sewar

sound effects with the analogue reverb

**We have recently published a number of articles featuring delay lines and the most popular was the Analogue Reverb Unit in Elektor 42 (October 1978). It would appear that this article was greeted with such enthusiasm by our readers that many have been encouraged to experiment further.
The following project has been designed as a 'front end' to the reverb unit with the purpose of allowing greater flexibility with reverb effects. It produces a variable rate clock signal together with five different modulation waveforms that can be used for phasing, vibrato and other effects. A random signal generator is also included for chorus effects. The composite output signal is intended to be connected to the external clock input of the analogue reverb unit.**

It will be seen, when referring to Elektor 42, that the Analogue Reverb Unit (ARU) uses the well known SAD 1024 shift register. As most of our readers will know, this device operates on the 'bucket brigade' principle. Briefly, this is analogous to a chain of buckets from input to output. The sampled signal at the input corresponds to the level of water in the first bucket. At the 'word of command' (clock pulse) this bucket is poured into the second bucket (which was of course empty). At the next word of command the second bucket is emptied into the third and so on for 512 times, the number of stages in one half of a SAD 1024. We should explain to newcomers to electronics that we don't really use water (at least, not yet) and the water level in our mythical bucket is in reality a charge packet on an almost mythical capacitor (they are physically very small).

Back in the real world, it will be apparent that the delay time is dependent mainly on two factors: the number of stages in the shift register (or registers), and the clock frequency. The first is a hardware design parameter and not easily altered, but the clock frequency is something that *can* be varied — and that is where we get to the point of this project.

1

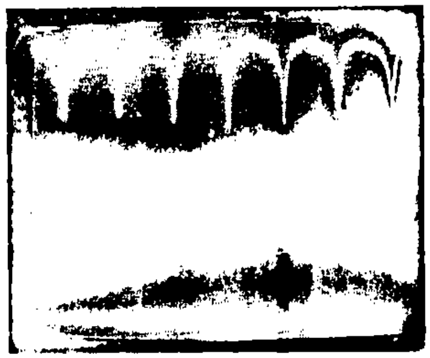


Figure 1. The oscillograph shows the comb-like structure of the phasing effect produced by adding a delayed to an undelayed signal.

A variable clock frequency has rather more going for it than might at first appear. If the output of the delay line is mixed with a 'clean copy' of its input signal the resulting periodic phase cancellation and addition will produce the so-called comb frequency response shown in figure 1. Now, if the clock frequency is raised and lowered the comb will 'open and close'. This in audible terms produces the phasing (or flanging) effect. A chorus effect is obtained by an entirely random variation of the clock frequency. The range of possibilities will now be apparent.

Before getting too deeply involved in this circuit some readers may prefer to become better acquainted with 'bucket brigade' shift registers, and for this the previously mentioned article in Elektor 42 should prove useful.

Table 1

Technical data

Clock pulse generator	
frequency range:	20 kHz to 250 kHz
waveform:	square
amplitude:	15 V p-p
Random modulation generator	
average fluctuation rate:	adjustable
average amplitude:	1.4 V p-p
Periodic modulation generator	
frequency range:	0.1 Hz to 10 Hz
waveforms:	sine, triangle, square, rising ramp sawtooth, falling ramp sawtooth
Power consumption:	± 15 V/50 mA

