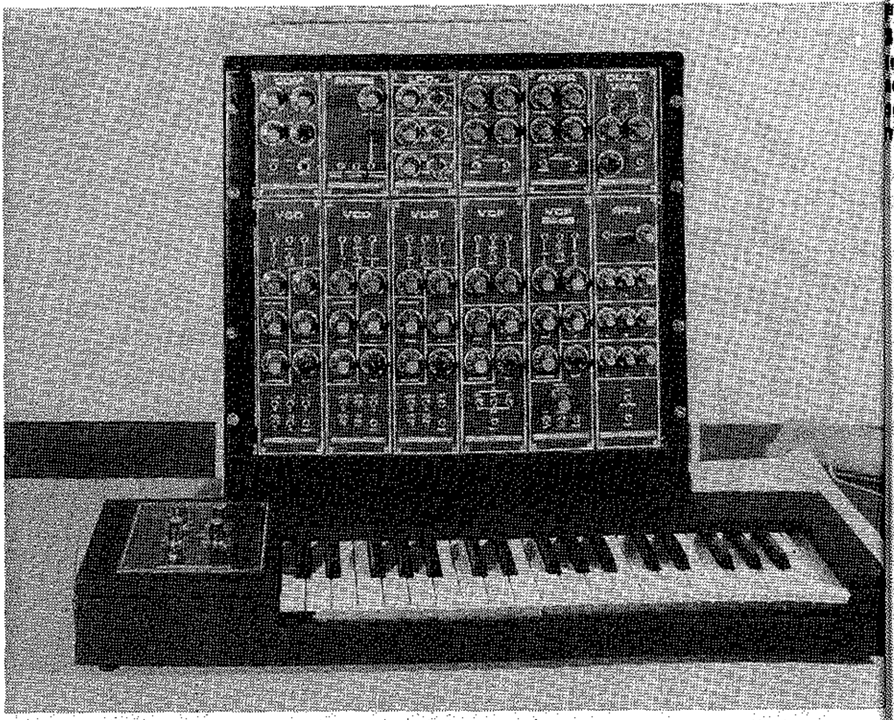
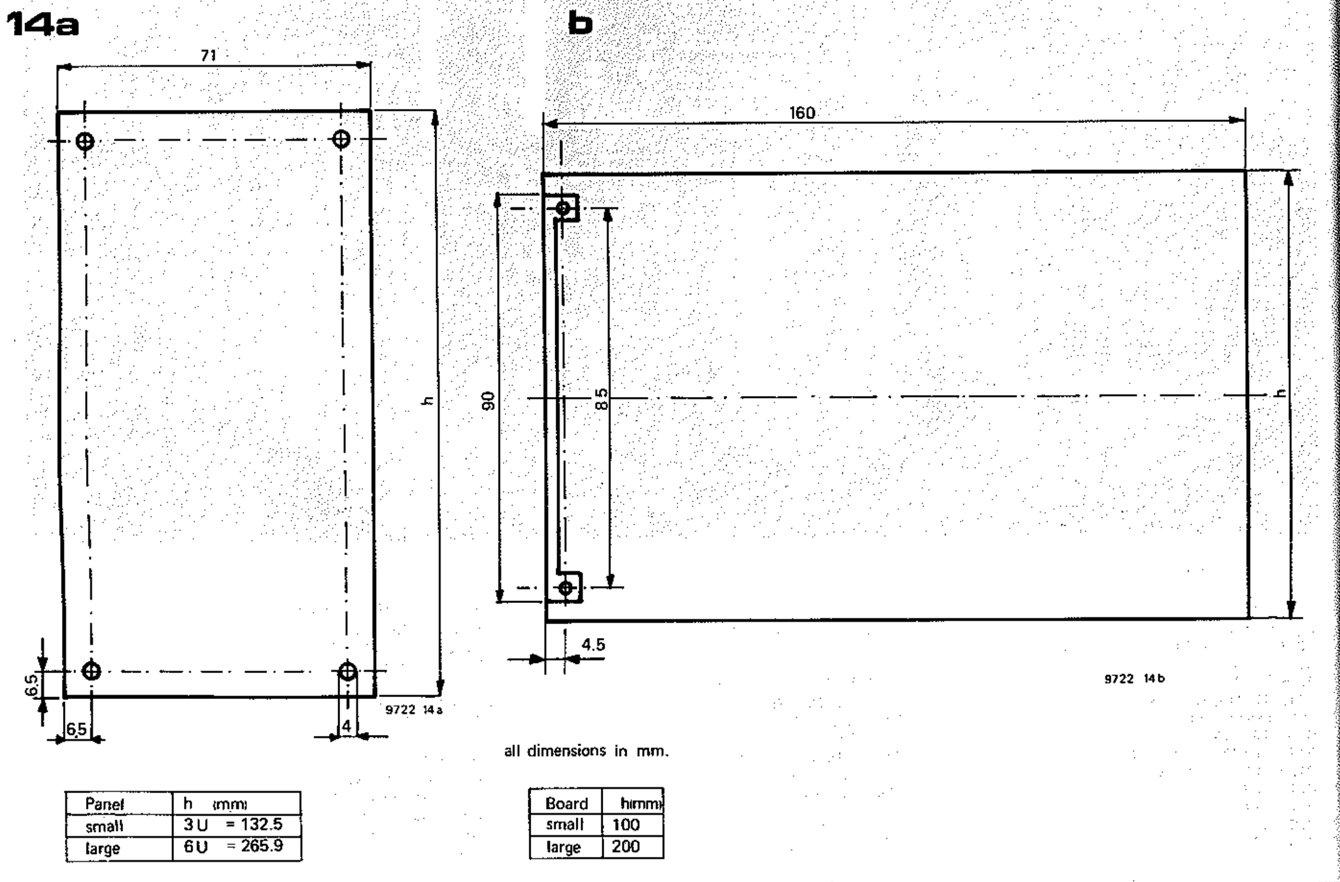


**Figure 13. A suggested layout for a 'basic' synthesiser.**



| Panel | h (mm) |
|---|---|
| small | 3 U = 132.5 |
| large | 6 U = 265.9 |

| Board | h(mm) |
|---|---|
| small | 100 |
| large | 200 |

**Figure 14. The dimensions of the Formant modules are compatible with the Eurocard rack system.**

power supply and the power transistors should be mounted on generous heatsinks, for example finned heatsinks of 100 mm x 50 mm with 30 mm high fins. The AC supplies to the stabilisers may be provided by a single transformer with multiple secondary windings (if available) or by a number of smaller transformers. In either case the transformer(s) should be generously rated, the one amp secondary current specified being the minimum acceptable.

Power supply connections to the voltage-controlled modules will be taken from the power supply by separate wires to each module. For this reason each power supply rail is equipped with a substantial connection 'busbar'. These are made from copper strip or pieces of copper laminate board, and are soldered to terminal pins pushed through the p.c. board. This arrangement can clearly be seen in photo 4.

Once the power supply unit has been built the output voltages can be set to their correct values. The –15 V supply should be adjusted to within 1% of its nominal value using a DVM, since the accuracy of this supply voltage has a direct bearing on the volts/octave characteristic of the keyboard. The +15 V and +5 V supplies need only be set to within 3% of their nominal values.

## Keyboard calibration

Once the synthesiser's own power supply has been tested and adjusted the offset compensation and volts per octave characteristic of the keyboard can b adjusted. The keyboard interface, inte face receiver and power supply ar connected as shown in figure 10.

### Offset compensation

The overall tuning is switched off (S 'off', i.e. position b). Depress the lowe key of the keyboard and hold it dow while adjusting P4 so that the KO output of the interface receiver is zero.

### Volts/octave characteristic

This should be adjusted to an accurac of at least 1% using a DVM. The overall tuning remains switched off. The KOV output is measured and P6 is adjusted so that alternately depressing keys one octave apart causes the KOV output to change by exactly one volt. The Formant keyboard is now compatible with any synthesiser that uses a standard 1 V/octave keyboard.

Finally, the offset compensation should again be checked and readjusted if necessary.

15

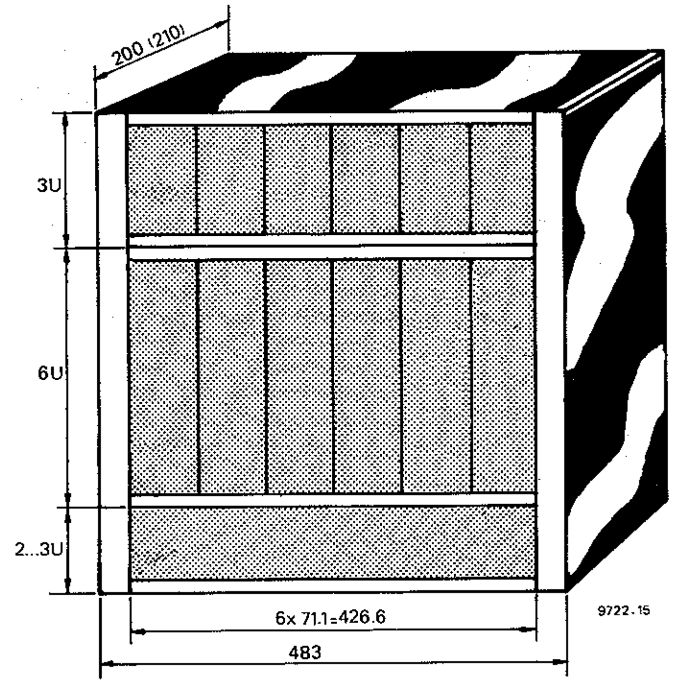


**Figure 15. A case containing one 6U and one 3U rack will accommodate six large and six small Formant modules. It can be useful to add a 2U or 3U bottom panel (using a larger case!), behind which amplifiers etc. can be mounted.**

### Gate delay

Accurate adjustment of the gate delay is not possible until the voltage controlled modules have been constructed, but an approximate adjustment will suffice until that time. P7 on the interface board should be set to about one quarter of its maximum resistance, and P8 on the receiver board should be set to minimum.

## Modular construction

A modular method of construction was chosen because it allowed the greatest flexibility in the final design. Each voltage-controlled circuit is constructed on its own p.c. board which plugs into a socket in the module housing that supplies power, control voltage and gate pulses. Interconnections between modules are made by means of patch cords.

The advantage of this system is that the synthesiser can be made as simple or as complex as is required. Provided sufficient space is left in the module housing for additional modules, it is possible to build a playable instrument with just a small number of modules, and to extend it as and when desired. This also means that every instrument can be tailored to the individual constructor's taste and is not fixed within rigid limits set by the designer. However, for those who require a little more guidance as to the right 'mix' of modules that should be adopted, a suggestion for a 'middle-of-the-road' instrument is given in figure 13. This utilises three VCOs, one 12 dB VCF, one 24 dB VCF, one RFM, one DUAL VCA, two ADSR envelope shapers, one LFO module, one NOISE module and one COM. Some readers may regard the extra (24 dB) VCF and RFM as slight luxuries, and indeed for the beginner or someone with a slightly limited budget, these modules could be initially omitted. However they do considerably enhance the tone-shaping capabilities of the Formant, and for this reason can justifiably be included in the 'basic' Formant system.

The module printed circuit boards and front panels are compatible with the Eurocard rack system. Two module heights are employed in Formant. A double-height (6U) module is used for the voltage-controlled modules (VCO's, VCA's and VCF's) while a single-height (3U) module is used for the ancillary circuits (envelope, shapers, noise generator etc.).

The basic dimensions of the modules are given in figure 14. The Eurocard rack system operates on a card spacing of 5.08 mm (0.2") or multiples thereof. Each Formant module occupies a panel width of approx. 71 mm, so the 426.7 of panel width available will accommodate six modules. A 6U rack and a 3U rack stacked together will thus accommodate six large and six small modules, as shown in figure 15. This corresponds exactly to the no. of modules in the 'basic' Formant system.

Of course some readers, especially those with previous experience of synthesisers, may already have a firm idea of the type of instrument they wish to build, and may like to construct a purpose-built case of wood or some other material. This is quite permissible, as the module housing does not require screening.

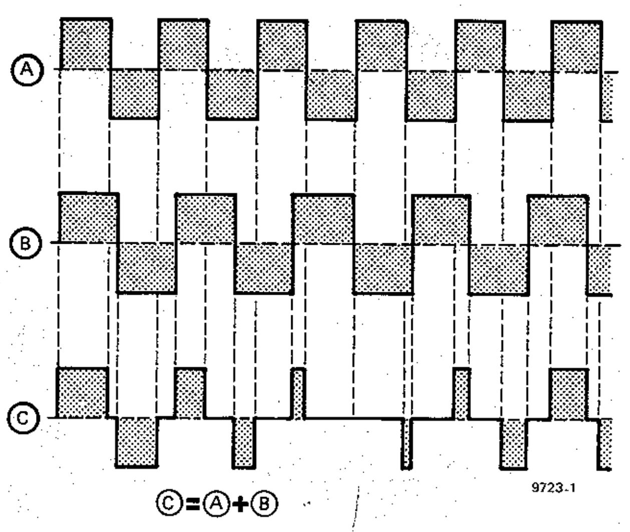
chapter 4

# voltage controlled oscillator

**The voltage controlled oscillators (VCOs) are the heart of any synthesiser. The quality of the VCOs ultimately determines the performance of the synthesiser. For this reason the next two chapters are devoted to their design and construction.**

The two principal requirements of a synthesiser VCO are stability and good tracking. Stability means that if the control voltage applied to the VCO remains constant, then the frequency of the VCO should also remain constant and not drift. Tracking means that the VCO must follow the prescribed logarithmic 1 octave/V characteristic as closely as possible. In particular, where several VCOs are used they should all have similar characteristics.

1



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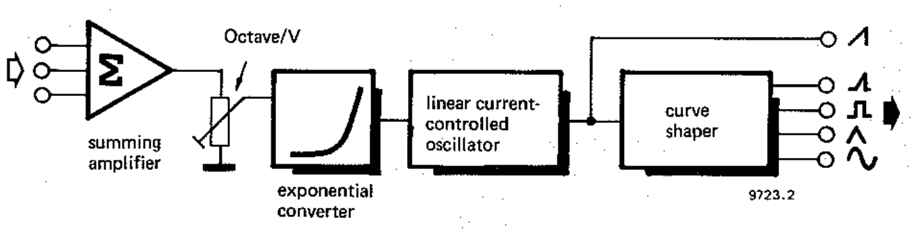


These parameters are particular important in a chording instrument suc as the Formant, where a number c VCOs are used simultaneously. In synthesiser using only one VCO sligl drift or deviation from the 1 octave/ characteristic might not be noticed since the ear is not particularly good judging absolute frequency, unless person has 'perfect pitch'. In an chording instrument however, eve slight mistuning is immediately apparen due to the formation of beat notes.

For example, if two or more VCOs ar tuned to the same pitch any sligh mistuning is audible as beat notes havin a frequency equal to the differenc between the two VCO frequencies. Slight mistuning between VCOs frequently employed deliberately. If th degree of mistuning is slight the bea frequencies are low and beat notes ar not audible, but a pleasing chorus c phasing effect is obtained, especially if several VCOs are used. This imparts a much more lively character to the sound which contrasts with the sterile sound of fixed phase instruments such as electronic organs based on a divider system (see figure 1).

However, if the VCO frequencies drift apart due to poor stability the beat notes quickly become obtrusive and unpleasant, and ultimately a discord is audible. A similar effect can be noted when the tracking of the VCOs is poor. If a chord is set up at a particular pitch then the musical intervals in the chord should be maintained when the chord is transposed to a different pitch. However, if the tracking of the VCOs is poor this will not be the case and a discord will result.

A good test of the VCOs in a synthesiser is thus to tune them together so that no beat notes are audible and check that the tuning is maintained over a period of time and with changes in such parameters as supply voltage, temperature etc. The tuning between the VCOs should also be maintained when the pitch is transposed.

Any VCO which cannot meet these criteria is useless for a synthesiser, and the design of a suitable synthesiser VCO is necessarily quite complex.

3

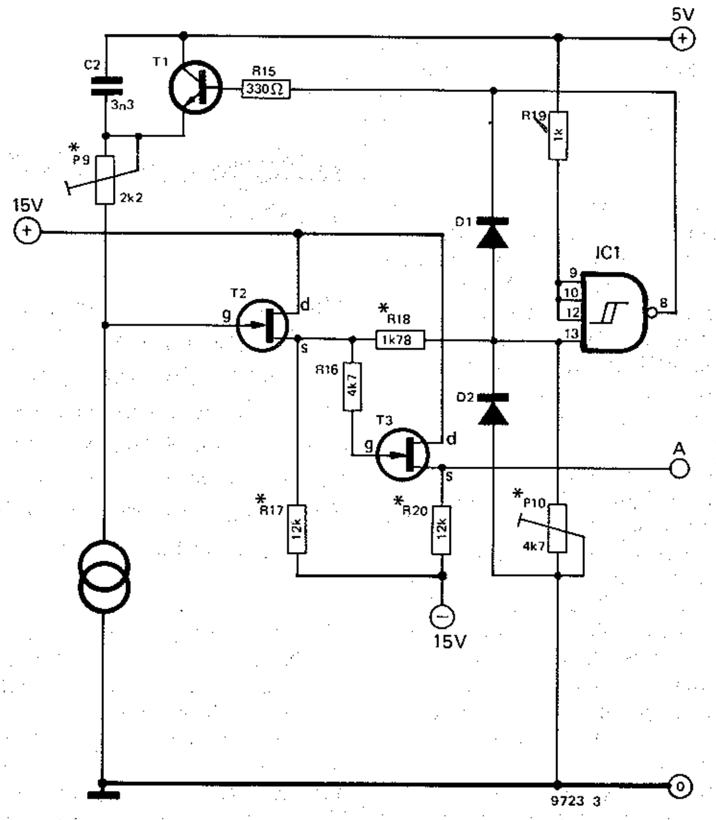


**Figure 1. When two notes of almost the sam frequency are played together, beat notes ar formed which produce a pleasing 'choru effect.**

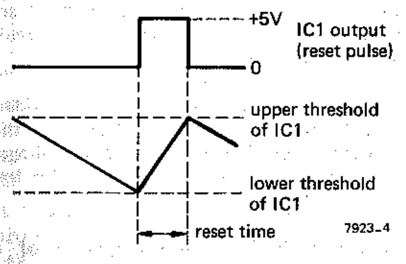
**Figure 2. Block diagram of the VCO, whic comprises an input summing amplifie exponential voltage-current converter, line current controlled oscillator and curve shap circuits.**

**Figure 3. The linear CCO is the heart of th VCO module. C2 charges linearly to the lowe threshold of IC1 before being discharged b T1, thus producing a sawtooth outpu waveform. The output of the exponenti converter, which determines the chargin current and hence the CCO frequency, represented by the current source symbol.**

**Figure 4. Detail of the sawtooth wavefor and the output of IC4 at the reset poi where T1 is turned on.**

**Figure 5. The exponential relationship b tween base-emitter voltage and collecto current of a bipolar transistor is exploited the exponential generator.**

4



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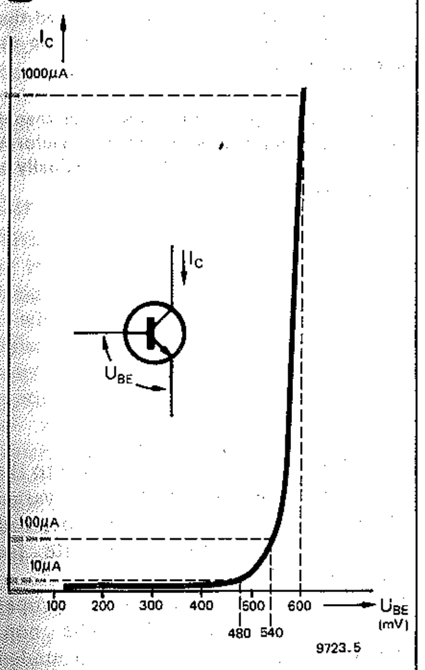


## Block diagram

The VCO circuit used in the Formant follows the form proposed first by Robert Moog (figure 2). The VCO input stage consists of a summing amplifier into which a number of control voltages may be fed. A potentiometer on its output sets the octaves/volt characteristic of the VCO. The resulting control voltage is fed to an exponential voltage-current converter, the output current of which doubles for every 1 V rise in input voltage. The output of this converter controls a linear current-controlled oscillator, which produces a sawtooth waveform. Finally, a curve shaper connected to the sawtooth output delivers four further waveforms: spaced sawtooth, squarewave, triangle and sinewave.

## Oscillator section

The CCO is the heart of the VCO circuit, as explained above.

The CCO section is shown in figure 3. The output of the exponential voltage-current converter that feeds this section is represented by the current source symbol at the bottom left of the diagram. This current is of course varied by the control voltage applied to the exponential converter.

FETs T2 and T3 are connected as source followers; their high input resistance ensures that no significant current is robbed from the current source, even at low currents, as this would spoil the sawtooth linearity and could affect the current-frequency linearity of the CCO. IC1 is a Schmitt trigger that senses when the sawtooth voltage has reached a predetermined level.

The circuit functions as follows: assume that initially C2 is discharged. The voltage at the gate of T2 will then be nearly +5 V, and since T2 operates as source-follower the voltage at the input of IC1 will be above the positive trigger threshold of this Schmitt trigger, so its output is low and T1 is turned off. As C2 charges from the current source the gate voltage of T2 will fall as the voltage across the capacitor increases. Since C2 is being charged from a constant current source, the voltage across it will increase linearly with time, in accordance with the equation

$$U_{C_2} = \frac{I.t}{C_2}.$$

When the voltage at the input of IC1 has fallen below its negative switching threshold the output of IC1 will go high, which will turn on T1 and discharge C2 until the input voltage of IC1 has risen above its positive threshold, when T1 will turn off and the whole cycle will repeat. A detail of the IC1 output and input waveforms during the discharge of C2 is shown in figure 4.

FET T3 is simply an output buffer stage. As mentioned earlier, the use of two buffer stages in cascade ensures that any load on the output cannot affect the linearity or frequency stability of the CCO.

The setting of P9 affects the high-frequency linearity of the CCO and is used to set the VCO tracking at high frequencies.

Since N-channel FETs are used for the source-follower buffers, the source voltage is always slightly positive with respect to the gate voltage, so that even when the gate of T2 is at zero volts there is always a slight positive voltage on the source. If the source of T2 were connected direct to the input of IC1 it would be possible that the source voltage of T2 (minimum, depending on FET tolerances, typically 1 V) might never fall below the negative threshold of IC1 (typically 0.85 V). For this reason T2 is connected to the input of IC1 via a potential divider comprising R18 and P10, the latter being adjusted to ensure that the oscillator starts reliably.

