procedure given in the previous chapter. The offset and octaves per volt adjustments can also be carried out using the procedure given there. During the offset adjustment P4 should be set to minimum and S3 should be set to the 24 dB position. During the octaves/volt adjustment of P8 the Q control, P4, should be set to maximum, as with the 12 dB VCF.

Using the 24 dB VCF

As can be seen from figure 12, the 24 dB VCF is connected between the 12 dB VCF and the VCA, so that the IOS output of the 12 dB VCF goes to the IC input of the 24 dB VCF instead of to the VCA, whilst the VCA receives its input from the IOS output of the 24 dB VCF. The 24 dB VCF also has inputs from the three VCOs.
In addition to the signal connections the 24 dB VCF must also be provided with supply to the VCF module in accordance with the standard practice for Formant. Provision of control voltage inputs from the ADSR envelope shapers will be discussed later.
For satisfactory operation of the 24 dB VCF the correct setting of the input level is important, even more than in the case of the 12 dB VCF. On the one hand, the input level should not be so large that distortion occurs, but on the other hand it should not be so small that the signal-to-noise ratio is degraded. The 24 dB VCF is designed so that the optimum input level is obtained using three VCOs set to maximum output, with one waveform selected per VCO. If more than three VCOs are in use or more than one output waveform is selected from each VCO then the VCO output levels must be reduced. On the other hand, if only one VCO is used then the signal level may be too low. In this case it is best to patch the EOS socket of the VCO to the ES input of the VCF, since this input has approximately three times the sensitivity of the hardwired VCO inputs.
The 24 dB VCF is capable of the same basic functions as the 12 dB VCF; driven by the KOV control voltage it will operate as a tracking filter, whilst the ENV and TM inputs allow dynamic modulation of the harmonic content of the VCF output. Due to the greater slope of the 24 dB VCF the setting of the ENV level control is more critical than with the 12 dB VCF, but if correctly adjusted then subtle nuances in the tonal character of the output signal are possible.
The question arises as to which ADSR envelope shaper should be used to control the 24 dB VCF, since only two are built into the basic Formant system, and control the VCA and 12 dB VCF respectively. Because of the modular construction of Formant it is, of course, perfectly feasible to build a third envelope shaper, which is the most versatile arrangement. The alternatives are to patch one of the other ADSR outputs to the TM input of the 24 dB VCF, or to hardwire the ENV input of the 24 dB VCF to the output of the envelope shaper that controls the 12 dB VCF. This latter arrangement is probably preferable, as it allows the ADSR signal to be fed to one or both VCFs by suitable adjustment of their ENV controls and also allows the possibility of patching the output of the other envelope shaper into the TM input of either VCF.

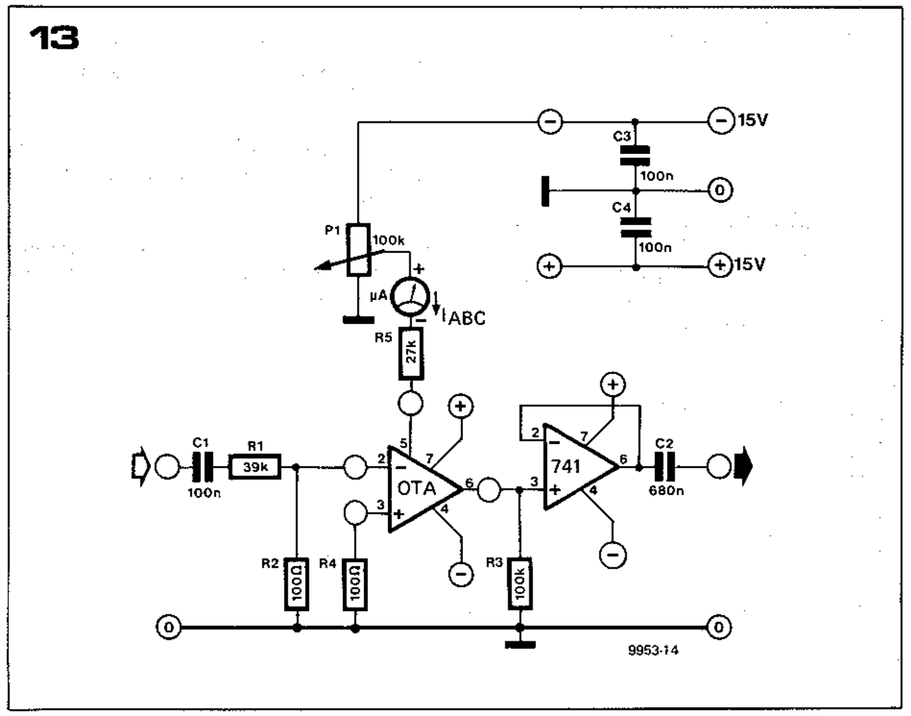


Figure 13. Test circuit for the selection of OTAs.

OTA selection procedure

Although not absolutely essential, it is well worth selecting OTAs with closely matched transconductance characteristics to ensure that the four filter sections track accurately.
A test circuit for the OTAs is given in figure 13. This should be fed with a sinewave signal of about 2 V peak-to-peak (or 0.7 V measured on an AC voltmeter) from a signal generator or from one of the VCOs. The output should be monitored on a 'scope or AC voltmeter. With a control current of 100 μA, measured on the multimeter in series with R5, the output voltage should be between 0.7 V and 1.3 V peak-to-peak. Without changing the input level or control current the OTAs to be tested should be plugged into the circuit one at a time and the output level for each OTA noted. The four OTAs whose output levels are most similar should be used in the VCF.
The circuit can also be used to check the linearity of the transconductance v. control current characteristic of the OTAs, e.g. doubling the control current should double the output of the test circuit and halving the control current should halve the output.

chapter 8

resonance filter module

In addition to an almost limitless variety of non-natural, wholly 'electronic' sounds, the Formant music synthesiser can, of course, be used to imitate the voicing of conventional (mechanical) musical instruments. The filter module described in this article is designed to allow more realistic simulation of natural musical instruments by providing the fixed bandpass resonances which are an important determining factor in the timbre of mechanical tone generators.

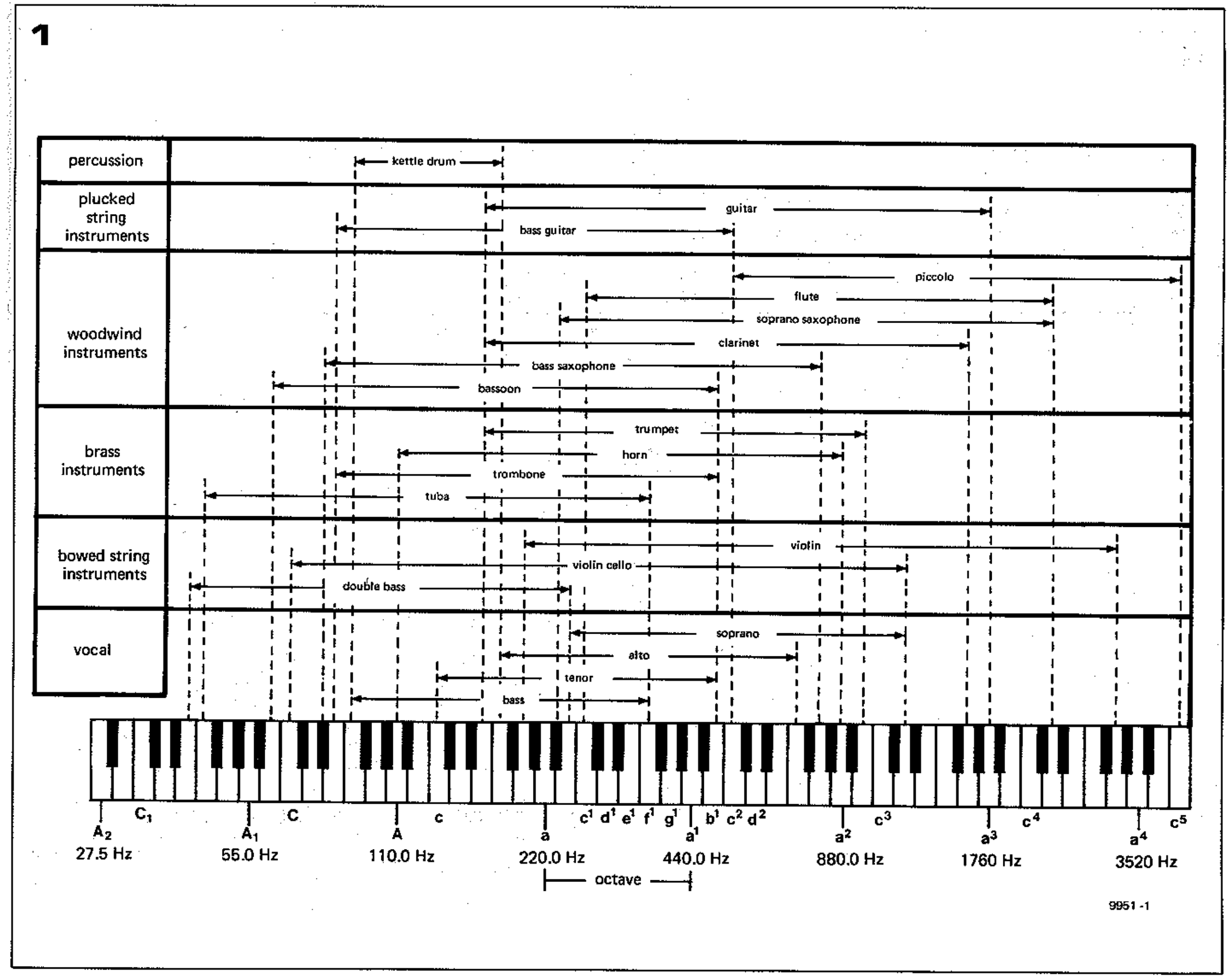
Although music synthesisers are capable of producing the most 'wierd and wonderful' electronic effects, it is a fact that they are frequently employed to imitate the sound of traditional acoustic instruments. Many commercially available synthesisers, for example, are provided with preset facilities for various common instrumental voices, whilst special units such as 'string-synthesisers', which are designed solely to reproduce the sound of a string section, are becoming increasingly popular.

As has already been explained, basic factors influencing the characteristics of a musical note are pitch, dynamic amplitude, and dynamic harmonic content. As the reader will be aware, pitch and dynamic amplitude characteristics are controlled by the VCO (Voltage Controlled Oscillator) and VCA + ADSR (Voltage Controlled Amplifier and Attack - Decay - Sustain - Release) modules in the Formant synthesiser, whilst the VCF (Voltage Controlled Filter) is used to vary the harmonic content of the signal.
However, in the case of mechanical tone generators, for example brass and woodwind instruments, an additional consideration is the existence of resonant areas in the instrument which possess free vibration periods of their own. These resonances, which are known as *formants* (whence the name for the Elektor music synthesiser!) are determined by the shape and mechanical construction of the particular instrument (the wooden back and belly of a violin, the pipes of an organ etc.). Unlike the variable pitch of, say, a violin string, they tend to reinforce the same harmonics, whatever the pitch of the note being played. The nature of the formants in an instrument is in fact one of the factors which govern its quality.
It should thus be apparent that, in order to realistically simulate the tonal characteristics of traditional instruments, one must be able to tailor the static harmonic content of the note accordingly. What is required is a number of resonant filters with independently variable centre frequency, gain and Q-factor. These features are present already in the state variable VCF of the Formant; however, that is only one filter, and more to the point, in this particular application there is no need for the filters to be voltage-controlled, since the filter parameters will be preset to suit whatever musical instrument is being imitated. This explains the reason for the separate manually-controlled resonance filter module described in this article.

Figure 1. The fundamental frequency range of a number of traditional musical instruments, with reference to that of a grand piano. *(From: 'Elektronik Taschenbuch, Band 1', Ferd. Dümmlers Verlag, Bonn; with kind permission from the publishers.)*

The uses of resonance filters

The effect of resonance filters can best be heard on 'bright' sharp VCO waveforms which have a high proportion of fairly intense upper harmonics. The effect on vocal sounds can be illustrated by taking a suitable signal with a frequency of around 200 Hz, setting the Q of the filter to a mid-value, and varying the centre frequency from minimum to maximum. At first 'dark' sounding tones, largely devoid of higher harmonics will be obtained; as the centre frequency is increased however, one by one the various vowel sounds can be distinguished until, at high centre frequencies, reedy flute-like sounds are produced. The higher the Q of the filter, the more pronounced the above effects – and vice versa.
All bandpass resonances of musical importance lie between roughly 100 and 2000 Hz. Table 1 lists the main fixed resonances of a number of common musical instruments and also indicates which VCO waveform is best suited to imitate the instrument in question. This table is, of course, merely intended as a rough guide, in the final instance the decision should rest with one's own ears. Unless otherwise indicated, the Q-control should be set to the mid-position. As a further aid, figure 1 shows the fundamental frequency ranges of vari-



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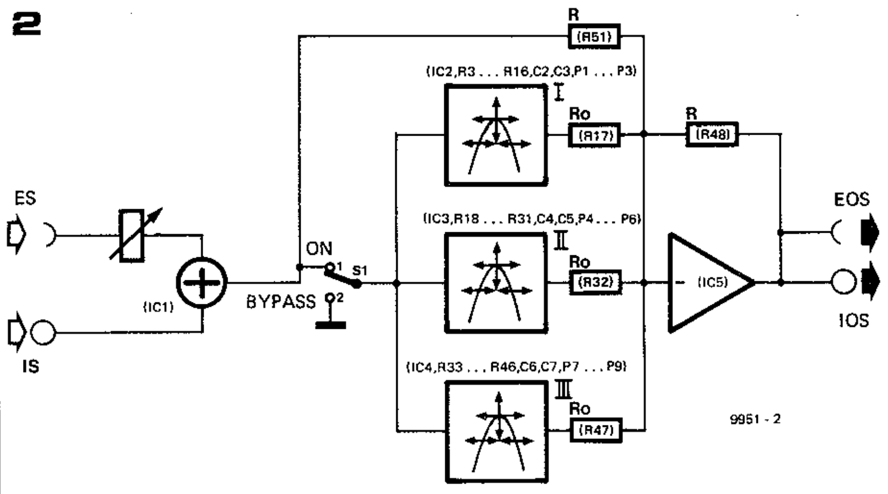

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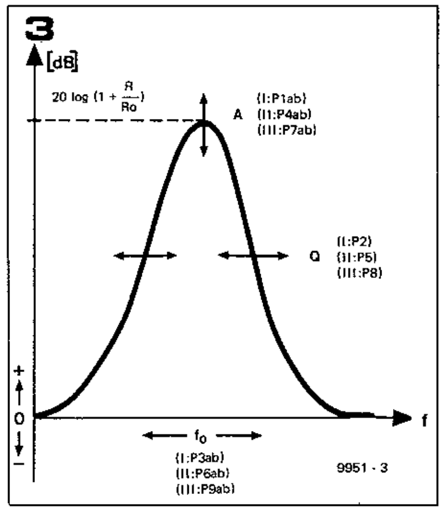

Table.

instrument	main resonance at	VCO signal
flute	approx. 800 Hz	fairly asym. squarewave
clarinet	1 . . . 2 kHz*	sym. squarewave
oboe	1300 . . 1700 Hz*	heavily asym. squarewave (pulse)
bassoon	approx. 440 Hz*	heavily asym. squarewave (pulse)
trumpet	approx. 1500 Hz	'spaced' sawtooth
bugle	approx. 1000 Hz*	sawtooth
trombone	approx. 600 Hz	'spaced' sawtooth
French horn	approx. 400 Hz*	sawtooth
tuba	approx. 250 Hz	sawtooth
violin	approx. 4000 Hz**	'spaced' sawtooth, sawtooth or heavily asym. squarewave (pulse)
cello	approx. 200 Hz**	
double bass	approx. 100 Hz**	

NB:
* with increased Q
** if possible, use several resonant filters (or a comb filter)

Figure 2. Block diagram of the resonant filter module. As can be seen, it possesses three independently variable filter sections.

Figure 3. The frequency response of one of the three filter sections contained in the resonant filter module. The figure illustrates how the filter parameters can be independently varied by means of the control potentiometers.

Figure 4. Detailed circuit diagram of the filter module.

Figure 5. Track pattern and component layout of the filter module p.c.b. (EPS 9951-1).

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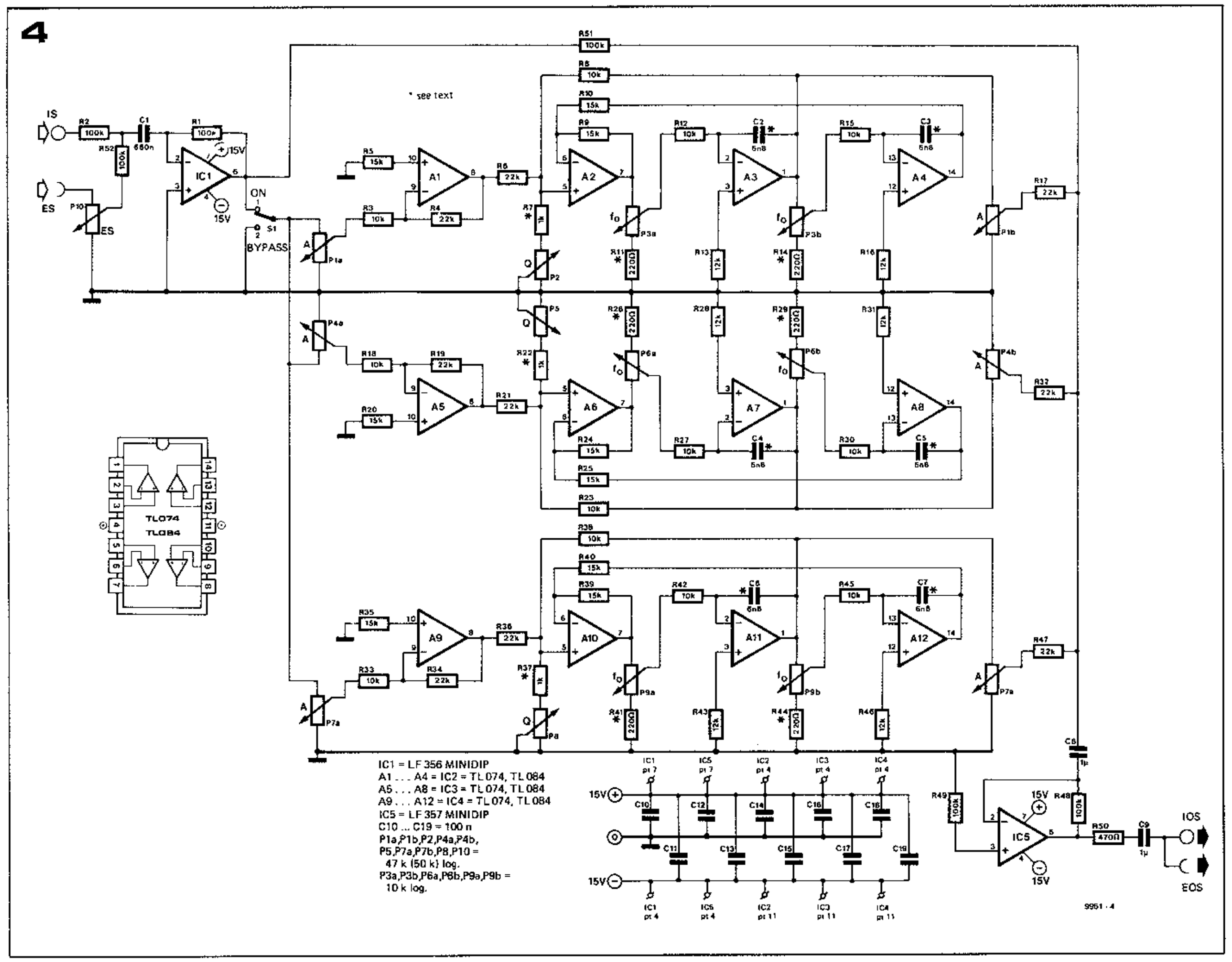

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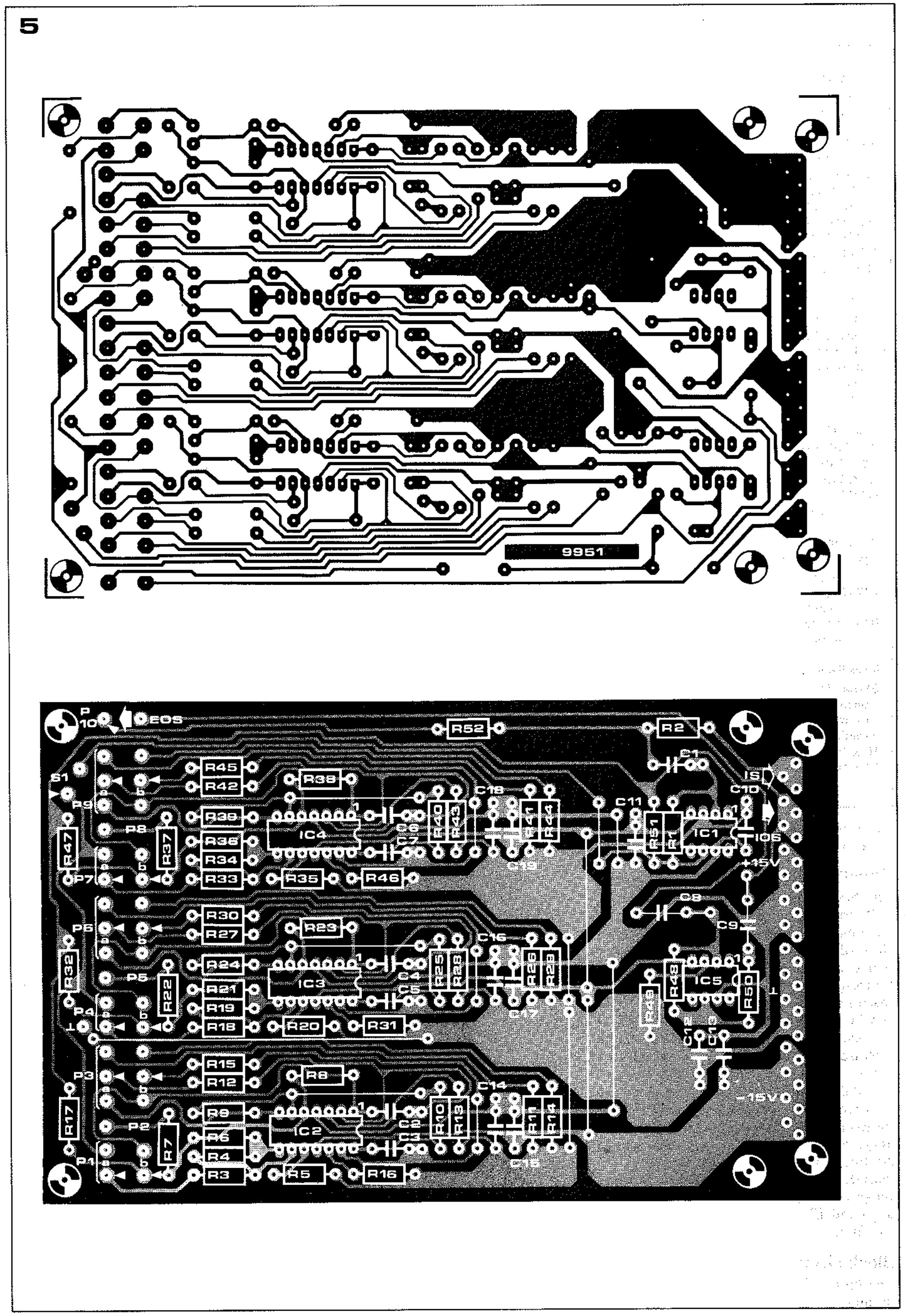

Parts list to figure 4 and 5.