2

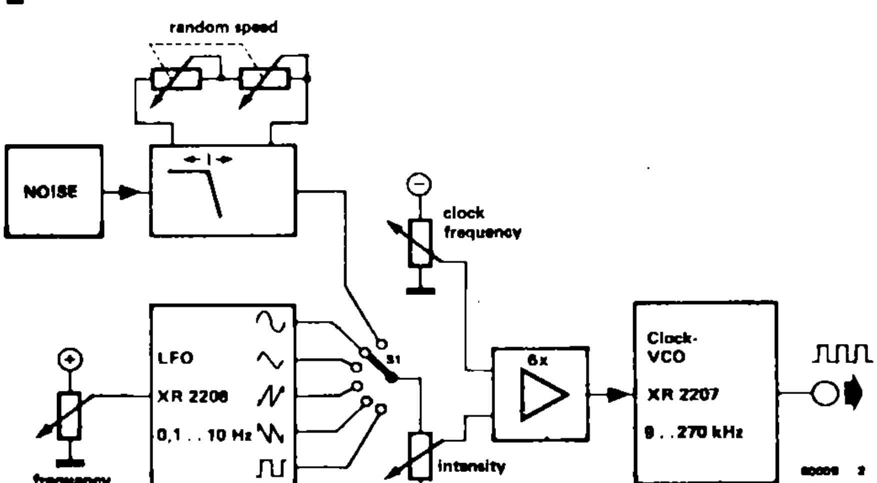


Figure 2. Block diagram of the clock pulse generator. Five different waveforms plus a random modulation signal can be selected.

The external clock

The design target was to develop the maximum in sound effect possibilities. The final concept is shown in the block diagram of figure 2.

The low frequency oscillator (LFO) is variable between 0.1 and 10 Hz and produces five different waveforms: sinusoidal, triangular, rising sawtooth, falling sawtooth and square. As a sixth possibility a noise source generates a random signal which is low pass filtered to limit the passband. The filter roll-off frequency is adjustable for variation of the average speed of the random signal. Switch S1 selects the required modulation waveform and the modulation depth is varied by the intensity control. After amplification the resultant signal controls the frequency sweep of the voltage controlled clock pulse generator (VCCPG?). Figure 3 shows the VCO output frequency as a linear function of the modulation control signal. The frequency modulated output of the VCO is connected to the input of the analogue reverberation unit thereby producing the various sound effects discussed in previous paragraphs.

Sewar Circuit

As can be seen from the circuit diagram of figure 4, the unit is built around three integrated circuits, a function generator (XR 2206), a VCO (XR 2207) and four FET input op-amps housed in one package (TL 084 or TL 074). The circuitry around the function generator (IC1) may be familiar to regular Elektor readers. The oscillation frequency is determined by components C2 + C3, R3, R4 together with potentiometer P4. Since availability of bipolar electrolytic capacitors may be limited, the required capacitance is made up from two 220 μF types connected back-to-back. The resultant 110 μF suffices to bring the frequency down to 0.1 Hz, the upper limit being 10 Hz.