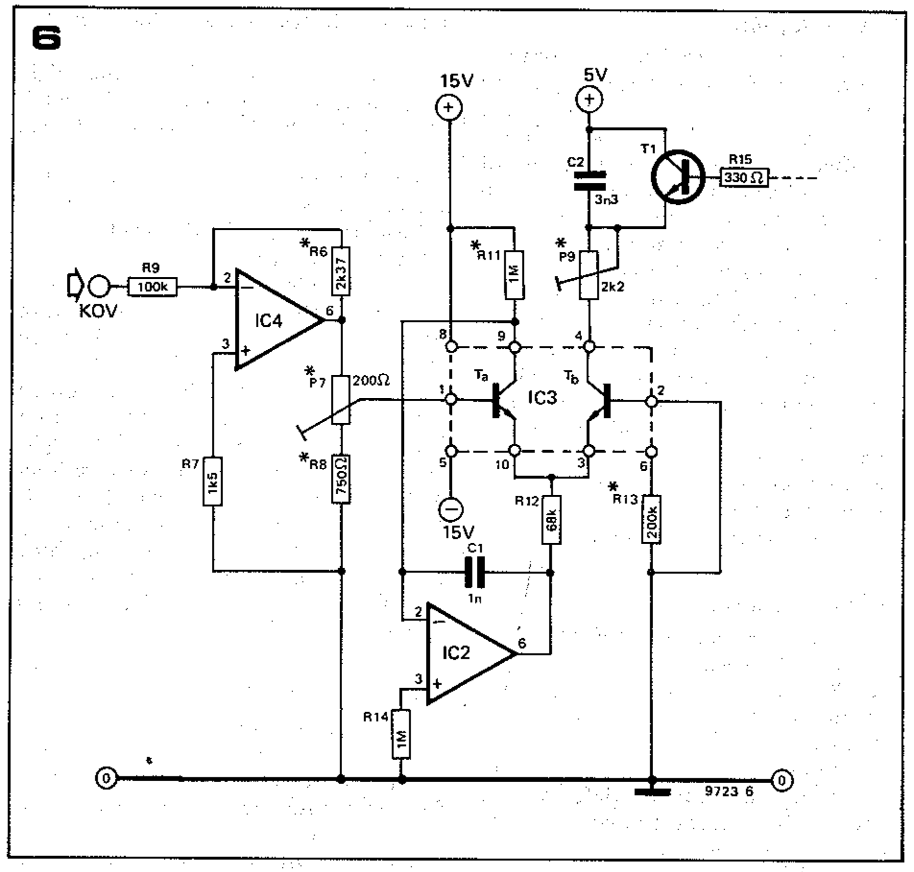
## The exponential converter

The exponential voltage-current converter doubles the output current fed to the CCO, and hence the CCO frequency, for every 1 V increase in control voltage.

In common with most such circuits, the exponential converter makes use of the (exponential) collector current versus base-emitter voltage characteristic of a bipolar transistor. Every transistor exhibits this exponential relationship, but not all transistors are suitable for use in exponentiator circuits. The reason is that collector leakage current can cause a deviation from the characteristic at low collector currents, and transistor base resistance can cause a deviation at high collector currents.

Special transistors for such applications are available, but even these have their limitations due to temperature dependence of the collector current. At around room temperature, collector current doubles for a $V_{be}$ increase of around 17 mV. However, a temperature increase of around 10°C will also double the collector current, so it is apparent that, unless some form of temperature compensation is employed, even small temperature changes will cause noticeable variations in the pitch of the VCO.

There are two methods of reducing the influence of changes in (ambient) temperature, both of which are used in the Formant VCO. The first of these is to use a matched pair of transistors in the exponential converter, one of which is used for temperature compensation. The second method is to keep the chip temperature of the transistors constant. By employing both methods absolute accuracy and stability of the exponential converter are achieved. Temperature stabilisation of the chip may sound like a complicated business, but fortunately a purpose-built IC is available, the μA726. It consists of two matched NPN transistors and also contains a tempera-



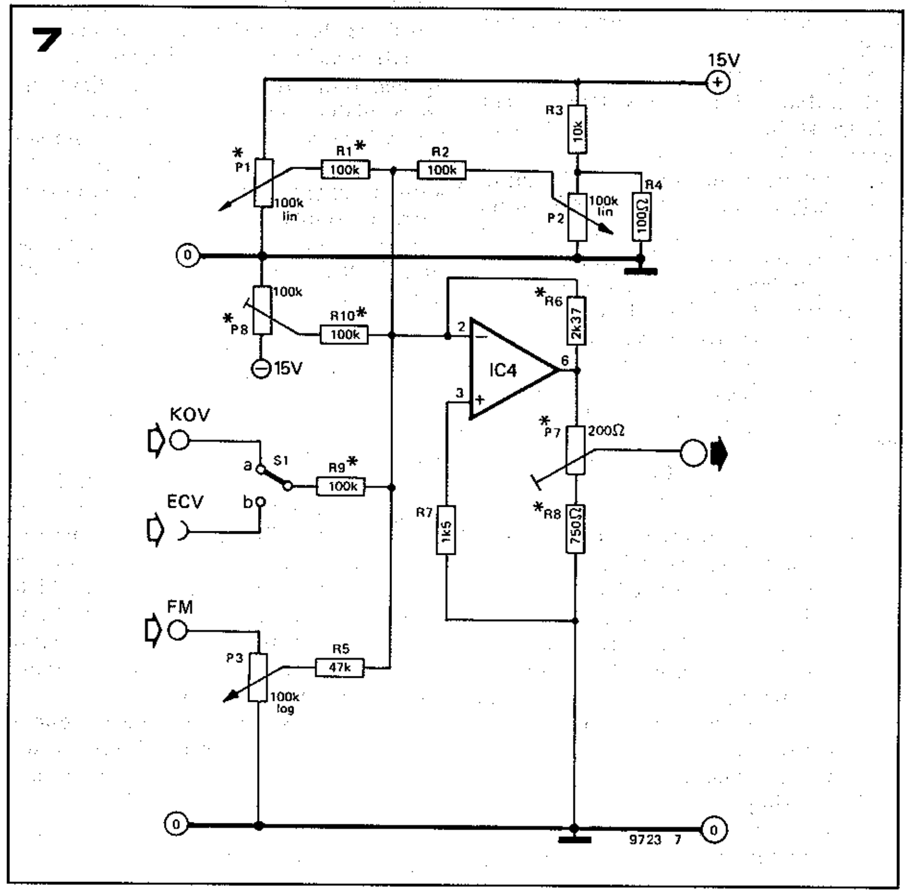


**Figure 6. Circuit of the exponential voltag current converter, which is both temperatu stabilised and compensated. IC4 and T1 a respectively parts of the input adder circu and the CCO.**

**Figure 7. Complete circuit of the input adde This will sum input control voltages from th keyboard or ECV socket, DC offset voltag for chording, and AC input signals for fr quency modulation of the VCO.**

**Figures 8 and 9. The musical quality of waveform depends on the harmonic conten The harmonic structure of two well-know waveforms is shown: sawtooth (figure 8) an squarewave (figure 9). In order to obtain th widest range of sounds from the Forma VCO, curve shaper circuits are provided th produce four waveforms in addition to th basic sawtooth.**

**Figure 10. Block diagram of the curve shape An output adder allows the various waveform to be fed to the output either individually in combination.**

ture control circuit that maintains a constant chip temperature.

The circuit of the exponential converter is given in figure 6. IC4 is not strictly part of the converter but is part of the summing amplifier section. At the operating temperature of the 726 a $V_{be}$ increase of between 19 and 23 mV is required for each doubling of collector current, so the 1 V/octave output of the keyboard must be attenuated.

IC4 is connected as an inverting amplifier with a gain of −0.0237. Since the KOV input is always positive the output of IC4 will always be negative, and will give an output of −23.7 mV per volt input. P7 allows the input voltage to th exponential converter to be varie between −18.7 and −23.7 mV per vo input, in order to compensate fo tolerances in IC3.

The exponential converter prope comprises IC2 and IC3. The no inverting input of IC2 is grounde through R14, so the inverting inpu should also be at (virtual) earth poten tial. For this to be the case, a constan current of 15 μA must flow throug R11, i.e. the collector current of T must be constant at 15 μA. The voltage to-current conversion can now b explained as follows.

If the input voltage KOV is increased b 1 V then the base voltage of $T_a$ will fa by around 20 mV (depending on th setting of P7). Since the collecto current of $T_a$ cannot decrease th output voltage of IC2 must fall in orde to reduce the emitter voltage of $T_a$ b 20 mV, maintaining the same base emitter voltage and thus the sam collector current. As the base of $T_b$ i grounded this means that the base emitter voltage of $T_b$ will fall by 20 mV and the collector current of $T_b$ wil double. The collector of $T_b$ is connecte to P9 in the CCO circuit, as shown i the top right hand corner of figure 6.

## Summing amplifier

The summing amplifier, part of whic was shown in figure 6, is given in it complete form in figure 7. Point KOV is permanently connected to th 1 V/octave output of the keyboar interface receiver, but the input of th summing amplifier can be switched between this point and an externa ontrol voltage socket (ECV). Poten-iometers P1 and P2 give coarse and fine djustment of a DC offset voltage to transpose the VCO pitch for setting up chords etc. Preset P8 is also provided as n offset control that compensates for the input offset voltage of IC4, and sets the lowest frequency of the VCO (around 15 Hz).

A frequency modulation (FM) input is provided, which can be fed with external (AC) signals to give vibrato effects etc. The modulation depth can be adjusted by P3, the maximum sensitivity being about 2 octaves/V with P3 turned fully clockwise.

As previously mentioned, the summing amplifier actually has a gain much less than one, so that the output voltage of IC4 is reduced to −23.7 mV per volt input.

## Curve shapers

Having ensured that the 'business end' of the VCO design is satisfactory, the design of the curve shaper section – which influences the musical characteristics of the VCO – may now be considered. The main processing of the synthesiser waveforms is done by means of voltage-controlled filters (VCFs) which remove certain frequencies from a harmonically rich waveform.

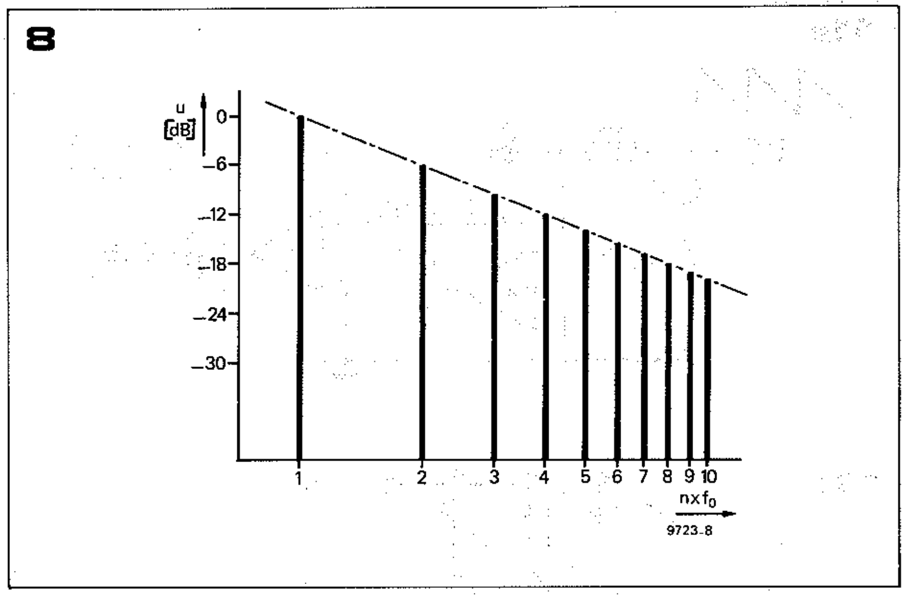
The spectra of two well-known harmonically rich waveforms are shown in figures 8 and 9 – the sawtooth, which contains all the odd and even harmonics of the fundamental, and the squarewave, which contains only the odd harmonics. However, this approach does have its limitations if only one waveform is provided at the VCO output. Using as an example the two waveforms just mentioned; no amount of filtering will generate the even harmonics necessary to turn a squarewave into a sawtooth, and it would be very difficult to filter out all the even harmonics from a sawtooth to make a squarewave. It is thus obviously useful to have several different waveforms available at the VCO output.

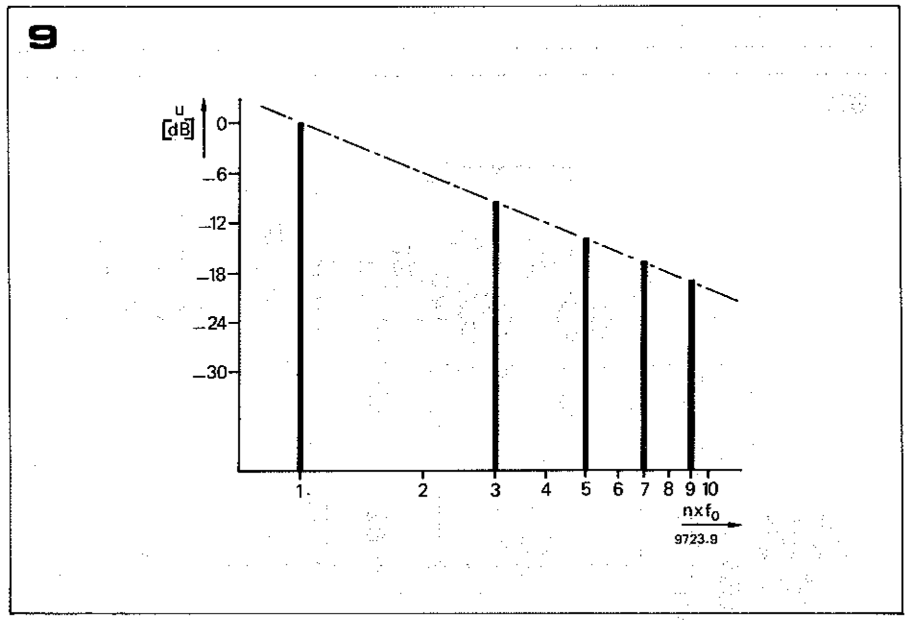
A block diagram of the curve shaper is shown in figure 10. The sawtooth output of the VCO is fed to curve shaper circuits, which produce respectively spaced sawtooth, triangle, sine and square waveforms. The pulse width of the squarewave may be modulated by an external control signal, as will be explained in the description of this part of the circuit.

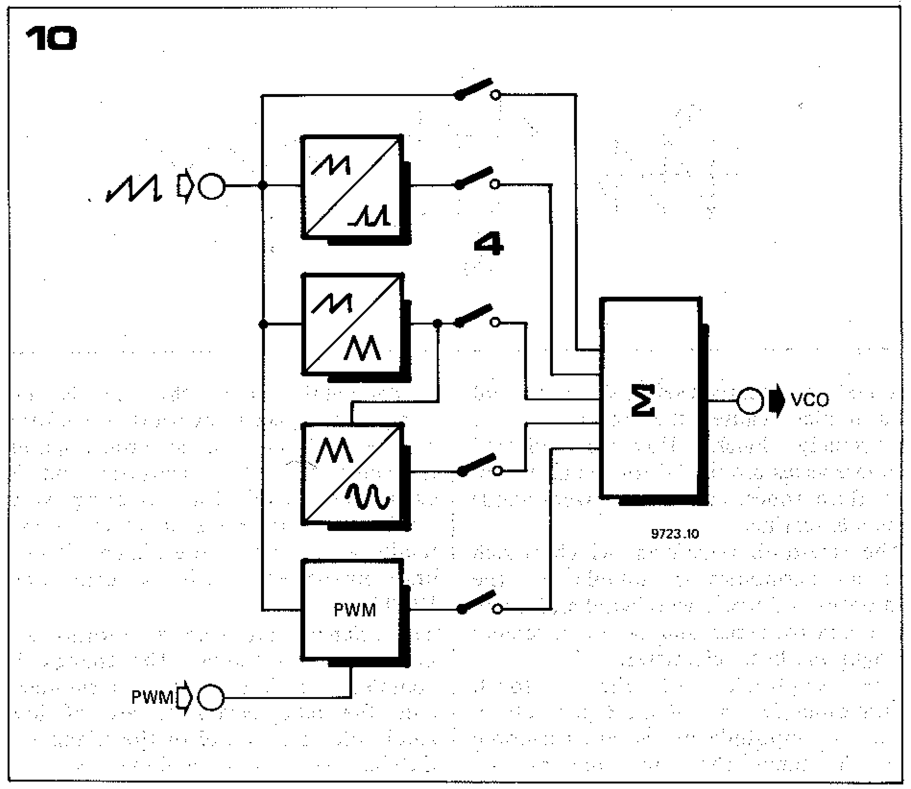
The five waveforms may be selected by means of switches to be fed, either singly or in combination, into a summing amplifier.

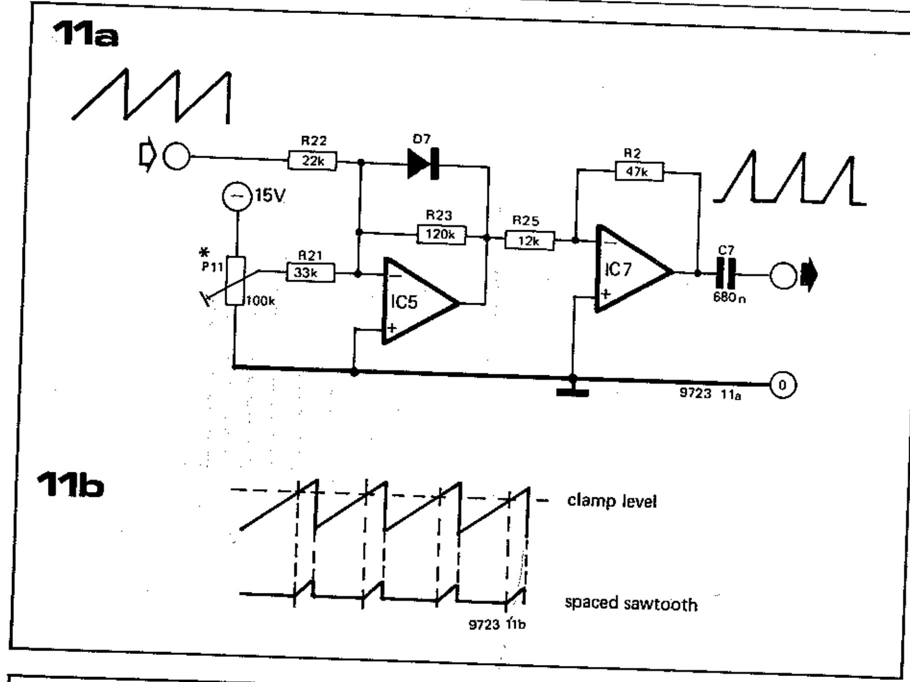
## Musical properties of the waveforms

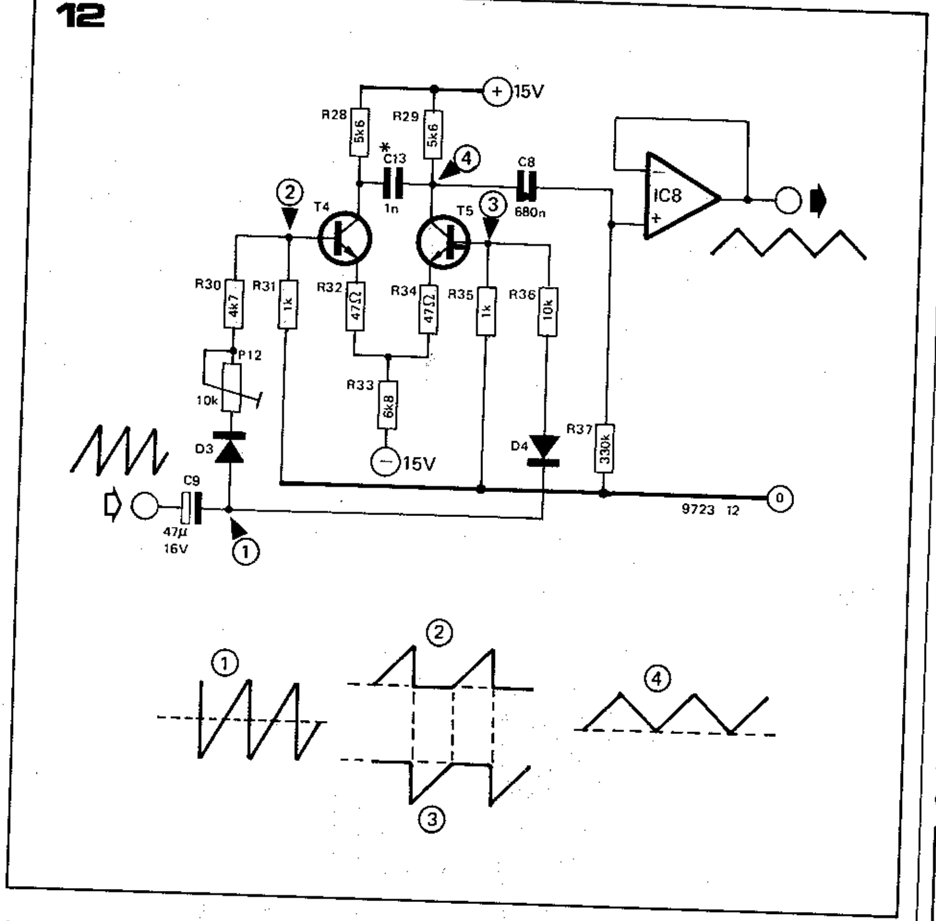
Each of the waveforms available at the VCO output has its own musical character, which is useful for particular applications. An unfiltered squarewave











is not particularly useful, since the odd harmonics cause the sound to be extremely harsh. However, filtered squarewaves are useful for the imitation of flute tones, and certain woodwinds such as clarinet.

The sawtooth waveform, which is rich in all harmonics is suitable for the imitation of brass, woodwind and many string instruments, and has an extremely bright and lively character.

The amplitudes of the sawtooth harmonics fall off at 6 dB per octave, i.e. the amplitude of the nth harmonic is 1/n times the amplitude of the fundamental. Where this fall is too abrupt the spaced sawtooth waveform can be used. This has an even brighter character than the sawtooth and is extremely useful for imitating very brilliant instruments such as the violin family and some of the higher pitched brass instruments such as cornet and trumpet.

