USING THE DIGISOUND 80 MODULAR SYNTHESISER

by

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Published by:
DIGISOUND LIMITED,

14/16 QUEEN STREET
BLACKPOOL
LANCS. FY1 1PQ

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PREFACE

When I prepared an article for 'Electronics Today International' on using the DIGISOUND 80 modular synthesiser the emphasis was on more unusual features, such as, synchronisation techniques. It became apparent, however, that many customers wished to start from basics and so I asked Bill Marshall if he would extend the original article. The result was increasing my mere 10 diagrams to 75 in the patching section and a proportional increase in text matter - with more to follow.

Almost all of our customers are experimenters who obtain a great deal of satisfaction from exploration and from generally doing things for themselves. This manual caters for this capability and the emphasis is entirely on patching the modules together and to making full use of the extensive control facilities provided. At this time we have largely avoided detailed patches aimed at simulating specific effects or conventional musical instruments. The manual is, however, also in a modular form and each section may be expanded, or new sections added, without destroying the pagination and general lay-out. We will, therefore, be pleased to receive from customers any patches of their own origination, including specific effects and sounds, if these are annotated with the control settings used. We will then publish the most useful of these as additions to sections, or as a separate section, acknowledging the author. We will also be adding specific patches of our own but the emphasis at this time must be becoming familiar with the synthesisers facilities and capabilities.

We have not delved into the theory of sound. There are many published books covering this aspect although we hesitate to recommend titles since they may be out of print or may not be exactly what you require. Furthermore none are fully comprehensive in terms of using a synthesiser and in view of their high cost they should be perused prior to purchase. We recommend, therefore, that when the opportunity arises you should take a look around some of the major bookshops - the music departments of Foyles in London and Blackwells in Oxford are known to be good sources but undoubtedly there are many others.

Glancing through the manual, as one is apt to do before getting down to using it, may reveal large patches using multiple VCO's and the like. Do not despair since if you progress steadily through the main using section (Section 4) in a sequential manner you will still be able to find plenty to do with a 'basic' synthesiser and at the same time learn how to use the functions provided on the modules. You should, however, study the complex patches since it will allow you to determine the most rewarding way of expanding the synthesiser. We are often asked which modules should be purchased but the answer must depend on what your specific aims are. Is your interest in using the synthesiser for music making; is there a greater emphasis on the fascinating electronic aspects; on combining the synthesiser with another instrument such as an organ or a guitar; for specific sound generation such as a study of percussive instrument effects; and so on. Only you know this and your expansion will be in the direction that assists you in reaching your goal. With the introduction of the 'ALPHADAC 16' computer controller the DIGISOUND 80 is already one of the most comprehensive systems available and subsequent additions to the range of products offered will ensure that this is maintained. For the beginner we recommend that he (or she) starts with a very basic unit, namely, a VCO, a low pass filter, two envelope shapers, the dual VCA, keyboard and power supply. One can then make a start with this manual and build the system up according to your aims.

We sincerely trust that this manual meets your immediate requirements and we look forward to its further expansion. Happy patching.

Charles Blakey

1

INTRODUCTION

It is assumed that the reader will have a knowledge of synthesiser terminology but most of this will have been picked up from the construction notes for the modules and some of this is re-iterated in Section 3 of this manual which describes the facilities found on each module. Likewise it would obviously help a great deal if the reader was acquainted with some of the basic aspects of sound but the level required is not very high - it is a bit like driving a car; as long as you know how to steer it, the knobs to touch and the 'rules' you will get along fine and you do not need to know how the thing works, although sometimes the latter helps!

The DIGISOUND 80 is fully described in the general leaflet on the synthesiser and its general specification will not be repeated here. The main point to remember is that it is designed with a 'plug in anything to anywhere' capability so that no damage will occur if you connect any of the 3.5mm sockets together. Thus you are free to experiment with safety and the worst that is likely to happen is some non musical outputs or perhaps no output at all until you locate the wrong connection.

Another aspect we wish to stress is that the control inputs to the modules mostly accept 0 to +10 volt signals and where panel space permits an attenuator is put on the input of the control signal. In some instances such attenuators have by necessity been omitted and hence we have the 80-5 Processor module which provides, among other things, external attentuators and distributors. While the control inputs can accept the full 10 volt range if you modulate them over this range with sharp signals, such as pulse and sawtooth waveforms, then some control breakthrough may occur. The same effect will usually happen with the most expensive electronic equipment, namely, if you rapidly take it back and forth over its full range then some undesirable effects are likely to occur. In other words, there is a difference from 'bench' patching, where you may simply be plugging an output of one module to the input of another to explore its effect, and patching for a musical effect where often a gentle modulation gives the best results. Likewise if you connect a 'sharp' waveform to a VCA or VCM in a static patch then there may be a small residual signal at minimum control level. Again, in practice, when one is playing the synthesiser the waveforms are often 'softened' by filtering which immediately reduces the level of residual signal. Furthermore the sound output is rarely allowed to fall to complete silence. If you require absolute silence then it can be obtained by various techniques. In a modular synthesiser you have the capability to push the modules to their limits, and beyond, whereas in a pre-patched synthesiser the same situation cannot arise since it limits the users range of control. In summary, keep a distinction between playing the synthesiser and playing with the synthesiser.

A monophonic synthesiser is one of the simplest instruments to play. A control voltage derived from the keyboard electronics is connected to a voltage controlled oscillator (VCO) to determine the pitch and the resulting waveform is fed into a voltage controlled amplifier (VCA) where the sound is shaped by an envelope shaper whose control signal also comes from the keyboard. This set-up will produce notes directly related to the keyboard but the resulting sound will be very uninteresting as we are all accustomed to listening to more complex spectra. We have to examine ways in which the resources of the synthesiser can be utilised to provide useful musical and other sound structures.

The first step is to alter the timbre of the note so that the sound will be pleasing to the ear and the simplest approach is selective filtering of the harmonics in the waveform from the VCO. Filtering is extremely effective, and often the only technique employed in small synthesisers, but the use of additive and subtractive synthesis techniques are advantageous. For example, the sine wave from the same VCO can be added to, or subtracted from, the filtered sawtooth wave to vary the level of the fundamental frequency in a more controlled manner. Alternatively, other VCO's may be used to boost or cut the partials or even introduce some inharmonics. It has been demonstrated that the warmth of a piano tone is due to the fact that the upper partials are not exact harmonic relationships of the fundamental frequency. Naturally the use of these more sophisticated techniques requires a larger synthesiser system and there is a trade-off between striving for near perfection, which costs money, and a sound which is acceptable. The latter applies whether one is concerned with imitative synthesis of a conventional instrument or creative synthesis of a 'new' sound.

Another feature to overcome is the monotonous nature of simple synthesiser patches, much of which arises from the precision of electronics in contrast to the craftsmanship, use of natural materials and environmental effects which have a major influence on the sound of conventional instruments. Again there are some very simple techniques which are incorporated into many synthesisers for imparting a more dynamic character to the sound. One can, for example, connect an envelope shaper to the voltage controlled filter (VCF) so that the timbre of a note is changing for at least part of the duration of a note. Other techniques include amplitude modulation (tremelo), frequency modulation (vibrato), phasing or pseudo-phasing, pitch bending and dynamic control of pulse width (pulse width modulation).

Reverting to the piano, a high quality grand will have about one hundred parts per key and the resultant sensitivity to velocity and pressure provide the player with a wide range of dynamic control. These keyboard features can be simulated electronically - at a price - but for many applications it is unnecessary. The important point to remember is that our aural responses have become accustomed to complex sound structures with a wide dynamic range. Thus if one does not have velocity and pressure sensitivity on the keyboard then it is a matter of using more cost effective resources to achieve a dynamic character for electronically created music.

The purpose of this manual is to provide the user of the DIGISOUND 80 synthesiser with information which will enable the best results to be obtained from the equipment, in particular exploring some of the aspects touched upon above. We present a brief outline of the principal use of each module and follow this up with diagrams of applications that involve several that are inter-connected or 'patched' together. The fact that one device can control another in some way is what allows a synthesiser to produce an exceptionally wide range of sounds. We do not attempt to provide a list of specific sounds, which the user will later be able to produce for himself, but rather to illustrate certain standard patches and techniques. This progresses to more advanced developments which may be used as building blocks for the creation of novel sounds as well as providing a basis for a more thorough understanding of sound synthesis in general.

Finally, we advise the reader to work through the manual from the beginning. This is particularly important in Section 4 which builds up in a progressive manner and if you attempt a patch part of the way through without at least studying the earlier parts then you may find it very difficult to interpret.

2

NOTES FOR GENERAL GUIDANCE

Some of the points in this section may appear trivial and obvious but please read it carefully and heed the advice given, which is intended to allow you to make good progress through Section 4.

RECALIBRATE the modules when necessary. This may be quite soon after initial calibration but thereafter the interval should be quite long. So often one comes across synthesisers which are not in tune, particularly amongst constructors whose main interest is inclined towards the electronic aspects of music making. In the latter circumstances the resultant output sounds awful. If you take the time to tune it then you will get much greater satisfaction from the results whatever your inclination.

BE SYSTEMATIC. Devise either a flow chart type of patch as used in Section 4 or draw up a tabular recording system with the outputs of the modules forming the rows and the inputs being the columns. In either case space should be allowed for recording the settings of the various control potentiometers. When a useful sound is obtained then make a record of the patch and describe the sound for future reference.

UNDERSTAND THE METHOD. One may be starting with a specific sound and trying to improve its qualities or modifying it to another sound. The patch may sometimes become so complicated that you have lost track of what is happening. When the latter point is reached then scrap the patch and start again from the original point.

USE REALISTIC METHODS. When experimenting with the sounds of musical instruments avoid playing notes singly. Even if one is lacking in keyboard skill at this time then still play a few notes at the appropriate tempo since the end result will be quite different and obviously more realistic. Likewise, when creating sound effects then ensure that the amplifier is set to the appropriate volume - one cannot emulate the sound of gunfire quietly!

IMPROVE THE QUALITY. Once a good imitation of a conventional sound is obtained, or a new sound created, then one can begin to explore ways of improving its dynamic quality. In other words avoid becoming too complex too quickly but start by developing a repertoire of sounds which may be improved upon at a later date as you become more familiar with the capabilities provided in the DIGISOUND 80.

UNDERSTAND THE SOUND. Try to work out what is happening to the sound as it is processed. An oscilloscope is certainly a useful aid but if one starts off with simple patches then the ear is just as good. After a relatively short time one should not be in a position of finding out what happens when, say, a high pass filter is used for treatment. Instead, one should be working towards the situation where the use of a particular treatment is a logical conclusion - this is what Section 4 is all about.

HAVE AVAILABLE an adequate number of patchcords of varying lengths and keep them separated in a suitable rack. If you do this then a modular synthesiser becomes a pleasure to use rather than an irritation since there are few things more frustrating than hunting through a heap of patchcords to locate one of the correct length. See later regarding further advice on patchcords.

EXPAND LOGICALLY. Do not be under the misapprehension that the modules provided by a 'basic' DIGISOUND 80 synthesiser will allow the creation of virtually any type of sound. On the one hand there is much scope for additional tonal treatment, reverberation and so on but do not complicate matters by obtaining these until the basic resources have been mastered. The other aspect, namely, increasing the number of modules has been touched upon elsewhere. If one starts small and then approaches synthesis in a logical manner then future expansion to meet specific needs should be obvious. There is, however, much in favour of a four 'voice' system, i.e., modules equivalent to four times the basic patch shown in Figure 4.1.1, with an ALPHADAC 16 computer controller. With such resources one may explore complex single voice patches and at the same time have the capability of playing in the polyphonic mode, sequencing, composing and using the other keyboard routines provided which are invaluable to both the skilled and unskilled keyboard player.

As readers will be aware the DIGISOUND 80 is an easily expanded system but the initial publication of the project ended with a 'basic' system and thus high signal levels at the final stage, the VCA, prior to either the 80-14 modular amplifier or an external amplifier. Such signal levels are not ideal for external amplifiers, mixers and the like since the input attentuators on the latter will be near their minimum and it is easy to overload the external circuits and cause distortion. As we write (Autumn 1981) there are, however, plans to expand the DIGISOUND 80 to include other sound treatments and some of these will be post-VCA modules. The output signal from the latter equipment will conform to the more usual requirements of external equipment of the type mentioned above. Interfacing the synthesiser will also be dealt with in a subsequent section (5) of this manual entitled the 'DIGISOUND 80 ON STAGE'.

The final part of this section deals largely with the subject of patchcords. We recommend that all jack sockets on the DIGISOUND 80 are grounded since this makes it a simple matter to connect up with external equipment of all types which is powered from another supply. Thus the use of screened cable is the obvious choice for patchcords. Screened cable also guards against picking up extraneous signals. It is worth noting, however, that the authors of this manual have used unscreened cable for interconnections within the DIGISOUND 80 without encountering any pick-up problems. The electrical environment will, nevertheless, vary from customer to customer and thus we cannot vouch that this will work for all. As a compromise one could use unscreened wire for control voltages, for example, the gate and keyboard CV while retaining the screened cords for audio signals. A particular advantage of using single wire is the ease of obtaining various colours which then aid identification of routing. This is particularly important when patching the ALPHADAC 16 in a polyphonic situation. In fact in this situation we recommend making up a harness of wires to conform to the basic patch of Figure 4.2.1 and routing this between the modules such that it does not interfere with the controls. This in no way decreases the flexibility of the system since jack plugs may still be freely removed at either end and, say, connected to another waveform from the VCO. Likewise the wires for Channel 1 within the harness may be disconnected completely and interconnections made with single cords in the usual manner when a complex patch is required for the monophonic channel of the ALPHADAC in the split mode. There are pros and cons concerning the use of a separate colour for each voice as against different colours to identify the gate and CV signals. In the latter case it is best to mark the jack plugs in some way to identify the voice channel and this may be with a self adhesive label with the number on or else a band of coloured tape. Although this last part has been centred around the ALPHADAC the same techniques of using coloured wires and identifying plugs are, of course, applicable to a monophonic system. One must, however, approach it logically since to use a different colour for every type of control voltage (keyboard CV, gate, LFO, ADSR output, etc.) will greatly increase the number of patchcords since you will also need a variety of lengths in each colour. The single wire for patchcords should preferably be of a similar gauge to the screened cable supplied but also flexible. The type of wire often used for test leads is ideal since it 'hangs' well. a 2.8mm diameter wire of this type is available from Digisound Limited and obtainable in five colours - if black is counted as a colour!

A further aid to keeping the system tidy is to use two leads connected to a single jack plug as shown in Figures 2.1(a) and 2.1(b). It will generally be necessary to slightly enlarge the hole on the top of the plugs body using a file or a drill. The type shown in Figure 2.1(a) is useful for taking, say, the keyboard control voltage to both a VCO and a VCF which may be widely separated in distance while the type of Figure 2.1(b) is ideal for a gate voltage to two ADSR's since the latter may well be cn the same module, as in the 80-8D and 80-18D.