3

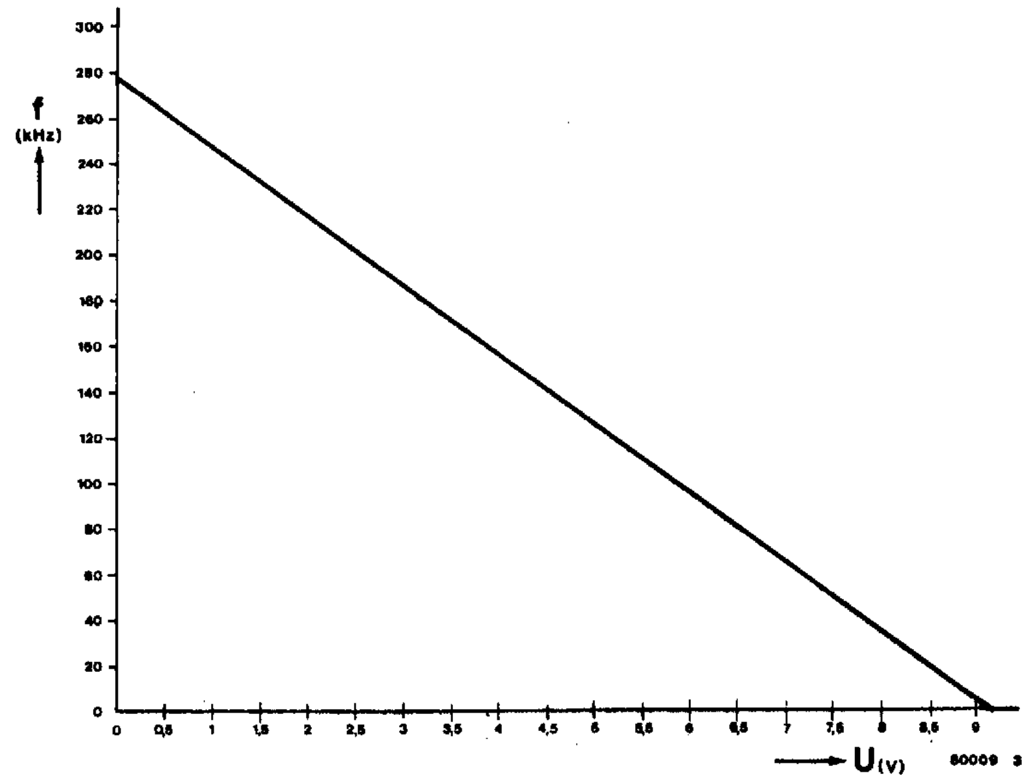


Figure 3. The graph shows the VCO output frequency as a quasi-linear function of the control voltage, in a ratio of approximately 30 kHz per volt.

The output waveforms and amplitudes are defined by the networks connected to various pins of the waveform generator. Switch S1a . . . S1d functions as follows.
Switch position 1 connects the filtered noise generator output to the VCO. The waveform generator is switched off and S1c contacts c1 shorts pin 11 to ground to suppress any stray radiation.
Switch position 2 corresponds to a sinusoidal waveform which is available from pin 2 of the waveform generator. The sinewave is produced by connecting resistor R2 across pins 13 and 14 via contact b2 while contact a2 shorts pin 1 to ground. The amplitude of the sinewave can be adjusted by means of preset potentiometer P3.
Switch position 3 corresponds to a triangular output at pin 2, by disconnecting R2 from pin 13. The amplitude of the triangular waveform can be adjusted by means of P1 which is connected to pin 1 via contact a3.
Switch position 4 corresponds to a positive going sawtooth waveform by removing the short from pin 11 and connecting this pin to the FSK input (pin 9) via contact c4. The positive going ramp of the sawtooth lasts for half of the triangle period, the negative going slope is determined by the resistance of R1 and is much steeper. The sawtooth frequency is therefore, practically twice that of the sinusoidal and triangular waveforms. The amplitude is again adjusted by means of P1.
Switch position 5 corresponds to a negative going sawtooth waveform by moving the bias at pin 1 from P1 to P2 via contact a5, thereby inverting the sawtooth polarity. The output amplitude is now controlled by P2.
Switch position 6 corresponds to a squarewave output. The generator output is now taken from pin 11 via R6 and S1d contact d6. It is clipped to 1.4 V p-p and made symmetrical with respect to ground by the network composed of R5, R6, R7 and the reverse-parallel connected diodes D1/D2. This symmetry obviates the need for a coupling capacitor which would otherwise distort the square pulse shape, especially at low frequencies.
Any DC component at pin 2 of the function generator IC is blocked by the coupling capacitor C1. This DC component is apt to surge when S1 is operated, and these surges cannot be sufficiently bled via the high resistance of P5 alone. However, the reverse-parallel connected diodes D3/D4 become conductive only on these surges and together with R8 speed up the discharge rate of the capacitor.
The random signal is generated as follows. Transistor T1 is used as a noise source. Its base-emitter breakdown comes into effect at around 8 V and makes the transistor behave like a very noisy zener diode. The resultant noise signal is greatly amplified by A1 and A2 in cascade which function as active low-pass filters due to capacitors C6 and C7 in their feedback loops. This combination gives a roll-off frequency of about 10 Hz. The random signal zero frequency component is offset by the bias control P8 at the non-inverting input of A2. The filtered output of A1 + A2 is passed through a further active low-pass filter, A3, with a 12 dB roll-off at an adjustable frequency controlled by P6. This sets the average fluctuation speed of the random signal. The final output is available at selector switch contact d1.
The sweep control signal from the modulation mode selector switch, S1, is attenuated by P5 to the modulation depth desired. This is applied to the non-inverting input of the 16 dB amplifier, A4, whose output determines the oscillator frequency of the VCO, IC2, as shown in the graph of figure 3. The VCO control signal is composed of the periodic or non-periodic waveform from the mode selector switch, plus a zero frequency component introduced at the inverting input of A4. The centre frequency of the VCO is then adjustable by P7 to between 20 kHz and 250 kHz. The stabilised voltage required for this is supplied by the network R20, D5 and D6. Capacitor C9 is the reactive component of the oscillator circuit and this capacitor determines the free-running frequency of the VCO. The power supply for the VCO is stabilised intern-

