Copy No: ....

HANDBOOK FOR SYNTHI 100

Serial No.: ....

Installed at..

....

Datanomics LTD.

Designers and manufacturers of special electronic equipment, "Dimple" micro-processors, plant and process control and UHF video systems. Manufacturers and suppliers of Datanomics' own range of products and the E.M.S. range of synthesizers. Sub-contractors to the trade for looms, P.C.B.'s, sub-assemblies, control equipment, testing rigs and sheet metal. Suppliers of inspection equipment incorporating nucleonics. Research & development projects a speciality.

Westminster Rd., Wareham, Dorset BH20 4SP. Tel: Wareham (09295) 6311 Telex 418480 Discus G. Cables 'Mediscus Wareham'

F O R E W O R D

Synthi 100 is only made in very small quantities, and because every aspect of its design is under constant review with an eye to improvement, each example of the machine is slightly different. This makes it impracticable to produce a definitive, once-for-all handbook, and for some time there has been no handbook at all, since we are aware that purchasers of this instrument are in the main already experienced in the use of voltage controlled equipment and the general techniques of electronic music, and in most cases will have had extensive demonstrations before taking delivery. At the same time we have been conscious that some users have simply not grasped the possibilities at their command and have seriously underrated and underused the capabilities which S.100 offers, partly because there is no other synthesizer with which to compare it. Apart from possessing a number of devices unique to EMS, the digital sequencer gives S.100 a degree of user control which can only be rivalled by fully computerised studios at many times the cost.

Because of the almost limitless possibilities of the studio, it is not easy to attempt a survey of its capabilities, and the following pages should be regarded as an "intelligent user's guide". We expect each owner to add his own notes and comments, individual calibrations of particular interest, etc. To take a particular example, the eight voltage controlled filters have always been built to an exacting specification, but because our research and development department considered there was room for improvement the circuitry has been slightly modified almost with each example manufactured. The settings and procedures given in this book may therefore not precisely apply to a given machine, but the method of taking response curves etc., is still valid and useful in all cases.

If in doubt about a particular device, look it up in the contents list and read the typical uses given, but regard this as a starting point rather than a complete survey of possibilities; the great strength of voltage control lies in the enormous variety of inter-dependencies available, and in S.100 these are so numerous that no handbook can begin to cover them.

We have divided the descriptions of the devices into groups of similar general purpose, such as Source or Treatment, Digital or Analogue, but some devices have several applications and therefore appear in more than one section.

If you consider that the presentation or contents of this handbook are not satisfactory, we would be pleased to know about this, and the purpose of the loose leaf method of binding is to make it possible to amend and add material easily. No doubt you will also wish to exchange views and ideas with other S.100 owners, and to this end we include a list of purchasers up to the time of issue of this copy - though we cannot of course guarantee that some of these machines have not changed hands.

VERY IMPORTANT

In earlier S.100 design, patch pins of 2K7 or shorting pins were often used. The patching system now employs virtual-earth summing nodes and 100K (white) pins are standard. A shorting pin is never used, since this would connect a high-current source directly to the summing node. Apart from gross distortion, damage may result to the input. Red (2K7) pins are used only for oscilloscope patching.

CONTENTS

SECTION 1:

Functional Principles and Layout, First Switch-On, Brief Checks, Input/Output Connections

SECTION 2:

Analogue Signal Devices (Sources and Treatments)

Signal Sources:

Audio Oscillators/Sub-audio Oscillators (1)
Input Amplifiers
White Noise Generators

Signal Source/Treatments (Dual Purpose Devices)

Filter/Oscillators (1)
Envelope Shapers (1)

Signal Treatments

Ring Modulators
Eight-Octave Filter Bank
Reverberation Units
Output Amplifiers

SECTION 3:

Analogue Control Devices and In-Built Peripherals

Control Devices:

Filter/Oscillators (2)
Envelope Shapers (2)
Sub-audio Oscillators (2)
Double keyboard
Joysticks
Random Voltage Generator
Slew Limiters

Peripherals:

Treatment Sends and Returns
Oscilloscope
Frequency Meter
Option

SECTION 4:

Digital Sequencer 1000 - Principles, Basic Patching and Editing

Description of Operation
Patches with Keyboards and Random Gen.
Editing

SECTION 5:

Analogue Signal-to-Control Devices, and Useful Additions to Standard Equipment.

Signal-to-Control Devices

Envelope Followers
Pitch-to-Voltage Converter

Additional Items

Jumper Leads
Extra measuring instruments
High-R Pins
Variable Pins
Key 4 remote relays
etc.

SECTION 1

SECTION 1

FUNCTIONAL PRINCIPLES AND LAYOUT, FIRST SWITCH-ON, BRIEF CHECKS, INPUT/OUTPUT CONNECTIONS

Functional Principles and Layout

The designer of a synthesizer can go about his work in two quite different ways. He can aim to make the instrument reasonably easy to use by reducing the number of decisions to be made by the user. This means fitting switches or permanent connections between various components in order to reduce the size of the patch (interconnection field with a large number of choices - like a telephone exchange) which the operator has to control for himself. The trouble is that the fixed decisions reflect what the designer would like, but not necessarily what you would like. Alternatively he can design a 'no compromise' instrument in which every decision has to be made by the user and a virtually limitless number of configurations are possible. Such machines are not particularly easy to use well, and the larger the machine the more difficult it becomes, but we make no apology for following this second course when we designed the S.100. On the other hand, after first making it difficult, we added a powerful assistant in the shape of the DIGITAL SEQUENCER, which is the most helpful aid to the synthesizer user ever invented. All the same there are 7,200 individual patch points in the S.100 (though we admit that not quite all of them are effectively operational), and thought and experience must be combined if really good work is to result (and the capability is there for such work).

For this reason we strongly recommend any potential user of S.100 to spend a lot of time 'training' on a small synthesizer such as our VCS3, Synthi A or AKS, so that the methods of using matrix board patching, the use of control voltages etc - in fact all the basic groundwork of EMS voltage controlled studios, is thoroughly mastered before attempting to use the S.100. In this handbook, therefore, it will be assumed that the reader is already familiar with the operation of the smaller machines.