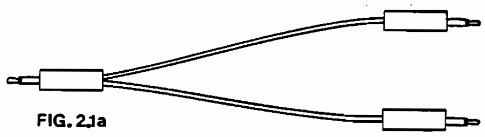


FIG. 2.1a

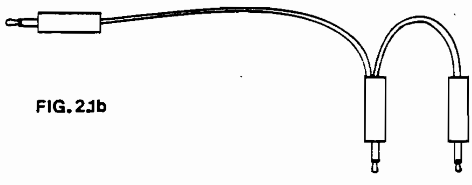


FIG. 2.1b

On the subject of tidiness we are often questioned as to the best lay-out of the modules. The short answer is that such an arrangement does not exist. With a 'basic' synthesiser then obviously the modules may be arranged in such a way that the patch of Figure 4.2.1 follows a logical flow but as the system expands then so arranging modules becomes more a matter of personal preference. In other words observe the patches you make most often and judge for yourself whether an alteration to lay-out will help -the modules are easy enough to move around at will. Another difficult question from customers concerns advice on a proposed system and the building of a nice cabinet in which all the space is spoken for. It is wrong to pre-judge the final size of a system and any module housings should allow for future expansion without detracting from aesthetic appearance which is why we have adopted a module housing which accommodates 12 standard modules and which neatly interlock together. If one is sometimes using the synthesiser for live performance then it is a good idea to accommodate the most widely used modules into one or two such cases and to house the less frequently used modules in another case, or cases. In this way one may perhaps avoid shipping the complete system around. Size of system and tidiness often go together since in a small unit one is usually using every module all of the time and hence the number of connections in a limited space looks complicated. On the other hand when one sees pictures of Wendy Carlos, Tomita and so on seated at their modular synthesisers then although there are numerous connections the overall effect is quite pleasing. The reason for the latter is simply that as the system grows then one will probably only use a proportion of the resources at any one time. A final comment on this subject is that although pictures of the DIGISOUND 80 show the front of the module housing in line with the rear of the keys it is better to step the module housing a few inches further back (suitably propping the module housing) since this reduces the likelihood of cords drooping onto the keys when one runs out of cords of the 'ideal' length.

Another type of patchcord is one which includes a diode connection, as shown in Figure 2.2, but one should take the precaution of marking which plug contains the anode of the diode (that to ADSR 2 in the figure). It will be evident from this arrangement that gate voltage 1 will cause only ADSR 1 to 'fire' but gate voltage 2 will operate ADSR 1 and ADSR 2 together. This is potentially a useful addition since without the diode the patch

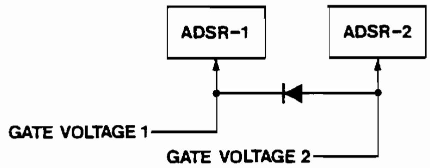


FIGURE 2.2

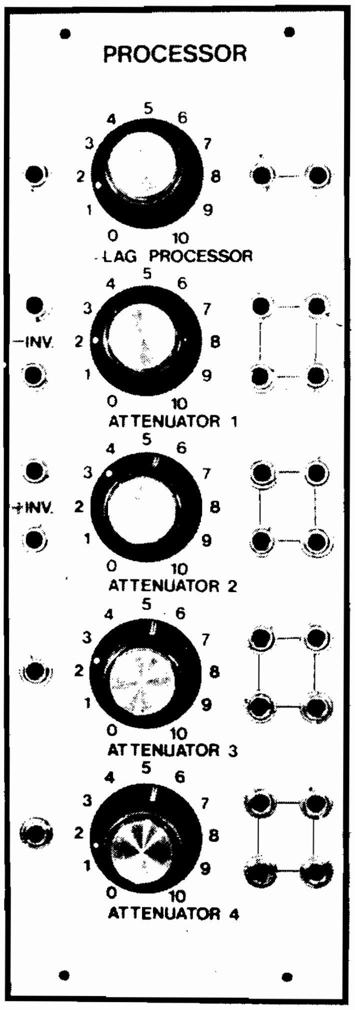
would be difficult to set up using conventional methods. The diode leads may also be used to advantage when sequencers are used for triggering functions and it will greatly simplify certain timing signals or gating applications.

3

THE MODULES

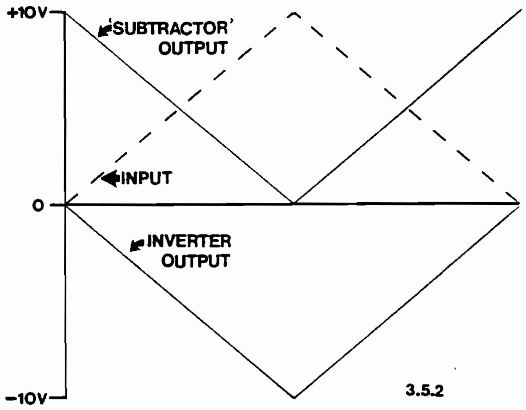
MODULE 80-5 PROCESSOR

The low output impedance and high input impedance of the DIGISOUND 80 modules allow one output to drive several inputs without overloading or introducing appreciable errors. In order, therefore, that a single output from a module may have individually adjustable levels to each of the modules that it is driving we have placed, whenever practical, attenuators on the inputs to modules. This arrangement also facilitates fading in of various effects, for example, if two modules are being modulated from another unit then one of the former two may be faded in and out without affecting the other. It was stated above that the attenuators are placed on the inputs whenever practical and in most cases it is the limitation of panel space which restricts the number that have been included. To overcome this problem of distribution we have included the 80-5 Processor module and this also includes a few other simple functions to aid synthesis.



Distribution of one output to four other inputs, with any level of attenuation on the combined output from the Processor, may be implemented in various combinations. For example, a single Processor may be used to distribute one signal to twelve other modules and with sets of three outputs at different levels of attenuation. For distribution purposes alone the Processor is invaluable and at least one is required for every ten other modules. Two of the distributors may also be used as 'inverters' or as a source of positive voltage for level shifting. To avoid confusion the term 'inverter' will be changed to 'SUBTRACTOR' since as constructed the effect is to subtract the input voltage from +10V. The main control signal in the DIGISOUND 80 is based on a 0 to +10V amplitude and thus if, say, the output of an envelope generator is taken via a 'subtractor' then the attack voltage will start off at +10V and decrease to zero instead of the normal response of going from 0 to +10V. The output from the 'subtractor' has an attenuator and thus the actual voltage excursion may be adjusted to the range desired. These 'subtractors' find wide application in synthesis patching, as is evident in the next section. It should be noted that signals are also inverted in phase when they pass through a 'subtractor'

Commonly the term 'inverting', especially as applied to operational amplifiers, means that the input voltage is inverted in polarity, e.g., a +5V input becomes a -5V output. There are some patches in synthesis with the DIGISOUND 80 which require this voltage inversion and it is recommended that one of the 'subtractors' is modified to an 'INVERTER'. The latter is simply achieved by removing the 130k resistor and 47k trimmer connected to the inverting input on one side of the op.amp. In addition to other uses the inverter also allows negative DC voltages to be obtained for offset purposes. If this modification is made then we suggest you mark the panel accordingly and probably the simplest way is to prefix the 'INV' by an 'S' or 'P' to denote 'Subtractor-Inverter' or 'Polarity-Inverter' respectively. The two modes are illustrated in Figure 3.5.2.



The other facility included in this module is the 'Lag Processor' which as the name implies is a signal delay device akin to the conventional portamento control on the keyboard. Thus control signals may be made to glide from one step to another and provide a more subtle transition. An example of this effect is a sawtooth waveform from a LFO being used to sweep a voltage controlled module. The sharp transition as the sawtooth reaches its peak voltage can result in an obtrusive 'plop' and the effect may be reduced, or totally eliminated, by the Lag Processor without detracting too much from the intended effect.

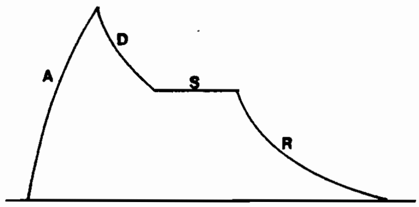
The Lag Processor is in essence a low pass filter with a manually adjustable cut-off frequency in the lower frequency range. Thus attempts to delay high frequency signals will also result in a decrease in amplitude of the signal. Nevertheless it does find application as a low pass filter especially in the treatment of white and pink noise sources.

MODULE 80-18
MULTI-FUNCTION ENVELOPE GENERATOR

Few synthesists appear to recognise the value of envelope generators other than for the usual applications of obtaining a sound contour when used in conjunction with a VCA or modifying the timbre during the course of a note when connected to a control input of a VCF. It should be noted that the combination of envelope generator plus VCA is often referred to as an envelope shaper. Envelope generators are, however, one of the most useful sources of control voltages, particularly since a single gate pulse can initiate a complex pattern of control voltages. It is hoped that this manual will illustrate some of the more diverse applications.

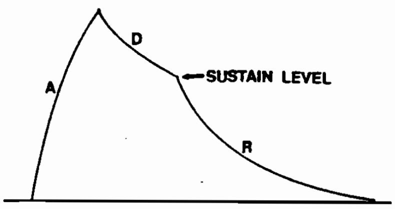
In view of the above there is a need for relatively low cost envelope generators to encourage their widespread use. At the same time, however, quality is important and especially the ability of the envelope output to return to near zero at the end of its cycle, irrespective of the settings used. For this latter reason the DIGISOUND 80-18 and 80-18A modules have been introduced to replace the 80-8. The 80-18 is a more versatile unit and has three operating modes which are simply selected via a single pole three-way switch. The three modes are:-

1. NORMAL. This is the conventional ADSR type of envelope, illustrated in Figure 2, in which the duration of the sustain period is determined by the presence of a gate voltage which in turn is equal to the period a key is depressed.

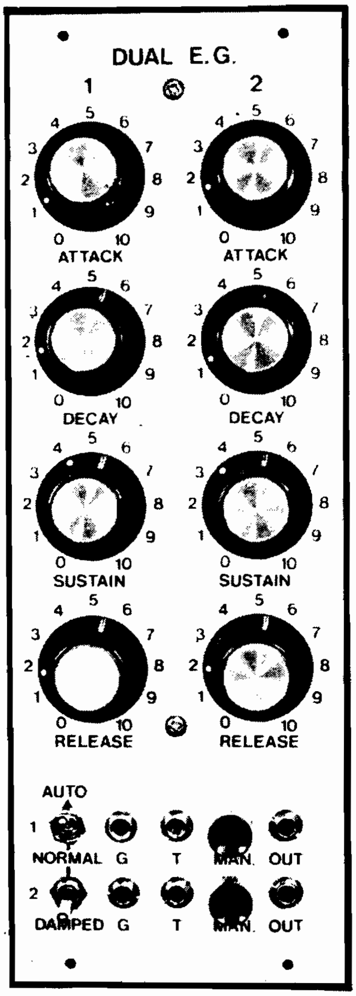


2. NORMAL ADSR ENVELOPE

2. AUTOMATIC. In this mode a short pulse will cause the envelope to cycle through a complete ADR envelope of the type illustrated in Figure 3. This mode is useful when the module is used in conjunction with programmable sound generators which normally only output a short pulse coincident with the start of a note. It will also be found useful by less skilled keyboard players since pressing a key momentarily will provide a complete envelope and one does not have to get the sustain period timing correct. It is also applicable to situations where long envelope times are set, since the user will have both hands free to manipulate the synthesiser while the contour is progressing through its cycle.



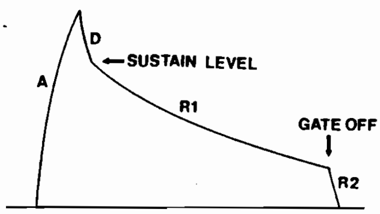
3. AUTOMATIC ENVELOPE



1. 80-18D PANEL

The AUTOMATIC mode is particularly beneficial when envelopes are being initiated from non-keyboard sources, for example, an LFO or the internal clock of the 80-12 Noise Generator/Sample & Hold module. A short pulse will now generate a complete ADR envelope and, by adjustment of the time constants, this type of envelope can be made to approximate the ADSR type, as is evident from Figure 2. Usually these external sources would only generate a limited AD type of envelope.

3. DAMPED. The objective of this mode is to more closely simulate the piano envelope which has a sharp attack, a brief initial decay, a long release and finally a very short release as the damper is applied to the string. This ADRR envelope is illustrated in Figure 4. In this mode release of the key, which is the end of the gate pulse, causes the final release, R 2, to occur. In other words releasing the note has the same action as applying the damper on a piano.



4. DAMPED ENVELOPE

The three timing functions (A, D and R) have ranges between two milliseconds and ten seconds, or more, and are exponentially scaled. The latter results in the most useful time ranges utilising the highest proportion of the associated control potentiometers. The attack voltage rises to +10V and the sustain level is also adjustable from zero to +10V.

The envelope generator has separate gate and trigger inputs and their respective jack sockets are marked G and T on the panel. A trigger pulse is not required to initiate any of the envelope modes but in the NORMAL and DAMPED modes the application of an external trigger voltage while the gate pulse is still present will re-start the attack cycle and thus allow generation of multiple peaked envelopes. The module accepts ground referenced gate and trigger inputs within the range of +3V to +15V. The gate input is of low impedance and thus one should avoid gating more than two 80-18's from an external module, such as the 80-3 VCLFO, whose outputs have a nominal impedance of about 1k0. The low impedance is not a problem for the normal gating sources, i.e., the keyboard via 80-15D2 or the 'ALPHADAC 16' since the output impedance of these is near zero.

In the AUTOMATIC mode very high sustain levels, about 90% or more, may cause the ADR cycle to latch-up in some circumstances. What will happen is the output will stay at the maximum sustain level. If this does occur then simply switch to NORMAL, which will release the cycle, and then back to AUTOMATIC having reduced the sustain level slightly.

The time constants may be trimmed to enable accurate matching of units in a polyphonic system.

The module may be manually gated using the push button marked 'MAN' and this facility is disabled when external gate sources are connected to the 'G' socket.

Other features of the 80-18 concern its use with the ALPHADAC 16 operating in the arpeggiation modes. First, the NORMAL (ADSR) envelope must be selected. More important is the fact that the very short pulses generated in the staccato mode will not re-trigger the envelope generator. The 80-18A should therefore be used for the monophonic voice (split keyboard) when, or if, the staccato effect is required.

MODULE 80-18A
ADSR ENVELOPE GENERATOR

The DIGISOUND 80-18A is a direct replacement for the 80-8 module but with improved quality in terms of control voltage feedthrough. This improvement ensures that the envelope returns to near zero irrespective of the settings used for the ADSR envelope.

