playing the formant

basic control settings and musical tips on how to play the formant

Because of its inherent versatility and scope, a synthesiser is not an easy instrument to master. Even relatively experienced musicians require a certain amount of time to sufficiently familiarise themselves with a synthesiser before being able to exploit its potentialities to the full. For the beginner, the impressive array of different controls on the front panel can be extremely confusing, and his first attempts to synthesise a particular sound can all too often lead to disappointment and frustration. For this reason the following chapters are intended to provide some basic tips on how to set about playing the Formant.
In addition to proving useful to Formant users, much of the advice is relevant to music synthesisers in general. In particular, the description of the resonance filter settings, which are of prime importance for the realistic simulation of conventional musical instruments, can in principle be applied to improve the sound of any electronic music instrument.
The review of basic control settings is divided into two parts: the first deals with the individual modules; how they are adjusted for satisfactory operation and how they can be used to best advantage.
The novice Formant user is thus spared the grief of coming to grips with the instrument on the basis solely of trial and error. The second part deals with combinations of basic settings used to synthesise the sounds of particular instruments. Thus, for example, the settings required to imitate the sound of a flute, trumpet, tuba, of string instruments, a piano, etc. are all described in detail and illustrated with the aid of diagrams. As already mentioned, particular attention is paid to the role of the resonance filter module in simulating the fixed bandpass resonances or formants of individual instruments, thereby improving the realism of the resultant sound.
With the aid of the following advice, the Formant user should be able to master the basic 'palette' of tone colours offered by the synthesiser. However it is only natural that the descriptions contained in this chapter bear the 'stamp' of the author; thus the reader should not feel constrained to limit his experiments with the Formant to those described here. They should rather be regarded as an initial stimulus to the reader, who is expected to go on to discover for himself the full musical potential of the Formant.

A final tip: one of the most fruitful activities the amateur synthesiser user can do is to listen to the results obtained by some of the acknowledged masters in this field, e.g. Walter Carlos, Isao Tomita, P. Moraz, etc. This way he will obtain some idea of the amazing capabilities of a sophisticated music synthesiser.

extending the formant 1

Before proceeding to a discussion of the basic control settings for each of the Formant modules, it is worth devoting a little attention to the equipment needed to reproduce the output of the synthesiser and the ancillary circuits which can be employed to further extend the Formant and enhance its tonal possibilities.

Loudspeakers

Readers building a synthesiser for home use will probably wish to play the instrument through an existing hi-fi set-up, at least to begin with. If this is the case, care should be taken not to overload the loudspeakers by keeping the volume reasonably low. Hi-fi speakers are designed to handle a much more broadly distributed power spectrum than that produced by a synthesiser, and it is quite easy to damage the tweeters with a sustained high frequency note. For serious use, a purpose-designed loudspeaker should be considered. Because of their high efficiency, horn systems are to be favoured, since they can then be used in conjunction with a relatively low power amplifier. A 35-40 W amplifier when used with horn speakers is more than ample for domestic purposes. A dealer specialising in electronic music systems should be able to offer advice on a suitable choice of loudspeaker.

Additional units

Although the Formant system so far described is a highly versatile instrument giving a performance comparable to commercial designs at a greatly reduced cost, it is nonetheless relatively unsophisticated compared to the larger commercial instruments. Fortunately, however, because of its modular conception, it is a simple matter to extend the system by adding extra VCOs, VCFs, VCAs, ADSRs, to obtain a more varied sound.
However, certain effects require the addition of completely new modules and ancillary units. One possibility which can be implemented immediately is the addition of the Elektor equaliser (January 1978) to allow presettable tailoring of the synthesiser spectrum. The equaliser p.c. board is of Eurocard format, compatible with the other Formant modules.
Phasing circuits are frequently used in synthesisers, and are particularly useful for realistic simulation of (orchestral) string tones. Another effects unit which is often used is a ring modulator. This circuit produces the sum and difference of two input frequencies at its output. The frequencies are often harmonically unrelated and strange, extremely 'unmusical' effects can be obtained. This type of circuit is extremely useful for synthesising bell, gong, and cymbal-like sounds. A circuit for a ring modulator, suitable for use in conjunction with the Formant, was published in the March 1979 issue of Elektor.
In its basic form the range of expression available from the synthesiser is slightly limited by the fact that it is played by a keyboard. However there are various ways in which this can be remedied. The addition of a 'pitchbender' joystick, which feeds a manually controllable DC voltage to the VCOs, allows modulation of the pitch of a note by hand in much the same way that a guitarist 'pulls' the strings of his guitar.
An interesting possibility is the elimination of the keyboard by playing the synthesiser via another instrument. This is accomplished by the use of a pitch to voltage converter, which produces an output voltage proportional to the pitch of the control instrument. This in turn controls the frequency of the synthesiser VCOs. An envelope follower produces an output voltage which follows the control instrument's amplitude, and this is used to control the gain of the VCAs. The result is a synthesiser which has the dynamics of the original instrument.
Other useful additions to the synthesiser are sequencers, sample-and-hold circuits, and in particular, reverberation echo units. Sequencers are used to store (either by analogue means or digitally) a sequence of VCO/VCF control voltages. These are then 'played back' into the synthesiser to automatically generate a note sequence which can, for example, be used to provide the backing for a manually played melody.
A sample-and-hold circuit is frequently employed to take sequential samples of the instantaneous voltage of a sawtooth waveform. This sequence of voltage samples is then used to control the synthesiser to generate pseudo-random sequences of notes.
Reverberation units can be used to great advantage to enhance the 'dry' somewhat artificial character of electronically synthesised sounds by allowing the notes to die away gradually, as opposed to being cut off abruptly when a different key is pressed. Long reverberation times or echo can transform the monophonic output of a VCO into a rich 'chorus' effect.
An especially useful and inexpensive item of equipment is a foot pedal, which can be used to control the tone-shaping modules of the Formant, the VCF and VCA. With the aid of the pedal, which should provide a variable DC voltage between 0 and 5 V, a variety of modulation effects can be obtained. For example, the duty-cycle of the VCO squarewave output can be modulated by pedal so as to obtain a type of phasing effect.

It is also important to have a sufficient quantity of different coloured patchcords of various lengths. These can easily be made using flexible single-core cable fitted with a 3.5 mm jack plug at each end. The cable is soldered to the centre contact (ball) of the plug, no earth connection being necessary as the earth return is made through the internal module wiring. In the interests of long life the patchcord wire should not be too thin, and some sort of strain relief should be used where the wire enters the plugs. To keep the front panel tidy, a good idea is to make the patchcords in different lengths, each designed for an interconnection between specific modules. Different colours of wire may also be used to simplify checking of complicated patch connections. Multiway patchcords, which are terminated at one end in several plugs, may also prove useful, allowing, e.g. the PWM inputs of several VCOs to be controlled by a single LFO output.
A high impedance pair of headphones will also prove extremely helpful, since they allow the output of individual modules to be monitored without having to reproduce them via the loudspeakers.
If an oscilloscope is available, then it is possible to follow the progress of a waveform from module to module, or to check the amplitude of a signal, etc. Such information is particularly interesting, since it permits one to actually see the envelope of signals, modulation effects, the existence of resonances, etc. All that is needed (in addition to the scope) to gain access to all the module outputs is a high impedance probe terminated in a jack plug.
Finally, having obtained a particular sound which one likes, how can the corresponding control settings be preserved for future reference? – a point which many Formant users have found to be a problem. One solution is the sketch of all the various Formant modules included at the end of this book. This can be photocopied and used for making a note of the control settings for each module.

control settings of individual formant modules 2

VCOs

Main characteristics of the VCOs

The Formant VCOs are electronic tone generators whose pitch can be varied over the range of audible frequencies. Assuming that they have been correctly set up during construction (V/octave characteristic and high frequency tracking), the VCOs should be accurately tuned and capable of tracking one another over a large number of octaves. Although the temperature drift of the VCOs should be so small as to be negli-

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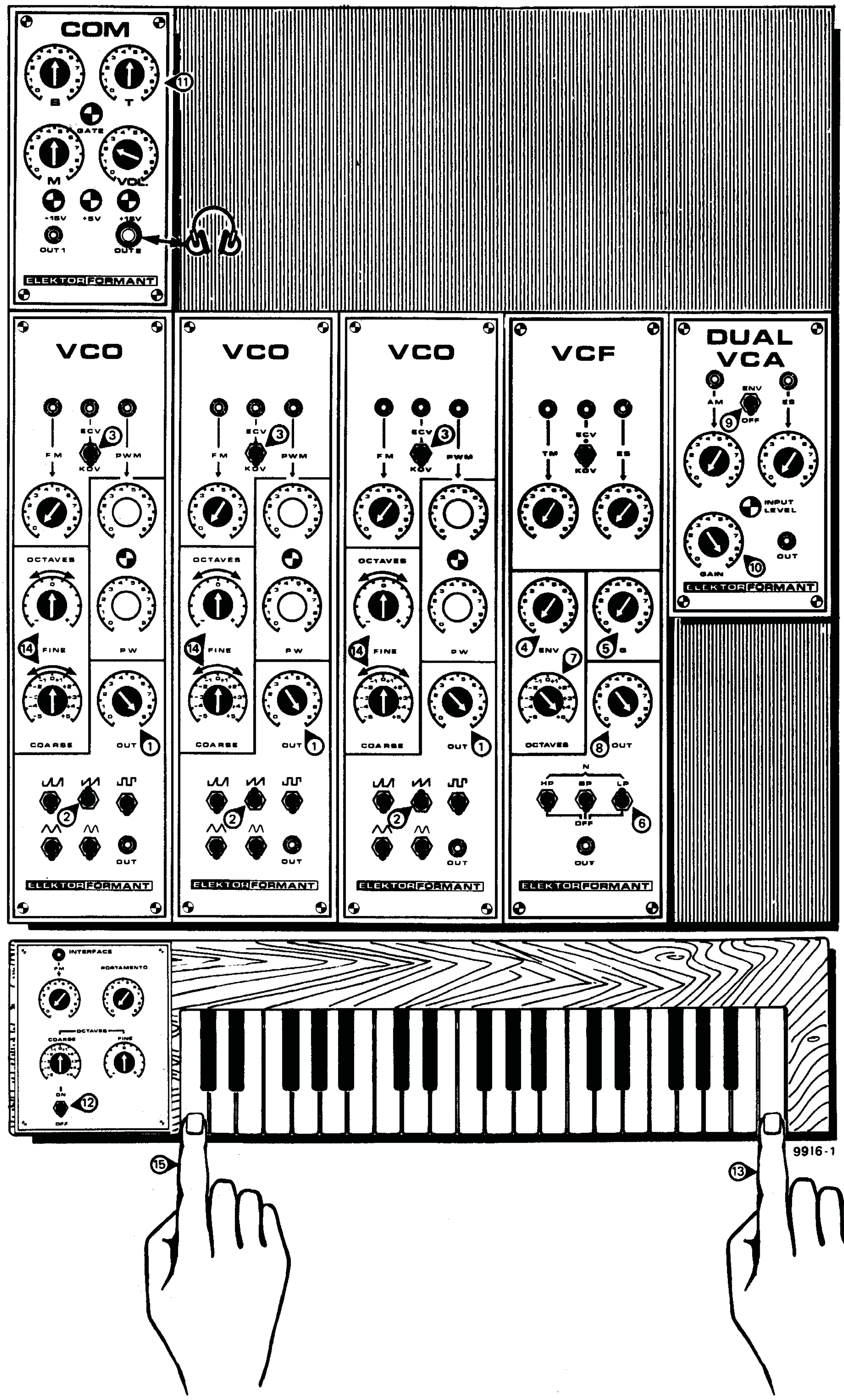
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Figure 1. Tuning the VCOs.

gible, the Formant should nonetheless be protected against sudden or excessive variations in temperature. Suitably positioned ventilation holes in the case of the instrument should help to reduce temperature fluctuations (and drift) to a minimum.
The approximate pitch of each VCO can be rapidly adjusted (over an extremely wide range) by means of the COARSE tuning control, whilst exact tuning is obtained with the aid of the FINE control.
With a choice of five different waveforms from each VCO (sinewave, sawtooth, spaced sawtooth, triangle, and pulse-width modulated squarewave) the range of basic tone colours for subsequent synthesis is extremely varied. The waveforms can be rapidly selected by means of toggle switches, and fed in any combination to the VCO output.
The output level control (OUT) determines the relative proportions of each VCO output fed to the VCFs. The capabilities of the VCOs are further enhanced by provision of an external control voltage input (ECV), a frequency modulation input (FM) for vibrato etc., and an input for pulse width modulation of the squarewave by external control voltages. The depth of pulse width modulation can also be varied by hand.

Tuning the VCOs

This section describes the tuning procedure for the VCOs which should be followed exactly. VCOs which are not exactly in tune sound 'rough' and dissonant, and are musically of little use. If in spite of carefully carrying out the procedure described here, the VCOs should continue to sound unsatisfactory, then the V/octave and high frequency tracking pots in each of the VCOs should be readjusted.
A 'bright' waveform, such as e.g. the sawtooth, which is rich in all harmonics, is best suited for tuning the VCOs, since beat notes are then particularly prominent. Before commencing the tuning procedure the other Formant modules must be set so that the output signals of the VCOs are fed in unchanged form to the power amplifier (figure 1).
First of all the output level controls are set to maximum (1) and the desired waveform, e.g. sawtooth, is selected (2). The VCO input is then switched to KOV – the keyboard output voltage (3). To ensure that the VCF(s) have no effect upon the VCO signals, the ENV and Q controls of the filter(s) are set to 0 (4) and (5), whilst the lowpass mode is selected (6) and the turnover frequency of the filter(s) is set to maximum (7). If the RFM is present it should be switched to 'bypass'. The output level control of the filter(s) is likewise set to maximum (8). The output of the VCOs is now passed unaltered by the VCF(s).
To ensure that the VCA likewise has no effect on the VCO signals, the toggle switch should be set to OFF (9) and the GAIN control set to maximum (10). The tonal character and volume of the VCO signal is set by means of the bass, midrange, treble and volume controls of the COM module (11). Finally, it is best to eliminate the effect of the COARSE offset control on the INTERFACE module by switching the latter out of circuit (12).
Before starting the tuning procedure proper, it is important to note that one should always start with the key on the extreme right (i.e. the 'highest' key) of the keyboard (13). In this way it is possible to ensure satisfactory tracking of the VCOs. Whilst the top key is pressed, the pitch of the VCOs is adjusted using the COARSE and FINE controls (14). If two or more VCOs are to be tuned to the same pitch, the FINE control should be adjusted until no further beat notes can be distinguished or else until the frequency of the beat notes is as low as possible. Of course it is also possible to deliberately mistune the VCOs very slightly so as to obtain phasing and chorus effects. The resultant sound is then less sterile or artificial.
Many users experience difficulties when attempting to tune three or more VCOs to obtain complex chords. This can be done as follows: First of all the lowest note in the chord is set up on one of the VCOs (the others are switched out). In the case of a simple major chord such as 'C'-'E'-'G', this would be 'C'. Once the 'C' key has been tuned, the key corresponding to 'E' should be pressed, and the pitch of the first VCO noted. The second VCO is then tuned by switching out the first VCO, pressing key 'C', and adjusting the COARSE control until the pitch of the signal is the same as that obtained from key 'E' with the first VCO. The third VCO is tuned for 'G' in exactly the same way.