Because the possibilities are so large, any attempt to provide lists of suitable patches, etc., would be fruitless. We shall explain exactly what each device does, and offer typical examples of its use, but after that it is up to your own musical imagination and your increasing experience with S.100 to build the sounds you think of.

Don't be timid about using the resources of the machine - or imagine that there is only one way to go about solving a problem. For example, it is easy to settle for only one or two ways of creating a timbre. In fact, even without combining several techniques (which can easily be done) there are at least eight different methods:
(1) Addition of shaped waveforms from one source; (2) Additive timbres from a number of oscillators locked by synchronising inputs and tuned to desired partials; (3) Subtractive timbre derivation by manually controlled filters (8-octave filter bank for formants, output filter on each channel);
(4) Subtractive timbre derivation by using dynamic voltage controlled filters (of which there are eight); (5) Using the filters themselves as sources, in oscillatory mode, with or without inputs as well; (6) Multiplicative timbre derivation by ring modulation, either with several oscillators (synchronised or not) or one source with frequency doubling and/or halving configurations (S.100 has three ring modulators, which may be cascaded or parallelled in many different ways); (7) Other modulation techniques (a.m. and/or f.m.) by multiple control of filters and oscillators; (8) Digital loops in the sequencer, by which controllable discrete phenomena are speeded up many hundreds of times by increasing clock rate so that they become audio frequency phenomena.

Any of these basic approaches can be further modified by e.g. reverberation (itself voltage controllable) or numerous control influences, some periodic, some asynchronous and some random. Or of course several techniques can be combined to produce timbre changes of great subtlety - and we haven't even considered any of the ways of incorporating a noise content into the timbre.

This is not an acoustics or musical textbook, so (as with practice on a small synthesizer) we must assume some previous knowledge of the nature of sound and how to manipulate it. Mental analysis of the contents of the mind's ear into likely configurations of actual devices comes with practice. Some will find block diagrams useful before translating these into a practical patch. Others will find it easier to work onto patch dope sheets (you will find a supply of these at Section 8). Others again may incorporate dope sheets into various forms of notation, or work straight on to the machine from an idea.

The above timbre example - a small fraction of a larger problem - was intended to encourage rather than daunt, in the sense that there are riches to be found if you take the trouble to find them.

The guiding principles behind the design of the S.100 were (1) to make a synthesizer in which there would be no compromise to versatility; (2) to extend and enhance the control principles already proved highly successful in our small synthesizers - i.e. matrix patching, bi-polar control voltages, direct coupled circuits which can be used both in sub-audio and audio roles; (3) to include more of everything found in the small synthesizers, but also greatly enhance the specifications of such devices as the Envelope Shaper, include many extra devices such as High Pass - to - Resonating Filters (as well as Low Pass), Envelope Followers, Slew Limiters, Pitch - to - Voltage Converter and Eight Octave Filter Bank; (4) to put the whole of this elaborate network of analogue controls under the command of a digital sequencer, giving an accuracy of performance hitherto undreamed of in analogue synthesizers.

Having decided upon the array of devices we would need, we were faced with nearly 120 inputs and the same number of outputs. Apart from the very high cost of a matrix board this size, it would have been extremely unwieldy, so instead of splitting one board into Signals and Controls (as in VCS3, Synthi A series), we specified two boards of 60 x 60, a Signal Board and a Control Board (left and right respectively). There were some difficult decisions - borderline cases where a device (e.g. an oscillator) may be a signal or a control. But provision is made for cross-patching, and it is also possible to use jumper leads from patch to patch. The input Amplifiers and Output Channels come up on both boards, and the first four Output Channel inputs appear on the Control Board as well as the Signal Board. Before going any further, and it is not necessary to understand all the devices at this stage, study the boards carefully, and identify all outputs to the patch (horizontal rows) with their appropriate panel controls. Find also the equivalent vertical columns referring to signal inputs (where applicable). The columns on the Control Board do not have any panel controls, because our design policy is to equip all outputs with level controls, and vary input levels from the controlling device. In cases where you wish to feed the same control to different devices at different levels, a selection of different pin resistances can be used (three are supplied as standard, and you can add more yourself).

In Section 2 and 3 you will find all the analogue signal and control devices described, with suggestions for learning their use. Beginning with simple patches, you can build up elaborate configurations in which many parameters are being dynamically controlled.

You may be puzzled to find that three devices - the Pitch-to-Voltage Converter and the 2 Envelope Followers, have inputs on the Signal Board and Outputs on the Control Board. This is because they are signal-to-control converters, and one of their main functions is in real time transformations of live instrumental inputs. They are described in Section 5.

Functionally, the sequencer is the most elaborate device in S.100, and it is a very powerful control instrument. The programming of analogue control is a matter of delicate adjustment, as you will know from your experience of small synthesizers. On S.100 you may easily have over 100 pins contributing to a patch, and to control all this with accuracy from the two keyboards, the two joysticks, or from the attack-dependent envelope shaper trapezoids or the various free running controls such as the slow oscillators (10, 11, 12) or the random generator, can be a difficult matter when a number of voltages have to change from event to event, or dynamically during them. The sequencer is digital, and operates by storing precise numbers which are converted into equally precise voltages to control to the devices. As you put data into the memory you also hear what you are playing, and on play back you hear it again exactly the same. When adding new voices you can continue to hear any sound previously recorded, and three completely different sets of voltages can be stored, controlling (if you like) quite different types of sound. If you don't want to 'play' your sequence in real time, you don't have to - you can put in data as slowly and carefully as you like. Mistakes can be singled out and removed or replaced without the least disturbance to any wanted event. Short of full scale computer control (and this is also possible, of course), there is no method of storing voltages which is so accurate and flexible. It makes the S.100, in spite of its huge possibilities of sound production and control, a reasonably docile creature to handle.

The sequencer operating controls are all at the right hand end of the machine - some on the vertical panel and some below, just above the output channels. The functions of all these controls are described in Section 4, and in Section 6 we describe some more elaborate ways of using the sequencer.

