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

Using the 24 dB VCF

As can be seen from figure 12, the 24 dB VCF is connected between the 12 dB VCF and the VCA, so that the IOS output of the 12 dB VCF goes to the IC input of the 24 dB VCF instead of to the VCA, whilst the VCA receives its input from the IOS output of the 24 dB VCF. The 24 dB VCF also has inputs from the three VCOs.

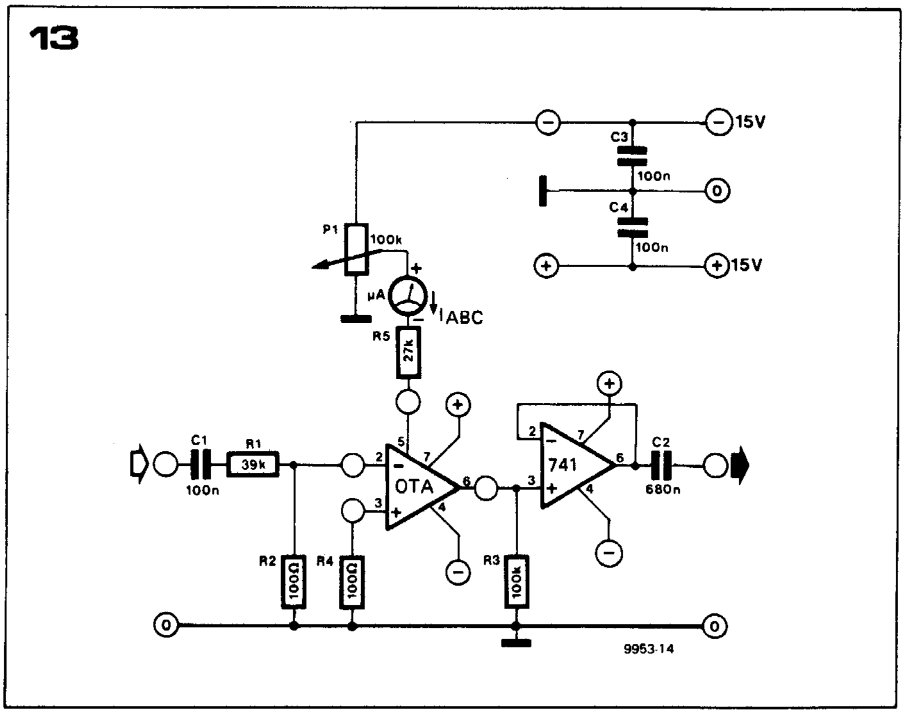
In addition to the signal connections the 24 dB VCF must also be provided with supply to the VCF module in accordance with the standard practice for Formant. Provision of control voltage inputs from the ADSR envelope shapers will be discussed later.

For satisfactory operation of the 24 dB VCF the correct setting of the input level is important, even more than in the case of the 12 dB VCF. On the one hand, the input level should not be so large that distortion occurs, but on the other hand it should not be so small that the signal-to-noise ratio is degraded. The 24 dB VCF is designed so that the optimum input level is obtained using three VCOs set to maximum output, with one waveform selected per VCO. If more than three VCOs are in use or more than one output waveform is selected from each VCO then the VCO output levels must be reduced. On the other hand, if only one VCO is used then the signal level may be too low. In this case it is best to patch the EOS socket of the VCO to the ES input of the VCF, since this input has approximately three times the sensitivity of the hardwired VCO inputs.

The 24 dB VCF is capable of the same basic functions as the 12 dB VCF; driven by the KOV control voltage it will operate as a tracking filter, whilst the ENV and TM inputs allow dynamic modulation of the harmonic content of the VCF output. Due to the greater slope of the 24 dB VCF the setting of the ENV level control is more critical than with the 12 dB VCF, but if correctly adjusted then subtle nuances in the tonal character of the output signal are possible.