The triangle and sine waveforms are musically very similar. The triangle is completely lacking in even harmonics, and the odd harmonics are of low amplitude. The sound of the triangle is flutelike, very smooth and mellow.

**Figure 11. Circuit of the spaced sawtoo converter. This clips the sawtooth wavefor passing only the peaks.**

**Figure 12. The triangle converter operates feeding the positive and negative half-wa rectified sawtooth to the inputs of a differe tial amplifier. The resultant difference outp is a triangle waveform.**

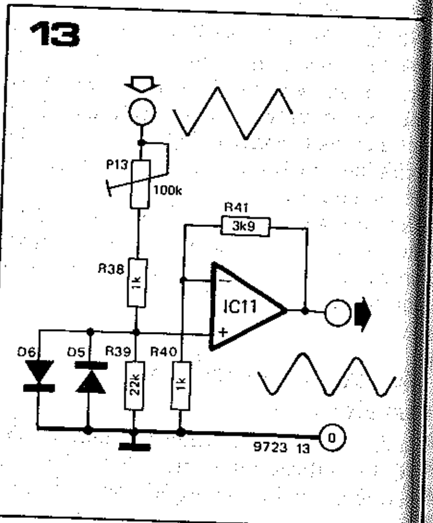
**Figure 13. The sine converter operates simp by 'rounding off' the peaks and troughs of t triangle to give an approximation to sinewave.**

**Figure 14. The PWM squarewave generator simply a voltage comparator whose outp switches at a certain point on the sawtoo waveform. The trigger level can be varie either by P5 or by an external input, th pulse width modulating the squarewave shown in figure 14b.**

**Figure 15. The output adder, which can used to combine the various output wavefor as desired.**

A pure sine waveform is, of cours completely lacking in any harmon content and sounds even smoother an more bland than the triangle – so far to be completely without character.

A low harmonic distortion of the sin waveform is not particularly importan for musical applications, provided th harmonic content is sufficiently lo that the sinewave sound contrasts wit that of the triangle. The sinewave thus derived from the triangle by a extremely simple diode shaper circuit.



## Spaced sawtooth converter

Figure 11a shows the circuit of the spaced sawtooth converter section. The sawtooth output of the VCO is fed into IC5 via R22. IC5 functions as an inverting half-wave rectifier, with a variable offset provided by P11. Depending on the setting of P11, the negative voltage at its slider causes a positive offset at the output of IC5 of between zero and about +14 V.

While the output of IC5 is positive D7 is reverse biased and the op-amp amplifies and inverts the positive going input sawtooth with a gain of about 5.5. However, this applies only so long as the output of IC5 remains positive. As the sawtooth voltage increases, a point on the waveform will be reached where the output of IC5 falls below zero. D7 will become forward biased and will clamp the output of IC5 to about –0.6 V.

The point on the sawtooth waveform at which clamping occurs depends on the setting of P11. With P11 adjusted to give an offset of zero the sawtooth will be clipped at a very low level. On the other hand, with P11 set to give a large offset voltage the sawtooth amplitude may never be high enough to cause the output of IC5 to swing negative, and the sawtooth will appear at the output of IC5 unclipped.

IC7 amplifies and inverts the output from IC5 with a gain of about –4, and P11 is adjusted so that the amplitude is the same as that of the sawtooth waveform, nominally 1.5 V p-p.

## Triangle converter

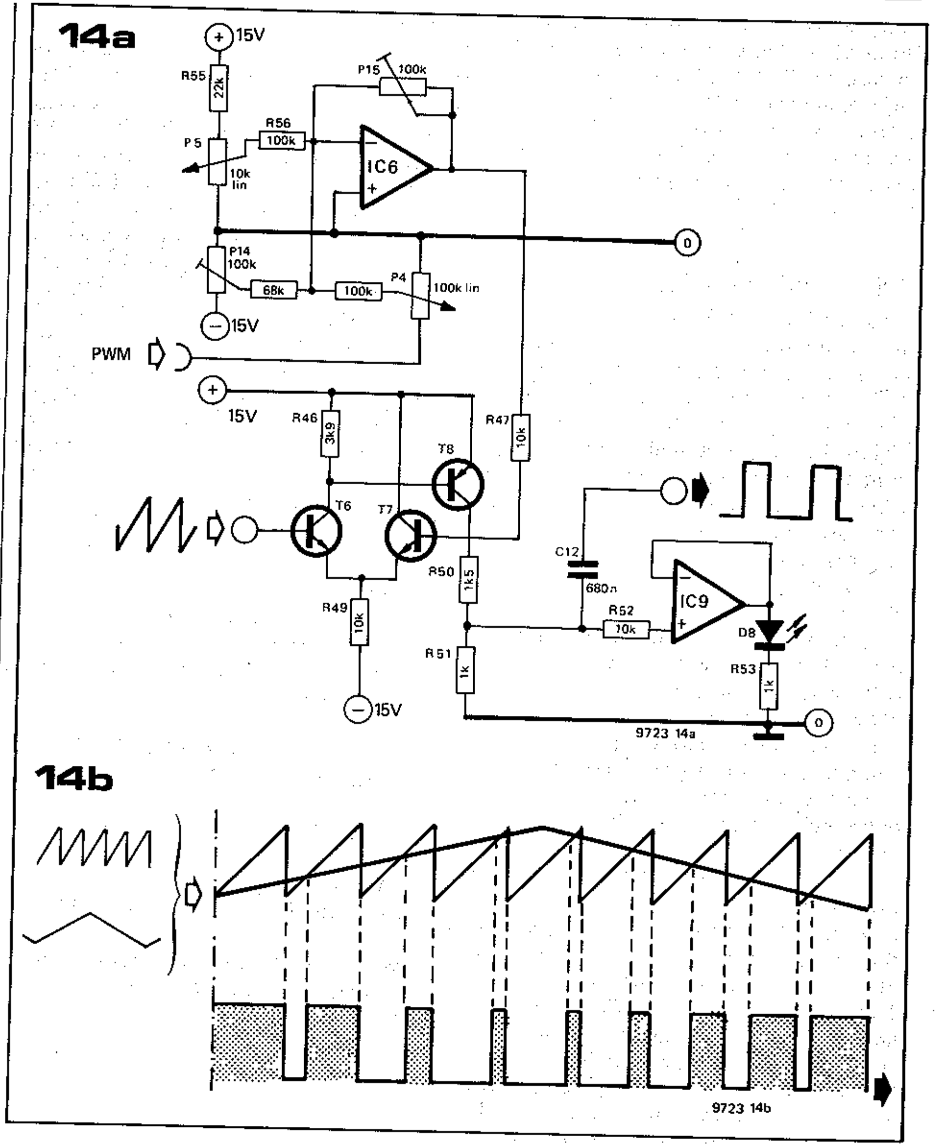
Half-wave rectification is again employed in the triangle converter, figure 12. The input sawtooth (1) is positive and negative half-wave rectified by D3 and D4, and the positive and negative half cycles are fed to the bases of T4 and T5 respectively (2) and (3). Since T4 and T5 form a differential amplifier the collector waveform of T5 is (2) - (3), which is a triangular waveform (4). IC8 is connected as a voltage follower to buffer the output.

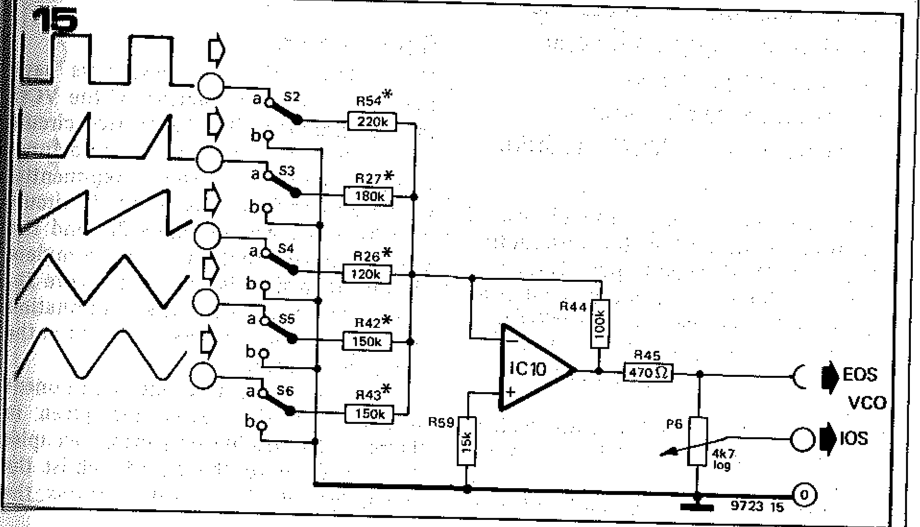
It may seem a little strange to use a discrete amplifier in this circuit when extensive use is made of IC op-amps elsewhere. The reason is that they have a limited slew rate, and this can result in a notch at the apex of the triangular waveform where the crossover from positive half-cycle to negative half-cycle occurs. This introduces harmonics that detract from the mellow sound of the triangular waveform. The discrete amplifier has a larger slew rate and is largely free from this defect. C13 also helps to filter out the spike, but it does cause a slight falloff of the triangle amplitude at high frequencies. The value of 1 n for C13 is by no means mandatory, and other values may be substituted to suit personal taste.

## Sine converter

As mentioned previously, the sine converter does not produce an extremely pure sinewave, but the circuit (figure 13) is simple and the output waveform is musically adequate. The triangle output from IC8 is fed to the non-inverting input of IC11 via P13 and R38. The positive and negative excursions of the triangle at the op-amp input are limited logarithmically by a matched pair of diodes D5 and D6, and the resulting approximation to a sinewave is amplified by IC11.

P13, R38 and R39 form an attenuator. The setting of P13 determines the triangle amplitude that would appear across R39 were D5 and D6 omitted, and hence the point on the triangle waveform at which limiting occurs. For example, with P13 set to maximum the voltage appearing across R39 will be very small, and D5 and D6 may conduct only on the peaks and troughs of the triangle, so the output will be too 'peaky'. On the other hand, with P13 set to minimum the signal will be





clipped very early in the waveform. Somewhere between these extremes is a setting of P13 that will give the best approximation to a sinewave. This setting can be found either by ear, or visually using an oscilloscope, or using a distortion meter to adjust for minimum distortion.

## Pulse width modulator

This section of the curve shaper generates a squarewave whose duty-cycle can be preset to any desired value from 0 to 100%, or which can be modulated by an external signal. T6, T7 and T8 (figure 14) form a high speed voltage comparator whose output will go high when the sawtooth input voltage exceeds the base voltage of T7, and which will go low on the trailing edge of the sawtooth.

The base voltage of T7 is set by the output voltage of summing amplifier IC6, which can be fed both with a DC voltage via P5 and with a signal from the PWM input. As the output voltage of IC6 becomes more positive the comparator will trigger later and later along the sawtooth ramp, so the output pulse will be narrower. This is illustrated in figure 14b, which shows the response to a low-frequency triangular PWM input signal.

P14 and P15 set the range of P5, so that this control can be used to preset the duty-cycle over the range 0 to 100%. The amplitude of the PWM input, and hence the modulation depth, is controlled by P4. IC9, which is connected as a voltage follower, lights LED D8 whenever the comparator output is high. This indicates that the VCO is functioning, and the LED brightness also gives an indication of the duty-cycle of the squarewave output.

## Output adder

The output adder circuit (figure 15) requires little explanation. When any switch is in the 'b' position then that input is open-circuit and the corresponding input resistor of the op-amp, IC10, is grounded. When a switch is in the 'a' position then the corresponding waveform is fed to the summing amplifier. Two or more waveforms may be summed by closing several switches simultaneously, which greatly extends the range of output waveforms available.

The adder stage has two outputs: external output signal (EOS), which is routed to the socket on the VCO front panel, and internal output signal (IOS), which is internally wired to the voltage-controlled filter (VCF).

As a suggestion for those experimenters who wish further to increase the flexibility of the VCO system, switches S2 to S6 may be replaced by potentiometers to form a mixer circuit in which the amplitude of each input waveform fed to the summing amplifier is infinitely variable.

## Conclusion

The discussion of the VCO module has now reached the stage where the description of all the circuit sections is complete, and the musical value of the various output waveforms has been given some consideration. The following chapter will deal with the constructional aspects of the VCO, including selection of components, assembly of the module p.c. board, testing and adjustment. When this stage is reached the synthesiser will at last start to become a playable instrument insofar as the VCO will produce an output signal of the correct pitch when a key is depressed, although the full musical potential cannot be realised until the rest of the synthesiser is complete.

*Literature:*

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*"Nonlinear Circuits Handbook" Analog Devices, Norwood, Mass. (USA) 1974*

*Schaefer, R.A.: "New techniques for electronic organ tone generation". JAES (Journal of the Audio Engineering Society), July/aug. 1971*

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## chapter 5

# construction of the VCO

**Having dealt with the theoretical circuits used in the VCO, this chapter goes on to discuss the selection of components and describes the practical construction, testing and adjustment of the VCO module.**

Care must be taken in the choice of components for, and in the construction of, the VCO, if reliable performance is to be obtained. The same general comments apply that were made earlier with regard to component quality. In addition, the following points should be noted:

1. Capacitor C2 should be a low leakage type – preferably MKM or equivalent.
2. Transistor T1 to T3 should be tested, as will be explained later.
3. Diodes D3 and D4 should be matched pair.

It is important that the reset transist T1 in the CCO section should be sele ted for low leakage current, as excessi leakage current means current lost fro C2 and non-linearity of the CCO at lo frequencies.

The test setup for T1 is shown in figure The PNP transistor T8 can be used the second transistor in the circuit, any similar transistor can be used. Th meter can be a multimeter set to th 1 mA range. The base of T8 is initiall left open-circuit to check that it is n leaky. The meter should read zero. Th base of T8 is then connected to the 0 rail via a 100 k resistor to check that has adequate current gain. The met should read at least 1 mA (i.e. full-scale The base of T8 is then connected to th collector of T1. Any leakage curren through T1 will be amplified by th current gain of T8 to give a deflectio on the meter. Only if the meter rea zero is the leakage current of T1 su ficiently low.

Finally, the current gain of T1 can b checked by connecting its base to +5 through a 2k2 resistor, when the met should again show full-scale deflection

FETs T2 and T3 can be tested using th circuit given in chapter 3 for testing th FETs in the keyboard interface. Unlik the keyboard interface circuit, FE which show a $U_S$ in the test circuit less than 0.5 V are not suitable for th VCO. However, FETs that have bee rejected for the keyboard interfa because their $U_S$ value was too high, ca be used in the VCO if the value of U lies between 1.6 V and 2 V. For FE with $U_S$ values between 0.5 and 1.5 the source resistors R17 and R20 shou be selected from table 1 in part 3. F FETs having a $U_S$ value between 1.6 and 2 V, R17 and R20 should be 4k7. Diodes D3 and D4 should be purchase as a matched pair or, if several diodes the correct type are to hand, a reaso ably matched pair may be selected b measuring the forward voltage dr versus forward current characteristics the diodes and selecting the pair that a most similar.

## Construction

Once these critical components hav been selected, construction of the VC may commence. On the printed circu board the VCO is split into two fun tional sections: the exponenti converter and CCO, the complete circu of which is given in figure 2a, and th curve shaper section, the complet circuit of which is given in figure 2 These two circuits are the combinatio of all the partial circuits discussed in th previous chapter.

Printed circuit board and componen layouts for the VCO are given i figure 4. The oscillator section occupie the top third of the board, whilst th remainder of the board contains th

curve shaper circuits. To avoid interaction between the two sections of the circuit they each have separate supply and ground connections. The only link between them is at the source of T3, which is the CCO output (point A in figures 2a and 2b). Assembly of the board poses no particular problems, the only point to note being that at this stage C13, R26, R27, R42, R43, R54, and the link joining pin 4 of IC3 to the gate of T2, are omitted for test purposes.

## Test and adjustment

The first test is to check that the CCO is functioning, and for this purpose a 1 M resistor is connected between the gate of T2 and −15 V to act as a current

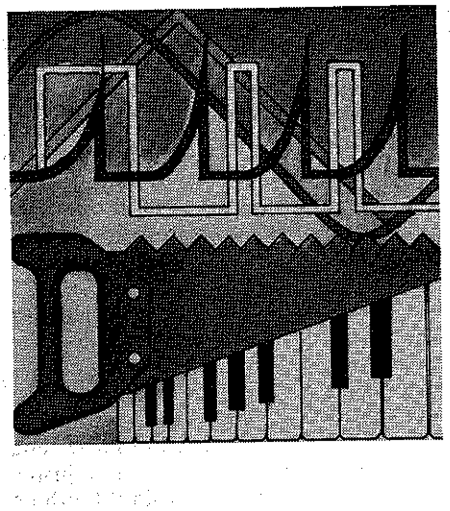
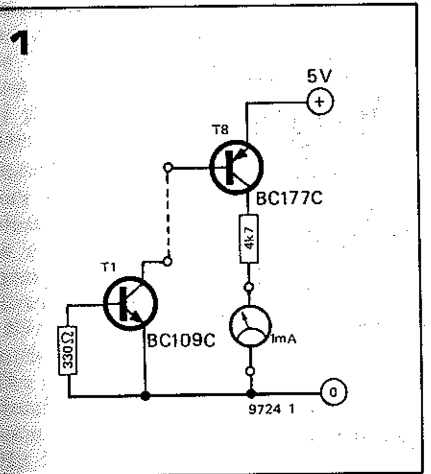
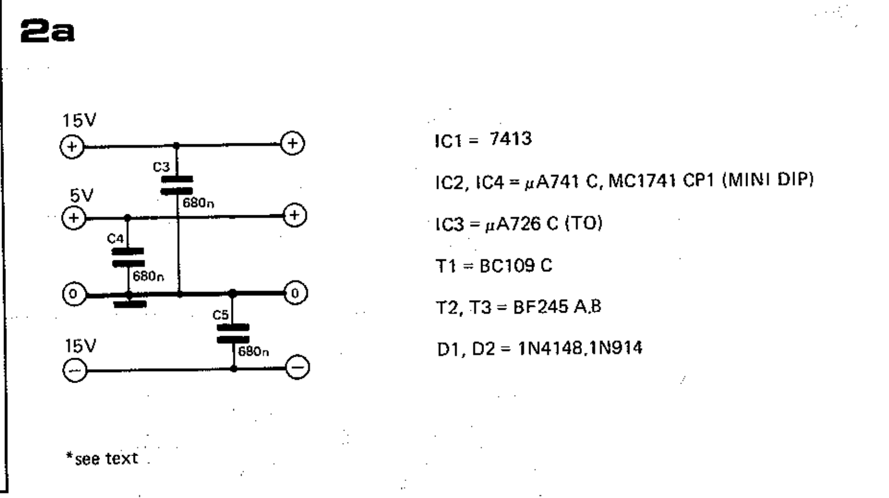


**Figure 1. Simple test circuit for selecting transistor T1 of the VCO.**

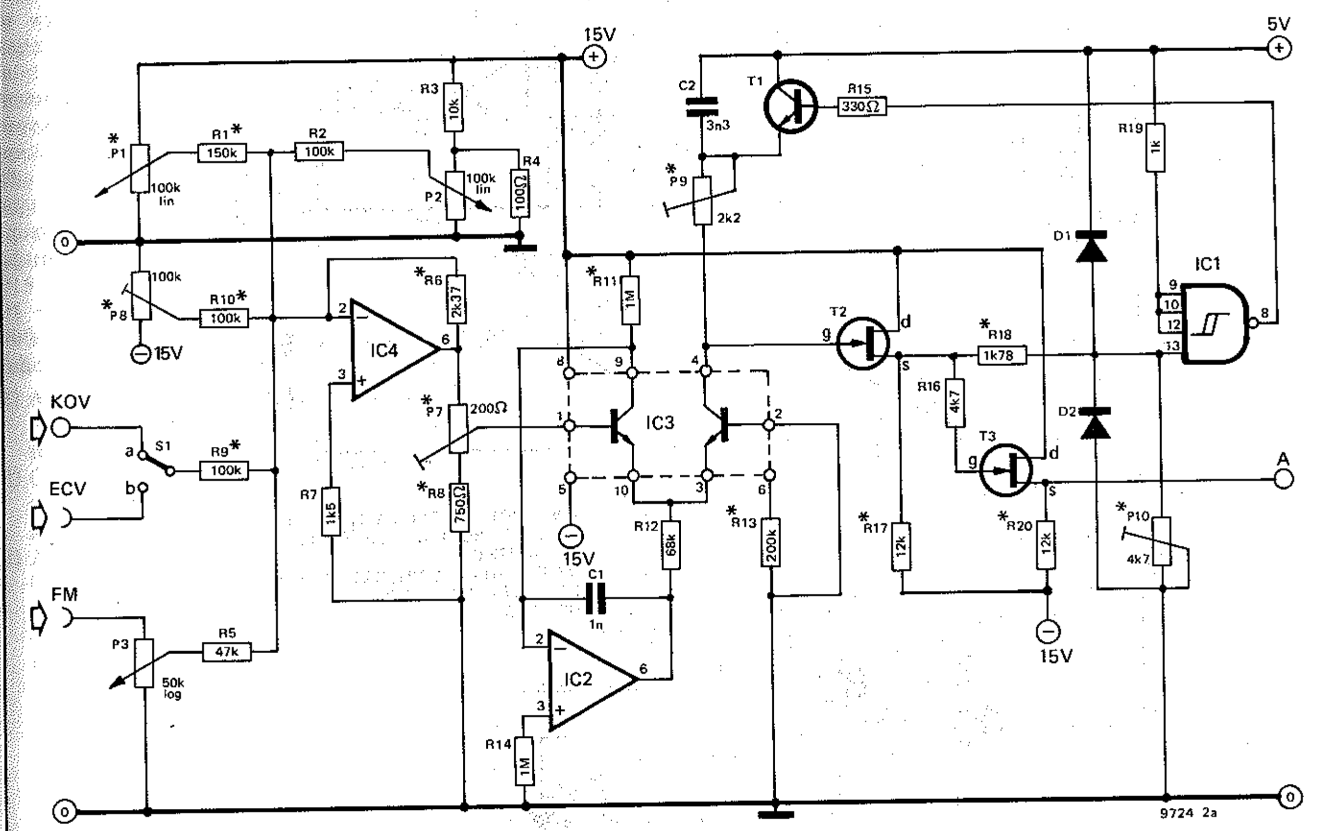
**Figures 2a and 2b. These two circuits constitute the complete VCO, and combine into two functional groups the partial circuits discussed the previous chapter.**





IC1 = 7413
IC2, IC4 = μA741 C, MC1741 CP1 (MINI DIP)
IC3 = μA726 C (TO)
T1 = BC109 C
T2, T3 = BF245 A,B
D1, D2 = 1N4148,1N914

*see text



source for the CCO. The CCO output can be monitored with an oscilloscope at point A.