The design of the S.100 does, we believe, allow for any method of work the composer wishes to use. It can be 'played' in real time or programmed slowly and carefully. It can be randomly or accurately controlled (or anything in between). Different types of sound can be set up at leisure one after the other but produced simultaneously. The devices include a very full capability for transforming live instruments, and the ideal studio location is near a performing area so that signals can be processed and sent back to the hall.

First Switch-on and Brief Checks

The S.100 will have been lined up, and is normally installed, by EMS engineers, and no attempt should be made to alter presets or make other internal adjustments on a new machine without specific instructions from a qualified installer. But the following points may be useful if for any reason you have to install it yourself, or take over a used machine.

1) Be quite sure, if the machine has not come direct from us, that the mains supply is correct for the machine. If your supply is 240V, and there is any doubt at all, first obtain a 1KVA 240-110V step-down transformer and run the machine through it. If the machine is already set for 240V the meter and panel lights will be very dim, and the sequencer clock display will not light up. If everything runs normally, you can continue to run the machine through the transformer, but if you prefer to convert it to 240V, all the power supplies must be changed, viz:
    (a) The power supplies for the S.100 (See technical manual for details)
    (b) The mains adjustment for the oscilloscope as fitted.

THE S.100 IS NOT SUITABLE FOR DC OR BATTERY OPERATION

2) The stand is normally on castors (though various special stands can be supplied), so the S.100 can be swung out for rear access. Where space considerations make this difficult it may be more convenient to mount it through a partition, giving access from behind, or in some studios it can be free-standing. Ventilation space must be provided above, as the sequencer and the scope both need to dissipate a certain amount of heat. Since one rarely sits at the S.100, some users prefer to raise the whole instrument higher to make it more convenient for standing operation. This also gives clearance to enable the keyboard to be stowed underneath when not in use.

3) After switching on, make the following checks:
   (a) Meter dial lamps on (no switch)
   (b) Illuminated panel lights (switch near mains switch)
   (c) Digital display (Sequencer Event Time). This will come on initially at any number. Pressing CLEAR/RESET or RESET buttons should bring it to zero, where it will stay. It will not run unless patched, so expect no result if you press START.
   (d) Envelope Shaper 'ON' lamps. Switch all three shapers to 'Free Run'; DELAY and ON to 0, ATTACK and DELAY about half way. The lamps should flash on and off.
   (e) Scope and Frequency Meter. The digital display on the frequency meter should be at zero (except certain types which may be in a time-counting mode). The scope should show two traces (if time base on and tube controls correct).
   (f) You can also check all source devices on the scope, but may prefer to do this audibly. A scope check of all oscillators, noise generators and filters (response controls to maximum to put them into oscillation), will, however, familiarise you with patch and panel locations, and scope controls.

All these simple checks can be done without connecting an output, but temporary listening arrangements are easily made by connecting stereo phones to Pan Outputs 1 - 4 Left, and 5 - 8 Left.

Put all Pan controls to Left. You can now listen to any output channel, but naturally in order to use the S.100 you must make proper input/output arrangements.

Input/Output Connections

Look at the back of the lower panel. There is a trough with an access tunnel at each end of the machine, and all leads can be brought in from the rear and concealed neatly in this trough. From left to right you will find:

4 Pan Outputs
8 Individual Outputs
4 External Treatment Sends
4 External Treatment Returns
8 Input Amplifiers (sometimes but not always there are ten sockets, Chs. 1 and 2 having extra sockets for MIC. amps).
2 Option sockets
2 71-way connectors (one male, one female) to which the whole Control Board is wired . 60 ways on each are used, leaving some spare ways which can be used for other purposes
1 Keyboard socket
1 Panel Light switch
1 Mains Keyswitch (key removable when ON as well as OFF), with main fuse and mains input socket beside it.

To take these in order:

Output Channels

FIGURE 1

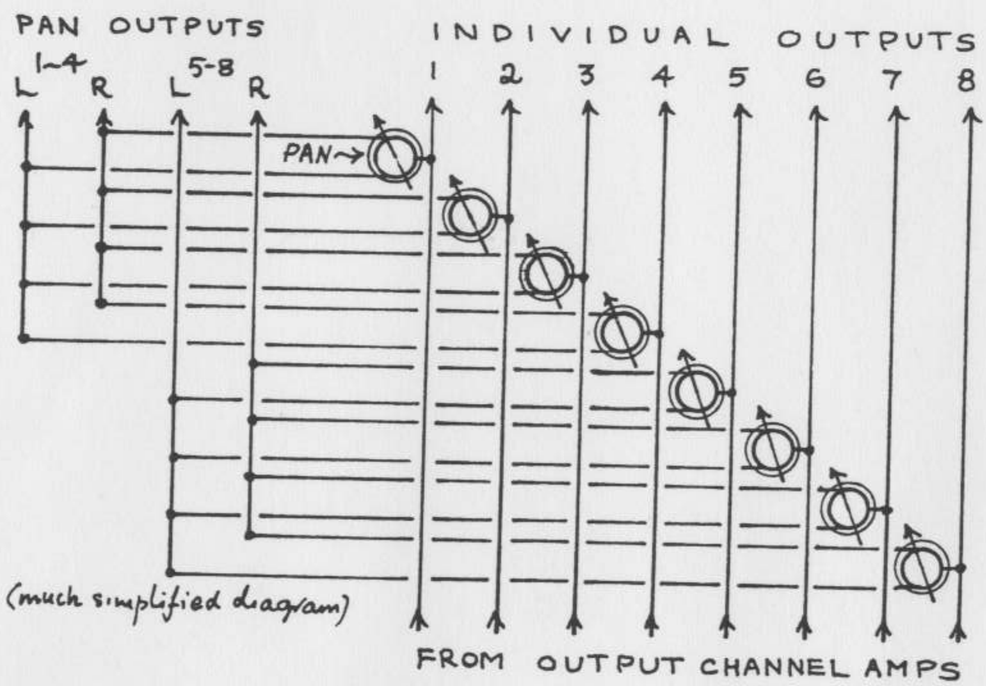
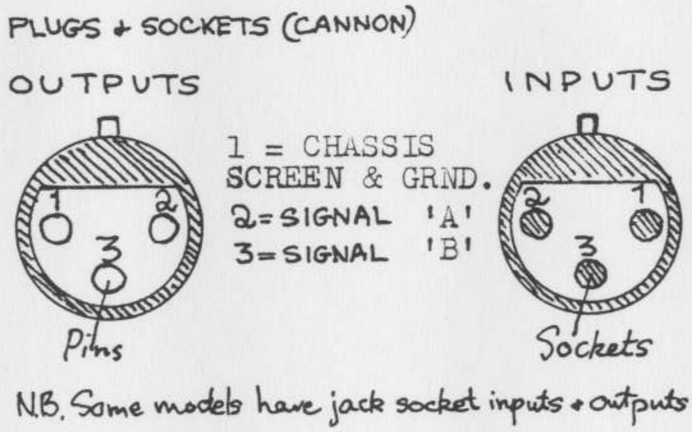