The question arises as to which ADSR envelope shaper should be used to control the 24 dB VCF, since only two are built into the basic Formant system, and control the VCA and 12 dB VCF respectively. Because of the modular construction of Formant it is, of course, perfectly feasible to build a third envelope shaper, which is the most versatile arrangement. The alternatives are to patch one of the other ADSR outputs to the TM input of the 24 dB VCF, or to hardwire the ENV input of the 24 dB VCF to the output of the envelope shaper that controls the 12 dB VCF. This latter arrangement is probably preferable, as it allows the ADSR signal to be fed to one or both VCFs by suitable adjustment of their ENV controls and also allows the possibility of patching the output of the other envelope shaper into the TM input of either VCF.

Figure 13. Test circuit for the selection of OTAs.



OTA selection procedure

Although not absolutely essential, it is well worth selecting OTAs with closely matched transconductance characteristics to ensure that the four filter sections track accurately.

A test circuit for the OTAs is given in figure 13. This should be fed with a sinewave signal of about 2 V peak-to-peak (or 0.7 V measured on an AC voltmeter) from a signal generator or from one of the VCOs. The output should be monitored on a 'scope or AC voltmeter. With a control current of 100 μA, measured on the multimeter in series with R5, the output voltage should be between 0.7 V and 1.3 V peak-to-peak. Without changing the input level or control current the OTAs to be tested should be plugged into the circuit one at a time and the output level for each OTA noted. The four OTAs whose output levels are most similar should be used in the VCF.

The circuit can also be used to check the linearity of the transconductance v. control current characteristic of the OTAs, e.g. doubling the control current should double the output of the test circuit and halving the control current should halve the output.

chapter 8

resonance filter module

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

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

As has already been explained, basic factors influencing the characteristics of a muscial note are pitch, dynamic amplitude, and dynamic harmonic content. As the reader will be aware, pitch and dynamic amplitude characteristics are controlled by the VCO (Voltage Controlled Oscillator) and VCA + ADSR (Voltage Controlled Amplifier and Attack - Decay - Sustain - Release) modules in the Formant synthesiser, whilst the VCF (Voltage Controlled Filter) is used to vary the harmonic content of the signal.

However, in the case of mechanical tone generators, for example brass and woodwind instruments, an additional consideration is the existence of resonant areas in the instrument which possess free vibration periods of their own. These resonances, which are known as *formants* (whence the name for the Elektor music synthesiser!) are determined by the shape and mechanical construction of the particular instrument (the wooden back and belly of a violin, the pipes of an organ etc.). Unlike the variable pitch of, say, a violin string, they tend to reinforce the same harmonics, whatever the pitch of the note being played. The nature of the formants in an instrument is in fact one of the factors which govern its quality.

It should thus be apparent that, in order to realistically simulate the tonal characteristics of traditional instruments, one must be able to tailor the static harmonic content of the note accordingly. What is required is a number of resonant filters with independently variable centre frequency, gain and Q-factor. These features are present already in the state variable VCF of the Formant; however, that is only one filter, and more to the point, in this particular application there is no need for the filters to be voltage-controlled, since the filter parameters will be preset to suit whatever musical instrument is being imitated. This explains the reason for the separate manually-controlled resonance filter module described in this article.