Should the oscillator fail to start then P10 can be adjusted until it does. It will probably be found that the oscillator stops as the slider of P10 approaches its two extreme positions, and P10 should be set midway between the positions at which oscillation ceases. At this stage, the frequency of the oscillator should be around 1 kHz, and the waveform will not be a perfect sawtooth, but will exhibit an exponential curvature due to the 1 M resistor being used in place of a constant current source.

Once the CCO has been checked, the 1 M resistor can be removed and the CCO connected to the exponential converter by soldering in the link between pin 4 of IC3 and the gate of T2.

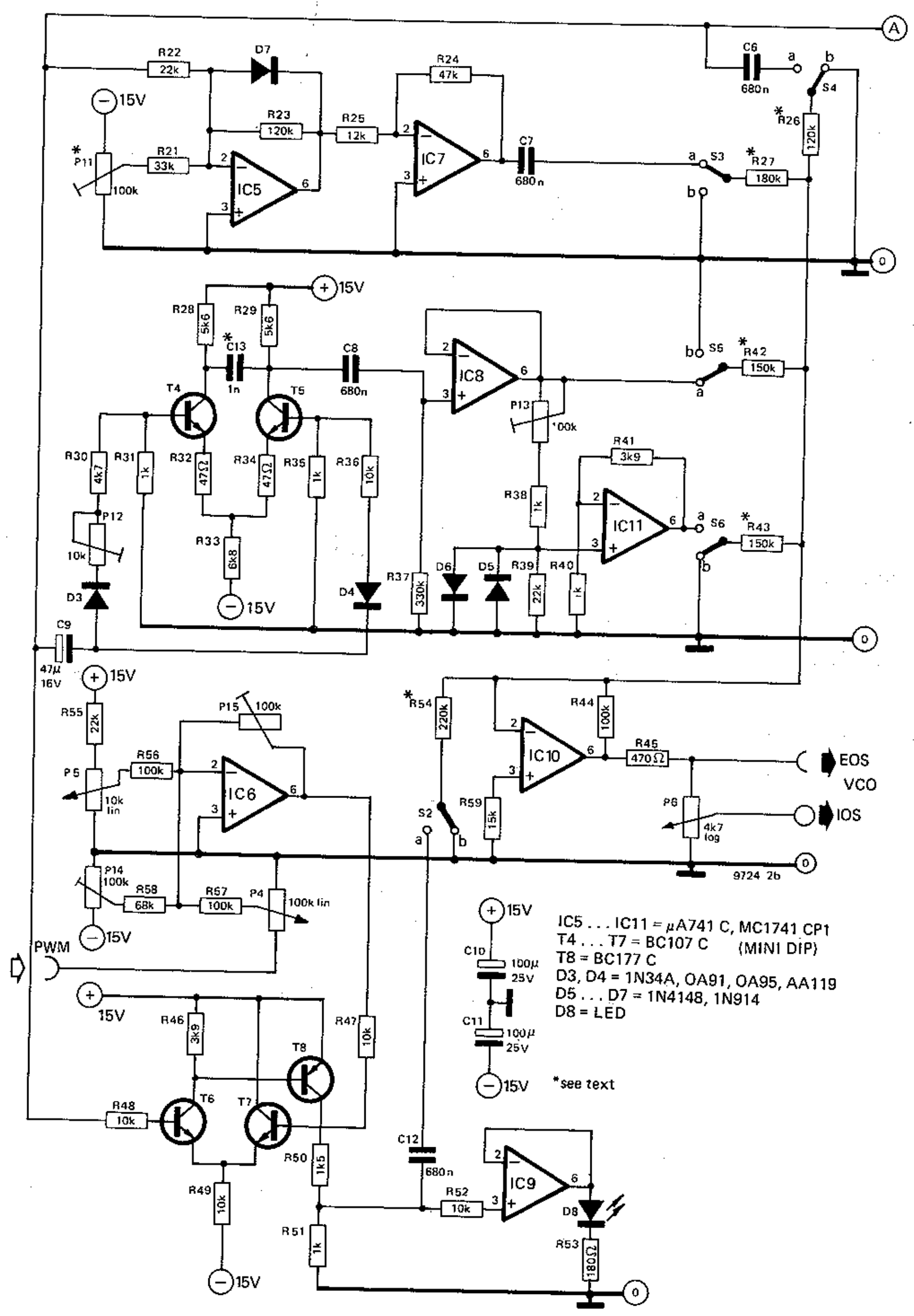
With the sliders of P2, P3 and P8 turned to zero volts and the KOV input grounded, it should now be possible to vary the VCO frequency by adjusting P1. If the exponential converter is operating correctly, the waveform at point A should be a perfect sawtooth. It may be found that at low frequencies the VCO will not oscillate reliably, in which case the adjustment of P10 will require further attention.

Once the VCO functions reliably over the entire audible range, P1 should be turned completely anticlockwise and the offset potentiometer P8 adjusted until the lower frequency limit (with no control voltage other than from P8) around 15 Hz. This adjustment does n need to be extremely accurate. desired, the frequency range of th VCOs can be extended beyond th existing 10 kHz. There are two poss bilities: either R12 can be reduced t 47 k, or else C2 can be reduced to 2n (or even 2n2).

## Curve shaper section

Adjustment of the curve shaper begin with the spaced sawtooth converte section. P11 adjusts the clamp level o this circuit and hence the 'spikeness' o the waveform, which affects the tona quality. The adjustment is a matter o taste, but as a guideline the peak-to peak amplitude of the waveform, viewed at point S3a, should be about 3 V.

**2b**



IC5 . . . IC11 = µA741 C, MC1741 CP1 (MINI DIP)
T4 . . . T7 = BC107 C
T8 = BC177 C
D3, D4 = 1N34A, OA91, OA95, AA119
D5 . . . D7 = 1N4148, 1N914
D8 = LED

*see text

Next, the triangle converter (T4, T5) can be adjusted. The symmetry of the triangle waveform is determined by the matching of diodes D3 and D4. P12 can compensate for slight mismatches in these diodes, but if the degree of mismatch is large the only answer is a better matched pair of diodes. The output waveform should be monitored at point S5a with P12 in its mid-position, and P12 should then be turned one way or the other to obtain a symmetrical triangular waveform. If notches are apparent at the peaks of the triangle waveform (especially noticeable at high frequencies) then capacitor C13 should be added. The value of 1 n is given as a guideline, but C13 should preferably be chosen experimentally to give the best compromise between elimination of the notches and attenuation of the signal at high frequencies.

Once the triangle waveform is satisfactory the sine converter may be adjusted. Ideally, diodes D5 and D6 should also be a matched pair in order to ensure symmetry of the sine waveform. However, a random pair of 1N4148s or 1N914s will usually prove to be a sufficiently close match in practice. The purity of the sinewave is adjusted visually by monitoring the waveform at point S6a and varying the resistance of P13 for best results. The sine converter output can be compared with the sine output of a signal generator, if available, or with a sine curve plotted on graph paper. The purists may like to adjust for minimum distortion using a distortion meter, though the simpler adjustment procedure is adequate from a musical point of view.

The final section of the circuit to be adjusted is the pulse-width modulated squarewave generator. The aim of this adjustment is to set trimmers P14 and P15 so that the adjustment range of P5 varies the duty-cycle from 1% to 99%. The setting-up procedure is as follows:

1) Adjust P14 until its wiper voltage is −5.5 V, and adjust P15 to maximum resistance.
2) Connect the voltmeter to the output of IC6 and monitor the PWM signal at point S2a with an oscilloscope.
3) Adjust P5 to give first maximum (approx. 99%) and then minimum pulse width (approx. 1%) of the PWM signal, and note the output voltage of IC6 for these two conditions thus: – $V_{max}$ = voltage for minimum pulse width, $V_{min}$ = voltage for maximum pulse width.

**3**

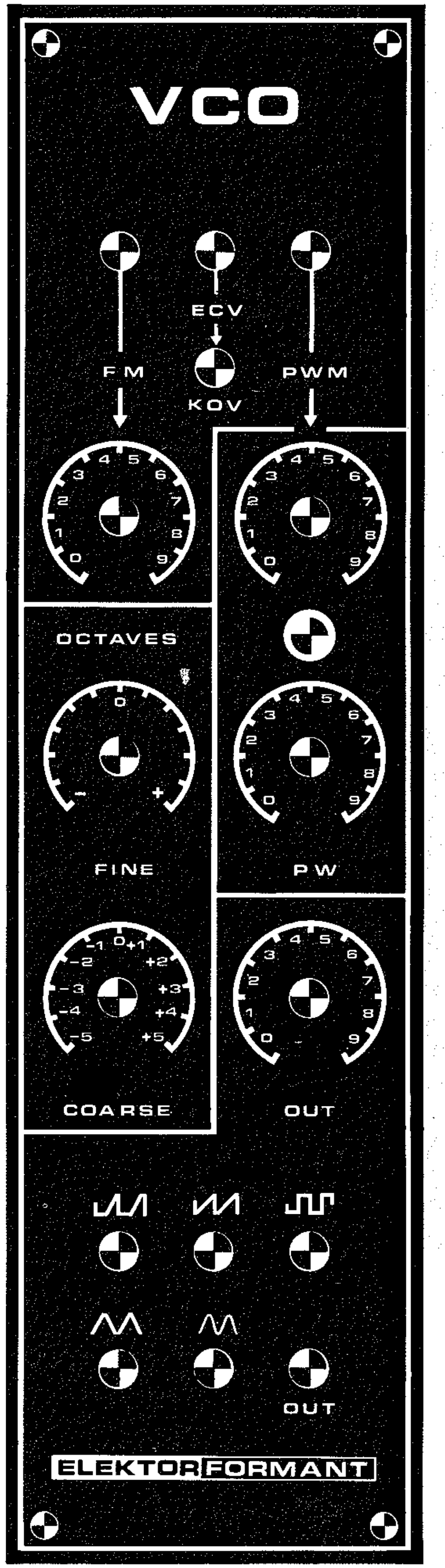


Figure 3. Suggested front panel layout for the VCO.

4

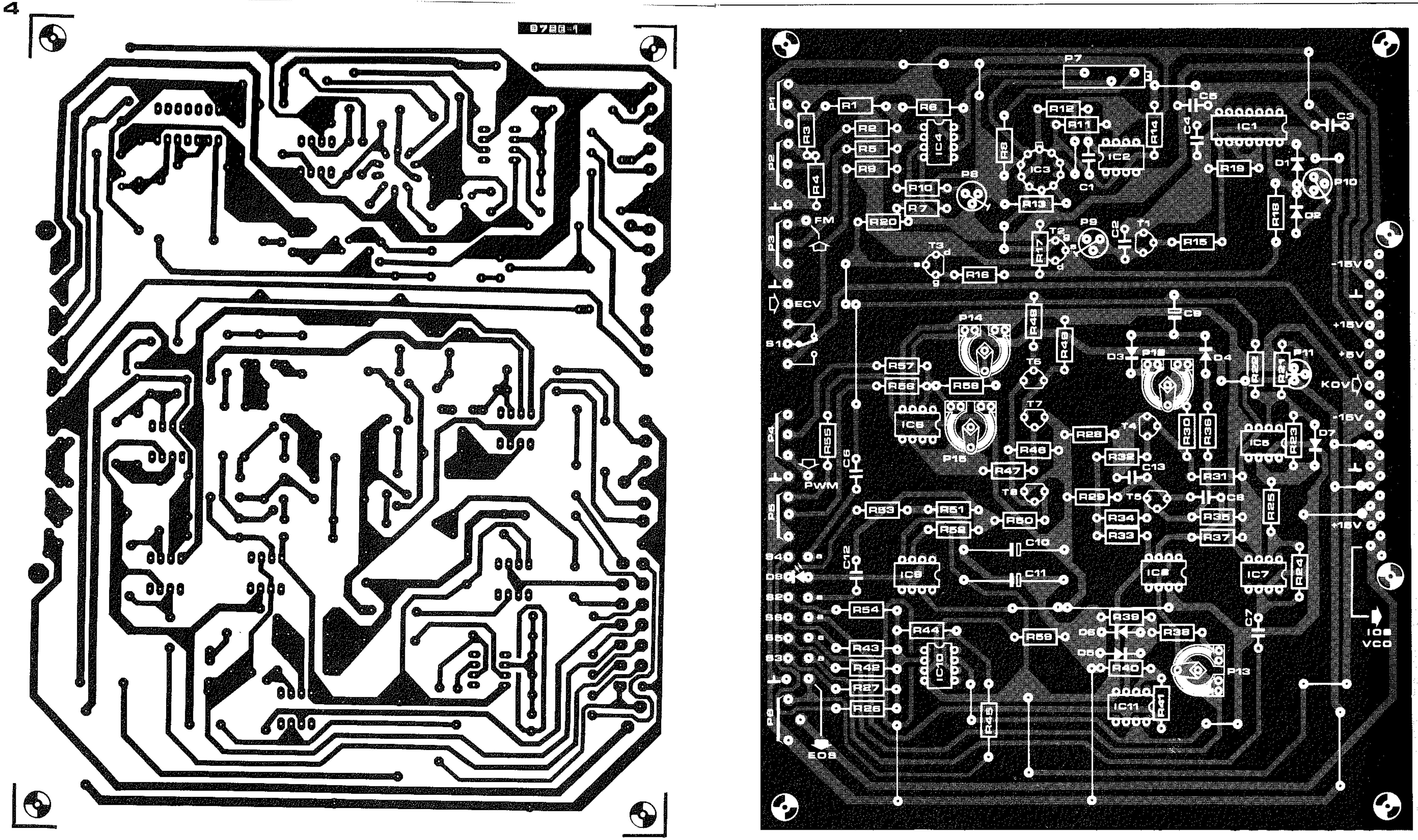


**Figure 4. Printed circuit board and component layout for the Formant VCO (EPS 9723-1).**

**Parts list for figures 2 and 4.**

Resistors:

a. 1% metal oxide
R1 = 150 k
R6 = 2k37[1]
R8 = 750 Ω
R9,R10 = 100 k
R11 = 1 M
R13 = 200 k
R18 = 1k78[1]

b. 5% carbon film
R2,R44,R56,R57 = 100 k
R3,R36,R47,R48,
R49,R52 = 10 k
R4 = 100 Ω
R5,R24 = 47 k
R7,R50 = 1k5
R12,R58 = 68 k
R14 = 1 M
R15 = 330 Ω
R16,R30 = 4k7
R17,R20 = 12 k (nominal)
R19,R31,R35,
R38,R40,R51 = 1 k
R21 = 33 k
R22,R39,R55 = 22 k
R23 = 120 k
R25 = 12 k
R26 = 120 k (nominal)
R27 = 180 k (nominal)
R28,R29 = 5k6
R32,R34 = 47 Ω
R33 = 6k8
R37 = 330 k
R41,R46 = 3k9
R42,R43 = 150 k (nominal)
R45 = 470 Ω
R53 = 180 Ω
R54 = 220 k (nominal)
R59 = 15 k

Presets:

a. Cermet
P7 = 200 Ω (or 220 Ω or 250 Ω) multiturn preset.
Note pinout, end pins spaced 5.1 mm and 7.6 mm from centre pin, which is offset by 2.5 mm.
P8,P11 = 100 k
P9 = 2k2
P10 = 4k7

b. Carbon
P12 = 10 k
P13,P14,P15 = 100 k

Potentiometers:

a. Cermet
P1 = 100 k lin

b. Carbon
P2,P4 = 100 k lin
P3 = 50 k log.
P5 = 10 k lin.
P6 = 4k7 (5 k) log.

Capacitors:

C1 = 1 n
C2 = 3n3 (MKM)
C3,C4,C5,C6,C7,
C8,C12 = 680 n
C9 = 47 µ/16 V
C10/C11 = 100 µ/25 V
C13 = 1 n (see text)

Semiconductors:

T1 = BC109C
T2,T3 = BF 245A, B
T4 . . . T7 = BC107C
T8 = BC177C
D3,D4 = OA91, OA95, AA118,AA119, or 1N34A
D1,D2,D5,
D6,D7 = 1N4148, or 1N914
D8 = LED, TIL209 or similar
IC1 = 7413
IC2,IC4,IC5,IC6,IC7,IC8,
IC9,IC10,IC11 = µA741C or MC1741 CP1 (MINI DIP)
IC3 = µA726C (Fairchild, TO package)

Miscellaneous:

31 pin (DIN 41617) connector
S1 . . . S6 = SPDT miniature toggle switch.
4 x 3.5 mm jack sockets

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[1] These are 'optimum' values. However, 2k4 and 1k8 resistors can be used for R6 and R18 respectively, provided they are 1% metal oxide types!!

4) Turn the wiper of P14 to zero volts and the wiper of P5 to maximum voltage. Now use P15 to adjust the output voltage of IC6 so that it is equal to the difference between the two previously noted values $V_{max}$ and $V_{min}$ i.e.
$V_{o,IC6} = V_{min} - V_{max}$.
The output voltage of IC6 will be negative since it is connected as an inverting amplifier.

5) Adjust P14 to give maximum pulse-width (99% duty-cycle) of the output signal. When the wiper of P5 is now turned to zero volts the pulse width should be minimum (1% duty-cycle). This completes the adjustment of the PWM stage.

Oscillograms of all the waveforms are shown in photos 1 to 7.

## Output adder

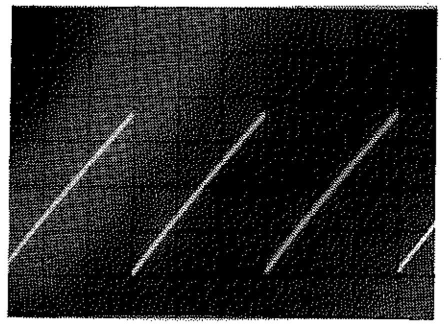
Once the various sections of the curve shaper have been adjusted the input resistors of the output adder may be selected (R26, R27, R42, R43 and R54). A 250 k potentiometer is connected in place of each resistor in turn, and the peak-to-peak amplitude of the relevant waveform is adjusted to about 2.5 V at output EOS. The resistance of the pot is then measured and it is replaced by a fixed resistor of the nearest preferred value from the E24 range.

## Front panel

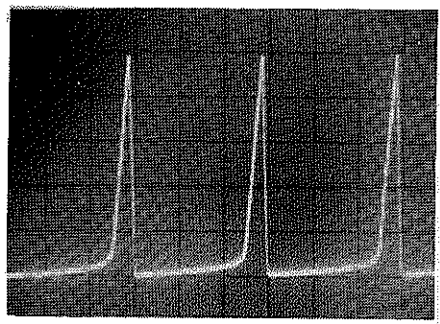
A front panel layout for the VCO is given in figure 3. The three inputs, FM, ECV and PWM are at the top of the panel, with the switch (S1) to select between ECV and KOV mounted below. Potentiometer P3, which controls the FM modulation depth, is mounted below the FM input socket, while P4 and P5, which control the pulse width modulation depth and duty-cycle respectively, are mounted below the PWM input socket. The coarse and fine tuning controls (P1 and P2) are also grouped together, on the left of the panel, while the output level control (P6) is grouped with the waveform selection switches (S2 to S6) and the output socket.

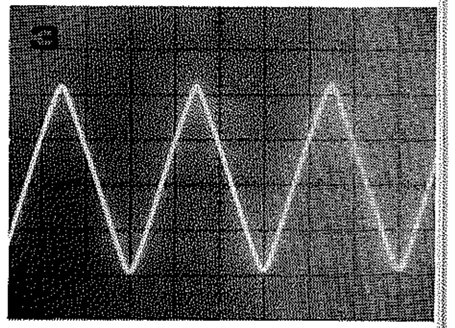
## Module construction

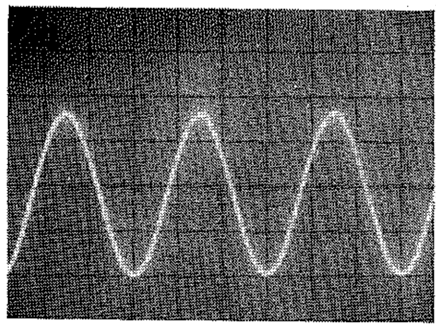
It is essential that the VCO module should be screened to avoid any interference pickup. To provide this screening, and to make the module mechanically rigid, the p.c. board is mounted on a carrier made from 16 or 18 SWG aluminium. The dimensions of the carrier are those of a large Eurocard (165 mm x 210 mm) so that the module will fit a Euro-standard card frame. A right-angle bend at the front edge of the carrier allows it to be secured to the front panel by means of the potentiometer mounting bushes. The p.c. board is mounted on the carrier using M3 screws and spacers. Photo 8 shows the completed module.

