Electronic Music and Creative Tape Recording

M. K. BERRY

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by

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CHAPTER 1

INTRODUCTION

Electronic music is the new music of the twentieth century. It covers a wide range of styles already, from avant-garde composers like Karlheinz Stockhausen and Pierre Henry through versions of classical pieces by Isao Tomita and Walter Carlos to electronic music in rock as composed and performed by such groups as Tangerine Dream and Kraftwerk. Electronic music also plays a large part in the remainder of pop and rock music, in fact there is scarcely a group without some sort of synthesiser or other effects generator.

Although music by Stockhausen and others may seem very unmusical to some people, most electronic music can be extremely expressive because it can be controlled very precisely in terms of pitch, amplitude and tone colour. This allows considerable scope for the composer/performer. With the addition of a two track tape recorder (stereo with independent control of each channel) or two seperate recorders, a complete composition can be made using simple electronic, and sometimes non-electronic, musical scores. One can experiment in one's own home and end up with some surprisingly good compositions. Electronic music, therefore, brings music-making into the home in a way which allows anyone to produce interesting music, even if they have never picked up a musical instrument before in their lives. Readers should take note, however, that to produce a few minutes of electronic music will require several hours work, and thus a great deal of patience is an asset.

This book sets out to show how electronic music can be made at home. It describes how the sounds are generated and then how these may be recorded to build up the final composition.

Of course it is not expected that the average enthusiast will own or have access to a professional recording studio with multi-track tape recorders and large mixing consoles. Thus this book will show how electronic music can be produced with the simplest and most inexpensive of equipment. The majority of homes now have a tape recorder of some description, and this is all that is required to begin composing tape music,as Chapter 2 explains.

For the constructor, several ideas are given to enable him to build up a small studio including a mixer and various sound effects units. All circuits shown in full have been built by the author, who is completely satisfied with their operation. Most of the circuits can be built by the beginner although it should be pointed out that the construction of the rhythm computer in Chapter 7 should not be undertaken as a first project, since it involves the use of CMOS integrated circuits which have to be handled with extreme care, lest they become damaged. Detailed precautions to be taken are given in Chapter 7.

Finally, the author wishes to thank Mr. Philip Marshall for checking, and correcting, the manuscript.

CHAPTER 2

MANIPULATION OF TAPE

The great advantage of tape over other recording mediums is that it can be cut and put back together in any way that is desired. Editing of tape is a technique which can be mastered quite quickly with a little practice. All you need is a sharp knife or pair of scissors, a reel of splicing tape and some blank leader tape. Ensure that the leader tape is the same width as the tape used on your recorder i.e. ⅛ inch for a cassette recorder or ¼ inch for an open reel recorder.

A very useful additional item, which is not too expensive, is a small splicing block, a device which holds the two pieces of tape to be spliced tightly so that a perfect joint may be made. Kits containing a splicing block, splicing tape and a knife can be bought quite cheaply.

To splice two pieces of tape together lay them, in line, on a flat surface, with the glossy side up, so that the two ends to be spliced are overlapping by about half an inch. Then with a sharp knife cut through both pieces of tape diagonally and remove excess pieces. Now butt the two ends together and lay a piece of splicing tape across the join. Press firmly on the join and then remove excess splicing tape with the knife so that there are no sticky edges. Fig. 2.1 shows how the join is made and also shows an alternative method used when the tape is required to be edited more precisely, whereby the tape is cut straight across at right angles to its path. This method though, should not be used too often as the finished splice is noisier than the diagonally cut splice.

It would be a good idea to practice splicing and editing with a cheap tape at first. Try recording a few notes with a microphone using any source which is available. Make the notes of a reasonable length and have a short pause in between each one. Now play back the tape and stop it precisely at the beginning of the first note you recorded.

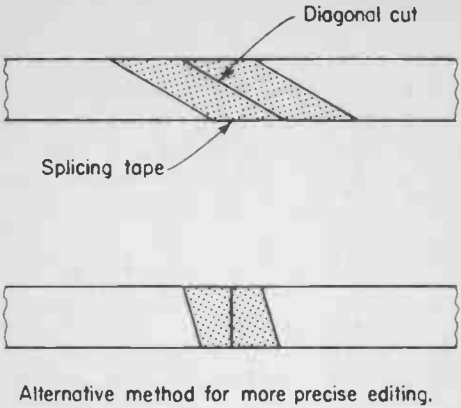


Fig. 2.1 Methods of splicing.

This will not be too difficult if the recorder has a pause button, but if it does not then you will just have to be quick with the 'stop' button. With the tape stopped, mark the tape on the glossy side two inches ahead of the point where it rests against the playback head. Use a coloured pencil or crayon to mark the tape. Now continue to play through the note and stop the tape at the end of it. Again mark the tape and then carry on to the beginning of the next note. Continue like this until you have marked the beginnings and ends of all the notes. Now remove the tape from the recorder and cut the tape at all the marks. You will be left with several pieces of tape. Some will have notes on them and some will be blank. Put the blank pieces to one side as these can be spliced together and the tape used again at a later date.

Now take the pieces of tape with the notes recorded on them. It might be an idea to select them so that they are in order of rising pitch, to make it easier when you rearrange them. How you rearrange them is entirely up to you. Remember that the notes can be of any length simply by cutting the tape to suit. The speed of the tape recorder needs to be known here if you want to splice in a note of a certain time length. If the speed of the recorder is 3¾ i.p.s. then a note 1 second long can be obtained by cutting a piece 3¾ inches long. Where a pause is required in the music, the required length of leader tape, or blank magnetic tape, can be spliced in. A length of leader tape, say 12 inches, should be spliced onto the beginning and also one at the end of the completed tape, so that the actual recorded part of the tape will not become damaged when fixing to spools. When the tape has been completed and the notes have been rearranged in the desired order, it can be played back through the receoder. The resultant sound may seem a little jerky and discontinuous, but as with all things, practice makes perfect, and the reader should soon find that his tape music becomes quite fluent.

Now that the basic method of manipulating tape has been mastered, there are one or two special effects that can be performed on a single tape receorder.

Speed Changing

Since most tape recorders have two or three different speeds, except cassette recorders, it is possible to record some material at one speed and play it back at another. The most common speeds are 1⅞, 3¾ and 7½ inches per second, but 15 i.p.s. is common on professional quality machines. Note that the speeds bear a direct relation to one another in that one is twice that of the next one below it and half that of the one above it.Thus if a certain pitch is recorded at 3¾ i.p.s. when it is played back at 7½ i.p.s., the pitch will have risen to a frequency twice that of the original. Those readers with an elementary knowledge of musical theory will know that where one note is twice the frequency then there is an octave difference in pitch between the two notes.

Thus by recording music at one speed and playing it back at the next highest speed it will be heard twice as fast and one octave higher in pitch. Now imagine that you have a rather complicated piano or guitar piece in front of you, and at present your piano or guitar playing is not that marvellous, so how do you record this piece?

Adjust the speed control on the tape recorder so that it is at one speed lower than the highest speed,that is, if the machine has 1⅞, 3¾, 7½ i.p.s., adjust it to 3¾ i.p.s. This is so that an unnecessarily low speed is not used, which would lower the quality of the recording. Now the piece of music to be recorded has to be played at half the intended speed and at one octave lower than it should be. It will be easy enough to play one octave lower, the only difficulties arising when having to play at half speed.If a mechanical metronome is available this can be set to half the intended final speed. Alternatively an electronic metronome can be constructed, details of which are given in Chapter 7.

When the piece has been recorded, change the speed to the highest and replay. You should be suitablypleased with the result. Note that there are slight differences in the reproduced tone of the musical instrument used, compared with the original tone. This is because some of the higher harmonics will have been lost and also the decay length of each note will have been halved. This last point will not cause problems in the case of organs, wind instruments and any other instrument which does not have a decaying characteristic, but a fixed amplitude instead.

This method, otherwise known as 'double speed' technique, enables anyone to play a piece of music on an instument which before he could not play fast enough.

Tape Reversing

Many musical instruments and natural sounds have a decay characteristic such that the note or sound rises quickly to a certain level and then dies away at a slower rate. A typical decay of a note is shown in Fig . 2.2 Alonside the diagram is another diagram of the same note, but shown in reverse, that is, with the original decay first and attack last. The effect would be that in the case of say, a guitar string being plucked, the note would gradually increase to a peak, and then suddenly die away. Yet how can this be done on a tape recorder?

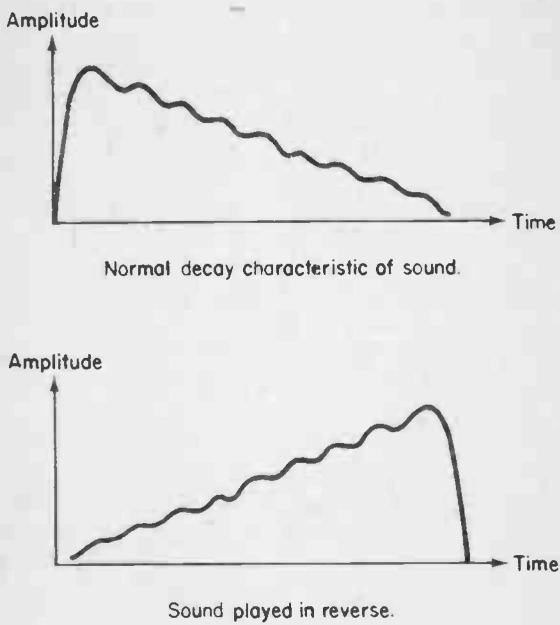


Fig. 2.2

Some open-reel tape recorders have a tape reversal switch which sends the tape in the opposite direction to which it usually travels. By recording a note on such a machine and playing back in reverse, the note will sound completely different.

However, on most machines, reversing, is not so simple. If a machine is a full track mono, or half-track stereo, then try swopping over the spools. In the case of the half-track stereo machine, a sound recorded on track 1 will play back in reverse on track 2 when the tape is reversed by changing over the spools. When splicing together as in the early part of this chapter it would not be possible to splice in any note in reverse.

However, if the tape recorder is a half-track mono or quarter track stereo, then it is not possible to use the above method, since on turning the tape over (changing spools), recording and playback is carried out on previously unused tracks. A method for reversing tape in this case has to be a complicated one of rethreading, and therefore is rather limited in its versatility. One method is shown in Fig. 2.3, although no doubt readers will be able to think of alternative methods. Note that an extra guide or two will have to be placed at various positions to assist in transporting the tape from one spool to the other. This can be any suitable object provided it has smooth surfaces where the tape makes contact, otherwise wear of the tape will occur.

Musique Concrete

This is a form of composition made up of sounds coming not from instruments or electronic sources but from sounds made by various other objects, including the natural sounds of wind, sea, thunder and rain. Connect up a microphone to a tape recorder and record as many sounds from natural sources as you like. Record things like dogs barking, birds singing, water running out of a tap, water running out of a bath, tapping milk bottles or wine glasses with spoons, doors creaking – in fact, anything. Record each one at various speeds and at various levels. Now when you get back to your workshop or studio, cut the tape into its individual sounds so that the task of rearranging can begin. This will be done according to individual taste, but all of the techniques explained so far can be used.

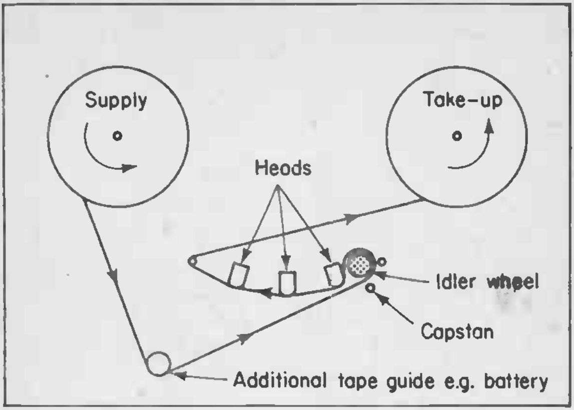


Fig. 2.3 Method of rethreading to achieve reverse replay.

Some of the more expensive tape decks are equipped with a vari-speed control which permits the tape speed to be continuously varied. Therefore, by experimenting, natural sounds can be reproduced at almost any pitch. An amusing example of this is a dog apparently barking Beethoven's Ninth Symphony!

Natural Echo and Reverberation

These should not be confused with echo and reverberation which is generated electronically, which will be dealt with later. Echo and reverberation are essentially the same treatment of sound, the difference being that echo is the repeat of a sound with a distinguishable time delay. In other words, you can hear each individual repeat of the original sound. However, reverberation is many echoes of the same sound coming quickly one after another and dying away gradually.

It is what is heard in large hard-walled buildings such as churches and cathedrals, when sound can be heard to fade away slowly. Reverberation can be introduced into your recordings by using a reverberation chamber. This can be any room which has hard walls, such as a bathroom. Make sure that all soft items of furniture have been removed else these will absorb sound thus reducing the maximum reverberation time.

The microphone or other source of sound is connected up to an amplifier and loudspeaker, which is placed at one end of the chamber. Another microphone is placed at the other end of the room and connected to the tape recorder. The reverberation can be altered by moving the microphone in the chamber. Also the reverberated signal can be mixed with the original sound to give varying degrees of reverberation. For this you will need to use some form of mixer and details are given in Chapter 6.

A general arrangement of the reverberation chamber is shown in Fig. 2.4.

Tape Loops

Record a few sounds on some tape and then cut out a length of about eighteen inches. Now join the two ends of the tape splice them together so that a loop is formed. Thread the tape loop through the recorder, and to ensure that it is kept under tension, arrange suitable guides on the deck.

On replay a repeating sequence of sounds should be heard, and thus you now have a basic rhythm to add to your tape music, although you cannot actually add it unless you use a second tape recorder or a multi-track technique, so for the moment just try experimentally with loops, long or short, at high or low speed, and see what interesting rhythm patterns you can get.

This chapter has dealt with the basic techniques of tape manipulation and the recording of natural sounds (as opposed

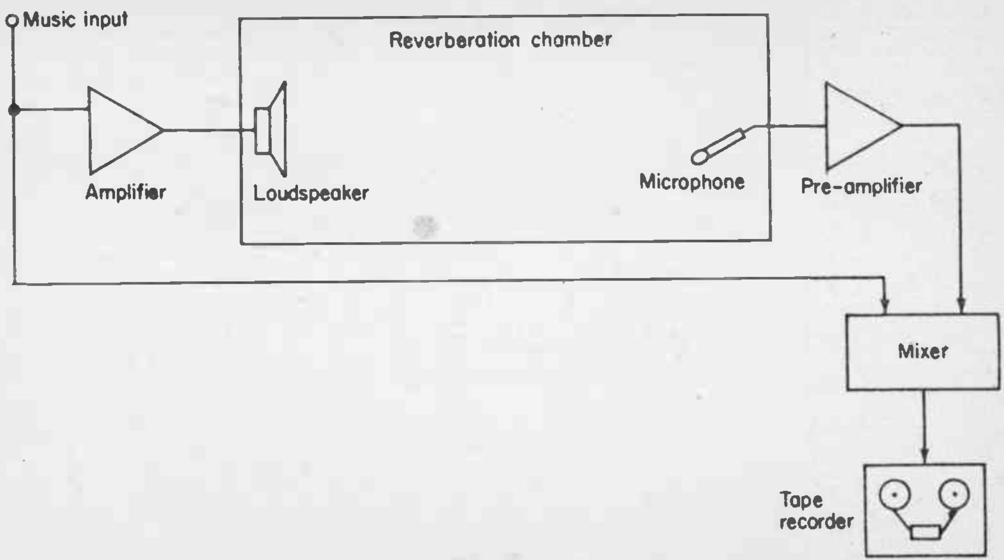


Fig. 2.4 Reverberation chamber arrangement.

to electronically generated sounds). Microphone technique is beyond the scope of this book. Now that the reader is well acquainted with these basic techniques he is ready to progress to electronically generated and processed sounds.