The second VCO is switched out and the pitch of the key corresponding to 'G' on the first VCO is noted. Upon switching to VCO 3 key 'C' is tuned by memory to 'G' of the first VCO.
This method is initially somewhat time consuming, and may need to be repeated several times, until the VCOs are tuned sufficiently accurately. Finally, the VCOs can be 'fine-tuned' to obtain the desired degree of phasing or chorus effect.
A useful aid for tuning chords is an electronic tone generator based on a crystal-controlled oscillator. Programmable dividers are used to provide a series of reference frequencies. The circuit of a suitable tuner was published in Elektor 15/16 (July/August 1976 – see also Elektor 17, p 9-57). The reference signals can be mixed with the VCO signals via the external input of the VCF or VCA, whereupon the VCO is zero beat tuned.
Of course the user is not constrained to tune the Formant in the equally tempered tonic scale, but is free to choose whichever scale he wishes.
A few additional remarks on chords:
As already mentioned, tuning several VCOs to roughly the same pitch produces a phasing effect which lends the resultant sound a harmonious, 'shimmering' quality. If the higher frequencies are filtered out using the VCF in the lowpass mode, a deep 'rolling' bass sound is obtained, similar to that of a church organ or grand piano. Passing the high frequencies and playing on the higher notes produces violin-like, or 'metallic' sounds. Tuning the VCOs at intervals of an octave causes the individual notes to merge together into an extremely rich overall sound, which, due to the large number of different frequencies it contains, has a great deal of 'body'. Interesting effects can be obtained by tuning VCOs several octaves apart and using different proportions of VCO signals and different waveforms for each VCO. This extends the range of basic tone colours available for subsequent synthesis. In certain instances simple major chords can sound extremely impressive, such as e.g. simulating fanfares of brass instruments. On the other hand, major chords can tend to sound rather monotonous and 'sickly-sweet', since the monophonic keyboard does not permit their resolution through chord changes. This is a problem with all chords which have a fixed, well-defined character. Because of the lack of chord changes they can rapidly become monotonous. One solution is to use a second, independent voice in the form of e.g. a MINI-FORMANT.
Minor chords, however, are particularly attractive on the Formant, since (as is the case with a guitar or piano) extremely complex phasing patterns are developed. The sound often benefits musically from using the lowpass filter to slightly attenuate the higher harmonics of the signal.

Pulse width modulation (PWM)

With the Formant VCOs the width of the squarewave output can be varied both by hand (using the PW control) and by means of an external modulation signal. In the latter case the modulation depth can be set by means of the control knob situated beneath the PWM input socket.

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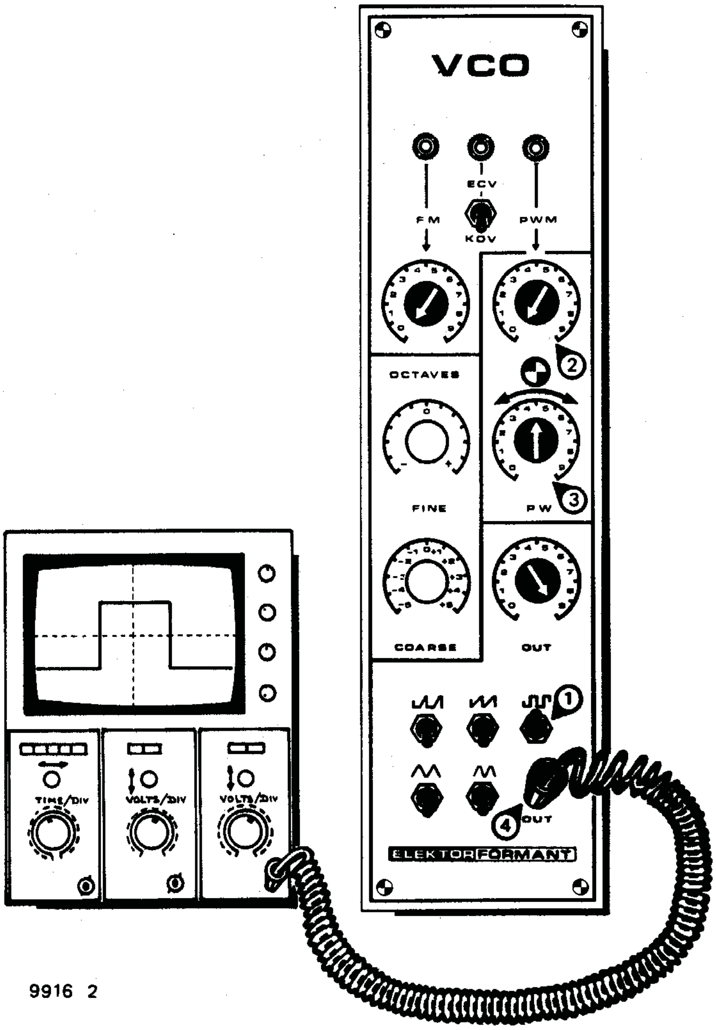

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To hear the effect of PWM, use only one VCO switched, naturally enough, to the squarewave output (see figure 2), and start with the modulation depth control (2) set to zero. Gradually increase the pulse width, from zero, by turning the PW control clockwise (3). The sound of the squarewave signal will change from a bright and rather 'thin' character to a point where it is distinctly hollow and clarinetlike. In this position the modulation depth is 50%, i.e. the squarewave has a 50% duty cycle. If one continues to turn the PW control to the right, the sound again becomes increasingly brighter and thinner. The change in the squarewave can be seen quite clearly if one connects the VCO output to a scope (4). Filtered squarewaves with a roughly symmetrical mark space ratio are useful for the simulation of a number of woodwind instruments and for piano-like sounds. Triangle waveforms are best suited as low frequency PWM signals. The resulting effect is similar to the phasing obtained when two or more VCOs are tuned to the same pitch.
Before looking at the use of the LFO module for pulse width modulation, it is worth first mentioning a couple of other interesting possibilities for external control signals.
If the output of an ADSR envelope generator is connected to the PWM input, the result is *dynamic pulse width modulation,* which varies during the course of each note. Depending upon the character of the envelope contour, extremely unusual, seemingly distorted sounds can be obtained, as well as sounds which gradually decay with a slow and pleasant phasing effect. The amplitude of the ADSR modulation signal is set by means of the modulation depth control directly underneath the PWM input socket (2), whilst the duty-cycle offset of the squarewave, i.e. at what mark-space ratio the pulse width modulation begins, is set by means of the PW control (3).
Another possible way of modulating the pulse width of the squarewave is to use the control pedal, which depending upon its position provides an output voltage of between 0 and 5 V. The latter is fed, like the ADSR output, to the PWM input socket. Once again, the initial mark-space ratio and the amplitude of the modulation signal are set by the PW and pulse modulation depth controls respectively. With the aid of multiway patchcords the pedal can be used to modulate several VCOs simultaneously.
Now to using the PWM input in conjunction with an LFO (figure 3). In order to obtain phasing over the entire range of modulation settings, the VCO should be set up as follows. The triangle output of the LFO is connected to the PWM input of the VCO (1). The modulation depth control is set to zero (2), and the initial mark-space ratio of the squarewave is set to 50% (3). At this stage the squarewave should sound hollow and clarinet-like. It may be necessary to adjust the initial pulse width slightly to obtain the desired effect. The LFO frequency should be fairly high, say approximately 1 Hz.
If the modulation depth control (2) is now turned fully clockwise to its right hand stop, on positive and negative peaks of the modulation signal the mark-space ratio of the squarewave will be 0 and 100%, i.e. the signal will be punctuated with regular gaps. To eliminate this effect the modulation depth should be reduced (2) until there are no longer any audible glitches in the output signal. Effective pulse width modulation can now be obtained for any intensity of modulation signal, and the user is free to vary the frequency of the LFO signal as desired. Best results are obtained if fairly slow modulation rates are used, since the frequency of beat notes is then low – of the same order as those produced when tuning several VCOs to approximately the same pitch.
In the case of LFO modulation, it is recommended using a separate LFO output for each VCO. By slightly varying the modulation frequency of each LFO it is possible to produce extremely complex phasing patterns when the squarewave signals are mixed together. The effect is that of multiple sound sources, i.e. as if there were more than three VCOs present. It is by independently modulating several VCOs in this way that one is able to obtain some of the most impressive orchestral effects (string ensemble etc.) from the Formant.

Ring modulation using PWM

In the March 1979 issue of Elektor a circuit for a ring modulator, which could be used in conjunction with Formant, was published. However it is also possible to obtain sounds containing a large number of intermodulation products simply by using PWM between two or more VCOs. The squarewave output of one VCO is fed to the PWM input of another (see figure 4). The OUT control of VCO 2 is turned right down (2), whilst the two PW controls on VCO 1 are set as for LFO modulation signals. The settings for these controls are not particularly critical in this case. The OUT control of VCO 1 is turned up full (3).
Depending upon the frequency settings of the two VCOs, typical ring modulator effects (reminiscent e.g. of the noise from short wave radios) are obtained at the output of VCO 1. Certain settings produce the basic timbre of gong and bell-like sounds, which are characterised by unrelated harmonic components.

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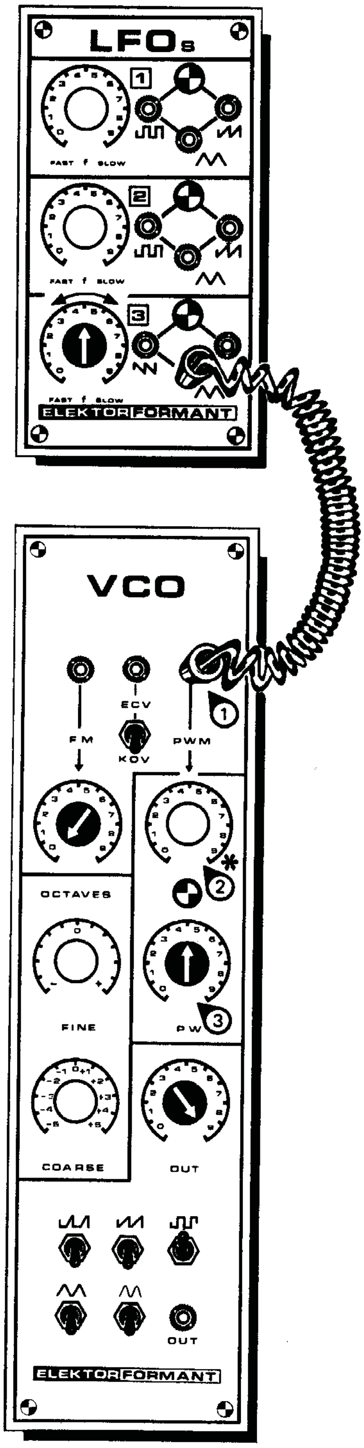

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* see text

Figure 2. Manual adjustment of pulse width.

Figure 3. Pulse width modulation using an LFO.

Figure 4. Ring modulator effects using PWM.

VCO Waveforms

A detailed description of the role of different waveforms in synthesising the tone colour of particular instruments is contained in chapter 3. At this stage suffice it to note that, thanks to the internal patching between VCOs, it is possible to switch between the different waveforms and mix the outputs together extremely rapidly and in a wide variety of different proportions. It is worth while experimenting with as many different combinations of waveforms as possible. With a little practice it is not difficult to familiarise oneself with the musical character of individual VCO waveforms. Sawtooth and spaced sawtooth, which are rich in both odd and even harmonics, have a 'bright' character, which renders them suitable for the imitation of brass and string instruments. In the case of the spaced sawtooth, the upper harmonics are particularly pronounced, so that they resemble the higher-pitched brass instruments such as the cornet. Filtering out the higher frequencies of a sawtooth around the middle octaves produces a more mellow, flute-like sound, whilst filtering of the fundamental gives a timbre similar to that of the oboe.
Symmetrical waveforms, such as the squarewave and triangle, contain only odd harmonics, and are characterised by a 'hollow' sound. The basic squarewave resembles a bright, strongly played clarinet. Varying the pulse-width (duty-cycle) of the squarewave so that the waveform is no longer symmetrical has the effect of introducing the even-numbered harmonics and producing a more 'reedy' timbre. Rectangular waveforms are also employed for piano sounds.
The triangle, which is also a symmetrical waveform, has a smooth mellow character, similar to that of a flute. This is due to the fact that the higher harmonics are of a very low amplitude, much lower than those of the squarewave. The sinewave, which in principle has no harmonic content apart from the fundamental, is even smoother still. However the relatively high harmonic distortion of the sinewave ensures that it does not sound completely expressionless and bland.
By judicious mixing of different combinations of waveforms and selective (resonance) filtering, it is possible to produce an enormous variety of different sounds, each with a unique timbre.

Frequency modulation

Each of the VCO front panels is provided with an 'FM' input socket for frequency

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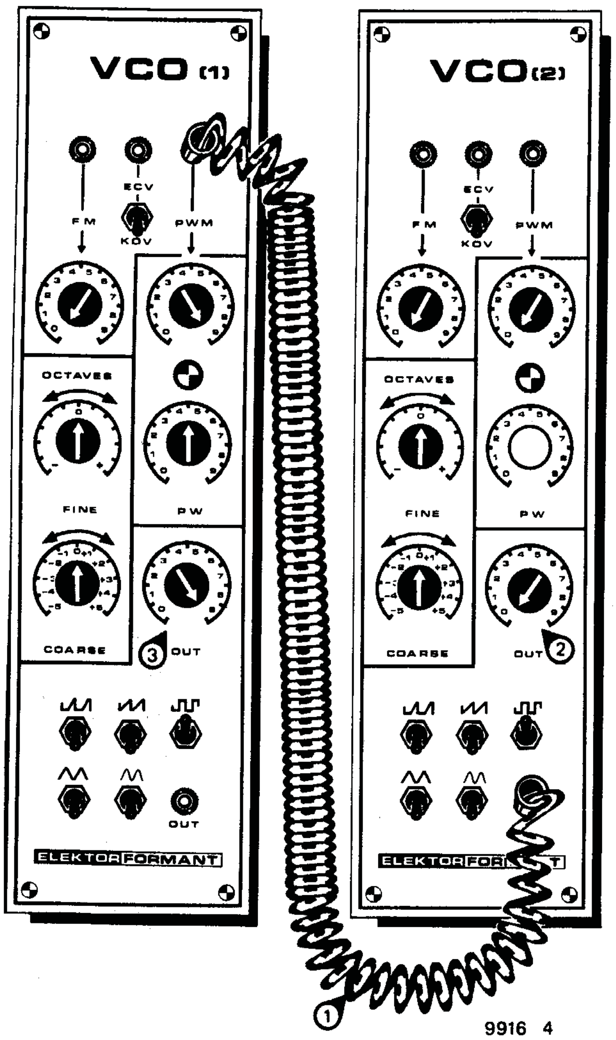

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modulation of the VCO by an external signal. The control situated directly beneath the FM socket determines the modulation depth, i.e. the maximum frequency deviation. As is the case with PWM, various different modulation signals can be used, the most obvious being the LFO outputs.