4

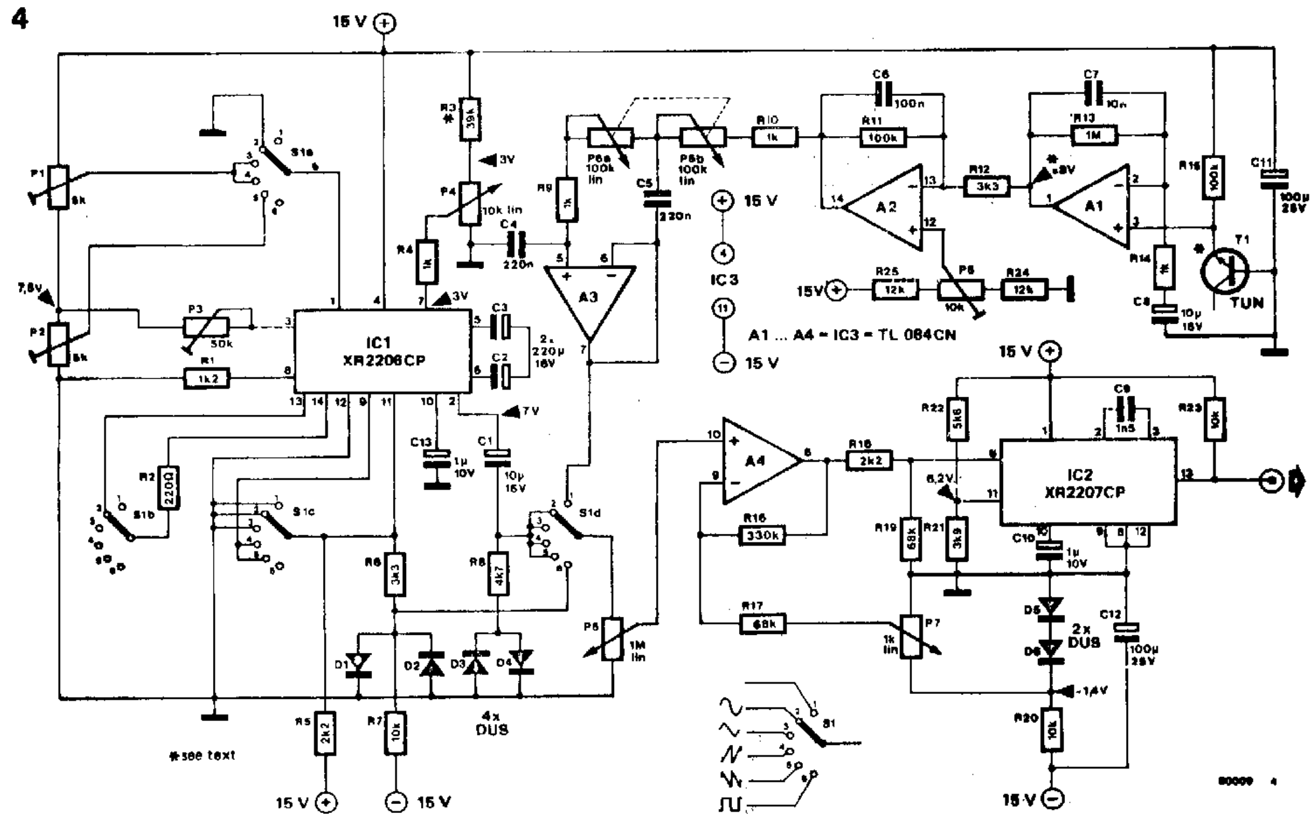


Figure 4. The complete circuit is made up from only three active integrated circuits plus a number of discrete passive components for efficient operation.