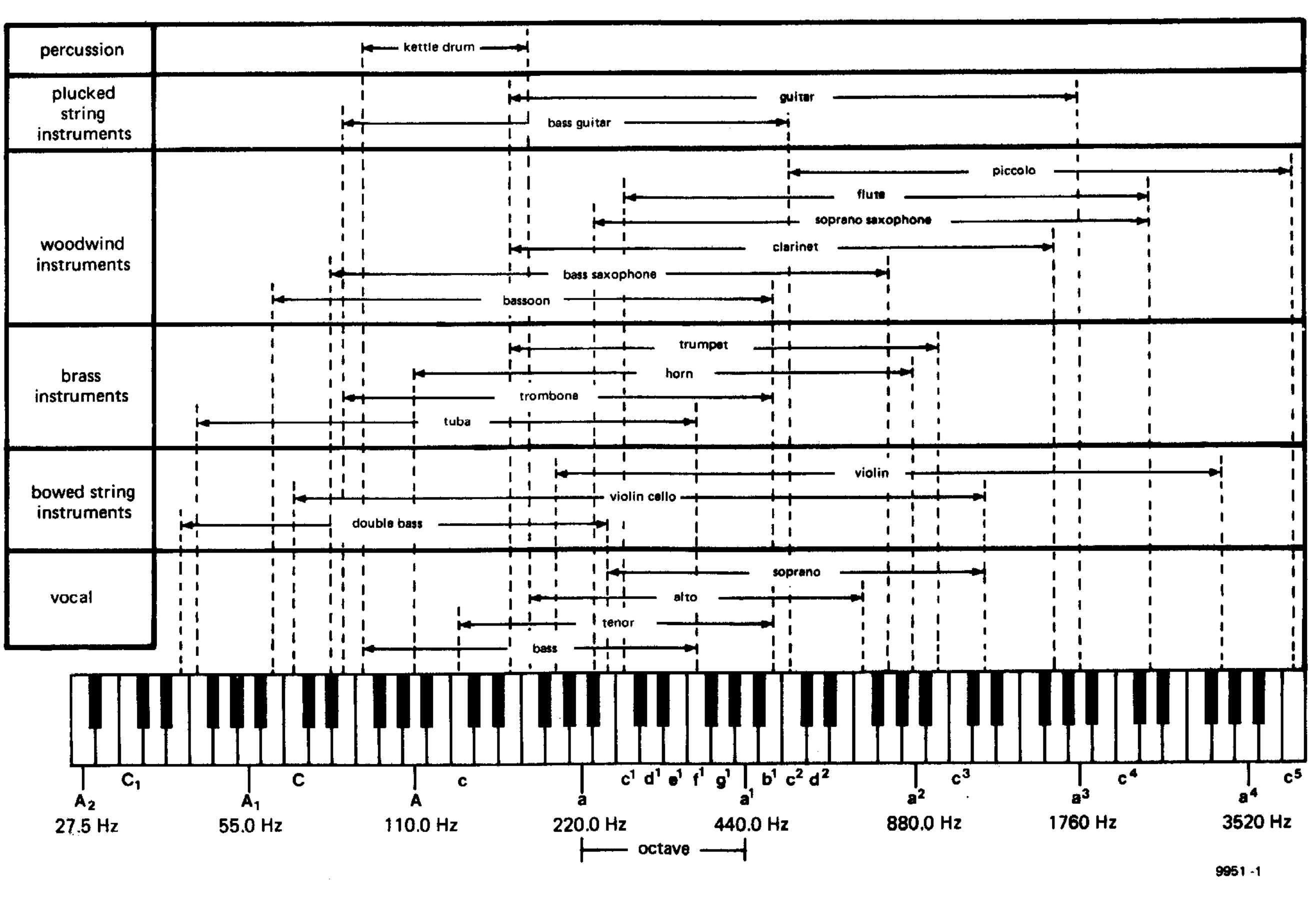
Figure 1. The fundamental frequency range of a number of traditional musical instruments, with reference to that of a grand piano. *(From: 'Elektronik Taschenbuch, Band 1', Ferd. Dümmlers Verlag, Bonn; with kind permission from the publishers.)*