CHAPTER 3

ELECTRONIC MUSIC GENERATORS

This chapter shows the main circuits that go together to make an electronic music synthesiser.

Any sound, musical or otherwise is made up of complex waveforms. These waveforms, or close approximations to them, can be broken down into component parts, such that it would be easy to reconstruct the sound by using different circuits to produce each different parameter of the sound. This is the basic idea of the synthesiser.

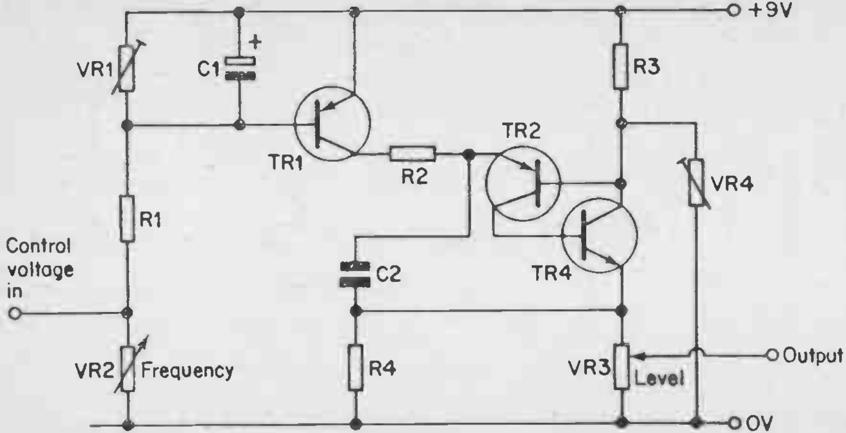
To start with, you need a note, or tone, and this has to have a certain frequency, or in musical terms, pitch. Second, you need to vary the amplitude, or volume of this note, and third, you want to vary the tonal quality, or timbre, of the note.

A basic synthesiser then, has an oscillator to generate the basic sound with a certain pitch, which is then followed by a variable gain amplifier to control the amplitude, and at some stage some sort of filter to alter the harmonic content, and thus the timbre.

Most synthesisers use at least two oscillators with associated controlled amplifiers and filters. The more duplication of circuits in a synthesiser, the more complex sounds you can get out of it.

Oscillators

The oscillator is the starting point for the generation of a sound. It is required to produce an output with a fixed harmonic content and fixed level but complete control over the frequency.



COMPONENTS

R1 220K
R2 27K
R3 470Ω
R4 12K
All ¼ ± 10% carbon

C1 2μF 12V elect.
C2 0·01μF

VR1 100K preset
VR2 1MΩ linear
VR3 50K lineor
VR4 10K preset

TR1 AC128
TR2 AC128
TR3 BC107

Fig. 3.1 V.C.O.

This control could be a variable resistor replacing a fixed resistor in a simple multivibrator circuit, but the most common type of control in synthesisers is voltage control. That is, the parameter – in this case frequency – is controlled by applying a varying voltage to the input of a voltage-controlled oscillator (VCO).

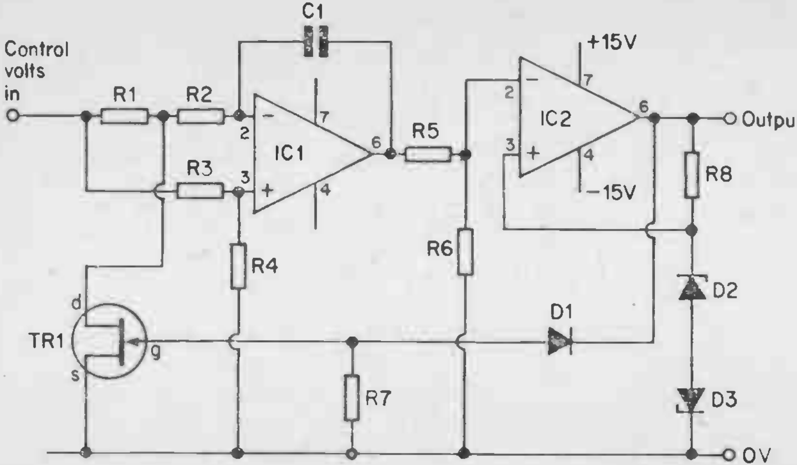
A very simple VCO is shown in Fig. 3.1 TR2 and TR3 form a relaxation oscillator whose frequency is controlled by the effective resistance between emitter and collector of TR1, which in turn is controlled by the voltage applied at the input. VR1 is a preset control which is adjusted for the lowest required frequency, and VR2 is a manual frequency control. Trimmer VR4 should initially be set at mid-track with VR2 fully clockwise (minimum resistance), then VR4 adjusted until the highest possible frequency is reached.

The output is taken from the slider of VR3 and is a square wave. The voltage applied to the input can be anything from OV to the supply rail (9V), or alternatively a variable resistance of some kind can be connected across the input.

The circuit can be constructed on a veroboard, or a printed circuit board can be designed. VR1 and VR4 are sub-miniature presets. A Venier dial can be fitted to VR2 to facilitate ease of tuning to other instruments, etc.

One disadvantage of this circuit, is that when VR2 is varied to change the frequency, it also changes the mark to space ratio of the square-wave output. This changes quite substantially the harmonic content and gives the tone a different sound for different frequencies.

A circuit which does not have this disadvantage is shown in Fig. 3.2. The output is a square wave which oscillates from +15V to – 6.3V, taken from the output of IC1, a 741. This output controls the f.e.t. switch, TR1, whose source and drain connect the inverting input of IC1 to ground. Thus when IC2 output is high, the inverting input of IC1 is shorted to



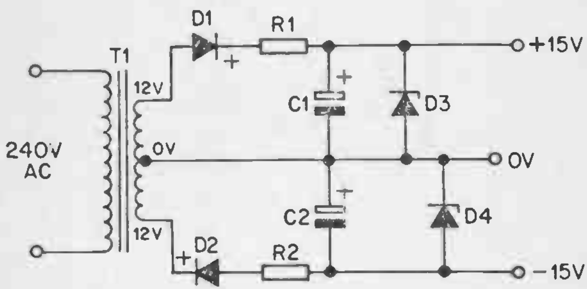
COMPONENTS

R1	5K6
R2	6K8
R3	6K8
R4	6K8
R5	10K
R6	12K
R7	6K8
R8	2K2
C1	820pF
D1	1N914
D2	BZY88 C5V6
D3	BZY88 C5V6
TR1	2N3819 f.e.t.
IC1	741 – 8 pin dil
IC2	741 – 8 pin dil

Fig. 3.2 V.C.O.

ground and IC1 gives a positive-going ramp at its output. IC2 remains high until this ramp causes the level of the inverting input to rise above 6.3V which is the level at the non-inverting input. IC2 then switches, causing the f.e.t. to become open circuit and IC1 begins on a negative-going ramp.
The frequency is linearly dependent on the input voltage, giving about 1kHz per volt.

The power supply required is ±15V dual supply, and as this is not easy to obtain from batteries, a suitable mains-powered supply is shown in Fig. 3.3. The ±15V type of supply is a common one used in synthesisers, and so this power supply can be used to drive other circuits.



COMPONENTS

R1	22Ω ½W	D1	1N4001
R2	22Ω ½W	D2	1N4001
		D3	Zener 1·3W BZX61 15V
C1	1000μF 25V elect.	D4	Zener 1·3W BZX61 15V
C2	1000μF 25V elect.		

T1 Transformer 240V primary 12−0−12 secondary at 250mA

Fig. 3.3 ±15 power supply unit.

Sources of Control Voltage

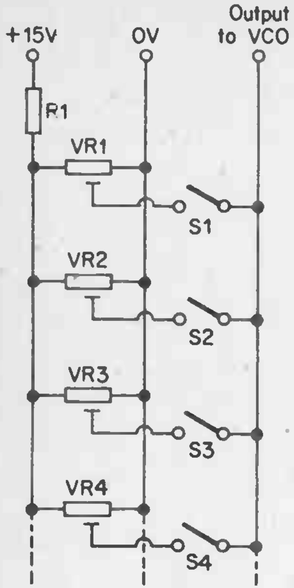
These two circuits and indeed any VC0 or other voltage-controlled circuit have to be controlled by some variable voltage source. This could be a potentiometer connected across the supply whose slider would then provide any voltage between the power rails.

Another common control voltage source is a keyboard provided with electrical contacts which provide different output voltages for each key pressed. If you wish the keyboard to be used for serious musical purposes, it must then be capable, together with its VCO, of producing an equal temperament scale. If the VCO is a so-called "logarithmic" VCO, this is easy, because the resistors in the divider chain which feeds the keys can all have the same value. However, in the VCO in Fig 3.2, the frequency was said to be linearly dependant on the control voltage and this makes things difficult. This oscillator gives increases in frequency in the same ratio as increases in control voltage are applied, and an equal temperament scale gives a logarithmic increase in frequency as you progress from each note up to the next. This is why it is easy to use a logarithmic VCO for equal temperament scales.

To generate on equal temperament scale using the second VCO therefore, you could try the circuit of Fig. 3.4. It is really several preset potentiometers connected across the supply, and each one selected by a key switch on the keyboard. Note that if you press two keys at once, the voltage they present to the VCO will be somewhere between the two separate voltages, and the same goes for the resulting frequency.

The keyboard itself can be anything from a converted dolls' piano to a full-size ready-made keyboard, or a stylus-operated using either printed circuit or veroboard edge strip as the keyboard. The choice depends on personal taste and, of course, funds.

Now that we have generated our basic sound, we will want to start playing around with it to make it more interesting. First we will look of ways of varying the amplitude or volume of the sound.



COMPONENTS

R1 4K7

VR1, 2, 3, 4...etc.
10K preset

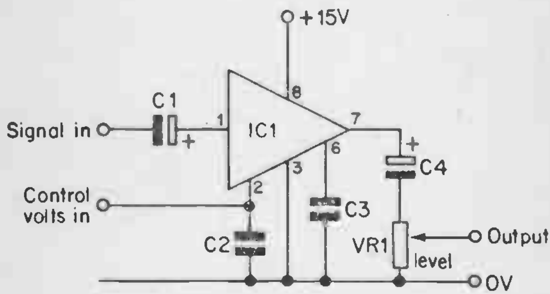
S1, 2, 3, 4...etc.
single pole switches
on keyboard

Fig. 3·4 Keyboard arrangement.

Amplitude Control

If you press the key of a piano, the sound builds up rapidly to a peak and then gradually fades away. The same happens with a guitar, a drum, and a harp, and in fact, a large number of instruments. Since we find these attack and decay parts of a sound to be pleasing, we would also wish to make sounds with these characteristics in our electronic synthesiser.

If you were quick enough, or if the attack and decay is slow enough, you could keep your hand on the level control and control the "shape" of the sound that way. But this is an electronic sysnthesiser and so we need to find a way to control the volume of the sound electronically, and this is where the VCA comes in. VCA stands for voltage-controlled amplifier and there is an integrated circuit called the MC3340P which, with a few additional components, controls the amplitude of a signal by way of a voltage applied to its control input. The circuit is shown in Fig. 3.5.



COMPONENTS

VR1 25K log.

C1 1μF tantalum
C2 0·1μF polyester
C3 680pF
C4 10μF 25V electrolytic

IC1 MC3340P electronic attenuator

Fig. 3.5 MC3340P voltage-controlled amplifier.

The control voltage is applied to pin 2 and as control voltage increases, so does the attenuation, until, with a voltage of 6V, the output level is down by 90dB. This voltage can be supplied either from a potentiometer, or even a keyboard (though not useful), or more usually from an envelope shaper.

An envelope shaper is a device which provides an output which is a simple d.c. voltage starting from ground level (OV) and increasing to a peak and then returning to OV again. The way it reaches and moves away from its peak can be controlled (attack and decay).

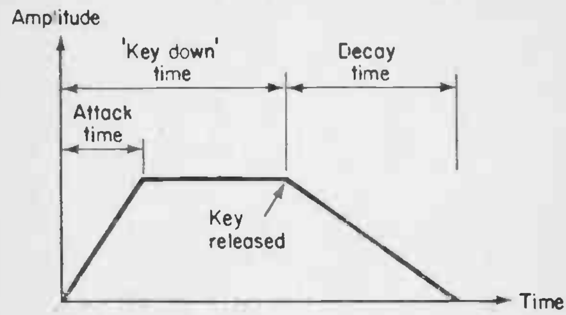
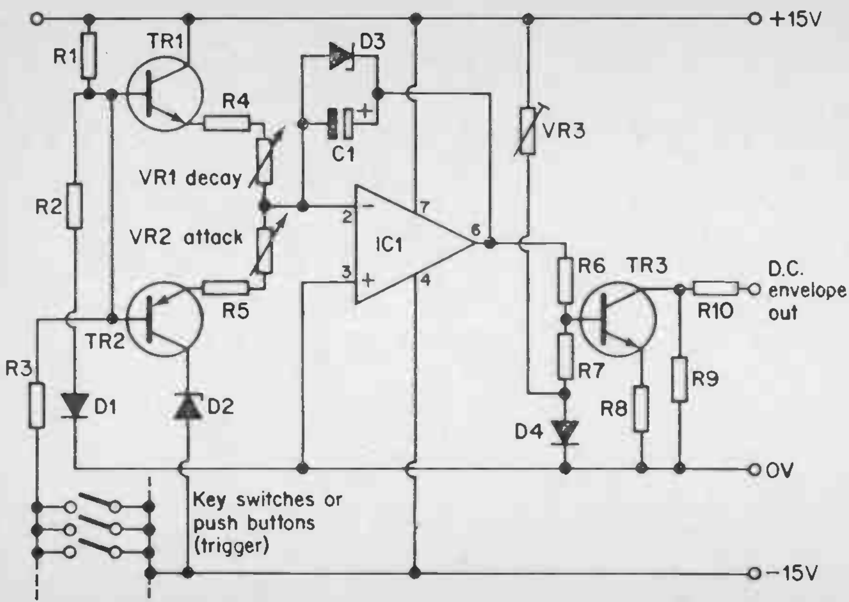


Fig. 3.6 Attack/decay output envelope.

The output waveform of a typical attack/decay envelope shaper is shown in Fig. 3.6. The two parameters which can be controlled are the attack time and decay time. An envelope shaper is usually triggered by a key switch on a keyboard or a push button. As soon as it is triggered it begins on the attack cycle until it reaches its maximum level (sustain level on the diagram). It remains here until the key is released and then immediately starts on the decay cycle. In fact the decay cycle always begins on the release of the key, whether the envelope is at the attack stage or sustain stage. An attack/decay envelope shaper is shown in Fig. 3.7. It can be powered from the ± 15V supply detailed in Fig. 3.3.