ally with the help of capacitor C10. The final squarewave output signal to be fed to the reverberation unit is taken from pin 13 of the VCO. The power supply for the clock generator (± 15 V 50 mA) can be derived from the supply for the reverberation unit.

Construction and setting up

The printed circuit and component layout for the ARU 'front end' is shown in figure 5. Assembly of the printed circuit board should not present any problems if suitable sockets are used for the integrated circuits. Electrolytic capacitors, particularly C1, C2, C3 and C8, should be low leakage types.

Special attention should be paid to the selection of transistor T1. With the circuit parameters given, its standing emitter voltage must lie between 6 V and 9.5 V, this voltage is the same as that of the DC component at the output of the unity gain amplifier A1. If the reading obtained lies outside this range a different device must be tried.

A multi-meter can be used for setting up the circuit parameters although an oscilloscope may be preferable. Test DC levels are indicated at a number of points on the circuit diagram to simplify setting up.

Prior to further measurements, the working range of P7 should be tested. This is done with P5 set to zero output. The voltage on the wiper of P7 should vary from 0 V to around 8.5 V, after which P7 is set to give an output of 5 or 6 volts. The actual figure will serve as a reference around which the modulation signals will swing symmetrically.

The first output test is on the squarewave, for which S1 is moved to the sixth position and P5 set to maximum output. With P4 set for the lowest oscillator frequency (its wiper fully towards R3) the meter reading will fluctuate between a low and a high reading, in a 3 to 5 second period, symmetrically about the reference level established previously. The peak-to-peak amplitude of the squarewave should be some 7 or 8.5 volts. The actual voltages obtained should be noted, since they will have to serve as a standard for the other waveform measurements.

Should the squarewave oscillation stop or the frequency rise too high when P4 is turned to the fully clockwise position, then the value of R3 should be altered. This can be done with the aid of a 47 or 50 kΩ trimmer and, once the correct value has been found, a fixed resistor can be substituted.

The next test is on the sinewave, for which S1 is set to its second position and P3 adjusted to give a sinewave output equal in amplitude to that of the squarewave.

To test the triangular waveform, with S1 in position three, P1 is adjusted for correct output amplitude. A similar procedure is followed for the two sawtooth amplitudes with corresponding switch positions and control adjustments.