Vibrato

This is the simplest and most common form of frequency modulation. The triangle output of an LFO is fed to the FM input of the VCO, and the modulation depth is gradually increased until low level modulation occurs. To obtain a pleasing vibrato effect, the frequency of the modulation signal will vary (between roughly 1 and 10 Hz) depending upon the type of instrument being simulated. Normally it will be in the region of 5 Hz.

Musically, vibrato lends the resultant sound a 'singing' character, which is particularly effective when simulating stringed instruments. If vibrato is applied to several VCOs using separate modulation signals, rich choral and orchestral effects can be obtained. The variety of vibrato effects can be rendererd even more expressive by gradually varying the modulation depth as one is playing.

'Deep' FM

If the modulation depth is increased (still with a triangular modulation signal), siren-like effects, sweeping up and down the entire range of audio frequencies, are obtained. Depending upon the waveform and frequency of the LFO signal, as well as the settings of the VCO controls, an amazing variety of different sounds and effects – often highly artificial and electronic in character – can be produced. It is also worthwhile experimenting with squarewave modulation signals to several VCOs. It is sometimes possible to generate rhythmic tonal structures which can be played from the keyboard.

Cross modulation

Purely 'electronic', 'shortwave static' type noises can be obtained by cross-coupling the outputs and FM inputs of two VCOs. By experimenting with different settings of the FM, OUT and COARSE controls of both VCOs, a variety of 'atmospheric' effects can be obtained.

ECV/KOV

With the ECV/KOV switch in the KOV position, the pitch of the VCO signal is determined by the output voltage of the keyboard interface receiver, i.e. by whichever key is depressed. If this switch is set to the ECV position, however, the keyboard output voltage ceases to have any effect upon the VCO signal. The frequency of the VCO will instead be determined by the voltage fed to the external control voltage socket above the ECV/KOV switch. The type of control voltage which might be fed to this socket is e.g. the output of a *sequencer*, or of a second keyboard (with its own interface).

If the switch is set to ECV, but no control voltage is fed to the socket, the VCO can be used as an audio oscillator, with which the volts per octave characteristic and high frequency tracking of the other VCOs can be checked. The adjustment procedure is identical to that described at the end of Chapter 5 in Part 1.

Interface

The COARSE and FINE tuning controls on the keyboard interface provide a variable DC offset voltage which allows the compass of the synthesiser to be extended beyond that of the keyboard, and provides the possibility of chord transposition. The COARSE control can be switched in and out by means of the ON/OFF switch, so that a certain transposition can be 'programmed' on the COARSE control and switched in when desired. With the aid of the COARSE and FINE controls the synthesiser can be easily tuned to the register of any desired musical instrument.

The PORTAMENTO control varies the difference in pitch between successive keys, permitting *glissando* effects, i.e. a continuous glide from one tone to another. Subtle variations of glissando can be obtained by varying the portamento with the left hand whilst playing the keyboard with the right hand. The remarks concerning the use of FM inputs and controls on the VCOs also apply to the interface circuit, bearing in mind that a signal fed to the FM input of the latter will affect all the VCOs and the VCF tracking filters, so that a form of 'master vibrato' is obtained.

VCFs and RFM

Main characteristics of the VCFs

In addition to the resonance filter module (RFM), the complete Formant system contains two separate voltage controlled filters (12 dB per octave and 24 dB per octave). The combined effect of the three filters gives the Formant great versatility, and makes it possible to obtain a wider and more detailed range of tone colours than is the case with most synthesisers of comparable size. The two VCFs are most commonly employed as tracking filters controlled by an envelope shaper (i.e. the ADSR), so as to provide dynamic variation in the harmonic content of a note. Due to the less severe filtering of higher harmonics, the 12 dB VCF is suited to

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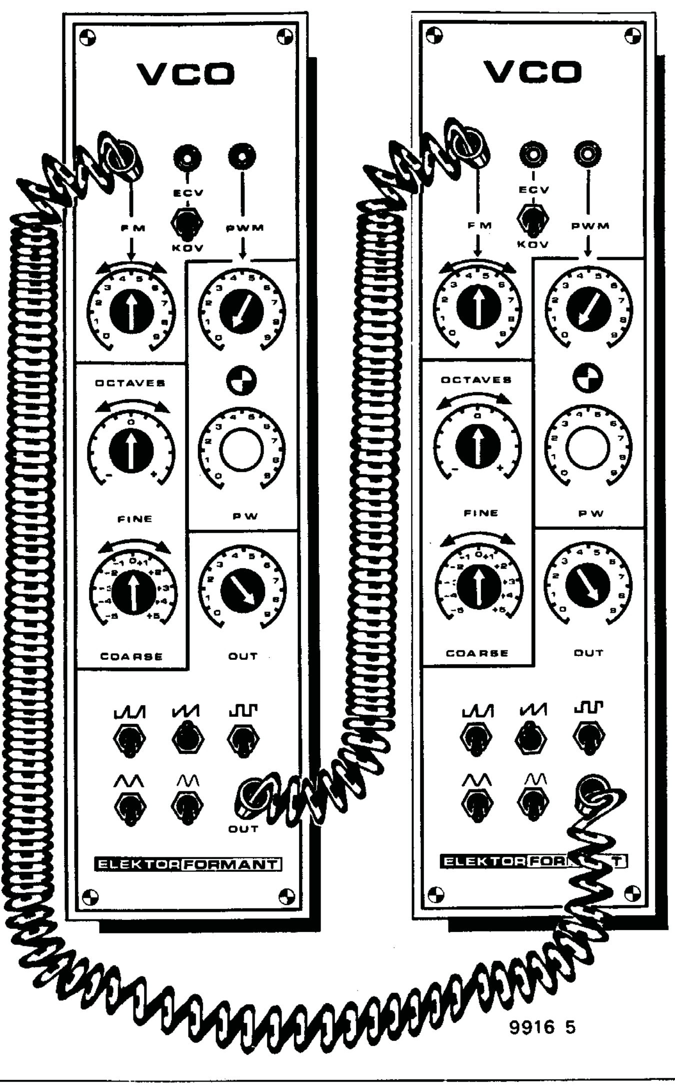

producing bright, 'transparent' sounds, whilst the steeper roll-off of the 24 dB VCF gives a fuller, more 'symphonic' sound which often has a more natural, less 'electronic' harmonic structure.
Both VCFs can also be used as static filters, simply modifying the tonal character of a VCO waveform to obtain the timbre of a particular instrument. For example either VCF can be used in the lowpass mode to filter out the upper harmonics of a squarewave and so simulate a flute-like tone.
Used in conjunction, precise tailoring of tonal characteristics is possible. Some typical effects which can be obtained using the two VCFs were described in the introduction to the 24 dB VCF module (see chapter 7, Part 1).
Since it can operate in the bandpass mode, the 12 dB VCF can also be used to complement the resonance filter, when simulating the *formant bands* of particular instruments. Indeed, in the absence of a resonance filter module, the 12 dB VCF can provide the resonance filtering necessary for realistic imitation of the voicing of conventional musical instruments. The turnover frequency of both filters can be varied over virtually the entire audio range with the aid of the OCTAVES control. The lowest turnover frequency is something below 15 Hz, which is sufficient to effectively block the VCO output when in the lowpass mode.
Both VCFs are normally patched internally to an ADSR. The amplitude of the envelope control voltage, i.e. the extent to which the cut-off frequency of the filter is shifted up and down, can be varied by means of the ENV control. Both filters also have a Q control, with the aid of which the selectivity of the filter can be adjusted. The turnover frequency of the VCFs can also be modulated by means of a low frequency signal fed to the TM (tone colour modulation) input socket, the modulation depth being set by the TM control. Finally, the external input socket (ES) and control allow the centre frequency/turnover point of the filter to be varied by a foot pedal, or some other external control voltage.
In the case of the 12 dB VCF, three toggle switches are used to select between the four available filter modes, 12 dB highpass, 12 dB lowpass, 6 dB bandpass and notch (the latter is provided by a combination of the highpass and lowpass functions). Unusual filter responses (e.g. elliptical response curves) can be obtained by experimenting with various combinations of switch positions and Q settings. In the case of the 24 dB VCF, toggle switches are used to select between highpass and lowpass modes and between different filter slopes (6, 12, 18 or 24 dB per octave).

Using the VCFs as tracking filters

As explained, the VCFs can be used as tracking filters, to provide dynamic variations in harmonic content during a particular note. The starting point are the basic control settings described in the tuning procedure for the VCOs, where the VCF initially has no effect upon the VCO output signal. The ECV/KOV switch is set to KOV (1), the filter switched to the lowpass mode (2), and the TM (3), ES (4) and Q (6) controls turned fully anticlockwise (see

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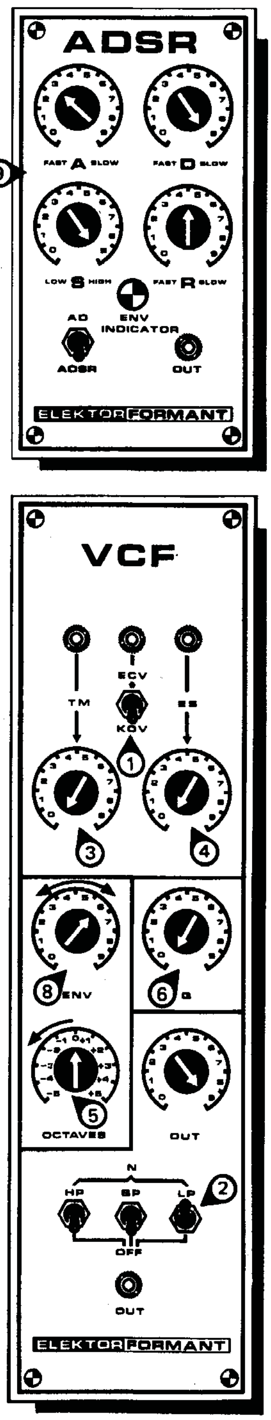

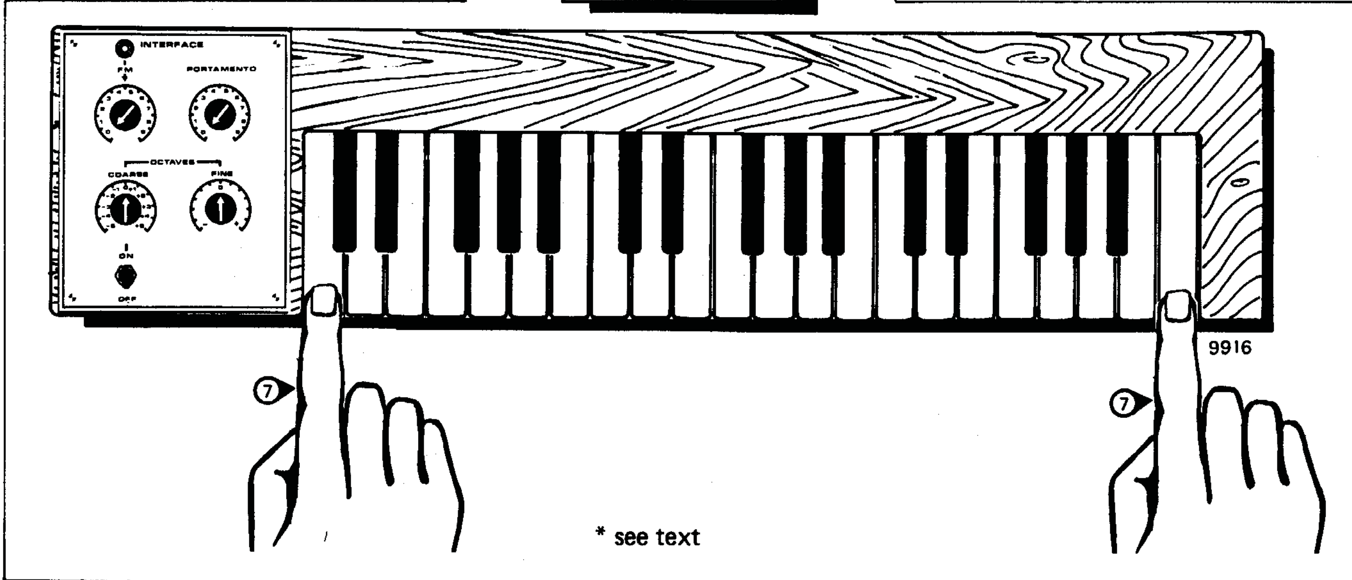

Figure 5. Cross modulation.

Figure 6. Setting up the VCF for use as a tracking filter.

figure 6). By slowly turning the OCTAVES control to the left (5), the turnover frequency of the filter is gradually lowered, and the tone colour of the VCOs should grow increasingly 'darker', as more and more of the higher harmonics in the signal are filtered out. Eventually, a point will be reached where the VCO signal is no longer audible. By pressing the top and bottom keys on the keyboard (7) one should check that the filter effectively blocks the VCO output signal regardless of the pitch of the note. It may prove necessary to readjust the OCTAVES control slightly at this stage. If a satisfactory result is still not obtained, the volts/octave characteristic of the filter needs to be readjusted. When the above procedure is completed, the VCF will 'open' (i.e. its turnover frequency will increase to pass more of the VCO signal) and 'close' (the turnover frequency falls to filter out the VCO signal) when a key is struck and released at a rate determined by the envelope contour of the ADSR.
Whilst experimenting with different keys, the ENV control on the VCF (8) is now gradually turned up (clockwise). The changes in harmonic content as the envelope control voltage is increased should be audible as different tone colours.
The timbre or tone colour of each note should grow brighter as the amplitude of the envelope voltage is increased and more of the harmonics are passed. With the ENV control turned fully clockwise a 'brilliant', strident tone is obtained, which may sound discordant. In the case of the 24 dB VCF, the level of the envelope control voltage is quite critical, since this filter is capable of very fine discriminations in harmonic content and tone colour. With percussive AD contours (rapid attack, slow decay) it is particularly important that the ENV control is not set too high, otherwise the VCF 'overshoots', and the peak of the AD curve is not reproduced in the dynamic harmonic content of the note.
The next step is to experiment with different ADSR envelope contours. With, for example, a sawtooth VCO waveform, it is possible to vary the sound from that of brass to string instruments; with additional waveforms the typical tonal character of clarinet, flute and other woodwind instruments can be simulated, whilst if the Q of the filter is increased, Wa-Wa effects and unusual speech like sounds are produced. For natural sounding tone colours the Q control should be turned almost fully anticlockwise.