The circuit generates an output envelope which varies from a minimum of –0.5V to a peak of 4.5V, measured at the output of IC1. TR3 is a stage that matches the envelope shaper to the MC3340P VCA. The only adjustment to be made is that of VR3 and this adjustment is made so that the following conditions are satisfied.



COMPONENTS

R1	100K
R2	3K9
R3	27K
R4	1K
R5	1K
R6	10K
R7	680Ω
R8	560Ω
R9	39K
R10	3K9
C1	10μF 40V
D1	1N914
D2	Zener BZY88 12V
D3	Zener BZY88 5·1V
D4	1N914
TR1	BC184
TR2	BC213
TR3	BC184
IC1	741 – 8 pin d.i.l.
VR1	1MΩ log
VR2	250KΩ log
VR3	22KΩ

Fig. 3.7 Attack/decay envelope shaper.

1. Voltage at the anode of D4 is 0.65V
2. With TR3 off, base should be at 0.6V
3. With TR3 on, base should be at 0.8V

Another type of envelope shaper, the ADSR shaper, is more sophisticated. ADSR stands for attack-decay-sustain-release and the envelope is shown in Fig. 3.8. When the key is closed and held down the envelope commences on its attack, it reaches a peak and immediately begins decaying again, but does not drop to zero. Instead it falls to a level that has been preset by the "sustain" control and stays there until the key is released when the envelope decays to zero at a rate set by the "release" control.

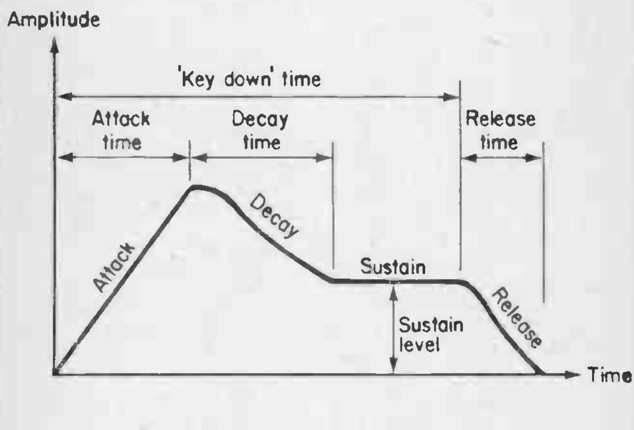


Fig. 3.8 ADSR output envelope.

Timbre Control

Most VCO's give an output which is always of the same waveform, that is, it may be always a square wave, as is the case with the two VCO's described earlier in this chapter. Some VCO's give a number of different waveforms at separate outputs like, sine, sawtooth, ramp and square. You may be

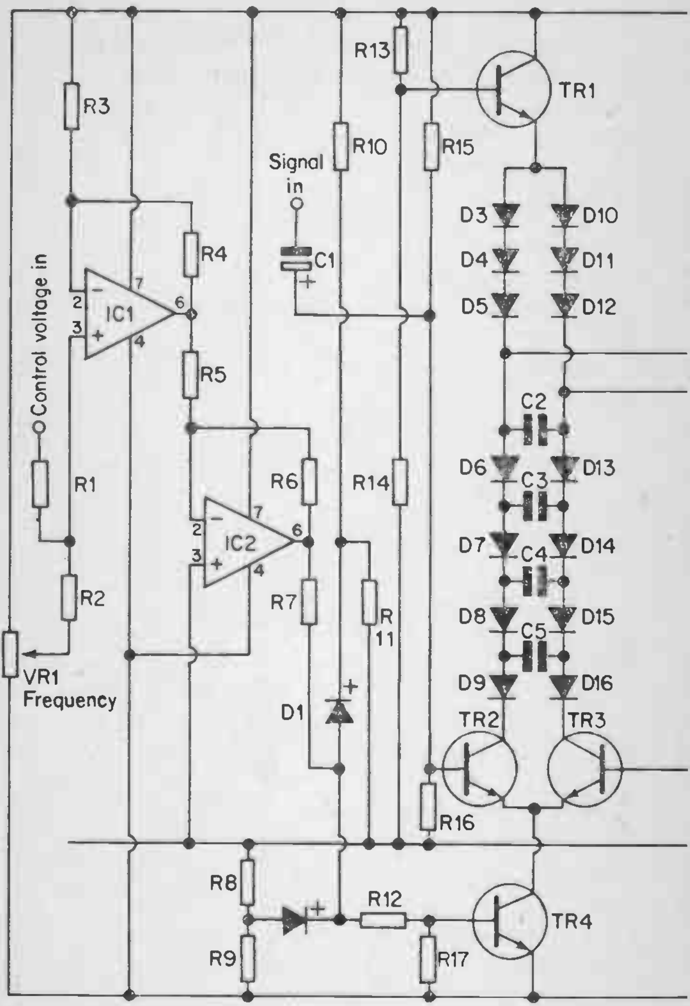
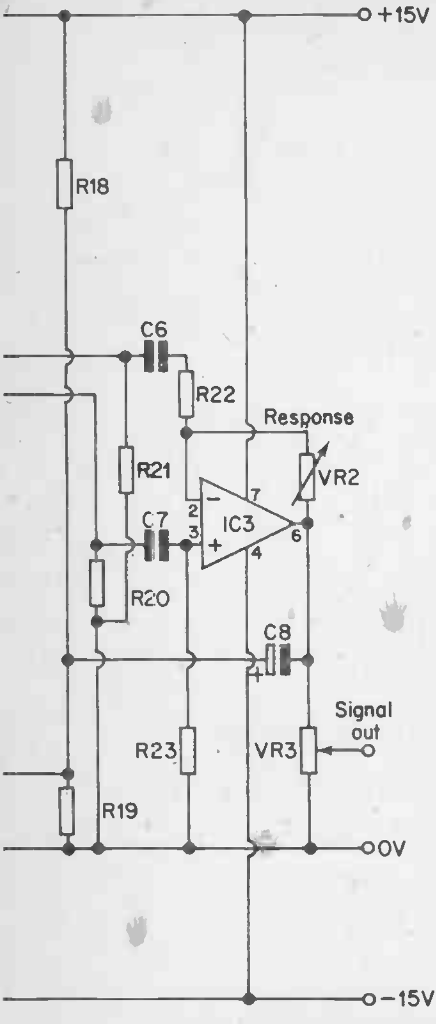


Fig. 3.9 Voltage-controlled filter



COMPONENTS

R1 3K3
R2 15K
R3 33K
R4 10K
R5 22K
R6 10K
R7 2K7
R8 2K2
R9 12K
R10 12K
R11 3K3
R12 12K
R13 22K
R14 39K
R15 12 K
R16 3K3
R17 560Ω
R18 12K
R19 3K3
R20 390K
R21 390K
R22 56K
R23 56K
All ± 10% 1/4W carbon

VR1 10K lin.
VR2 500K lin.
VR3 5K log.

C1 4·7μF 25V elect.
C2 0·01μF
C3 0·01μF
C4 0·01μF
C5 0·01μF
C6 0·22μF
C7 0·22μF
C8 4·7μF 25V elect.

TR1–TR4 BC109C

D1–D16 1N914

IC1–IC3 741 8 pin d.i.l.

wondering why we need a variety of waveforms. Well, what do you think is the difference between a flute and a trumpet playing exactly the same note, apart from the volume? The answer is the harmonic content, harmonics being those pitches at one, two, three . . . etc. octaves above the fundamental pitch.

The harmonic content is also directly indicated in the waveform. A sine wave is a pure fundamental with no harmonics, whereas a square wave is rich in harmonics.

Thus a useful circuit in any synthesiser is a filter – and wouldn't it be nice if that could be voltage-controlled as well. In fact, this is usually the case. The device is called a voltage-controlled filter or VCF.

The usual type of VCF employed is a low-pass type so that high harmonics can be cut off. The control voltage varies the cut-off frequency usually over the range 10Hz to 15kHz.

The circuit of a VCF with a passband variable over the range 100Hz to 5kHz is shown in Fig. 3.9. The part that does the work of filtering is the diode ladder, D3–D16 and C2–C5. Above the cut-off frequency C2–C5 have reactances small compared to the resistance (dynamic) of the diodes and so any signal does not make it to the take-off point at the top of the ladder. However, at low frequencies, the reverse applies, and the signal climbs straight up the ladder. The cut-off frequency is changed by varying the dynamic resistance of the diodes, which is done by varying the current through the ladder.

Once again, this circuit can be powered from the ±15V supply.

The response control varies the filtering characteristics of the circuit such that at one end the filter is a straight-forward low-pass, then a low-pass filter peaking towards the cut-off frequency, and at the other end the filter begins oscillation.

The cut-off frequency can be varied either by the manual frequency control VR1 or by the control voltage via R23. This could be devised from a keyboard such that every note on it will have the same or similar harmonic content. This is where the VCF frequency "tracks" the VCO frequency. Alternatively the control voltage could come from a low frequency oscillator to give a Wah-Wah effect. Another alternative source could be an envelope shaper where the quality of the sound could be made to change during operation. This method can produce some very interesting effects and will be dealt with in some more detail in the next chapter.

Other Types of Filter

There are various other types of filter which you may come across which can be used for sound effects and recording processes They are described in terms of what frequencies they pass and graphic descriptions of what they do are given in Fig. 3.10.

1. **High pass.** As its name suggests, frequencies above this filter's cut-off frequency are passed, whereas those below are not. Useful for eliminating low frequency noise (e.g. rumble), or, as an interesting exercise, to remove the fundamental frequency of a note and leave the higher harmonics.

2. **Band reject.** This one passes all frequencies except for a band centred around the notch frequency. The width of the band and depth of the notch depend on the efficiency of the filter. Although not very useful as an effect, it can be used to remove unwanted notes.

3. **Band pass.** Essentially the reverse of the band reject filter, this one only passes a band centred around one frequency. This is useful in sound treatment, because a collection of them with centre frequencies at 1/3 octave, or one octave apart, can be used to control the frequency content of any signal.

Commercial equipment comes in the form of a panel with slider controls side by side to control the boost or cut of each centre frequency. The equipment is known as a "graphic equaliser".

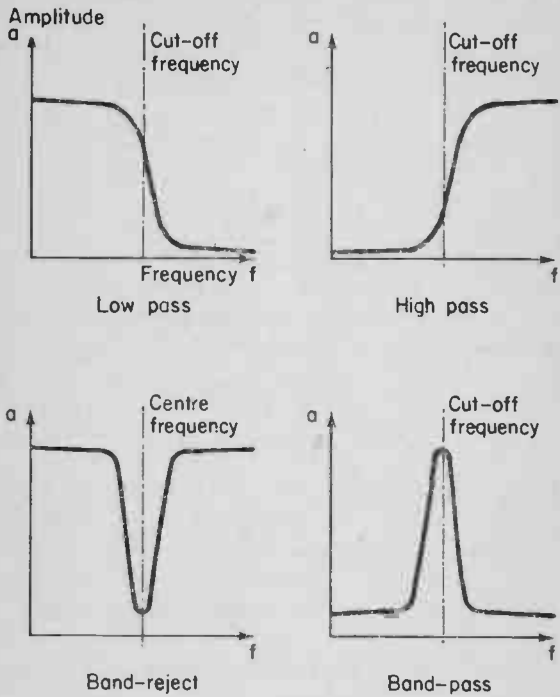


Fig. 3·10 Graphs showing functions of various filters.

CHAPTER 4

RECORDING ELECTRONIC MUSIC

In this chapter we are concerned with actually using the types of circuit described in the last chapter, and getting their resulting sounds down on tape, so as to make your first electronic music composition, as opposed to the bangs and scrapes we dealt with in Chapter 2, otherwise known as musique concrete.

To enable us to make more sophisticated compositions than those in Chapter 2, we'll need to have some less restricting recording techniques.

It is always better to use two separate recorders, because you can get better quality and lower noise levels on re-recorded material. But then not everyone (well, hardly anyone) is going to possess two good quality machines, so if you can't get together with a friend, you can use the first method described, which requires a stereo recorder, preferably half-track, so that you can use the reverse play facility.

What we are trying to achieve here is a scaled-down version of the now widely used technique in professional studios–multi-tracking. In the big studios they have big tape machines with up to 24 tracks which can be recorded or replayed independently. The tape itself has to be much wider than ¼ inch – 2 inches for 24 tracks. Separate instruments and voices are recorded on each of the tracks, each of which can then be modified at will – and also each track can be recorded at a different time, thus avoiding the necessity for having all the artists in the studio at the same time. When the recording engineer is satisfied that he has all the required material he mixes all the tracks together, giving each an appropriate level relative to the others, and also placing each in a certain part of the stereo field. The final stereo tape is the master tape, which is sent along to the disc cutter where the record production process begins.

To adapt this technique to the amateur's situation, where only 2 tracks are available per machine, means that we will have to mix each new recording with the previous recordings, restrict the number of re-recordings therefore to about 3 or 4, to reduce noise build up, and we will probably have to stick to recording in mono.

First, then, a method which only requires one stereo recorder.

Now the first track can be called track 1. and the second track 2. This may seem an obvious statement, and some people may be wondering why they can't be called Left and Right, since it is a stereo machine. Well, we are not going to record in stereo, but use the two channels as separate mono.

If the machine has three heads, and you record something on track 2 which is supposed to be in time with what you are playing back on Track 1. then when you replay both simultaneously, the signal on track 2 will be slightly ahead of that on track 1. This is because of the physical distance which separates the record and replay leads on 3-head machines. This is a point to watch as it can mess up your timing, and the composition will not sound right. You could try fixing up a connection so that you can get a replay signal from the record head. Note that on two head machines this problem does not arise.

Now to do some actual recording. Record a signal on track 1. This can be anything for the moment – a microphone or an electronic source – it's up to you.

The next thing to do is to play this back, and mix the output with a new signal, and then the combination of the two signals is recorded on to track 2.

Now I can hear some of you saying – "What do you mean by mixing?" Well, it is simply what it says, that is, two (or more) signals are added together in a combination you decide. To do this you need a mixer, and for the moment a simple passive one will do the trick. A diagram is shown in Fig. 4.1, together with the set-up required for the recording process just described.

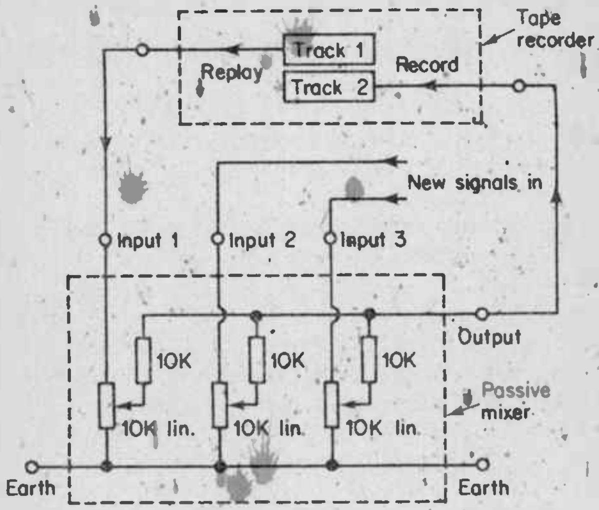


Fig. 4.1 Passive mixer with set-up for recording from track 1 to track 2.