The PCB and panel are identical for the 80-18 and 80-18A modules and so the latter may subsequently be converted to a multi-function envelope generator, if required. A jack socket placed in the hole used for the function switch will improve the appearance of the module while in its simplified form.

The gate impedance is 10k but other characteristics are the same as the 80-18 operated in the NORMAL mode. Reference should therefore be made to the description of the latter module.

The 80-18A will be re-triggered in the staccato mode of the ALPHADAC 16 arpeggiation routines.

4

STEP BY STEP GUIDE TO SYNTHESIS

4.1 INTRODUCTION

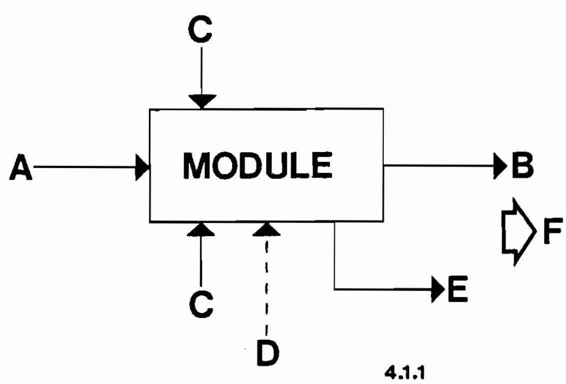
The information presented in this section of the manual is designed to encourage experimentation by the user and in view of this we have omitted actual control settings from our patching diagrams. Thus techniques illustrated are to be used as constructional blocks for the creation of your own sounds or as a foundation on which to build more complex patches.

The layout and format of this section dealing with the applications of modular synthesiser devices proved to be a very difficult task owing to the fact that similar techniques could be included in several of the sub-sections. As a compromise, therefore, we have tried to create a flowing text which takes into account the most basic of patches while each sub-section leads onto more advanced methods. Because of this approach it is necessary to work through the sections sequentially.

Although the layout is unorthodox it is fairly concise. Obviously the entire point of owning a DIGISOUND 80 synthesiser is for its great versatility and so if we were able to list all of the patches, that could vaguely be described as musical, then the versatility of your synthesiser would be very much in question. In other words, it would be a task next to impossible. The information is presented as a step by step guide to the practical understanding of electronic music techniques.

The patching configuration is as follows and illustrated in Figure 4.1.1. All modules will be indicated by rectangles and the audio inputs (signals) will enter the rectangle from the left (indicated by 'A'). Treated signals (B) will logically follow from the right of the rectangle. Control voltage inputs (C) will enter the module from either the top or the bottom while control voltage outputs (E) will follow the same routes. Gate, trigger or other timing signals (D) will be shown as a dotted line if confusion is likely to arise by using a solid line. Arrows obviously indicate the direction of the signal or control voltage. In many instances the patches have been simplified by omitting the output stages which may include some further treatment, at the discretion of the user, and this is signified by a bold arrow(F). Lastly, in cases where more than one type of input (or output) to a module is used then the precise input (or output) will be indicated.

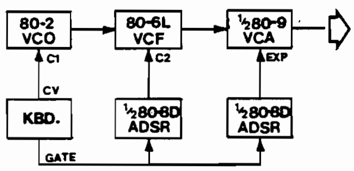
As regards terminology, signals and control voltages will often be referred to simply as signals without distinction. The reason being that with the DIGISOUND 80 synthesiser the two are compatible and so only their intended use determines what they should actually be known as.



4.1.1

4.2 BASIC KEYBOARD PATCHES

As a starting point the DIGISOUND 80 should be patched in the same manner as most 'mini' synthesisers, as illustrated in Figure 4.2.1. Note that the 80-6L may be substituted by an 80-7 in the LP4 mode or an 80-10 may be used for each half of the 80-8D. In the arrangement shown the frequency of the VCO is determined by the keyboard control voltage, which is scaled at 1V/octave. The switch on the VCO should be in the 'off' position, thereby disabling the octaves manual control, while the 'fine' control should be set to zero or tuned to A=440Hz. The



4.2.1

frequency scale can be transposed down by one or two octaves or up by one, two or three octaves by using the octave shifter of 80-15E. Note that two envelope generators are used, ADSR-1 and ADSR-2, one for the filter and one for the VCA. ADSR-1 is connected to the VCF using Control Input 2, which has its own attentuator, and the envelope shape allows the cut-off frequency and hence timbre to be varied during the course of a note. The output of the VCF goes to the AC coupled input on one side of the 80-9 Dual VCA (it is normal to use the AC input for audio signals) and the sound shaped using ADSR-2 patched to the 'EXP' control input. Both envelopes are gated simultaneously when a key is pressed.

Set the patch up as follows:-

i. VCO with square wave output, i.e., pulse output with manual pulse width control (PWM) at setting 5 to produce a 50% duty cycle.

ii. ADSR-1 off by putting Control 2 potentiometer on the VCF fully anti-clockwise.

iii. ADSR-2 set to a piano type envelope: fast attack, high sustain and slow release and one could start with A=5%, D=30%, S=80% and R=90%.

iv. Resonance control on the VCF to about 70% rotation.

Now play a sequence of notes at the correct tempo while adjusting the 'octaves' control on the VCF. When the resultant sound resembles a conventional musical instrument, or something near, then adjust the envelope of ADSR-2 to suit. We can now begin to examine some of the aspects of basic synthesis and this is best achieved by conducting a series of simple experiments. Other keyboard sounds can be obtained by proceeding in a systematic manner.

Modify the patch of Figure 4.2.1. as follows:-

i. Examine the effects of a 'tracking' filter. This is done by taking the control voltage line from the keyboard to the VCF Control Input 1 as well as to the VCO, as shown in the patch. The effect of this is such that the VCF's cut-off frequency is directly related to the frequency of the VCO. The actual difference is set by the coarse (octaves) and fine controls on the VCF and various settings should be tried. Suppose the VCF is set 4 octaves higher than the VCO then since both modules are scaled to 1V/octave the harmonic content of the notes will remain constant.

ii. Connect an attentuator (from the 80-5 Processor) to the line between the keyboard control voltage and the VCF Control Input 1. This will allow the degree of tracking to be varied by the potentiometer. In effect you are now altering the volts/octave relationship to the VCF which will result in a greater amount of harmonics to be present the higher the note played on the keyboard.

iii. Interpose a 'subtractor' (also from the 80-5, refer to module description) in the keyboard control voltage line to the VCF and examine the effect at various attentuation levels. You will find it to be the reverse of the situation described in (ii).

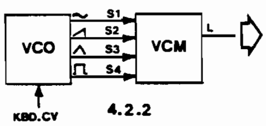
iv. Assess the influence of ADSR-1. Special attention should be paid to the level of the envelope, as determined by the Control Input 2 attenuator, since high settings may cause the filter to exceed its dynamic range. In other words the harmonic content may not change for some of the higher notes if the filter is also tracking. Examine different cut-off frequencies of the VCF, with and without the filter tracking and also with different levels of envelope control. Also experiment with different shapes from the ADSR starting with simple AD contours and progressing to full ADSR envelopes. Normally the ADSR-1 envelope will not exceed the duration of the ADSR-2 envelope otherwise only part of the formers contours will be effective.

v. Carefully examine the effect of the VCF's resonance control. As the control is rotated clockwise a point is reached when the filter breaks into oscillation (not the case if you are using an 80-7) and the point immediately before it does so provides a very harsh but pleasant electronic 'ring' to the sound. This feature will be put to use later in this section.

vi. Examine the effect of pulse width control of the VCO. A characteristic 'phasing' sound will be heard as the pulse width (PWM) control is turned.

vii. Repeat experiments (i) to (v) using different waveforms from the VCO. Generally the sawtooth waveform will be found most useful while the sinewave should exhibit poor response since there are only weak harmonics (there would be none in a pure sine wave) for the filter to extract.

viii. Try mixtures of waveforms, using the 80-4 Voltage Controlled Mixer (VCM) as illustrated in Figure 4.2.2, to alter the resultant timbre of the note going to the VCF. The mixtures are simply obtained by manually varying the levels of each of the individual shapes.

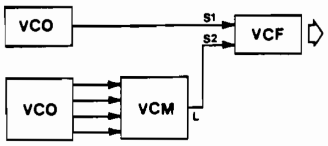


4.2.2

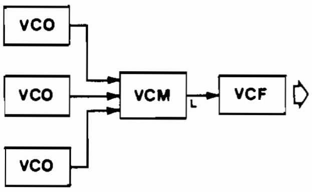
Each of the above steps should initially commence with the patch at its original starting point but one soon learns how to arrive at the initial values of the controls. Each of the steps should also be tried in conjunction with one another and other possibilities should be evident, such as, using the 80-5 'subtractor' on ADSR-2. Similarly, the 80-5 'subtractor' inverts the phase of a signal - what happens when a waveform (try different ones) goes direct to the VCM and is then mixed with the same signal after passing through the subtractor? The object of this programme is to familiarise yourself with the basic keyboard patch and the influence of all the principal controls used in the creation of a basic sound. If you are new to synthesis then spend plenty of time on this section.

The above exercises will yield some potentially useful sounds and reference should always be made back to the original patch in Figure 4.2.1 if in doubt when applying the techniques to follow. We now look towards ways of perfecting the basic sounds and introducing greater dynamic control over their quality. The advantages of the modular approach in terms of the ability to add or interpose other modules will soon become apparent.

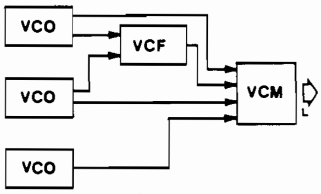
4.3 ADDITION OF PARALLEL MODULES



4.3.1



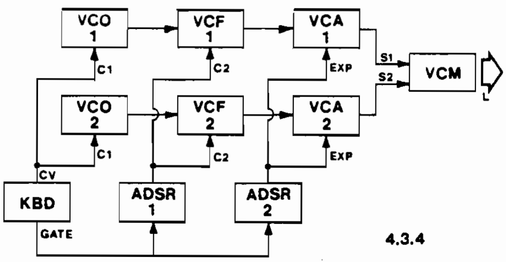
4.3.2



4.3.3

One of the most important ways of adding to the resources of a synthesiser is to add more VCO's to enrich the sound. Two VCO's controlled by the same keyboard voltage could be taken to Signal Inputs 1 and 2 on the VCF and tuned together in unison, near unison, or to any interval such that the beat frequency is not obtrusive. There are various others ways we can connect additional VCO's and a few of these are illustrated in Figures, 4.3.1, 4.3.2 and 4.3.3. As you can see, the versatility of the modular approach to synthesis is becoming apparent and the patching possibilities very numerous. Each of the examples demonstrate often subtle but noticeably different effects.

The concept can be extended a stage further by duplication of modules. In the next example (Figure 4.3.4) the same ADSR's are used to control both 'voices' and variation in the envelope amounts and the filter cut-off frequencies can be adjusted, as previously described, to alter the timbre of each voice. Note that to realise this example the Processor module is being used as a 'patchcord splitter' and for such a basic function it is not shown in the block diagram.



4.3.4

Other techniques, similar to those outlined previously, can also be added to this patch. As an aid we offer a few experimental guidelines:-

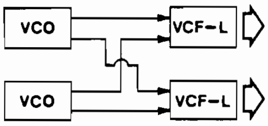
i. ADSR-1 to control VCF-1 and VCA-2.

ii. One voice, or only the VCF or the VCA, is inverted in phase with respect to the other, i.e., a 'subtractor' sub-module is put into the control line to the module(s) in question.

iii. Filter tracking.

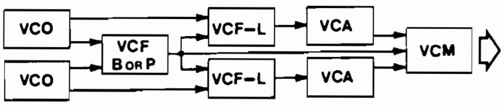
iv. Bring in unfiltered sound from the VCO's directly to the other channels of the VCM.

v. Taking another waveform from each of the VCO's to the opposite filter as illustrated in Figure 4.3.5.



4.3.5

Adding a third filter, Figure 4.3.6, further increases the possible effects obtainable from a single patch. Note that in these last two patches we have not shown the ADSR's, etc. and they should be extended as in Figure 4.3.4.



4.3.6

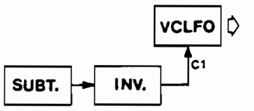
Again combinations of earlier examples can be applied here and a staggering range of possibilities may be effected. None of the patches are, however, modulated in any way other than from the envelope generators controlling the filters. There are three main types of modulation we require to study before we can make an in depth study of sound synthesis. The three are frequency modulation (FM), pulse width modulation (PWM) and amplitude modulation (AM) and are discussed in the following sections.

4.4 FREQUENCY MODULATION

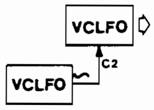
To demonstrate frequency modulation start with the basic patch shown in Figure 4.2.1 and connect the triangular output (the +10V output will give more meaning to your settings) from an 80-3 VCLFO to the Control Input 2 of the VCO. Experiment with different frequencies of modulation, different modulating waveforms and different settings of the Control 2 attenuator.

It will be noticed that increasing the amount of modulation with Control 2 attenuator determines the 'swing' or amount of deviation from the set frequency. Also experiment with temporarily putting the switch(es) on the VCO(s) to the ON position to bring in the use of the coarse control. Variation of the coarse control will alter the frequency which is being modulated. An obvious point you may say but now progress by altering the VCO frequency in such a way that the deviation from VCO frequency to peak modulation frequency is one octave, or more, by careful adjustment of Control 2 on the VCO. A square wave VCLFO output may yield the best results. Now tune in a second VCO in the same way, perhaps even using different VCLFO's for each VCO and modulated at different rates. A very pleasing rich sound will normally result and one must continue to experiment in this way to fully utilise the possibilities that exist.

The effective frequency range of the VCLFO may be altered by application of a negative voltage to the Control Input 1. The 'subtractor' is a source of positive voltage (refer to 80-5 module description) and when passed through the 'inverter' becomes a negative one. This may then be used to manually extend the VCLFO's frequency, see Figure 4.4.1.

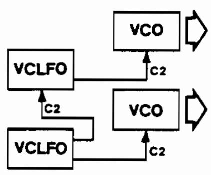


4.4.1

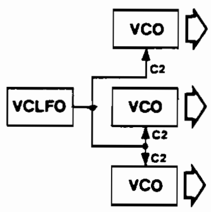


4.4.2

Another interesting technique to explore is modulating the modulator as shown in Figure 4.4.2. The concept may obviously be extended to modulating the modulator which modulates another modulator! Remember it is both the subtleness and richness of these effects which makes the modular synthesiser such a powerful instrument. Pay special attention also to the use of high modulating frequency settings since very complex and unusual effects can be obtained owing to the frequency harmonics of the modulating source being 'superimposed' on the main audio signal, especially at high levels of modulation. Figure 4.4.3 shows a simple patch that can yield some unusual sounds using two VCLFO's. Note once more that the large arrow indicates that the patch would normally be extended in the usual way with the keyboard control voltage going to VCO Control Input 1, the output going to a VCF and onto a VCA and both of the latter modules controlled by ADSR's gated from the keyboard. Figure 4.4.4 shows a simple method of adding a pleasing richness to the sound from two or more VCO's, with or without the ranges of modulation being present as discussed above. In this case gentle modulation works best which is achieved by adjustment of the Control 2 attenuator.