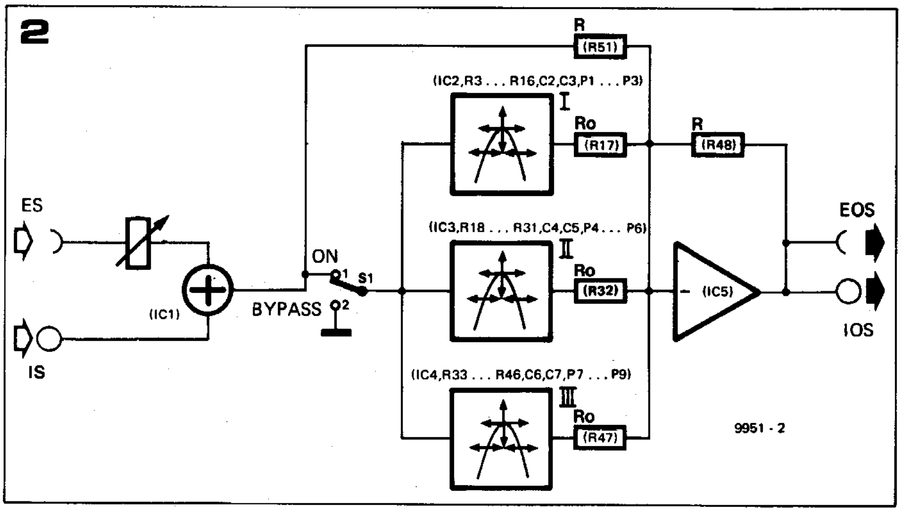
The uses of resonance filters

The effect of resonance filters can best be heard on 'bright' sharp VCO waveforms which have a high proportion of fairly intense upper harmonics. The effect on vocal sounds can be illustrated by taking a suitable signal with a frequency of around 200 Hz, setting the Q of the filter to a mid-value, and varying the centre frequency from minimum to maximum. At first 'dark' sounding tones, largely devoid of higher harmonics will be obtained; as the centre frequency is increased however, one by one the various vowel sounds can be distinguished until, at high centre frequencies, reedy flute-like sounds are produced. The higher the Q of the filter, the more pronounced the above effects – and vice versa.

All bandpass resonances of musical importance lie between roughly 100 and 2000 Hz. Table 1 lists the main fixed resonances of a number of common musical instruments and also indicates which VCO waveform is best suited to imitate the instrument in question. This table is, of course, merely intended as a rough guide, in the final instance the decision should rest with one's own ears. Unless otherwise indicated, the Q-control should be set to the mid-position. As a further aid, figure 1 shows the fundamental frequency ranges of vari-

1





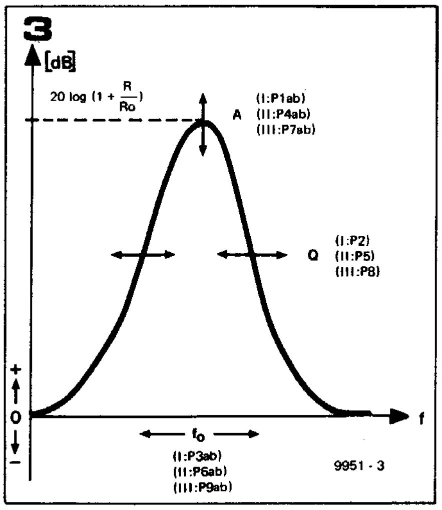


Table.