When used as a tracking filter the VCF is almost always best switched to the lowpass mode. However interesting effects can be obtained by selecting the highpass response and 'inverting' the envelope contour, whilst phaser like effects can be produced by selecting the notch response in the case of the 12 dB VCF.

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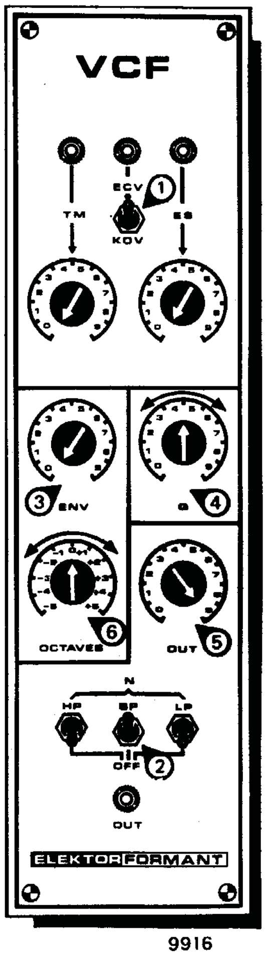

Figure 7. Setting up the VCF for use as a resonance filter.

Figure 8. The 12 dB VCF used to provide pedal controlled Wa-Wa.

Using a VCF as resonance filter

In the absence of a resonance filter module, it is possible to use the 12 dB VCF to tailor the static harmonic content of a note so as to reproduce the *formants* (fixed bandpass resonances) of mechanical tone generators such as brass, woodwind and string instruments. As has already been explained, this allows the voicing or timbre of these conventional instruments to be simulated with a much greater degree of realism.
When used as a resonance filter (see figure 7) the ECV/KOV switch should be set to ECV (1), and the bandpass function selected (2). The ENV control is turned right down to zero (3), whilst the Q control is set to the mid-position (4). As was the case when tuning the VCOs and experimenting with the VCF as a tracking filter, a 'bright' waveform which is rich in harmonics (e.g. the sawtooth) should be selected. The OUT control should be turned up full (5), so that the resulting sounds can be heard clearly. The frequency of the VCO signal should be set at roughly 200 Hz. If the OCTAVES control of the VCF is now turned gradually from left to right, the effect of variations in the centre frequency of the bandpass response should be audible. The results should be identical to those obtained with the resonance filter module (see chapter 8 Part 1). Initially 'dark' sounding tones are produced; then as the centre frequency of the filter is increased, one by one the various vowel sounds can be heard ('u', 'o', 'e', 'i' and 'a' should each be distinguishable), until at high centre frequencies, sharp, reedy sounds are produced. These vowel-like sounds are obtained over the frequency range of roughly 100 to 2000 Hz, and the majority of bandpass resonances of musical importance lie within this region. A table listing the main fixed resonances of the most common conventional musical instruments along with a rough guide to the most appropriate type of VCO waveform, was given in Part 1, in the chapter on the RFM.
The greater the Q of the filter, the more pronounced are the variations in tone colour, since the formant bands which are being enhanced are then even narrower. The lower the Q, the less noticeable the effect of resonance filtering.
How resonance filtering can be employed to simulate the timbre of specific instruments is described in greater detail in chapter 3.

Other filter settings

Modulated bandpass filter

The above settings for the resonance filter mode are left untouched, whilst a very low frequency triangle waveform is fed from one of the LFOs to the TM input. The TM control is set to roughly the mid-position. The OCTAVES and TM controls should together be adjusted until a suitable modulation depth (i.e. range of frequencies over which the centre frequency of the filter is modulated) is obtained. The intensity of the resonance filtering is then set by means of the Q control. The continual variations in tone colour which result, produce a sound which is extremely pleasant, and somewhat reminiscent of phasing.

Pedal controlled Wa-Wa

With the VCF in the basic resonance filter mode it can be used to provide pedal controlled Wa-Wa. The pedal, which can easily be constructed, should provide a 0 to 5 V DC supply, which is

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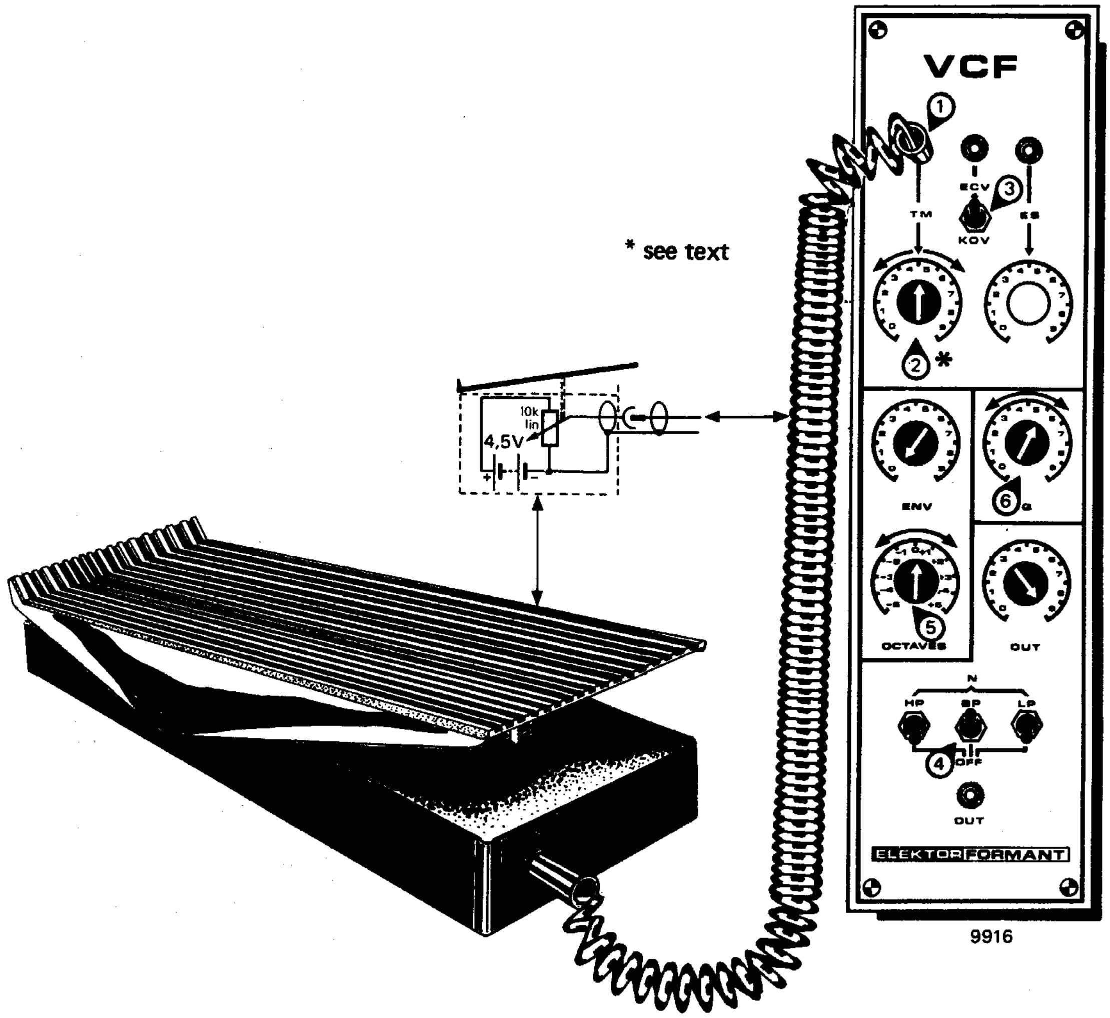


fed via two-core cable and a jack plug to the TM input of the VCF. A pedal allows much faster and more flexible control of the VCF; the ADSR envelope contours are in general too 'fixed' and slow to set up. (In this respect it is also worth while experimenting with pedal control of the VCFs when used as tracking filters, i.e. the ADSR contour is replaced by the output of the pedal)

The basic settings of the VCF for pedal controlled Wa-Wa are shown in figure 8. The output of the pedal is connected to the TM input socket (1), but initially the TM control is turned down to zero (2). As before, the external control voltage input (ECV) is selected (3), and the filter switched to a bandpass response (4).

Various notes from the desired range of frequencies are then played on the keyboard, whilst the OCTAVES control is adjusted until the resulting tones are sufficiently 'dark' (i.e. the 'W' of the Wa-Wa is obtained). With the pedal fully depressed, the TM control is then adjusted to determine how far the Wa-Wa sound 'opens'. The procedure should be repeated for several different notes, and generally a setting is chosen at which – with the pedal hard down and a medium to high Q factor – a distinct vowel-like sound, similar to 'a' (as in 'man') is obtained. Different types of Wa-Wa effect can be produced by varying the Q factor of the filter whilst playing a note, and by switching to a lowpass filter response. By carefully adjusting the relevant parameters (OCTAVES, Q and TM controls, low-pass or bandpass filter responses) it is possible to accurately simulate the tonal characteristics of most of the commercially available types of Wa-Wa circuits.

It is also possible to use the VCF to provide Wa-Wa for an electric guitar. The amplified guitar signal (which should have as rich a harmonic structure as possible) is fed to the external signal (ES) input socket, and the ES control turned up full.

RFM

The resonance filter module contains three separate resonance filters, whose parameters (gain, centre frequency and Q) can all be independently varied. Since resonance filtering involves certain *predetermined* frequency bands being enhanced, regardless of the pitch of the note being played, there is no need for the filters to be voltage controlled. The RFM thus frees the VCF(s) for use as a tracking filter, whilst its ability to pick out more than one resonant frequency band is extremely useful in such applications as the simulation of string sounds (see chapter 3).

The remarks made in the previous section (on the use of a VCF as resonance filter) regarding the choice of centre frequency and Q of the filter are obviously also applicable in the case of the RFM itself.

The front panel controls for the RFM are quite straightforward. The centre frequency of the filters can be varied over a nominal range of from 50 to 2300 Hz. Normally the filter parameters will remain fixed, once a particular setting has been chosen. However by rapidly varying the centre frequency and Q controls, phasing type effects can be obtained.

The RFM is provided with a bypass switch which allows the module to be switched quickly and simply out of circuit when desired.

VCA

Main characteristics of the VCA

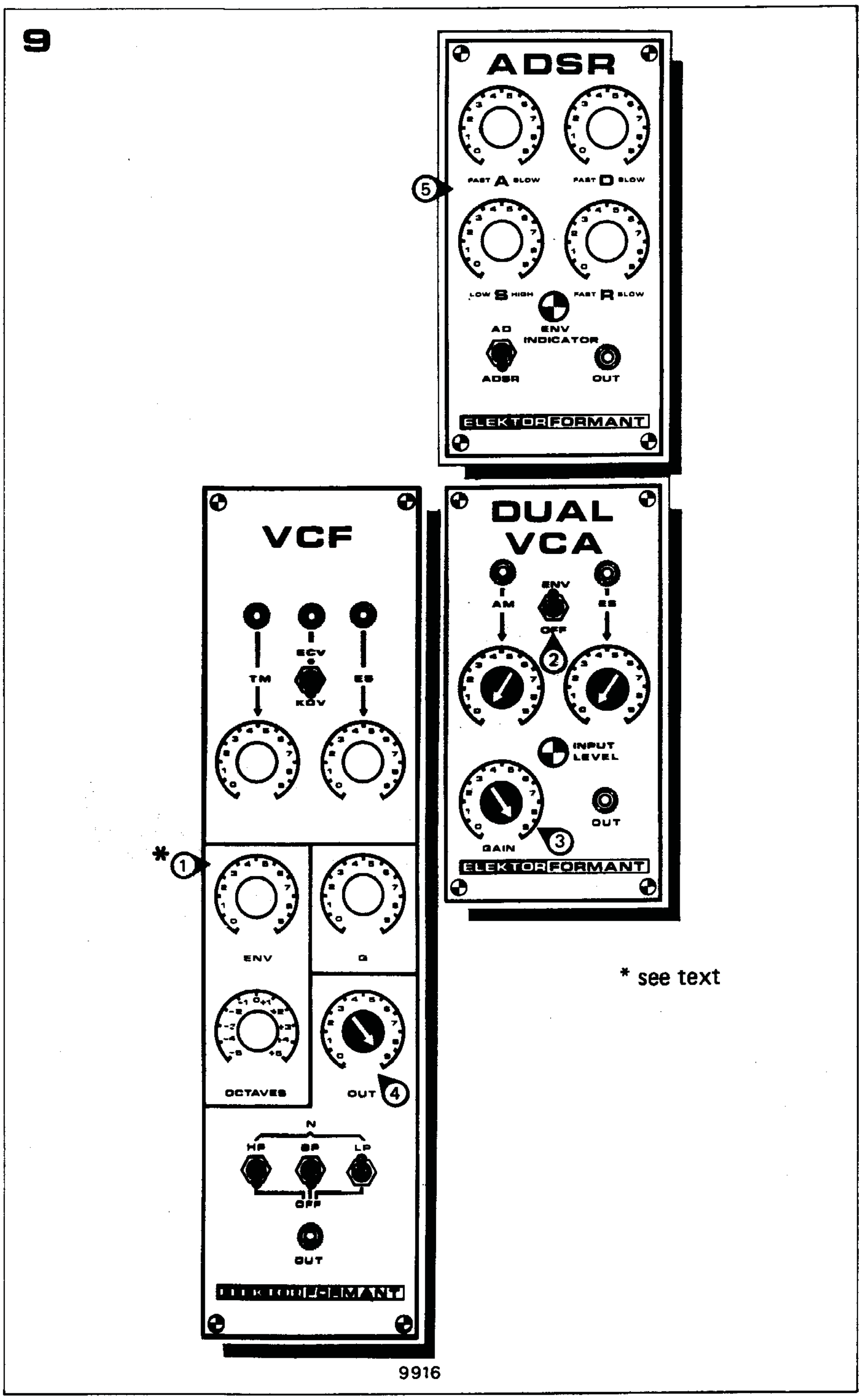
The two independently variable gain stages of the VCA provide exponential envelope (ADSR) contours for realistic simulation of the dynamic amplitude characteristics of different instruments, and periodic (linear) amplitude modulation (tremolo) of the signal waveform. The VCA is internally wired to the output of one of the ADSR envelope shapers. ADSR control is selected by means of a toggle switch; the ability to switch out the envelope control voltage is useful when tuning the synthesiser, since it allows signals to pass through the VCA without being affected by the envelope shaper. The VCA also has a modulation indicator, which allows the input signal level to be adjusted for the optimum compromise between low distortion and good signal-to-noise ratio. The linear amplitude modulation can be continuously varied between 0 and 100% modulation depth. The AM input also allows the VCA to be controlled via a pedal.