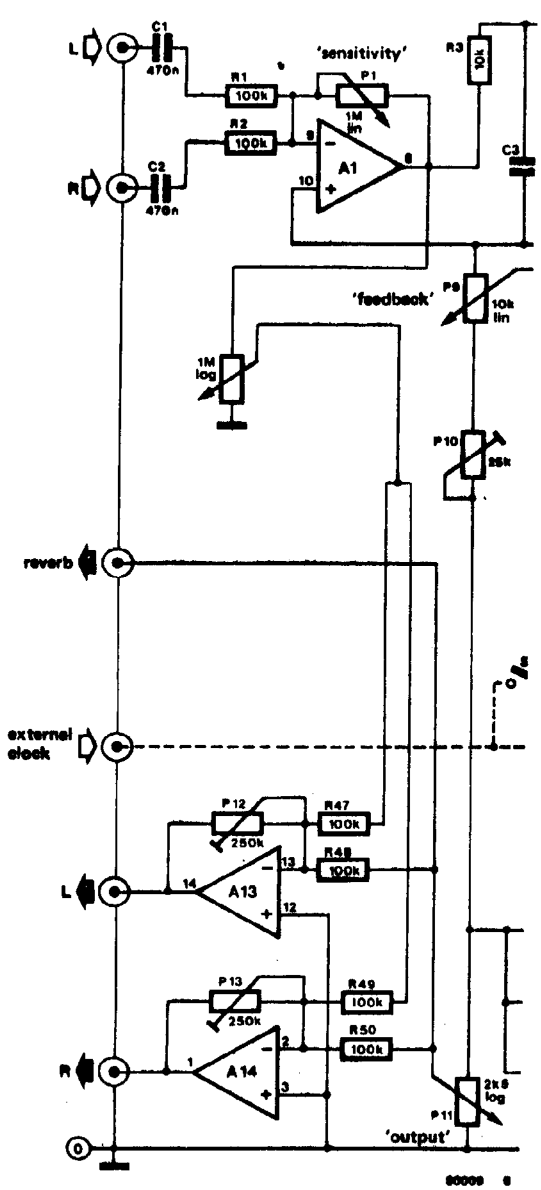
The final adjustment to complete the setting up procedure is the random signal setting – with S1 in the first position and P5 at maximum. To reduce the noise amplitude to a comfortable level, a 1 μF capacitor is used to bridge the emitter of T1 to ground (capacitor positive terminal to emitter). Potentiometer P8 is used to adjust the DC output component to match the reference level established in the preliminary operation. If the meter reading appears to be somewhat erratic, due to the extremely high gain in the noise amplification circuit, the output should be adjusted so that its average reading approaches that of the reference level. The 1 μF bridging capacitor is now removed, and the circuit is ready.

ARU + Sewar

So far, the circuit is just a front end that supplies a sequence of clock pulses at a controlled variable rate. Its effect will only be audible when connected to an electronic reverberation system and associated equipment, such as that described in the Obtober '78 issue of Elektor. Consequently, some adaptations are necessary to the reverb circuit board.

The reverberation unit must use the SAD 1024 integrated circuit. To prepare the unit for a high clock rate, a wide LF band is required, which is made possible

6



7

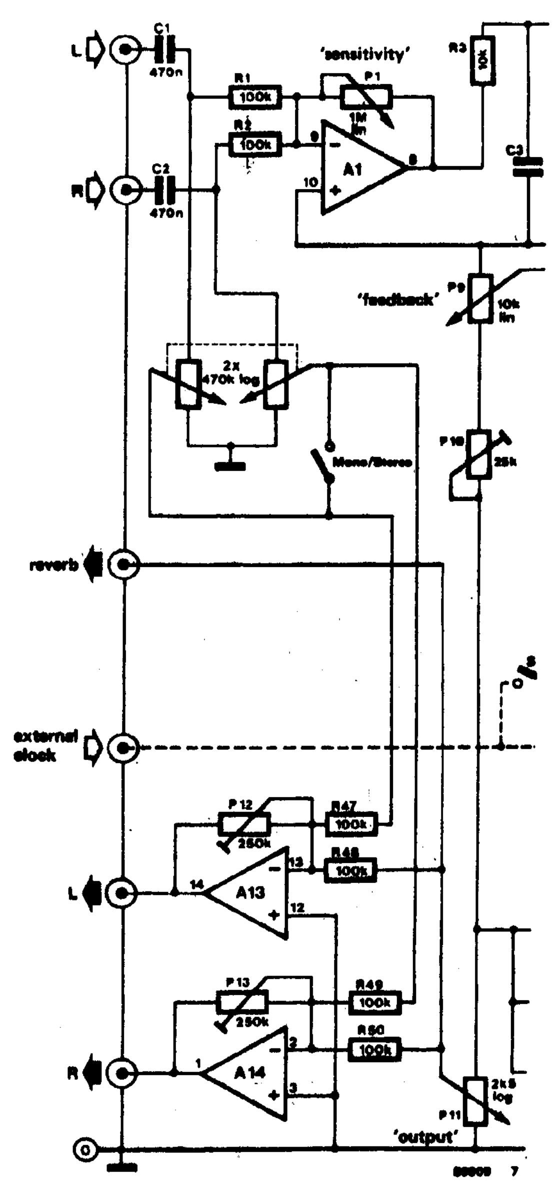


Figure 6. For the phasing effect an additional control is required to blend the delayed to the undelayed signal. A single control is needed for mono operation.