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

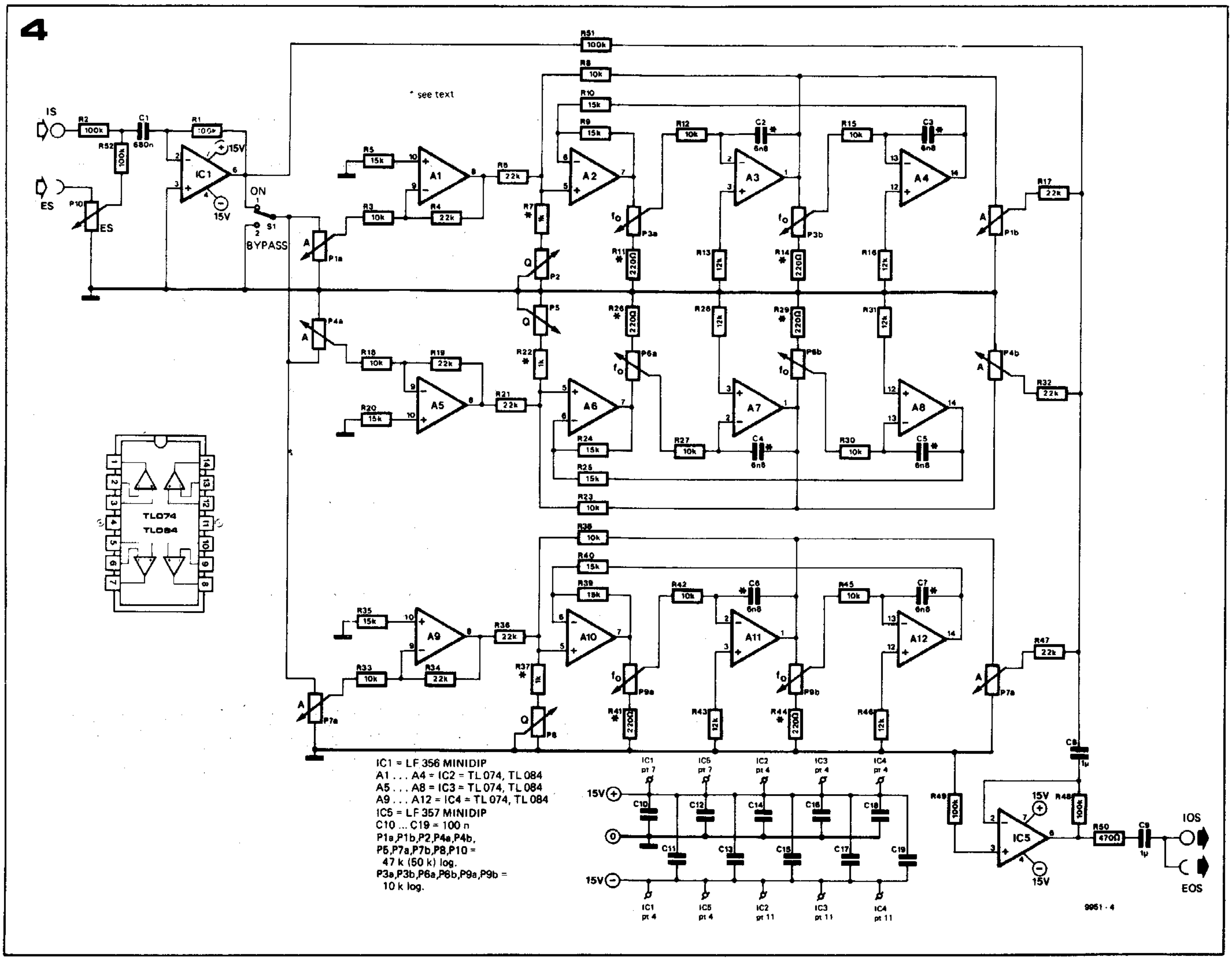
NB:
* with increased Q
** if possible, use several resonant filters (or a comb filter)