Resistors:
R1,R2,R48,R49,R51,R52 = 100 k
R3,R8,R12,R15,R18,R23,
R27,R30,R38,R42,R45 = 10 k
R4,R6,R17,R19,R21,
R32,R34,R36,R47 = 22 k
R5,R9,R10,R20,R24,
R25,R35,R39,R40 = 15 k
R7,R22,R37 = 1 k (see text)
R11,R14,R26,R29,
R41,R44 = 220 Ω (see text)
R13,R16,R28,R31,R43,R46 = 12 k
R50 = 470 Ω

Potentiometers:
P1,P4,P7 = 47 k (50 k) logarithmic, stereo, dia 4 mm
P2,P5,P8,P10 = 47 k (50 k) logarithmic; dia 4 mm
P3,P6,P9 = 10 k logarithmic, stereo; dia 4 mm

Capacitors (all Siemens MKM, MKH or other polycarbonate/polyester type)
C1 = 680 n
C2,C3,C4,C5,C6,C7 = 6n8 (see text)
C8,C9 = 1 μ
C10 . . . C19 = 100 n

Semiconductors:
IC1 = LF 356 (National Semiconductors), Mini DIP
IC2,IC3,IC4 = TL 084, TL 074 (Texas Instruments)
IC5 = LF 357 (National Semiconductors) Mini DIP

Miscellaneous:
31-way DIN 41617 edge connector or terminal pins
S1 = miniature SPDT
2 miniature sockets 3.5 mm dia.
10 x 10 mm collet knobs (with pointer)
1 front panel

ous traditional instruments, with reference to a piano keyboard.

Circuit

The basic requirements of the filter circuit are, independently variable centre frequency, Q and gain. Since the function of the filter is essentially to enhance a particular band of frequencies (corresponding to the formants of the instrument in question), the circuit is of the boost-only type, i.e. provides selective gain. Without the need to provide a selective cut (below the 0 dB line) the circuit design is considerably simplified.
A total of three resonant filters forms an acceptable compromise between the number of settings required for reasonably realistic imitation and the constraints of space and economy. Of course, it is quite possible to double the range of control facilities by connecting a second filter module in cascade with the first.

Block diagram

The block diagram of the resonant filter module is shown in figure 2. The figures

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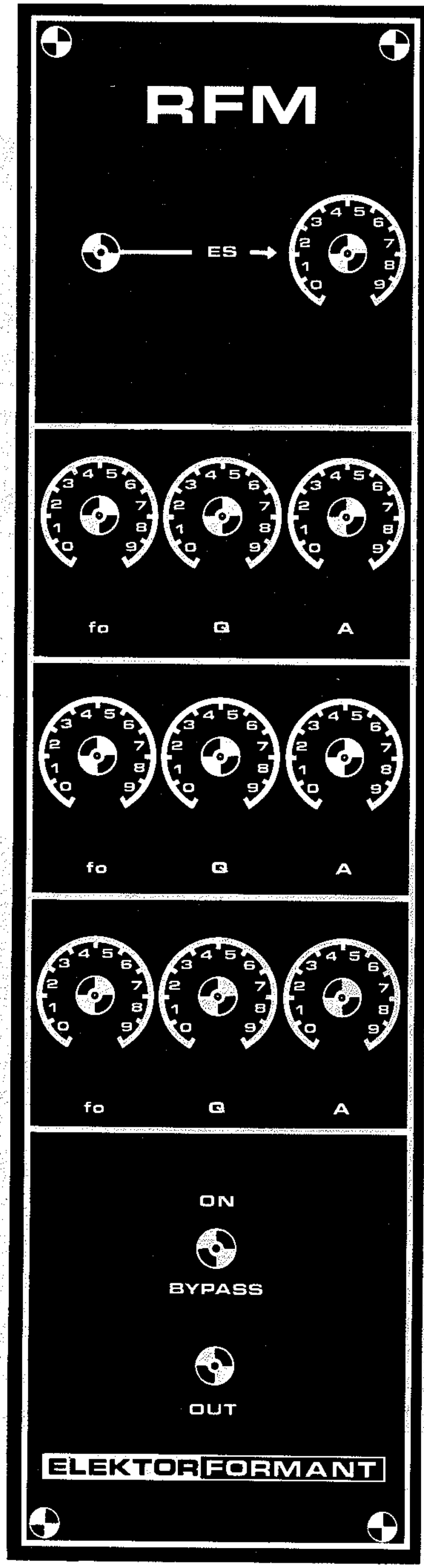

in brackets indicate which components in the final circuit are associated with the different sections of the circuit. Signals can be fed in via the panel-mounted socket (ES) or via the hard-wired input (IS). A portion of the signal is fed direct to the output summing amplifier via R (R51 in the complete circuit) and the input signal is also fed to three bandpass filters whose gain, centre-frequency and Q can all be varied. The outputs of these filters are also summed in IC5 via resistors R_O. The output of the filter module will thus consist of a portion of the original input signal plus signals boosted around the centre frequencies of the three filter stages. Two outputs are provided from the filter module, an internal hardwired output (IOS) and an output to a front panel socket (EOS). A bypass switch is provided, which allows the three filter sections to be switched out, in which case only the original signal appears at the output, and the gain is frequency independent, being unity.
The amount of boost that can be provided by a filter section relative to the gain obtained in the 'bypass' condition is determined by the gain of the filter sections and the ratio R/R_O. If it is assumed that the filter gain can be varied between zero and one then the maximum amount of boost (in dB) is $20 \log (1 + \frac{R}{R_O})$.
The frequency response of a filter section is shown in figure 3. The figures in parentheses indicate which controls in the complete circuit vary the different parameters of the filter.
The complete circuit of the filter module is shown in figure 4. IC1 sums and inverts the two input signals, whilst the three filter sections are of the state-variable type. The resonant gain of the filters is set by means of P1, P4 and P7 respectively. One gang of the pots is connected at the input, the other at the output of the filter. This has the effect of improving the dynamic range, since it means reduced noise and less chance of overloading. Finally, there is the inverting summing amplifier round IC5, which also cancels the phase shift introduced by IC1.

With the values for R and R_O given in the circuit diagram, the maximum gain of the filter is approx. +15 dB. The quality factor, Q, can be varied by P2 (P5, P8) between roughly 0.8 and 5. The centre frequency can be varied between approx. 50 and 2300 Hz, which is more than sufficient for normal use. The frequency range can, however, be modified by altering the value of a number of components; the necessary changes are detailed in the appendix.
Maximum Q is obtained for the minimum resistance of the Q-potentiometer. The maximum Q can therefore be increased by reducing the value of R7 (R22, R37); in this way a Q of between 20 and 30 can easily be obtained. A high Q is useful when processing waveforms such as squarewaves, which have

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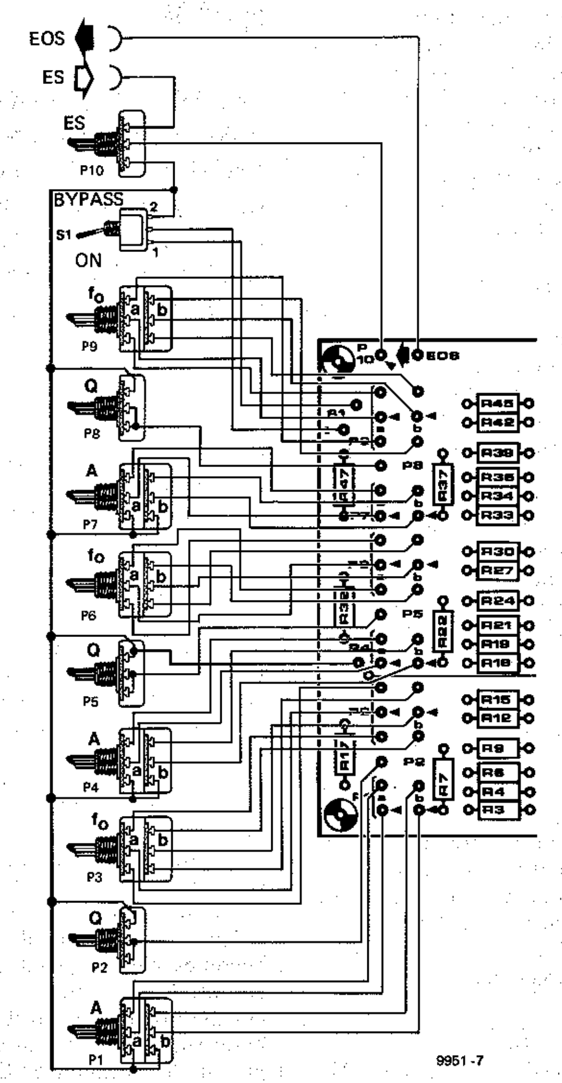


Figure 6. Because of the large number of controls, the front-panel for the resonant filter module is clearly different from the other Formant front-panels (EPS 9951-2).

Figure 7. Wiring diagram for the components mounted on the front-panel.

very steep edges. These tend to set the filters 'ringing' at their resonant frequencies, and produce percussive effects. For R7 (R22, R37) = 470 Ω, a Q of 11.3 is obtained; R7 = 330 Ω gives a Q of 15.8, and R7 = 220 Ω a Q of 23.4. The higher the Q, the more pronounced the percussive effect.

Construction

The printed circuit board for the resonant filter module is shown in figure 5.
As far as the selection of components is concerned, the usual criteria apply. The only difference is that in view of the large number of front-panel controls (10 potentiometers) it is strongly recommended that miniature components (miniature pots with 4 mm diameter spindles) be used. In this way the controls can be arranged in functional groups of three to a row.
The front panel for the filter module is shown in figure 6, and the details of the wiring for the front-panel controls are illustrated in figure 7. In contrast to the

other Formant modules, the resonant filter module requires no calibration or adjustment procedure. The operation of the circuit can be checked by feeding in a white noise input from the noise module. Varying the three filter parameters should produce clearly audible changes in the resulting sound. It will also be apparent that rapid variation of the Q- and f_0 controls produces effects similar to phasing, thus the filter module can be used to provide manual phasing.
The scale on each of the f_0 potentiometers on the front panel is calibrated with five nominal frequencies. The three middle settings in particular should be viewed as rough guidelines, since the resistance curve of logarithmic potentiometers can exhibit fairly wide tolerances.
The filter module should be placed between the COM-module and the power amp. However, if one wishes to use the headphone output on the COM-module, the resonant filter module can be connected directly before the latter.

Appendix

With the component values given in the circuit diagram, the centre frequency of the filters can be varied between roughly 50 and 2300 Hz. To calculate the correct values for higher frequencies than this, the procedure is as follows:
Firstly, the desired maximum frequency of f_0 can be used to calculate the value of C2 = C3 = C4 = C5 = C6 = C7 = C from the following equation:

$$C = \frac{16}{f_0 \max}$$

where C is in nanofarads and f_0 in kHz.
Secondly the value of resistor R (see figure 2) can be determined on the basis of the desired minimum centre frequency f_0 min:

$$R = \frac{16}{C \cdot f_0 \min}$$

where C is in nanofarads, R is in kΩ, and f_0 in kHz
The value of R_0 = R11 = R14 = R26 = R29 = R41 = R44 can be calculated from:

$$R_0 = \frac{10}{R - 2}$$

where R and R_0 are in kΩ. These equations can be used to check the values of figure 4.

chapter 9

ADSR

The ADSR (Attack-Decay-Sustain-Release) shaper is used to control the VCF and VCA modules and thereby determine the dynamic harmonic structure and dynamic amplitude characteristic of the VCO signals.

It is often not realised, even by musicians, how much the character of an instrument is determined by the dynamic amplitude and harmonic behaviour, rather than by the steady-state harmonic content of the instrument. If the attack and decay periods of a note are artificially modified, then the whole character of the sound is altered. An interesting and amusing experiment is to record the sounds of several musical instruments, but to remove the attack and decay periods by bringing up the recording level after the note starts and fading it down before the note ends. Then ask some musical friends to identify the instruments. They will no doubt be amazed how characterless the sound of an instrument becomes when robbed of its particular amplitude envelope.
On the other hand, starting with a single basic waveform such as the triangle output of the Formant VCO, a whole range of instrument sounds can be produced simply by varying the amplitude envelope, ranging from 'soft' sounds such as flute and some organ voices, to 'hard', percussive sounds such as piano and xylophone.
Envelope control of the harmonic content using the VCF allows even greater variation in the character of the sound.

Types of envelope curves

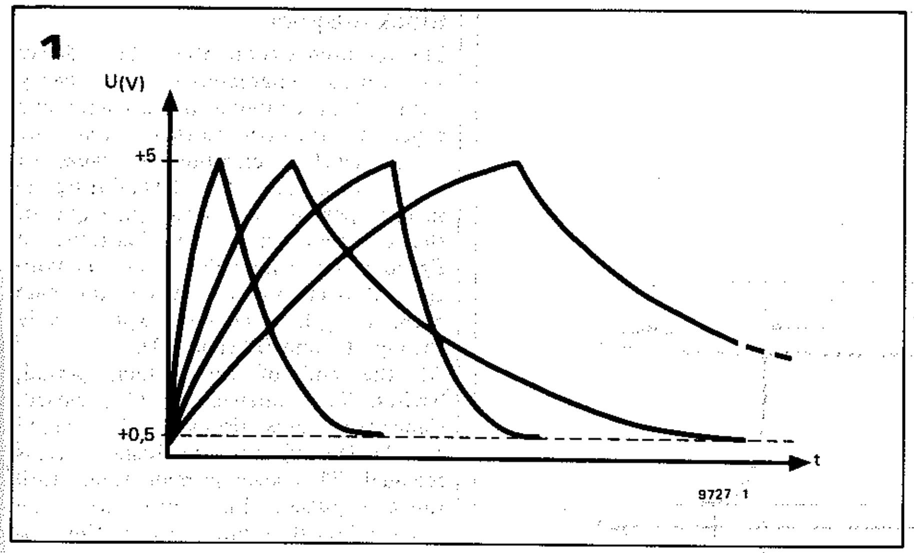
The envelope shaper of the synthesiser must be able to simulate the envelope contour of conventional musical instruments when the synthesiser is used in an imitative capacity, and also to produce envelopes that are purely synthetic in character (i.e. not found in sounds produced by normal acoustic methods). Fortunately, there are relatively few types of envelope contour that are musically important, and these are all fairly easy to generate electronically.

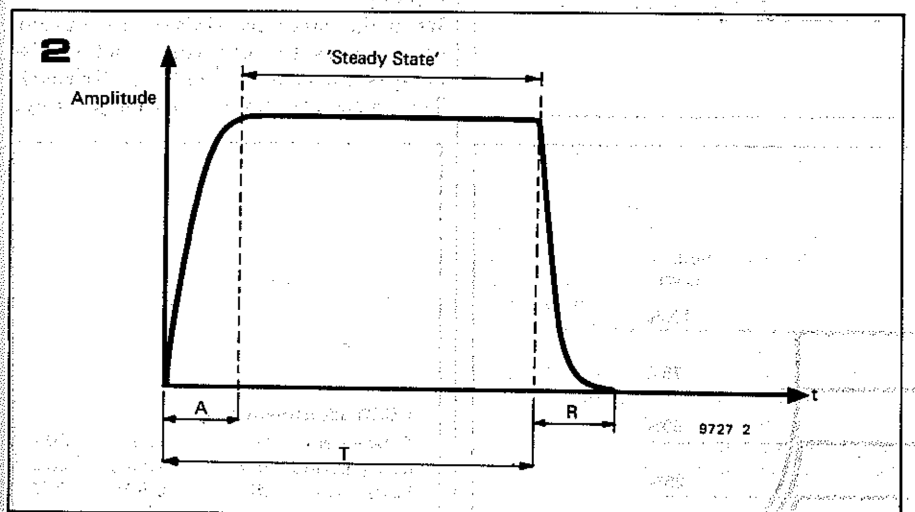
1. Attack/decay contour

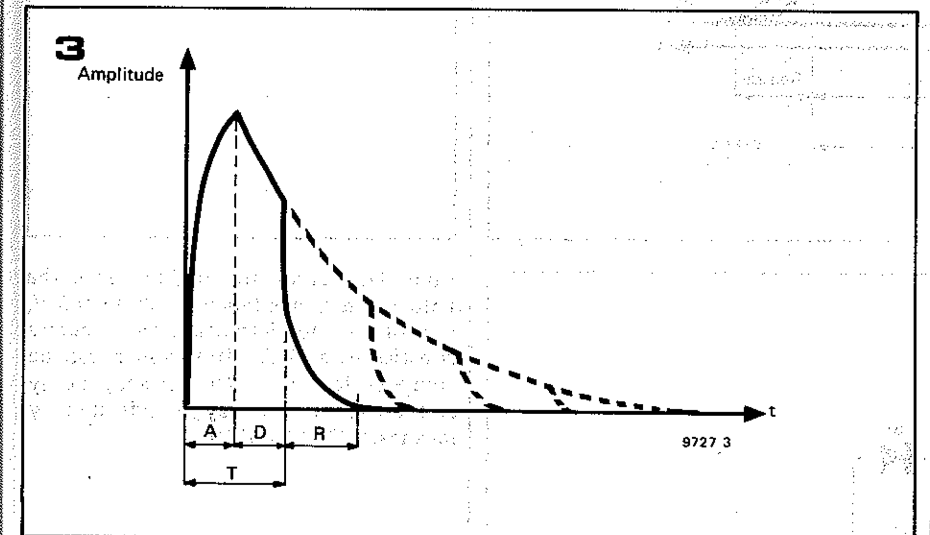
The simplest type of envelope curve is that consisting only of attack and decay periods. The envelope contour rises to a peak when the note is played, and begins to decay immediately the peak is passed (see figure 1). By varying the attack and decay times a wide variety of sounds can be produced.
For example, if a rapid attack and slow decay is applied to the VCA control, then a percussive sound like a piano results. Applied to the VCF in the low-pass mode, the same envelope contour can produce very bright, metallic sounds, depending on the input waveform.
If the attack period is made long and the decay period short, then applying this to the VCA will produce completely synthetic 'fantasy' sounds similar to those obtained by playing a recording backwards. Applying this type of envelope contour to the VCF can produce sounds similar to those of a brass instrument played staccato.
However, the main use of this type of envelope curve is for the production of percussive sounds such as xylophone, marimba, glockenspiel, bells and gongs, cymbals, and struck or plucked strings such as guitar, banjo, harp, other string instruments played pizzicato, harpsichord, and of course, piano.