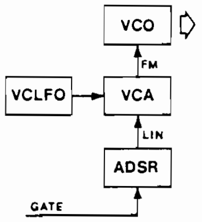


4.4.3

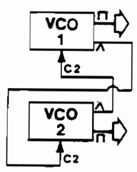


4.4.4

In the above examples the modulation has been applied to Control Input 2 which is scaled at one volt per octave or proportions thereof according to the setting of Control 2 potentiometer. The 80-2 VCO's also provide for linear modulation and the input and associated attenuator is marked 'FM'. Using this input will result in a linear change in frequency with applied voltage. This input has some special uses. First by using an envelope generator (ADSR plus VCA) the modulating waveform may be shaped. This is shown in Figure 4.4.5. A more usual application is the arrangement of Figure 4.4.4 and if the VCO's are set to different intervals then a type of chorusing effect will be obtained since the tracking of the oscillators has been affected. The FM input should be explored but unless specifically stated frequency modulation (FM) in the rest of this text will refer to modulation of the exponential input (C1 or C2).

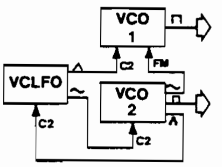


4.4.5

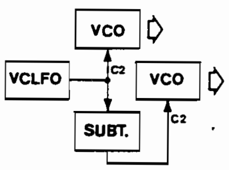


4.4.6

Cross modulation techniques are now quite popular but the effects are quite complex and the following patches should be studied with care so as to determine what is happening. Cross modulation is achieved when we patch the output of module B to the input of module A and then the output of module A to the input of module B. One such patch is shown in Figure 4.4.6. Surprisingly the output waveforms are stable since the modules settle into a complex equilibrium. Figure 4.4.7 is an extension of the method and note the use of the FM input here. When these patches are controlled by the keyboard (to the VCO's) the resultant sounds, at different pitches, are often of totally different timbres and sometimes do not seem to have any relation to each other.



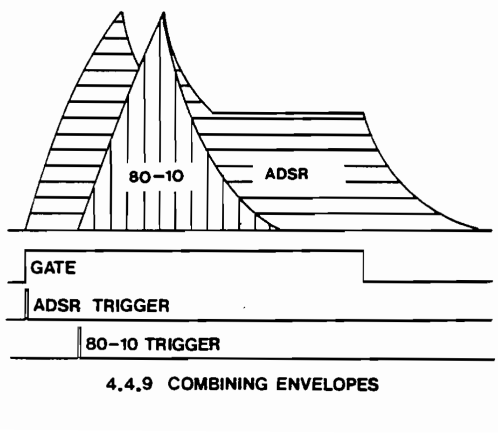
4.4.7



4.4.8

Frequency modulation, however, need not confine itself to automatic repetition of sounds, such as vibrato which is frequency modulation at around 7Hz, or involve itself in complex cross modulation patches. Some very simple, yet effective, ways of modulating the frequency of a VCO may be achieved with minimum effort. Take, for example, Figure 4.4.8. Instead of both VCO frequencies rising one is going up as the other goes down. At slow modulation rates the effect is more noticeable, especially if the modulation and the VCO's are 'tuned'. Some novel variations of this can be implemented when a keyboard is introduced by applying a subtracted version of the keyboard control voltage to only one of the VCO's leaving the other in the normal mode. Bring in cross modulation to the patch and you have a rather unpredictable keyboard!

Frequency modulation may be accomplished in many ways, one simply looks around the synthesiser for sources of control voltages which may be useful for such purposes, e.g., the External Input or the Sample & Hold as will be discussed later. One of the most useful control sources is the ADSR envelope shaper and only a few synthesists seem to realise its application beyond the conventional VCA and VCF shaping techniques. Patch an ADSR output to the Control Input 2 of the VCO and gate the patch, as usual, from the keyboard. The frequency will follow the ADSR contour with the range of frequency being adjusted by the Control 2 attenuator on the VCO. The classic, now cliche, drum synthesiser sound can be imitated by using an ADSR patched in this way. A fast attack and a moderate decay with the sustain level set to zero will result in a 'one-shot' type of contour which is similar to the drum synthesiser sound. Since the sustain level is zero the position of the Release time control is irrelevant. By applying this AD one-shot to a VCO, VCF and VCA the patch is complete. Experiment with the resonance control. As the filter goes into oscillation, or approaching same, at high Q settings (high resonance) a secondary tone will merge with the original tone. The frequency of the secondary tone, which is a sine wave, may be adjusted using the filter frequency controls. In fact the filter alone can provide the sound when it is used in the high Q mode. Experiment with different ADSR contours, inverted and subtracted contours, and the mixing of two, or more, contours. The re-trigger or delay modes on the 80-10 VCEG can be used to advantage in these situations. Figure 4.4.9 illustrates the latter and the same combination may be used in many other applications of envelope contours. In fact this dual peak type of contour is often more realistic when simulating the sounds of specific instruments. Although

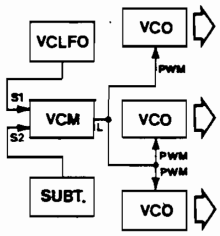


4.4.9 COMBINING ENVELOPES

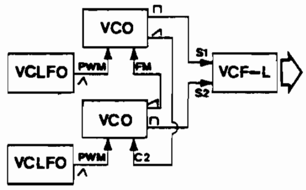
the example is a simple one, that may be approximated using the re-triggering facility on the 80-10 VCEG alone, the use of two or more ADSR contours, with the addition of delay and re-triggering when required, can be used to generate unusual and complex contours. To obtain a better feeling for the contour shape you are generating it is best to set them up using a VCO rather than a VCA since small changes in frequency are easier to detect than small changes in amplitude. At some stage you should go back to the earlier sections and evaluate these contours in some of the more useful patch settings that you have evolved.

4.5 PULSE WIDTH MODULATION

Pulse width modulation (PWM) is a well known technique but surprisingly many synthesisers, including very costly units, do not provide the facility. The DIGISOUND 80 VCO and VCLFO are provided with both manual and external pulse width control which allows adjustment of the pulse width over its full duty cycle. At zero setting (PWM Control at 0 indicates zero percent duty cycle) no sound will actually be produced from the pulse output. As the PWM control is turned clockwise the pulse wave gets wider and becomes a square wave (50% duty cycle) at setting 5. Further rotation widens the pulse until at full rotation the output is virtually a DC voltage. As mentioned in the section concerned with the basic keyboard patch a very pleasant phasing sound will be produced as the pulse width control is turned. The flexibility of the facility is, however, more fully realised by automatic control of pulse width. Patch the DC triangle wave from a VCLFO, set to about 2 or 3 cycles per second, to the PWM socket of the VCO. The PWM Control potentiometer is now disabled. The phasing effect will now be automatic but there will also be times when there is no sound output, albeit momentarily. The reason is that since the triangle is ramping between 0 and 10 volts the pulse width output is cut off at these two extremes, as described for the manual adjustment. This can be used to advantage, especially at high PWM frequencies when a form of amplitude modulation will be superimposed on the output. The effect is quite different from other types of amplitude modulation discussed in the next section. For more conventional use of PWM, however, it will be necessary to lift the 'floor' of the modulating waveform above zero to provide a smooth and uninterrupted pulse width modulated output. This may be accomplished using a positive voltage derived from the 80-5 'subtractor' sub-module and adding it to the modulating waveform of the VCLFO using the voltage controlled mixer. The arrangement allows the level of modulation to be precisely adjusted over any useful range of values. The patch may then be extended by taking the output from the VCM and gently pulse width modulating three or more VCO's, possibly adding attenuation (from the 80-5) to control the level of PWM to each VCO. The patch is shown in Figure 4.5.1 and it may be used to achieve a very warm choral effect.



4.5.1



4.5.2

An unusual type of modulation occurs with the patch of Figure 4.5.2. We have assumed the use of two VCLFO's although in many cases the effects will apply if only one is used. Alternatively, if only one VCLFO is used then the method may be explored using the patch shown in Figure 4.5.3. For these experiments use the VCO's in the free-run mode, that is, without using the keyboard and with the octave switch on the VCO's to the ON position to allow manual adjustment of their frequency. The VCO's are cross modulated, one VCO using the linear FM input and the other using the exponential input via Control Input 2. Both are modulated by sawtooth waves. Certain rules should be applied to obtain the best effects:-

i. Pulse width modulate the VCO's one at a time and use high frequency settings of the VCLFO's such that the output seems to suddenly lock and produce a smoother tone than that obtained at other settings. The VCO's should initially be tuned to within a few hertz of each other and also set to fairly low audio frequencies. Note that no positive offset is applied to the PWM control from the VCLFO's since such a step is unnecessary at high frequency settings.

ii. Fade in the frequency modulation to each VCO, one at a time, until similar locking effects are obtained.

iii. Next adjust the cut-off frequency of the VCF to obtain the best sound.

There are several other aspects of the patch which should be observed:-

4.5.2

i. A very powerful sound results which carries a great deal of weight at the bass end of the spectrum.

ii. The resultant waveform is of an extremely complex nature.

iii. The success of the patch largely depends on the settings of level for both the pulse width and frequency modulation.

iv. Each VCO should produce a different timbre from the other and by mixing at the filter stage yet another timbre is developed.

The patch may now be experimented with in the following manner:-

i. Exchange linear to exponential inputs and vice versa on each VCO thus altering the cross modulation.

ii. Only use the linear control input (FM) for cross modulation of frequency.

iii. Only use the exponential input (C2) for cross modulation.

iv. Experiment with different waveforms from the VCLFO's which are being used for PWM.

v. Experiment with different waveforms for cross modulating and also different VCO outputs to the filter.

vi. After carrying out the above steps, and combinations thereof, choose a few of the most interesting and examine the influence of the frequency control settings on both the VCLFO's and the VCO's.

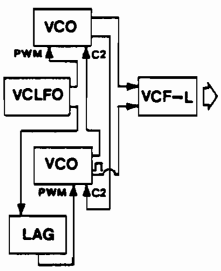
vii. Add a positive voltage offset to both PWM inputs, as described earlier.

viii. Cross modulate the VCLFO's.

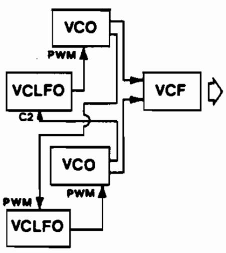
There are numerous other ways in which the basic patch of Figure 4.5.2 may be configured. Figures 4.5.3 to 4.5.6 show some variations that should be tried and notes made of the effects obtained.

Figure 4.5.6 illustrates a method for using the VCM module to control the heart of the effect and their variations which is achieved by alteration of the input levels and panning controls. Another variation is the substitution of a VCM for the VCF in the initial patch of Figure 4.5.2 and also taking the VCLFO outputs to the final audio mix.

Study these patches carefully and try adding some of your own. There are several reasons for introducing such complex patches at this stage: 1. As an introduction to complex patching. It is a difficult task and one which requires a great deal of practice if the 'spaghetti' networks are to produce interesting or even musical sounds. 2. They teach that precise setting of controls is necessary in the creation of useful effects and as a result one learns to be patient, which is essential for good creative synthesis. 3. They teach method and progression. One soon learns the best way of quickly arriving at initial settings of controls and which modules to bring into the patch and in which order. 4. They encourage experimentation and after a while a similarity in some of the effects will be observed. At this stage, therefore, one should be beginning to form a sound picture in the mind as to how the various degrees of complexity can be produced with the minimum of effort and how the addition of another module will affect the result.



4.5.3



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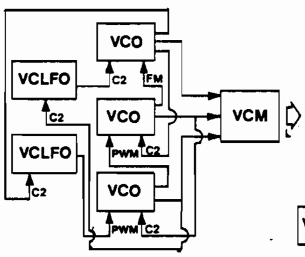

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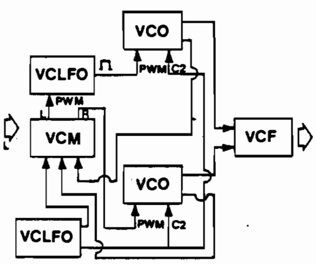


4.5.6

4.6 MODULATION OF FILTERS

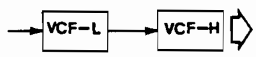
Many of the characteristic sounds of the synthesiser are largely due to filtering. In the following discussion we will make use of the four types of filter offered in the 80-6 series, namely, low pass, high pass, band pass and phase shift. In most cases the 80-7 state variable filter may be substituted for the 80-6 filters, for example, LP4 on the latter gives the same response as the 80-6L. Also if using the 80-7 filter one should examine the difference obtained with the two pole outputs. In the following examples the filter will be designated by VCF followed by suffixes -L, -H, -B and -P denoting low pass, high pass, band pass and phase shift respectively.

The most common type of filter used in synthesisers is the low pass type and when a filter is unspecified, i.e., merely written as VCF, then the filter type in nearly all synthesiser texts will be a low pass version.

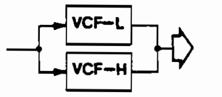
Refer back to the basic patch of Figure 4.2.1. As shown the most widely used method of modulating the filter is by the introduction of an envelope contour from an ADSR unit which varies the cut-off frequency of the filter as the note progresses and finally decays. Introduction of the keyboard control voltage, or a portion of it, produces a sound with more dynamic character and is a technique which should be widely used. The keyboard control voltage is normally injected into Control Input 1, which has no attenuator, and is available for filter tracking purposes. If the full degree of tracking is not required then the keyboard voltage may be attenuated by using Control Input 2 and its associated potentiometer or if this input is being used for another purpose an 80-5 attenuator to Control Input 1 may be used.

In the first series of experiments, evaluate the various filter types in the basic keyboard patch and study the effects of using different levels of envelope and resonance. Also introduce filter tracking as discussed above.

Refer to the patch in Figure 4.6.1. The addition of a high pass filter to the basic keyboard patch, in series with the low



4.6.1



4.6.2

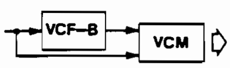
pass filter, is the only modification required. A little thought here will reveal that if the cut-off frequencies do not overlap then the arrangement creates a band pass filter where the width of the band can be made very wide. Experiment with controlling these two filters as if they were one by tracking them and also modulating them from a single envelope generator. Unusual timbres can result if the envelopes are introduced in different proportions to each filter.