Tone colour and amplitude dynamics

Both the VCF and the VCA can be controlled by an ADSR envelope shaper, so that dynamic variations in harmonic content and amplitude can be obtained during the playing of a note. The majority of 'natural' sounds produced by mechanical tone generators have both a characteristic dynamic harmonic structure and amplitude envelope. However when taking one's first steps with the Formant, it is recommended that only one of these parameters be simulated. One should experiment in order to discover which factor had the greatest influence on the resulting sound, and depending upon whether it be the tone colour or amplitude dynamics, use the VCF (as tracking filter) or VCA accordingly. Several practical suggestions in this respect are given in chapter 3. If envelope shaping of signal amplitude only is required (fixed tone colour), the VCF is set to pass the VCO signal unaltered, or it can used for additional resonance filtering.

The basic control settings of the Dual VCA are quite straightforward (see figure 9).

The preceding VCF is switched to 'allpass' or to a bandpass response for resonance filtering (1), whilst the ENV/OFF switch is used to select the ADSR control input (2). The gain of the VCA is set to maximum (3), and the output level control (4) is adjusted such that the modulation indicator LED just starts to glow, or that the brightness of the LED varies in sympathy with the beat frequency of the VCOs. The output of the ADSR to which the VCA is wired now controls the dynamic amplitude envelope of the output of the VCA. The desired attack, decay times etc. are set up on the ADSR controls (5). One should then experiment with different ADSR time constants and note the difference in the resulting notes. The tonal differences caused by a slower attack, compared to those obtained with corresponding alterations in the control envelope fed to the VCF (tracking filter) are particularly striking. It is well worthwhile feeding the same ADSR envelope to the VCA and a VCF, and then switching between the two to hear the changes in tonal character which result. In this way one can obtain a better idea of when to use the VCA, and when to use a VCF to obtain certain types of sound.



Linear amplitude modulation

Regardless or whether or not the VCA is used with envelope shaper control, the linear gain stage of the VCA can be employed to provide periodic amplitude modulation of the output waveform (tremolo). The low frequency modulation signal (e.g. a triangle waveform provided by one of the LFO modules) is fed to the AM input socket, and the GAIN control is set to the mid-position, thus allowing the maximum modulation depth of 100% to be obtained if desired.

Figure 9. DUAL VCA under envelope shaper control.

Figure 10. DUAL VCA used to provide an 'expression' or 'swell' pedal.

Figure 11. Setting up a simple attack-decay envelope on the ADSR module.

The actual modulation depth is determined by the AM control, and can be continuously varied down to 0.

Expression and swell pedal

If a pedal is connected to the AM input socket of the VCA it can be used either as an 'expression' pedal or to provide 'swell' effects. In the former case the pedal simply varies the amplitude (i.e. the volume) of the output signal as desired whilst playing. The dynamics of the notes are still controlled by the envelope contour fed to the exponential gain stage of the VCA and/or by a tracking filter. In the case of a 'swell' pedal, the envelope control input is switched off and the tracking filter set to pass the VCO signal unaltered. The pedal alone is then used to control the amplitude envelope of each note. A swell pedal is often used to vary the envelope contour of an electronic guitar signal. By providing a much slower attack phase, the guitar can be made to imitate the sound of a violin.
Figure 10 illustrates the basic front panel settings for pedal control of the VCA. Depending upon the type of effect required, the envelope control input is switched either on or off (1), and the output of the pedal connected to the AM input socket (2). The GAIN control is turned down to zero (3), whilst the AM (modulation depth) control on the other hand is set to maximum (4).
If the output of an ADSR envelope shaper is connected to the AM input via a patchcord, a linear amplitude envelope is obtained. Thus by slightly turning up the GAIN control, there is the possibility of providing an initial variable 'offset' to the amplitude envelope.

ADSR envelope shapers

Main characteristics of the ADSR module

The ADSR modules of the Formant are internally hardwired to the VCF(s) and DUAL VCA. They provide a varying control voltage with exponential attack and decay characteristics, which can be used to vary the dynamic amplitude and harmonic content of a note.
The four parameters of the envelope contour which can be modified are: attack time, decay time, sustain level, and release time. The envelope shaper has two basic modes of operation, ADSR and (triggered) AD, and these are selected by means of a toggle switch on the front panel. The envelope contour can be monitored by means of an LED indicator, the brightness of the LED varying in accordance with the envelope voltage. Three main types of envelope contour can be obtained with the Formant ADSR module, these are now described in turn.

Attack-decay contour

The simplest envelope contour is that consisting only of attack and decay phases, i.e. when a key is depressed, the envelope voltage rises to a peak, whereupon it immediately begins to decay. For this type of contour the ADSR is operated in the triggered mode, i.e. once a note is struck, a fixed attack-decay sequence is initiated, regardless of when the key is released. AD contours are typically used to imitate percussive sounds, such as those produced by a vibraphone, xylophone etc.

The basic settings for an AD envelope are shown in figure 11. The triggered AD mode is selected (1) and the sustain control is set to zero (2), whereupon the desired attack and decay times are set up on the corresponding controls (3, 4). Depending upon whether the envelope control voltage is applied to the DUAL VCA or to a VCF (connected as a tracking filter), a variety of different tone colours can be produced by experimenting with different attack and decay periods. Interesting effects can be obtained by inverting the normal envelope of a note to provide very long attack times and relatively short decay times. Totally unnatural sounds, similar to those produced by playing a tape backwards, are the result. When playing

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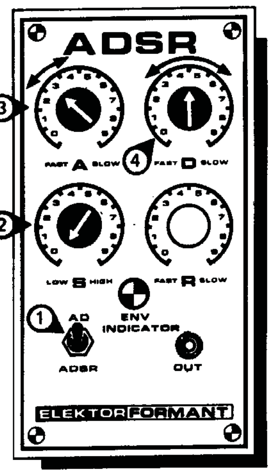

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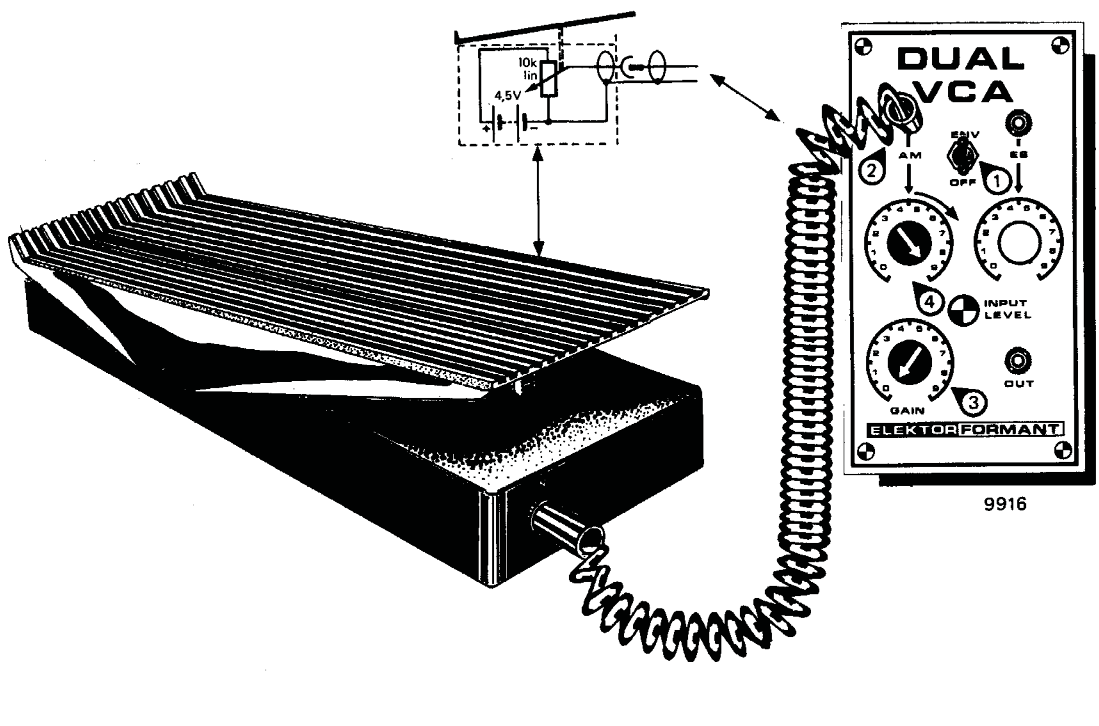

with the ADSR module in the triggered AD mode it is necessary to use a slightly different keyboard technique to that employed when the sustain-release portion of the envelope is determined by the moment the key is released.
It is important to ensure that the subsequent key is struck only *after* the previous key has been released.
With certain types of instrument which have a percussive AD contour, releasing the key during the decay phase will cause the note to die away much more rapidly than would be the case were the key held down. Thus there is a relatively slow decay period, followed by a much faster release. To simulate this type of envelope contour the ADSR/AD switch is set to the ADSR mode and the sustain level set to 0% (sustain control turned fully anticlockwise). A very short attack time is selected, with a relatively long decay time; the release time should be set shorter than the decay time.

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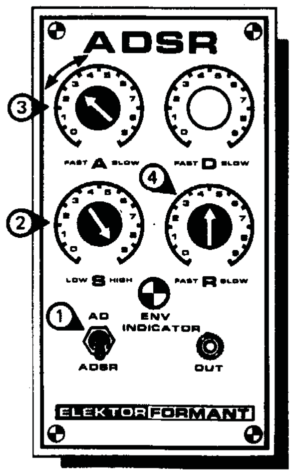


Figure 12. Setting up an attack-sustain-release envelope on the ADSR module.

Attack-sustain-release contour

With an attack-sustain-release contour, once the note reaches its peak it is sustained at a 'steady-state' level until the corresponding key is released. This type of amplitude contour is produced by e.g. a pipe organ. To generate an ASR envelope, the AD/ADSR switch (1 figure 12) is set to the ADSR position, the sustain level is set to 100%, and the desired attack and release times are selected as required.

Attack-decay-sustain-release contour

The most complex voltage envelope which can be provided by the ADSR module consists of separate attack, decay, sustain and release phases. Once it has reached its peak value, the envelope voltage decays slightly before settling at the steady-state or sustain level. When applied to a VCF, this type of contour is particularly useful for simulating the changes in harmonic content which occur in the course of a note. A good example is the dynamic harmonic structure of a trumpet note, where the harmonic content initially increases to a maximum, before falling away slightly as the note is established. The basic control settings for an ADSR curve are identical to those for an ASR curve, with the exception that the sustain level is reduced to whatever value is desired, and a suitable decay time is selected. Note that the full ADSR contour will only be obtained if the attack + decay time is longer than the period for which the key is depressed, and if a sustain level of greater than 0% is chosen. If the key is released before the sustain level is reached, then the release period is initiated prematurely, and either AD or ADR curves may be produced. If the sustain level is 0%, then, once again, only AD or ADR contours may be produced, depending on when the key is released.

LFOs

The LFO module contains three independently variable low frequency oscillators, each of which provides a choice of three output waveforms. The frequency range of each LFO extends from roughly 5 mHz (i.e. 1 cycle every 3 minutes) to 20 Hz. Many comparably-sized synthesisers have only one low frequency oscillator, so that a VCO often has to 'double up' as an extra LFO. By having three separate LFOs each offering a choice of waveforms, a wide variety of complex (cross-)modulation effects can be obtained.
LFOs 1 and 2 are identical, and provide squarewave, triangle and sawtooth (positive-going ramp) waveforms. LFO 3 produces a triangular waveform and two sawtooth waveforms in antiphase, i.e. with positive and negative going ramps respectively. The triangle output in particular is suitable for 'musical' modulation effects (vibrato, tone colour modulation, tremolo, etc.), whilst all three waveforms can be used for 'electronic' modulation effects, in which pronounced, 'non-musical' signal modulation occurs. The output of each LFO can be monitored by means of an LED indicator, the brightness of which varies in sympathy with the (triangle) output.

NOISE module

The noise module has two low frequency outputs which provide white and pink noise (white noise with the bass components of the noise spectrum boosted) respectively, and a third extremely low frequency 'random voltage' output, the fluctuation rate of which can be varied manually. The noise signals can be patched into the Formant signal path via the ES socket of a VCF or the VCA. The noise sources can be used (with suitable filtering) to simulate the sound of cymbals and brushes, as well as unmusical sounds such as rain, wind, surf, thunder, etc. The attack transients of many orchestral instruments have large amounts of filtered white noise.
The random voltage is useful for adding controlled 'doses' of random modulation so as to produce slight variations in an otherwise steady-state note. The fluctuation rate of the random voltage can be monitored by a front-panel LED similar to those of the LFOs.

COM module

The COM contains conventional tone (bass, mid-range and treble) and volume controls, which can be used to complement the VCF(s) and tailor the overall frequency response of the synthesiser to suit individual requirements (type of loudspeaker used, room characteristics, etc.). In addition, the COM module also monitors the 3 supply voltages of the Formant and the presence (or absence) of a gate pulse.

control settings for the simulation of different instruments 3

The control settings discussed in the previous chapter were largely limited to a description of how each module in the Formant system was operated. This chapter starts where the previous left off, and by presenting a series of basic control settings for different instruments, attempts to give the reader an insight into how the various modules can be used *together* to tailor the parameters of a note in order to achieve the desired sound.
By no means are all the significant tone settings described; rather the chapter

illustrates the various ways in which the tonal characteristics of a note can be modified and controlled. The following settings are only intended as an example, and the reader is encouraged to experiment with variations so as to appreciate the fine discriminations in tone colour of which the Formant is capable.
Considerable stress is laid upon the improvement of instrument timbre by means of resonance filtering. The examples in this respect are valid not only to the Formant, but can be applied with success to other synthesisers, and indeed practically any electronic instrument (electric guitar, string synthesiser, etc.).