On track 2 you now have a combination of two signals. One is freshly recorded, the other has been recorded twice. This latter recorded signal is called a second generation recording and will have a higher noise level than a first generation recording (freshly recorded signal). Thus we do not want to re-record the first signal too many times, and so we limit the recordings to fourth generation. In other words only four independent signals can be put onto tape, that is, if you only add one at a time. It would be possible to add two or more at one go, and an extra input has been provided in the mixer.

So, to sum up, here is how you put four signals on tape.

(1) Record signal 1 on track 1

(2) Mix track 1 with signal 2 and record on track 2

(3) Mix track 2 (signals 1 + 2) with signal 3 and record on track 1

(4) Mix track 1 (signals 1 + 2 + 3) with signal 4 and record on track 2.

The final composition – signals 1 + 2 + 3 + 4 – can be replayed on track 2.

As this might lead to some confusion over which signals are on which tracks, see Fig. 4.2 for a diagrammatic explanation of the process.

Remember that whenever you record in a new signal, whether mixed in with others or not, if you are not satisfied with it, rewind the tape and try again. A track from which you are taking signals already recorded will not be erased if you are just replaying it in order to mix it with new signals.

So now you have one way of doing some "multi-tracking". With some stereo recorders it may not be possible if the play/record switch is the same for both channels, so that you could not have one track recording while the other track is playing back. A much better system would be to have the two tracks on separate tape recorders. Then the overall quality would be better and you would have more control over levels, tone etc. This leads to the second method, one which uses two separate mono recorders.

This time, instead of recording from one track to another on the same machine, you record from one machine to another.

To start with, signal 1 is recorded on machine 1, which is then replayed, mixed with signal 2, and recorded on machine 2. This is then replayed, mixed with signal 3 and recorded on machine 1 again – and so on until the composition is com-

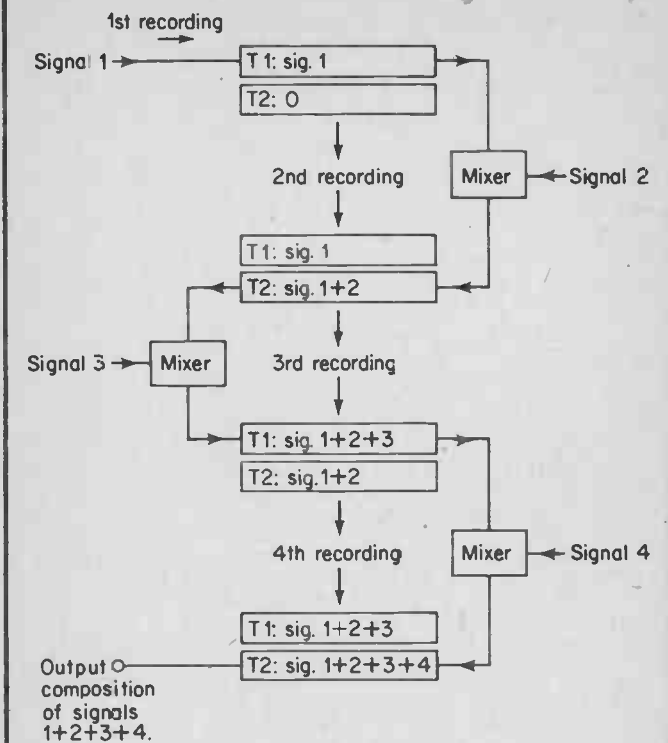


Fig. 4.2 Diagramatic representation of track to track recording.

plete. A diagram, Fig. 4.3, shows how this technique works. Once again you have to restrict the number of re-recordings. You can find out by experiment how many re-recordings you can get away with before the noise level becomes noticeable.

Another point that will have to be watched for is one concerning tape speed. No two machines have exactly the same capstan speed, so a tape recorded on one machine may have slightly different pitch when played back on another. So, when using this multi-track technique, or any other using two tape machines, make sure that all tape stays with one machine, that is, tape recorded on a certain machine should always be replayed on that machine.

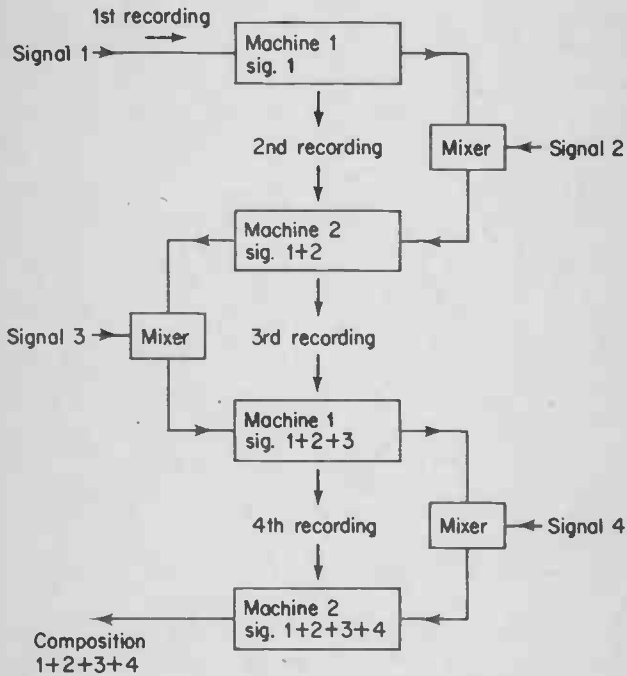


Fig. 4.3 Technique for recording from mono machine to mono machine.

The third method uses two stereo machines and it is possible to make stereo recordings by using the method described for mono machines. A stereo signal is recorded onto one machine using both channels to achieve stereo. Then this is played back and mixed with a new stereo signal which is then recorded on to the second machine, and as you can see the process continues as before up to fourth or fifth generation recordings.

Note however that you would need a stereo mixer – two of the mixers in Fig. 4.1 are adequate – so that each stereo channel can be mixed separately and at the same time.

A more effective way of using two stereo recorders, if you are not too fussy about having stereo compositions, is as follows:

(1) Record signal 1 on to track 1 of machine 1.

(2) Record signal 2 on to track 2 of machine 1, while monitoring signal 1 through recording head (not through replay head if a 3-head machine for reason stated before).

(3) Play back signals 1 + 2, mix to taste and record on track 1 of machine 2.

(4) Record signal 3 on to track 2 of machine 2 while monitoring track 1.

(5) Play back, mix and record, continuing like this until you achieve your desired result.

You may find you can go further than 4th or 5th generation, but experiment first.

The great advantage of this method is that you can take a lot more trouble over the mixing, and of course you will not be playing an instrument while the mix is being made, because both signals are on tape, whereas before only one signal was on tape, while the other came from a source you were trying to operate.

Now that you have some effective methods of multi-tracking, you can now start experimenting with electronic sources of music (or acoustic, or a mixture of both).

If you want to maintain some sort of musical structure to your composition, then it would be as well to make the first signal that you record some sort of rhythm track, so that subsequent recordings will have "beat". This can be any kind of rhythm you care to choose. It could be just a simple click or drum beat (tin cans and cardboard boxes can sound as good as drums with careful positioning of the microphone), or it could be a complex electronically-generated rhythm structure, which we will deal with in Chapter 7.

Once you have got your rhythm down on tape, you can start adding pieces generated by your various tone generators, VCO's, VCF's, envelope shapers and so on. It is best, before you actually record something, to experiment with these circuits, and connect them up in various ways with various control settings, just to see how many different sounds you can obtain.

Here then, are a few ideas and hints on how to use the circuits.

Effects from VCOs only

If the VCO has different waveform outputs, try mixing some or all of them together. This will give a wide variety of sounds with different qualities.

For instance, brassy type sounds are rich in harmonics and a large proportion of square wave can be used, whereas woodwind instruments can be imitated using a large proportion of pure sine wave with a little triangular or square wave to give it a slightly reedy quality.

You must remember that a VCO or similar tone generator is monophonic, i.e. it can only play one pitch at a time. Thus, if you have two or more VCOs you can do far more than just one. For a start, chords can be made by carefully tuning each VCO in the set-up to give each separate note in the chord – a four-note chord would need your four VCO's.

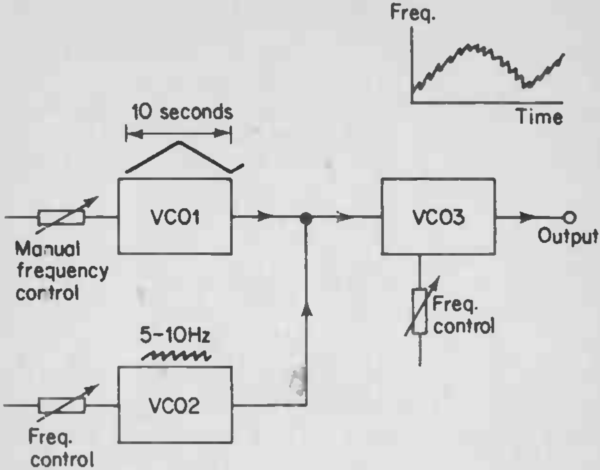


Fig. 4.4 Intermodulating VCOs.

Also with two or more VCOs it is possible for one VCO to modulate another. Consider the situation shown in Fig. 4.4. VCO1 is generating a very slow output at 1 cycle every 10 seconds, and VCO2 is generating a faster output at somewhere between 5 and 10Hz. These two outputs are added together and fed to the control input of VCO3. The manual control of VCO3 should be adjusted to give the required frequency range. The resultant sound will be a note with vibrato and also steadily rising and then falling and rising again. Vibrato is the effect caused by VCO2 and is the variation of frequency about a centre frequency, and the output from VCO2 should be adjusted so that this variation is not too large. Once you have got a varying note coming from VCO3, you can start varying the manual controls of each VCO to see what sort of effects you can get. As soon as you get something you like, note all control settings and/or get it recorded on tape.

Effects from Envelope Shapers and VCAs

The normal method of use of these is to put them after the VCOs so that their output is controlled in level only. In this mode, though, it is possible to imitate many instruments by adjusting the attack and decay controls, and on ADSR shapers, the sustain and release controls as well. A guitar sound, for example, would have a fast attack rate and slow decay rate.

Another way of using an envelope shaper is to let the envelope itself become the control voltage of a VCO, so that the frequency rises and falls at rates determined by the setting of the attack and decay controls.

If you take the set-up of Fig 4.4, and put in an envelope shaper with VCA after VC03, a further interesting effect can be obtained by taking the output of the envelope shaper (note that this is the d.c. envelope only) and connecting it to the

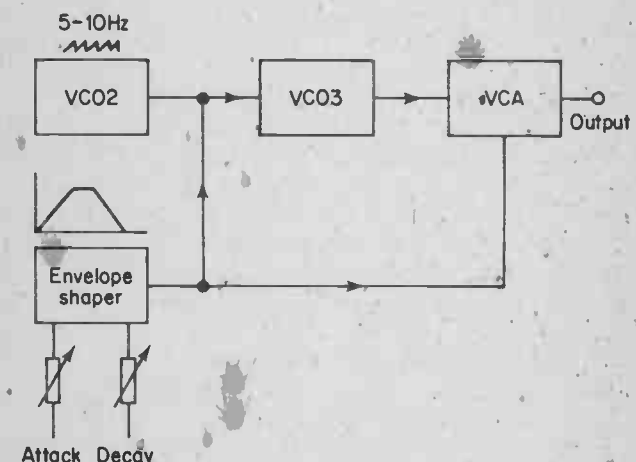


Fig. 4.5 Arrangement using intermodulating VCOs and envelope shaper.

input of VCO3 instead of VCO1's output. Fig 4.5 shows how this is done.

When the trigger is applied to the envelope shaper, quite a complex sound results varying not only in frequency, but in amplitude as well.

Effects from VCFs (low-pass)

The control voltage applied to the VCF controls the cut-off frequency, above which all frequencies are rejected. This is often used to remove some of the higher harmonics and thus changes the timbre of the sound. The cut-off frequency can be kept constant, but the most useful method of use is to let the cut-off frequency track the frequency of the signal at the input. This is done by the arrangement in Fig. 4.6, where the control voltage which is fed to the VCO is also taken to control the VCF. Thus, each note will have approximately the same harmonic content.

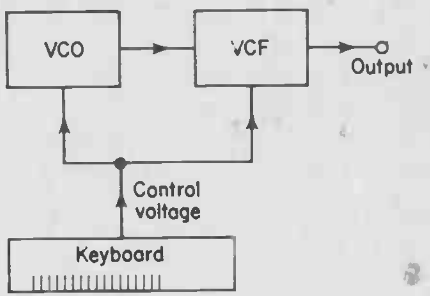


Fig. 4.6 VCF tracking VCO.

Another interesting way of using a VCF is to vary the pass-band during the length of the sound. This achieves an effect sometimes known as Wah-Wah, because of the characteristic sound it gives. Use the arrangement of Fig. 4.7. for this effect. With some experimenting with the attack and decay controls of the envelope shaper and also the level of the envelope, you should be able to get the synthesiser to "talk", or at least you will get some amusing sounds. If you have ever listened to synthesised music by Isao Tomita or Walter Carlos you will know what this "talking synthesiser" sounds like. However, I'm afraid you are not likely to get these sounds just yet, because, as with most of the really complex synthesised sounds, you need to duplicate circuits, so that each can work on a certain part of the sound – in terms of frequency and amplitude.

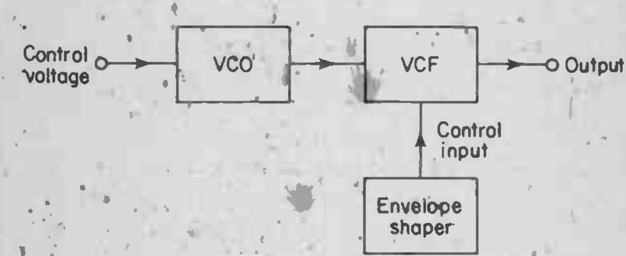


Fig. 4.7 Circuit for Wah-Wah effect.

One of the advantages of a synthesiser as a musical instrument is that you can play it as you build. You could start off by building one VCO, – you would be able to use it straight away. Then perhaps an envelope shaper and VCA, then a VCF, then another VCO, and so on, adding more circuits as the pocket allows. At all stages in the building – which need never stop – it is possible to play the instrument.

Composing Electronic Music

So far we have seen all of the mechanics and technical methods of producing electronic music. It would be ridiculous now to set out a list of exactly what you must do to produce a piece of electronic music. What you eventually produce will be a combination of what you thought of doing at the start plus some parts that you discovered while experimenting. A very few people will have a clear idea of what they want, and most people will just start experimenting until they find a sound that they like, record it, and carry on experimenting to find the next sound. Undoubtedly you will get half way through, decide that what you have recorded so far doesn't sound right and so you erase some or all of it and start again. Try not to be too impatient – your first efforts may sound somewhat amateurish – but keep at it and you will soon find the best way of putting a piece together.

At best then, I can give a few hints as to how to go about producing/composing a piece of electronic music. These hints are really for those who do not know quite what they want when they start.

First and foremost, put some sort of rhythm track down on tape. This has been mentioned earlier in the chapter, but it is important because it gives you something to tie all of your other sounds in with. It does not have to be a predominant sound, it can just be kept in the background if you like.