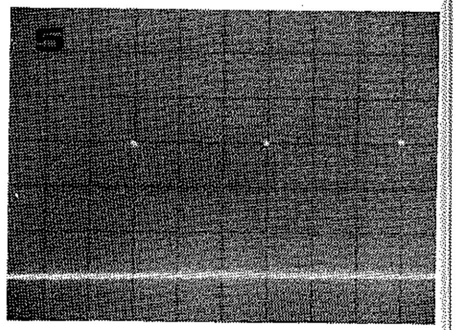


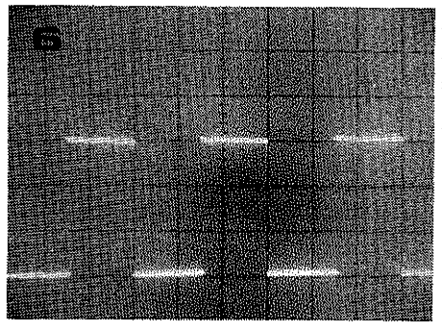
| | |
|---|---|
| ECV | = External Control Voltage, i.e. front-panel input to VCO. |
| KOV | = Keyboard Output Voltage, i.e. permanently wired input to VCO from interface receiver. |
| FM | = Frequency Modulation input |
| PWM | = Pulse Width Modulation input |
| EOS | = External Output Signal from VCO (front panel output) |
| VCO/IOS | = Internal Output Signal from VCO, will be permanently wired to one VCF input. |

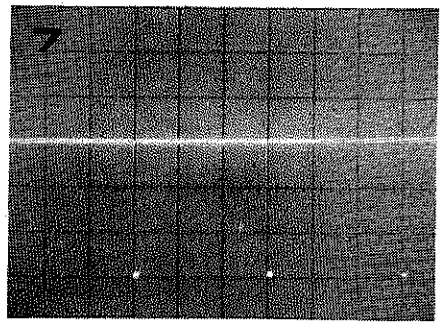












**Photos 1 to 7. These oscillograms give an indication of the waveforms that should be available at the curve shaper outputs:**
**1. Sawtooth 2. Spaced sawtooth 3. Triangle 4. Sinewave 5. Squarewave, minimum duty-cycle 6. Squarewave, 50% duty-cycle 7. Squarewave, maximum duty-cycle.**

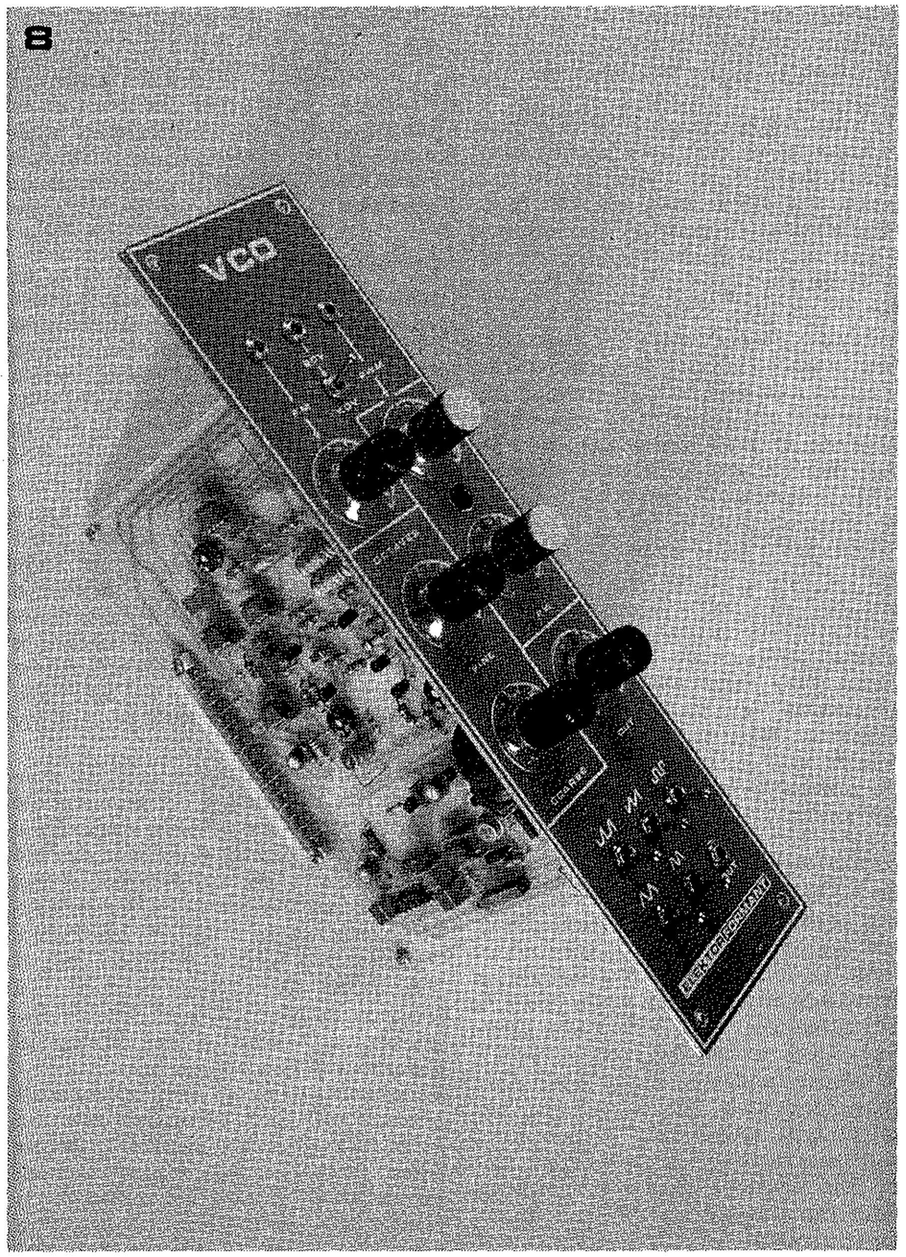


**Photo 8. The completed VCO module.**

## Octaves/Volt adjustment

The most critical adjustment made to the entire synthesiser is the setting up of the octaves/volt characteristic of the VCOs, as this adjustment determines the accuracy of the synthesiser tuning.

There are two methods of adjusting the VCO. The simpler method requires the use of a frequency counter and digital voltmeter, while the second method requires an audio signal generator with a calibrated frequency scale.

Before commencing the adjustment procedure power should be applied to the VCO for several minutes to allow the temperature (especially of IC3) to stabilise.

To adjust the VCO using frequency counter and DVM, all inputs and controls of the VCO input adder are set to zero volts and P9 is set in its centre position. The connection between the wiper of P1 and R1 must be unsoldered, and the free end of R1 connected to ground, whilst the wiper of P1 is connected to the KOV input with S1 in position 'a'. The frequency counter is connected to the VCO output and the DVM to the wiper of P1. With P1 turned fully anticlockwise the frequency counter will read around 15 Hz, which was set previously by means of P8. P1 is now turned slowly clockwise until the DVM reads 1 V, when the VCO frequency should be twice what it was with P1 set to zero, e.g. if the zero frequency was exactly 15 Hz the frequency should now be exactly 30 Hz. Of course, initially this will not be the case, and some adjustment of P7 will be required.

P1 is then turned until its wiper voltage is exactly 2 V, when the VCO frequency should be four times the zero voltage frequency, e.g. 60 Hz. This procedure is repeated at 1 V steps over the entire range of P1,checking that the correct frequency is obtained at each step. Thus if 0 V = 15 Hz, then 1 V = 30 Hz, 2 V = 60 Hz, 3 V = 120 Hz etc. P7 is adjusted to obtain the best accuracy possible over the widest frequency range.

At high frequencies (greater than 3 kHz) P9 can be used to correct any deviations from the 1 octave/volt characteristic.

To adjust the VCO using the beat note method, the outputs of an audio oscillator and the VCO must be fed into the left- and right-channels of a stereo amplifier, or via an audio mixer into a mono amplifier, so that the beat notes can be heard via the loudspeakers. The VCO is connected to the KOV output of the previously calibrated keyboard.

The audio oscillator is set to a frequency between 400 and 500 Hz, and the main tuning of the keyboard is switched off. The top note of the keyboard is then depressed, and the VCO tuning controls P1 and P2 are adjusted until the audio oscillator and VCO are in tune with zero beat.

Next, the key an octave lower is depressed, when a dissonance or very rapid beat note will be heard. P7 is then adjusted until zero beat is obtained between the audio oscillator and the VCO note one octave lower.

The top key is again depressed, when it will be found that, due to the adjustment of P7, a beat note is again heard. Using the VCO tuning controls, readjust for zero beat, then depress the key an octave lower, which will now be slightly out of tune due to adjusting the VCO tuning controls. P7 must therefore be readjusted to obtain a zero beat.

This procedure is repeated several times until the oscillator is perfectly in tune with both the top note and the note an octave lower. The tuning is then checked two octaves and three octaves below top C, and if necessary P7 is readjusted to obtain the best tuning over the entire keyboard range.

The higher ranges of the VCO must now be adjusted using P9. For this purpose the audio oscillator is tuned to around 2 kHz, the bottom note of the keyboard is depressed, and the coarse and fine tuning controls of the VCO are adjusted for zero beat. The key an octave higher is then depressed, and P9 is adjusted for zero beat using the same technique as for the previous adjustment procedure using P7. The tuning is then checked two octaves and three octaves above bottom C.

This completes the adjustment of the VCO.

## chapter 6

# 12 dB VCF

**This chapter introduces the first of the tone-shaping modules which process the 'raw' output of the VCOs to provide a wide variety of different tone colours and amplitude dynamics. The module presented here is a 12 dB per octave voltage-controlled filter (VCF) which is used to tailor the frequency spectrum of the VCO signal.**

Before looking at the VCF circuit in detail, it is worth examining the ways in which the VCF is used. Four filter functions are available. A lowpass filter with a rolloff of −12 dB per octave above the turnover point, a highpass filter with a rolloff of −12 dB per octave below the turnover point, a bandpass filter with variable Q and minimum slope of −60 dB per octave on either side of the centre frequency, and a notch filter. The turnover point – or centre frequency in the case of the band filters – is the same for all four filter functions, and can be varied by the application of a control voltage.

### Lowpass filter

The simplest use of the VCF is what might be called static tailoring of a VCO output using the KOV output of the keyboard to control the VCF. Suppose (to give a simple example), it is required to filter out a large proportion of the harmonics of the squarewave signal to produce a flutelike tone. The lowpass function of the VCF would be used and the turnover point would be set so that when a particular key was depressed the desired tone colour was obtained. If a higher note is depressed then the VCO pitch will increase. However, since the KOV output is also applied to the VCF the turnover point of the VCF will increase with the VCO frequency, so that it always remains in the same octave relationship to the VCO frequency. The same harmonic structure of the output waveform is thus maintained, – i.e. the VCF is being used as a tracking filter.

If the VCF is used simply as a tracking filter then the harmonic content of the output remains fixed for the duration of each note. However, dynamic variation of harmonic content during a note is also possible by controlling the VCF from the envelope shaper.

For example, to provide a good imitation of a trombone sound the note should initially start off with only a weak harmonic content. As the loudness of the note builds up the harmonic content also increases, i.e. the note becomes 'brighter'. Similarly, at the end of the note it is the harmonics which die away first.

This is achieved by using the VCF in the lowpass mode as a tracking filter with ADSR control, i.e. with inputs from KOV and from the envelope shaper. When a key is depressed the turnover point is initially determined by the KOV input, and is set so that the harmonics are filtered out. As the envelope shaper output voltage rises (attack) the turnover frequency of the VCF is increased to pass more of the harmonic content. At the end of the note (decay) the envelope shaper output falls and the turnover frequency of the VCF is reduced to filter out the harmonics once more.

These two simple examples relate to the imitative capability of the synthesiser, since most people will have a 'feel' for the sound of conventional musical instruments. However, it must once again be stressed that the synthesiser is not limited merely to an imitative role. It can also produce sounds that are unique to itself, that do not occur naturally and are totally 'electronic'.

### Highpass filter

So far only the use of the lowpass filter has been discussed. The highpass filter has the opposite effect to the lowpass filter, i.e. it can be used to attenuate the fundamentals of notes while retaining the harmonics. This is obviously useful for sounds which have only a weakly developed fundamental or a bright tonal character, such as harpsichord and spinet type sounds, and certain string and brass instruments. When controlled by the envelope shaper the highpass filter can also give an 'ethereal' character to a sound.

### Bandpass filter

In addition to the fundamental and harmonic series produced when a particular note of the instrument is sounded, brass and many woodwind instruments exhibit a number of fixed bandpass resonances, which are determined by the particular mechanical construction of the instrument. Use of the VCF as a bandpass filter with fixed centre frequency (KOV input switched off), together with a second VCF as lowpass tracking filter, allows these instruments to be more accurately imitated.

### Pedal controlled Wa-Wa

Using the VCF in the bandpass mode with a fairly high Q-factor, A Wa-Wa effect can be obtained by controlling the VCF with a 0 to 5 V DC supply from a pedal-controlled potentiometer (such Wa-Wa pedals are available commercially or are easily home-made).

### Notch filter

By sweeping the centre frequency of the notch filter up and down the spectrum, either manually using a potentiometer or automatically using a low-frequency oscillator, phaser-type sounds can be produced. If this is done using a white noise input instead of a VCO then interesting 'jet-aircraft' noises can be obtained.

## Design of the VCF

As far back as 1965, R.A. Moog designed 24 dB/octave lowpass and highpass filters, and no satisfactory alternative to these was found for several years, although they were periodically 're-invented' by others. It was not until the introduction of a specific type of integrated circuit, the operational transconductance amplifier (OTA), that a viable alternative became possible.

The Formant VCF is developed from the two-integrator loop shown in figure 1. Although a complete mathematical analysis of this circuit is beyond the scope of this book (those interested are referred to the bibliography), the basic concept is fairly simple to grasp.

The two-integrator loop can be considered as an analogue computer for the solution of a second-order differential equation. If the input resistor R1 and potentiometer $P_Q$ are removed, it can be seen that the circuit bears a remarkable resemblance to a quadrature oscillator. In fact, if the loop gain of the circuit is sufficient then it will function as an oscillator – at the frequency for which the differential equation solution holds.

$P_Q$ provides damping so that the circuit does not oscillate, but merely acts as a filter. Highpass, bandpass, and lowpass filter functions are available simultaneously at outputs (1), (2) and (3) respectively. At the turnover or centre frequency of the filters there is 90° phase shift between the integrator inputs and outputs. Thus between point (1) and point (3) there is 180° phase shift in all. By combining outputs (1) and (3) using a voltage follower A4 a notch function can be obtained. Since the two inputs are 180° out of phase at the centre frequency there is a null at the junction of the voltage follower's two input resistors at this frequency.

Of course the centre/turnover of this filter is not voltage-controlled, but is fixed by the integrator constants R and C, so to achieve voltage control one of these elements must itself be voltage-controlled. Voltage control of capacitance is impractical in this application. Voltage controlled resistors are possible in the form of LED/LDR combinations or FETs, but unfortunately both these methods suffer from disadvantages such as unpredictable performance due to wide tolerances, small control range, poor linearity, and breakthrough of the control signal.

Figure 1. The two-integrator loop used in the Formant VCF provides 12dB/octave highpass, bandpass and with the addition of A4, a notch filter.

Figure 2. Instead of normal op-amps, OTAs are used in the Formant VCF. The output current change is $g_m$ times the input voltage change, but $g_m$ can be varied by feeding in a control current $I_{ABC}$.

Figure 3. The OTA integrator used in the Formant VCF. The integrator time constant is controlled by the current $I_{ABC}$. A high impedance buffer ensures that all the output current of the OTA flows into the integrator capacitor.

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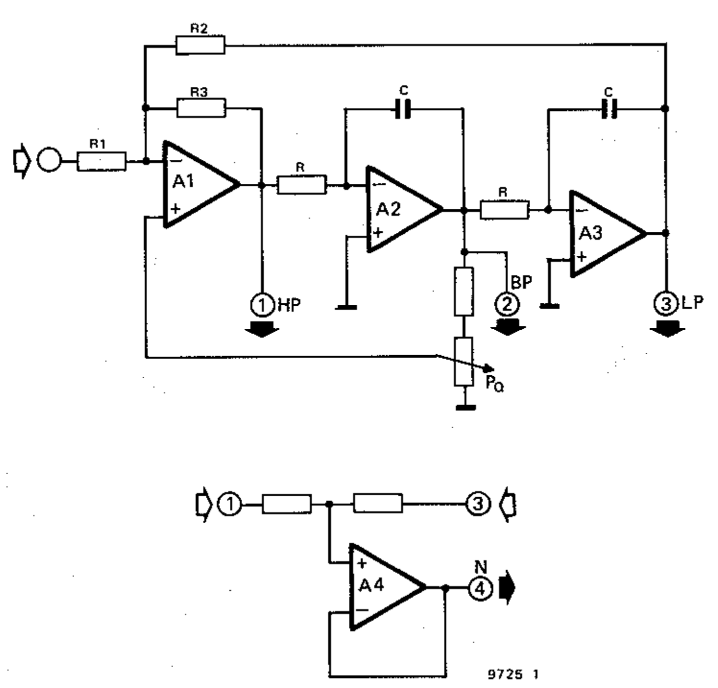

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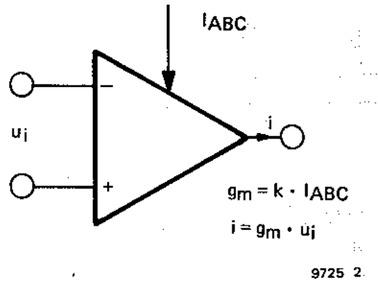

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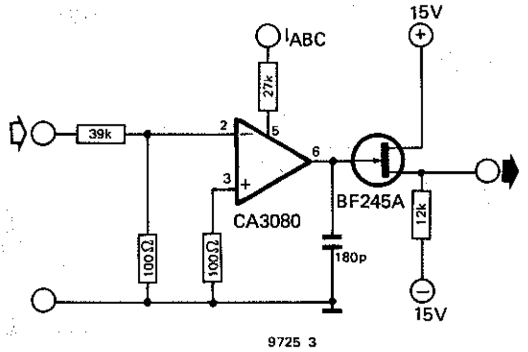


An alternative solution can be found by re-thinking the basic integrator design. The classic op-amp integrator consists of a differential-input voltage amplifier with the non-inverting input grounded. An input resistor connected to the inverting input (which is a virtual earth point) converts the input voltage into a proportional current. Since this current cannot flow into the inverting input it must flow into the feedback capacitor, and a voltage appears across the capacitor (and hence at the op-amp output).

It is fairly obvious that the op-amp is functioning simply as a voltage-to-current converter, and an equivalent circuit for an integrator would be an amplifier with a voltage-controlled current output, with a capacitor connected, not in a feedback loop, but between the output and ground. Varying the voltage-current transconductance of the amplifier would then effectively vary the 'resistance' constant of the integrator.

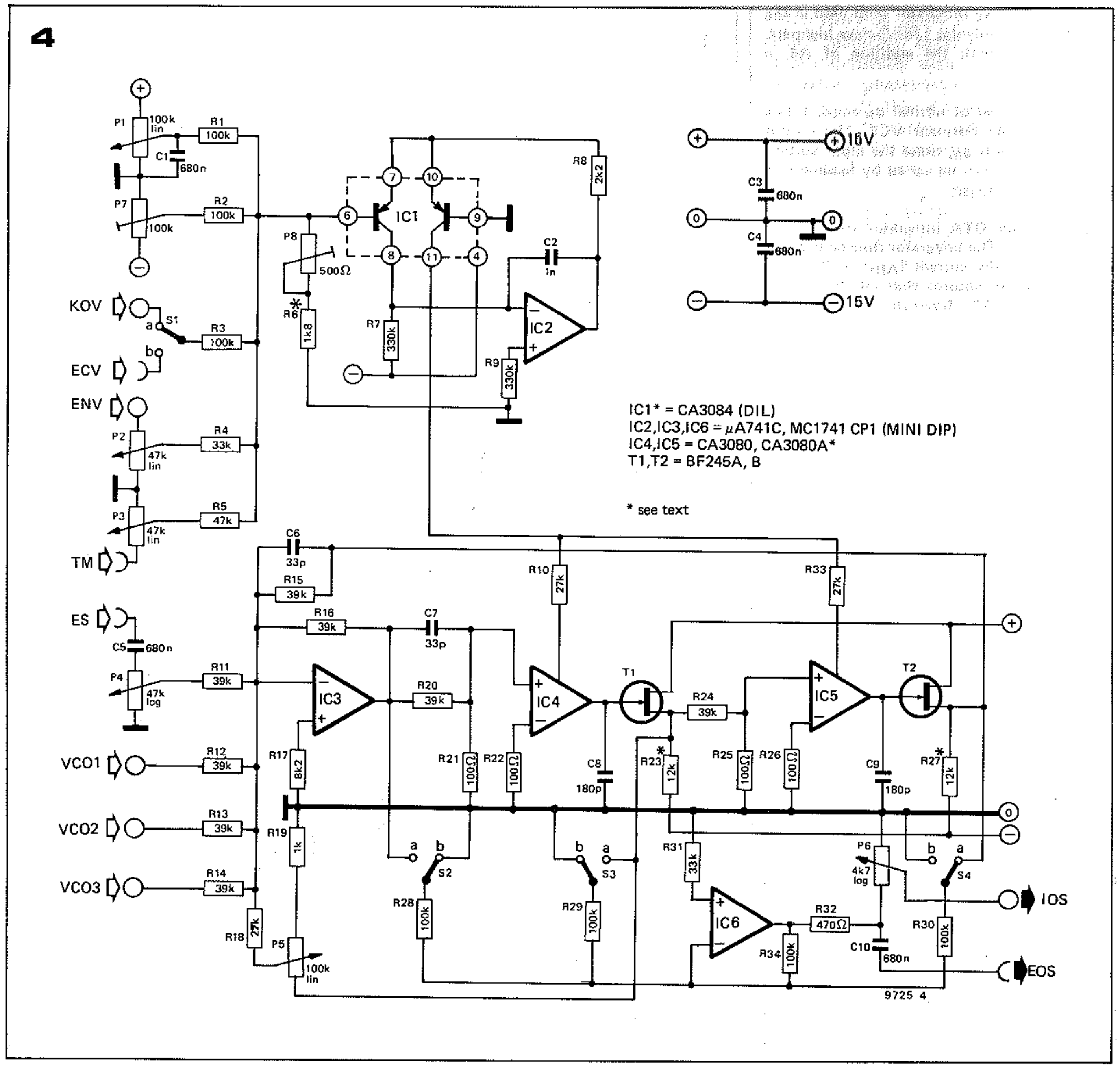
A suitable device exists ready-made in the shape of the operational transconductance amplifier or OTA. This is

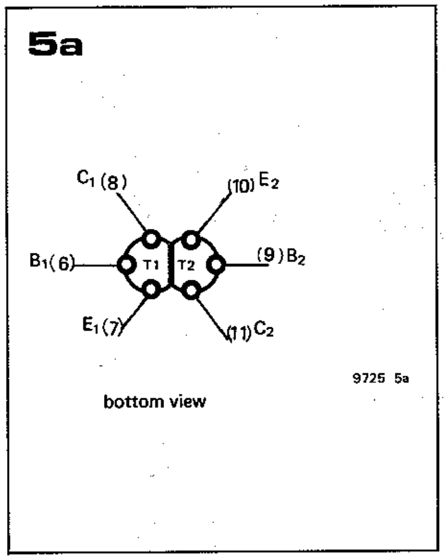
Hardwired inputs:
KOV = Keyboard Output Voltage (from interface receiver).
ENV = Envelope shaper control voltage (from ADSR unit).
VCO 1, 2, 3 = From VCOs 1, 2 and 3.

Front-panel inputs:
ECV = External Control Voltage.
TM = Tone colour ('Timbre') Modulation input.
ES = External Signal, e.g. noise, input.

Outputs:
VCF/IOS = Internal Output Signal from VCF, (will be hardwired to a VCA).
EOS = External Output Signal from VCF (front panel output).