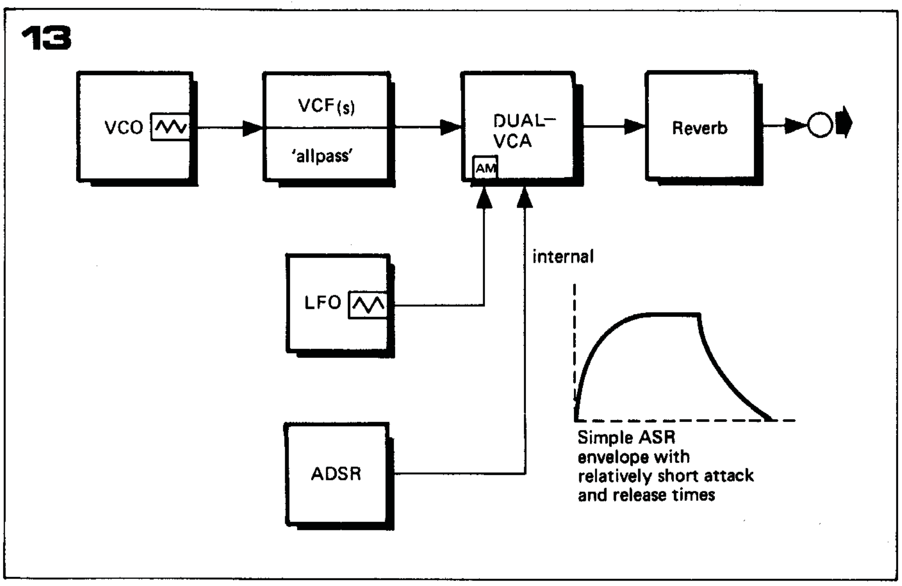
Synthesiser size and musical performance

It is sometimes said that a synthesiser with only one VCO is of little use. Such a small instrument is extremely limited in its capabilities. However this statement is almost completely unjustified, particularly when one considers that a large number of musical instruments are monophonic (including the majority of synthesisers, be they ever so large) and have only a single sound source, take for example a trumpet, clarinet, saxophone, flute, oboe, etc.
If the Formant is to be used to simulate the sound of one of the above-mentioned instruments, then it is *recommended* that, as in the case of the original, only one sound source, i.e. one VCO, is used, since as a result of the phasing effect caused by slight mistuning, several VCOs would normally alter the tone of the instrument. Thus there are a variety of applications, for which only a single VCO is required. If additional phasing or chorus effects are desired, then these can easily be provided by pulse width modulation of the squarewave output. Using only one VCO also allows the effect of the toneshaping modules in the Formant, i.e. the VCF(s) and the VCA, to be better appreciated. Thus in a large number of the following control settings for simulating certain common instruments, only one VCO is employed.
Another reason why playing with a single VCO has its advantages is that because of the monophonic keyboard, the interval of any chord will remain the same. If the VCOs are tuned in a chording group, playing for extended periods in this fashion can have rather a monotonous and 'listener-fatiguing' effect.
It can be seen, therefore, that the synthesiser enthusiast with a limited budget can start off quite satisfactorily with a 'mini-Formant', consisting of a single VCO, one state-variable VCF, one DUAL-VCA, and a single envelope shaper.
On the other hand, for the more experienced synthesiser player who wants to create finely differentiated and complex tone colours, there is virtually no limit to the number of modules which can be incorporated into the Formant. Walter Carlos, who is probably one of the most well-known synthesiser artists, normally used 8 VCOs, 3 VCAs, and 8 ADSR envelope shapers, and he felt that 8 envelope shapers was *too few!*
More than with any other instrument perhaps, the synthesiser places great demands upon the proficiency of the person playing it. On a vibraphone, for example, it is very difficult to produce a sound which is downright discordant or unpleasant. With a synthesiser, on the other hand, nothing could be simpler! Whether the synthesiser contains a small number or large number of modules, the performance of the instrument is determined by the knowledge and skill of the player.

Simple flute sounds

The tone-shaping process used to simulate basic flute-like sounds is illustrated in the block diagram of figure 13. The flute is a monophonic instrument with a single sound source; its timbre is rather 'hollow' and dark. A single VCO is thus used as the tone generator, and the triangle waveform, which has a mellow, 'woodwind-like' character, is chosen. The notes of a flute do not exhibit marked variations in dynamic harmonic content, and a reasonably accurate imitation can be obtained by using the ADSR (via the DUAL VCA) to control their dynamic *amplitude* contour only. The VCF(s) is (are) therefore set to pass the VCO signals unaltered ('allpass').

Figure 13. Block diagram illustrating the tone shaping process for simple flute sounds.



The dynamics of a flute note are characterised firstly by a soft, non-percussive attack; the note is then sustained for as long as the flute continues to be blown, and then decays fairly rapidly once the supply of air is terminated. Thus a simple attack-sustain-release envelope with reasonably (but not excessively) short attack and release times, should be set up on the ADSR module. In addition, flute tones are often characterised by a varying degree of tremolo. This can be simulated by feeding a triangle signal from one of the LFOs to the AM input of the DUAL VCA. The intensity of the tremolo can be set by means of modulation depth control beneath the AM socket.
As has already been mentioned, a reverberation unit is an extremely useful adjunct to the Formant, since it eliminates the rather 'dry' and 'clinical' sound to which synthesisers are prone. By allowing notes to decay much more gradually, the music becomes 'livelier' and considerably more realistic.
It is worthwhile experimenting with variations on the basic settings shown in figure 13. For example, if the basic configuration is preserved, but an attack-decay envelope contour is selected (ADSR is switched to triggered AD mode), then with 0% sustain level, a short attack time and a relatively long decay time, increasing the reverberation intensity will produce sounds similar to an electric piano or distinctly percussive sounds akin to those of a vibraphone.

Simple brass sounds

The following example demonstrates how a tracking filter can be used to simulate the variations in dynamic harmonic content which are typical of brass instruments. Most brass instruments have a bright, hard, 'shiny' timbre, which is due to the comparatively large proportion of fairly intense higher harmonics in the note. Thus a suitable VCO waveform would be either the sawtooth or spaced sawtooth.
As is the case with the flute, brass instruments are monophonic, and

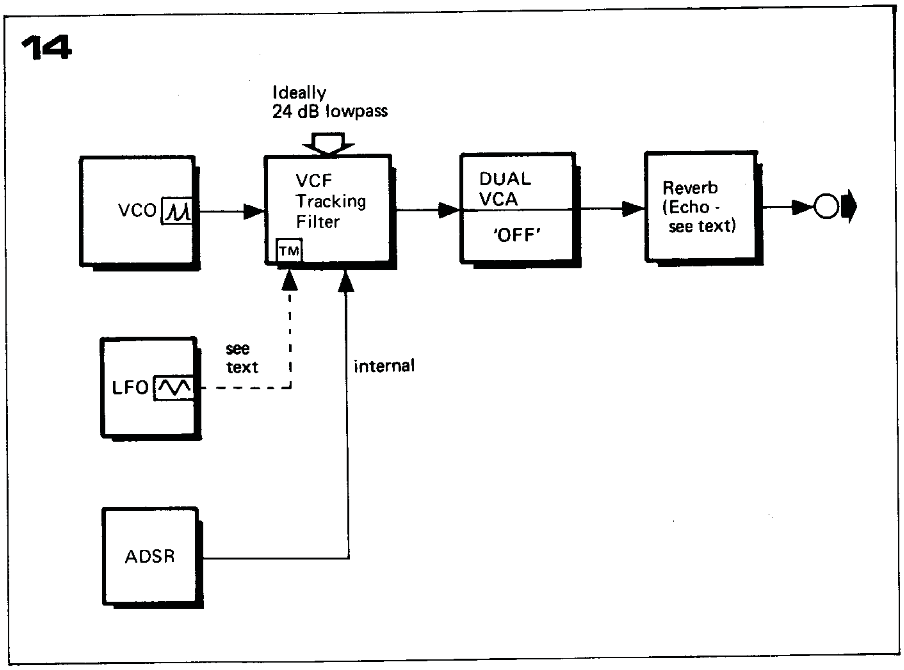


Figure 14. Block diagram illustrating tone shaping process for simple brass sounds.

Figure 15. Improved brass sounds.

Figure 16. Tone shaping process for synthesising clarinet, bassoon, and oboe sounds.

comprise a single sound source. A distinguishing feature of brass instruments generally is a pronounced build-up of harmonics during the attack period of the note. To reproduce these marked changes in tone colour, the ideal choice is to use the 24 dB VCF as a tracking filter controlled by an ADSR (see figure 14). The VCA is switched to 'OFF', i.e. is no longer under envelope control and passes the VCF output unaltered. In the absence of a 24 dB VCF, it is of course perfectly possible to employ a 12 dB VCF; however for brass instruments in particular, the severe filtering provided by the 24 dB VCF produces a more realistic sound.

To simulate basic brass-type sounds an envelope contour with a relatively long attack constant (assuming one is not playing too quickly) should be chosen. With a simple attack-sustain-release contour the note is sustained until the supply of air is cut off, whereupon it decays fairly rapidly (short release time). The Q of the VCF should be set to zero, or to only a very small value, whilst, particularly in the case of the 24 dB VCF, the ENV control should be adjusted extremely carefully. The realism of the sound can again be considerably enhanced by a certain amount of reverberation being added. Using a sawtooth waveform and turning up the ENV control slightly, it is possible to imitate a French horn or flugelhorn with considerable accuracy.

Even more detailed simulation can be obtained by employing complex attack-sustain-decay-release contours to reproduce the slight drop in harmonic content once a note is established which is characteristic of many brass instruments (see figure 5, chapter 9). 'Ethereal' brass-type sounds can be produced by 'shallow' timbre modulation using the triangle output of an LFO, and turning up the ENV control fairly high. If adjusted correctly, the instrument will sound like a mixture of a softly-played trumpet and a brightly played flute. If an echo unit is used, the brass sounds of the Formant are lent a chorus-like, multi-voice character.

Realistic tone colour through resonance filtering

Although the above control settings realistically simulate the dynamic tone colouration of brass instruments, the sound still remains slightly unsatisfying when compared to the 'real thing'. The natural sound of a trumpet is brighter and more intense, the trombone is 'fuller', more rounded, whilst the tuba has a richer bass. The corresponding synthesiser sounds seem to have less character, are a slightly 'pale' imitation. The reason for this is that the trumpet, trombone, French horn, tuba, etc. are distinguished not only by their different pitch and dynamic harmonic content, but also by differences in their resonance modes. Due to its particular shape and size, each instrument possesses different bandpass resonances (formants), which greatly influence its basic timbre.

In the case of the tuba, for example, the predominant bandpass resonance is much lower (around 250 Hz) than that of the trombone or the trumpet, and this is partially responsible for the rich bass character. In the case of the French- and flugelhorn, because of their greater Q factor, the primary resonances cause a much more marked colouration of timbre than is true of the other brass instruments.

The importance of resonance filtering is not limited to brass instruments; the resonant frequency bands or formants of woodwind instruments such as bassoon, oboe, clarinet, flute, etc. have just as much effect on their tone colour. In the case of string instruments, the situation becomes more complicated. For although bandpass resonances play just as significant a role in determining timbre, the large number of formants which string instruments possess make them difficult to simulate with limited resources.

The following examples illustrate how resonance filtering can provide a more realistic timbre when synthesising brass, woodwind and string instruments. For simple resonance filtering, either the resonance filter module itself, or the 12 dB VCF (bandpass mode) can be used. For simulating several bandpass resonances, such as when synthesising string sounds, the resonance filter module is a 'must'.

Improved brass sounds

The basic control settings described in the section on 'Basic brass sounds' are retained, the difference is that a resonance filter is introduced into the signal path (see figure 15). The main fixed resonances of a number of brass instruments are listed in the table shown below (see also the table contained in the chapter on the resonance filter module). When the table was compiled, the author experimented with different centre frequencies of the resonance filter for a variety of different Q factors, until the resultant sound was closest in each case to the original instrument. The results obtained largely coincided with the actual resonant frequencies (SIRKER 1974).

Instrument	Main resonance
trumpet	1500 Hz
flugelhorn	1000 Hz +)
trombone	600 Hz
French horn	400 Hz +)
tuba	250 Hz

+) increased Q factor

Initially the centre frequency of the resonance filter is set to the nominal value listed in the table. The Q of the filter is adjusted to roughly its mid-position. For trumpet and trombone sounds, spaced sawtooth waveforms are particularly suitable, whilst ordinary sawtooth will suffice for tuba, French horn and flugelhorn. The ENV control should be turned fairly far up for trumpet and trombone, and even further up for the other brass instruments mentioned above. The Q of the filter should also be increased for the flugelhorn and French horn. For the resonance filtering to achieve the desired effect,

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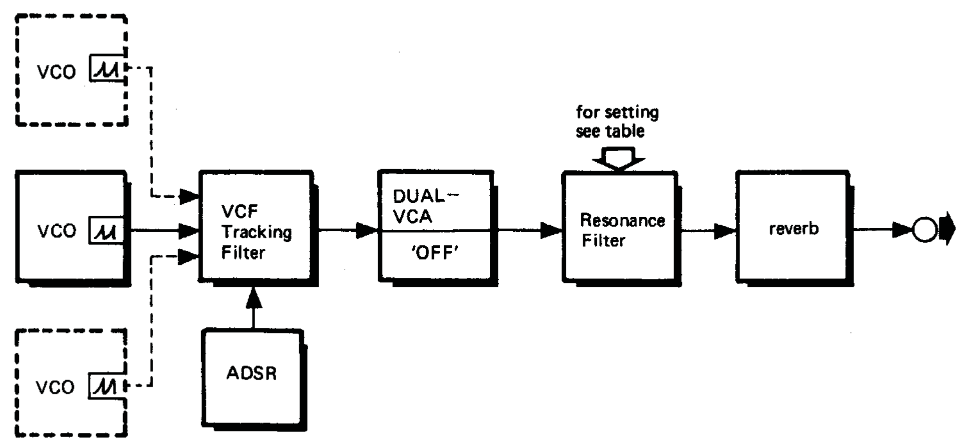

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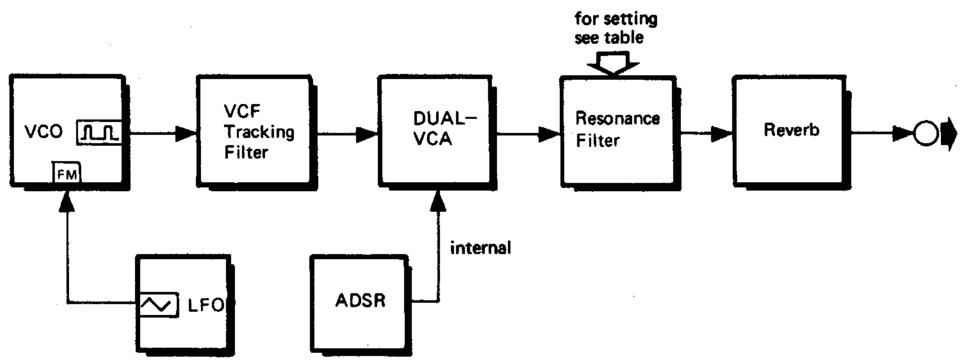

the synthesiser should be played within the natural frequency range of the instrument concerned. In each case care should be taken to ensure that the pitch of the notes played are on average below the centre frequency of the filter. If this is not the case, the filter will fail to enhance various upper harmonics of individual notes, and the desired tone colouration will be lost.
To assess the effect that resonance filtering has on the timbre of the sound, one should play several tunes which extend over a number of octaves. The fixed centre frequency of the filter will cause the tone colour of the sound to vary for different pitched notes; the overall tonal character of an instrument is after all determined by how it sounds at different pitches. Think for example of the sonourous timbre of a cello at the lower end of its scale compared to how it sounds on the higher notes. If sufficient care is paid to exact adjustment of the Q factor and centre frequency of the resonance filter, and to the careful setting of the ENV control of the tracking filter, the resulting sounds should have a thoroughly convincing musical quality. The improved brass sounds are thus a good example of the type of musically satisfying effect that can be obtained using only one VCO.