Second, start experimenting with VCOs to get the right sort of frequency range and tone that you would like for the first sound. If you want to shape the sound, or imitate another instrument, plug in an envelope shaper and VCA and experiment with that. If you still have not got what you want, put a VCF on the end. With all these circuits, you should be able to find some sort of sound to record. When you have found it, record it using one of the techniques described so that the rhythm track is maintained at some level. Then look around for your next sound. As you go on, you will need to make sure that each new sound fits in with the others already recorded. Don't be tempted at first to keep adding a vast number of sounds – just experiment with 3 or 4 playing at the same time.

If you are the type of person who would like to plan out your composition before you start you may be wondering how you can write electronic music. Most people are aware of the conventional method for writing music, and many find it difficult to read. Conventional written music is too precise for electronic music. It has only a limited set of fixed pitches, it can give no indication of tonal quality and in general is too restricting.

There is no set way of writing electronic music, but a common method is to use different symbols, shapes, colours for various types of sound. Frequency is only indicated generally by dividing the music into 3 frequency bands, written one above the other. Perhaps, when talking about written electronic music, we should say that it is "drawn". In some cases, if the music does not sound right, at least the script looks decorative! A sample of electronic music script is shown in Fig. 4.8.

The script is divided horizontally into 3 arbitrary frequency bands, and vertically into time divisions, although these are by no means to be strictly adhered to, but are for guidance only.

Section AB

In each frequency band there is a fast improvisation, the black dots represent the notes, which will sound as short stabs. These notes have no set pitch and this is left up to the performer.

Section BC

In the high and low frequency bands, the thick horizontal lines represent definite pitches, although which actual pitches they are is not given, but each note will last for a reasonable length of time, say 1–2 seconds. If the line is tapered, the note decays away. The large and small rings in the centre band represent a Wah-Wah sound, with a single "Wah", and the length of the ring governing the actual length of the sound.

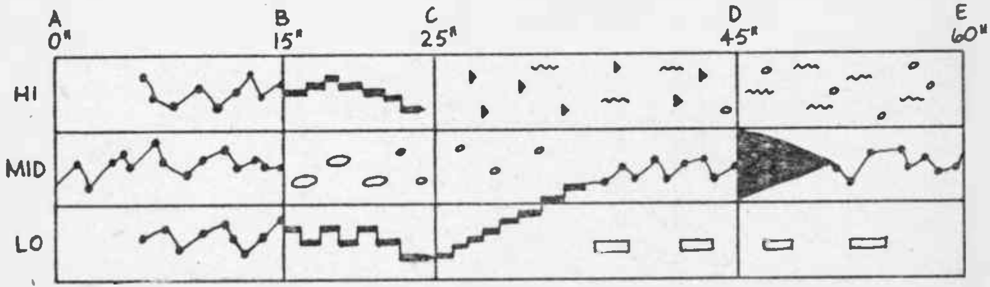


Fig. 4.8 Example of "written" electronic music.

Section CD

In the lower band, the notes step up their pitch into the middle band where they break into a fast improvisation. Later on in the low band are open squares. These also represent a "Wah" sound, except that instead of gradually varying the harmonics that create the effect, they are suddenly changed at the start and suddenly changed back at the end of the sound. In practice this would be achieved by manually controlling the frequency of the VCF instead of letting it be controlled by an envelope shaper.

In the top band there are two types of sounds. The wavy line is simply a VCO being modulated by another in order to produce a vibrato sound. The small black triangular symbols represent filtered blips of white noise.

Section DE

The only new symbol is the large black triangle in the middle band, and even then this is only an enlarged version of the small black blips in the previous section. It is a loud section of white noise that has been passed through a filter and an envelope shaper.

CHAPTER 5

MIXING AND MIXERS

The previous chapter illustrated the need for a piece of equipment known as a mixer and also gave a simple design for a passive mixer. It was passive because it included no stages of amplification, and the process was subtractive rather than additive. All professional mixers are of the active type, where each signal is amplified before and after it is mixed with other signals, such that the overall gain of the instrument is unity. Thus the output signal is at the same level as each of the inputs.

The basic task of a mixer is to take a number of inputs and mix them together into a final output signal. Each of the inputs can have a particular level in the final mix, and in stereo mixers, a particular position in the stereo field.

In addition to this basic task, a modern professional mixer has many other functions including tone control, meter monitoring, headphone monitoring and facilities that allow the musicians to hear certain instruments only, known as talkback.

The layout for a typical studio mixer is shown in Fig. 5.1.

There are several inputs which are all identical, although some could be specially for microphones or other low level source. Each input signal is amplified, then passes through tone control stages, and the signal is then fed to the channel fader. This is the control which sets the level at which the particular input is heard in the final mix. After the channel fader comes the pan pot which puts the signal into a particular position within the stereo field. This it does by sending the signal into the slider of the pot, the two ends of which are connected to two busbars, one for the left channel and one for the right channel. These busbars are connected to all of the input channels in the same way. On the output side of the mixer, two output

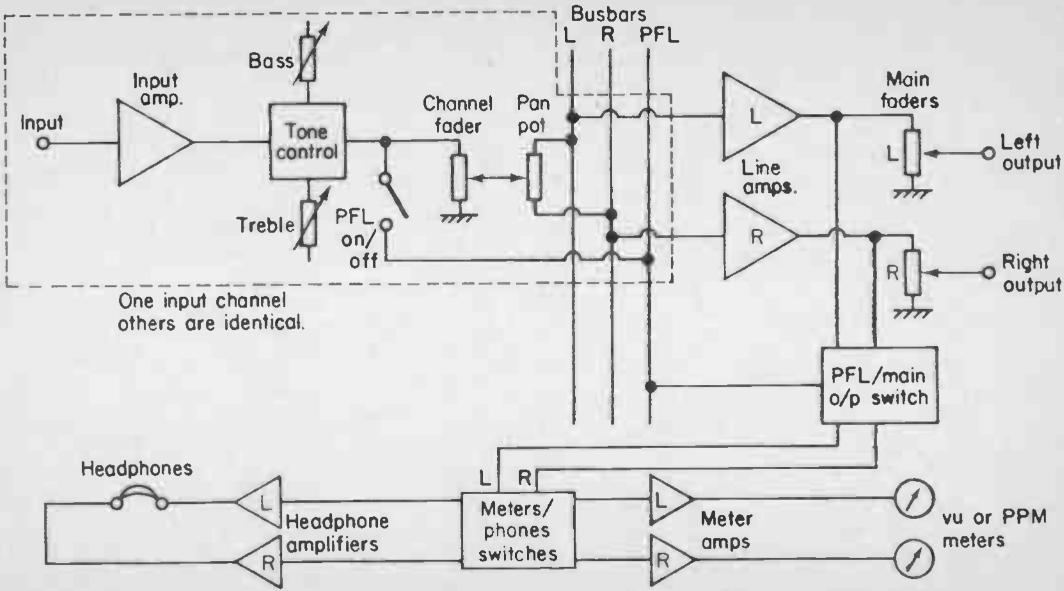


Fig. 5.1 Typical studio mixer layout (simplified).

amplifiers take the mix from the busbars and, via main faders, give the final stereo output.

Apart from the two main busbars, there are often others that are required to perform ancillary functions. One would be the PFL, or pre-fade listen. This allows you to listen to any input whether it has been faded into the main mix or not, so that it is possible to adjust tone controls, or check for overloading and distortion, before the signal is faded into the mix.

The PFL signal is taken from just before the channel fader as indicated in the diagram. The PFL busbar can be connected to either a meter or headphones or both, whichever is most useful for monitoring purposes.

Below is a list of the various functions to be found on a mixer.

Channel Fader

Usually takes the form of a slider control, and sets the level of the respective input signal in the final mix.

Preset Gain

When used in conjunction with the channel fader it sets the level of an input signal while the channel fader is at zero. This level can be checked via PFL. When the input is then required in the main mix, the channel fader is fully opened and the signal enters the mix at the preset level. This is an advantage since you don't have to vary the level of a signal when it is in the main mix in order to find an appropriate level for it. Instead, this level is found before the signal is mixed using the preset gain control.

Panoramic Pots (Pan pots)

There is usually one for each input channel, and its job is to position its respective input in the stereo field. It is normally positioned after the channel fader so that it is not possible to set this control while using PFL. (Remember that the PFL is extracted before the channel fader). Full rotation of the control either way will put the signal into the Left or Right busbars, while setting the control at halfway will send equal amounts of the signal to each busbar.

Group Select Switching

In the mixer described in Fig. 5.1. there is only one output group of two stereo channels. It is possible, however, to have more than one output group of stereo signals. A switch or set of switches is provided after the pan pots on each input channel to select the group into which that input will feed its signal, which by now is a stereo one. For every output group, there will be one pair (Left and Right) of busbars. Each group is then amplified separately and finally the groups are mixed together via the Group Faders.

Equalisation

Each input channel will be provided with at least a bass and a treble control, and possibly a mid-range control, each control boosting or cutting its particular frequency band by ± 15dB typically. In addition to these controls, there may be switched filters that can roll off either high or low frequencies.

PFL (Pre-fade listen)

On each input channel a switch is provided to allow the signal to be connected to the PFL busbar. At the output section there are switches that direct the PFL signal to either meters or headphones. Sometimes the PFL signal can be routed to a performer's headphones, and then the facility is known as "Talkback".

Foldback

This facility involves another pair of busbars. Any input module can have its output switched into these busbars, and in some mixers its level can be adjusted separately from the channel fader. The output from the foldback busbars is amplified and taken to a socket. This can then be connected to another input channel. Thus a whole block of signals can be processed as one in the main mix.

Monitoring

Monitoring is provided in the form of meters or headphones. A few types of metering are available such as VU (volume unit) and PPM (peak programme meter). The monitoring circuits can be switched either to the PFL busbar or to the main mix output.

Main Output

When all input channels have been mixed, all groups have been mixed, a single stereo signal is fed via main faders to the main output. This section may also include equalisation controls which would operate over all of the inputs. As stated before, the gain of the whole mixer should be unity, and so this output should have the same level as the input signals.

A Basic Mixer Design

The following mixer design includes up to 8 input channels, stereo output, panning, PFL, and preset gain on all inputs, meter and headphone monitoring, and tone control on the main output, although the same circuit could be fitted onto each of the input channels. A block diagram is shown in Fig.5.2.

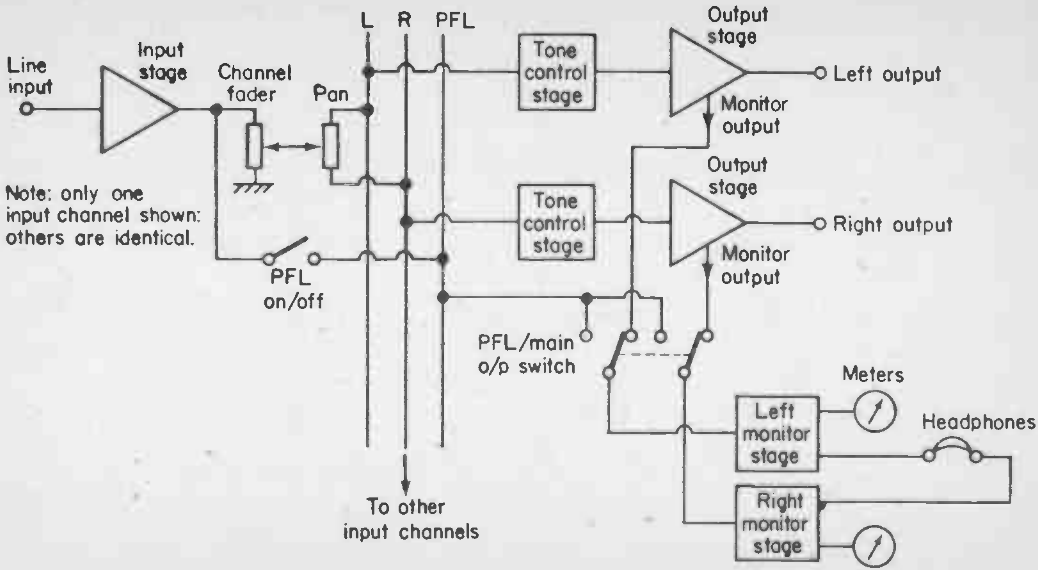
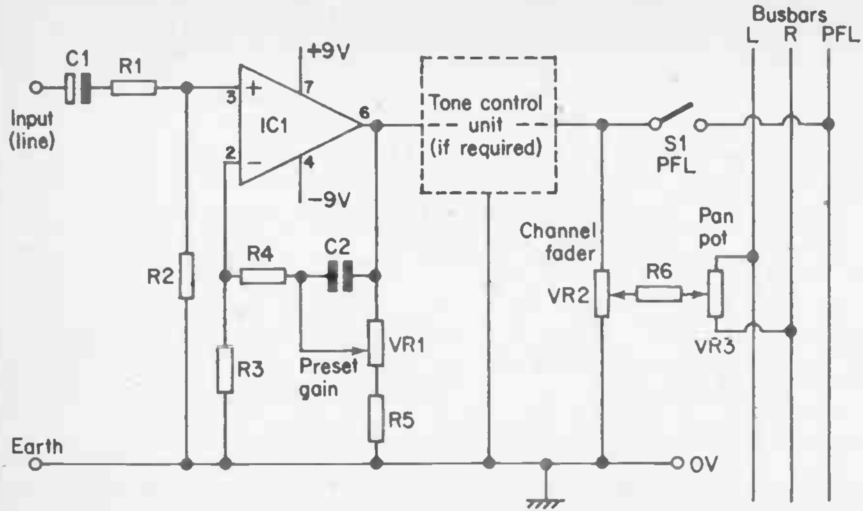


Fig. 5.2 Layout of basic mixer design.



COMPONENTS

R1	100K
R2	100K
R3	100K
R4	100K
R5	100Ω
R6	100K
All 5% ¼W carbon	
C1	1µF 35V tantalum
C2	820pF ceramic
VR1	10K log. carbon
VR2	10K log. slider
VR3	10K lin.
IC1	741 – 8 pin d.i.l.
S1	single pole on/off

Fig. 5.3 Circuit diagram of one input channel.