2. Attack-sustain-release contour

The attack/decay characteristic previously described is typical of instruments where the sound is initiated by a short pulse of energy (e.g. by striking or plucking a string), after which the sound dies away since there is no further excitation to sustain it. The envelope contour shown in figure 2 is typical of instruments in which a note is sounded and sustained, such as a pipe organ, woodwind instruments, and bowed string instruments. In a pipe organ the note builds up fairly rapidly after a key is depressed as standing wave modes are established in the pipe, and the note is sustained by virtue of the fact that air is continuously blown into the pipe. When the supply of air stops on releasing the key the note terminates more or less rapidly.
The same basic contour applies to woodwind instruments and to string instruments played with a bow, since the note is here again sustained by blowing or bowing. However, with such instruments much greater expression can be obtained by modulation of the







steady-state level, since this is determined by the player, and not by a mechanical blower as is the case with a pipe organ.
With a synthesiser, a degree of expression can be obtained by modulating the VCA using the low-frequency oscillators or noise source.

3. Attack-decay-release contour

A variation on the attack-decay contour is shown in figure 3. Here the slow decay is allowed to continue for only a certain time, and the note is then terminated by a more rapid release. The most common example of this type of contour is provided by our old friend, the piano. When a note is sounded and the key remains depressed, then the damper is held off the string and the note decays over a period of a few seconds. If, however, the key is released after playing a note, the felt damper contacts the string and the note terminates after about 500 ms.

Figure 1. The attack-decay envelope contour is the simplest contour found in music.

Figure 2. The attack-sustain-release contour is used to simulate instruments where the note can be sustained at a constant level, such as organ, woodwind, and bowed string instruments.

Figure 3. Instruments such as the piano can be simulated using the attack-decay-release contour. As long as the key remains depressed the decay path is followed, but once the key is released the note is ended more abruptly, following the release contour.

4. Attack-decay-sustain-release contour

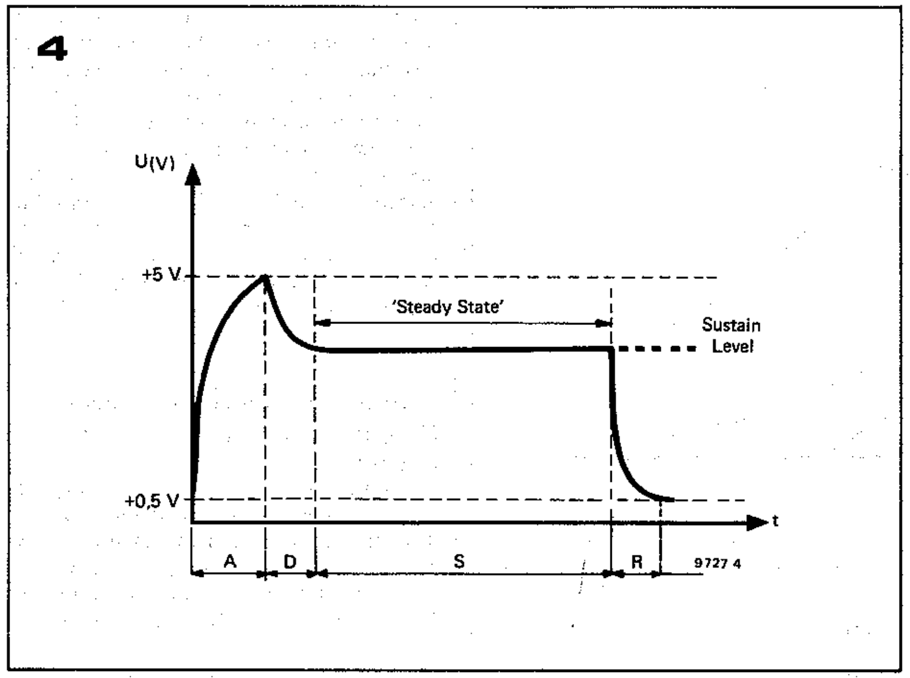
Most of the examples given so far relate to envelope control of the VCA, since the amplitude contour of a sound is somewhat easier to visualise than its dynamic tone colour behaviour. However, the most complex envelope contour, shown in figure 4, is a good illustration of envelope control of the VCF.
Many brass instruments, such as the trumpet, are characterised by a rapid build-up of harmonics during the attack period of the note, which gives the instrument a very strident sound. Once the note is established, however, the harmonics die away somewhat, and the tone is much more mellow during the steady state period. Finally, during the release period at the end of the note, the note dies away fairly rapidly.
This type of characteristic can be obtained by using the VCF in the low-pass mode and controlling it with an envelope contour similar to that shown in figure 4. As the control voltage rises during the attack period, so the turnover frequency of the VCF increases, passing more harmonics. During the decay period the VCF turnover frequency falls until the steady-state value is reached, and finally, during the release period the VCF turnover frequency drops very rapidly.

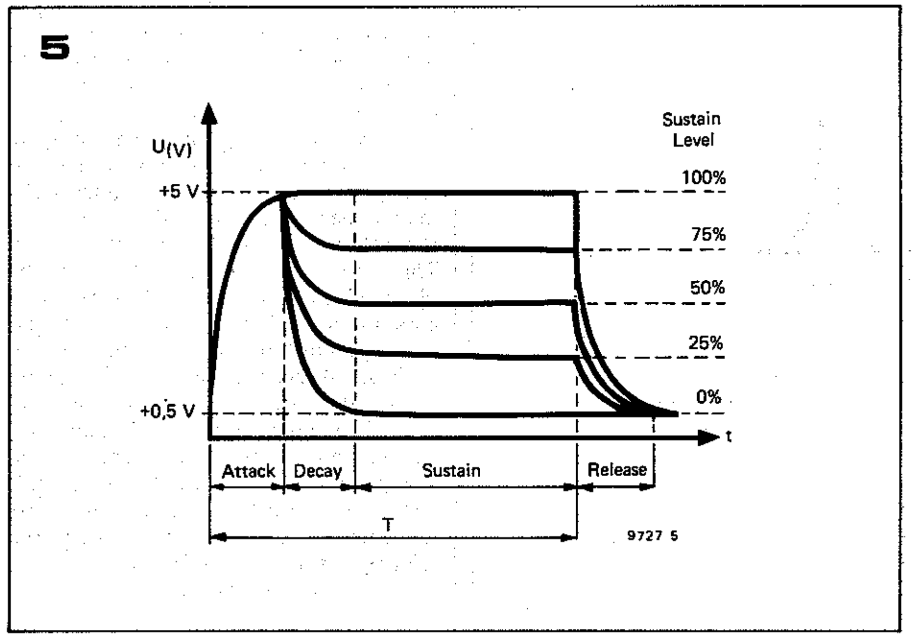
Envelope shaper requirements

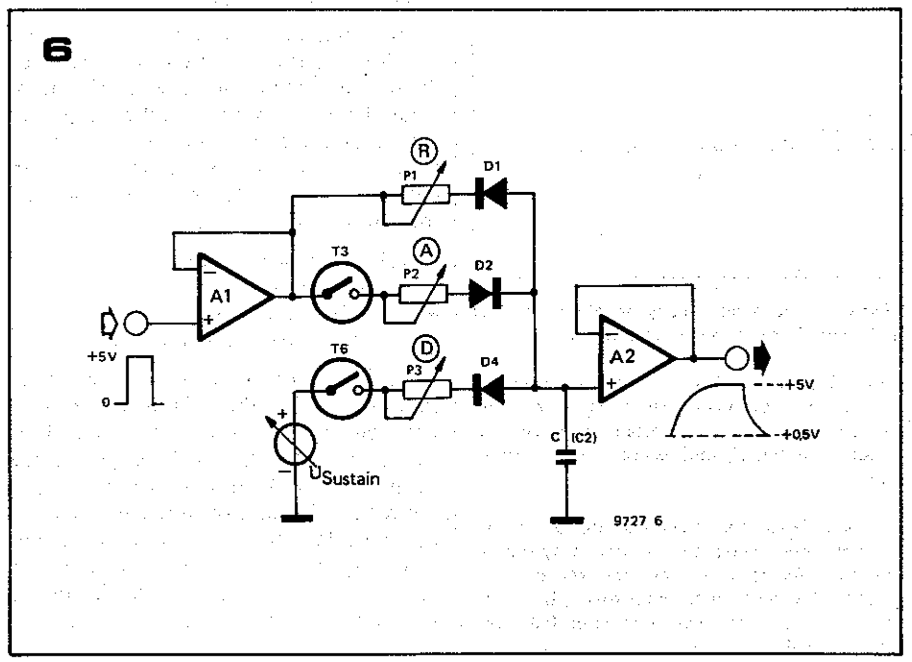
It is apparent from figure 5 that the envelope contours shown in figures 1 to 3 are merely special cases of the more general attack-decay-sustain-release contour illustrated in figure 4. Any of the four contours can be generated by an envelope shaper having the following four functions:

- variable attack time (A)
- variable decay time (D)
- variable sustain level (S)
- variable release time (R)

These four parameters can be preset manually using the ADSR controls of the envelope shaper. The envelope shaper is controlled by the gate pulse output of the keyboard. When a key is depressed the gate output goes high and this initiates the attack-decay sequence. The output of the envelope shaper then remains at the sustain level until the key is released, when the release period begins.







Block diagram

The required exponential attack, decay and release characteristics are easily obtained by charging and discharging a capacitor through resistors, and the sustain level by clamping the capacitor voltage to a preset D.C. level during the sustain period. The basic principle of the envelope shaper is illustrated in figure 6. The gate pulse is fed to a voltage follower A1, and when the gate pulse is high C charges exponentially through P2 and D2 (and T3).

At the end of the Attack period, 'switch' T3 is opened and T6 is closed. Capacitor C now discharges through D4 and P3 (Decay), until the Sustain level is reached. This level is maintained until the gate pulse finishes, either when the key is released or when a preset time has elapsed.

When the gate pulse finishes, the output of A1 goes to zero volts, and C discharges through D1 and P1 (Release). The capacitor cannot discharge fully, since D1 ceases to conduct once the voltage on C has fallen to about 0.5 V, but this is not important as it merely constitutes a D.C. offset which can be compensated for. The attack, decay and release times may be adjusted by means of P2, P3 and P1.

ADSR adjustment ranges:

Attack period	(A)	10 ms . . . 20 s
Decay period	(D)	10 ms . . . 20 s
Sustain level	(S)	0.5 V . . . 5 V
Release period	(R)	10 ms . . . 20 s

Complete circuit

The complete circuit, which is shown in figure 7, is, of course, more complicated.

The envelope shaper has two modes of operation, ADSR and AD, which are selected by means of S1. With S1 in position 'b' (ADSR) the circuit operates as follows:

When a key is depressed the gate pulse output goes to +5 V. IC1 has a gain slightly greater than unity, so about +6 V appears at its output.

The leading edge of the gate pulse also triggers monostable T1/T2, which produces a short pulse to set flip-flop T4/T5 (T5 turned on and T4 turned off). The collector voltage of T4 thus rises, turning on T3 and allowing C2 to charge from the output of IC1 through T3, P2, R17 and D2. This is the attack period.

The voltage on C2 is fed to voltage-follower buffer IC4, which is connected to the outputs EOS and ENV and also to the non-inverting input of IC3. This IC functions as a comparator, with its inverting input held at about 4.7 V by R24 and R25. When the voltage on C2, and hence at the output of IC4, exceeds this value, the output of IC3 swings positive, resetting flip-flop T4/T5, turning off T3 and terminating the attack period. T6 is turned on, initiating the decay period when C2 discharges through D4, R21, P3 and T6 into the output of IC2 until the sustain level, set at the output of voltage follower IC2 by P4, is reached.

The output of the envelope shaper then remains at the sustain level until the key is released, when the output of IC1 goes to zero volts and C2 discharges through D1, R13 and P1 (release period).

Diode D7 protects C2 in the event of the output of IC1 going negative for any reason, when the voltage across C2 is clamped to a maximum of −0.7 V.

A LED indicator constructed around IC5 allows visual monitoring of the envelope contour. The brightness of the LED follows the envelope voltage.

Two outputs are provided from the envelope shaper; an external output to a front panel socket (EOS), and an internally wired output (ENV).

The full ADSR envelope contour is, of course, produced only if the key is depressed for a period longer than the attack plus decay time, and if the sustain level is greater than 0%. If the key is released before the sustain level is reached then the release period is initiated prematurely, and either AR or ADR curves may be produced. If the sustain level is 0% then only AD or ADR curves may be produced, depending on when the key is released. If the sustain level is 100% then, of course, only AR or ASR curves may be produced, depending on when the key is released, since the decay period is inhibited.

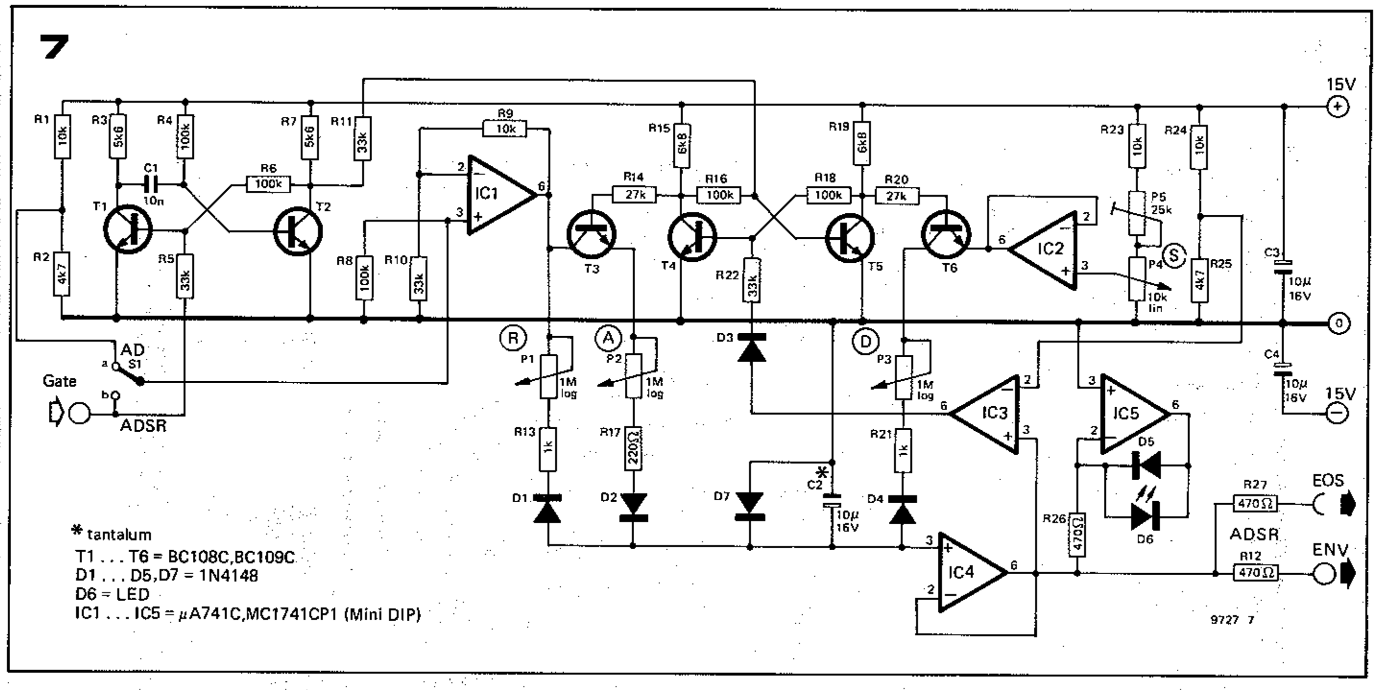


Figure 4. The attack-decay-sustain-release contour is the most complex envelope shape provided by the Formant envelope shaper. When applied to the VCF it is useful for imitating brass instruments, where the harmonic content of the note rises initially to a large value, then reduces to a lower level during the steady-state part of the note.

Figure 5. By varying the sustain level the envelope contour can be changed from an AD contour at 0% sustain, through various ADSR contours to an ASR contour at 100% sustain. T is the time for which the key remains depressed.

Figure 6. This simplified diagram illustrates the basic principle of the envelope shaper. C charges through D2 and P2 during the attack period. It then discharges through D4 and P3 to the (adjustable) sustain level; finally, it discharges through D1 and P1 during the release period. P1, P2 and P3 can be used to vary the release, attack and decay times.

Figure 7. Complete circuit of the Formant envelope shaper.

Triggered AD mode

It is sometimes useful to be able to produce AD envelope contours that are unaffected be releasing the key, that is to say, once the key is depressed, a fixed attack-decay sequence is initiated, which is completed whether the key is released or not. This triggered AD contour is obtained by setting S1 to position 'a' and selecting 0% sustain level. The input of IC1 is now connected to the junction of R1 and R2, so its output is permanently at about +6 V, irrespective of the gate input.

When a key is depressed, the gate signal triggers the monostable, setting the flip-flop and turning on T3. At the end of the attack period, comparator IC3 resets the flip-flop, turning on T6 and initiating the decay period. C2 will now discharge through D4, R21, P3 and T6 to the 0% level (sustain is set at 0%).

Even if the key is released before this sequence is complete, the release period is inhibited since the output of IC1 is permanently at +6 V, so C2 cannot discharge through D1, R13 and P1.

Construction

There are no special requirements with regard to resistor tolerances in the envelope shaper circuit, and ordinary, good-quality 5% carbon film components are quite adequate; C2 should be a tantalum electrolytic capacitor for low leakage, and C1 the usual

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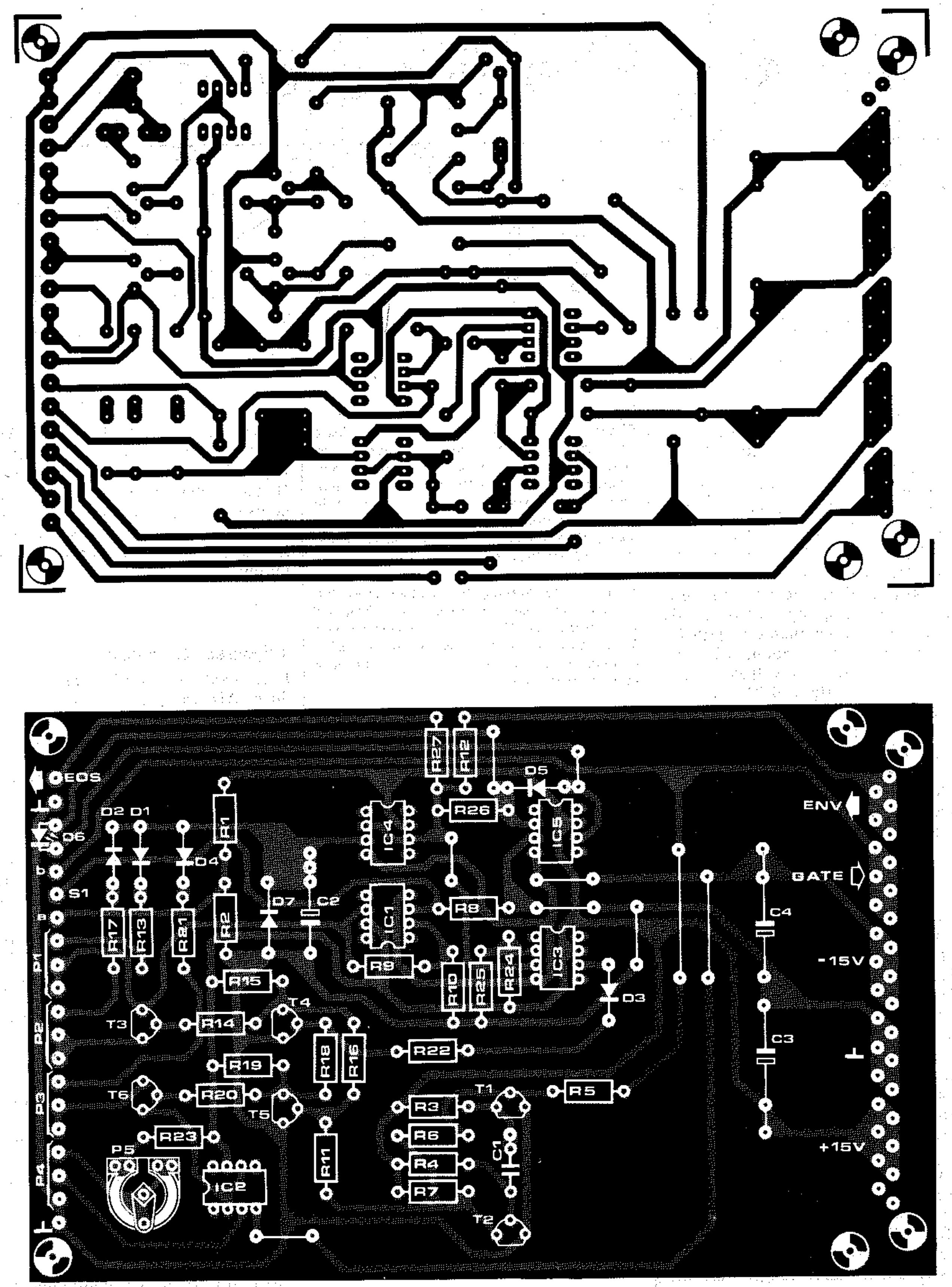