Front-panel controls:
OCTAVES = P1, coarse frequency adjustment.
ENV = P2, sets envelope shaper control voltage.
TM = P3, sets tone colour modulation level.
ES = P4, sets external signal level.
Q = P5, Q-factor adjustment.
OUT = P6, sets VCF/IOS output level (not EOS!).
ECV/KOV = S1, selects external or internal control voltage input.
HP = S2, selects high-pass output.
BP = S3, selects bandpass output.
LP = S4, selects low-pass output.
N = S2 + S4, selects notch (band-stop) output.





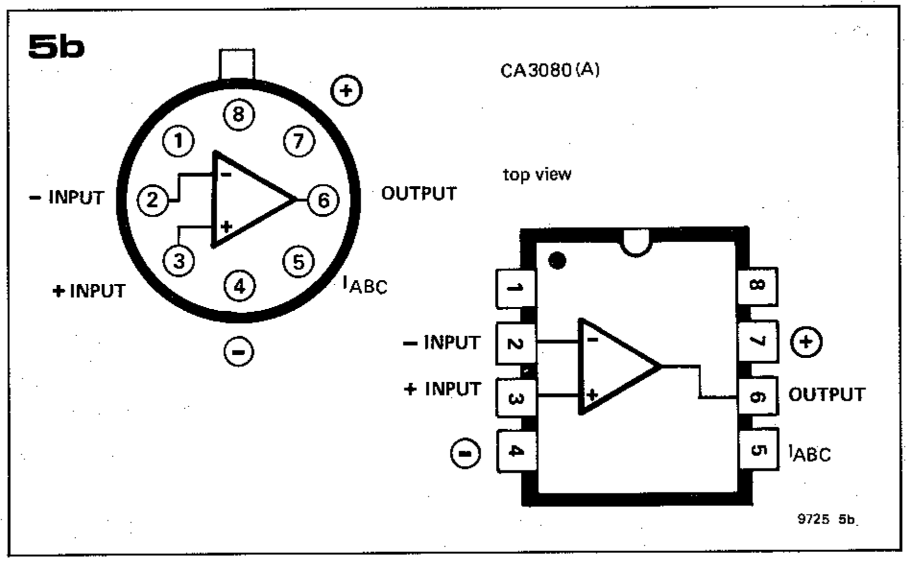


**Figure 4. Complete circuit of the Formant VCF, which consists of a voltage-current exponential converter and a linear current-controlled filter.**

**Figure 5a. Two well-matched PNP transistors may be used in place of IC1 for greater economy. The pin numbers shown correspond to the pinout of IC1.**

**Figure 5b. The CA 3080 is available in two packages. If the TO- package is used the leads must be bent to fit the DIP layout on the p.c.b.**

an amplifier that produces an output current which is proprotional to the input voltage, i.e. $i = g_m \cdot u_i$, where i is the output current, $u_i$ is the input voltage and $g_m$ is the transconductance. The feature of the OTA which makes it ideal for the VCF is that the transconductance $g_m$ is determined by a control current $I_{ABC}$, thus $g_m = k \cdot I_{ABC}$, where k is a constant. This is illustrated in figure 2.

For the CA 3080 OTA used in the Formant VCF the constant k is $19.2\ V^{-1}$ at an ambient temperature of 25° C, and so $g_m = 19.2 \times I_{ABC}$ mS (milliSiemens = milliamps/volt). This IC is particularly suitable because of the outstanding linearity of its transconductance characteristic over three decades of control current, and because of its relatively small tolerance in the value of 'k' (2:1 for the 3080 and 1.6:1 for the 3080A). However good linearity is achieved only for small input signals, and the input voltage must be attenuated to about ± 10 mV when used in the VCF.

Figure 3 shows the circuit of the integrator used in the Formant VCF. The input voltage is attenuated by the potential divider connected to the inverting input, and across the output is connected the 180 p integrating capacitor.

To maintain correct operation of the integrator the total output current of the OTA must flow into the integrator capacitor, which means that a buffer stage with a very high input impedance is required on the OTA output to avoid 'current-robbing'. A FET connected as a source-follower is used for this purpose.

The control current $I_{ABC}$ is fed in through a 27 k resistor. The integrator time constant is inversely proportional to the control current, so the VCF centre/turnover frequency is directly proportional to the control current.

## Complete circuit of the VCF

Figure 4 shows the complete circuit of the VCF. The actual filter circuit has a linear frequency characteristic and is current controlled. It must therefore be preceded by an exponential converter that converts the input control *voltage* into an exponentially related control *current*, so that the VCF tracks with the same 1 octave/V characteristic as the VCOs.

The exponential converter occupies the upper portion of the circuit, and is essentially similar to that of the VCOs. However, the control characteristic of the VCF does not need to be so accurate as that of the VCO, since a small error will only introduce minor, unnoticeable errors in amplitude response, whereas the same error in the VCO characteristic would cause unacceptable tuning errors.

For this reason the VCF exponential converter is provided only with a passive input adder (cf. figure 2a of the last chapter), and temperature stabilisation of the exponentiator is dispensed with, thus saving the cost of a not inexpensive μA726 IC. However, temperature compensation is retained in the form of a matched transistor pair. The circuit differs here from the VCO since the exponentiator must source current into the OTAs rather than sinking it as in the VCO, so PNP transistors are used.

Since temperature stabilisation is not used, a number of options are open for the choice of the matched transistor pair. Those who have access to a good transistor tester or curve tracer can select a matched pair of any small signal medium gain ('B' spec) transistors such as the BC 179B, BC 159B, BC 557B etc. These are then glued together with epoxy adhesive for good thermal tracking as shown in figure 5a, taking care that there is no electrical contact between the cases if metal-can types are used. (Note that the pin numbers given in figure 5a correspond to the IC pinning in figure 4).

The preferred solution is to use a CA 3084 transistor array, which is what was used in the prototype, but if this is difficult to obtain then almost any dual PNP transistor, such as the Analog Devices AD 820 . . . AD 822, Motorola 2N3808 . . . 2N3811 or SGS-ATES BFX 11, BFX 36, will do.

Note that the value shown for R6 (1k8) is correct when using the CA 3084. If a dual transistor is used, it is advisable to reduce the value of R6 to 1k5.

The current-controlled filter consists of IC3, IC4 and IC5. It will be noted that the integrators IC4 and IC5 are non-inverting. This does not affect the operation of the circuit, since non-inversion has the same effect as the double inversion that takes place in figure 1. However, it does ensure that the three outputs of the filter are in the same sense, whereas in figure 1 the bandpass output is inverted with respect to the other two outputs.

IC6 functions as an output buffer, and also as a summing amplifier for the highpass outputs to provide the notch function. By setting S2, S3 or S4 in position 'a', highpass, lowpass or bandpass functions respectively may be selected. By setting both S2 and S4 in position 'a' the notch function is obtained. Since IC3 is connected as an inverting amplifier and IC6 also inverts, this double inversion means that the output signal is non-inverted with respect to the input signals. The overall gain of the VCF (in the passband) is x 1 (0dB).

## Inputs, controls and outputs

The exponential converter section is equipped with a coarse octave tuning control P1 (note the absence of a fine control as compared with the VCO) and two presets P7 and P8 to adjust the offset and octave/V characteristic.

KOV and ECV control inputs are provided, as for the VCO. The input for envelope shaper control (ENV) is adjustable by means of P2. The tone colour modulation input controlled by P3/(TM) is analogous to the FM input of the VCO, i.e. it allows the centre/turnover frequency of the VCF to be modulated. There are four signal inputs, three internally-wired VCO inputs and one external signal (ES) input, whose amplitude can be controlled by P4. The Q-factor of the filter is controlled by P5.

Switches S2 to S4 select the desired filter type, as has already been described. Two outputs are provided, an uncontrolled output EOS which is brought out to a front-panel socket, and an internal output IOS, which is controlled by P6.

## Construction

A printed circuit board and component layout for the VCF are given in figure 6. The same considerations of component quality apply to the VCF that apply to all parts of the synthesiser. As mentioned earlier, two basic versions of the CA 3080 are available. The CA 3080A has better specifications as regards tolerance, and extended temperature range, but the basic CA 3080 is quite adequate (assuming that the synthesiser is not to be used in Antarctic blizzards).

The CA 3080 is available in two packages, TO- can and mini-DIP, both of which are shown in figure 5b. The p.c. board is laid out for the mini-DIP version, but the TO- version can easily be accomodated by splaying out the leads to conform with the mini-DIP pinning (in fact some TO- package 3080s are supplied with this already done).

The FETs T1 and T2 must be tested as detailed in chapter 3 and their source resistors R23 and R27 selected in accordance with Table 1 of that chapter.

A front panel layout for the VCF is given in figure 7, and a wiring diagram for the front-panel mounted components is shown in figure 8.

## Testing and adjustment

During assembly, it is convenient to use IC sockets so that the current-controlled fiter section of the circuit can be tested independently of the exponential converter. To test the CCF, IC1 is removed and a 100k log potentiometer is connected 'back-to-front' between ground and -15V (i.e. so that the end of the track approached by clockwise rotation of the wiper is connected to ground).

A multimeter set to the 100 μA range is

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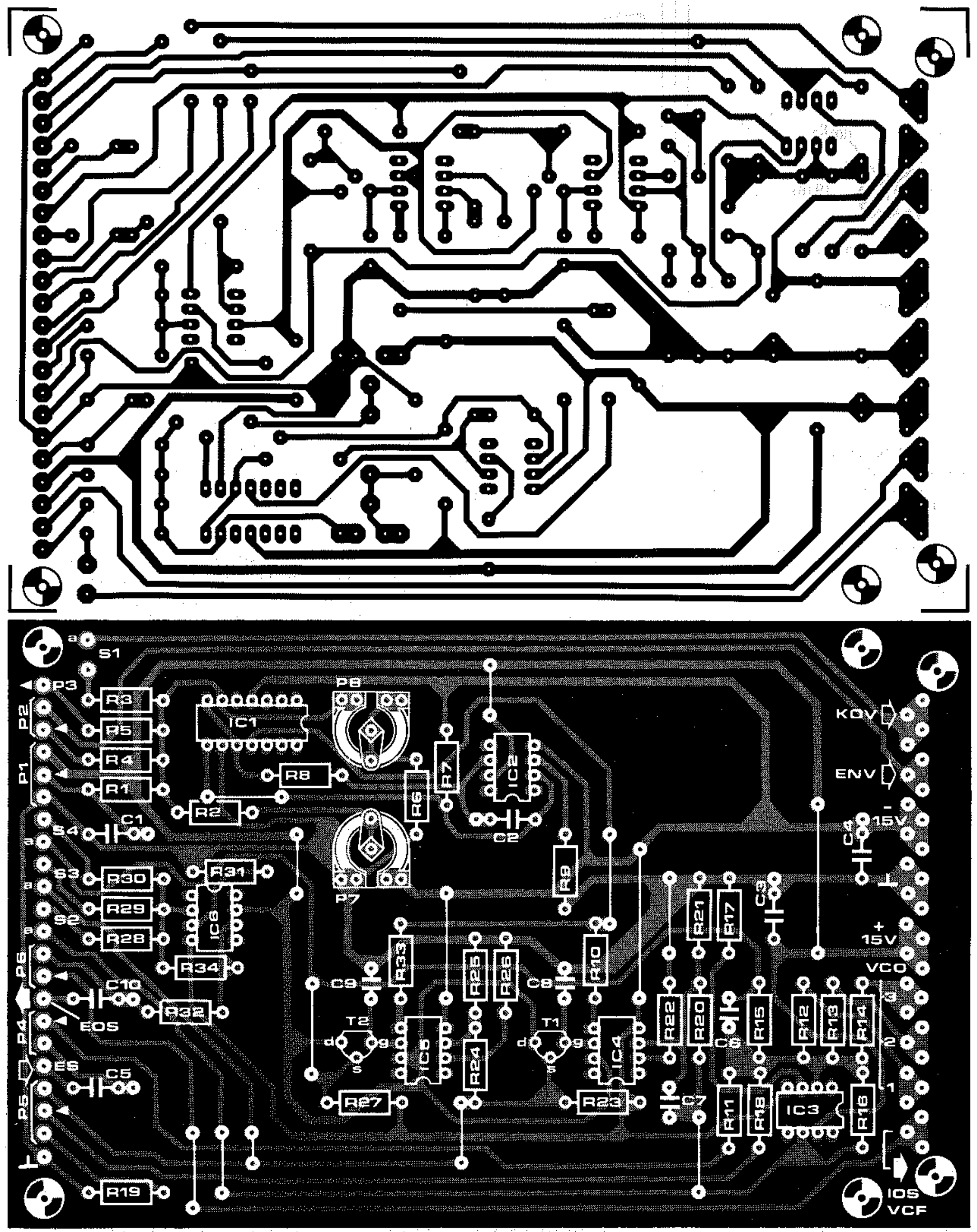

**Figure 6. Printed circuit board and component layout for the VCF. (EPS 9724-1).**

**Parts List**

**Resistors:**
R1, R2, R28, R29,
R30, R34 = 100 k
R3 = 100 k (1% metal oxide)
R4 = 33 k
R5 = 47 k
R6 = 1k8 (see text)
R7, R9 = 330 k
R8 = 2k2
R10, R33 = 27 k
R11, R12, R13, R14,
R15, R16, R20, R24 = 39 k
R17 = 8k2
R18 = 22 k
R19 = 1 k
R21, R22, R25, R26 = 100 Ω
R23, R27 = 12 k (nominal value, see text)
R31 = 33 k
R32 = 470 Ω

**Potentiometers:**
P1, P5 = 100 k lin
P2, P3 = 47 k (50 k) lin
P4 = 47 k (50 k) log
P6 = 4k7 (5 k) log

**Presets:**
P7 = 100 k
P8 = 470 Ω (500 Ω)

**Capacitors:**
C1, C3, C4, C5, C10 = 680 n
C2 = 1 n
C6, C7 = 33 p
C8, C9 =180 p

**Semiconductors:**
IC1 = CA 3084 (DIL) see text.
IC2, IC3, IC6 = μA 741 C (Mini DIP), MC1741 CP1 (Mini DIP).
IC4, IC5 = CA 3080 (A)
T1, T2 = BF 245a, b.

**Miscellaneous:**
31-way plug (DIN 41617)
S1 – S4 = miniature SPDT toggle switch

connected between the wiper of the potentiometer and the junction of R10 and R33, an input signal is provided to the VCF from a sinewave generator or from the VCO, and the Bandpass output is monitored on an oscilloscope. The test then proceeds as follows:

1. Set the Q-factor of the filter to maximum (wiper of P5 turned towards R19).
2. By means of the 100k log potentiometer set the control current to 50 μA on the meter.
3. Slowly increase the generator frequency from about 300 Hz to 1500 Hz; somewhere in this range the VCF output should peak as its resonant frequency is reached (i.e. there will be a sharp increase in output at a particular frequency with a fall-off on each side). Note the frequency at which resonance occurs.
4. Increase the control current to 100 μA and check that resonance now occurs at twice the previously noted frequency.

Note. Tests 2 to 4 are intended to check the linearity of the filter frequency v. control current characteristic. The tolerance in the absolute value of filter frequency for a given control current is due to OTA tolerances and is unimportant provided linearity is maintained i.e. the filter frequency doubles for each doubling of control current.

5. Set the generator to about 50 Hz and check that it is possible to obtain resonance at this frequency by varying the control current with the 100 k potentiometer. Repeat this test at 15 kHz.

The exponential converter can now be tested after inserting IC1 and removing IC4 and IC5. A multimeter set to the 100 μA range is connected from the bottom end of R10 to -15V and the wiper voltage of P1 is monitored with a voltmeter.

The test and adjustment now proceed as follows:

1. Set P8 to its mid-position, and turn P1 fully anticlockwise so that its wiper voltage is zero. Adjust P7 until the mircroammeter reading is 50 μA.
2. Turn P1 clockwise until its wiper voltage is 1V, then adjust P8 until the microammeter reads 100 μA.
3. Repeat the procedure for 2V, 3V, 4V etc. on the wiper of P1, checking that the exponentiator output current doubles for every 1V increase.

### Offset adjustment

Now that the two sections of the VCF have been checked, IC4 and IC5 can be re-inserted so that the entire VCF can be checked as a functional unit, as follows:

1. A squarewave with 50% duty-cycle at a frequency of about 500 Hz is fed to one of the filter inputs. P1 is turned fully clockwise and P7 is turned anticlockwise.
2. The lowpass output of the VCF is

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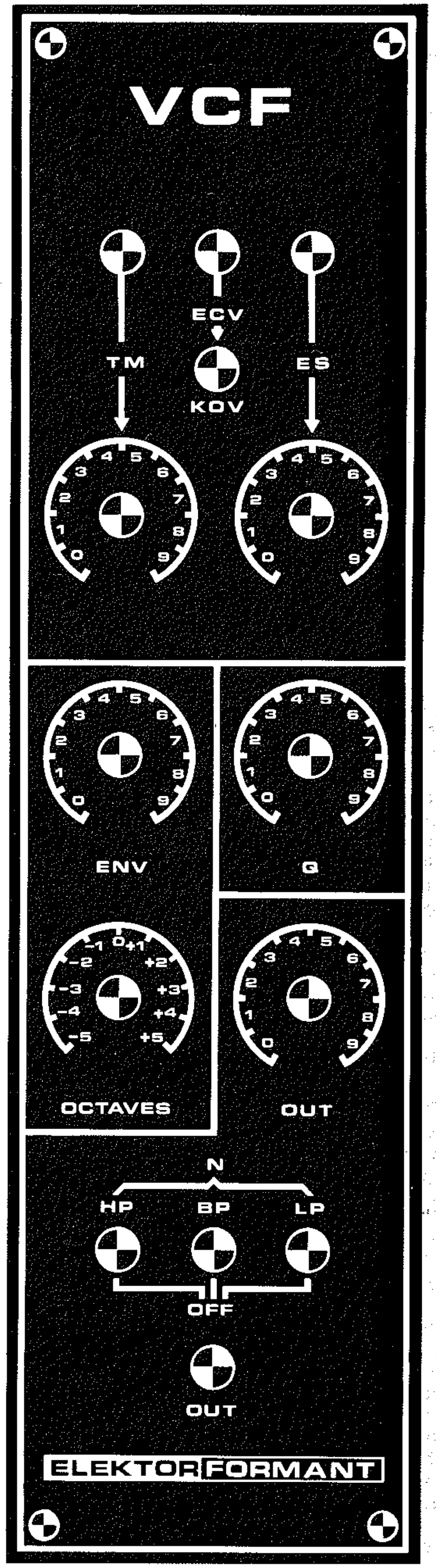

**Figure 7. Front panel layout for the VCF.**

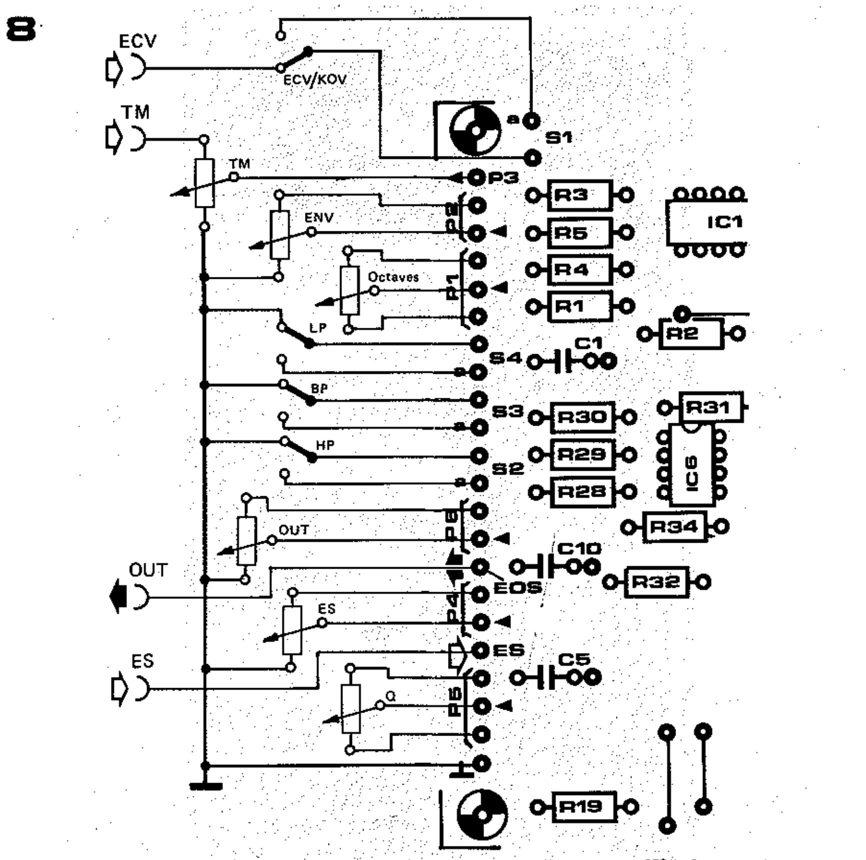

Figure 8. Wiring diagram for the panel mounted components.

monitored on an oscilloscope, and at this stage should appear at the output without degradation.

3. If the wiper of P7 is now turned slowly clockwise the leading edge of the squarewave will start to be rounded off as the turnover point of the filter is reduced. To carry out the offset adjustment with P7 its wiper is turned as far clockwise as is possible without significantly degrading the square waveform (just a slight rounding of the top corner is acceptable, but this adjustment does not have to be particularly precise).

### Octaves/Volt adjustment

The octave/V characteristic of the VCF can be adjusted by seeing how well it tracks against a previously calibrated VCO. To do this, the KOV input is connected to the VCO and the VCF, and the sine output of the VCO is connected to the VCF input. The adjustment procedure is as follows:

1. Switch off the main tuning of the keyboard, depress top C of the keyboard and use the octaves control of the VCO to set its frequency to about 500 Hz.
2. Set the Q control, P5, of the VCF to maximum, monitor the bandpass output of the VCF and adjust P1 until the VCF output peaks. As the filter is easily overloaded at high Q-factors it may be necessary to reduce the VCO output voltage.
3. Depress the key two octaves lower and adjust P8 until the VCF output again peaks.
4. Depress top C again and if necessary readjust P1 so that the output peaks.
5. Repeat 3 and 4 until no further readjustment is necessary for the output to peak when changing from one note to the other.
6. The offset adjustment may have been disturbed, so check this and if necessary readjust P7 as described in the offset adjustment procedure.
7. Repeat 3 onwards until no further improvement can be obtained.