Figure 1 shows the output arrangements, from which you will see that the Pans are in two separate groups, and can be taken to four outputs if desired. The limitations are that you cannot cross-pan from Outputs 1 - 4 to Outputs 5 - 8 or vice versa. The pan outputs are very useful when you have only a limited number of monitor amplifiers and speakers available. If necessary you can hear all 8 channels on two output monitors (same as stereo phones connection mentioned above). With four amplifiers use all four pan outputs. When a full studio mixer is available, complete with its own panning arrangements, it is best to use the individual channel outputs, and this applies also when using multi-track tape recorders unless panning direct from S.100 to a tape

recorder. The socket connections are identical on all 12 outputs, and you can change quickly from one type to the other. Figure 2 shows the pin connections for all input and output plugs and sockets, which are balanced lines. Signal levels should be within the range 0 - 16 dBM.
When feeding to an unbalanced source, use earth as pin 1, Signal to pin 3. Pin 2 is left open circuit. (Do not earth).
When feeding to the S.100 from unbalanced, separately earthed equipment, connect signal earth to pin 2, signal high to pin 3. (This will avoid hum loops).

FIGURE 2



Treatment Sends and Returns

These are general purpose output and input lines which come up on the Signal Board, and are very useful for interfacing with other studio equipment, such as extra filters, reverberation plates, other synthesizers. They can also be used as extra output or input lines. They are individually wired (i.e. there is no special relationship between Send 1 and Return 1), and are fitted with level controls (5K) on the panel. No amplifiers are built in to this facility, but of course an output amplifier can precede a Treatment Send. If extra gain is needed for a Return, one of the Input Amplifiers should be used instead.

Input Amplifiers

These eight sockets (ten if special microphone amplifier inputs are provided) feed eight identical input amplifiers. They are of line sensitivity (0 - 16 dBu) or 10VDC Max. for 0.1% THD) and their direct coupling makes them suitable for signals or controls. Low output microphones need external amplifiers, and in some models Inputs 1 and 2 have special additional microphone sockets with provision for powering these preamplifiers (which are also supplied in these cases).

Inputs can be from a mixer, from other electronic music equipment, from tape recorders, electric instruments, radio, phonograph etc. In some studios it would be most convenient to take all the output, send and return, and input lines to a main studio patch for distribution.

Option Sockets

These two 8-way sockets are not wired, and are included (like the Option knob at the right of the right hand panel) for the convenience of the user. In any studio there must be room for expansion, and although these sockets may never be used it was felt desirable to include some provision for extra facilities. One obvious use would be to plug in a button box for remote control of tape recorders.

Control Board Connectors

This plug and socket enable the entire control board to be operated from a remote point. If the S.100 is interfaced with a computer, or its control patch made part of an even larger patch, this facility makes it easy to do. It can, of course, be partly used, in the sense that there might be selected controls you wish to operate remotely (e.g. envelope or sequencer key inputs). There are also a number of unused ways on these plugs, so extra connections can be made.

Keyboard Socket

To plug in the double 5-octave dynamically proportional keyboard provided.

Panel Light and Mains Input

The lights under the overhang give working illumination to the lower panels. The mains keyswitch, note, can be removed when the machine is on as well as off. This is valuable when a sequence has been recorded and work interrupted, in case the S.100 is accidentally switched off.

Peripherals

Apart from the equipment already suggested - i.e. studio patch, general purpose mixer, tape recorders, monitor amplifiers and speakers - nothing more is essential to operate the S.100 efficiently. Two excellent measuring devices - a double beam oscilloscope and a frequency meter - are incorporated in the machine itself. A digital voltmeter is a useful accessory in certain kinds of work where it is essential to monitor very accurately the voltages being sent to the sequencer or supplied to devices, and is supplied as an in-built facility on certain later models. A reverberation plate is a useful refinement, and this can be connected through the Treatment Sends and Returns. Any studio acquires extra equipment for which there is ample provision in the input/output arrangements of S.100. Many special circumstances can occur, and any competent studio manager will devise suitable methods of control. A single example will show what we mean: The sequencer has a special Key output (Key 4 - see Section 6), and it might be a requirement to make this key operate (a) a remote attack in another synthesizer, (b) an electronic stepping device so that successive keys perform different functions, (c) a mechanical relay to e.g. start a tape recorder. All these things can be done - it is a matter of devising the relatively simple peripheral arrangements you need.

The following two Sections deal respectively with the analogue signal device (sources and treatments) and the analogue control devices. In fact you will have to refer across from one description to another fairly frequently, because it is a practical impossibility to treat each device in isolation.

NOTES:

SECTION 2

S E C T I O N 2

ANALOGUE SIGNAL DEVICES

Signal Source

- Audio Oscillators
- Sub-audio Oscillators (1)
- Input Amplifiers
- White Noise Generators

Of the twelve oscillators in the S.100 (apart from the eight Filter/Oscillators), nine are intended largely as audio sources, and three as control sources. But since all outputs are directly coupled (able to pass DC) these roles can be reversed, and it is largely a matter of the frequencies required. Simply switch to high/low range as required. The nine 'audio' oscillators divide into two types, but all have the same frequency range. All have a control voltage slope of 1V per octave. All can be locked by synchronising inputs, which ensures perfect tracking of unisons, octaves, fifths or any other simple ratio partial.