Parts list for figures 7 and 8

Resistors:
R1,R9,R23 = 10 k
R2,R25 = 4k7
R3,R7 = 5k6
R4,R6,R8,R16,R18 = 100 k
R5,R10,R11,R22 = 33 k
R12,R26,R27 = 470 Ω
R13,R21 = 1 k
R14,R20 = 27 k
R15,R19 = 6k8
R17 = 220 Ω

Potentiometers:
P1,P2,P3 = 1 M log.
P4 = 10 k lin.
P5 = 25 k preset

Semiconductors:
T1 . . . T6 = BC108C, BC109C or equivalent
D1 . . . D5,D7 = 1N4148, 1N914
D6 = LED (TIL 209 or similar)
IC1 . . . IC5 = μA 741C, MC1741 CP1 (MINI DIP)

Capacitors:
C1 = 10 n
C2 = 10 μ/16 V tantalum
C3,C4 = 10 μ/16 V

Miscellaneous:
31-way Euro connector (DIN 41617)
1 x 3.5 mm jack socket
4 x 13 . . . 15 mm collet knobs with pointer

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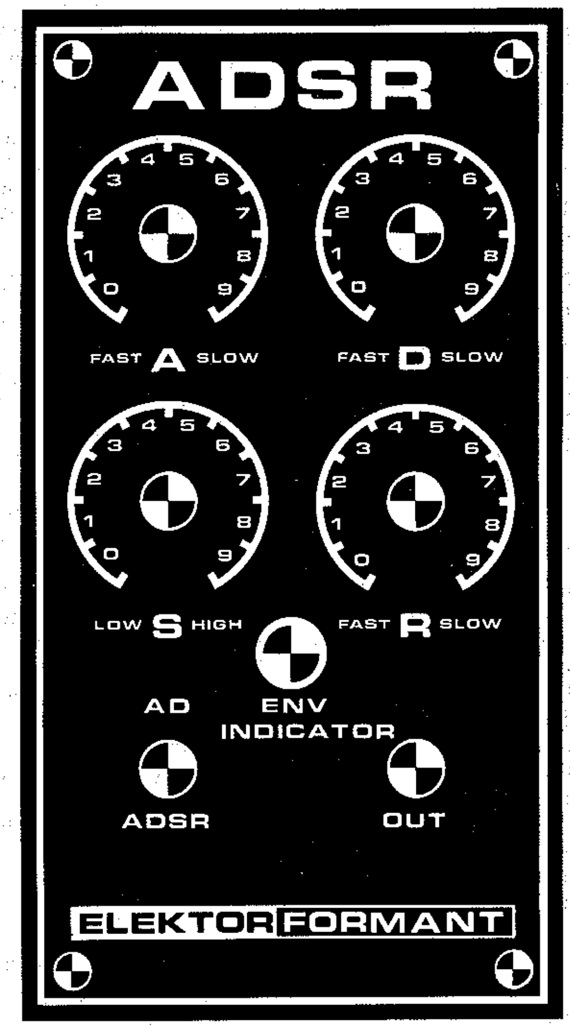

polyester or polycarbonate type. It is a good idea to test T3 and T6 for leakage, using the method detailed in chapter 5.
A printed circuit board and component layout for the envelope shaper are given in figure 8, and a front panel layout is shown in figure 9. Connections to the front panel are fairly simple, the only front panel-mounted components being the four potentiometers for attack time, decay time, release time and sustain level, switch S1, the external output socket and the envelope indicator LED.

Testing and adjustment

To test the envelope shaper a gate pulse must be available from the 'GATE' output of the interface receiver board. The EOS output of the envelope shaper is monitored on an oscilloscope with the Y sensitivity set to about 1 V/div and the timebase set to about 10 ms/div.
For the first test, the sustain level is set to zero, S1 is set to the 'AD' position and the attack and decay potentiometers are set to 'fast'. The release potentiometer has no effect during this test. If a key is depressed at short intervals then a short AD envelope curve will be seen, which rises and falls between about 0.5 V and 5 V. The output of IC3 can also be monitored, to check that it swings briefly between −15 V and +15 V when the peak of the attack curve is reached.
The only adjustment required to the envelope shaper is to set the 100% sustain level, using P5, to correspond with the voltage on C2 at the end of the attack period. If it is too low, then there will always be a decay, even at 100% sustain level; if it is too high then the calibration of P4 will be inaccurate, since 100% sustain will be reached before maximum rotation of the potentiometer.
To make the adjustment, the sustain level is set to 100% and medium attack and decay times are selected. Preset P5 is then adjusted until there is just no decay after the attack period (i.e. the attack period blends into the sustain level with no dip). The adjustment can be checked by turning P4 slightly to the left, when a slight dip after the peak of the attack period should be noted. As P4 is turned further anticlockwise then the decay down to the sustain level will become greater and greater, until finally, at 0% sustain level, pure AD curves will be produced.
The envelope shaper is now ready for use.

Figure 8. Printed circuit board and component layout for the envelope shaper (EPS 9725-1).

Figure 9. Front panel layout for the envelope shaper module.

chapter 10

voltage controlled amplifier (VCA)

This chapter continues the discussion of the tone-forming circuits with a description of the Dual VCA module, which can be used in conjunction with the envelope shaper for dynamic control of signal amplitude, and also for periodic amplitude modulation of the signal waveform (tremolo).

The voltage controlled amplifier module is called a 'Dual VCA' because it contains two cascaded, but independently controlled, amplifiers. The gain of the first amplifier is voltage controlled via an exponential converter, and is used for envelope shaping. The second has a linear gain-control input and is used for periodic modulation of signal amplitude (tremolo). The VCA is provided with a modulation indicator, which allows the best compromise to be obtained between signal-to-noise ratio and overload margin.

Connection of the VCA in the synthesiser system

Figure 1 illustrates how the VCA fits into the synthesiser system. The VCA takes its input from the output of the VCF, which in turn takes its input signal from the VCOs.

The VCF and VCA can both be controlled by the ADSR envelope shapers, so allowing dynamic variation of tone colour and amplitude during the playing of a note. However, the VCF has a KOV input from the keyboard to allow it to function as a tracking filter, but the VCA lacks this, since there is no pitch related control of signal amplitude.

Using the VCA and the VCF

It may be interesting at this point to spend a little time comparing and contrasting the effects produced by the VCA and VCF, and discussing how they are used to complement one another in the synthesiser system. As an example, consider the case where the VCA and VCF are both controlled by the same waveform from the envelope shaper, consisting of a rapid attack and a relatively slow exponential decay, as shown in figure 2a, and are fed with a 440 Hz sawtooth waveform.

If the VCF is used alone in the lowpass mode and the cutoff frequency of the filter is initally set very low, the input signal will be completely suppressed. However, during the attack phase of the envelope control waveform the cutoff frequency of the filter will rise very rapidly, and the amplitude and harmonic content of the note will both increase as first the fundamental, then the harmonics, are passed. During the slow decay phase the note will die away slowly as the cutoff frequency falls, starting with the higher harmonics, then the lower harmonics, and finally the fundamental. The variation in turnover frequency of the filter is illustrated in figure 2b.

The tone thus produced is not unlike that of a clavichord, or of a piano which has had drawing pins stuck into the hammers to produce a jangly, honky-tonk effect.

If the same signal and control waveforms are fed to the VCA, the signal amplitude will rise rapidly as the gain increases during the attack phase, and will fall away slowly during the decay phase. However, the harmonic content of the signal will remain unaltered. The sound thus produced is similar to that of percussion instruments such as the piano and xylophone.

By varying the attack and decay times of the envelope shapers a wide variety of tone colour and amplitude dynamics can be produced using the VCF and VCA in conjunction.

VCA design considerations

The dual VCA contains two amplifiers whose gains are independently voltage-controllable, and the design of the VCA poses certain problems, the principal one being that of obtaining adequate dynamic range, as is illustrated in figures 3a to 3d.

Figure 3a shows a control contour from the envelope shaper. At the peak of the control contour the VCA must have a finite maximum gain, which, for the purposes of the discussion, it will be assumed is unity, or 0 dB. At the beginning and end of the note the signal must be inaudible, which means that the gain of the amplifier should ideally be infinitesimally small at these moments in time. In practice, if the gain is around −70 dB then this will be adequate.

What happens if the dynamic range is inadequate is shown in figure 3b. Suppose the gain of the amplifier can be varied by a range of only 40 dB or so, and is set to 0 dB on the peak of the control contour. At the start and end of the note the signal will only be 40 dB down, and if the note is being played fortissimo then this residual signal will still be quite audible.

Another fault of badly-designed VCAs is illustrated in figure 3c. In this example, the VCA cuts off completely below a certain level of control voltage, and so misses part of the attack and decay period of the note. This might be said to be the opposite fault to that of figure 3b, though it is not directly related to dynamic range, but rather to extreme non-linearity of the control characteristic.

Returning to the example of the VCA with only 40 dB dynamic range, if the gain is adjusted so that the signal is inaudible at the beginning and end of the note (i.e. some 70 dB down), it will only be able to increase by 40 dB when the control voltage is applied, instead of the 70 dB required to reach the 0 dB level. The result is an amplitude plateau, as shown in figure 3d.

As mentioned briefly earlier, control of the envelope shaping section of the VCA is carried out exponentially. This is to compensate for the logarithmic loudness response of the human ear. On the other hand control of the periodic amplitude modulation section (tremolo) is linear, since this gives the 'softest' and 'sweetest' sound to the tremolo effect.

Principle of the Formant VCA

The VCA in Formant uses the CA3080 OTA as the controllable amplifier, as in the VCF. The principle of the Formant VCA is illustrated in figure 4. The input voltage U_i is converted to a proportional output current $I_o = g_m \cdot U_i$. However, since we are interested in voltage amplification this output current must be converted into an output voltage, and this is done simply by feeding the current through a load resistor R_L to produce an output voltage $U_o = g_m \cdot U_i \cdot R_L$.

The transconductance of the amplifier, g_m, may of course be varied by a control current I_{ABC}, as explained in chapter 6, and the gain of the VCA may thus be controlled – although at this stage of course it is a CCA!

The output of the OTA may not drive any external load in addition to R_L, as this would lower the load impedance and alter the gain, so the output of the OTA is connected to a voltage follower/buffer with a high input impedance.

Both sections of the VCA operate on the same principle. However, only the output of the second OTA is buffered, since it is this output that is connected to any external loads. As the output of the first OTA has no external connection it is simply connected to the input of the second OTA.

The OTA has one disadvantage that cannot be ignored. As mentioned in the previous chapter, its linearity is good only for small input signals (typically ± 10 mV) which is why a large degree of input signal attenuation is required. This means that the signal-to-noise ratio is not exceptionally good, and for this reason it is best to use the VCA with the largest possible input signal consistent with low distortion. A modulation indicator is provided, which allows the best compromise to be obtained between excessive noise, at low input levels, and distortion at high input levels.

Figure 1. Block diagram illustrating how the VCA fits into the Formant synthesiser system.

Figure 2. Envelope control of the VCF and VCA. The attack-decay contour of figure 2a, when applied to the VCF, varies the turnover frequency of the filter, which provides dynamic alteration in the tone colour of the sound (figure 2b). When applied to the VCA, the envelope contour alters the gain of the VCA, and thus the amplitude of the sound (figure 2c).

Circuit of the VCA

The complete circuit of the VCA is given in figure 5. The exponential converter built around IC1 and IC3 will immediately be recognised, since it is very similar to that used in the VCF. The input configuration, however, is much simpler, there being but one external input, ENV, from the envelope shaper. If required this can be switched out by setting S1 in position 'a', in which case a fixed gain results.

The gain/control voltage characteristic of the VCF is roughly 12 dB/volt, but as the use of the word 'roughly' suggests, the accuracy of this characteristic is relatively unimportant, unlike the octave/volt characteristics of the VCO and VCF. The ear is much less critical of amplitude errors than it is of frequency errors. The dB/volt characteristic of the VCA may be adjusted by P2, whilst P1 is an offset trimmer. The output current of the exponential converter controls the gain of the first OTA, IC6.

The linear voltage-current converter is constructed around IC2, which is connected as an inverting, summing amplifier. An input signal may be fed to P4 via the AM input socket, and a DC input voltage is available from P3 ('Gain'). Both these input voltages cause proportional currents to flow through R12 and R13, and since these currents cannot flow into the inverting input of the op-amp they flow round the feedback loop through T1, and into the control input of IC7.

The audio signal to the VCA comes either from the permanently wired internal signal input (IS) or from the external signal socket (ES) on the front panel of the VCA module. The amplitude of the external input signal is controlled by P5, whereas the amplitude of the internal signal is controlled at the IOS output of the VCF, by P6 of the VCF module.

IC4 functions as a summing amplifier with a gain of −1, and the signal level at the output of IC4 is monitored by the modulation indicator constructed around IC5. This is a non-inverting amplifier feeding a bridge rectifier D1 to D4, the output of which drives the modulation indicator LED D5. Once the

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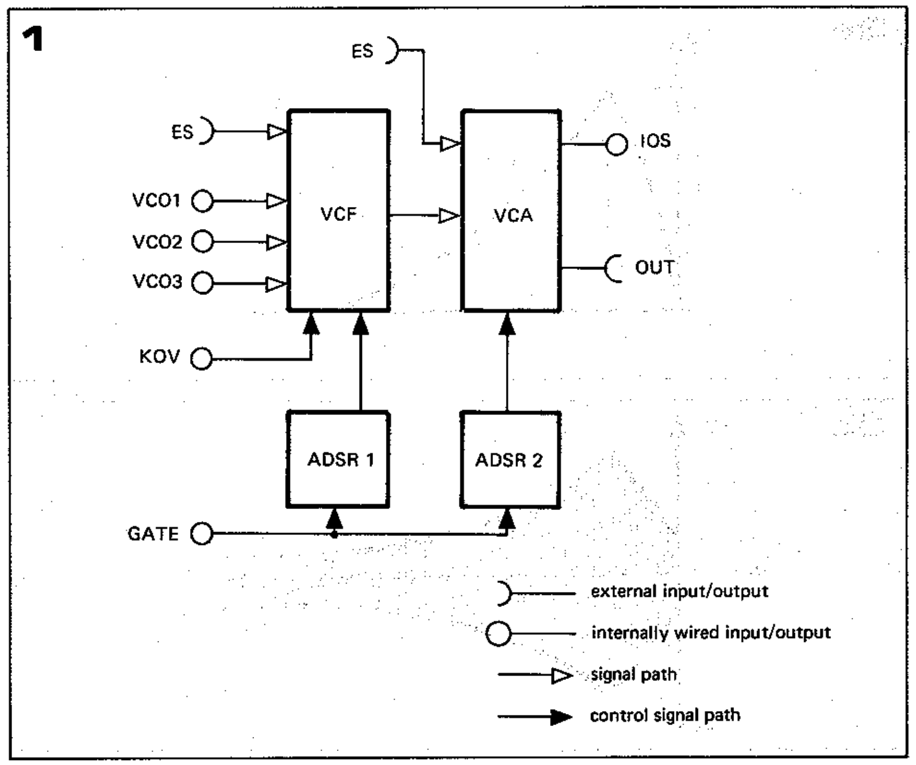
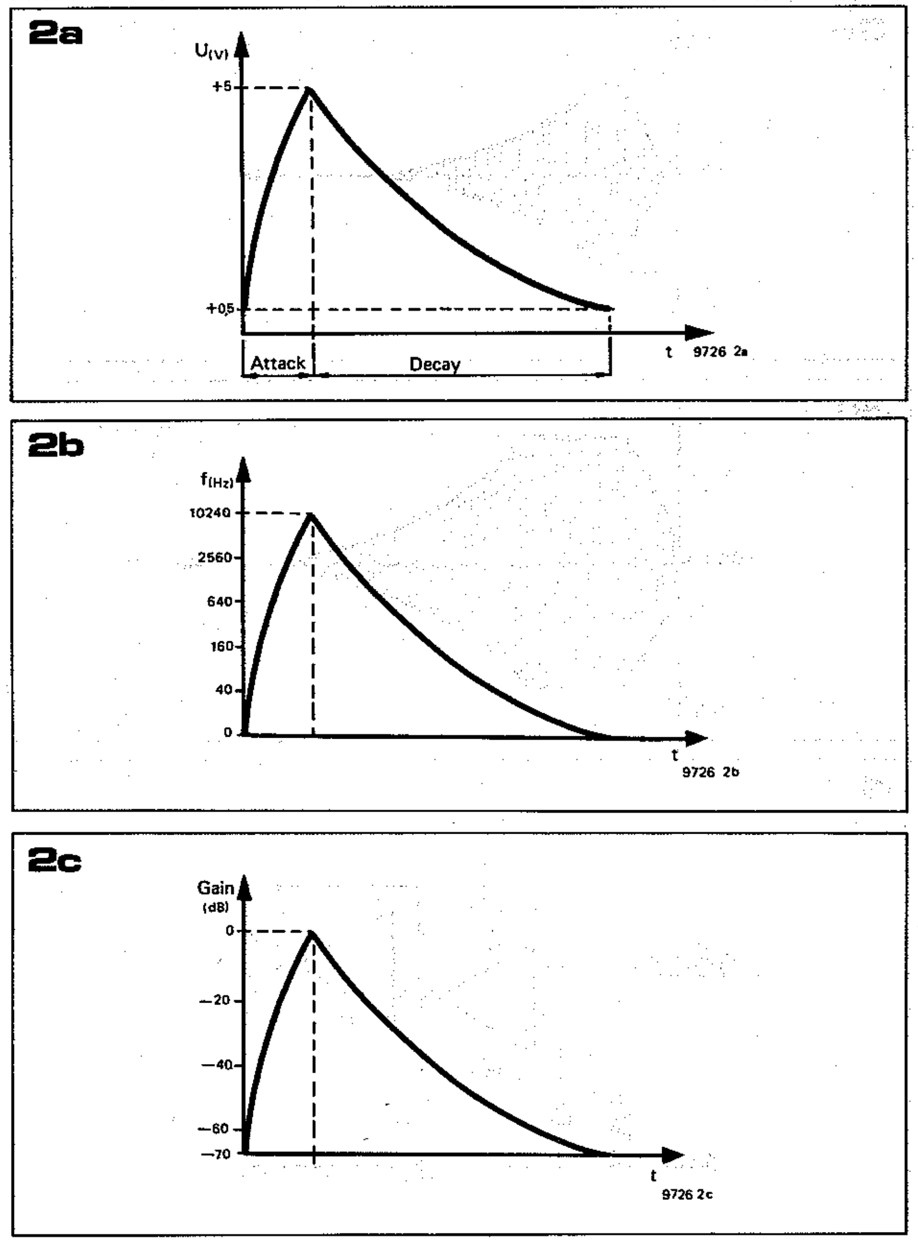
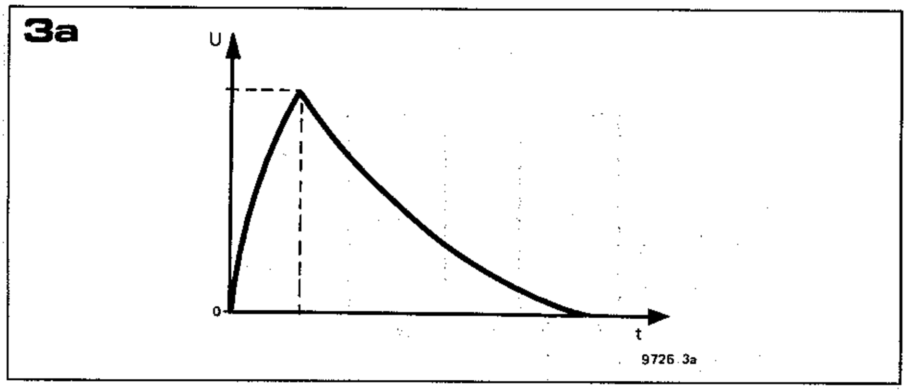