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

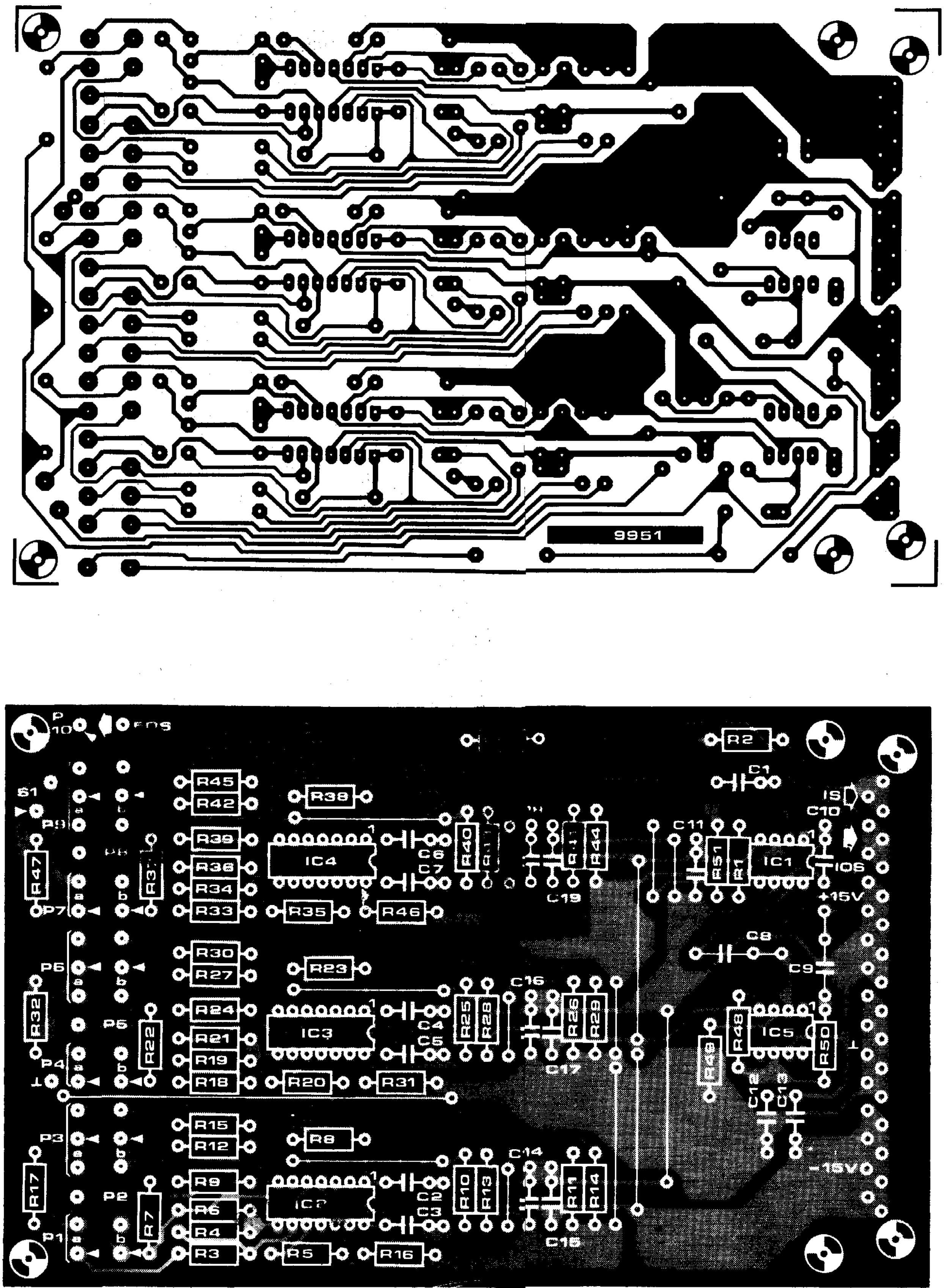
Figure 3. The frequency response of one of the three filter sections contained in the resonant filter module. The figure illustrates how the filter parameters can be independently varied by means of the control potentiometers.

Figure 4. Detailed circuit diagram of the filter module.

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



5


9951
R45
R42
R38
R39
R36
R34
R33
R35
R46
IC4
C6
C7
R40
C19
R44
R2
C1
C11
R51
R1
IC1
IC6
C10
+15V
C8
C9
R30
R27
R23
R24
R21
R19
R18
R20
R31
IC3
C4
C5
R25
R28
C16
R26
R29
C17
R48
IC5
R50
R49
C12
C13
R15
R12
R8
R9
R6
R4
R3
R5
R16
C2
C3
R10
R13
C14
R11
R14
C15
-15V
R47
R37
R32
R22
R17
R7
P1
P2
P3
P4
P5
P6
P7
P8
P9
S1

Parts list to figure 4 and 5.

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

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

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

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

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

ous traditional instruments, with reference to a piano keyboard.