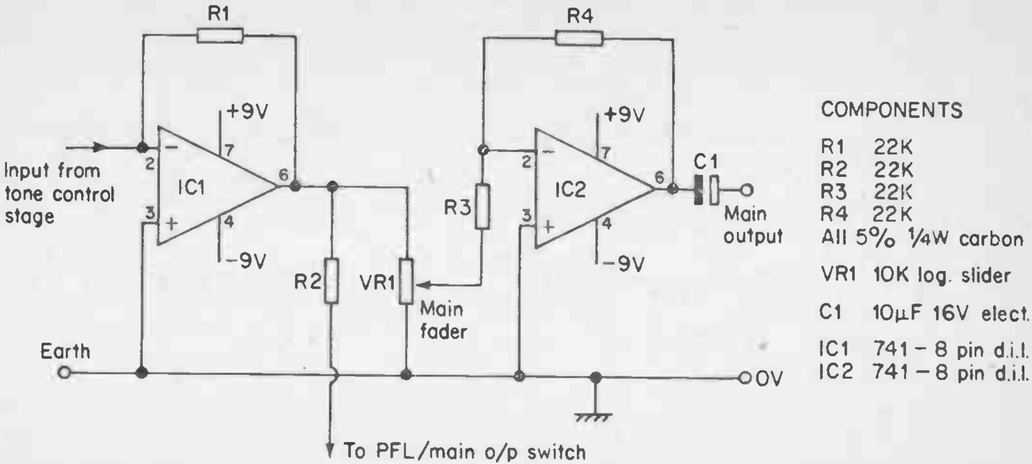
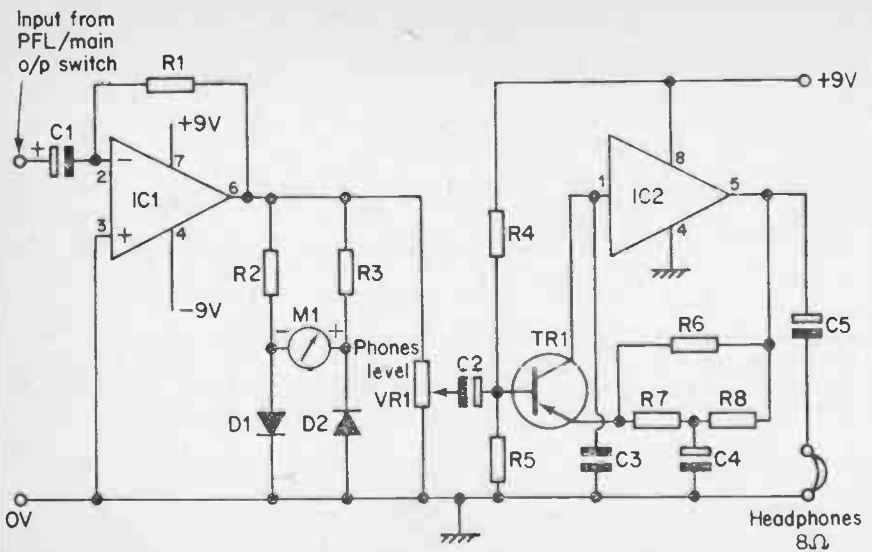


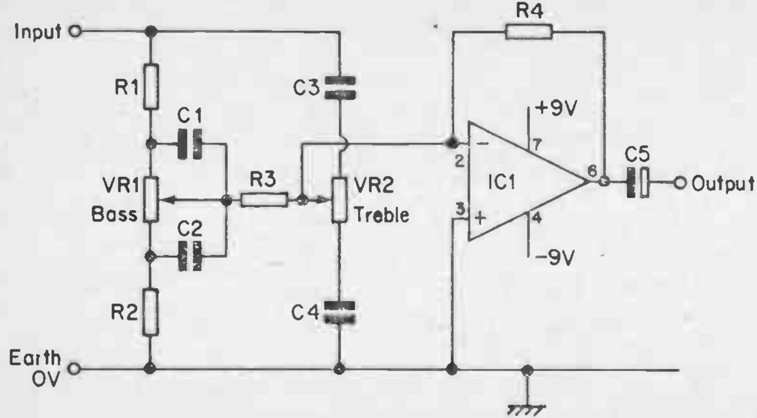
Fig. 5.4 Circuit diagram of output stage (2 required).



COMPONENTS

R1	100K
R2	1K5
R3	1K5
R4	56K
R5	33K
R6	10K
R7	18Ω
R8	1K
VR1	25K log.
C1	1μF 63V elect.
C2	4·7μF 63V elect.
C3	0·0047μF polycarb.
C4	100μF 25V elect.
C5	22μF 25V elect.
IC1	741 – 8 pin d.i.l.
IC2	MC3360P ¼W amp.
D1	OA90
D2	OA90
TR1	2N3905
M1	Vu meter

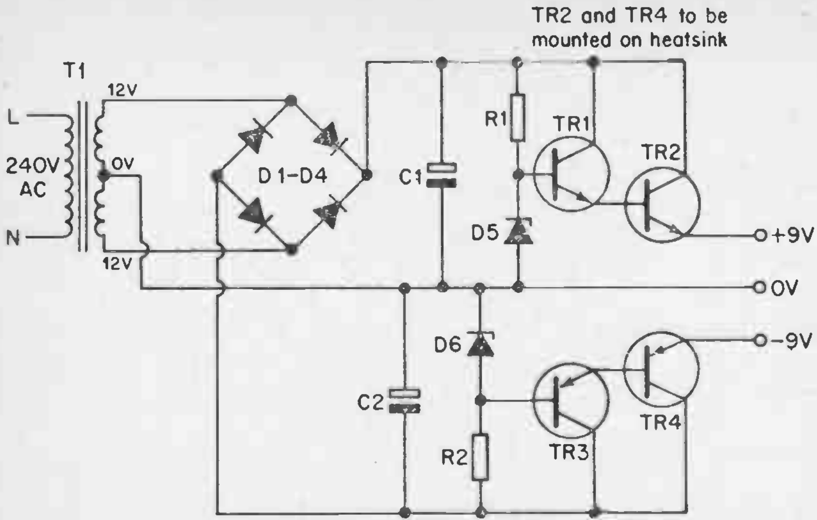
Fig. 5.5 Circuit diagram for monitoring stage (2 required).



COMPONENTS

R1	22K
R2	1K
R3	5K6
R4	100K
VR1	100K log.
VR2	100K log.
C1	0·015μF polyester
C2	0·15μF polyester
C3	1000pF polystyrene
C4	0·01μF polyester
C5	10μF 16V electrolytic
IC1	741 – 8 pin d.i.l.

Fig. 5.6 Circuit diagram for tone control unit.



COMPONENTS

R1 1K8
R2 1K8
10% 1/4W carbon

C1 1000µF 25V elect.
C2 1000µF 25V elect.

D1–D4 bridge rectifier W01
D5 BZY88 10V zener
D6 BZY88 10V zener

TR1 BC107
TR2 BD131
TR3 BC213L
TR4 BD132

T1 12–0–12V at 6VA. miniature type.

Fig. 5.7 Power supply for mixer.

The various line amplifiers, mixer and amplifiers all use a 741 operational amplifier, thus keeping costs down. A complete input channel is shown in Fig. 5.3, including the preset gain control, channel fader and pan pot. For connection to busbars, refer to Fig. 5.2, the block diagram. You can have as many input channels as you require up to 8. Each has exactly the same circuit and is connected to the busbars in exactly the same way.

The circuit for one output stage is shown in Fig. 5.4. Two are required, one for the left output and one for the right. A master fader is included in this stage. If desired, the faders from each channel can be combined into one dual potentiometer. If left separate, however, it is possible to use the two channels separately, which may be an advantage in some cases.

The metering and headphone circuits are shown in Fig. 5.5. Both can be switched either to the PFL busbar or to the main output. Once again, there are two headphone amplifiers and two meters for each stereo channel. There is, however, only one PFL busbar, thus when the monitoring circuits are connected to this busbar, both channels are fed with the PFL signal so that both sides of the headphones get the same signal, avoiding the effect when only one earpiece is connected. When the monitoring circuits are connected to the main output, each channel receives the appropriate left or right signal, giving the full stereo sound.

Fig. 5.6 gives the circuit diagram for the tone control module. One of these is included in each output channel. In addition, and if desired, one tone control module can be included in each input channel, so that each signal can be equalised separately before being mixed. The mixer layout diagram (Fig.5.2) shows where the module is fitted in.

The final circuit necessary to ensure satisfactory working is of course the power supply. This is a stabilised supply providing dual voltage rails of ± 9V. The circuit is shown in Fig. 5.7.

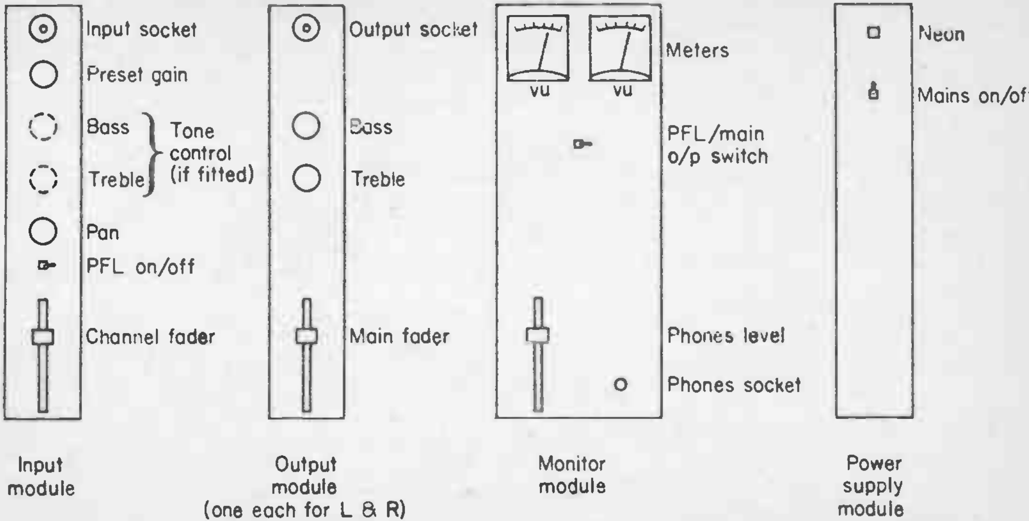


Fig. 5.8 Front panel arrangements for mixer modules.

Construction of the instrument should be done according to a modular system. Thus each input channel is constructed in a separate module, which includes a front panel. This panel takes the form of that shown in Fig. 5.8. The panel supports below it the boards which hold the circuitry for that channel only. Also shown in the diagram is a similar module for each of the other circuits – the output stages, and the monitoring circuits together with meters. The p.s.u. can be built onto a front panel of its own, with just a mains switch and neon on the panel. The modules, all of which are the same length, can then be fitted into a box such as that shown in Fig. 5.9, where the ends of each module are screwed to flanges which run along the front and back of the cabinet.

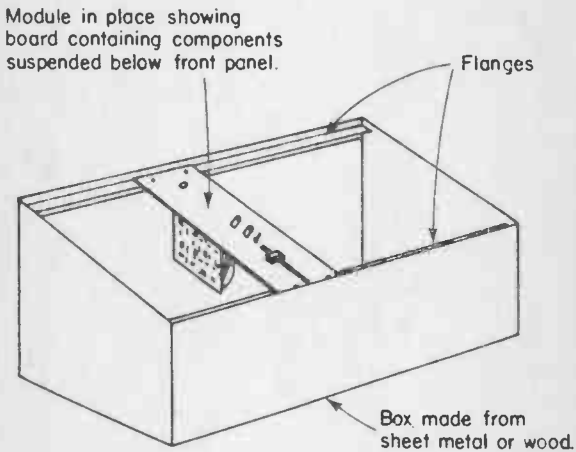


Fig. 5.9 A suitable box for the mixer.

The advantage of this method of construction is that you can build up the mixer gradually, and not have to wait until it is completely finished before you can use it. Also, the mixer can easily be extended at any time, just by putting in more modules.

Note that the modules are linked together by busbars which should be run along the base, front or back of the box. The wires from each module are terminated in plugs which connect with these busbars, thus enabling any module to be easily removed for servicing and repair work.

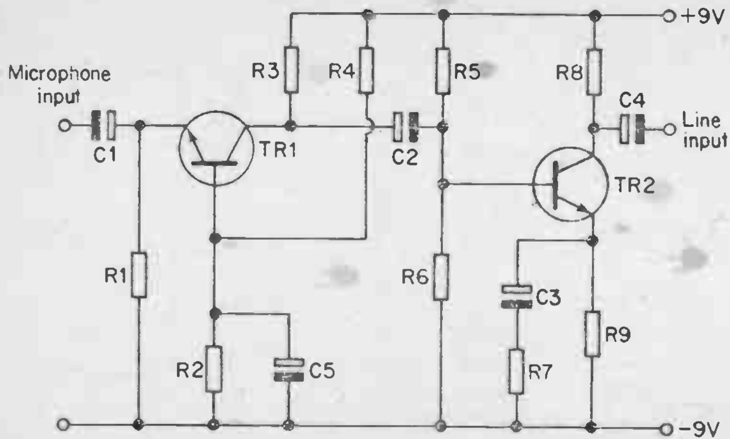
Using the Mixer

With the information given in the earlier part of this chapter, you should be able to make full use of the mixer's various functions. This mixer can now, and should be, used instead of the passive mixer described in Chapter 4 for the various recording techniques. With the extra inputs it is possible to use several sources for each recording. Try taking signals from different parts of a synthesiser set-up, such as taking the outputs from the VCO, VCF, etc. as well as taking the final output from the VCA.

Note that as this mixer is a stereo one, you will have to arrange that it works in mono when you use the recording techniques in Chapter 4 (except, of course, if you use two stereo recorders and record from one to the other in stereo, so as to achieve a final composition in stereo). To do this, turn all pan pots hard left, and then just use the left-hand output.

Microphone Inputs

As it stands, the mixer will accept line inputs between 50mV and 1V. If you wish to feed a dynamic microphone into the instruments, it will be necessary to use a preamplifier. Such a preamplifier is shown in Fig. 5.10 and can either be constructed within the mixer or in its own box, run by two 9V batteries The latter is probably the more versatile solution.



COMPONENTS

R1	1K2
R2	12K
R3	12K
R4	120K
R5	120K
R6	12K
R7	1K
R8	10K
R9	1K
C1	250μF 25V elect.
C2	10μF 25V elect.
C3	50μF 25V elect.
C4	10μF 25V elect.
C5	50μF 25V elect.
TR1	BC109C
TR2	BC109C

Fig. 5.10 Circuit diagram of microphone pre-amplifier.

CHAPTER 6

ELECTRONIC SOUND EFFECTS

In addition to the basic effects provided by the usual VCO's, VCF's and envelope shapers found in synthesisers, there are a host of other sound effects that are formed by electronic means. The major types of effects are described in this chapter. Many of them were originally designed to be used as sound treatment devices for electric guitars, but all can be used to modify any sound, whether it be from an electronic instrument or an acoustic instrument via a microphone.

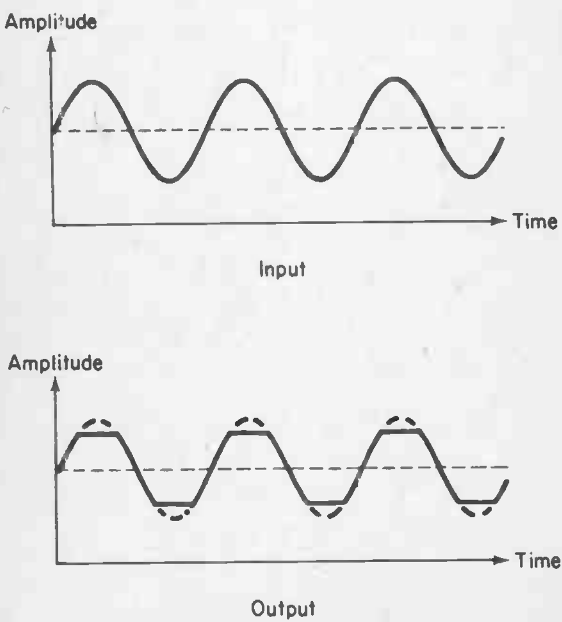


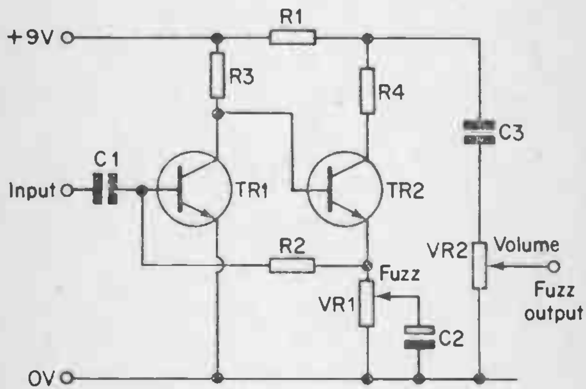
Fig. 6.1 How a signal is clipped to produce FUZZ.