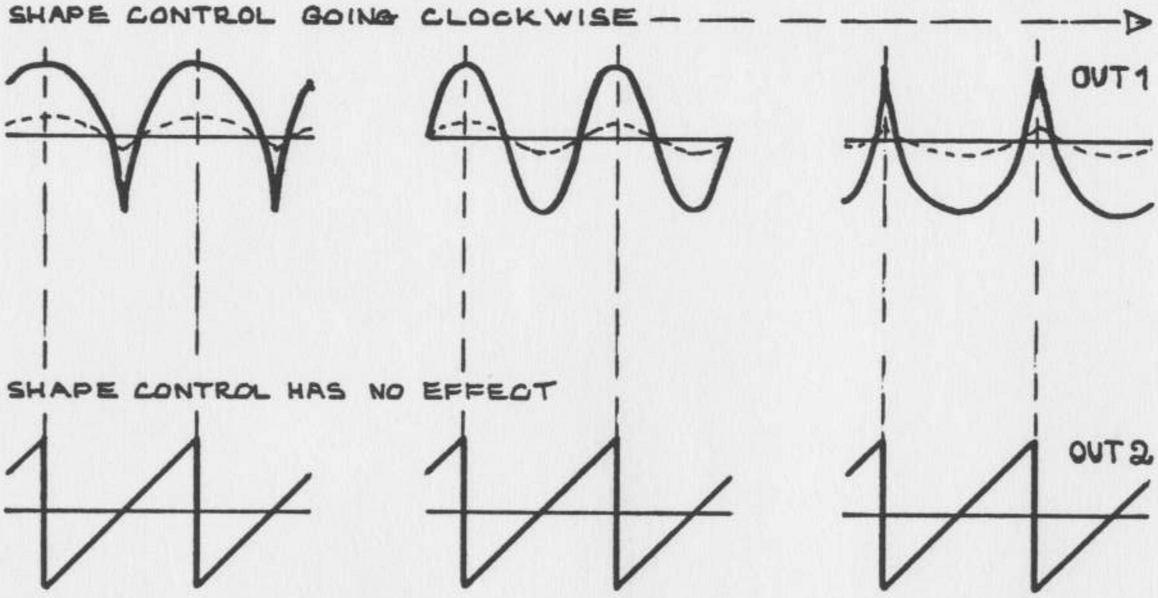
<u>Oscillators 1 - 12</u>

Output 1 - Sine + Sawtooth (with sine Shape Control to give variable amounts of even harmonic distortion)

Output 2 - Sawtooth Ramp + Pulse, with Pulse mark-space ratio variable.

These 12 identical oscillators have two in-phase outputs with shapes and phase relationship as shown in Fig.3. Find these outputs on the Signal Board and the control inputs on the Control Board.

FIGURE 3.



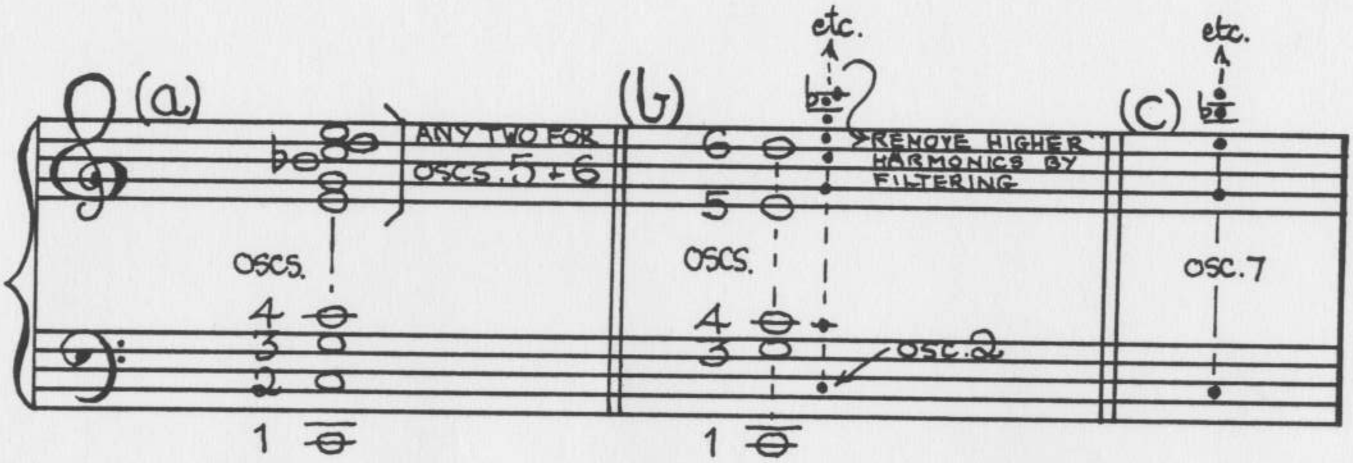
(Amplitudes separately controllable)

With the small synthesizers, the amount of timbre control possible without using filters is limited by the fact that two oscillators can never be tuned so that they are locked in phase, and beats will always occur. Try now putting Oscs. 1 and 2 sines to output, and Osc. 1 ramp to synch. input 2. For the moment keep Osc. 1 ramp level at zero, and tune the two oscillators an octave apart, Osc. 2 being the higher. Monitor both on Y_1 (two pins); however carefully you tune, the picture will slowly drift as the relative phase changes, and slow beats will occur. Now bring up the ramp output of Osc. 1; at a certain point the waveform (and the sound) will lock, and there will be the illusion of one sound source instead of two (particularly if Osc. 2 sine level is set somewhat lower than Osc. 1, to simulate a 'natural' second harmonic). This facility enables timbres to be constructed additively, by adding sines to build the desired harmonic structure. There is no reason why you cannot use ramps as well, but the harmonic structure is more difficult to predict (but see below).