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## chapter 7

# 24 dB VCF

**Because of the greater range of tonal possibilities they offer, VCFs with an extremely steep slope seem to have a particular appeal for most synthesiser enthusiasts. The design presented here is for a VCF offering a choice of lowpass or highpass functions and a filter slope of 6, 12, 18 or 24 dB per octave.**

### New possibilities

It should be stated at the outset that the 24 dB VCF is not intended to replace the 12 dB design. On the contrary, the two filters are complementary to one another and can be used in combination to provide greatly increased possibilities for tailoring the harmonic structure of the sounds produced by Formant.

For example, the 12 dB VCF can be used in the bandpass mode together with the steep filtering of the 24 dB VCF to produce selective tone coloration. The two filters can be controlled by the same envelope shaper or by different envelope shapers, and may be connected in cascade or in parallel. The latter arrangement offers several interesting possibilities. For example, hard, metallic sounds can be produced by applying a short, steep envelope voltage to the 12 dB VCF and a longer, shallower contour to the 24 dB VCF.

If the filter inputs are connected in parallel then interesting effects may be obtained by connecting one VCF output to one input of a stereo amplifier and the other VCF output to the other input. This gives rise to a very distinctive dynamic amplitude characteristic and stereo imaging, particularly if the two VCFs are controlled by different envelope shapers.

The audible differences between the 12 dB VCF and the 24 dB VCF are quite prominent. The 12 dB VCF produces sounds that are distinctly 'electronic', which can have a slightly fatiguing effect on the listener during extended playing sessions. The sounds produced by the 24 dB VCF, on the other hand, are much more 'natural', and can be listened to for extended periods without fatigue. This effect is probably due to the more severe filtering of higher harmonics which the 24 dB VCF provides when used in the lowpass mode, since these harmonics tend to make the sound of the 12 dB VCF much more shrill than that of the 24 dB VCF.

The effect of the steeper filter slope of the 24 dB VCF is illustrated in figure 1, which shows the different outputs from the 12 dB VCF (dotted line) and 24 dB VCF (continuous line) when fed with a sawtooth waveform. It is apparent that, due to the almost complete removal of the harmonics of the sawtooth, the output of the 24 dB VCF is practically a sinewave, whereas the original waveform is still apparent at the output of the 12 dB VCF since the harmonics are only partially removed.

It is clear from the foregoing that a 24 dB VCF greatly extends the musical possibilities of a synthesiser and is virtually a must for the serious user.

### Design of the 24 dB VCF

The design of the basic filter section shown in figure 2 is very similar to that of the 12 dB VCF, which was described in detail in the previous chapter. However, advantage has been taken of recent developments in FET op-amp technology to simplify the design slightly. As has been explained, the basic filter section is an integrator or 6 dB/octave lowpass section consisting of an OTA driving a capacitor. The voltage/current transconductance ($g_m$) of the OTA can be varied by an external control current and hence, via an exponential voltage/current converter, from an external control voltage. This control current alters the time constant of the integrator and hence the turnover frequency of the filter section.

The output current of the OTA must all flow into the capacitor, otherwise the integrator characteristic will be less than ideal. This means that the output of the OTA must be buffered by an amplifier with a high input impedance. In the

**Figure 1.** This illustrates the difference between the outputs of a 12 dB/octave VCF and a 24 dB/octave VCF having the same turnover frequency, when fed with a sawtooth input. The 24 dB VCF removes practically all the harmonics giving a sinewave output, whereas the original waveshape is still distinguishable at the output of the 12 dB VCF.

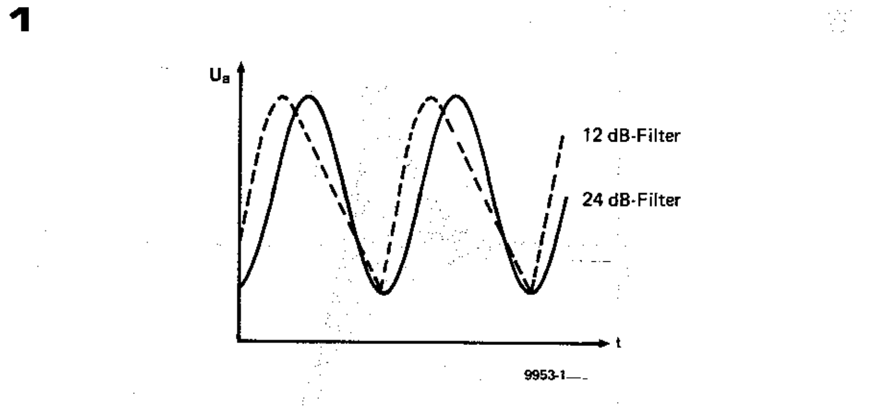


**Figure 2.** The basic filter section of the 24 dB VCF is the same as that of the 12 dB VCF, i.e. an OTA integrator followed by a FET op-amp buffer.

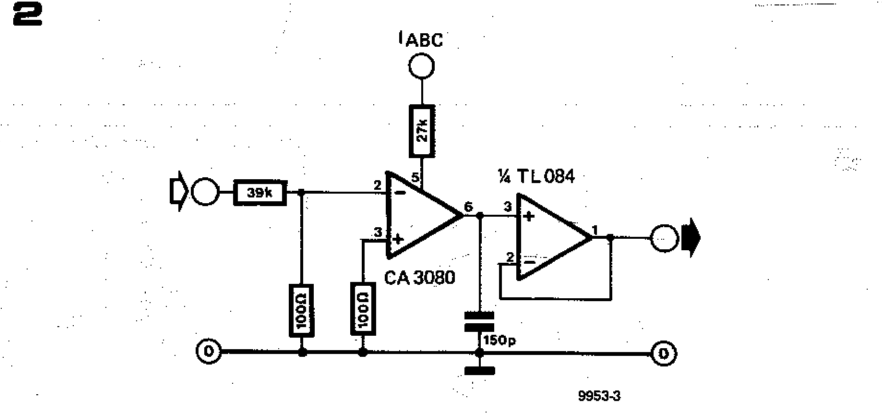


**Figure 3.** The highpass function is obtained by connecting the 6 dB lowpass section in the feedback loop of an operational amplifier.

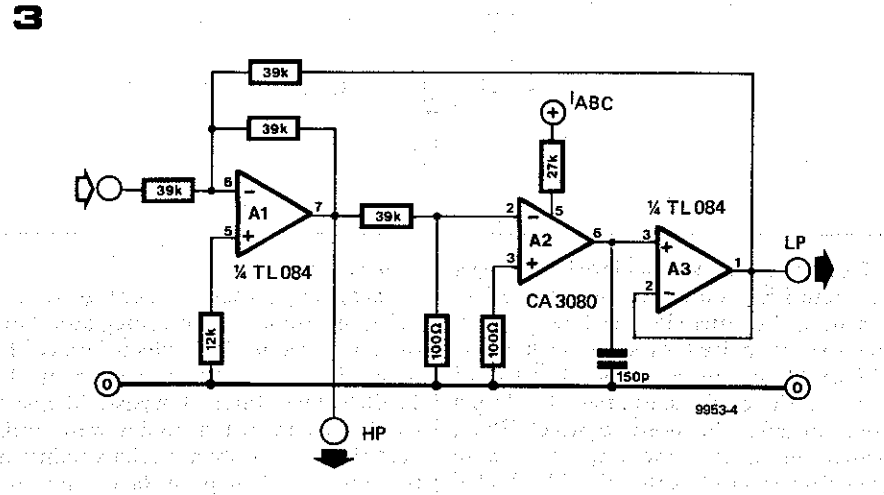
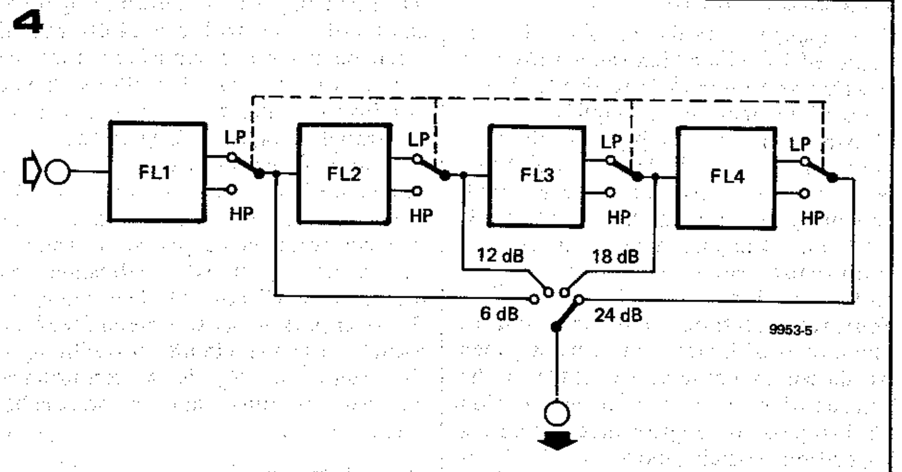


**Figure 4.** To obtain a 24 dB/octave filter, four 6 dB/octave sections are cascaded.



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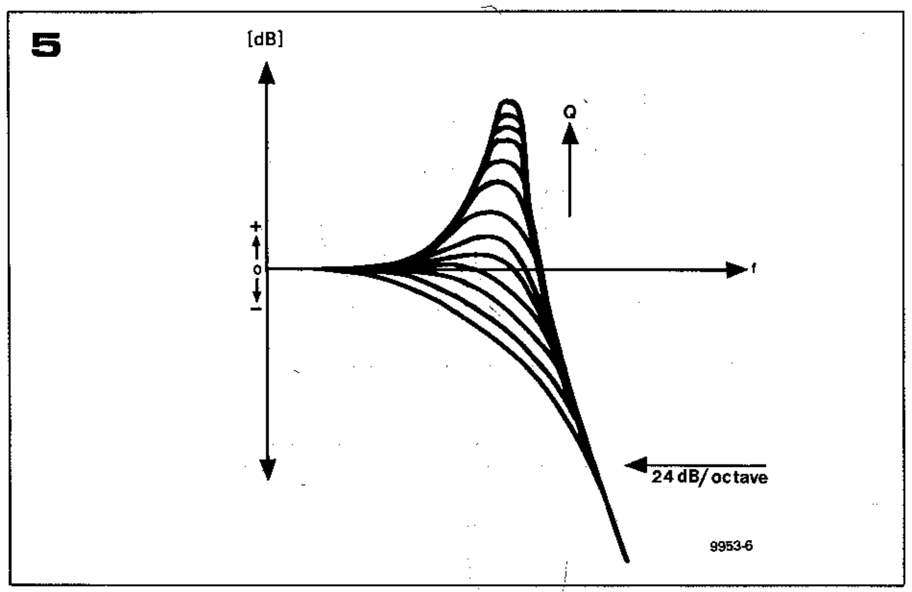

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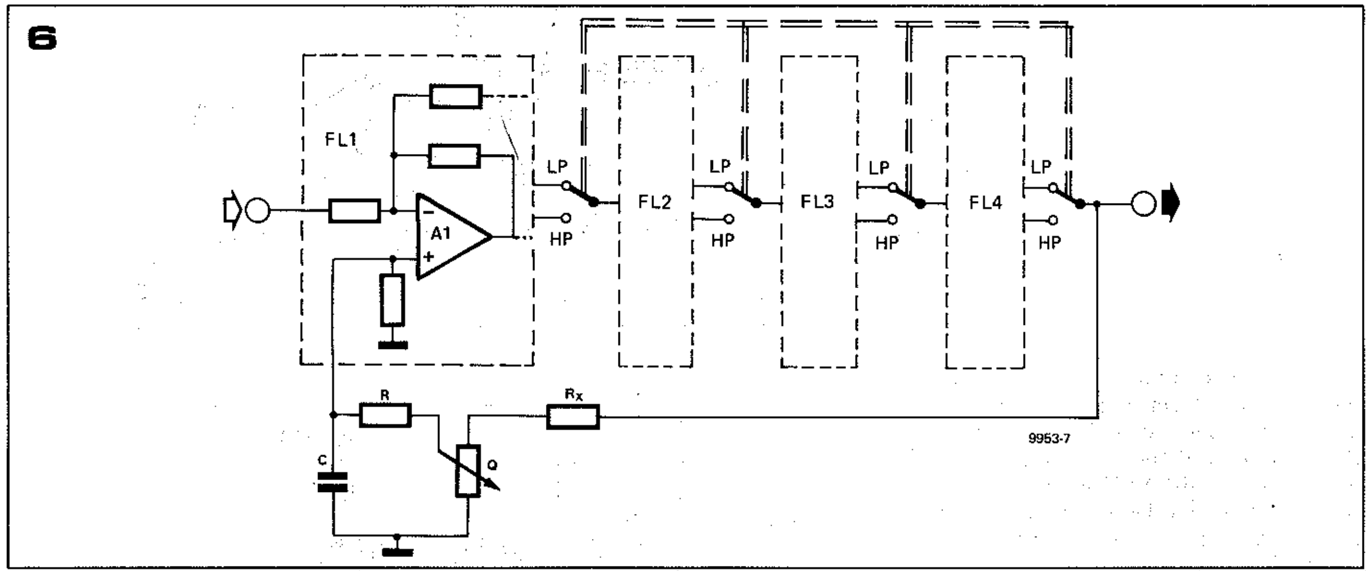


**Figure 5. Positive feedback around the entire filter allows the response to be boosted about the turnover frequency. The degree of boost can be varied by a 'Q' control.**

**Figure 6. Block diagram of the 24 dB/octave filter, showing how the Q control is incorporated.**

**Figure 7. Complete circuit of the 24 dB VCF. The exponential voltage/current converter is identical to that used in the 12 dB VCF.**

12 dB VCF this was achieved by using a discrete FET source follower and a 741 op-amp. Fortunately, relatively inexpensive quad FET op-amps such as the Texas TL084 are available. The use of one of these ICs simplifies the design and obviates the need to select FETs, which becomes something of a chore when one considers that the 24 dB VCF uses four integrator stages.

## Highpass function

The highpass mode of the filter is achieved by connecting the 6 dB/octave lowpass section in the negative feedback loop of an operational amplifier, A1, as shown in figure 3. A highpass filter response is then available at the output of A1 whilst a lowpass response is simultaneously available at the output of A3. Of course, this arrangement gives only a 6 dB/octave slope per section, and in order to obtain a 24 dB/octave filter four filter sections, built according to the circuit of figure 3, must be cascaded as shown in figure 4. Switching at the output of each section allows selection of highpass or lowpass mode, whilst a 4-position switch allows 1, 2, 3, or 4 filter sections to be switched in to give 6-, 12-, 18-, or 24 dB/octave slopes respectively.

It is apparent that this arrangement is different from the two-integrator loop or state-variable filter which formed the basis of the 12 dB/octave filter. In the 12 dB/octave filter, lowpass, highpass, bandpass and notch modes were available simultaneously at various points in the circuit, though in fact only one function at a time could be selected at the output.

An interesting effect, shown in figure 5, can be obtained with the 24 dB VCF if a feedback loop is connected from the output of the cascaded filters to the non-inverting input of the first stage as illustrated in figure 6. Due to the phase shift around the turnover frequency this causes positive feedback, which boosts the gain of the filter around the turnover frequency as shown in figure 5. The degree of boost is adjustable by means of a 'Q' control. The choice of $R_X$ is important as too much feedback would cause the circuit to oscillate, so the value of $R_X$ is a compromise between stability and a reasonable degree of boost.

## Complete circuit

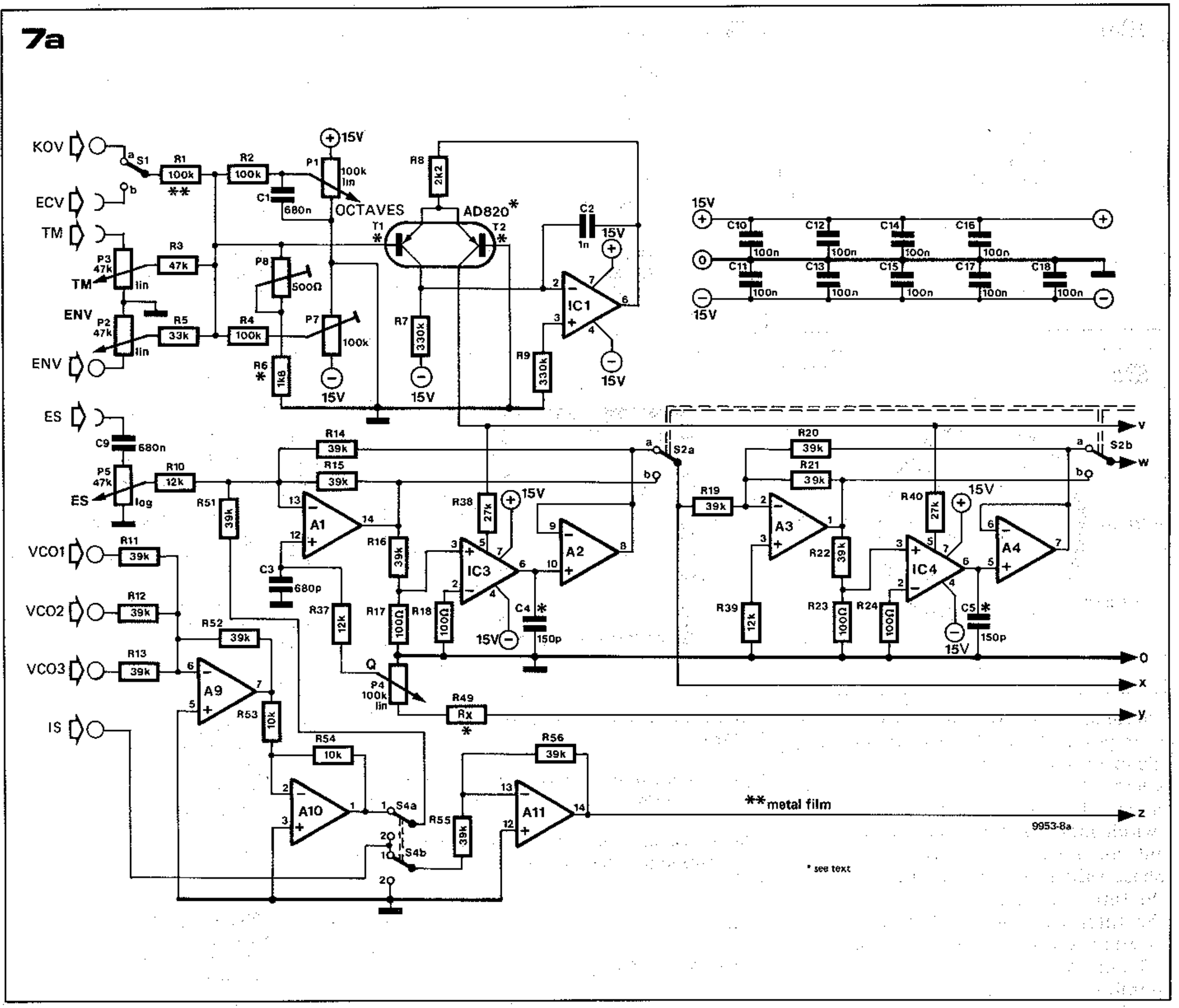
The complete circuit of the 24 dB VCF is given in figure 7. The exponential converter, constructed around T1, T2 and IC1, is identical to that used in the 12 dB VCF and gives the same 1 octave per volt characteristic to the turnover frequency of the filter. The control voltage inputs are also the same as for the 12 dB VCF, and are listed in table 1.

Since the 24 dB VCF must have the option of being connected in parallel or in cascade with the 12 dB VCF, the input switching arrangements are a little complicated. A9 and A10 form a non-inverting summing amplifier for the three VCO inputs, whilst the output of the 12 dB VCF is fed in via the IS connection. With S4 in position 2 the output of A10 is disconnected, so the VCO inputs are inhibited. The output of the 12 dB VCF is fed to the input of the 24 dB VCF via S4 and R51, so that the two VCFs are in cascade.

With S4 in position 1 the output of A10 is connected to the inputs of the 24 dB VCF, whilst the output of the 12 dB VCF is routed through A11. The output of A11 and the output of the 24 dB VCF are added together in the output summing amplifier A12, i.e. the two VCFs are connected in parallel.