Figure 7. Stereo operation necessitates the addition of a tandem control plus a mono/stereo switch.

by adapting the low-pass filters to a 15 kHz roll-off. The method of doing this has been explained, together with other modifications, in the October '78 article. The cable connecting the clock unit to the reverberation unit must, of course, be screened.

In order to obtain the desired phasing effect, an additional control is required for blending the delayed to the undelayed signal. This modification is suggested in figure 6, for mono, and figure 7 for stereo operation, the latter featuring a mono/stereo switch and a 500 (470) kΩ tandem volume control. The phasing effect is most pronounced when the delayed and undelayed components are of approximately the same intensity.

Selecting and setting the clock rate and its frequency sweep is a fairly simple matter. The first action is to set control P5 to minimum, cutting out all frequency modulation, and to adjust P7 to set the clock rate to the delay required. The required modulation mode is then selected and the modulation depth can then be adjusted by increasing P5. If the sweep gets too wide with respect to the centre frequency, which shows up as an audible whistle, the setting of P7 will have to be altered — normally around halfway. For some effects the equipment may be used without any modulation at all i.e. with P5 set at minimum.

The effects obtainable are described in more detail in the May '79 issue of Elektor, pages 5-18 . . . 5-24. They are recapped in Table 2. Quite unusual reverb/phasing and reverb/vibrato effects can be found by using the variable feedback possibility of the reverberation system. Apart from these, the triangle and sawtooth modulation waveforms permit a wide variety of experimental sound effects, which must be heard to be believed.

References:

Formant (4)
Elektor E30 October 1977, 10-40 etc.
Analogue reverberation unit
Elektor E42 October 1978, 10-44 etc.
Delay lines (2)
Elektor E49 May 1979, 5-18 etc.
Simple function generator
Elektor E33 January 1978, 1-40 etc.