Figure 4.6.2 shows the same two filter types in parallel, which effectively creates a band reject (notch) filter. Repeat the series of experiments applied to Figure 4.6.1 and pay particular attention to the cut-off frequencies of the filters and also to the resonance control, all of which will influence the sharpness of the notch.

The use of the filter combinations described above offer greater synthesis possibilities than the use of a band pass or notch filter alone. The reason being the ability to vary the width of the pass or reject bands.

It will be obvious from earlier discussions that the VCLFO may be used to modulate the cut-off frequencies of VCF's and for this purpose the waveform should be connected to Control Input 2 so that the attenuator may be used. Try different waveforms and also the effects of differing amounts of modulation to each filter in the examples of Figures 4.6.1 and 4.6.2 when configured within the basic patch. Some very interesting and useful results should emerge.

The band pass filter may be used to enhance some intermediate frequencies of a waveform. Most conventional musical instruments have certain natural resonant frequencies and to simulate this effect these frequencies may be boosted using band pass filters, usually more than one being required for realistic simulation. The effect is obtained by setting the band pass filter to a frequency coincident with the resonant frequency of the particular instrument being simulated and combining the resultant signal with the unfiltered signal in a VCM, or at the input of another filter, and subjecting the output to normal processing via a VCA etc. Figure 4.6.3 shows the use of a VCM for the patch. Adjustment of the levels of signals at the mixing stage will determine the degree to which the resonant frequency is being boosted and the controls allow complete adjustment from resonant frequency only (band pass output) to no resonant frequency (untreated signal). Increasing the Q factor of the filter by turning the resonance control clockwise will effectively reduce the bandwidth of the frequencies enhanced as well as providing additional boost. Generally moderate to high levels of Q are most effective but any tendency to oscillation should be avoided in this situation.



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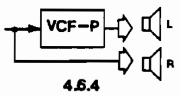


4.6.4

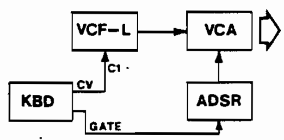
As inferred earlier, the latter technique may be extended by using two or more band pass filters in parallel to produce two or more resonant peaks which can be made to track the keyboard and produce a type of voltage controlled resonator.

A similar type of patch may be used for the removal, or attenuation, of some unwanted intermediate harmonics. One simply removes the band pass filter in Figure 4.6.3 and substitutes it with the band reject patch of Figure 4.6.2 or by using the 80-7 filter in the notch mode.

Again we can experiment further with these patches and most usefully with the resonant frequency boosting type. Start by examining the effects obtained from envelope shapers, keyboard voltages and VCLFO's to shift the resonant peak. Next take portions of the envelope waveform, keyboard voltage or VCLFO waveform through an attenuator or 'subtractor' and thence to the appropriate control input for external voltage control of the signal levels entering the VCM. The result of the latter experiment will be dealt with in more detail later in this manual but one should see, or rather hear, that many more possibilities of dynamic control are available as soon as we begin to modulate the mixing levels, hence the timbre, of the treated sound. As a primer on dynamic control methods, apply the techniques of VCLFO, keyboard voltage or envelope output to the mixing levels of the patches illustrated elsewhere in this manual, in particular to those shown in Figures 4.3.3, 4.3.4, 4.5.1, 4.5.5 and 4.5.6. These techniques, however, are really a form of amplitude modulation which is dealt with in greater detail in the next section.

The 80-6P phase shift filter has two deep notches in its output signal. The phasing effect will be obtained by modulating the filter. It is better to have more control over the proportion of treated and untreated signals and to take the outputs to a stereo pair in which only one side has the phasing effect, see Figure 4.6.4. An 80-6P filter in series with the 80-7 in the notch mode will produce an even greater effect. Additional experiments with this combination are applying different modulation rates to each filter using the ADSR or VCLFO waveform outputs.

As mentioned in the description of the modules, the 80-6 filters can be made to resonate, i.e., to break into oscillation at high Q (resonance) settings. We can therefore add another technique to our steadily increasing repertoire of sounds by making use of the filters ability to oscillate. Figure 4.6.5 shows a patch which does not require a signal to be supplied to the VCF. The 80-6L works best in this application. The effect being produced is to shock the filter into oscillation and the resultant sine wave is shaped by the ADSR connected to the VCA in the usual manner. The frequency of the sinewave output is determined by the keyboard control voltage plus the setting of the coarse and fine controls on the filter. The resonance control should be set fully clockwise. If we now mix in a little of the envelope voltage via the Control Input 2 on the filter we again obtain the cliche drum synthesiser sound.



4.6.5

All of the filters have provision for external control of resonance. Look back over your experiments and choose those where the setting of the resonance control had a useful effect. These should now be repeated and the resonance controlled using voltage controlled sources such as VCLFO, keyboard voltage or ADSR output. If the filter has provision for simultaneous manual and external control of resonance then remember that the manual control will determine the initial Q of the filter and to begin with this should be set to zero. Furthermore the modulating voltage should have an attenuator in-line and the 80-6 filters may be arranged so that the manual control is disabled by insertion of a jack plug and the potentiometer for the manual control may then be used for the external control voltage.

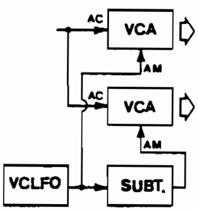
4.7 AMPLITUDE MODULATION

Amplitude modulation is quite simply the modulation of the level of a particular signal. In a small synthesiser the tremelo effect, which is amplitude modulation at about 7Hz, is usually confined to applying modulation from a low frequency oscillator to a control input on the VCA - although many lack even this basic facility. This is a sad omission since we will demonstrate that amplitude modulation is a very powerful tool which may be used in a large number of situations. The 80-9 Dual VCA has a 0 to 100% amplitude modulation control which therefore allows a wide range of control. The usual method of amplitude control applied to the VCA does have one draw-back, namely, that the whole sound envelope is modulated. The latter effect is far from a natural tremelo which does not commence immediately nor is constant in frequency for the duration of the note. We will limit ourselves to discussion of the various applications of amplitude modulation and to outlining a few methods of transforming the usual boring and unrealistic synthesiser tremelo effect into a more interesting sound. Some of the effects are quite subtle but when used in a musical piece it is often the small changes which provide character and interest to the music.

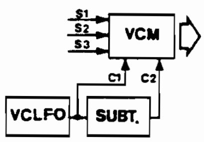
The first variation in amplitude modulation arises when the signal from a VCA (one half of an 80-9) is amplitude modulated by a (VC)LFO and the output taken to another VCA (the other half of the 80-9) where it is amplitude modulated by another LFO at a slightly different frequency. Envelope shaping of the input signal is carried out in the first VCA in the normal manner via the EXP input. If necessary, the two modulating frequencies may be derived from a single LFO if the Lag Processor (80-5) is used in the control path to the second VCA. Sine or triangle waves are normally used for AM purposes but experiment with other waveforms, their combinations, and with different modulating frequencies. Remember that the 80-9 requires a 0 to 10 volt input to the AM control.

Another interesting effect is split phase tremelo in which the same signal source goes to separate VCA's (both halves of the 80-9). The VCLFO is connected direct to VCA-1 while for VCA-2 the LFO waveform is phase inverted by the 80-5 'subtractor' sub-module. The two outputs are taken to separate channels of a stereo amplifier. The arrangement is shown in Figure 4.7.1. Envelope shaping can be applied to both VCA's in the normal manner or by using another VCA on the signal input before it is split.

In the essence the 80-4 VCM module is a quad VCA and in the previous section we saw how it could be used to alter the timbre of a sound by additive synthesis techniques. This approach will now be extended. If a number of signals are being mixed in the VCM then amplitude modulation of one or more of the signals can be obtained via the appropriate voltage control input. This will result in the creation of new timbres depending on the patch used. Envelope shapers may be used for the modulating voltage or indeed a proportion of the keyboard control voltage which will then give a degree of mixing in direct relationship to the notes being played on the keyboard.

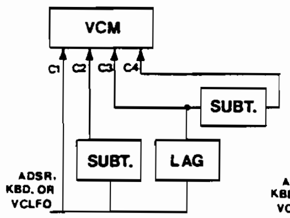


4.7.1



4.7.2

The following examples of amplitude modulation may be applied to any previous patch shown in the manual where the VCM has been used. Figure 4.7.2 is a variation of the split phase tremelo patch since the mixing levels of signals 1 and 2 will rise and fall in opposite phase. Signal 3 (and 4 if used) is unaffected in this situation. The keyboard voltage could be substituted for the VCLFO in this patch, or used on the other two channels. Figures 4.7.3 and 4.7.4 carry this a stage further by using envelope contours initiated from the keyboard. Subtractors, lag processors and attenuators may be used to advantage in these patches which are in addition to the attenuators on the input signals which allow for different initial signal levels. The signal input paths to the VCM are omitted for reasons of clarity.



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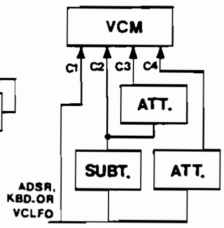
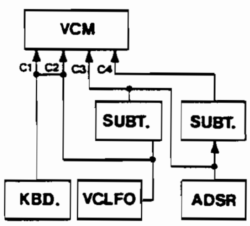


4.7.4

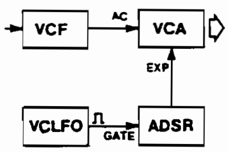
Figure 4.7.5 shows a patch which uses three modulating voltages simultaneously. In practice you will require attentuators in the control lines since the maximum control input to the VCM is +10 volts and combination of VCLFO with the keyboard control voltage or the ADSR will exceed this level. Reverting to Figure 4.7.4 the response to control voltages and hence the signal outputs from the four channels will be as follows:-

MODULATING VOLTAGE	C1	C2	C3	C4
Low (0V)	Low	High	Mid	Low
High (10V)	High	Low	Low	Mid

The versatility of this technique should now be apparent and the variations are almost limitless. Having given the guidelines you should experiment with further combinations.



4.7.5



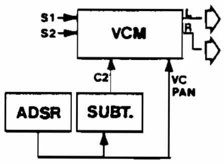
4.7.6

The 80-10 VCEG allows all four of the A, D, S and R functions to be controlled by external voltages. By applying a VCLFO to the external sustain control via one of the attenuators on the 80-10 it is possible to amplitude modulate the signal only during the sustain phase of the envelope. This is very effective in playing modes which use a long sustain period since the tremelo only occurs during the sustain. Variations of this can be made by referring to Figure 4.4.9 which would allow the amplitude modulation to occur, say, only during the period of the second envelope. Additionally, although not a part of amplitude modulation, is the ability to alter the time constants of the 80-10 VCEG by applying external control voltages to the appropriate control input. The most obvious application is to make, say, the attack or release time proportional to the keyboard control voltage although some very strange envelopes can be realised by using different VCLFO modulating frequencies at each of the time constant and sustain inputs.

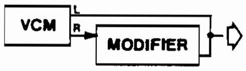
Percussive effects are also a form of amplitude modulation which use a repeating envelope contour in place of a simple waveform. In Figure 4.7.6 the ADSR unit is gated automatically from the pulse output of a VCLFO. The pulse width in this application determines the gate 'on' time, hence the time before release occurs. Short percussive envelopes work best, such as, simple AD contours without sustain. The patch may be extended by applying a proportion of the VCLFO or ADSR voltages to the filter control inputs. Furthermore, if the 80-10 is used in this technique then the time constants may be subjected to modulation, as indicated above.

In all the applications where a VCA is used with an ADSR the resultant envelope may be partly buried, thereby reducing the output level, by applying part of the keyboard voltage to the VCA's AM or Linear control inputs. Another result of burying the envelope is to reduce the release time. The latter can be put to good use since although one can still have a long release time it will sound sharper since only the fastest part of the exponential decay is heard. Another method of obtaining automatic control of loudness is to use the voltage controlled panning facility of the VCM. If only one output is taken from the VCM then the amplitude of the mixed signals is proportional to a voltage applied to the VC pan input. Similarly, if the other output of the VCM is used with the same panning control voltage (the pan control must be either fully clockwise or anti-clockwise) then the amplitude of the mixed signals will be inversely proportional to the modulating voltage.

By using both outputs from the VCM the result is, of course, automatic panning where the signal moves from the left output to the right output and vice versa in time with the modulating voltage. This is an effective technique but one which should be used sparingly. To provide variations of the technique we could patch the keyboard voltage to the VC pan input so that the sound moves, say, from left to right as the keyboard is ascended. This is an excellent way to increase the 'sound stage' over which the synthesiser will perform in a stereo set-up and the technique should be thoroughly explored. A similar effect, although a more pronounced one, will occur if an ADSR contour is employed. What we are aiming to do is point out some of the many possibilities. We do not suppose that you will ever wish to play a conventional tune in which the sound is panned around by an envelope shaper but for an original composition or effect it may give just the results you are seeking. A further development of panning is a situation in which the signals 'appeared' and 'disappeared' as the mix is panned from one channel to the other. In Figure 4.7.7 the input signal 2 can be made to fade out and in as the envelope voltage dies away and the sound has travelled from left to right and back again. Removing the 'subtractor' from the patch causes the reverse of the situation regarding signal 2. Unusual situations arise if the modulating voltage is derived from a VCLFO run at fairly high frequencies.



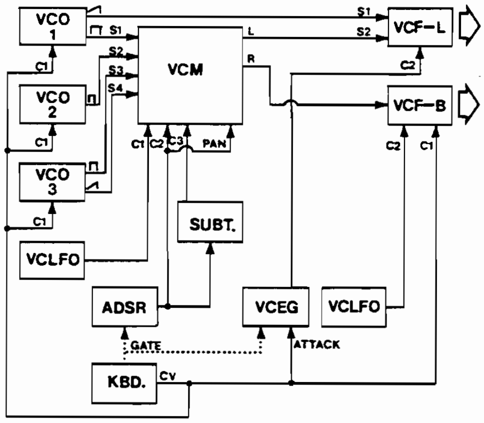
4.7.7



4.7.8

Since the VCM has two outputs, left and right, and the signal mix can be panned between them we can use this facility as a means of providing voltage control or timbre. Each output may be taken to a separate modifier chain and processed individually and by voltage control of the panning the resultant signal will demonstrate a more dramatic change in timbre than has been described so far, especially when the envelope generator is used as the control voltage. The sound can then be made to totally change in quality as it is shaped. Figure 4.7.8 shows the basic arrangement of this voltage controlled timbre modulator and the MODIFIER is any means of signal modification from simple filtering onwards.