The main thing to remember when using synch. inputs is that they cannot lock unless they are near to a simple frequency ratio (2:1 in the above example, but 1:1, 3:1, 8:1 etc). If you are way off tune the oscillator being controlled will give an unstable output. Try this by moving Osc. 2 frequency slowly. As soon as it is far enough off for Osc. 1 to be unable to lock the octave, it will switch rapidly from one frequency to another (a complicated function of both the frequency and amplitude of the synch. input). As soon as it finds the next simple ratio it will lock again (the fourth below the octave or the fifth above, etc). So you can synchronise a whole harmonic series, within reason.

For a more complete study, tune all six oscillators as follows: FIRST PUT IN CONTROL PINS from some suitable control source. Unless you do this before tuning you will have to do it all again. For this experiment take a joystick output, which will enable us to track our constructed timbre through several octaves (a static timbre at this stage, of course, with no internal variations or envelope). Put white (100K) pins in at C114/26, 27, 28, 29, 30, 31. Set lefthand joystick (q.v.) vertical range control to give about 3-4 octaves when the stick is moved up and down, and leave stick towards the low pitch end (towards you). Tune Osc. 1 to a low note, put all six sines to one output, and insert a synch. pin from Osc. 1 ramp to Osc. 2 synch. input. Follow the series shown in Fig. 4(a) - i.e. tune Osc. 2 to the octave above, setting the frequency control in the centre of 'locked' area, and the ramp output at or near maximum. Continue to put in synch. pins after tuning each oscillator as near as possible to successive harmonics, then check them. Tune slave oscillators on low side if anything, as synchronisation tends to raise frequency.

FIGURE 4

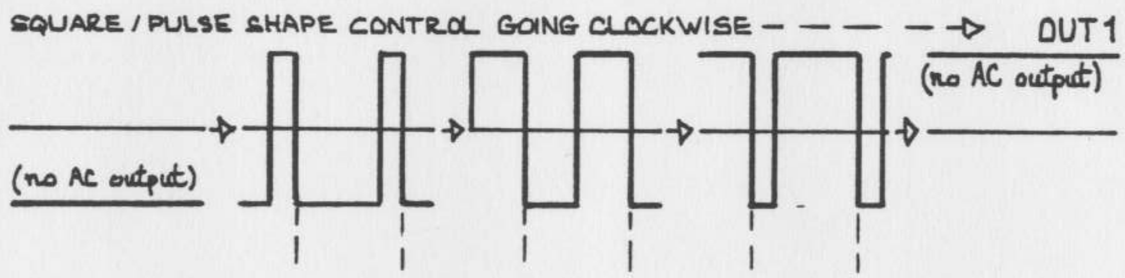


When you get to Oscs. 5 and 6, you may choose various configurations from harmonics 5-10 or higher (or you can miss out one or two of the lower harmonics and use more higher ones). Locking the higher harmonics becomes more difficult as they are much closer together and the ratios larger. Now adjust sine levels so that the fundamental is loudest and progressively reduce the level of each harmonic (this for a start - you can alter the timbre a great deal by changing internal amplitudes). Try moving the joystick. Some of the oscillators may go out of lock at some point. Readjust for best holding. It is asking a great deal to expect synchronisation of, say, the 7th harmonic for five octaves or more, but it should be possible over at least four octaves. It is also worth trying different synch. pin arrangements. For example State 2 on Specimen Patch 1 locks Oscs. 2 and 3 from 1, but 4 and 5 from 2 (the octave of 1), and 6 from 4 (the double octave). This may work better, but you have to try different arrangements. Sometimes a double-locked arrangement is best, with mutual synch. pins between two oscillators. Note that the synch. input is driven from the output you are not using for signal - otherwise you could not reduce the signal level for the sound you want without losing the synch. input level.

Look at Fig. 4(b). This shows an alternative timbre structure using a sawtooth output on the second harmonic (change synch. also if you are using State 2). The waveform itself now contributes to the timbre (coming up on even harmonics of the fundamental), and the sound will be richer, but buzzy at the top. Remove some of the upper harmonics by putting the whole complex through the Eight Octave Filter Bank (q.v.). Set all the lower controls of the Filter Bank at No. 10, and slope off to 0 at the highest octave.

This method of timbre construction can give extremely rich organ-like tones, and you can experiment with emphasising certain harmonics. You can also, of course, try different series altogether, but you will not achieve lock unless the relationships are fairly simple.

FIGURE 5.



These three oscillators are also identical with each other and have the same frequency range and control voltage slope as Oscs. 1-6, Fig. 5 shows that the two independent shapers and two outputs give a very wide range of shapes. A symmetrical waveform either has no harmonics (sine), or odd harmonics only. The symmetry position of both shapers, therefore, giving respectively a 'square' wave (meaning a precisely equal time lapse between positive- and negative-going fronts) and a triangle, produce a series consisting of harmonic 1 (fundamental), 3, 5, 7, 9 etc, but the amplitudes of successive harmonics of the triangle fall off much more rapidly than those of the square. So the triangle is a much smoother-sounding waveform - indeed next to the sine it is the most gentle sound of all the ready-made waveforms in the untreated state. Do the same listening and looking tests as with the sine/ramp oscillators, first one shape at a time and then together. Note that as soon as either waveform is the least out of symmetry an even harmonic content is noticeable. The square output shaper gradually shortens one of the half-cycles until it completely disappears. Just before this point the amount of fundamental is very small and we hear only a thin buzz of upper harmonics.

You can build waveforms in a similar fashion with these oscillators (they can all be synchronised) but because they have an inherently large built-in harmonic content the timbre structure will be rich and you may feel the need of filters to add a subtractive treatment to the additive effects of synchronised oscillators. Fig. 4(c) shows the effect of replacing the original Osc. 2 waveform of Fig. 4(b) with a symmetrical waveform (try both square and triangle and synchronise Osc. 7 from Osc. 1). As a final experiment in timbre control at this stage, send oscillators through different output channels, bringing these round again to one single final output channel. Then put on various dynamic controls to the output channels, modulating the internal structure of the timbre. If this is too difficult at this point, come back to it later. Some of the controls can be free-running, some envelope dependent.