3a


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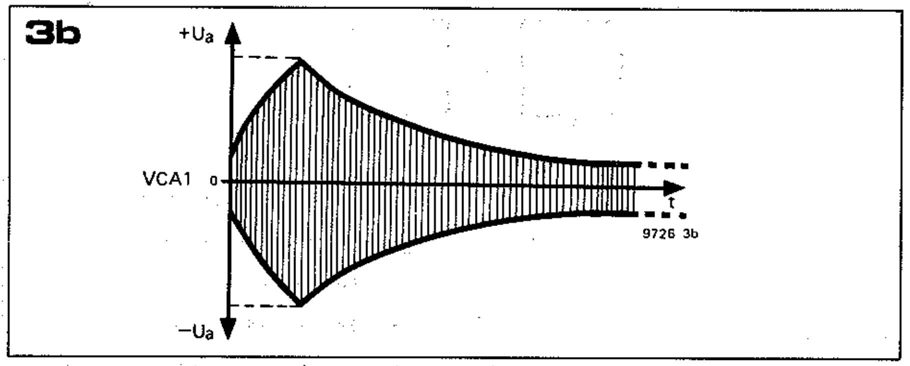
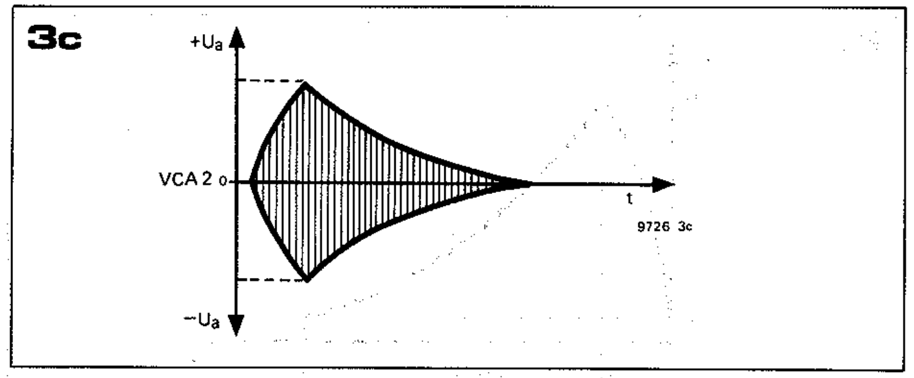
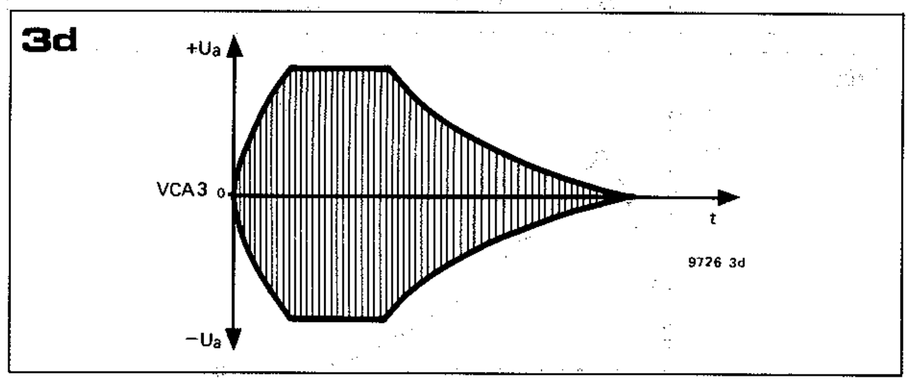


3c


3d


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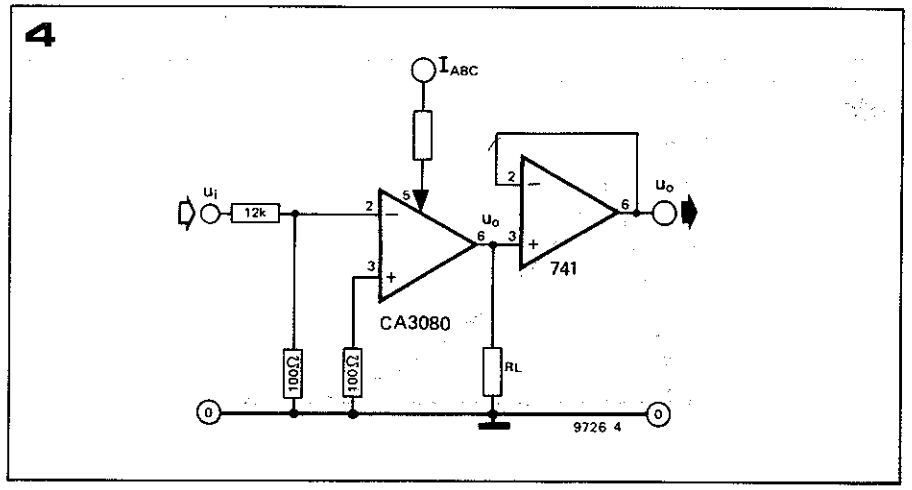

peak signal level at the output of IC5 exceeds the combined knee voltages of D1 plus D5 plus D4 (or D3 plus D5 plus D2) then the LED will start to glow and will glow brighter as the signal level increases. P6 is used to adjust the gain of IC5 so that D5 starts to glow at the signal level where overmodulation begins to occur.
The output signal from IC4 is attenuated by R19 and R20 down to a level which the OTA, IC6, can handle. The output of the exponentially controlled OTA, IC6, is fed via a second attenuator R25/R26, to the input of the linearly-controlled OTA, IC7. The output of IC7 is buffered by voltage-follower IC8 and two outputs from the VCA are provided, an internally wired output, IOS, and an output to a front panel socket, EOS. Potentiometers P7 and P8 are provided for trimming the offset voltages of IC6 and IC7.

Construction

The comments with regard to component quality that have been made in previous chapters apply equally to the construction of the VCA, and will not be repeated. A printed circuit board and component layout for the VCA are given in figure 6, and a front panel layout is given in figure 7.

Figures 3a to 3d. Some typical faults of badly-designed or badly-adjusted VCAs are illustrated here. None of the amplitude envelopes in figures 3b to 3d follows the control contour of figure 3a.
In figure 3b there is feedthrough of the signal after the control contour finishes; in figure 3c the signal is still cut off for some time after the control contour starts, and cuts off again before it finishes; in figure 3d the VCA has insufficent headroom and limits causing a 'plateau' on top of the envelope curve.

Figure 4. The principle of the Formant VCA is illustrated here. The OTA produces an output current proportional to the product of the input voltage and the control current I_{ABC}. This causes a voltage drop across the load resistor R_L, and the output is buffered by an op-amp voltage follower. The input attenuator is necessary to avoid overloading the OTA.

Figure 5. Complete circuit of the Formant Dual VCA. This contains two, cascaded, voltage-controlled amplifiers with independent control inputs; exponential control for envelope shaping and linear control for amplitude modulation (tremolo).

Testing and adjustment

For optimum performance the VCA must be matched to a particular envelope shaper, and thereafter the VCA and envelope shaper should be used as a pair. This is not necessary in the case of the VCF, which may be used with any envelope shaper.
To test and adjust the VCA, the completed keyboard and interface receiver must be available, together with VCOs, VCF and the envelope shaper to which the VCA is to be matched. The IOS output of the VCO is connected to one of the VCO inputs of the VCF, and the IOS output of the VCF is connected to the IS input of the VCA. The GATE output of the interface receiver is connected to the GATE input of the envelope shaper and output ENV of the envelope shaper is connected to input ENV of the VCA.
For the initial test, the sawtooth output of the VCO is selected and the output level is set to maximum. The VCF is set to the lowpass mode, but the turnover frequency is set to maximum by turning the octaves control fully clockwise. The Q control is set to minimum, the KOV input is switched off and the output level is set to maximum.

- At the IOS output of the VCF, the sawtooth signal from the VCO should now be available in phase with, and at the same amplitude as, the VCO output (about 2.5 V p-p).
- At the output of IC4 of the VCA, the signal should be available at the same level, but inverted.
- With S1 of the VCA in position 'a' (ENV input switched off) and P7 and P8 in mid-position, the sawtooth signal should be available at the output of IC6 in phase with the VCO output, and the amplitude should be adjustable by P1.
- At the output of IC7 the signal should again be in phase, and both P1 and P3 should vary the amplitude.
- Finally, the signal should be available at outputs IOS and EOS.

This concludes the basic functional check of the VCA, and the adjustment procedure can now be carried out.

Modulation Indicator

Using the same input signal, P6 is adjusted until the modulation indicator D5 just begins to glow. Increase the signal amplitude by switching in the second and third VCOs, when the LED should glow brighter.
After this test, the second and third VCOs should be switched off again.

Offset adjustment

Turn the output level of the VCF to zero and short the IS input of the VCA to ground. Set S1 of the envelope shaper to 'AD' and the A, D, S and R controls to minimum (shortest attack and decay, and 0% sustain). Turn P5 of the VCA fully anticlockwise, set S1 of the VCA to position 'b' (ENV) and observe the DC output voltage of IC6 on an oscilloscope.
When a key is depressed, a step output voltage will be observed at the output of IC6. This is the offset voltage of the IC, which is amplified as the gain of IC6 increases under the influence of the envelope control voltage; if it is not nulled out then it will break through to the output as 'cracks' or 'plops'. P7 is adjusted until the step voltage is as small as possible on the most sensitive range of the oscilloscope.
The offset nulling procedure must then be repeated for IC7. S1 is switched to the 'off' position, P3 is turned fully anti-clockwise and the external output of the envelope shaper is connected to

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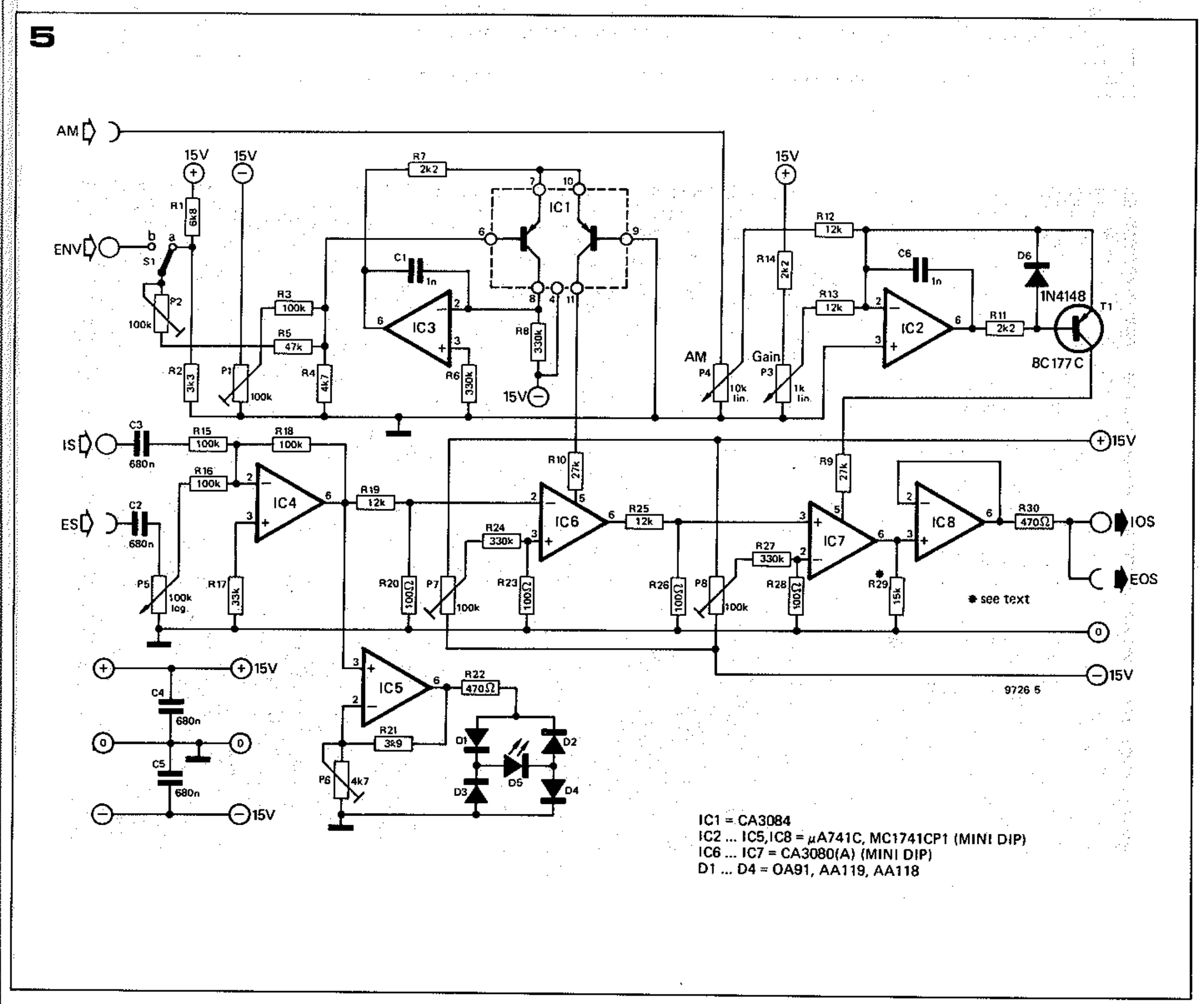

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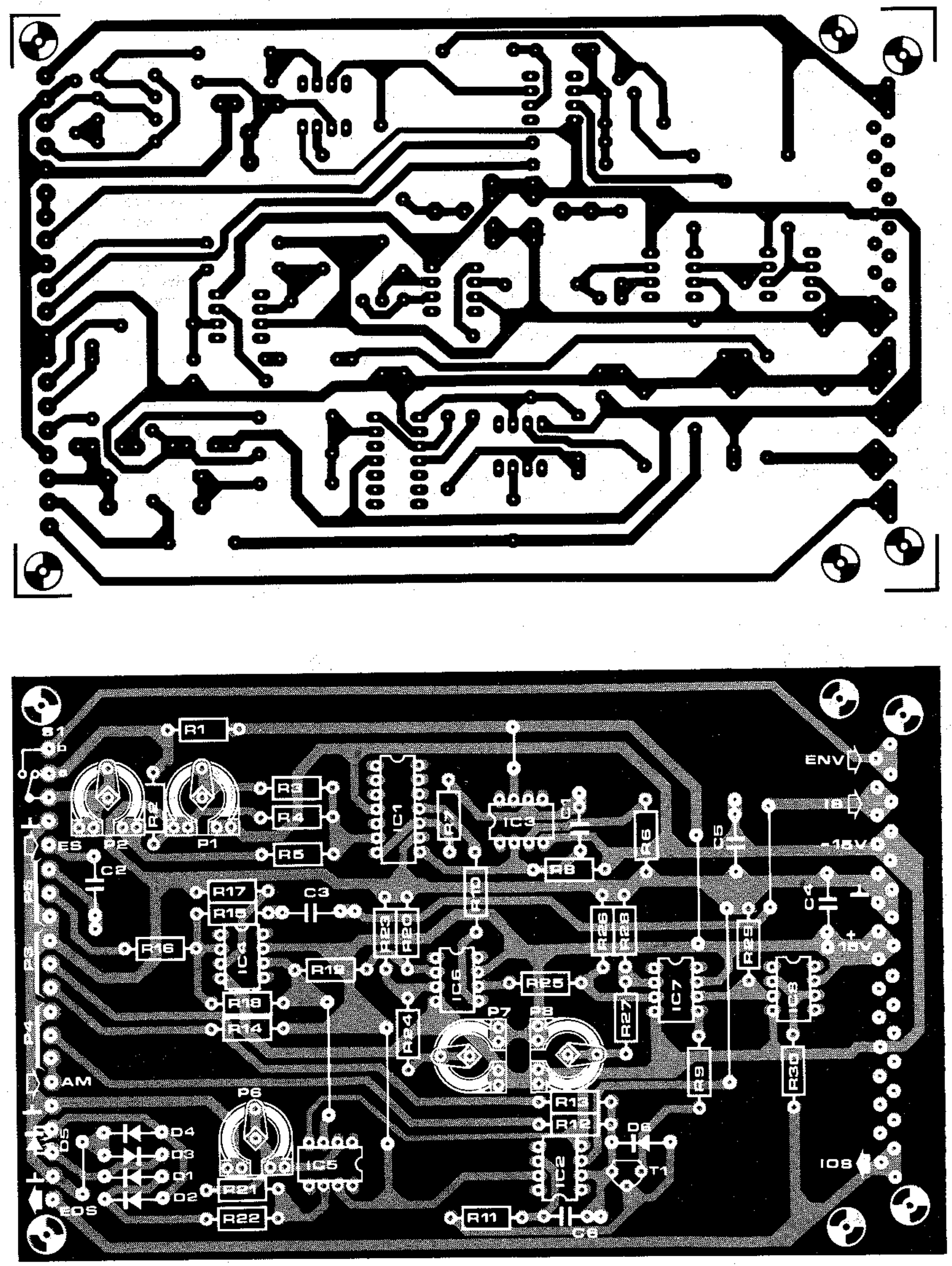

Parts list for figures 5 and 6

Resistors:
R1 = 6k8
R2 = 3k3
R3,R15,R16,R18 = 100 k
R4 = 4k7
R5 = 47 k
R6,R8,R24,R27 = 330 k
R7,R11,R14 = 2k2
R9,R10 = 27 k
R12,R13,R19,R25 = 12 k
R17 = 33 k
R20,R23,R26,R28 = 100 Ω
R21 = 3k9
R22,R30 = 470 Ω
R29 = 15 k (nominal value, see text)

Potentiometers:
P1,P2,P7,P8 = 100 k preset
P3 = 1 k lin.
P4 = 10 k lin.
P5 = 100 k log.
P6 = 4k7 (5 k) preset

Capacitors:
C1,C6 = 1 n
C2,C3,C4,C5 = 680 n

Semiconductors:
IC1 = CA 3084 (DIL package)
IC2,IC3,IC4,IC5, IC8 = µA 741C, MC 1741 CP1 (Mini DIP)
IC6,IC7 = CA 3080 (A)
T1 = BC 177B, BC 178C, BC 179C, BC 557B, BC 558C, BC 559C
D1 . . . D4 = DUG (OA91, AA118, AA119)
D5 = LED (TIL 209 or similar)
D6 = 1N4148

Miscellaneous:
31-way Euro connector (DIN 41617)
S1 = SPST miniature toggle switch
3 off, 3.5 mm jack socket
3 off, 13-15 mm collet knobs with pointer

Figure 6 Printed circuit board and component layout for the VCA (EPS 9726-1).

Figure 7 Front panel layout of the VCA. S1 is located between the AM and ES input sockets. Immediately below these sockets are the respective input level controls: P4 sets the AM modulation depth and P5 is the external input level control Below these again are the modulation indicator (D5), the manual gain control (P3) and the output socket (EOS).

the AM input of the VCA. The IOS output of the VCA is monitored on the oscilloscope and the offset nulling procedure is repeated, this time using P8.

Adjustment of exponential gain control

The exponential converter must be adjusted so that the required gain control range of IC6 is obtained from the +0.5 V to +5 V range of the envelope shaper.

S1 of the envelope shaper is set to the 'AD' position and fairly short attack and decay times are selected. The short circuit across the VCA input is removed, the VCF level control is turned to maximum and a signal is fed in from one of the VCOs. P2 on the VCA board is initially set to its mid-position.

The output of IC6 is now monitored with an oscilloscope and a key is repeatedly depressed, when AD envelope curves should be seen. P1 is then adjusted for minimum feed through when the key is not depressed, less than one or two millivolts will be acceptable. The Y sensitivity of the oscilloscope is now adjusted so that the entire envelope curve can be seen when a key is depressed. P2 should then be adjusted until a good attack/decay curve without limiting (seen as a flat top or plateau as shown in figure 3d) is just obtained. Since P1 and P2 interact to some extent, it may be necessary to repeat the adjustment procedure several times to obtain the best results.

Adjustment of overall gain

The overall gain of the VCA should be 0 dB (unity) at maximum modulation of IC6 and IC7. To achieve this it may be necessary to alter the value of R29, which is nominally 15 k. Set the gain control P3 to maximum, and the envelope shaper to the 'ADSR' mode with 100% sustain. A key is now depressed and held down, and the output level of the VCA (at IOS or EOS) is compared with the input level at IS. These levels should be the same; if the output level is too low, then R29 must be increased in value, and if the output level is too high then R29 must be reduced. A 3 dB difference (x 0.707 or x 1.414) between the input and output levels is acceptable.

This completes the adjustment of the VCA.

Use of the VCA

The input signal level to the internal input of the VCA is controlled by the output potentiometer of the VCF. In

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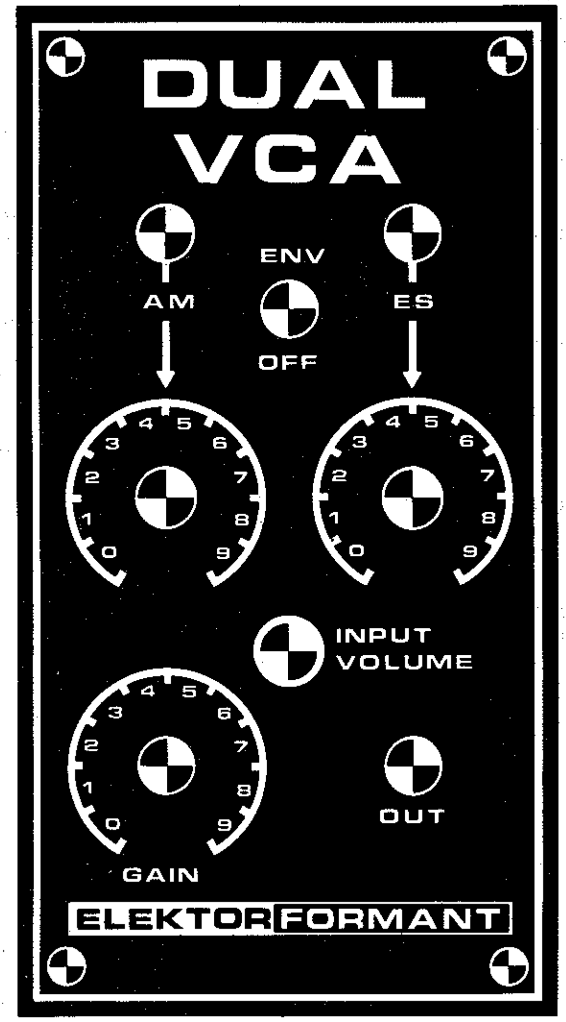

normal use this control should be adjusted so that the LED just begins to glow, which occurs at a nominal level of 2.5 V p-p with one VCO input signal, less if more than one VCO is connected. If the LED glows brightly, then the VCA is being overmodulated and distortion may occur. This is not to say that this should never be allowed to happen, since the deliberate introduction of distortion can be used to produce 'fuzz' effects. If the LED does not glow, then this indicates under-modulation and the possibility of a poor signal-to-noise ratio.