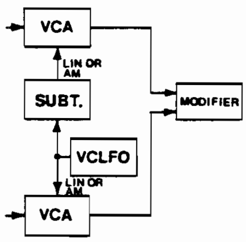
The patch of Figure 4.7.9 combines a few of the suggestions made this far. As an exercise work out what is happening in the patch as soon as a key is pressed and over the duration of the envelopes. Do this first without reference to the synthesiser and then check your conclusions with an audible demonstration. Adding final envelope shaping to VCA's added to both outputs will yield a fully configured patch.



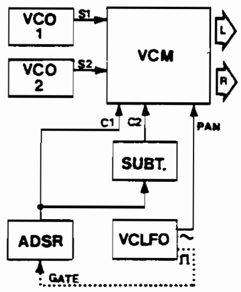
4.7.9

As indicated, the VCM is a particularly useful module since it may be used in many different roles. Since, however, amplitude modulation of the outputs provide the panning effect described it is the entire mix which is subjected to the overall panning effect. Referring back to Figure 4.7.9 we can see that VCO-1 sawtooth output is by-passing the VCM to remain fixed in the sound stage. This approach of taking one, or more, signals which by-pass some treatment stages is an extremely important one to recognise.

Panning, of a type, may also be accomplished with the patch of Figure 4.7.10. In essence this is a split phase tremelo patch but using two different sound chains instead of the same signal feeding each VCA. The VCLFO may be replaced by an envelope generator or, to lesser effect, by the keyboard control voltage. This opposite mode from panning is known as SEQUE and its use should be apparent.



4.7.10



4.7.11

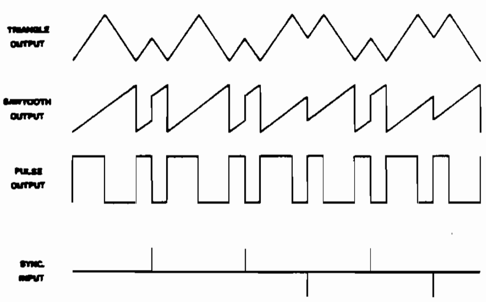
All of the examples may, of course, be expanded by combining them with earlier techniques. They serve to illustrate the variety of sounds that can only be obtained with a modular synthesiser and which will prove beneficial to the user once the basic techniques have been mastered. Figure 4.7.11 is another patch incorporating some of the techniques discussed above.

4.8 SYNCHRONISATION

When two waveforms of very nearly the same frequency, or multiples thereof, are mixed together a periodical inter-amplitude modulation of the sound occurs. This effect is known as 'beating' and can serve to increase the richness of the sound or can, on the other hand, be a totally undesirable component of the mixed waveforms.

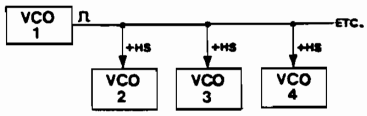
Synchronisation of two or more oscillators, set to ratios to produce complex waveforms, is required when the beat frequency has to be eliminated from the sound. The effect of synchronising oscillators is to lock the harmonic relationships between two, or more, together so that the combination sounds more like one oscillator with a very complex waveform. This effect must remain when the frequencies of each oscillator are altered simultaneously, for example, from the keyboard. The technique may, however, also be used in its own right to produce some pleasing, or unusual timbral effects. It will be evident that synchronisation applies to the use of two or more VCO's but the technique may be explored by using a VCLFO/VCO combination especially when the LFO is set to higher frequencies.

The effect of HARD SYNCHRONISATION is illustrated in Figure 4.8.1. A positive going synchronisation pulse will cause the triangle waveform to reverse direction only during the rising portion of the triangle whereas a negative going synchronisation pulse will cause reversal only during the falling portion. The effect of the synchronisation pulses on some other waveforms is also illustrated and since the sine wave is derived from the triangle wave in the VCO it will also influence the sine wave shape in a complex manner.



4.8.1

POSITIVE HARD SYNCHRONISATION may be implemented as illustrated in Figure 4.8.2. VCO-1 (or a VCLFO for experimental purposes) is referred to as the master oscillator and the pulse output from this is connected to an attenuator on the 80-5 Processor module. The four outputs from the attenuator allow up to four oscillators, known as slave oscillators, to be synchronised by connection to the +HS input on the VCO's.



4.8.2

Additional oscillators may be synchronised from the same master by using one of the attenuators outputs connected to the input of another attenuator to increase the number of distribution outlets. Although Figure 4.8.1 shows the synchronisation pulses as spikes the VCO's respond to the positive going edge of the pulse and so the duty cycle of the actual pulse is not critical. The 80-5 attenuator is used solely for distribution purposes and although synchronisation may occur at lower settings the potentiometer should be set fully clockwise at the start. Some experiments with synchronisation should now be conducted:-

i. Examine various frequency ratios of master and slave oscillators and note the influence that synchronisation has on the harmonic content of the output. It may be best to start with one slave VCO and then add others, if you have them. The frequency of the slave VCO's should be higher than that of the master VCO otherwise the slave(s) will simply follow the frequency of the master.

ii. Repeat step (i) using the keyboard to control the VCO's.

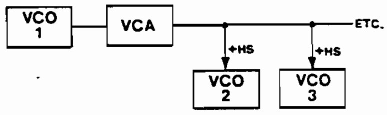
iii. Repeat step (ii) but with the keyboard voltage only connected to the slave VCO(s).

iv. Note the effect of frequency modulating the slave VCO(s) since this technique is capable of yielding some very pleasing timbral effects.

4.8.2

It is evident from Figure 4.8.1 that NEGATIVE HARD SYNCHRONISATION (-HS) may also result in the generation of complex waveforms and in many instances it will be impossible to audibly discriminate between the two hard synchronisation techniques. To obtain negative hard synchronisation, the pulse output from the master VCO is connected to an 'inverter' on the 80-5 module (refer to module description) prior to being distributed further, if required, by an attenuator and then connected to the -HS input on the slave VCO's. It is possible to use both positive and negative hard synchronisation simultaneously but this will require two master VCO's set to different intervals. The latter arrangement, as well as the use of other sources of synchronisation pulses, may be interesting but their complexity is such that they are unlikely to be rewarding in the early stages of experimentation.

SOFT SYNCHRONISATION causes premature reversal of the waveforms from the slave VCO's with the result that their oscillation period is an integral multiple of the pulse period of the master VCO. Soft synchronisation (SS) requires negative going pulses (positive pulses should be avoided entirely) and so may only be implemented in the same manner as described for negative hard synchronisation. For soft synchronisation, however, the attenuator on the 'inverter', or the distribution attenuator, should initially be set mid way and then gradually increased, if necessary, until synchronisation occurs. The nominal pulse amplitude for SS is -5 volts maximum. When setting up for this harmonic locking effect the master VCO should be tuned for the correct interval and the fine control adjusted to an exact frequency lock, which will be audibly evident. It will be obvious that turning the attenuator control on the Processor anti-clockwise will stop the synchronisation from taking place. We can develop some simple but effective techniques from this ability to suddenly take VCO's in and out of synchronisation. In Figure 4.8.3 a simplified diagram is shown of a basic patch for +HS and it may easily be converted for -HS or SS by using the methods outlined above. The VCA may be amplitude modulated by another pulse to take the VCO's in and out of synchronisation, thus providing some unusual effects. Alternatively, an envelope voltage contour may be applied to the VCA so that synchronisation only occurs for part of the contour. The best



4.8.3

type of envelope for the latter is one with near minimum attack time and high sustain levels (to effect the sync.) and thus the VCO's come out of synchronisation as soon as the release phase commences. It will also be best to have a fairly short release time. This effect must, however, be precisely set up in the overall patch otherwise the results will be jittery as soon as the key is released. It is nevertheless a useful approach for obtaining automatic control of the synchronisation function and it should be examined.

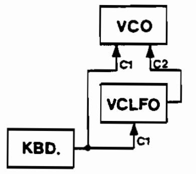
It should also be borne in mind that the VCLFO's modulation frequencies can be synchronised in the same way as a VCO and as a final experiment in this section apply the synchronisation techniques to the cross modulation patches described earlier in this manual. It may also be applied to other patches where multiple VCO's and VCLFO's are used.

4.9 FURTHER DYNAMIC CONTROL METHODS

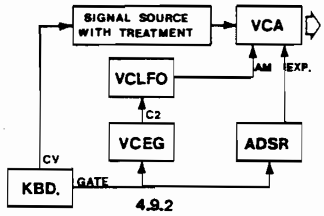
We have already explored several ways of altering the periodic and tiresome nature of some electronically produced sounds. In this section we introduce additional ways in which properties of a sound can be varied with time and the emphasis will be on the keyboard to provide the dynamic control methods.

At this stage, we should take a fresh look at situations in which use is made of a VCLFO. The patch of Figure 4.9.1 shows how the keyboard voltage is also used to control the frequency of the VCLFO. This quite simply means that the modulation frequency of the VCO is entirely dependent on which key is depressed, e.g., the higher the note played on the keyboard the higher the rate of modulation. The use of this technique contributes a great deal to creative synthesis, more than will initially be apparent. For the first series of experiments in this section we should apply the technique to patches and situations described earlier in the manual, paying particular attention to the patches shown in Figures 4.4.2, 4.4.3, 4.4.4, 4.4.7, 4.4.8, 4.5.2, 4.5.3, 4.7.1, 4.7.2, 4.7.3, 4.7.9 and 4.7.10. These experiments should be repeated using an envelope generator gated from the keyboard to provide the control input to the VCLFO. As an example of a working situation Figure 4.9.2 shows a patch which provides a more realistic simulation of the tremelo effect previously mentioned in the manual. No explanation of the patch should be necessary since it is merely a combination of techniques which were introduced earlier. It should, however, be noted that the 80-10 VCEG is in the delay mode and the built-in timer set such that an AD envelope is generated coincident with the end of the attack period of the other envelope generator (ADSR). The AD envelope from the VCEG is obtained by setting the sustain level to zero, as previously described, and the envelope is also used to vary the frequency of the VCLFO during the remaining portion of the note. The reader should also now be in a position to modify this patch for himself and so enhance its effect.

Experiments can also be tried using a VCO modulating another VCO, e.g., by patching the keyboard control voltage to the 'normal' signal VCO and a portion of it



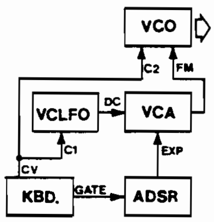
4.9.1



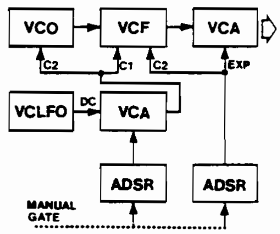
4.9.2

to the modulating VCO via Control Input 2. This will yield somewhat similar effects to the cross modulation patches described earlier except that in this case somewhat strange, though still quite tuneful, sounds will occur without too much wandering from the keyboard related notes.

Since the Voltage Controlled Amplifier (½ of 80-9) can be DC coupled further methods of dynamic control are available using this module.



4.9.3



4.9.4

Examine the patch of Figure 4.9.3. The arrangement is a technique known as 'dynamic depth frequency modulation'. The title really explains the action. The amount of VCO modulation present varies in proportion to the envelope contour. The technique may also be used for modulation of filters, PWM, etc. and again we should examine some of the previous patches in the manual and apply Dynamic Depth Modulation wherever applicable. To provide a variation, use can also be made of the keyboard control voltage to alter modulation levels in a more subtle way by patching to the VCA AM input(s). The X-Y Controller may also be utilised in several situations where a subtle change in frequency or depth is required, or indeed the initial pulse width of the VCO may be set using the keyboard control voltage such that the higher the note the wider the pulse. To provide a final example, a patch is shown in Figure 4.9.4 which may be used for the creation

4.9.2

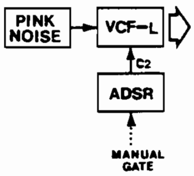
of a sound which is best described as a space war weapon! This is simply the extension of the previous patch. The release time of ADSR 1 must be shorter than the release time of ADSR 2. The manual gate may be simply performed by pressing the two manual gating buttons on the ADSR units simultaneously or by patching in the keyboard gate and pressing the key. The patch will result in modulation of the signal mostly during the period when the gate is 'on'.

4.10 THE USE OF NOISE IN ELECTRONIC MUSIC

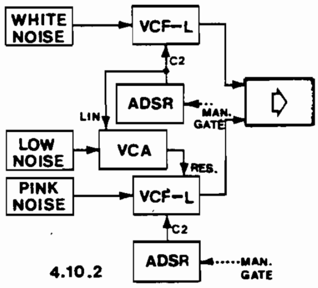
We have now reached the point in synthesis where we need to create sounds other than musical ones and one of the most important modules for the production of such sounds is the Noise Generator. The sounds of electronically created wind and howling gales, utilising a Noise Generator, are to be found on hundreds, if not thousands, of albums and therefore we will attempt to progress a stage further and use this module for other applications. We must first learn, however, to create the basic sounds.

The effect of wind may be created simply by patching the white noise output to the input of a voltage controlled low pass filter. Simple manual control of the cut-off frequency of the filter is all that is required to generate a wind sound. Careful setting of the resonance control on the 80-6L filter can produce a 'whistle' in the audio output which enhances the effect. The best setting of this latter control is where the filter is just about to break into oscillation. Substituting white noise with pink noise will create a sound of deeper intensity, more like the sound of the sea. We may experiment with these basic patches by adding the keyboard control voltage to the filter's control input and so effectively 'play' the wind or sea. Adding an envelope generator to the filter's control input while still using the keyboard voltage can, if careful attention is paid to envelope shape, result in a breaking wave and this may be brought in when required by manually gating the envelope generator.

Figure 4.10.1 shows a simple patch which may be used to simulate gunfire. Ricochet may be added by turning up the resonance control until the filter actually goes into oscillation during the sweep time. Longer envelopes may be used to simulate explosions. Now try the following refinement. Patch the low noise output to the external resonance control of the filter. The result is a very powerful deep rooted modulation that greatly enhances the rumbling of the explosion. You should also examine the use of this low noise modulation technique with some of the cross modulation patches described earlier in the manual since some very impressive effects are possible. Figure 4.10.2 is an extension of



4.10.1



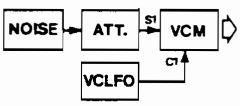
4.10.2

the previous patch which allows the low noise to be faded out of the filter as the envelope progresses. The manual gating buttons should be used for this application.