5

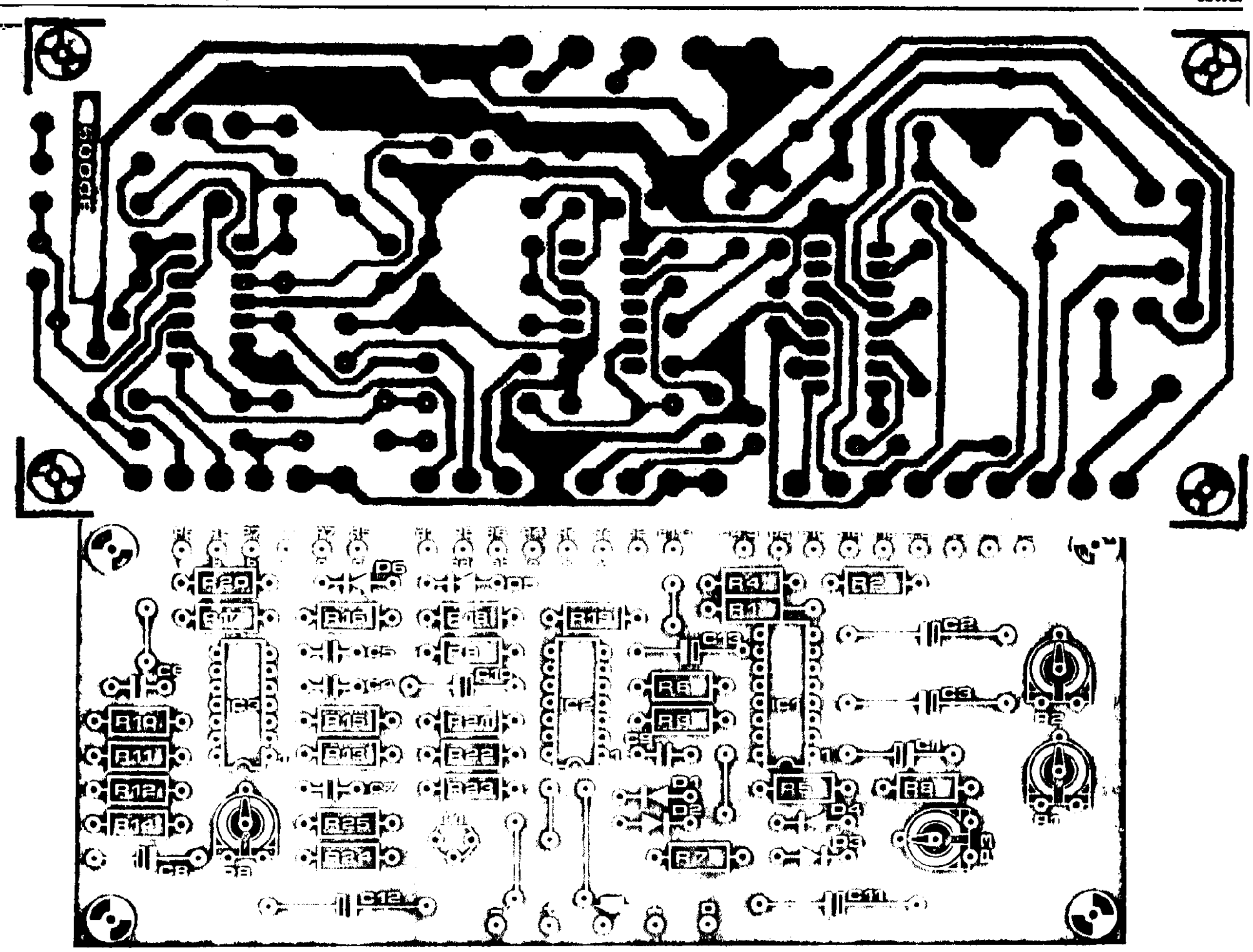


Figure 5. Printed circuit board and component layout for the clock pulse generator.

Table 2

Sound effects	effect	phasing	vibrato	chorus	random phasing	random vibrato
	modulation waveforms	sine or triangle	sine or triangle	random	random	random
	undelayed signal amplitude	maximum	zero	zero	maximum	zero
	delayed signal amplitude	maximum	maximum	maximum	maximum	maximum

Parts list

Resistors:
R1 = 1k2
R2 = 220 Ω
R3 = 39 k
R4,R9,R10,R14 = 1 k
R5,R18 = 2k2
R6,R12 = 3k3
R7,R20,R23 = 10 k
R8 = 4k7
R11,R15 = 100 k
R13 = 1 M
R16 = 330 k
R17,R19 = 68 k
R21 = 3k9
R22 = 5k6
R24,R25 = 12 k
P1,P2 = 4k7 (5 k) preset
P3 = 47 k (50 k) preset
P4 = 10 k lin
P5 = 1 M lin
P6a,P6b = 100 k lin stereo
P7 = 1 k lin
P8 = 10 k preset

Capacitors:
C1,C8 = 10 μ/16 V
C2,C3 = 220 μ/16 V
C4,C5 = 220 n
C6 = 100 n
C7 = 10 n
C9 = 1n5
C10,C13 = 1 μ/10 V
C11,C12 = 100 μ/25 V

Semiconductors:
IC1 = XR 2206
IC2 = XR 2207
A1,A2,A3,A4 = IC3 = TL 074, TL 084
T1 = BC 548B, BC 108B, BC 547B (TUN)
D1,D2,D3,D4, D5,D6 = 1N4148, 1N914 (DUS)

Sundries:
S1a+S1b+S1c+S1d = 6-way 4-gang rotary switch