Tremolo

To produce tremolo effects a low-frequency oscillator signal (LFO) can be fed into the AM input socket. The Formant LFOs, described later in the series, have an output voltage swing of ± 2.5 V, and if the GAIN potentiometer P3 is set in its mid-position this will give a modulation depth of 100%. Reducing the LFO input signal by means of the AM potentiometer P4 allows the modulation depth to be varied down to 0%.

Expression Pedal

An expression pedal may also be connected to the AM input. This can be a pedal fitted with a logarithmic potentiometer and battery, whose output can be varied from zero to about +5 V with the pedal fully depressed.

Tuning

The ENV/OFF switch S1 is particularly useful when tuning the synthesiser, since it allows signals to pass continuously through the VCA, unaffected by the envelope shaper when in the OFF position.

Outputs

The external output of the VCA has an impedance of about 500 Ω, and this output may be fed to other equipment such as tape decks and external amplifiers, or to high impedance headphones for monitoring.

The internal output signal (IOS) is taken to the Formant amplifier module, which will be described later. This is fitted with tone and volume controls and a small power amplifier for monitoring purposes. It will drive low impedance headphones and loudspeakers, and can also be used to drive spring line reverberation units or other external equipment.

chapter 11

LFOs and noise module

The low frequency oscillators and noise generator are invaluable components in a synthesiser system. The LFOs allow amplitude and frequency modulation of the VCO outputs to provide tremolo, vibrato and other effects. The noise sources can be used for random modulation of the VCO signals, and in addition can be used as signal sources themselves.

Mention has already been made of the fact that conventional instruments exhibit more 'life' and variation in tonal character than electronic instruments due to the way in which they are played. For example, string instruments and woodwind instruments can exhibit marked tremolo and/or vibrato due to variations in the bowing or blowing. The keyboard of a synthesiser provides a relatively inflexible and expressionless means of playing that does not allow these nuances to be introduced into the sound, and in order to make the sound more 'lively' amplitude and frequency modulation must be introduced using the LFOs and noise source.

The noise source also provides the basic material to produce a whole spectrum of sounds that do not have a defined pitch. White noise can be used to produce sounds such as wind, rain and surf. 'Coloured' noise, which is white noise with the low frequency components boosted, is used for sounds having a strong bass content, such as the rumbling of thunder.

In addition to modulating the VCO signals, noise can also be added to these signals to simulate wind noise in organ pipes and woodwind instruments.

The LFO module

The Formant LFOs are basically low-frequency function generators that produce three different waveforms. Each LFO module contains three LFOs, two of which are identical and produce square, triangle and sawtooth waveforms. The third LFO produces a triangular waveform and two sawtooth waveforms in antiphase with each other, i.e. one with a positive-going ramp and the other with a negative-going ramp.

The circuit of LFO1 is shown in figure 1a; LFO2 is identical. The basic oscillator circuit consists of two op-amps IC1 and A3 connected respectively as an integrator and a Schmitt trigger. When the output of A3 is positive a potential of about +2.5 V (depending on the position of the wiper of P3) is applied to R9. The full positive output voltage of A3 is applied to P1, so a current (dependent on the wiper position of P1) flows into the integrator through R1. The output of IC1 ramps negative until it reaches about −2.5 V, when the voltage on the non-inverting input of A3 will fall below the voltage on the inverting input (zero volts) and the output of A3 will swing negative. The voltage applied to R9 is now −2.5 V, and the full negative output voltage of A3 is applied to P1. Current will flow out of the integrator through R1, and the integrator output will ramp positive until it reaches about +2.5 V, when the voltage on the non-inverting input of A3 will rise above zero and the output of A3 will swing positive. The whole cycle then repeats.

The output from IC1 is thus a triangular waveform with a peak-to-peak voltage of about 5 V, while at the wiper of P3 a squarewave of the same amplitude and frequency is available. P3 presets the trigger threshold of A3 and hence the signal amplitude. P1 is used to adjust the frequency of the LFO by varying the voltage applied to the integrator input, which alters the integrator input current and hence the rate at which the integrator ramps positive or negative.

The triangular wave output is taken direct from IC1 via R13, whilst the squarewave output is buffered by voltage follower A4. The sawtooth waveform is derived from the triangle by A2. When the output of A3 is positive and the triangle output is on its negative-going slope, T1 is turned on, grounding the non-inverting input of A2. A2 thus functions as a unity-gain inverting amplifier, producing a positive-going ramp. When the output of A3 is negative and the output of IC1 is positive going, T1 is turned off and A2 functions as a unity-gain non-inverting amplifier (voltage follower), again producing a positive-going ramp. The positive- and negative-going ramps of the triangular waveform are thus converted into a series of positive-going ramps. Since every half-cycle of the triangle is converted into a full cycle of the sawtooth, the frequency of the sawtooth is twice that of the triangle and square waveforms, as illustrated in figure 2.

To indicate that the LFO is functioning a LED indicator, constructed around A1, is connected to the triangle output.

Figure 1a. Circuit of LFO 1, which is identical to LFO 2 and produces triangle, sawtooth and square waveforms.

Figure 1b. LFO 3 also produces three waveforms, but instead of a squarewave output a negative-going sawtooth is provided.

The third LFO circuit, shown in figure 1b, is similar to the first circuit, with two exceptions. Firstly, no squarewave output is provided; secondly, a sawtooth with negative-going slope is provided by A8, which inverts the positive-going sawtooth from A6.

Construction of the LFO module

Figure 3 shows the printed circuit board and component layout of the LFO module, which of course contains three LFOs. The components for LFO2 are identical to those for LFO1, being distinguished on the board and in the components list by an apostrophe (').

The board layout is fairly cramped, and care should be taken when soldering components to avoid solder bridges. A front panel layout is given in figure 4.

Adjustment of the LFOs

Each LFO requires four adjustments:

- P3, P3' and P7 set the signal amplitude.
- P2, P2' and P5 null the offset of the integrators.
- R16, R16' and R17 must be selected to set the lowest frequency of the LFO.
- P4, P4' and P6 adjust the LED indicators.

The adjustment procedure, which is identical for all three LFOs, will be described for LFO1.

Amplitude adjustment

1. Monitor the triangle output on an oscilloscope; set P2 to its mid-position and P1 for maximum frequency.
2. Adjust P3 to give a peak-to-peak output of 5 V.
3. Check the amplitude and waveform of the other outputs.

Offset adjustment

1. Disconnect R1 from the wiper of P1 and ground it.
2. Monitor the output voltage of IC1 with a multimeter. It will probably exhibit a tendency to drift positive or negative, and the voltage will settle at +15 V or −15 V. Reset the

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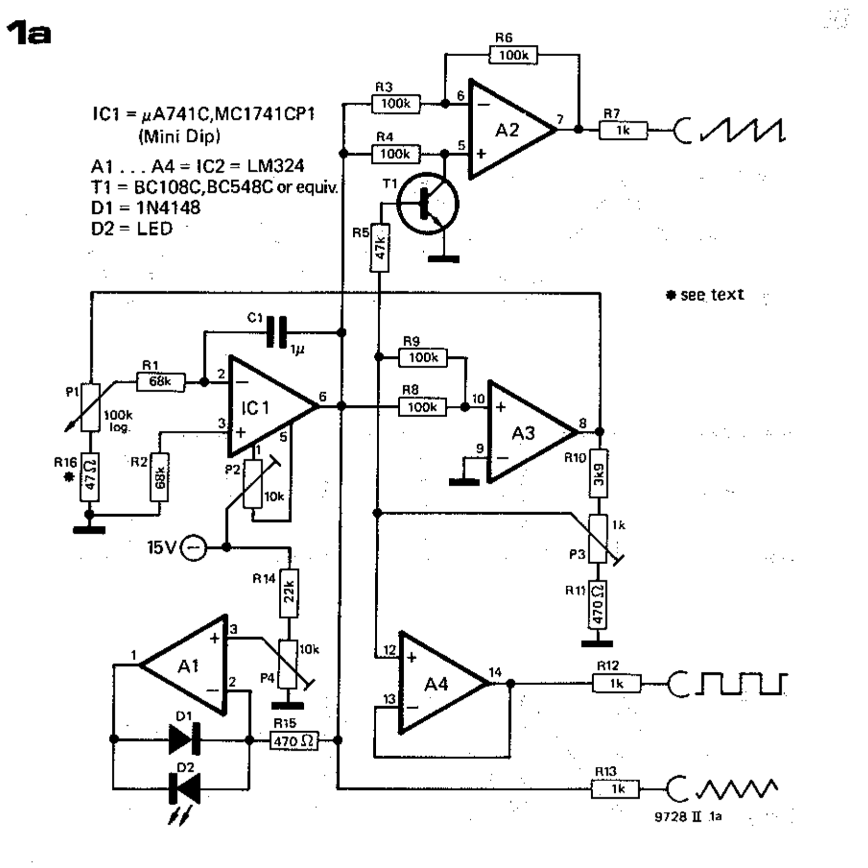

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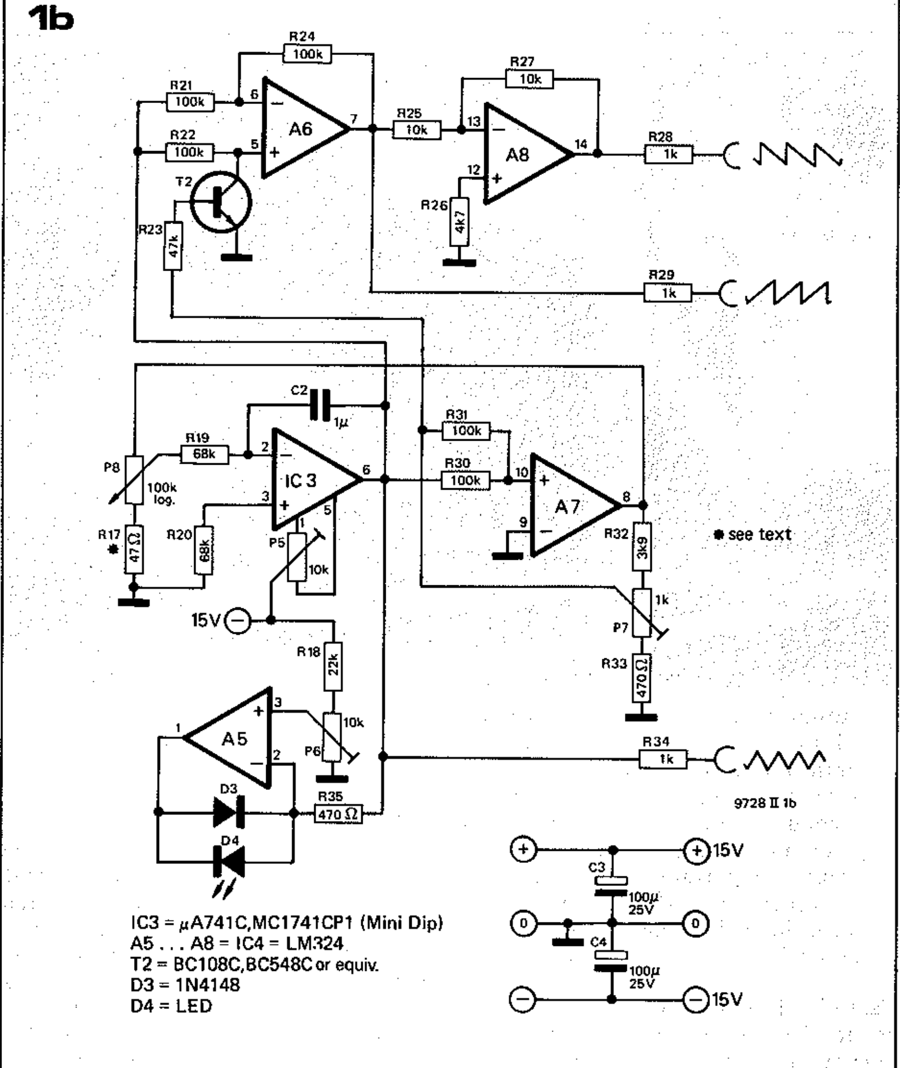

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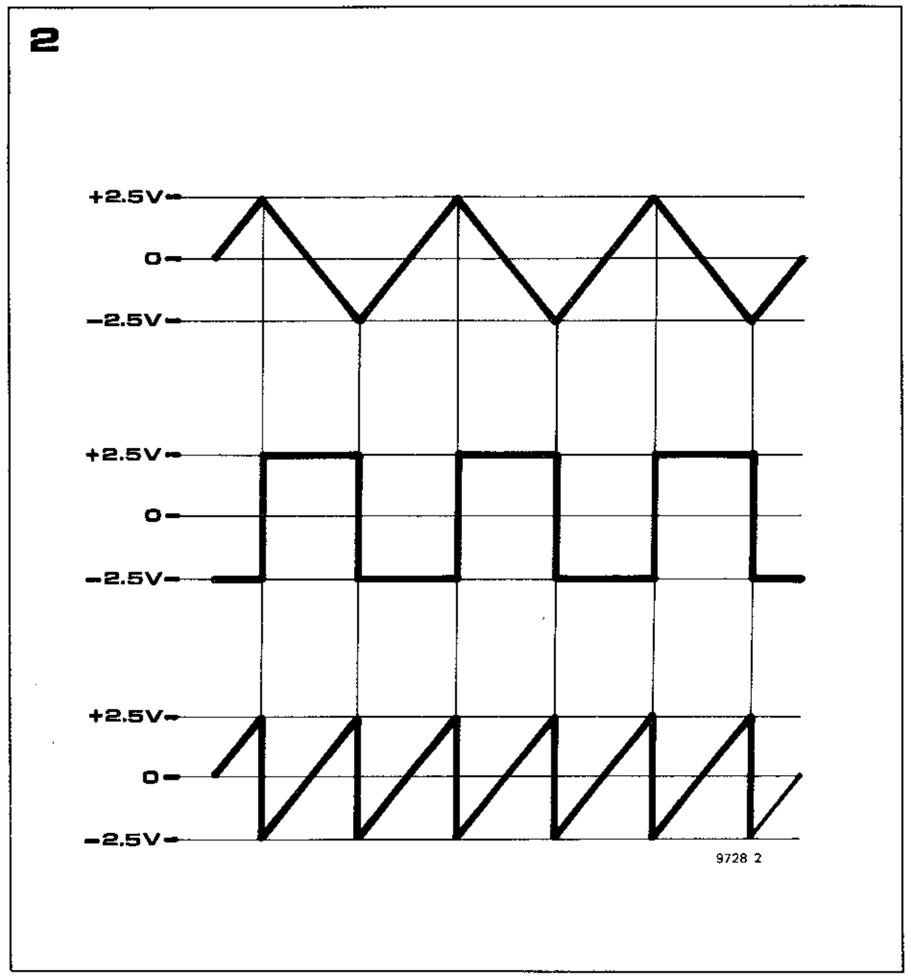
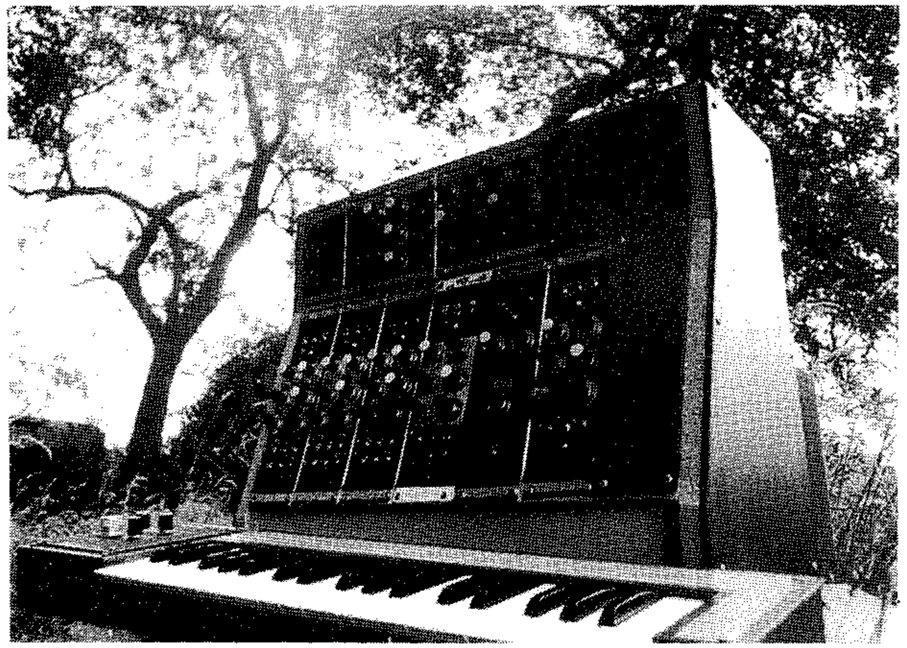


Figure 2. Showing the phase relationship of the triangle, square and sawtooth waveforms. Since the sawtooth is derived by inverting alternate half-cycles of the triangle waveform, its frequency is twice that of the other waveforms.



output voltage to zero by discharging C1 through a 1 k resistor. Adjust P2 until the voltage remains stable at zero volts for a period of several seconds (without the discharge resistor in circuit). Repeat this adjustment, progressively switching the multimeter to more sensitive ranges until the drift is only a few hundred millivolts in several seconds.

Careful adjustment of the offset is vital, as it determines the minimum frequency at which the LFO will operate reliably and the symmetry of the waveforms at low frequencies.

Selection of R16

The value of R16 determines the minimum integrator input voltage that can be set by P1, and hence the minimum frequency of the LFO. The value of R16 must not be chosen too high or the minimum LFO frequency will be too great. On the other hand it should not be chosen too low, or the integrator input current at the minimum setting of P1 will be comparable with the input currents of IC1. This will result in unreliable operation of the oscillator at low frequencies.

R16 should be chosen so that the minimum frequency of the LFO is about one cycle every three minutes, but the value of R16 should not be less than 10 Ω. If it is not possible to obtain this low frequency then the input currents of IC1 may be too high, or C1 may be leaky.

The maximum LFO frequency is about 20 Hz.

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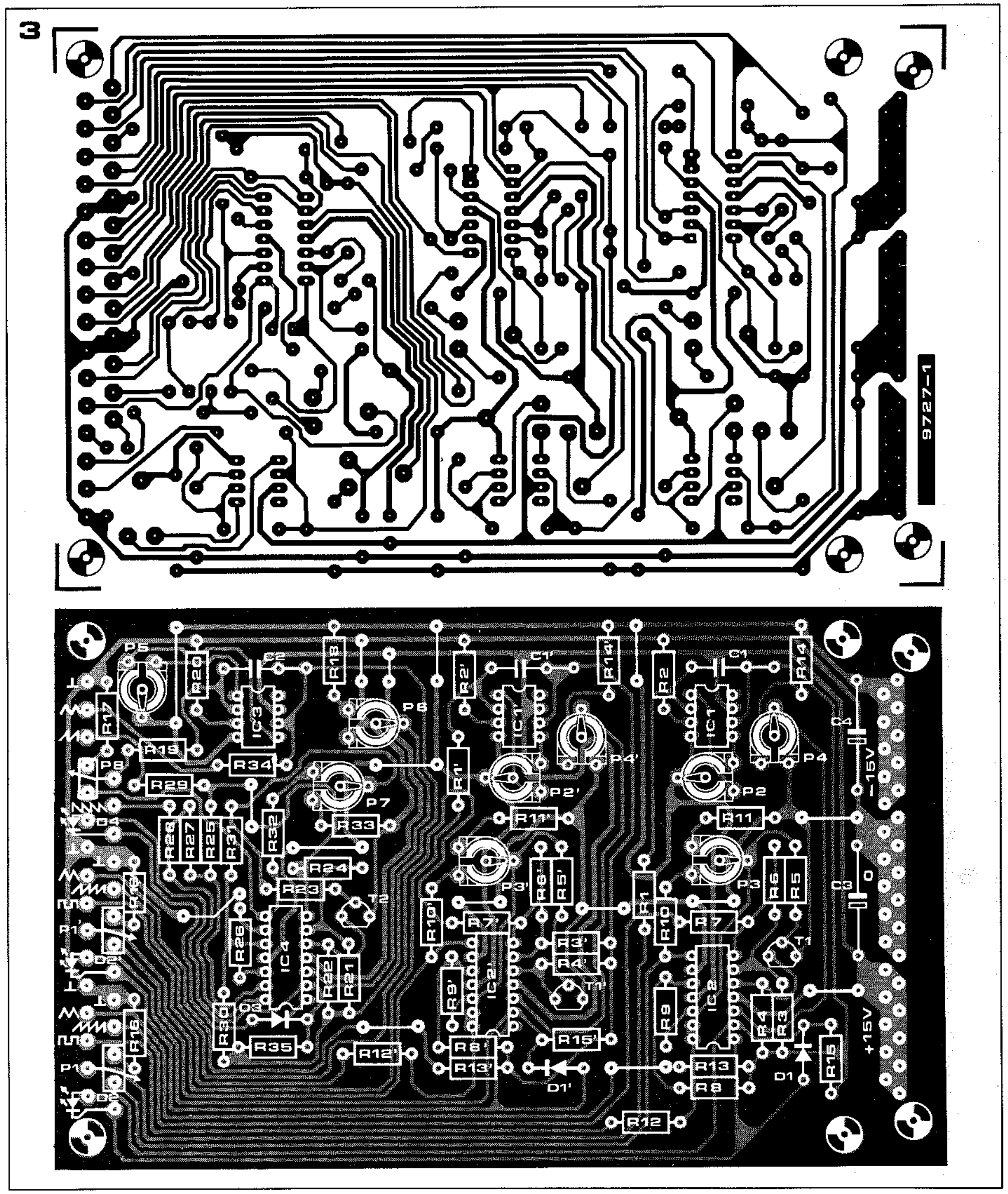

Figure 3. Printed circuit board and component layout for the LFO module (EPS 9727-1).