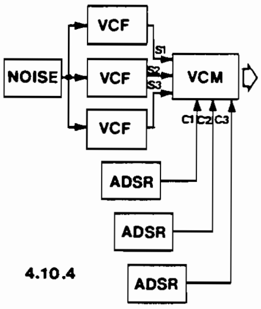
Next experiment with the direct injection of noise to the VCF control input, via the attenuator, which will result in a strange and often 'nasty electric edge' to the sound. It will be noticed that when the low noise source is used in this way a very pleasing random filtering effect will be given to the sound, especially with high levels of attenuation so as to avoid saturation of the control input. The reader should also revert to some of the earlier patches in the manual and experiment with the injection of noise into a control input of a VCO. Now try treating noise through various types of filter, including the lag processor on the 80-5 module, and, in particular, the 80-6P phase shift filter if available. Also mix in some conventional audio input, i.e., a normal synthesised sound. Make a note of the results obtained for future reference. Next revert to the introduction of noise to the control input of the filter when the signal input of the filter also comes from the noise generator. The use of an 80-6P filter with a pink noise input and low noise for control of frequency and/or resonance will produce some interesting results.

Automatic sweeping of filters using the contours from VCLFO's or ADSR's may be used effectively with noise inputs. Figure 4.10.3 shows an initial patch for the automatic creation of surf sounds. As an exercise, try perfecting the patch using some of the techniques discussed earlier. The following suggestions should help:-

i. Interpose a lag processor between the VCLFO and the VCM to modify the



4.10.3



4.10.4

transition in sound which occurs when the sawtooth wave reaches its peak amplitude.

ii. Add the low frequency noise source to Signal Input 2 of the VCM and adjust the sound level of this signal manually. The use of low noise provides a background rumble, often heard with strong seas, and it is necessary to obtain a good balance between the intensity of the two noise sources now being used.

iii. Instead of a sawtooth waveform use an envelope voltage to control the sound contour of the noise sources. The envelope generator may be gated manually or from the keyboard. The time constants should be set to provide a long attack, fast decay and a fairly long release time so that the latter simulates the effect of the receeding waves. A medium sustain level should be used. The conventional envelope gives a fast initial build up of sound but slows as the attack proceeds. An improved contour would be one with a slow build up and rising to a crescendo as the wave breaks. This type of contour may be obtained using the 80-10 VCEG in the following manner:

a) Connect the output of the 80-10 to an attentuator on the 80-5 module with the potentiometer fully clockwise since it is to be used solely for distribution purposes.

b) Take one output from the attentuator to the '-' input on Attentuator 1 of the 80-10 and from 'A1' to the external attack control 'A'.

c) Adjust Attentuator A1 and the attack control potentiometer to produce a concave attack response of desired shape and time.

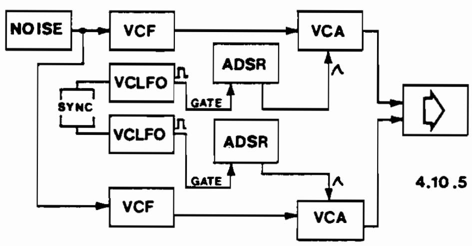
d) If waves of greater ferocity are to be simulated then the white noise may be high pass filtered. This filter should be directly after the white noise source and not on the final sound since the latter arrangement would result in the low frequency noise being filtered out.

e) Try the patch substituting the white noise with pink noise.

The above demonstrates the method of approaching sound synthesis. That is, building up a picture of the sound in your mind which in the above case is one of gradually increasing intensity of sound as the wave approaches and the fairly sharp decay as the wave crashes on the shore. Thus initial sawtooth shaping of white noise can give a reasonable imitation but this may be greatly improved by slight modification to both contour and nature of the sound source.

A further exercise is the development of the patch shown in Figure 4.10.4. Also experiment with variations in the technique to produce a series of sound effects which may prove useful at a future date.

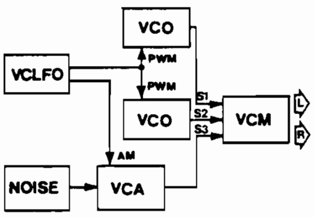
Noise is also very useful for creating percussive effects and may be tried using noise as the signal in the patch of Figure 4.7.6. By using synchronisation techniques to ensure exact timing relationships this patch may be expanded into dual, triple, quad, etc. 'voice' and the frequency of one VCLFO may be set to produce timing pulses which are integral multiples of another, thereby producing a basic rhythm. The example in Figure 4.10.5 shows a two voice percussion patch. Synced VCLFO's are used and indicated by the markings on the diagram and, for clarity, it is not normal to show the type of synchronisation used. A moments thought, however, will reveal that Soft Synchronisation (SS) will work best here. Refer back to Section 4.8 if in doubt.



4.10.5

The noise source may, of course, be replaced by two VCO's to provide 'pitched' percussion using the above patch or even one VCO and a noise generator. Conduct some experiments with percussion techniques and extend your ideas into multi-voiced patches if resources allow.

Noise may also be used to simulate the sounds of certain engines or other pieces of machinery and the resultant effect may be used on its own or added to another synthesised sound to provide a background for the final mix. Automatic control of the noise generator is usually required for such applications and for purely 'mechanical' sythesis the use of ADSR units may often be omitted. The example in Figure 4.10.6 uses a VCLFO to 'open' and 'close' a VCA to provide amplitude modulation of the noise signal. Use is also made of the ability to achieve a 0% duty cycle (zero output) of the pulse waveforms on the VCO's thus causing the sound to be 'off' for part of the VCLFO waveform cycle. The three sound sources may be mixed in any proportions in the VCM, after which further treatment may be applied.



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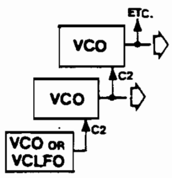


4.10.7

It will be observed that certain patches, at some settings of the controls, will produce an output which may be referred to as 'noise' in spite of the fact that periodic events are occuring within the complex waveform of the output. Such patches may be used in place of a noise generator and are quite effective for some sounds, for example, 'mechanical' type of sounds. It is generally advisable to use as few modules as possible to create this 'periodic noise' since they may be required for some subsequent treatment of the effect. A band pass filter, finely tuned, works well in this application since often only certain parts of the sound are required at the output and may provide the entire effect that is sought.

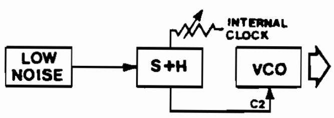
As a starting point for other patches which will create a complex output try the patch illustrated in Figure 4.10.7 with high settings of both frequency and modulation depth.

The nature of a true noise signal consists of a mixture of all audio frequencies with white noise having equal power at all frequencies and pink noise equal power per octave. This fact alone may be used to create totally unpredictable events and some of these are applied in the next section dealing with the Sample & Hold section of the 80-12 Noise Generator.

4.11 APPLICATIONS OF SAMPLE & HOLD

In the previous section we examined the use of a low frequency noise source to provide some random events but these had a severe limitation, namely, that they could only be altered in terms of level. The principal use of the 80-12 Sample & Hold unit is the production of random voltages which, to a certain degree, may be manipulated by the user so that, for example, their speed may be altered at will. By having a module which produces random voltages at useful rates we can utilise the facility to provide a means for the synthesiser to play itself automatically.

The basic patch is shown in Figure 4.11.1. The low noise from the same module is connected to the S&H input, the output from the S&H to a VCO and an appropriate waveform from the VCO direct to an amplifier. A series of discrete pitches which are random in frequency are produced and their tempo is determined by the setting of the internal clock. The mid frequency of the output is set by the coarse control on the VCO while the total range of pitch is adjustable with the attenuator on the S&H input or the Control Input 2 attenuator on the VCO, or both. The effect is psychologically very powerful and in a more simplified manner this is generally the only application of Sample & Hold in a small synthesiser and thus has become a well-worn product of electronic music.



4.11.1

The random voltage from the Sample & Hold unit may also be used to control the frequency of a filter. The 80-6L low pass filter works best here especially when the initial frequency (set by the coarse control of the VCF) is set to almost totally filter out the audio input. Furthermore, by carefully increasing the resonance the filter will begin to oscillate and the result is a series of random sine wave tones accompanying the modulated audio frequency. If the step voltage change from the S&H is too obtrusive for a particular application then the random voltages from the S&H output may be taken via the 80-5 lag processor to 'soften' the edges of the sharp transitions without interfering too much with the overall effect. The latter is similar to a portamento control.

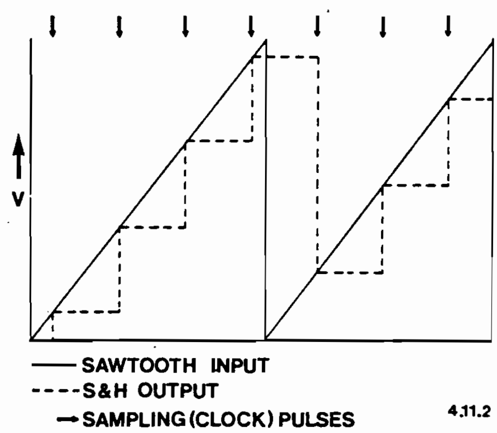
We should now experiment with various applications of random voltages and the following provides some guidelines:

i. Try the pink and white noises as the material for sampling.

ii. Apply filtering to the noise sources prior to sampling, with emphasis on the band pass filter which will reduce the bandwidth of the source.

iii. Introduce random frequency control to VCLFO's with particular attention being paid to patches that utilise some form of dynamic control. Random, or random frequency, pulse width modulation is another technique which should be fully explored.

In a modular synthesiser, however, the scope of application of the S&H unit is virtually unlimited. The S&H input will accept signals from virtually any source and these may be pre-recorded material, suitably amplified by the 80-13 External Input module. Generally, however, the result of sampling most external material is still to produce a series of random pitches. A more useful signal source for the S&H is the sawtooth waveform from a VCLFO. The latter will produce discrete pitches which are gradually rising in scale until the peak of the sawtooth is reached after which the stepwise scale repeats, although not necessarily with identical frequencies. The illustration of Figure 4.11.2 makes the general principle clear.



4.11.2

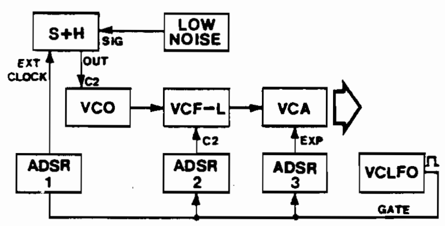
4.11.2

At low input frequency several notes may be obtained during one cycle of the sawtooth waveform and by adjustment of the S&H input (or Control Input 2 on the VCO) the steps may be of widely varying frequency intervals. As the frequency of the sawtooth is increased one begins to obtain arpeggiation effects and then more complex sound patterns are created as the frequency of the sawtooth exceeds that of the clock. Note that at low frequencies the 'scale' of pitches may be made to rise in an exponential manner by interposing the 80-5 lag processor between the VCLFO and the S&H input. Another effect is to modulate the VCLFO which is generating the sawtooth so as to produce changing patterns of 'scales' and 'arpeggios'. This may even be extended a stage further by taking a proportion of the random voltage back to the control input of the VCLFO used to modulate the VCLFO being sampled. A note of the effect should be made.

The S&H unit may also be used with an external clock, for example, the pulse waveform from a VCLFO. To obtain the discrete frequencies referred to above it is necessary to set the duty cycle of the pulse output near to its minimum. This limitation on pulse width has been purposely designed into the module in order to increase the range of application. For example, when sampling a sawtooth waveform if the duty cycle of the VCLFO clock pulse is increased then a gliding (portamento) effect between notes will be obtained and the degree of this effect is related to the duty cycle, the VCLFO sampling frequency (clock rate) and the frequency of the sawtooth being sampled. To increase variety the pulse width may be automatically modulated so as to produce a variation from 'clean' notes to notes which are slewed to various degrees. Increasing the sampling time by increasing the pulse duty cycle can also be effective when sampling other signal sources.

Thus far the effects have been at a uniform tempo using either the internal clock or a LFO. Since, however, the LFO is voltage controlled this restriction may be avoided. A simple means of achieving variation in tempo is to modulate the clock VCLFO with another VCLFO which will cause the clock to speed up and slow down in a controlled manner. Alternatively the output from the S&H module may be connected to the Control Input 2 of both the VCO (to generate the pitches) and the VCLFO (to vary the clock speed). In this patch a high output from the S&H will effect both a high pitch and at the same time speed up the VCLFO so that it takes the next sample faster.' The effect is therefore more musical since the higher the note the shorter the duration of the note. Clearly one may also obtain the opposite effect by interposing an 80-5 'subtractor' between the S&H output and the VCO and/or the VCLFO control input.

In the above patches the S&H directly drives a VCO which is connected to an amplifier without any intermediate modification to timbre or shape. Having created sounds using the keyboard programme described earlier the user may wish the Sample & Hold unit to generate some of these sounds automatically. To accomplish this the arrangement would be similar to Figure 4.1.1 but with the keyboard replaced by the S&H unit. The main shortcoming, however, is a lack of control over the duration of the gate pulses which determine the sustain period of a note. The clock output from the S&H unit, or the pulse from a VCLFO, may be used as an independent trigger for the 80-10 envelope shaper but they should not be used for gating this module without first attenuating them to +5V (unless the 80-10 has been modified to accept higher gate voltages). Either type of clock may,



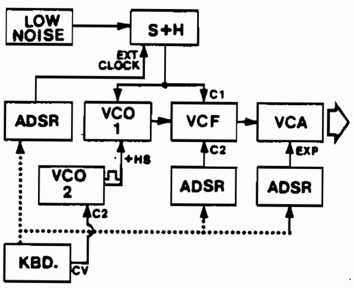
4.11.3

however, be used to gate the 80-8 envelope generator directly. Owing to the short duration of the clock pulses in normal operation they are only suitable for percussive (AD) type envelopes. This limitation is overcome with the patch of Figure 4.11.3 in which the pulse width from the VCLFO may be varied over its range to generate the gate time for the 80-8 ADSR's 2 and 3. The controls on the 80-8 ADSR 1 are all set to minimum so as to generate a sharp pulse for sampling which is independent of the pulse width of the VCLFO. In this arrangement the output from the VCO may be treated to generate a variety of sounds as described in Section 4.2. It will be evident that practically all of the techniques so far described for the S&H and for the keyboard generated sounds may be applied to this patch. A further possibility is to use the S&H output for PWM of the VCLFO clock.

In the patch of Figure 4.11.3 the VCLFO may be replaced by the keyboard. In this situation the pitch is still determined independently from the signal being sampled but the tempo and duration of notes is controlled by the player. Various combinations of keyboard and S&H are worth exploring and a useful combination is shown in Figure 4.11.4 in which the keyboard gate output is used to control the envelope generators and the keyboard control voltage is employed to control the frequency of a second VCO. This latter VCO is used as a master oscillator providing positive hard synchronisation pulses to the VCO whose frequency is being controlled by the S&H. Adjustment of the various control inputs will result in a situation where just pressing the same key will generate a series of tones whose pitch is unpredictable but are nevertheless musically quite pleasing. This musical effect is enhanced by playing the keyboard in the conventional manner. The best results are obtained when the VCF is tracking the VCO and for this arrangement the S&H output goes direct to Control Input 1 on both VCO and VCF while the pitch spread is set by the attentuator on the S&H input. We can simplify the technique illustrated to provide a method of deriving a random voltage to accompany each note played. Figure 4.11.5 shows a typical keyboard patch which provides a random accompaniment to modulate the VCF and VCLFO.