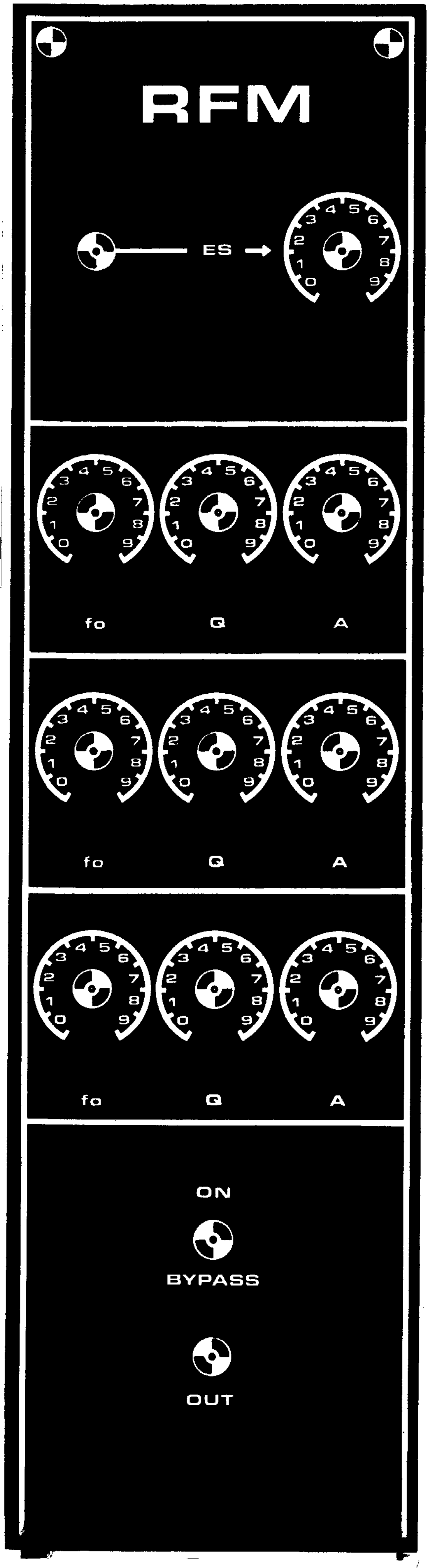
Circuit

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

Block diagram

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

6



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

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

7

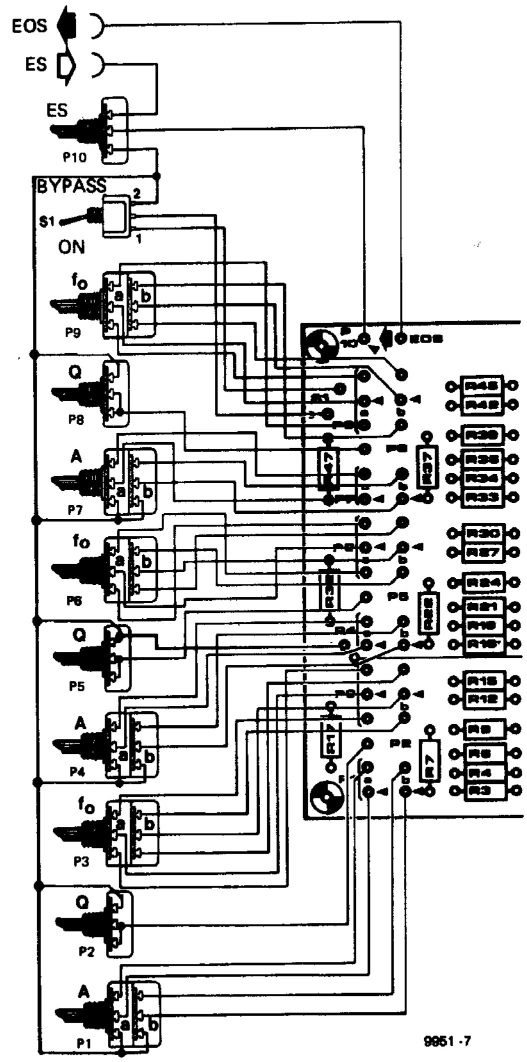


Figure 6. Because of the large number of controls, the front-panel for the resonant filter module is clearly different from the other Formant front-panels (EPS 9951-2).

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

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

Construction

The printed circuit board for the resonant filter module is shown in figure 5.

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

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

Appendix

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

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

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

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

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

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

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

chapter 9

ADSR

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

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

Types of envelope curves

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

1. Attack/decay contour

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

2. Attack-sustain-release contour

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