Parts list for LFO module

Resistors:
R1,R1',R2,R2',R19,R20 = 68 k
R3,R3',R4,R4',R6,R6',R8,R8', R9,R9',R21,R22,R24,R30, R31 = 100 k
R5,R5',R23 = 47 k
R7,R7',R12,R12',R13,R13',R28, R29,R34 = 1 k
R10,R10',R32 = 3k9
R11,R11',R15,R15',R33, R35 = 470 Ω
R14,R14',R18 = 22 k
R16,R16',R17 = 47 Ω (see text)
R26 = 4k7

Potentiometers:
P1,P1',P8 = 100 k log
P2,P2',P4,P4',P5,P6 = 10 k preset
P3,P3',P7 = 1 k preset

Semiconductors:
IC1,IC1',IC3 = μA 741C, MC 1741CP1 (Mini DIP)
IC2,IC2',IC4 = LM 324 (DIP)
T1,T1',T2 = BC 108C, BC 548C or equivalent
D1,D1',D3 = 1N4148, 1N914
D2,D2',D4 = LED (e.g. TIL209)

Capacitors:
C1,C1',C2 = 1 μ (polyester or polycarbonate)
C3,C4 = 100 μ/25 V

Miscellaneous:
31-way connector (DIN 41617)
9 x 3.5 mm jack
3 x 13 . . . 15 mm knobs

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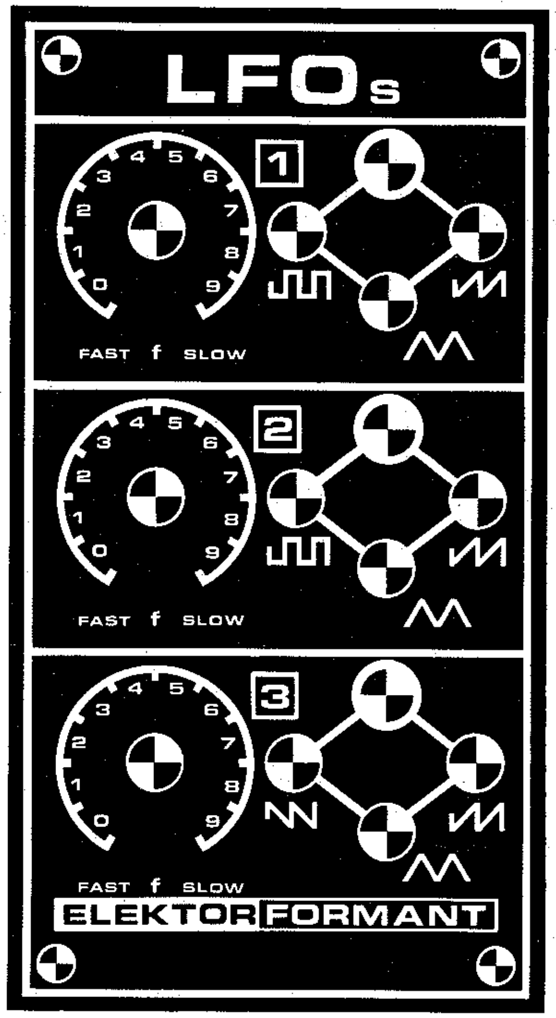

Figure 4. Front panel layout of the LFO module.

Figure 5. Circuit of the noise module.

Figure 6. Printed circuit board and component layout for the noise module (EPS 9728-1).

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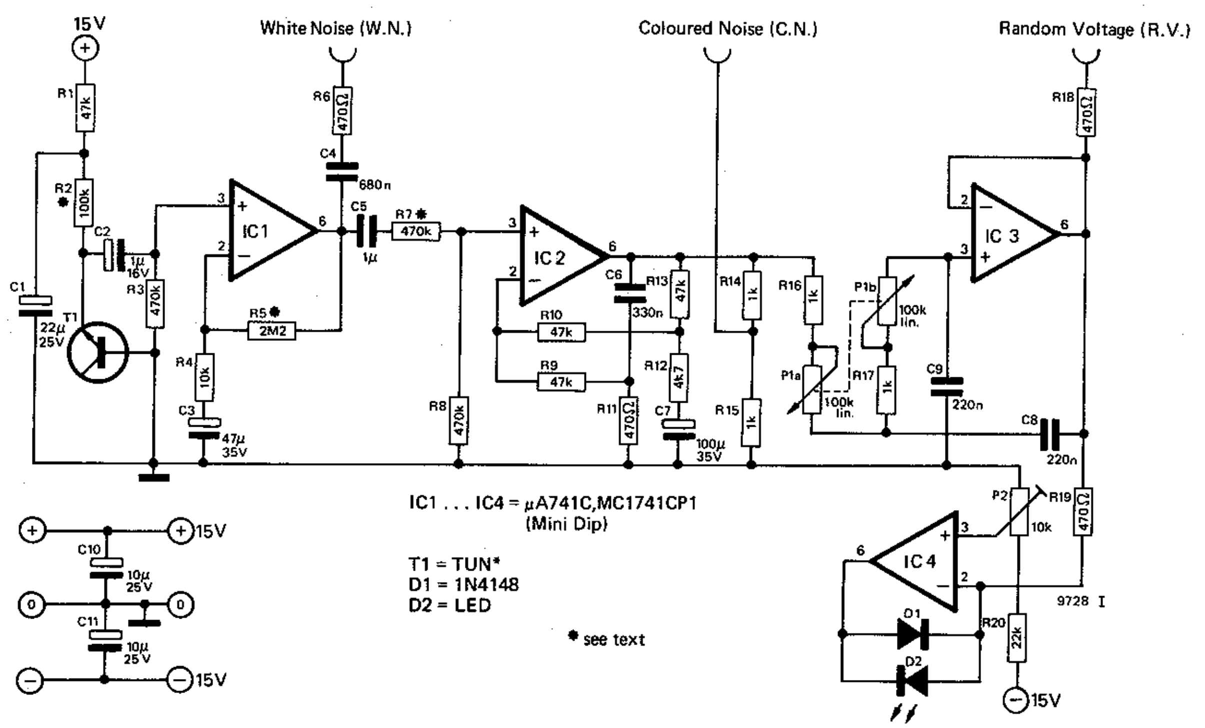

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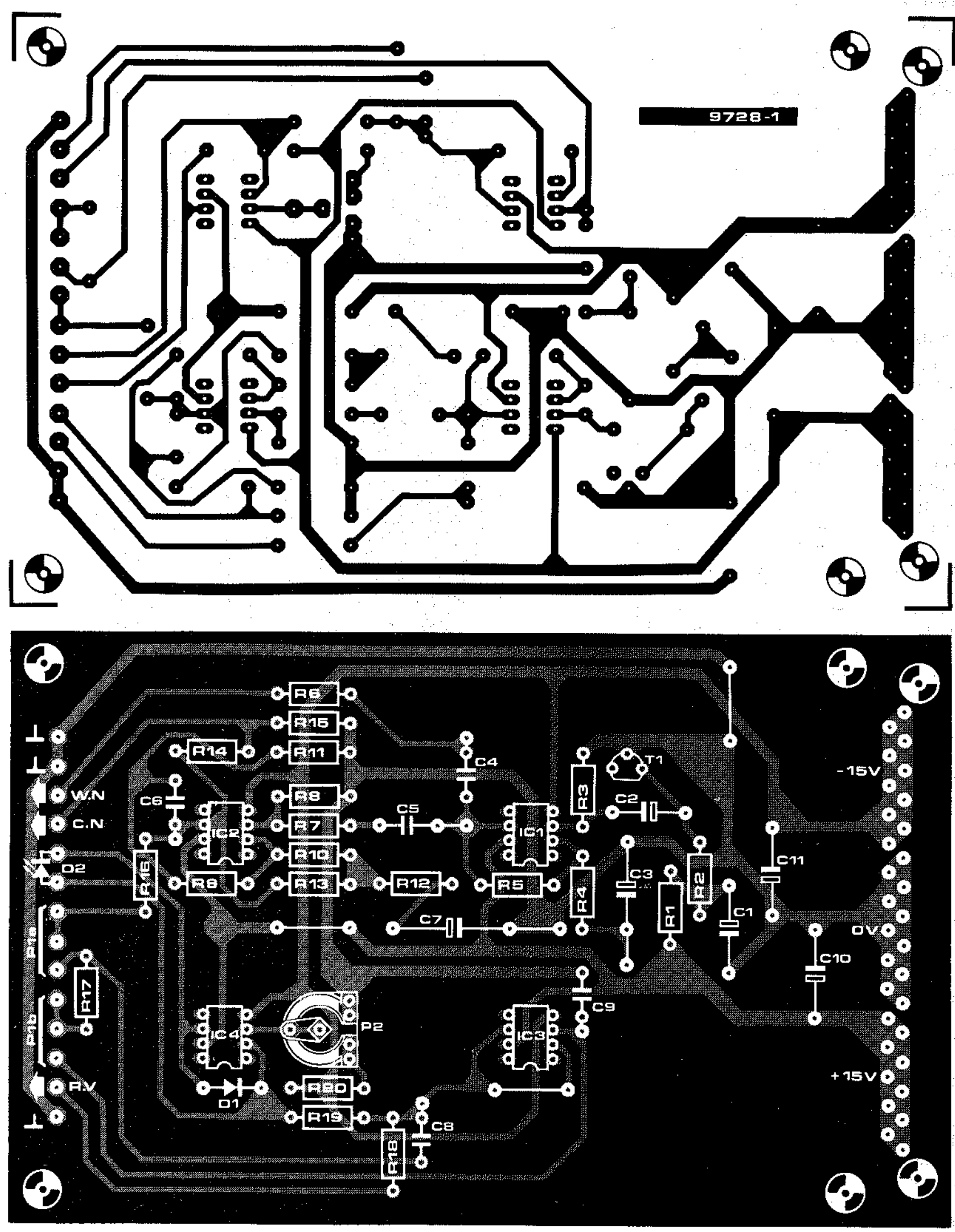

Parts list for noise module

Resistors:
R1,R9,R10,R13 = 47 k
R2 = 100 k (see text)
R3,R7,R8 = 470 k
R4 = 10 k
R5 = 2M2 (see text)
R6,R11,R18,R19 = 470 Ω
R12 = 4k7
R14,R15,R16,R17 = 1 k
R20 = 22 k

Capacitors:
C1 = 22 μ/25 V
C2 = 1 μ/16 V
C3 = 47 μ/35 V
C4 = 680 n
C5 = 1 μ (polyester or polycarbonate)
C6 = 330 n
C7 = 100 μ/35 V
C8,C9 = 220 n
C10,C11 = 10 μ/25 V

Semiconductors:
IC1,IC2,IC3,IC4, = μA 741C, MC 1741CP1 (Mini DIP)
T1 = TUN (selected)
D1 = 1N4148, 1N914
D2 = LED (e.g. TIL 209)

Potentiometers:
P1 = 100 k lin. ganged potentiometer
P2 = 100 k preset

Miscellaneous:
1 x transistor socket
1 x 31-way connector (DIN 41617)
3 x 3.5 mm jack sockets
1 x 13 . . . 15 mm knob

Adjustment of the LED indicator

P4 should be adjusted so that the brightness of the LED follows the amplitude of the triangle output, i.e. the LED should be at minimum brightness when the triangle voltage is at its most negative, and at maximum brightness when the triangle is at its most positive. P4 should adjusted so that the LED brightness does not reach maximum before the peak of the triangle, but on the other hand it should not extinguish completely before the trough of the triangle.

The noise module

The complete circuit of the noise module is shown in figure 5. The noise is produced by the base-emitter junction of an NPN transistor T1, which is reverse-biased. The noise is amplified to a level of about 2.5 V peak-to-peak. This white noise output is fed out via C4 and R6.

The white noise is also fed into a filter constructed around IC2, which has two frequency dependent elements in the feedback path. These two elements interact as follows. On its own, the feedback network comprising R10, R12, R13 and C7 would produce a 6 dB/octave rise in the gain of IC2, from 0 dB at zero Hz via 3 dB at 9 Hz to approximately 20 dB at 90 Hz. The feedback network R9, R11, C6, on its own would produce a 6 dB/octave fall in gain from 0 Hz to 1 kHz, above which the gain would remain constant at 0 dB.

The combined effect of these feedback networks is that below 90 Hz the 6 dB/octave rise and 6 dB/octave fall cancel out, giving a gain of 20 dB. Above 90 Hz the gain falls at 6 dB/octave to 0 dB at 1 kHz, above which it remains constant. The result is that the bass end of the noise spectrum is boosted, and 'coloured' noise is available at the output of IC2. The coloured noise output is taken from the junction of R14 and R15.

The coloured noise output is also fed to a second filter built around IC3. This is a 12 dB/octave lowpass filter with variable turnover frequency, which passes only the very low frequency components to produce an extremely low frequency 'random voltage'. The fluctuation rate of this random voltage is adjusted by means of P1, which varies the turnover frequency of the filter. Fluctuations of the random voltage are displayed on a LED indicator, which is identical to those used in the LFOs.

Construction and adjustment of the noise module

A printed circuit board and component layout for the noise module are given in figure 6, and the front panel layout is given in figure 7.

As not all transistors are suitable noise generators, a socket should be fitted in the T1 position on the board so that different transistors may be tried. Measuring with a multimeter on a suitable AC voltage range at the white noise output, a voltage of 0.5 V to 0.8 V should be present. Alternatively, if an oscilloscope is used to monitor the output, a noise signal of about 2 V to 2.8 V peak-to-peak should be obtained. It may be necessary to try several transistors before a suitable one is found. Varying the value of R2 between 33 k and 150 k may also help.

If the transistor produces too high a noise level this can be reduced by making R5 smaller, thus reducing the gain of IC1.

The amplitude of the coloured noise output should also lie in the same range as the amplitude of the white noise output. If it is too small then R7 should be reduced and if it is too large R7 should be increased.

The random voltage output should vary between about +2.5 V and −2.5 V with P1 in the 'fast' position.

The final adjustment to the noise module is to set P2 so that the LED brightness indicates the amplitude of the random voltage output in a linear manner. This adjustment is carried out in exactly the same way as the adjustment of the LFO indicators.

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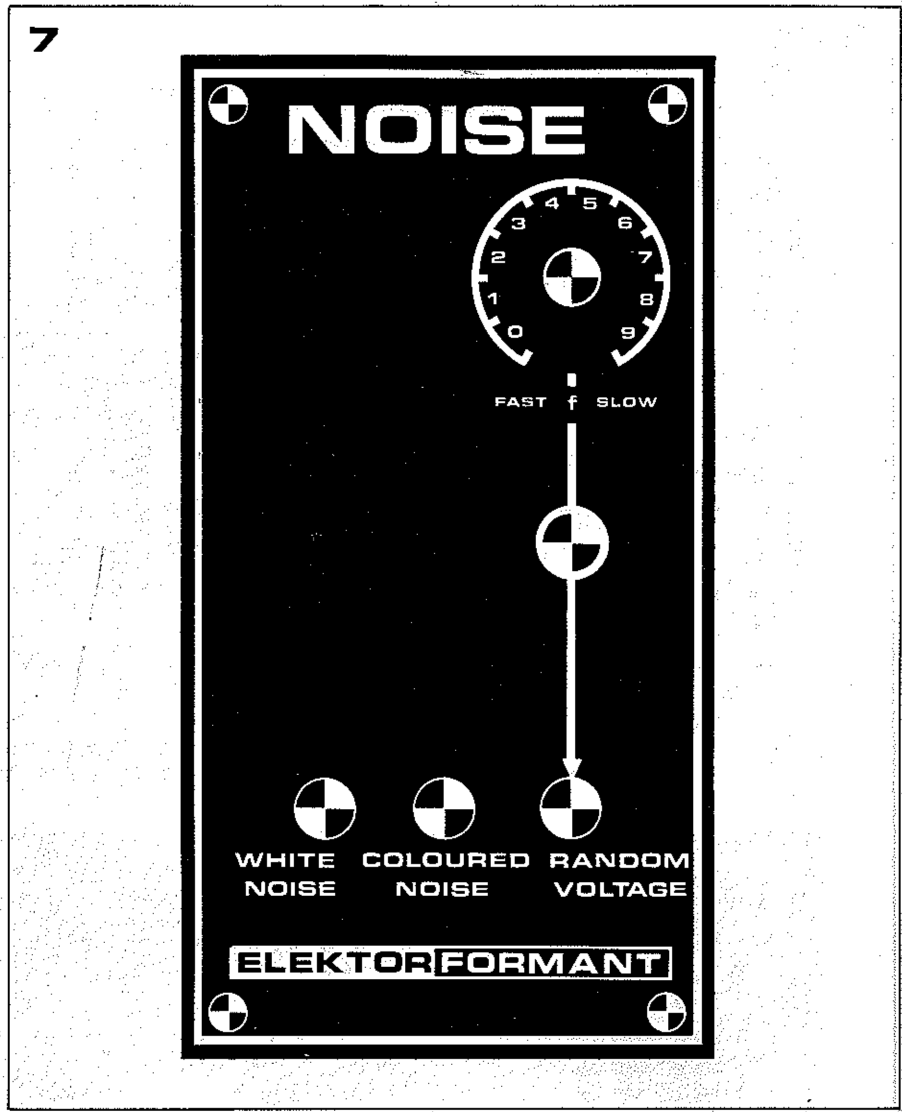


Figure 7. Front panel layout of the noise module.

chapter 12

COM

With a description of the COM (Control and Output Module), and an overall wiring diagram for the 'basic' Formant system, this chapter brings Part 1 of the book, which has dealt with the design and constructional aspects of the various Formant modules, to a close.

The COM contains a tone control amplifier with bass, middle, treble and volume controls, and an output buffer capable of driving high impedance (> 600 Ω) headphones for monitoring or practice purposes. The COM front panel also contains the indicator LEDs for the three power supply voltages and the gate signal. These indicators should not be regarded merely as a gimmick but as an important aid to monitoring the state of the Formant system. A fault in any of the supply voltages is immediately indicated by one of the LEDs, as is the absence of a gate pulse.

COM circuit

The complete circuit of the COM is given in figure 1a.