Methods of limiting the range of the S&H output should now be familiar and these should be applied to the patch of Figure 4.11.5. The basic technique of random vibrato can easily be deduced and the method can be used for a variety of situations and not just with the dynamic depth frequency modulation shown.

One of the most interesting patches for the S&H is illustrated in Figure 4.11.6 which allows the creation of a wide variety of rhythmic patterns of a quality which is normally only available using a sequencer. Start by manually adjusting the frequency of VCO 1 slightly higher than VCO 2. Adjust the S&H clock for a rhythm tempo and adjust both input level to S&H and Control Input 2 on VCO 2 to give a good dynamic range while avoiding very low and very high frequencies. All of these adjustments have a major effect on the resultant sound. What happens in this patch is that sometimes the S&H output is taking the frequency of VCO 2 above that of VCO 1 and producing a variety of timbral effects through synchronisation whereas at other times VCO 2 frequency is lower than that of VCO 1 and so the output is determined by VCO 1. The resultant sound is a series of changing frequencies but with some of the notes also changing significantly in timbre. The sound may be made more rhythmic by replacing the low noise signal with a sawtooth waveform, with or without slewing by the lag processor. With the latter the result is an 'easy-to-listen-to' rhythm which may be used as a background track. Because of the interesting nature of the output this apparently simple patch can consume several hours in learning how to obtain a variety of rhythms. This patch may obviously be extended, e.g., modulation of the sawtooth and treatment of the VCO 2 output as described earlier.



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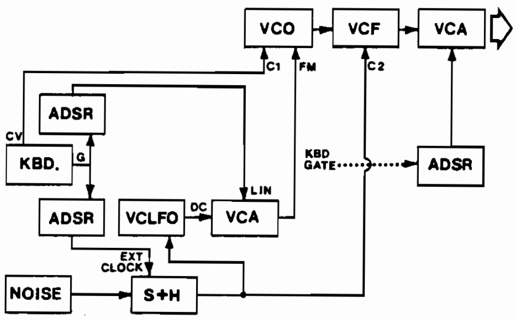

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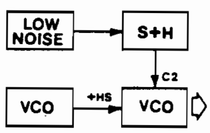
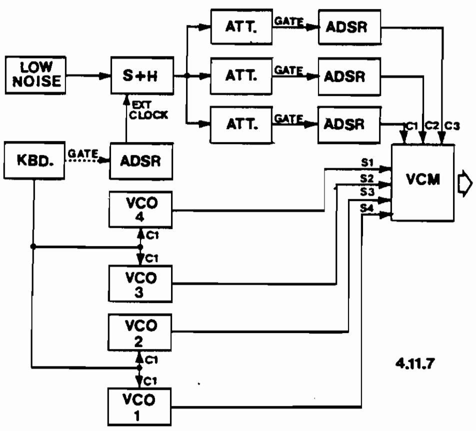


4.11.6

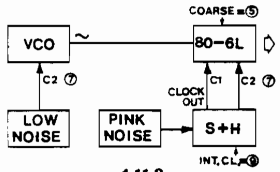
A more advanced technique of random voltage applications is shown in Figure 4.11.7. Again with this patch a great deal of patience is required to obtain useful results. The Sample & Hold is clocked by the keyboard in the manner previously described. The output from the S&H must be restricted so that some of the random output voltages will fall below the minimum voltage required to gate the ADSR envelope shapes. Thus attenuators have been placed in the lines, as illustrated, to facilitate different voltages being required to gate each ADSR. As may be seen from the patch, a normal note will first be sounded followed by either VCO's 2, 3 or 4 and each different note will yield a different combination. Each VCO may, of course, be treated by filtering and so on prior to entering the VCM and this can result in some very interesting and unusual effects.



4.11.7

This basic idea may be used without the random influence by using the keyboard via a 'subtractor' (to generate higher voltage levels) and the voltage from the 'subtractor' used in the same way as the voltage from the S&H in the patch of Figure 4.11.6. The normal (non-subtracted) keyboard voltage would be used to control the VCO's in the same way. This patch will result in a lesser amount of VCO's being brought into use the higher the note on the keyboard. For the best results with this technique one should avoid playing slow pieces for the simple reason that somewhere on the keyboard two notes a semitone apart will be the difference between one or two VCO's being sounded. The latter occurs for three different points on the keyboard. For slow pieces the technique of using four different ADSR/VCA combinations (if four VCO's are being used) is much simpler to set up since the attack times only need to be adjusted for the fading in of different VCO's (or voices). The VCM may even be used to perform the task of the four VCA's thus offering a saving on module count. The ADSR units are gated from the keyboard in the normal manner.

Figure 4.11.8 shows the S&H in a 'working' patch. Note the use of the clock output of the S&H. The clock output, being a narrow pulse, may be used as a trigger input for DIGISOUND 80 envelope generators but in this application it is being used to 'blip' the filter each time a sample is taken. The VCO is set to maximum frequency and the sine wave output is utilised. Normally this will not filter well but in this patch the frequency



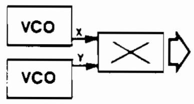
4.11.8

modulation via the Control Input 2 adds sufficient harmonics for the desired effect to be created. Note also that the settings of the control potentiometers have been included and are indicated by a number within a circle. The resonance control of the filter should be set to the point where oscillation is about to take place. The initial frequency of the VCF is set to pass only very low audio frequencies. This patch is reminiscent of a 'water-drop' effect at low to moderate sampling rates. Increasing the frequency of the clock and also the cut off frequency of the VCF will give an effect similar to running water.

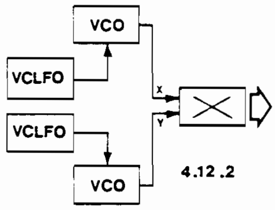
4.12 RING MODULATION

The ring modulator may be used with two input frequencies, X and Y, and its output is the sum and difference of these two frequencies, that is, X + Y and X - Y. With sinewaves the resultant outputs are well defined. When one or both of the inputs is, however, a waveform with a high harmonic content then the resultant output becomes extremely complex. The first experiment is illustrated in Figure 4.12.1 in which the ring modulator module is denoted by the multiplication sign in the box. Try all combinations of waveform outputs from the VCO's including the $\pm$5V sine and triangle outputs. Next try controlling one of the VCO's from the keyboard after setting the rest of the patch up in the same way as Figure 4.1.1. The result of these experiments will normally be a series of heavily modulated sounds ranging from deep gong effects to less harsh chimes. It will also be noted that in many instances only a small change in frequency will have a pronounced effect on the quality of the sound.

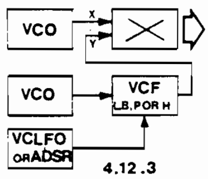
Experiments should progress in a logical manner and Figures 4.12.2 to 4.12.6 illustrate some basic variations which should be further developed. Note that Figure 4.12.3 allows for altering the harmonic structures, hence the ring modulation effect, of the input to the ring modulator. The patch of Figure 4.12.6 will hold certain frequency ratios constant and a further development (phase locked loop) is shown in Figure 4.12.7.



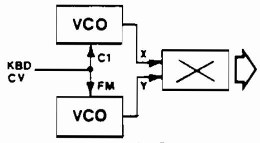
4.12.1



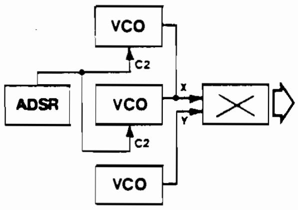
4.12.2



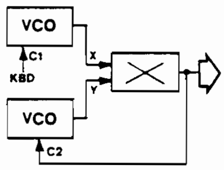
4.12.3



4.12.4



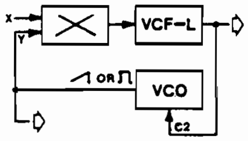
4.12.5



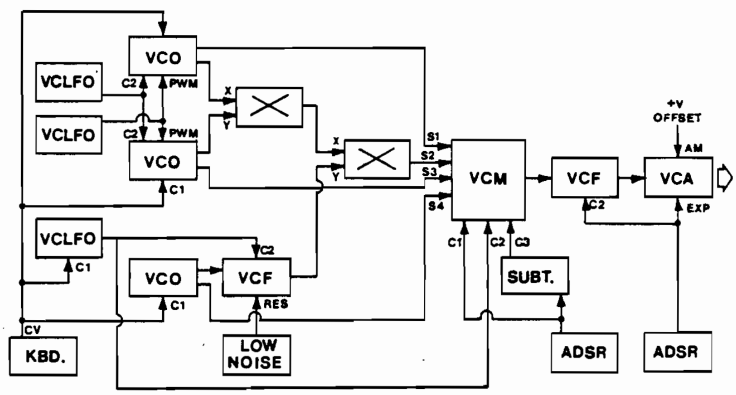
4.12.6

A patch which has been used in a working situation is shown in Figure 4.12.8 and this illustrates the extent to which patch development may be taken.

Returning once more to the dual percussion patch of Figure 4.10.5 we now show a variation (Figure 4.12.9) which may be used to provide a wide range of ring modulation effects. Note that the ADSR's should be adjusted so that there are always two signals being applied to the ring modulator. This patch, in

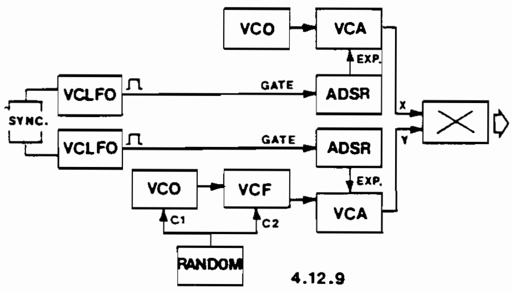


4.12.7



4.12.8

4.12.2

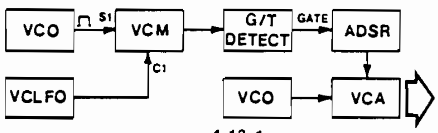


4.12.9

common with many others, requires patience in setting the controls in order to achieve the desired effect. There are, of course, many simpler ways of obtaining a percussive type sound with different timbral qualities on each 'beat'. The advantage of the method illustrated here is that the random steps may be synchronised to one of the LFO's so that the timbre changes will be initiated after a certain number of 'beats'. The patch yields some unusual results.

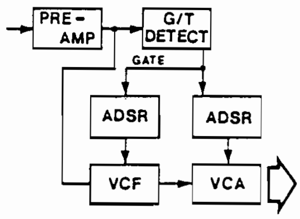
4.13 THE EXTERNAL INPUT

Rhythmic patterns from the synthesiser for use as background tracks are, as already inferred, normally obtained by using a sequencer which controls both the tempo and VCO frequency. In the absence of a sequencer, or the ALPHADAC 16 computer controller, one may use the Sample & Hold module as described in Section 4.11 or the External Input module. The results from the latter modules are, however, very limited in comparison with the aforementioned controllers. A patch using the 80-13 module is shown in Figure 4.13.1. In this patch the VCLFO is amplitude modulating the output from a VCO in the 80-4 VCM to form a well defined series of peak voltages which are subsequently detected by the GATE/TRIGGER input of the 80-13 External Input module. The derived gate pulses are connected to an 80-10 VCEG for shaping the sound output from VCO 2 in the VCA. VCO 2 could be a sawtooth output from VCO 1 but this reduces flexibility.



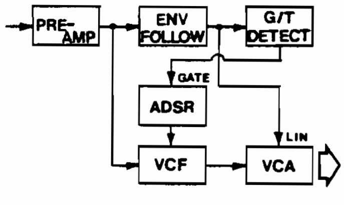
4.13.1

The range of rhythms may be extended by modulating the VCLFO with another VCLFO, or a VCO at low frequency. Tone shaping may be accomplished by interposing a low pass VCF between VCO 2 and the VCA and the centre frequency of the latter varied with a triangular waveform from the VCLFO. This latter VCLFO may be the modulating VCLFO illustrated but again using the same module for two purposes obviously imposes limitations on the variety of sound obtainable. The 80-10 VCEG can make use of both the gate and trigger outputs when its function switch is in the DELAY mode but the effectiveness depends on the degree and variety of amplitude variation going to the GATE/TRIGGER extractor of the 80-13 module. Furthermore one has to ensure that the gate and trigger pulses are adequate for driving the VCEG and this is indicated by the status LED's on the 80-10. If necessary the input to the 80-13 should be adjusted via the gain control on the VCM.



4.13.2

To use the 80-13 for interfacing with external equipment one should use a patch of the type shown in Figure 4.13.2. This patch works best when the input material has a series of well defined peaks and it allows for filtering and envelope shaping of the input signal, or other voltage controlled modifications that may be required. When the peaks are difficult to extract from the source material then a modification to the patch will be necessary. Figure 4.13.3 shows one alternative which takes the external input through the envelope follower and so smooths out the waveforms amplitude variations. The output from the envelope follower may be used to directly control a VCA (linear input) since the DC voltage obtained is directly proportional to the amplitude of the external input signal. Depending on the nature of the input material there may be a residual voltage mixed with the envelope which prevents the VCA from completely cutting off but this is easily remedied by partially 'burying' the envelope, as described in the 80-9 VCA description notes. The output of the envelope follower also goes to the GATE/TRIGGER detector in order that an ADSR contour may be derived, as in Figure 4.13.2, which in turn is used to control various voltage controlled functions. In the patch of Figure 4.13.3 it is shown controlling a VCF.



4.13.3

Experiments should be made using different types of external sources such as voice, guitar, pre-recorded music, etc. and the various types of signal modification applied. With some external signals it is worth filtering them after the pre-amplification stage so as to facilitate subsequent extraction of their envelopes or their gate and trigger peaks.

Finally, if one simply requires to modify the external signal after the pre-amplification stage using the facilities provided with the DIGISOUND 80 then the following guidelines should be of assistance. Voice signals respond well to ring modulation and phasing and also treatment by random filtering. For ring modulation the voice is used as one input while a VCO provides the modulating signal. The resultant sound can be 'Dalek' in character depending on the VCO frequency used. A more characteristic 'alien' sound may be obtained by subsequently 100% amplitude modulating the signal in a VCA. Guitar signals respond well to additional filtering and amplitude modulation and again the ring modulator can give rise to some strange effects which are perhaps best mixed with some of the original signal. By this stage in the manual you should, however, be well acquainted with the facilities offered by the various modules and thus in a position to decide for yourself the best treatment to apply to an external signal in order to obtain the desired effect.