Multi-voice brass sounds

As a conclusion to experimenting with the synthesis of brass sounds, one should try employing several VCOs, tuned to e.g. simple major or minor chords. Depending upon the control settings, fanfare-like effects, or background 'brass section' sounds can be produced. By varying the envelope contour of the signals, either short, rhythmic notes, or notes which rise and decay extremely slowly, can be obtained. Multi-voice brass sounds are ideally suited to provide background accompaniment to solo instruments such as a mini-Formant or electric guitar. The invariant chords are not perceived as irritating of fatiguing, but rather provide a pleasant contrast to the foreground melody-line.

Some typical synthesiser effects with no natural equivalents

If one retains the basic control settings of figure 15 but varies the envelope contour controlling the VCF (tracking filger) to provide very short attack times and slightly longer decay times, extremely attractive *metallic sounds* are obtained. By varying the centre frequency and Q of the resonance filter, a spectrum of sound from very dark, 'bassy' tone colours through a variety of timbres including that of metal strings being struck, to very bright, sharp string sounds, can be distinguished.
The majority of these sounds have no natural equivalent, and appear more or less 'artificial' to the ear. The 'electronic' sound can be further enhanced by increasing the Q of the tracking filter. It is also worthwhile experimenting with the highpass, lowpass and notch filter functions of the VCF used as resonance filter. Starting with the basic configuration shown in figure 15 it is relatively simple to generate a variety of different 'fantasy' sounds, which although they do not resemble conventional instruments nonetheless can be extremely pleasurable.

Oboe, bassoon and clarinet

A number of conventional instruments also have pronounced tone colouration, e.g. the oboe, bassoon and clarinet. As was the case with the flute, the variations of harmonic content during a note are of secondary importance. As far as their synthesis is concerned, this means that the DUAL VCA is used to determine the dynamics of the individual notes. The amplitude dynamics are similar to those of the flute, i.e. a simple attack-sustain-release curve is all that is required

for a reasonable simulation. Each of the instruments, however, has a highly individual timbre, which cannot be accurately imitated without resonance filtering.
The sound of a symmetrical squarewave is quite similar to that of a clarinet, but without the presence of formants, the 'synthesised' clarinet is unconvincing at the lower end of the scale; sounding 'sterile' and electronic, with the typical 'nasal tone' missing on the lower notes. The timbre of the oboe and bassoon is not similar to that of any of the VCO waveforms, nor can it be derived with the aid of a tracking filter. In other words, although it is possible to imitate basic brass sounds fairly successfully without resonance filtering, the same is *not* true for the clarinet, whose timbre can only be accurately synthesised with the aid of a resonance filter. The effect of resonances on the timbre of the oboe and bassoon is even more marked than is the case with the clarinet.
Realistic clarinet, oboe and bassoon, not to mention saxophone and flutes, are a distinguishing feature of the Formant – due to the emphasis on resonance filtering as part of the basic design concept.
The following table lists the main fixed resonances of the clarinet, oboe and bassoon (see also the table in chapter 8 Part 1).

Instrument	Main resonance
clarinet	1000-2000 Hz
oboe	1300-1700 Hz
bassoon	440 Hz

For the synthesis of bassoon and oboe sounds, a heavily asymmetrical squarewave is the most suitable type of VCO waveform, whilst a symmetrical squarewave (obtained by careful adjustment of the PW control on the VCO) is the obvious choice for the clarinet. For all three sounds a relatively high Q factor is needed, especially in the case of the oboe and bassoon. The tracking filter is set to 'allpass', although by suitable adjustment of the OCTAVES control the higher harmonics of each instrument should be slightly muted. The dynamics of the individual notes are controlled by means of the DUAL VCA (see figure 16). If the resulting sound is frequency-modulated slightly, a 'singing' effect, which reproduces the typical vibrato of these instruments, is obtained. Once again, the use of controlled amounts of reverberation is recommended. With careful adjustment of the relevant parameters, the Formant will imitate the three above-mentioned instruments with a remarkable degree of accuracy. In the case of other reed instruments, the importance of resonance filtering is equally crucial. By varying several of the parameters in figure 16, sounds covering e.g. the range of soprano to baritone saxophones can be produced.

String instruments

For brass and woodwind instruments the role of bandpass resonances in determining the basic timbre of the instruments was vital. In the case of string instruments however, the importance of resonance filtering is less important. The acoustic response of string instruments is much less coloured by the effect of a main fixed resonance. This is on account of the particular shape of most string instruments, which possess not one or two, but a large number of resonances extending over the entire audio range. This means that a single bandpass filter is insufficient; a resonance filter bank such as the Formant RFM, containing several independently variable bandpass filters is required. Even then however, the simulation of individual string instruments remains a problem. The note from a violin or cello is initiated when the bow comes into contact with a string; that corresponds to an envelope contour with a slow attack constant fed to the DUAL VCA. As the string begins to stop oscillating, so the note begins to decay gradually. However during normal playing, the string will be maintained in a state of oscillation by further movement of the bow. The resulting envelope of the note is extremely complex, too complex to be accurately reproduced using a simple envelope shaper.
However, a *reasonable* approximation to a string instrument can be obtained with a straightforward attack-sustain-release envelope, leaving aside the 'steady-state' dynamics of the note and the (important) nuances of multi-voiced orchestral string sounds for the moment. The question may be asked whether, in view of the difficulties involved, one should renounce the attempt to synthesise string instruments. The answer is certainly not! However one should simply be aware of the fact that 'string sounds' will not have quite the convincing realism as the other types of instrument which have already been discussed. Nonetheless there are many musical applications which call for a sound which is similar to that of a cello or violin, and these the Formant is perfectly capable of synthesising.
No block diagram is given for the synthesis of string sounds, since they can in fact be produced in several different ways. Suitable types of VCO waveform (one should begin with a single VCO) are sawtooth, spaced sawtooth or asymmetrical squarewave. The tracking filter is set to the 'allpass' mode, to pass the VCO signals unaltered, with the exception of the upper harmonics of the signals, which, by adjusting the OCTAVES control, are slightly attenuated. In practice a slight amount of vibrato has proven useful.
The following table gives an approximate guide to the centre frequency of the resonance filter module.

double bass	Main resonance*
violin	approx. 400 Hz
cello	approx. 200 Hz
double bass	approx. 100 Hz

As has been explained, the tonal significance of the primary resonances listed in the table is less profound than was the case with brass or woodwind instruments. Nonetheless, useful improvements in tonal realism can be obtained with the aid of the resonance filter module. The additional use of a phaser or flanger is also recommended. The comb-filter like response of the phaser provides a number of fixed 'peaks' or resonances in the signal, which, if carefully situated, will considerably enhance the timbre of the sound.
The DUAL VCA is controlled by a simple attack-sustain-release envelope, with a fairly slow attack time and relatively fast release. A healthy amount of reverberation is also very important. To get as close as possible to the natural sound of string instruments, it will be necessary to experiment with varying the parameters of the resonance filter, the placing of the peaks and notches provided by the (unmodulated) phaser, as well as the attack and decay times of the envelope contour and the amount of upper harmonic attenuation introduced by the tracking filter.
Once the optimum settings – particularly of the resonance filter – for double bass, cello, violin and viola have been determined, switching in a second VCO which is tuned in unison with the first will produce an 'orchestral' effect. This is due to the creation of beat frequencies caused by very slight mistuning between the two VCOs, and lends the sound a lush, 'choral' character. The effect can be further intensified by switching in a third VCO, also tuned in unison with the other two, and by independently pulse width modulating the squarewave output signals of one, two or all three VCOs. Another possibility is to frequency-modulate (vibrato) each VCO, when the latter are tuned in unison or to chording intervals. By carefully adjusting the vibrato frequency and the modulation depth (intensity) it is possible to obtain a convincing simulation of a note played by a large string section. Slow modulation of the phaser response (i.e. gradually sweeping the peaks in the response up and down the audio spectrum) can further reinforce the orchestral timbre of the note.
Finally, it is worthwhile experimenting with tuning the VCOs in chording groups. If acceptable orchestral sounds have been successfully obtained, one should try synthesising minor chords. These more complex types of sound are ideally suited as an orchestral background for other instruments.

Improved flute sounds

At the beginning of the chapter the control settings for synthesising simple flute sounds were described. However

*Ideally reinforced by resonance filter bank or comb filter

the attentive listener will have noticed that there were a number of deficiencies in the tonal character of the resultant sound. These can be largely eliminated by employing a slightly more elaborate tone-shaping process. The basic timbre of the simple flute sounds was determined *solely* by the character of the triangle waveform; the resulting sound being *fairly* similar to that of the (old) *woodwind* flute, but quite foreign to the 'silvery' timbre of a modern (German) flute. In addition, the characteristic enhancement of certain upper harmonics as a result of the instrument's resonances was also missing. A further inadequacy was the simulated tremolo, which in the case of a real flute is more than simple amplitude modulation. With a modern flute in particular, the amplitude modulation is accompanied by clearly discernible periodic fluctuations in tone colour. As the amplitude of the note increases during tremolo modulation, so the timbre of the note becomes brighter. This effect is especially noticeable on the lower notes of a German flute.
The question therefore arises, how does one go about simulating these characteristics? First of all, if one wishes to imitate the timbre of a modern (metal) flute, then a slightly asymmetrical squarewave is a more appropriate waveform to start with than the triangle. The intense upper harmonics of the squarewave must be severely suppressed with the aid of the 24 dB VCF. Depending upon the type of flute to be simulated, resonance filtering to create the corresponding formant bands is also necessary. As a rough guideline for the position of the main fixed resonance of a German flute one can start with a centre frequency for the bandpass filter of approximately 800 Hz.
The amplitude dynamics of the individual notes are tailored to a simple attack-sustain-release contour with the aid of the ADSR and DUAL VCA. To provide a more realistic tremolo effect, dynamic timbre modulation, if necessary controlled by hand, is used in place of amplitude modulation. The resulting sound should be much more 'lively' as a result of this step. In comparison, amplitude modulation alone sounds expressionless.
The basic set-up for improved flute sounds is illustrated in figure 17. A few points on the nuances of the control settings: initially the *symmetrical* squarewave, which sounds similar to a clarinet or woodwind flute, is selected. The OCTAVES control on the 24 dB VCF (switched to the lowpass mode) is gradually turned anticlockwise, whilst playing several notes on the keyboard, until a comparatively soft, mellow, flute-like timbre is obtained. Using the triangle output of an LFO, a slight amount of timbre modulation is introduced (TM input and control on the VCF); the frequency of the LFO signal should be kept fairly low. With a medium to high Q factor, the centre frequency of the resonance filter is then set to a roughly 800 Hz, thereby ensuring that the lower notes of the flute in particular lose their 'electronic' character. The DUAL VCA is used only to control the amplitude envelope of each note; the AM control is turned down to zero.
Once the basic control settings have been carried out, there comes an extremely important adjustment — the PW control of the VCO is turned very slightly, only 1 . . . 2 mm, off-centre (the direction is irrelevant). This has the effect of providing a slightly asymmetrical squarewave, and changes the timbre of the sound from that of a woodwind flute to the desired 'silvery' tone colour of a metal flute. The setting of the PW control should be carefully checked *by ear*, for it is of critical importance for a realistic simulation of the flute tones. It is also important that the OCTAVES control of the tracking VCF be turned down far enough to eliminate any 'buzzy', 'electronic' edge to the sound caused by the upper harmonics of the squarewave. On the other hand, it should not be turned down so far that the resultant sound becomes sterile and lifeless. It is recommended adjusting the TM control by hand whilst playing, so that at certain points during a note or tune the tremolo is further enhanced.
Finally, a tip for those bitten by the experimenting 'bug': by reducing the pitch of the flute and its resonant frequency to 400 Hz, or even as low as

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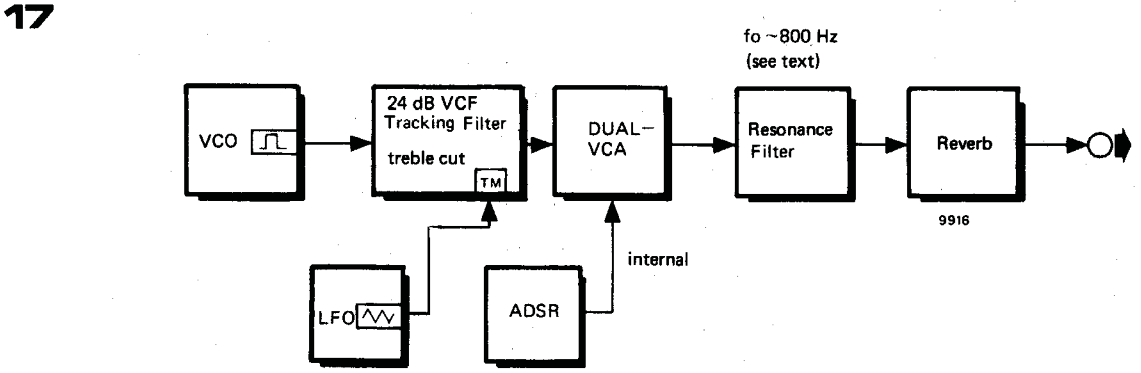

Figure 17. Control settings for improved flute sounds.

Figure 18. Control settings for organ sounds.