1b

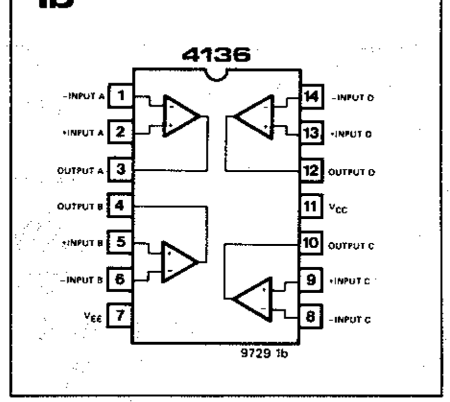


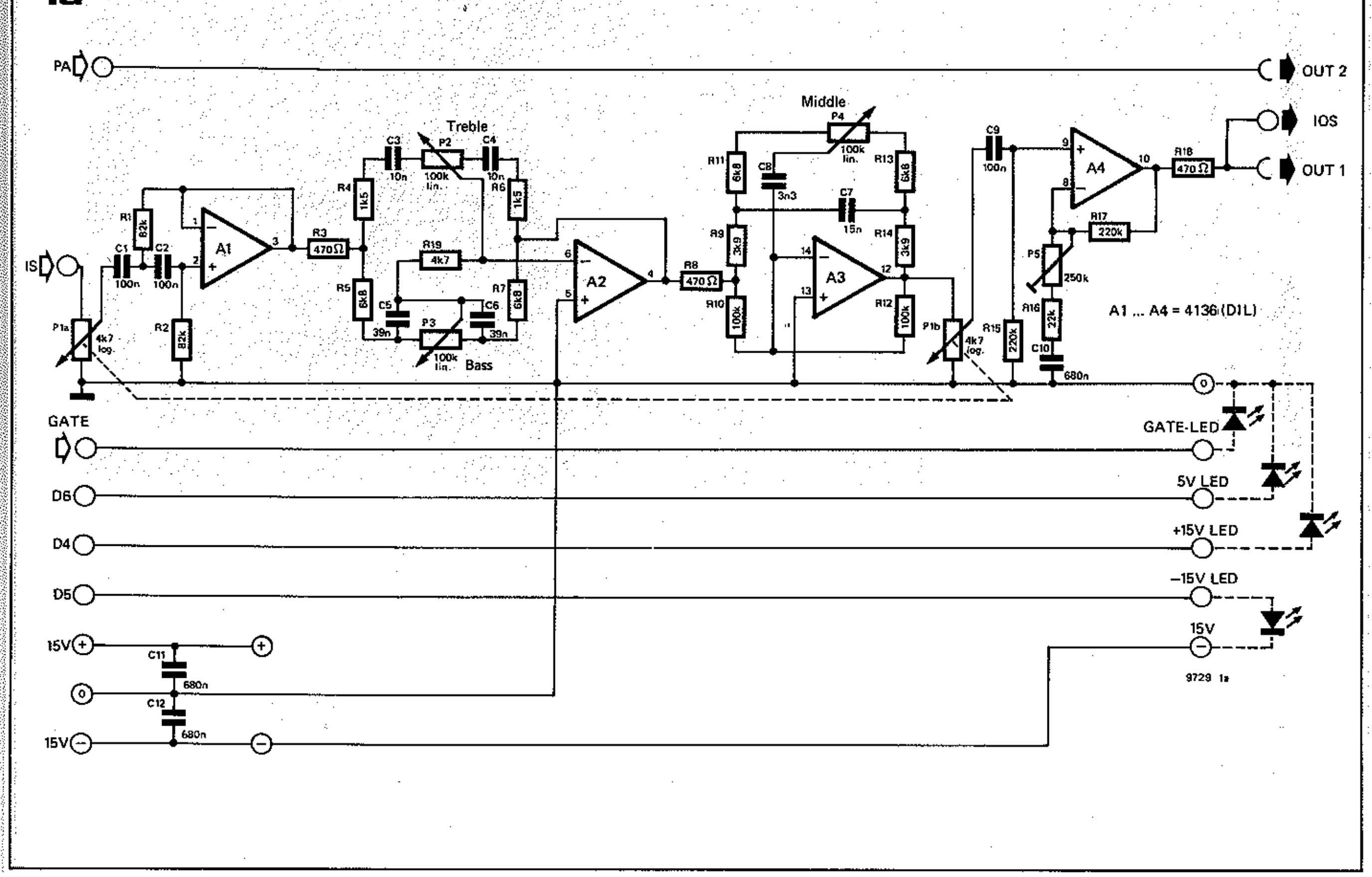
Figure 1a. Circuit diagram of the COM, which consists of a tone control/headphone amplifier and indicator LEDs for gate pulse and the three power supply voltages.

Figure 1b. Pinout of the 4136 IC.

The input signal is fed to a volume control P1a and thence to an 'anti-plop' filter built around A1. This is a 12 dB/octave highpass filter with a break frequency of around 20 Hz. It suppresses low-frequency transients and rolls off the bass response of the system to reduce 'listener fatigue' which can be caused by the low bass notes of electronic music, especially with full bass boost. By rolling off the bass response the filter also helps protect the bass drivers of the loudspeakers against excessive, very low-frequency signals. Indeed, if the synthesiser is to be used with small 'bookshelf' speakers it may be advisable to raise the turnover point of the filter to 40 Hz by changing the value of R1 and R2 to 39 k.

The treble and bass controls, built around A2, are a conventional Baxandall network. To avoid the middle control interacting with the bass and treble controls it is constructed separately around A3. The output of A3 then feeds into a second volume control P1b. The use of a ganged volume control on a single signal channel may seem a little unusual, but it does have several advantages. A volume control at the input to the COM prevents any possibility of overloading A1, whatever the signal level. On the other hand, the provision of a volume control later in the circuit allows a better signal-to-noise ratio to be maintained at low settings of the volume control, since noise (principally from A1) is attenuated along with the signal as the control is turned down. The fact that this control produces a 'double

1a



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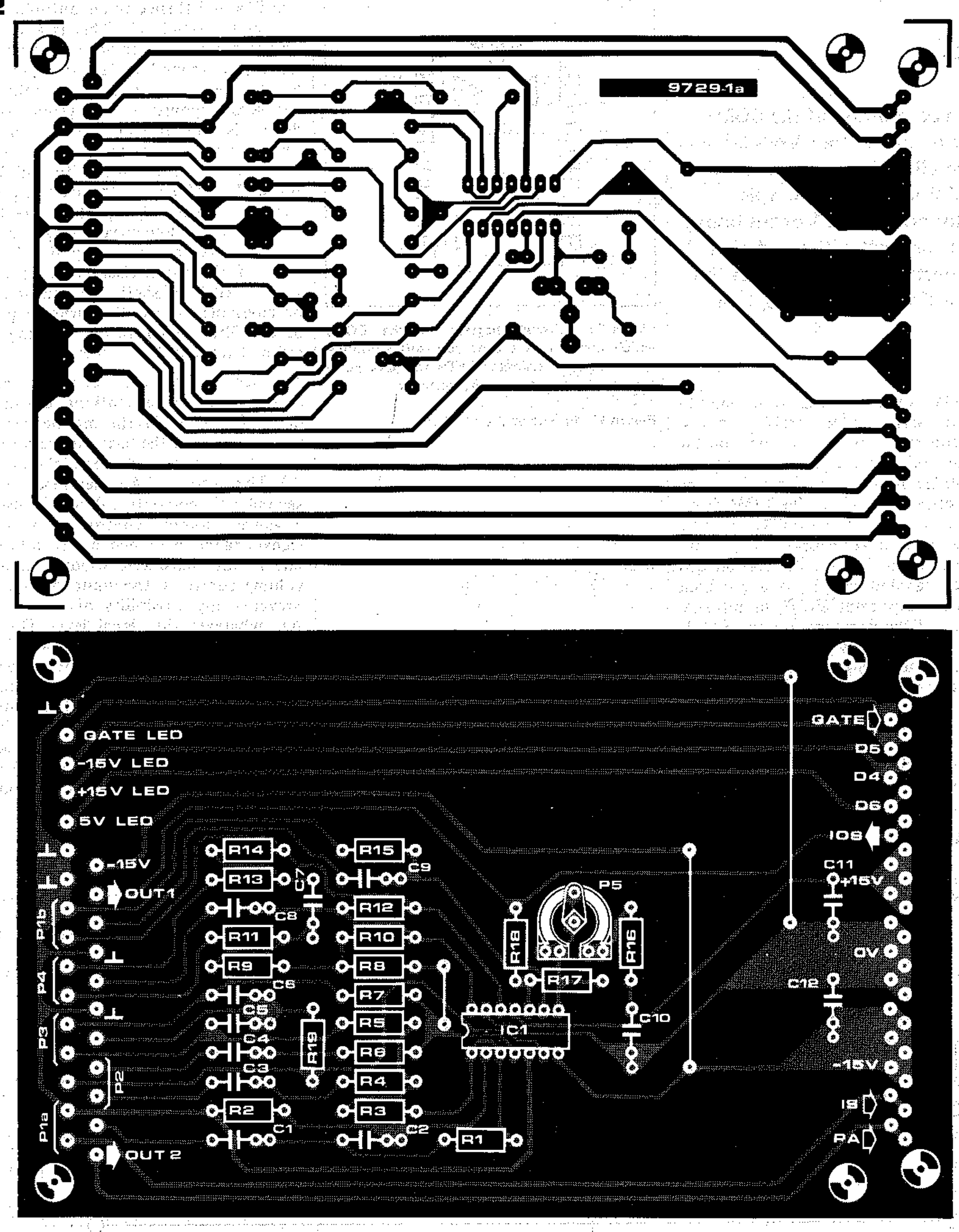

Parts list for figures 1 and 2.

Resistors:
R1,R2 = 82 k
R3,R8,R18 = 470 Ω
R4,R6 = 1k5
R5,R7,R11,R13 = 6k8
R9,R14 = 3k9
R10,R12 = 100 k
R15,R17 = 220 k
R16 = 22 k
R19 = 4k7

Potentiometers:
P1a,P1b = 4k7 log ganged pot.
P2,P3,P4 = 100 k lin.
P5 = 220... 270 k preset.

Capacitors:
C1,C2,C9 = 100 n
C3,C4 = 10 n
C5,C6 = 39 n
C7 = 15 n
C8 = 3n3
C10,C11,C12 = 680 n

Semiconductors:
IC1 = 4136 (DIL package) EXAR, Fairchild, Raytheon or Texas.

Miscellaneous:
31-way connector to DIN 41617
3.5 mm jack socket
6.3 mm jack socket
4 collet knobs, 13...15 mm diameter, with pointer.

Figure 2. Printed circuit board and component layout of the COM (EPS 9729–1).

Figure 3. Front panel layout for the COM (EPS 9729–2).

Figure 4. Wiring diagram for the front panel mounted components.

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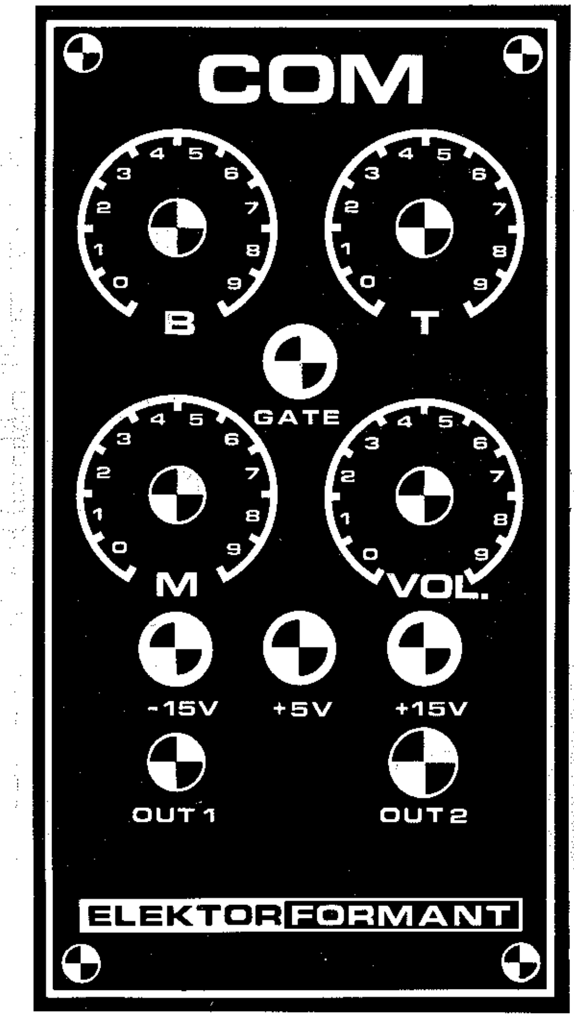

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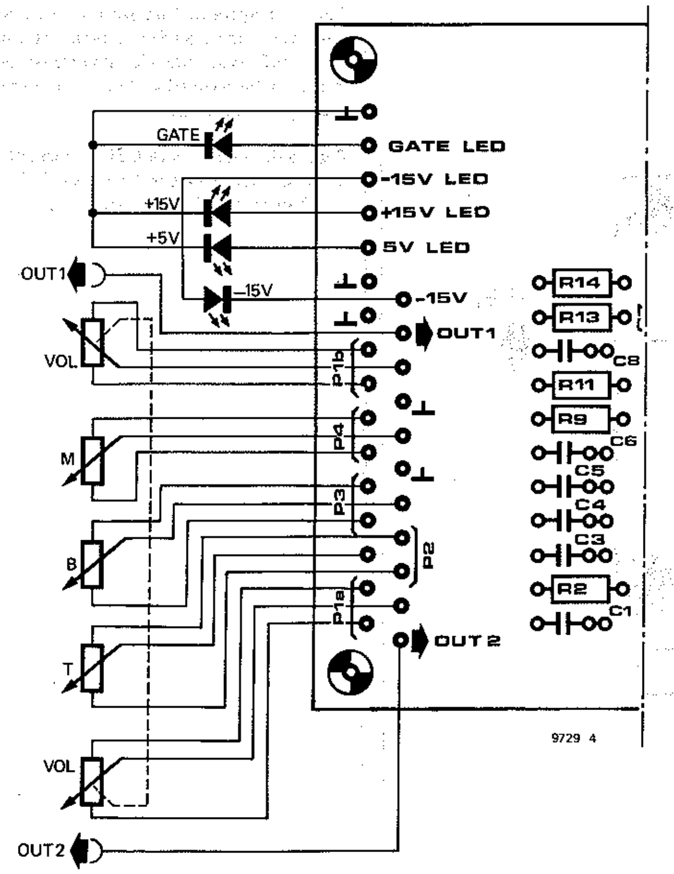

logarithmic' characteristic does not cause any inconvenience in operation.

No power amplifier is built into the COM as the heat generated in the output stage could cause temperature drift problems in other circuits in the system. However, the COM is provided with an internal output to a separate power amplifier, IOS. The output of the amplifier may then be brought back through the COM via the PA input connection on the COM board edge connector to a socket on the COM front panel (OUT 2). The COM output is itself also brought out to a socket on the front panel (OUT 1) into which high impedance headphones may be plugged. Note that a 6.3 mm jack socket is used for OUT 2. The four indicator LEDs also receive their power via the COM edge connector from the appropriate circuits, and are also mounted on the COM front panel.

Construction and testing of the COM

A printed circuit board and component layout for the COM are given in figure 2, a front panel design is given in figure 3 and wiring to front panel mounted components is shown in figure 4. Screened leads should be used for the connections to bass, middle and treble potentiometers B, M, and T.

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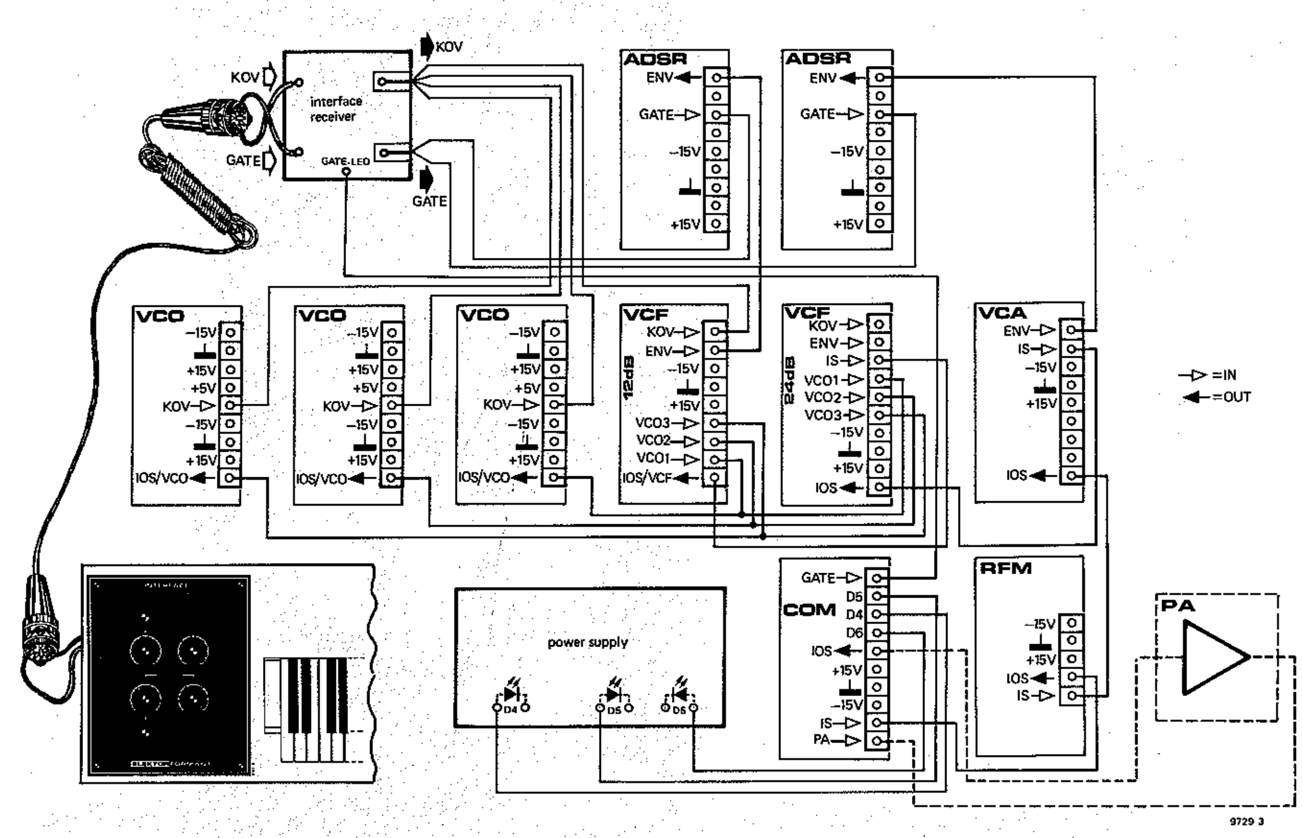

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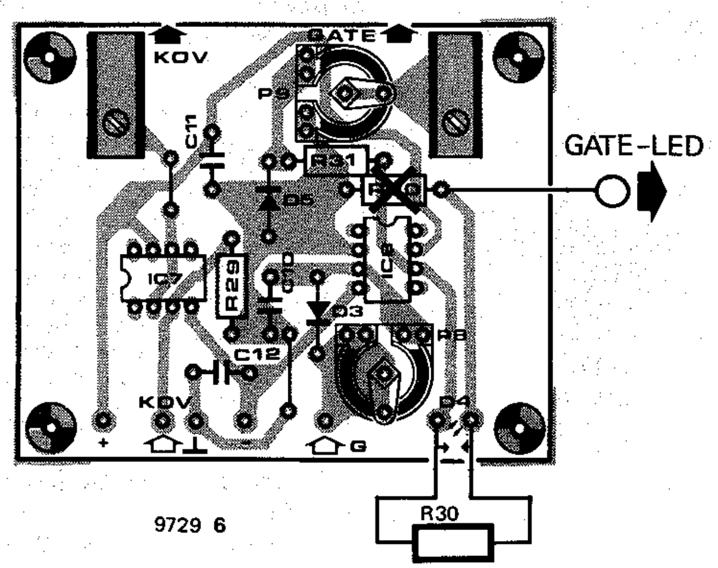

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Figure 5. Inter-module wiring for the basic Formant system. Supply voltage connections have been omitted for reasons of clarity. The LFO and noise modules have been omitted as the only hardwired connections they have are supply connections.

Figure 6. The 'gate-LED' output of the interface receiver can be simplified by mounting R30 in the 'D4' position.

Some readers may not wish to bring the output of a power amplifier back through the COM to output 2, since this may not be convenient especially if the synthesiser is to be used with, say, an existing hi-fi setup. In this case two options are open. Output sockets 1 and 2 can simply be connected in parallel or alternatively output socket 2 can be wired direct to input IS to provide an output signal unaffected by the tone and volume controls.

It is not intended to provide a design for an output power amplifier since several good designs have already been published in Elektor. However, a few hints on the mounting of such an amplifier will not go amiss. As mentioned earlier, the power amplifier should not be mounted in a plug-in module since it may then cause thermal problems. It should preferably be mounted at the back of the module cabinet with the output transistors mounted on heatsinks whose fins are external to the module housing. The Formant power supply is not intended to supply current for a power amplifier, so a separate power supply will be required. The mains transformer should be mounted as far away as possible from the Formant modules to reduce hum pickup (the same applies to the Formant mains transformer).

The COM can be tested by feeding in a signal from one of the VCOs and monitoring it on an oscilloscope to check that the waveform is undistorted. The gain of the COM output stage, A4, can be varied between about 1.8 and 11 by means of P5. This preset should be adjusted so that full drive of the headphones or power amplifier is obtained with the volume control turned fully up (clockwise).

Complete wiring diagram

The interwiring between modules for the basic Formant system is given in figure 5, but readers wishing to build a more extensive system can expand this as required.

For clarity the supply wiring is not shown, but the wiring method already mentioned must be adhered to, i.e. each module should have separate supply leads from its socket back to the 'star' connection points (busbars) on the power supply module. The temptation to simplify the wiring by simply linking between the supply pins of the modules should be avoided as this will cause interaction between modules.

The 'Noise' and 'LFOs' modules are not shown in figure 5, since the supply wiring is the only connection to these modules.

Again for clarity, the full pinout of each module edge connector is not shown, but the connections are shown in the correct sequence working down from the top edge of each module.

One small modification is required to the interface receiver printed circuit board (chapter 3) in order that the gate LED can be wired with only a single link. R30 on the interface receiver board is mounted in the space provided for D4 as shown in figure 6. A single wire is then connected from the lower pad to which R30 was originally connected to the appropriate pin of the COM socket. Without this modification two leads would have to be brought out to D4.

Patchcords

Due to the hardwired interconnections between modules, Formant is perfectly playable without any of the front panel patching sockets being used. However, for effects such as vibrato and tremolo, patchcords are used to connect the outputs of the LFO module to the VCOs or VCA. These can easily be 'home-made' – see chapter 1 of Part 2.