These nine oscillators are the basic (but not the only by any means) tone sources of the S.100, and you are ready to experiment further with them. All nine oscillators can also be used as controls if you wish, and for this purpose can be cross-patched to the other board either by jumper leads or via some of the output channels (which can be disconnected from the actual output circuits when used in this way so that unwanted control clicks etc cannot appear

Input Amplifiers

We have already discussed these briefly in Section 2, and they are listed here as sources because as soon as a signal appears on Rows 1-8 it becomes a source from the patch point of view. These are not voltage controlled amplifiers, and it is only necessary to check that levels are suitable (maximum input levels are + 16dBU or $\pm$ 10VDC) for use in the synthesizer. The gain controls come at the middle of the applifier so large signals can be reduced before reaching the amplifiers, whilst obtaining a good noise figure. The simplest check on a known signal is to put the input amplifier straight through to an output channel and listen for overall distortion levels. If the input is a known waveform such as sine or ramp it can be checked on the scope for clipping or other unwanted effects. It is worth checking in this way, because if a signal is already in a poor state (distorted or with high noise content) it may be a waste of time to apply elaborate treatments to it. Because the inputs may be signals or controls, these amplifiers come up on both boards (at the same location). The inputs of the input amplifiers, however, do not appear on the patch, and there are occasions when you may need to connect an output amplifier to an input amplifier by using a lead connecting a socket to a plug on the rear interfacing panel. It is a good idea to make up one or two leads with plugs and sockets for this purpose.

White Noise Generators

There are two of these because it is useful to be able to treat each differently. With 'colour' control central the frequency content is flat $\pm$3dB from 100Hz to 10KHz. The colour control is a low-pass - to - flat - to high-pass filter which tilts the spectrum towards 'dark' or roaring types of noise or 'light' and hissy sounds. It is a relatively gentle treatment, and normally one would use other filters as well, of course, but it is surprising how useful it is in controlling the general character of the final output. Apart from being the source for a very large range of unpitched sounds, noise can be used as a random control, and as a contributing factor in otherwise pitched timbres. Because noise almost invariably involves the use of other treatments, it is more convenient to bring it up under our discussions of the relevant treatments, and so we do not suggest any specific exercise at this point. But find the outputs on the patch, pin to an output and experiment with the effect of the colour controls.

SIGNAL SOURCE/TREATMENT (DUAL PURPOSE DEVICES)

Filter/Oscillators (1)
Envelope Shapers (1)

These eleven devices are more often used as signal treatments (filters and envelope shapers) or control sources (envelope shaper trapezoid outputs), but it is worth remembering that they all have a perfectly valid function as audio signal sources. These devices are further dealt with (as Controls) in Section 3.

Filter/Oscillators 1 - 8

Filters 1 - 4 are Low Pass to Resonating Filters with an additional oscillatory mode.
Filters 5 - 8 are High Pass to Resonating Filters with a similar oscillatory mode.

All have a range better than 5Hz-20KHz, and a slope (in LP or HP mode) of 24dB per octave. As Resonating Filters they have a maximum Q (magnification factor) of 20.

N.B. Filter designs have been modified considerably in successive S.100 productions, and some details of the following notes will not apply to all examples. The general points made are still of value, however, and users should make their own calibrations.

A. Filter/Oscillators as Oscillators

Of the eight audio sources available, Filters 1-4 (Low Pass) deliver a sine of fairly high purity - higher than that from Oscs. 1-6. There is no control of shape, but very high purity can be produced by filtering one of these sources (using another filter as a resonating filter). The four High Pass

Filters (5-8) also perform well as oscillators, but the waveform is not so sinusoidal, containing mostly low-numbered even harmonics. However these can also be made into very good sines with appropriate filtering.

Find the eight filters on the panel (top left) and the inputs and outputs on the signal board together with the eight control inputs. Put Filter 1 output to an Output Channel, and the frequency control knob at about 6. When you turn up the Response control from 0 (open up the output channel as well of course) there will come a point when the device oscillates spontaneously. It is worth making a note of the point at which each filter goes into oscillation, for future reference. Continue to turn Response to maximum. The tuning knob will now give you well over the audible range of frequencies, and the frequency is near (but not quite the same as) the centre frequency of the device when used as a resonator - in fact the oscillating frequency is a little lower, and you will notice that when just beginning to oscillate the tone is sharper than when Response is turned to maximum. Check the scope picture, which should be near to a perfect sine, and shows a smooth waveform of different character from the "segmented" waveform used in Oscs. 1-6.

Now take an output from all four LP Filters (1-4) set to oscillate as above, send them to one output channel, and put in a control from one of the keyboard pitch outputs. Play middle F-sharp and set all four oscillators to the same middle range note. There will be some beating because the filters are not synchronisable, but good tracking should be achieved over most of the keyboard range.

The four High Pass Filters will also oscillate in the same way, but the waveforms are not so sinusoidal, and the sound consquently 'rougher', though if filtered will still provide excellent sine source. In order to do this, take a high pass filter output and put it through one of the low pass filters with Response set at about 4, when it will be a resonating filter. Tune frequency control (of LP filter) for maximum amplitude, which will also be resonance and maximum purity. The same procedure can be used with the sine outputs of Oscs. 1-6 - passing it through a resonator will 'tidy it up' and bring it to very nearly a perfect sine. Only rarely do we have a musical application for pure sine, but when the S.100 is used for research, sines of even greater purity can be made by cascading resonating filters, taking great care that the extra gain achieved at each stage does not cause clipping at the next stage.