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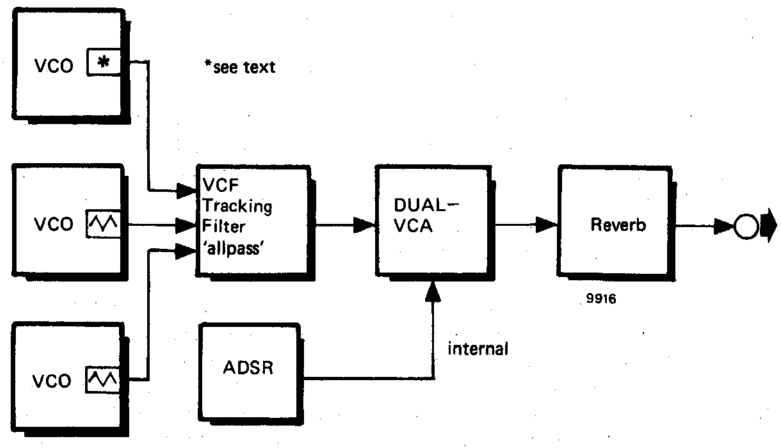

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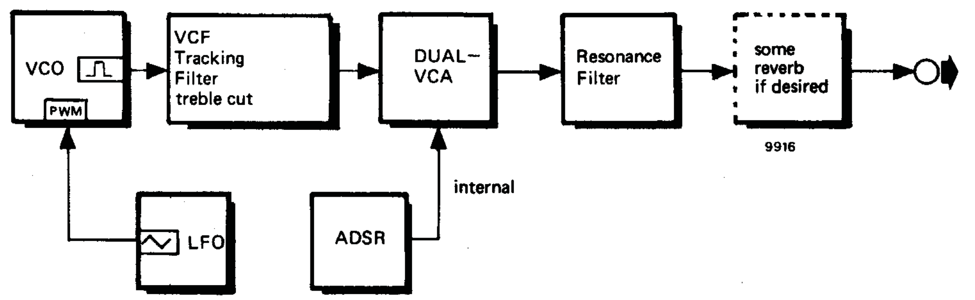


250 Hz, sounds similar to those of a bass flute can be obtained.

Organ sounds

In contrast to the above example of string sounds, the simulation of an organ is extremely simple (see figure 18). First of all the three VCOs are tuned to intervals of an octave and the triangle or sinewave is chosen as the output waveform for the two lower VCOs. If a church organ sound is desired, a symmetrical squarewave should be used for the top VCO; if a similar but less strident type of sound is required, the triangle waveform is more suitable. For a trumpet-like organ register, the sawtooth or spaced sawtooth should be used for the top octave. The proportions in which the individual octaves are mixed are adjusted using the OUT controls on each VCO. A simple attack-sustain-release envelope, which is fed to the DUAL VCA, determines the dynamics of the notes. Relatively short attack and release times are used. At the top end of the scale, where the fundamental frequency of the top octave signal is above 2 kHz, the organ tones have an agreable 'silvery' character, which is reminiscent of the sound produced by the smallest pipes of a conventional organ. For all types of organ sounds it is recommended that a considerable amount of reverberation be added.

Piano- and piano-related sounds

The range of keyboard instruments related to the piano is extremely wide. It extends from the grand piano, with its deep, soft bass notes (when played 'pianissimo') through honky-tonk type piano voices, to the bright highly percussive sounds of the harpsichord, spinet, etc. One could also include the wide variety of sounds obtained from different types of electronic piano in this group.
With the important restriction of being monophonic, the Formant is capable of synthesising the majority of the above-mentioned instruments. Disregarding peculiarities of individual keyboards, most piano-related sounds have in common the fact that, by depressing a key, a hammer strikes one or more strings, causing it (them) to vibrate at a given frequency. The individual notes have a fast, percussive attack, then decay relatively slowly. When the key is released, dampers, which were raised the moment the key was struck, fall back onto the string(s) and terminate the note comparatively rapidly.
A second characteristic of the majority of piano type sounds is that the variations in dynamic amplitude described above are generally not accompanied by pronounced variations in harmonic content. A further feature of piano-like sounds is the presence of periodic 'notches' in their response; for this reason an asymmetrical squarewave is the most obvious choice of VCO output waveform. For certain clavichord and electronic piano sounds the sawtooth and spaced sawtooth are also suitable.
In spite of the less marked dynamic harmonic behaviour of piano-related sounds the two tracking filters in the Formant are not left idle. The 24 dB VCF can usefully be employed to suppress the intense upper harmonics of the asymmetrical squarewave when imitating a pianoforte, removing the 'fuzzy' electronic edge to the notes. The resonance filter is used to reproduce the main fixed resonance of the piano (fairly low Q factor). A typical feature of the voicing of a piano and grand piano is the somewhat 'soft' bass notes, which sound slightly dry, with no suggestion of bass 'drone'. The middle register on the other hand sounds richer (generally speaking the hammer strikes several strings, as opposed to the bass notes where each hammer strikes only a single string), whilst the upper notes again sound relatively soft (assuming one is not playing 'forte'). This points to a bandpass resonance between the middle and lower end of the piano scale (depending upon the type of sound desired). The importance of a low Q factor has already been mentioned.
Widely differing resonance filter settings will be required for other piano-related instruments such as the spinet, harpsichord, and in particular the clavichord. The same is true for electronic piano sounds, where personal taste will largely dictate the parameters of the resonance filter.
The following description of control settings applies to the simulation of pianoforte and grand piano sounds (see figure 19).
An asymmetrical squarewave (roughly 10 to 20% mark-space ratio) is selected, whilst a relatively low frequency triangle waveform is fed as modulation signal to the PWM input of the VCO. The PWM control is only turned up very slightly, so that the phasing effect is *just* audible. With the aid of the 24 dB VCF (ENV control turned right down to zero) the upper harmonics of the squarewave are gradually attenuated (treble cut) until the desired timbre is obtained. First of all, however, a suitable envelope contour should be set up on the ADSR module and fed to the DUAL VCA (see description of attack-decay-release envelope in previous chapter). Finally the centre frequency of the resonance filter (low Q) is adjusted until the higher piano notes are sufficiently soft and the bass notes suitably 'dry', but without being too sterile.
For honky-tonk type sounds, two VCOs are slightly mistuned with respect to one another. In the case of a mini-Formant, where only one VCO is available, the frequency of the PWM modulation signal must itself be modulated. The percussive attack characteristics of honky-tonk notes can be reinforced by using an attack-decay envelope (very short attack- and decay times) to control the 24 dB tracking filter; the ENV control is turned up slightly, whilst the other settings of the VCF remain unaltered. The result is a characteristically 'jangly' sound, with very bright attack characteristics.

Gong, bell and glockenspiel sounds

All the various sounds described so far had in common the fact that they possessed a *musically coherent* harmonic structure. The last category of sound to be discussed in this section (see figure 20) is that of gongs, bells, and percussive instruments generally, which have a musically unrelated harmonic structure. This type of effect can be useful in a

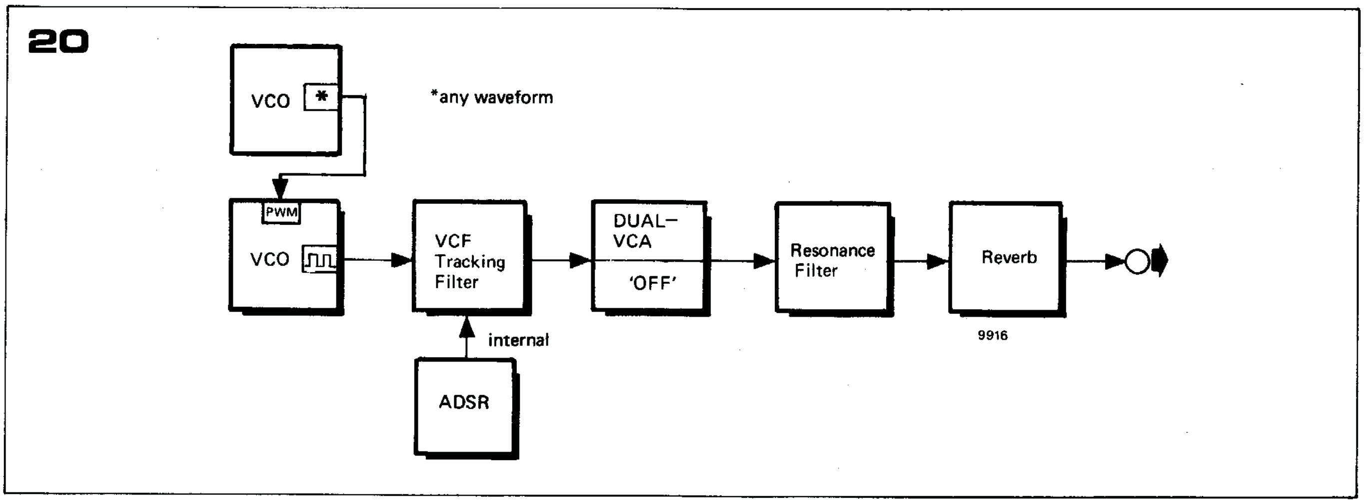


Figure 19. Control settings for simple piano-like sounds.

Figure 20. Tone shaping process for gong and bell-like sounds.

variety of applications from stage sound effects to 'experimental' music.
The basic control settings involved are comparatively simple. Two VCOs are employed, one providing a pulse width modulation signal for the other (this ring modulator like configuration is described in chapter 2). The 24 dB tracking filter under ADSR control is used to tailor the dynamics of the notes. A percussive attack-decay contour with a very short attack time and long decay time should be employed. The two VCOs are then tuned until the timbre of the resultant sound is similar to that desired (i.e. bell or gong-like); note that only one VCO output control is turned up.
Assuming the dynamic tracking filter has been switched in, the simplest types of sounds to set up are those of bells. Somewhere between the many 'short-wave' radio noises typically produced by ring modulators can be found a wide variety of differentiated tone colours covering the timbre of every possible type of small and large bell, gongs, clock chimes, etc. The more resonant, 'darker' timbre of large bells, and the lighter tone colour of small(hand) bells, can be usefully enhanced with the aid of the resonance filter.

Further tonal possibilities

The full capabilities of the Formant have by no means been exhausted – particularly if one has a large system with many modules. Many of the more complex control settings have been omitted for reasons of space, whilst comparatively little has been said on the subject of synthesising wholly 'electronic' sounds, the use of noise for special effects, or on the techniques of portamento playing. However this merely means that the Formant user has ample scope to experiment on his own and to enjoy the pleasures of discovering for himself the full range of tonal possibilities which the Formant can realise.

Using the FORMANT with other instruments

The Formant can not only be played as a solo instrument, but is well suited to be used in conjunction with a wide variety of other instruments. For enthusiasts of electronic music, there is the obvious possibility of using several synthesisers, e.g. a combination of a 'full-scale' and mini-Formant (consisting of only one VCO, VCA, VCF and ADSR). The large Formant is employed to provide an orchestral background, whilst its 'little brother' plays the melody line (generally speaking the two synthesisers would then be played by separate people). Another useful addition to the Formant is a good electronic piano or sophisticated organ. As a rule synthesisers gain much from being used in conjunction with other instruments. They can be likened to a large room full of different instruments, but with only one player available to use them. One way of circumventing this problem is to use tape, although there is the drawback of the relatively high cost of good quality tape machines. The flexibility and scope of a synthesiser are best demonstrated when used in a band, where its range of tonal possibilities can be exploited to contrast with the differently structured sounds of the other instruments. Such interplay with other types of instrument can also prove musically highly stimulating for the Formant user.
In conclusion, all that remains is to wish the prospective Formant user a fruitful 'voyage' of musical discovery and many happy hours of experimenting!

Literature:

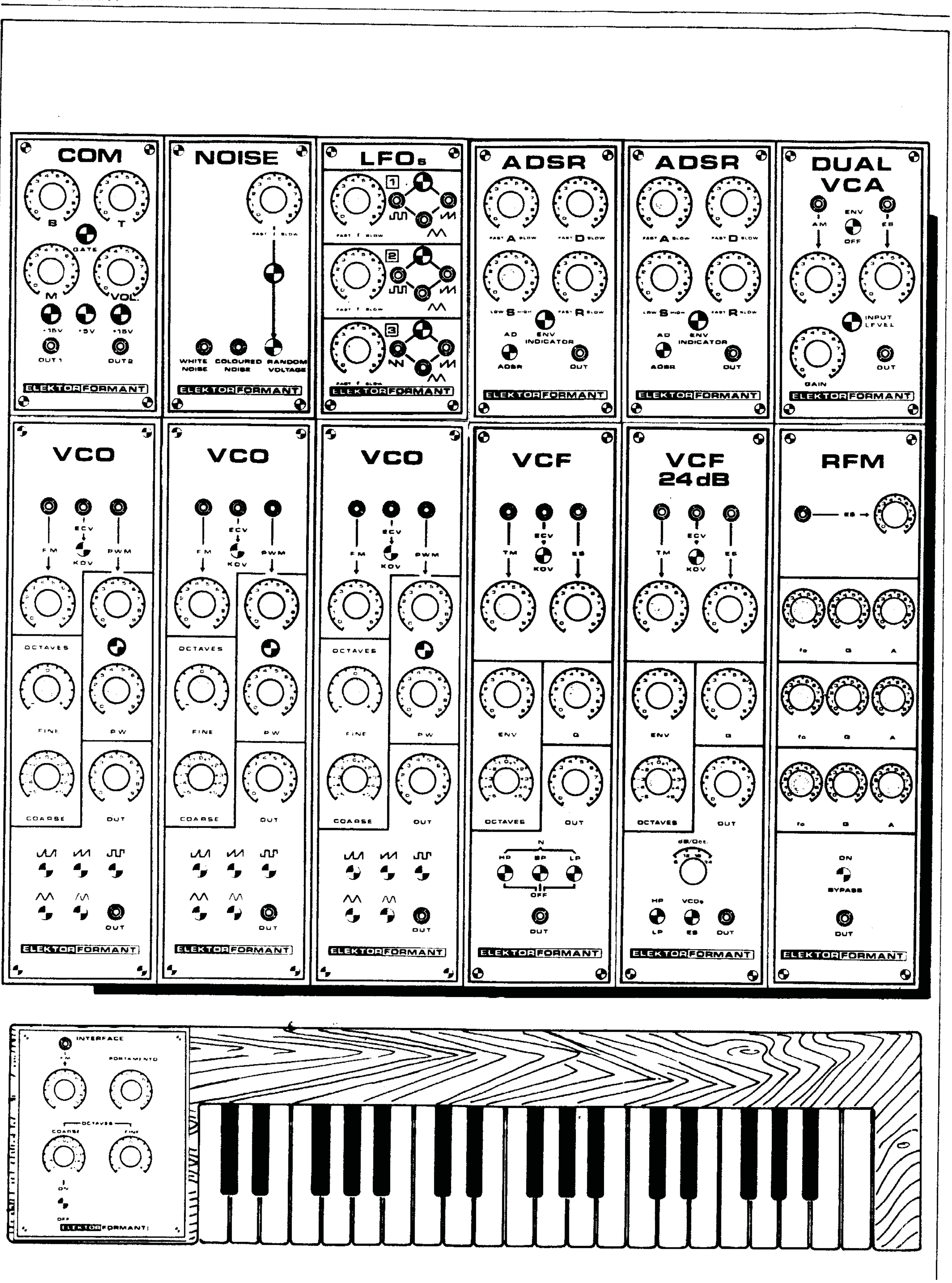
CARLOS, W.: 'WALTER CARLOS on synthesisers' letter published in WHOLE EARTH CATALOGUE 1974

HUTCHINS, C.M.: 'Instrumentation and methods for violin testing'. JAES, Vol. 21, Sept. 1973, No. 7, 563-570

STRONG, W. and CLARK, M.: 'Synthesis of wind-instrument tones'. JASA, Vol. 41, No. 1, 1967, pp 39-52

CHAPMAN, C. and DÜREN, W.: 'Synthesizer-Spezial'. Fachblatt-Musik-Magazin Dec. 1977 to May 1978

SIRKER, U.: 'Strukturelle Gesetzmaßigkeiten in den Spektren von Blasinstrumentenklängen'. ACUSTICA, Vol. 30, No. 1, 1974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NOISE
LFOs
ADSR
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DUAL VCA
VCO
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RFM
ELEKTOR FORMANT
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