Fuzz

One of the first treatments for electric guitars, fuzz is merely the adding of distortion to the basic guitar sound, thus giving an output rich in harmonics.

This is usually achieved by feeding the input through a high gain amplifier, so that the signal is clipped or squared off (Fig. 6.1.). A gain control varies the amount of distortion or "fuzz" in the signal. A simple fuzz circuit is shown in Fig. 6.2.

In more complex fuzz circuits, tone controls may be fitted, so that you can have varying types of fuzz.



COMPONENTS

R1 470Ω
R2 100K
R3 68K
R4 6K8
All 10%
¼W carbon

C1 0·22μF polyester
C2 33μF 16V elect.
C3 0·22μF polyester

VR1 1K lin.
VR2 10K log.

TR1 BC107
TR2 BC107

Fig. 6.2 Fuzz effect circuit.

Wah-Wah

Another of the early effects, this one is where a band-pass filter has a varying centre frequency. The sound can be imitated by repeating the word "Wah-Wah". This effect has been used a lot with guitars and is often used in the form of a foot pedal, whereby the centre-frequency of the filter is varied by movement of this pedal. As shown in Chapter 4, this effect can be achieved by using a voltage-controlled filter and an envelope shaper.

Swell and Sustain

A swell pedal is just a foot-operated volume control, and is often used to change the signal of electric guitars by cutting off its attack to give it a much smoother envelope. This allows the instrument to imitate other instruments like the violin or even the bagpipes, with a little help from a fuzz circuit and some tone controls.

An extension of the swell effect is sustain, where the output level is automatically kept at the same level while the input signal may be decaying. The inexpensive method of doing this is to use a circuit similar to a fuzz circuit, which holds the output constant by squaring off the signal. The drawback though is that it too introduces distortion. Whereas this is desireable in the fuzz circuit, it does not enhance the sustained signal.

A better arrangement is to use a VCA to control the amount of amplification of the input signal. The output signal is tapped off and fed to a rectification stage to give a d.c. voltage which is proportional to the peak output voltage. This d.c. voltage is then used to control the VCA. With this method little distortion is introduced.

A guitar fed through a sustain circuit can be made to produce some very long notes, but towards the end, the sound will become noisier, because the pick-ups will tend to collect more noise and hum than string vibrations when the strings are virtually still. Therefore there is usually a limit on the sustain time to stop this from happening.

Tremolo

This effect is amplitude modulation, and must not be confused with vibrato, which is frequency modulation. Tremolo is generated by controlling a VCA through which a sound is passing with a low frequency oscillator, usually between 1 and 10Hz. This gives a continually changing volume of the sound. The frequency of the oscillator defines the "rate" of tremolo, and the amplitude of the modulating waveform defines the "depth" of tremolo.

Vibrato

As stated above, this is frequency modulation, that is, the pitch of a sound is varied up and down. This effect is simply produced by using a VCO and controlling it with a low frequency oscillator (which could be another VCO). Vibrato is often built into electronic keyboard instruments such as pianos and organs. With most other conventional musical instruments it is possible to apply vibrato to any note by using certain playing techniques.

Electronically, the effect is achieved using phase shifting, and usually this phase shifting is voltage-controlled so that a low frequency oscillator can be used to make the effect automatic With this system, any sound can have vibrato added. Note that the first system described (using a VCO) generates its own sound and cannot be used to treat sounds from other instruments.

Ring Modulation

The output of a ring modulator consists of the sum and difference frequencies of two input signals. Thus, for two input signals with frequencies a and b, the output signal will have frequencies (a + b) and (a – b). In a synthesiser the two inputs are usually provided by two VCOs. By carefully tuning each VCO so that their frequencies are closely related (e.g. in unison, or tuned apart by an octave or a fifth), then if the output of the ring modulator is mixed with the original input signals from the VCOs, the resulting sound is a four note chord which is musically concordant. The sound appears to be comparatively complex considering the system from which it was generated.

Noise

It may be surprising to learn that the noise that we spend so much time trying to reduce in hi-fi systems is actually a very good electronic music generator. The type of noise that is used is known as white noise, and is what can be heard when tuning in between stations on the VHF FM radio band.

White noise consists of all frequencies in the wave spectrum. It is called "white" because of the analogy to white light, that is, consisting of all colours in the spectrum. A particular type of noise is pink noise, which has an emphasis on the lower frequencies, producing a sound that resembles distant rolls of thunder.

There is currently a diode on the market specifically intended for white noise generation. It is the noise diode Z5J which, using the simple circuit of Fig. 6.3, will produce an output of some 50mV peak to peak.

Pink noise can be produced by filtering off some of the higher frequencies. In fact, interesting percussive effects can be produced by using a filter and also an envelope shaper with VCA to give some sharp attacks.

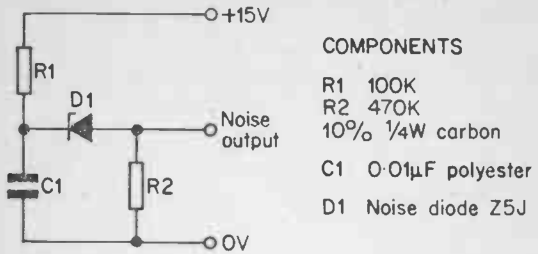


Fig. 6.3 Noise generator circuit.

Phasing

If a signal is split into two equal parts, and one part is delayed slightly so that it is out of phase with the other, then when the two are remixed, certain frequencies will be cancelled – a comb filter is in operation. If the delay time of the one part of the signal is then gradually varied about the point where the two signals are exactly in phase, so that is starts as a delay and then becomes ahead of the fixed phase signal, the familiar phasing sound results.

In practice this can be done by using two identical recorders playing back the same signal where one machine is equipped with varispeed facility so that the delay between the two signals can be changed and the phase shift effected.

Electronically, the signal is split and one part goes through a short delay line to be remixed with the original signal.

Echo

Echo is the repetition of a signal, either once or more than once, with a distinguishable delay time. The easiest and most common method of producing this effect is to use a tape loop as shown in Fig. 6.4.

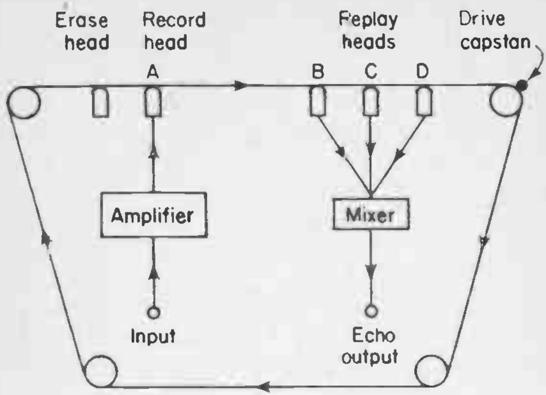


Fig. 6.4 Echo tape loop.

The signal is first recorded on the tape via the recording head A. The tape then passes across the heads B, C, D, which replay the recorded signal. Thus three echoes are heard a short time after the original was heard. The actual delay time depends on the distance between the heads and on the speed of the tape. Therefore the delay time can be varied by changing the speed of the tape.

The only disadvantage with this system is that the tape tends to wear out very quickly. Cartridges have been used which are really just large tape loops.

There can also be more than just three replay heads. Commercial models with 7 or 8 are not uncommon. The output from each head is mixed together with the others to form the final output.

Recently, digital delay lines have been produced, where a signal is fed in at one end of a series of digital registers. The signal is clocked through this chain and emerges at the other end after a delay time which depends upon the clocking speed.

This delayed signal forms the first echo and can be fed through the same delay line an infinite number of further times.

It should be fairly apparent where echo can and cannot be used. One trick that you could try is to set the delay time to something less than a tenth of a second. Then, using just the first echo, it will seem as though the piece of music is being played on the same instrument at the same time.

Reverberation

This effect was mentioned in Chapter 2, and a method of producing it was described. This method used a reverberation chamber, and we will now see some electronic methods.

The first is a digital delay line, described above, which can be used for both reverberation and echo.

Normally, electromechanical means are used. In studios, large plates of metal have a transducer fixed to one side and microphones scattered around the other sides. The plate is suspended on springs to allow it maximum vibration.

A signal is amplified and fed to the transducer, which then sends vibrations across the plate. The microphones pick these up and turn them back into electrical signals which are amplified. When the original signal finishes, the plate continues vibrating, but with an ever decreasing amplitude. Thus the output from the microphones will be an effect very much like actual reverberation.

The cheaper version of this system is the spring-line, where the metal plate is replaced by a spring with transducers at each end. Fig. 6.5. shows how the system is arranged. It works on exactly the same principle as the metal plate. Note that there is a potentiometer marked "Reverberation". This mixes the reverberated output with the original signal, so that you can have varying degrees of reverberation.

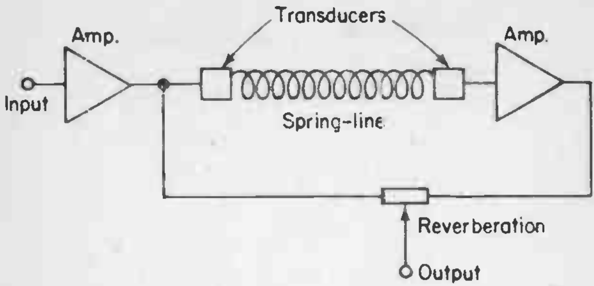


Fig. 6.5 Reverberation – spring-line method.

Reverberation is very useful in electronic music because it can add a great deal of life to what may otherwise seem to be very clinical. For instance, if your composition is building up to a grand climax with lots of different sounds to it, you will not want it to finish suddenly by immediately dropping in volume from the height of the music to sudden silence. So you add reverberation, and then you get that effect when an orchestra comes to the end of the 1812 overture while playing in a large concert hall. The sound takes a few seconds to die away.

CHAPTER 7

RHYTHM

There has been much debate as to whether electronic music needs a rhythm base to it. In the early days, there was very little rhythm in compositions, and this tended to make them sound alien to most people. A few people would "freak out" to them of course, but the vast majority of ordinary people just could not understand them. To them, they sounded disjointed, and with no direction at all. Of course, adding a recognisable beat to electronic music takes it away from the original intentions which were to break away from the restrictions and limitations of conventional music, and this is what the purists did not like. Naturally, electronic music can be produced in that form, and only that form, where there is no rhythm, no fixed scales and no pattern to the music, but it defeats its own object by ruling out these constituents.

The job that rhythm performs, in electronic music and all other types, is to provide a base on which to build all of the melodies, harmonies, and so on. In addition, electronically generated rhythm can be much more compulsive than conventional percussion, and provides a hook to the music, a fact reflected by recent success in the pop charts by records using programmable sequencers (see later) to provide rhythm.

This chapter describes the various methods of generating rhythm, using tape and/or electronic means.

Rhythm Using Tape

Tape loops were described in Chapter 2, so no further information is required, except for the fact that you can now include a tape loop rhythm in your composition by using one of the recording techniques described in Chapter 4. The rhythm track should always be the first one you record anyway, so first record the drum beats or clicks, and make up the tape loop. Then replay this on one machine while recording it on another as the first signal in the recording process. Record for as long as you need the rhythm plus a bit more. Remember that when the second signal is added, you can fade out the. rhythm track whenever you want to, using the channel fader on the mixer. It is probably as well, therefore, to record enough rhythm track to last the entire length of the composi-

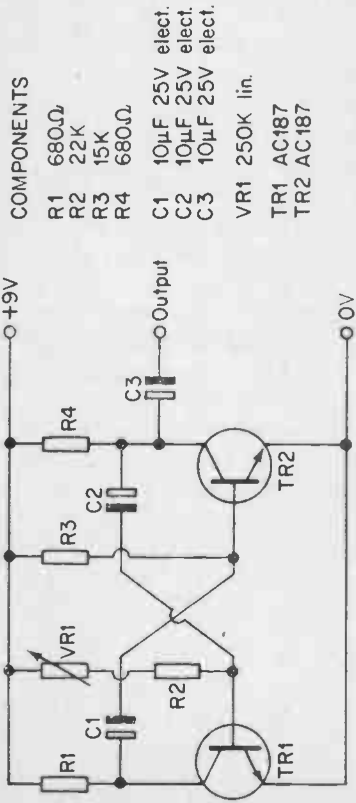


Fig. 7.1 Electronic metronome.

tion, since even if it is not heard in the final production, it is helpful to keep all the various "instruments" in step with one another.

Another use of tape for rhythm is often found in commercial electronic organs. A pre-recorded cassette or cartridge contains recordings of actual percussion instruments played in a variety of different tempos and styles. It is fine for playing all of the usual type of music, but not very useful as far as electronic music is concerned.

Electronic Metronomes

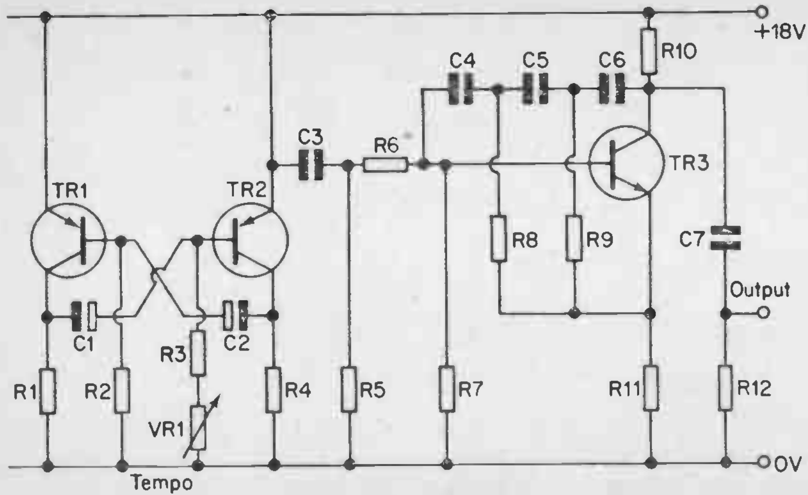
Mechanical metronomes for keeping time have been around for a long time. An electronic version is easy to make, and with a little imagination can actually be used as a rhythm track.

A very basic metronome is shown in Fig. 7.1. It is simply a multivibrator with variable frequency between 40 and 200 beats per minute. It produces a click for every beat. This click can, however, be used as a trigger for an envelope shaper in a synthesiser set-up.

Fig. 7.2. shows a similar metronome, this time with the addition of a voicing circuit. With the component values shown, the circuit will produce a beat which has a sound that resembles a clave, with frequency 1500Hz. To change the pitch of this beat, the value of R8 should be changed slightly.