The four 6 dB/octave filter sections

7a



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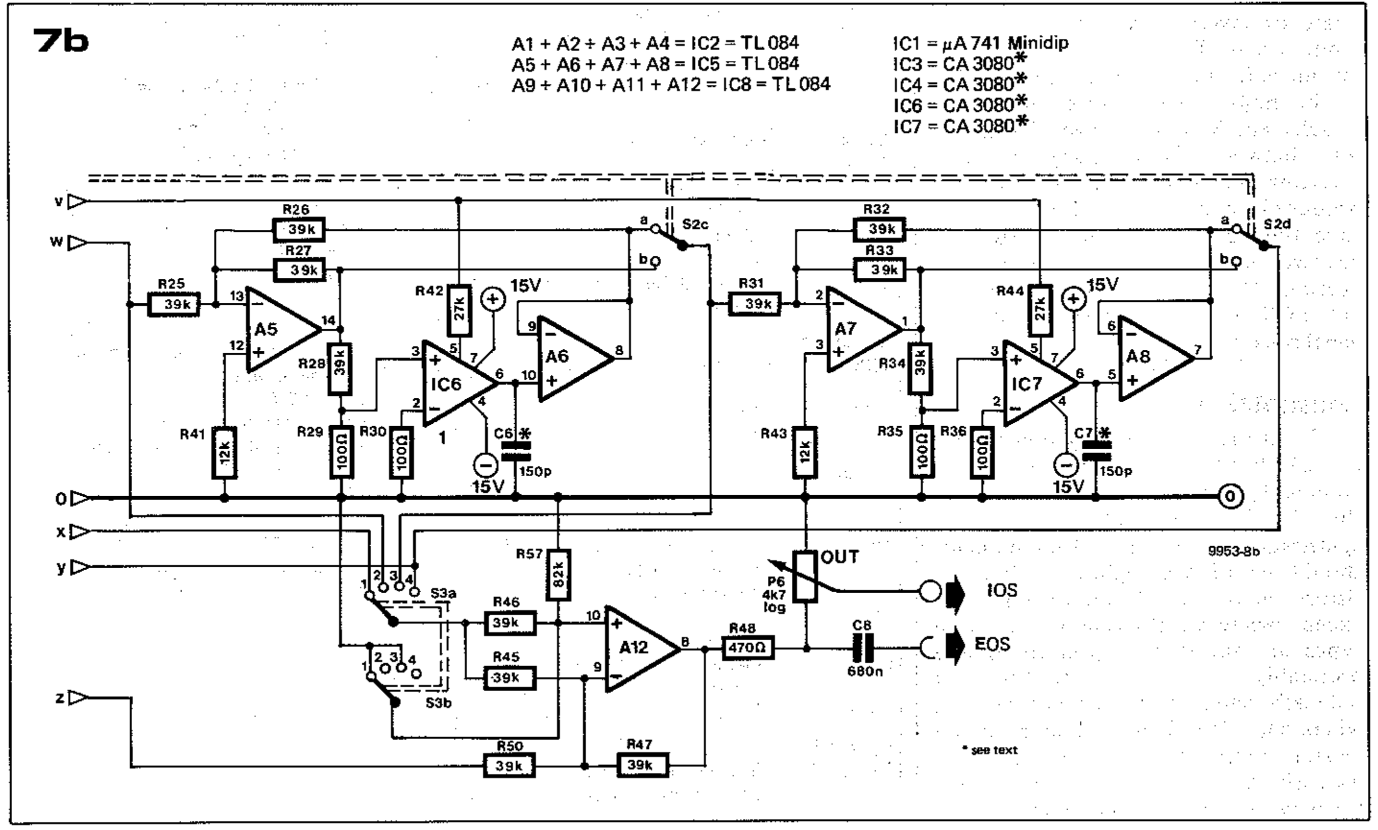


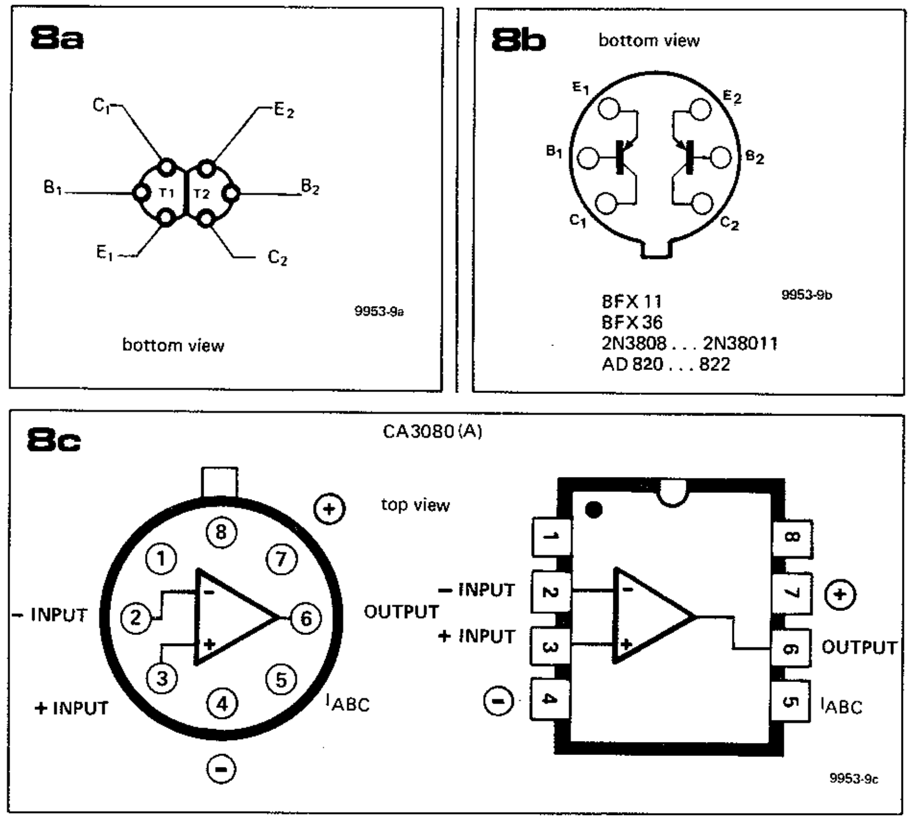


**Figure 8. Pinouts for the dual transistors and CA3080.**

comprise A1 to A8 and IC3 to IC7. The four poles of switch S2 select between highpass and lowpass modes, while S3 selects the filter output and hence the slope. The reason that S3 is a two-pole switch may not be immediately apparent, but is easily explained. Ignoring the phase shift introduced by the action of the filter, i.e. considering only signals in the filter passband, each filter section inverts the signal fed to it, since A1, A3, A5 and A7 are connected as inverting amplifiers. This means that the outputs of alternate filter sections are either in phase or inverted with respect to the input signal. To ensure that the filter output is in the same phase relationship to the input signal whatever filter slope is selected, S3b is arranged to switch A12 between the inverting and non-inverting modes to cancel the inversions produced by the filter sections.

Like the 12 dB VCF, the 24 dB VCF has two outputs, a hardwire output connection IOS and an uncommitted output, EOS, which is connected to a front panel socket.

## Construction

As far as the choice of components for the 24 dB VCF goes, the same general comments apply that were made about the 12 dB VCF and the Formant synthesiser in general. All components should be of the highest quality; resistors should be 5% carbon film types except where metal oxide or metal film types are specified; capacitors should preferably be polyester, polystyrene or polycarbonate, and must be these types where specified. Semiconductors should be from a reputable manufacturer.

As with the 12 dB VCF the dual transistor may be any of the types specified in the parts list, or may be home-made by gluing together two normal transistors, though in this case thermal tracking will not be quite so good. The CA3080 should preferably be in a MINIDIP package to fit the hole spacings on the p.c. board, though the metal can type can be made to fit by splaying the leads. The pinouts for the dual transistors and the CA3080 are given in figure 8.

Although not absolutely necessary, it is a good idea to select OTA's with approximately the same transconductance, since the four sections of the filter will then have almost the same turnover frequency. The CA3080 is available in two versions, the standard version, in which the ratio between the maximum and minimum $g_m$ is 2:1, and the CA3080A, in which the spread in $g_m$ is only 1.6:1. A test circuit and test procedure for selecting ICs with similar $g_m$ are given at the end of the chapter and it is certainly worthwhile buying a few extra OTAs and selecting the four with the most similar $g_m$. The 'reject' devices are perfectly acceptable for use in the 12 dB VCF or VCA, and need not be wasted.

The other ICs in the circuit should all be TL074 or TL084 quad **BIFET** op-amps, although for IC8 it is permissible to use an LM324. Thanks to the use of quad op-amps it is possible to accomodate the 24 dB VCF on a standard Eurocard-size (160 mm x 100 mm) p.c. board, although the control connections are not all on the front edge of the board. The printed circuit pattern and component layout for this board are

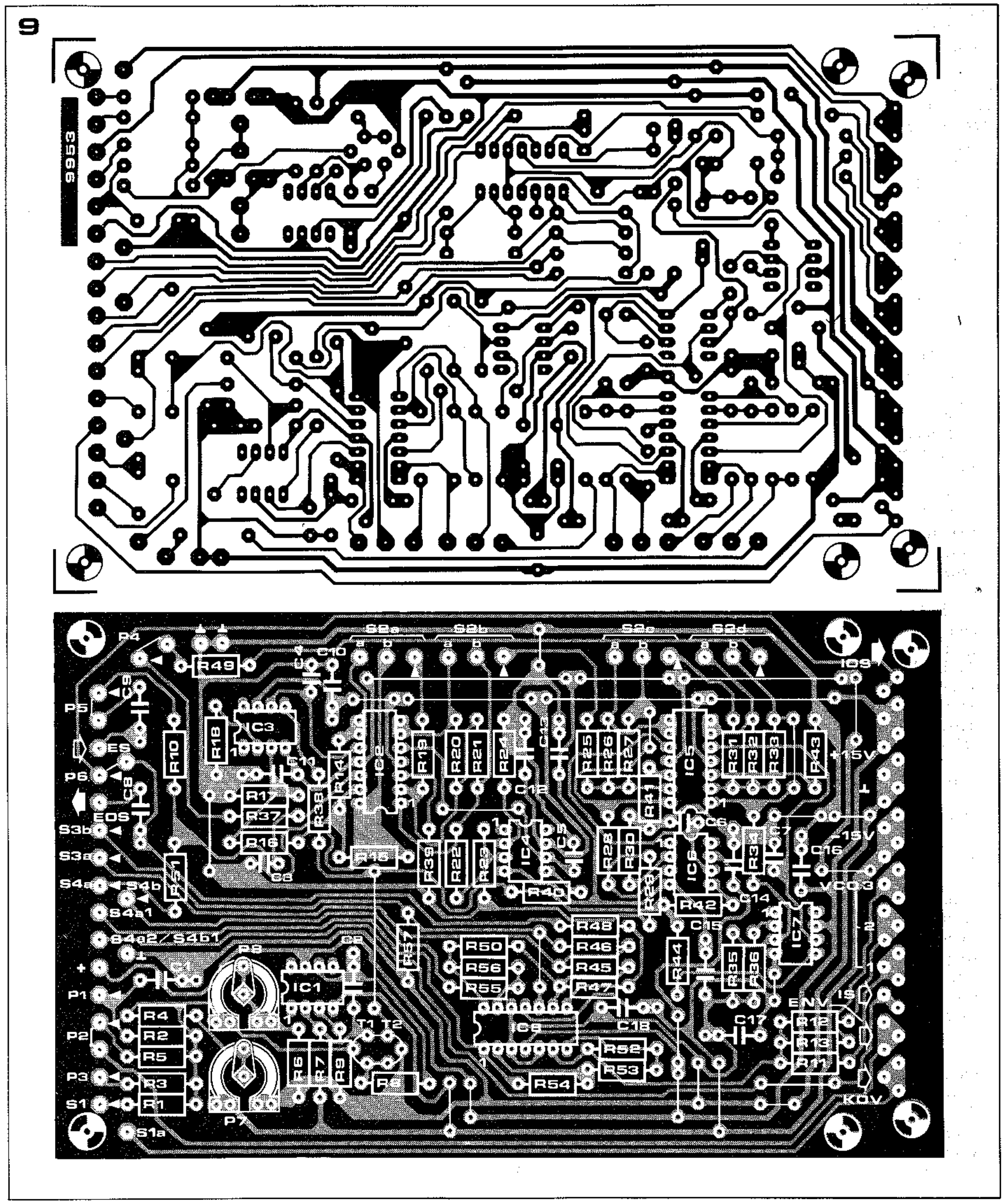


**Figure 9. Printed circuit board and component layout for the 24 dB VCF. (EPS 9953-1).**

**Table 1. Summary of the control functions and input/output connections of the 24 dB VCF.**

**Table 1**

a) hardwired inputs (not on the front panel)
KOV = Keyboard Output Voltage (from interface receiver)
ENV = Envelope shaper Control Voltage (from ADSR unit)
VCO 1,2,3 = Signals from VCOs 1, 2, 3
IS = Internal signal from the 12 dB VCF

b) external inputs (sockets on front panel)
ECV = External Control Voltage (for exponential generator of the VCF)
TM = Tone Colour Modulation input
ES = External Signal (from e.g. noise module)

c) outputs
IOS = Internal Output Signal (from VCF to VCA)
EOS = External Output Signal (socket on front panel)

d) controls
TM = P3; sets tone colour modulation level
ES = P5; sets external signal level
ENV = P2; sets envelope shaper control voltage
OCTAVES = P1; coarse frequency adjustment
Q = P4; sets level of peak boost around turnover frequency
OUT = P6; sets IOS output level

e) switches
ECV/KOV = S1; selects external or internal control voltage input

**Parts list to figures 8 and 10**

Resistors:
R1 = 100 k metal oxide
R2,R4 = 100 k
R3 = 47 k
R5 = 33 k
R6 = 1k8
R7,R9 = 330 k
R8 = 2k2
R10,R37,R39,R41,R43 = 12 k
R11 . . . R16,R19 . . . R22, R25 . . . R28,R31 . . . R34,R45, R46,R47,R50,R51,R52,R55, R56 = 39 k
R17,R18,R23,R24,R29,R30, R35,R36 = 100 Ω
R38,R40,R42,R44 = 27 k
R48 = 470 Ω
R49 = 100 k (see text)
R53,R54 = 10 k
R57 = 82 k

Potentiometer:
P1,P4 = 100 k linear
P2,P3 = 47 k (50 k) linear
P5 = 47 k (50 k) logarithmic
P6 = 4k7 (5 k) logarithmic
P7 = 100 k preset
P8 = 470 Ω (500 Ω) preset

Capacitors:
C1,C8,C9 = 680 n
C2 = 1 n
C3 = 680 p (polystyrene, not ceramic)
C4,C5,C6,C7 = 150 p (polystyrene, not ceramic)
C10 . . . C18 = 100 n

Semiconductors:
IC1 = 741
IC2,IC5 = TL084, TL074
IC8 = TL084, TL074, LM 324
IC3 . . . IC6 = CA 3080, CA3080A (MINIDIP or TO; see text)
T1,T2 = AD 820 . . . 822, 2N3808 . . . 3811, BFX 11, BFX 36 (see text) or 2 x BC 557B

Miscellaneous:
31-pin DIN 41617 connector or terminal pins
S1 = SPDT
S2 = 4-pole double throw
S3 = 2-pole 4-way; index angle approx. 30°
S4 = DPDT
4 minature sockets, 3.5 mm dia.
7 13 . . . 15 mm collet knobs with pointer (to match existing synthesiser modules).

given in figure 9, while a front panel layout is given in figure 10.

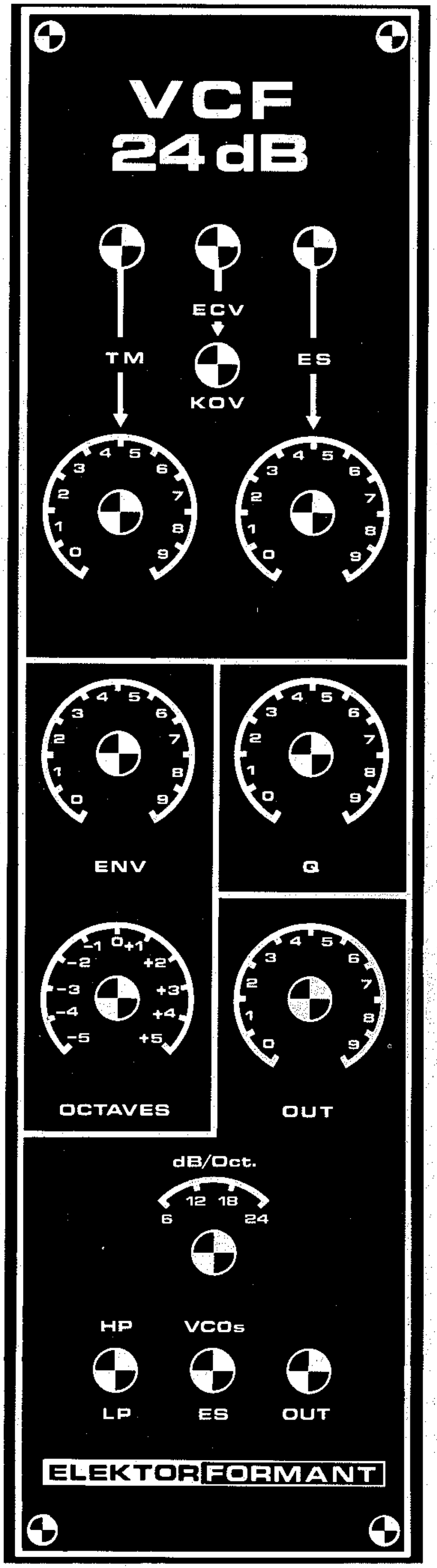
### Test and adjustment

To enable the exponential converter and the filter section to be tested separately they are joined by a wire link which runs across the board from T2 to a point adjacent to R15. This link should be omitted until the VCF has been tested.

To test the filter section it is necessary to provide a temporary control current. This is done by connecting a 100 k log potentiometer between −15 V and ground, with its wiper linked to the junction of R39 and R4 via a multimeter set to the 100 $\mu$A DC range. The test then proceeds as follows:

1. Turn the wiper of P4 fully towards ground, select 24 dB slope with S3 and adjust the control current to 100 $\mu$A.
2. Feed a sinewave signal into the ES socket and adjust either the sinewave amplitude or P5 for 2.5 V peak-to-peak measured on an oscilloscope at the wiper of P5.
3. Monitor the filter output on the 'scope and check the operation of the filter by varying the sinewave frequency and checking that the signal is attenuated above the turnover frequency in the lowpass mode and below the turnover frequency in the highpass mode.
4. The function of S3 should now be checked. Set S3 to the 6 dB position and S2 to the LP position. Increase the frequency of the input signal until the output of the filter is 6 dB down on (i.e. 50% of) what it was in the passband where the response was level. Now switch to 12 dB, 18 dB and 24 dB and check that the response is respectively 12, 18 and 24 dB down, i.e. is reduced to 25%, 12.5% and 6.25% of its original value. The exact results of this test will depend upon the matching of the OTAs.
5. Set the Q control, P4, to its maximum value, when the circuit should show no sign of oscillation. If the circuit does oscillate it will be necessary to increase the value R49. If it does not oscillate then the Q range can be increased by decreasing R49, taking care that instability does not occur.
6. Finally, the linearity of the turnover frequency v. control current characteristic should be checked. Adjust the input frequency until the response is a convenient number of dB down (say 6 dB). Double the control current then double the input frequency and the response should still be 6 dB down.
7. To check the exponential converter connect a 27 k resistor in series with a multimeter set to the 100 $\mu$A range between the collector of T2 and the −15 V rail. Then follow the test

10



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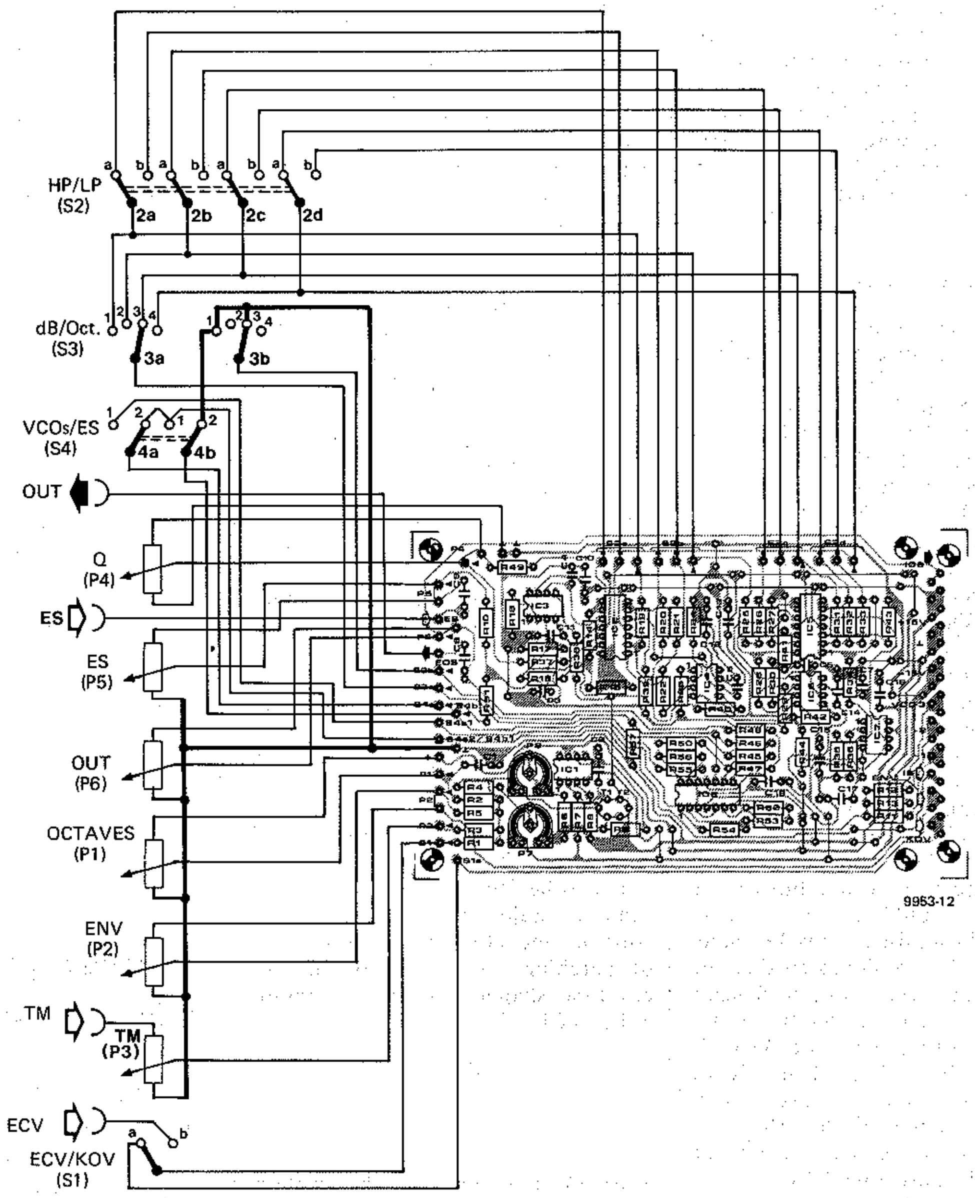

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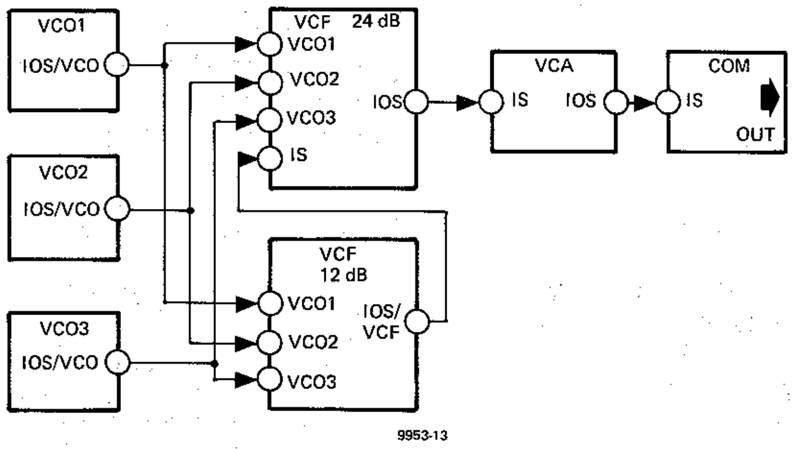

**Figure 10. Front panel layout for the VCF. (EPS 9953-2).**

**Figure 11. Showing the wiring between the p.c. board and the front-panel mounted components.**

**Figure 12. The 24 dB VCF is connected into the Formant system between the 12 dB VCF and the VCA.**