modules to a point immediately adjacent the left hand Vero endplate and secured to the connector mounting rail by a tie of lacing cord. From this point they should run along the end plate and be trimmed so that they protrude about three inches beyond the front face of the mainframe. Terminated in 1mm miniature plugs, they can be mated with their respective sockets on the Ring Modulator circuit board when the finished module is being inserted into the mainframe.

A fully comprehensive revision of all module interconnections will appear in part nine of the current series.

For the benefit of constructors who may wish to explore the possibilities of the MFC6040, Fig. 7.12 shows two possible methods of providing external control with the resistance attenuation curve and the control voltage attenuation curve of this very versatile device.

In Part 3, VRZ is 100Ω.

Next month: The Envelope Shaper will be described.