Pre-programmed Electronic Rhythm Generators

These rhythm generators have a number of pre-programmed rhythms such as Tango, Waltz, March etc. and a number of electronically simulated percussion instruments; bass drum, snare drum, bongos etc. Usually, any one selected rhythm will automatically choose its own instruments so that, for example, for a march, the snare drum, bass drum, and cymbals are used,



COMPONENTS

R1	12K
R2	120K
R3	22K
R4	12K
R5	150K
R6	390K
R7	68K
R8	27K
R9	27K
R10	15K
R11	2K2
R12	2K2
All 10% ¼W carbon	
C1	1µF 63V elect.
C2	1µF 63V elect.
C3	0·1µF polyester
C4	2200pF polystyrene
C5	2200pF polystyrene
C6	2200pF polystyrene
C7	500pF polystyrene
VR1	1MΩ log.
TR1	AC128
TR2	AC128
TR3	BC109

Fig. 7.2 Metronome with pitched beat.

and no others. Other generators, though, do allow for freedom of choice of percussion.

Recently, due to the advancing technology of large scale integration. some IC's have been brought on to the market. Their job is to accept a clock pulse and provide a string of pulses for the beats, including an accented down beat. They also have to send the right pulses to the right instrument simulators according to which rhythm they have been told to perform. Fig. 7.3. shows a block diagram of the associated circuitry to go with these IC's. An external generator provides the input pulses to the IC, and its speed can be varied, thus changing the tempo. Selector switches tell the chip which pre-programmed rhythm it has to make out of the clock pulses. The chip then feeds various pulses to the voicing circuits. Whenever these receive a pulse, they transform it into the sound of the percussion instrument they are supposed to be simulating. Finally, the outputs from all of the varying circuits are mixed together and fed to an amplifier and loudspeaker.

Programmable Sequencers

Since most circuits in an electronic music synthesiser rely on voltage control, a device which can remember a sequence of voltages would be a very useful thing, and this is exactly what the sequencer does. In the simplest type, voltages are set up on potentiometers and then a clock sends each voltage in turn to the output. When it reaches the last potentiometer, it resets itself and begins stepping through the potentiometers again. The speed at which it does this can be varied by changing the speed of the clock. The output, a sequence of voltages, can be taken to a VCO. This arrangement can produce some very melodic rhythm patterns and at some very rapid tempos.

More complicated sequencers involve the use of solid state memories, and each note is stored in the memory direct from a keyboard, so there is not a large bank of potentiometers to set-up. Typically, the number of notes that can be stored is 128 or 256, so that this type of sequencer does not stop at

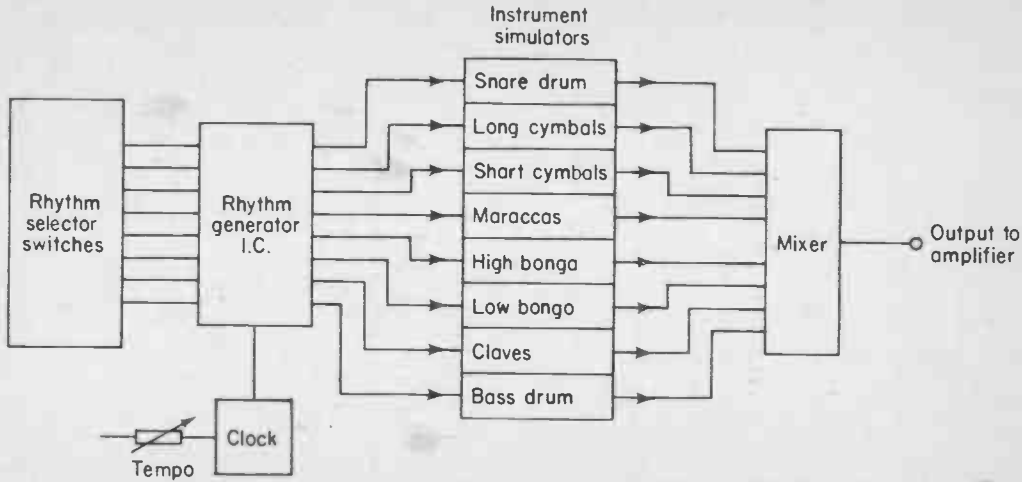


Fig. 7.3 Block diagram of pre-programmed rhythm generator.

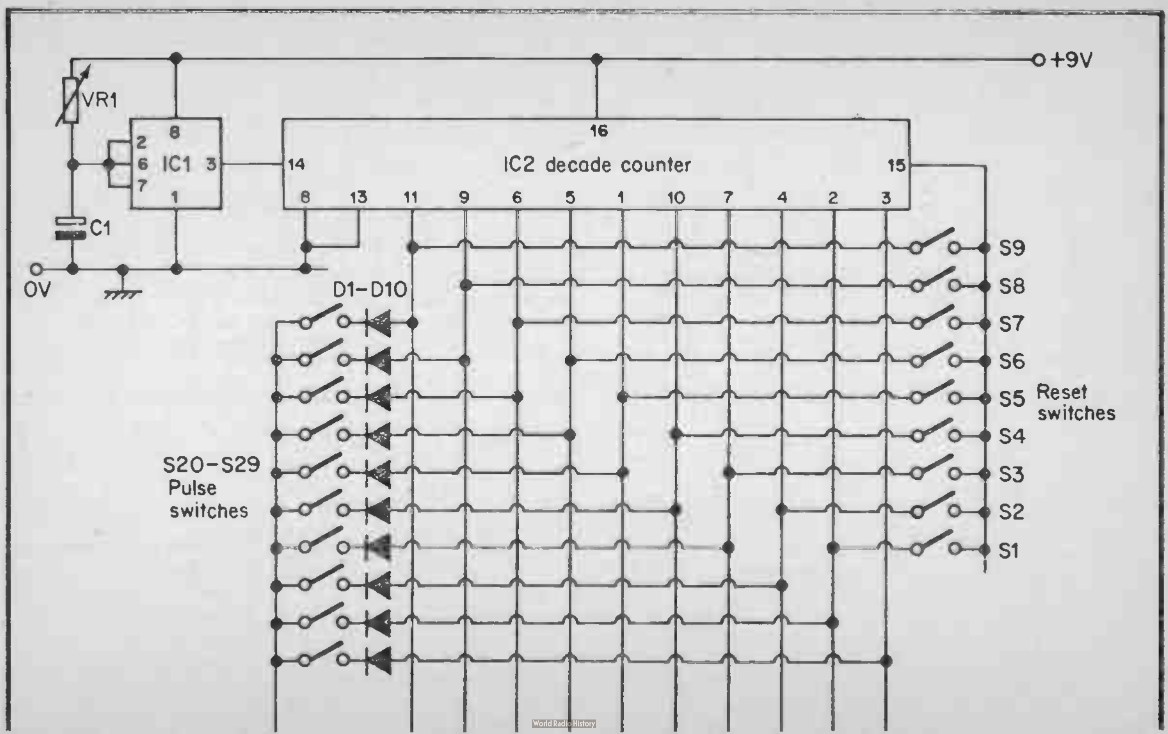
simple rhythms, but can store complete tunes – a rival to the recorder perhaps!

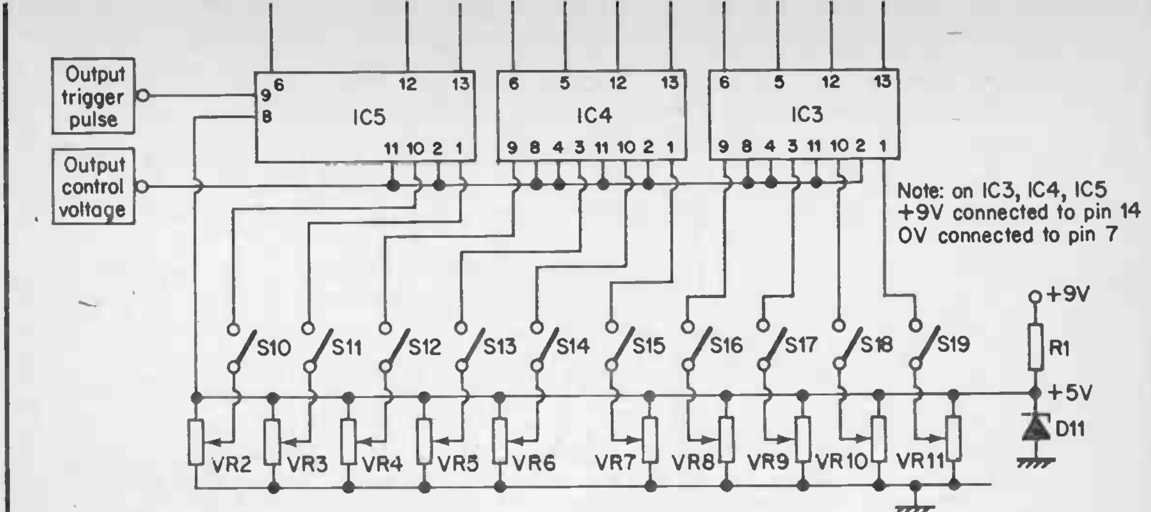
In addition to providing a control voltage output, most sequencers can provide a pulse that can be used to trigger envelope shapers. A pulse can be written in to the sequencer's memory at any time during the voltage sequence.

A 10-Note Programmable Sequencer

This design is of a sequencer that can have up to 10 notes whose control voltages are varied using potentiometers. The sequence can be reset to the beginning at any of the 10 notes, but will automatically reset after the tenth if left to its own devices. Any one channel can be switched out so that no voltage is sent from that channel when the channel is connected to the output. A pulse is provided (switchable) on each channel to enable envelope shapers to be triggered.

The circuit diagram is shown in Fig. 7.4. IC1 is the clock pulse generator, whose frequency is controlled by VR1. It drives the decade counter IC2, which sends a pulse out on 10 different pins one after the other. If any of these pins are connected to the reset pin 15, then the sequence immediately recommences from the beginning. Switches S1–9 perform this function. The ten outputs of IC2 control solid state switches IC3-5, which switch in and out the ten preset voltage devices. These are merely potentiometers working off a reference voltage provided by a Zener diode. The trigger output for envelope shapers is provided by taking each output of the decade counter, via a switch and then to one of the spare switches in IC5. The output pulse is +5V, although the additional circuit of Fig. 7.5. will invert it to about –8V. This requires a –15V supply, but the synthesiser circuits described in Chapter 3 also required this power rail, so an extra power supply does not have to be provided. The inverted trigger pulse can be used for the envelope shaper of Fig. 3.7.


+9V
VR1
IC1
2
6
7
8
3
1
C1
0V
16
IC2 decade counter
14
15
8
13
11
9
6
5
1
10
7
4
2
3
D1–D10
S20–S29
Pulse
switches
S9
S8
S7
S6
S5
S4
S3
S2
S1
Reset
switches



COMPONENTS

R1 – 2K2 10% ¼W carbon
C1 – 10μF 25V electrolytic
VR1 – 1MΩ lin. carbon
VR2–VR11 – 10KΩ lin. carbon (10 off)
S1–S29 – single pole on/off (29 off)

D1–D10 – 1N914 (10 off)
D11 – BZY88 5·1V zener diode
IC1 – NE555 timer
IC2 – 4017 decade counter
IC3–IC5 – 4016 quad switches (3 off)

Fig. 7.4 10-note programmable sequencer.

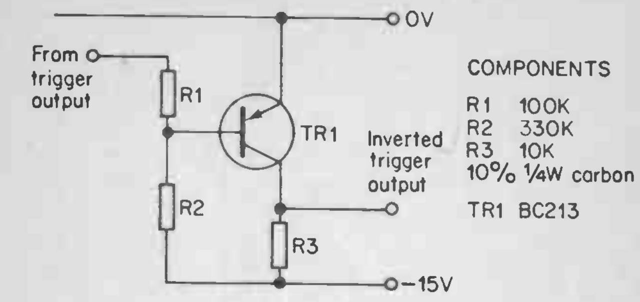
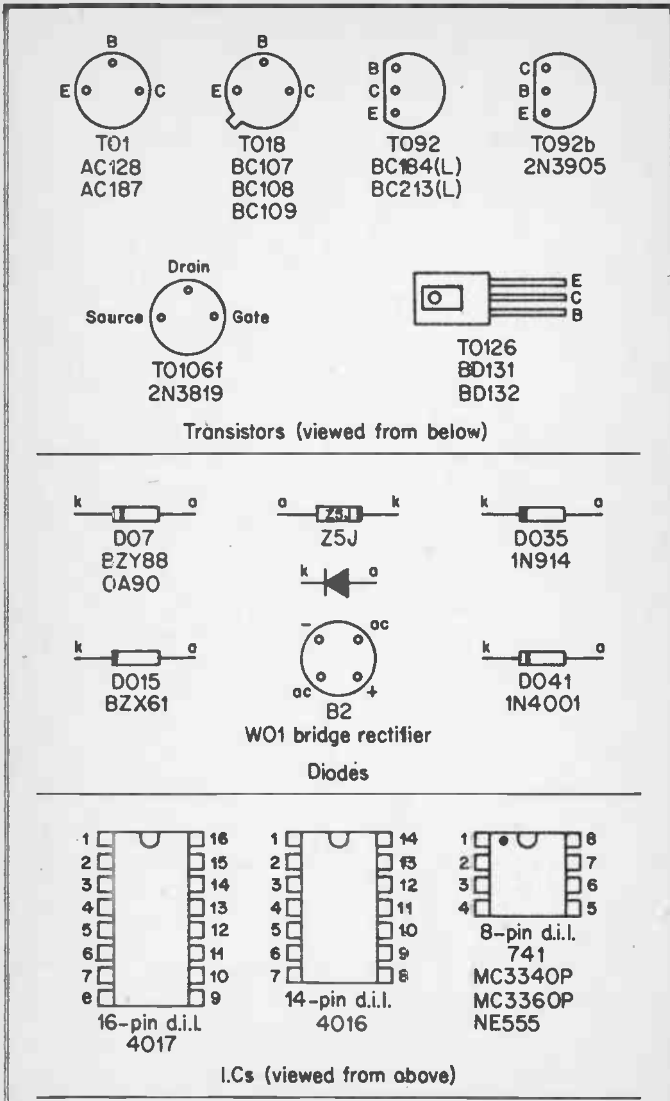


Fig. 7.5 Trigger pulse inverter.

The sequencer should be constructed with care because all of the integrated circuits used are of the CMOS variety. CMOS devices are particularly prone to damage by static electricity, and this is easily generated in the home. The "chips" are provided in either a conductive film or aluminium foil, the object of which is to short-circuit the pins. Therefore never take off the protective packing until you are ready to solder the "chip" into the circuit. Always work on an earthed metal tray and wear an earthed metal wrist-band. Endeavour to use D.I.L. sockets with all CMOS IC's, but if you must solder them in, ensure that the soldering iron is well earthed.

As far as the rest of construction is concerned, the sequencer can be built on Veroboard and encased in a box with a sloping front so that the 10 channels and their controls are arranged rather like a mixer. The whole instrument can be run inexpensively from a small 9V battery.

The control voltage output is normally used to control a VCO, or even two, but in addition it can be used to control a VCF. You should find that by using VCO's, VCF's and envelope shapers you will obtain some very interesting rhythmic patterns. With this sequencer it is possible to change voltage settings while it is running, so that you hear the effect of the changes immediately.



Appendix 1. Semiconductor base connections.

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