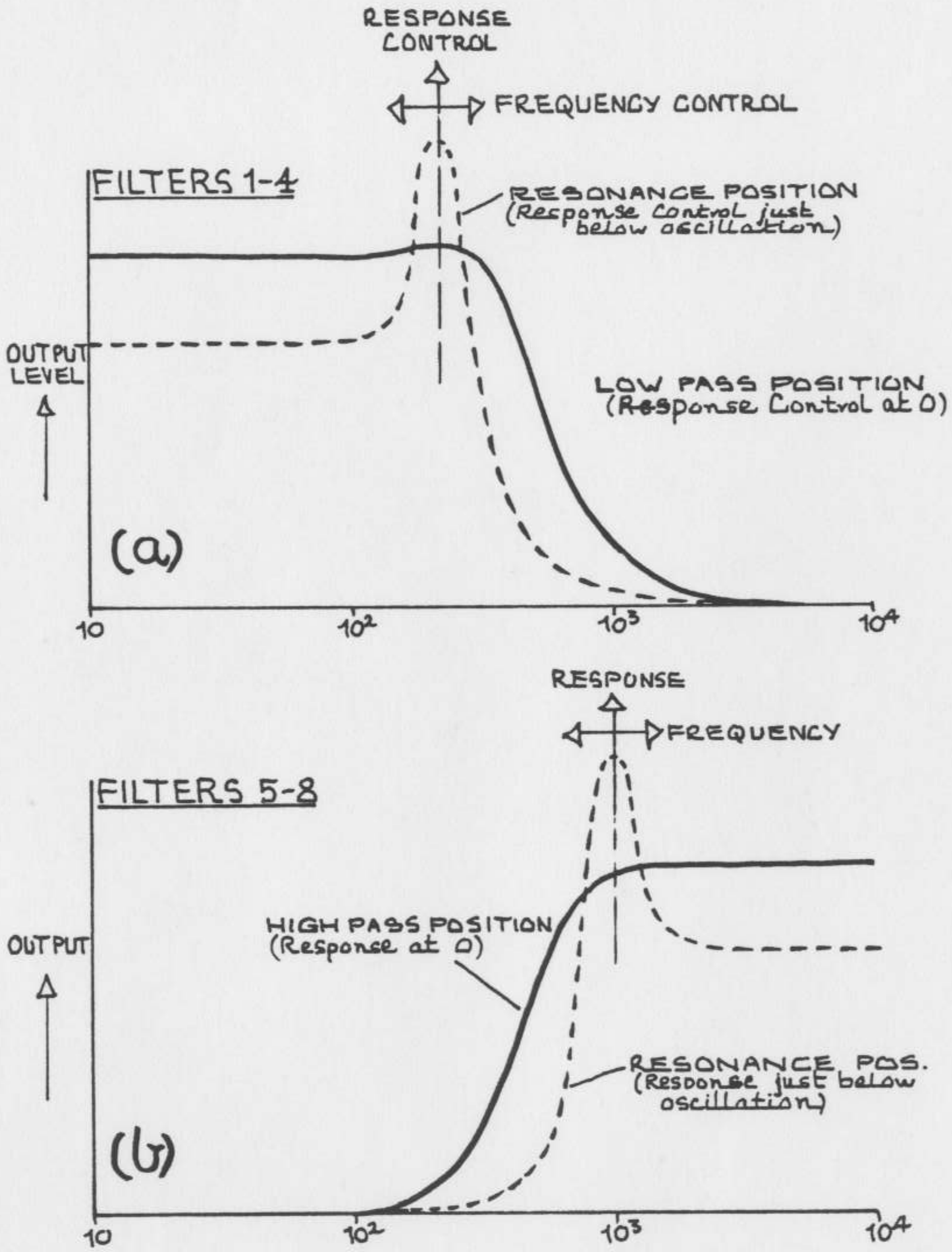
If sine filtering of this kind is required over a range of frequencies, the oscillator and filter control inputs can be tracked together by the same control voltage so that the centre frequency of the filter follows the frequency of the oscillator.

Before discussing the main function of these devices - as dynamic filters - there is one "borderline case" which should be mentioned. The production of pitched sounds particularly high in noise content sometimes presents difficulties. The kind of sound I refer to varies from sounds like quiet singing through the teeth or breathing and singing at the same time to heavy whistling and roaring noises like steam locomotives and storms of wind. They cannot be achieved by filtering noise in the normal way because these filters are not meant to be sharp enough to derive a single pitch from noise (though special filters can be designed to produce sine tone from white noise). On the other hand, although in theory it is possible to produce noise by randomising tone sources, one would need a very large number of sources to do this (but by all means experiment - some very interesting results can be got by multiple f.m. of many oscillators). But when the filters are set at the critical point at which oscillation is just about to begin, and given noise as an input, a whole range of these sounds emerge.

For a simple experiment take one of the noise outputs (colour control halfway) to a low pass filter and bring up the Response until tone begins to emerge. Over a fairly short range of the control the tone gradually becomes stronger, until at full oscillation it dominates the noise (it is a form of noise modulation of the filter). Attach a keyboard to the filter control, and try also putting in an Envelope Shaper (q.v.). Having done this basic experiment you can elaborate it in a number of ways, using the two noise generators with different colour settings, and several filters to give a complex of 'noise pitches'. You could also use a Slew Limiter (see Section 3) to produce portamenti in these pitch changes (an effect to which this kind of sound specially lends itself).

B. Filter/Oscillators as Filters

FIGURE 6.



Set all 8 Response controls to 0. This is the basic Low Pass or High Pass position (shown in Fig. 6(a) and (b)). The 'elbow' or hinge point is moved sideways by changing the frequency control either manually or by voltage control. Simple filtering in this mode will produce the effect expected by looking at the curves, and don't forget that if the filter is given a signal entirely in its no-pass area you will hear no output. Experiment with static and dynamic timbre control by using, say, two synchronised oscillators (giving a before-filtering choice of three basic waveforms and all their variants of level and shape). Guidance is given by Specimen Patch 2.

The double beam oscilloscope fitted to S.100 has a separate handbook, and if you are unfamiliar with the normal adjustments for these instruments (Brightness, focus, shift controls, amplifiers, time base etc) study this book first. For the filter display we shall use a fairly slow vertical time base (at right angles to normal), because the X input will not accept DC and for reasons given below the display must be slow. Since the analysis is just as valid in both directions, we will use a triangular timebase instead of the usual sawtooth, and this also gives us a better 'paint' with a slowly moving spot. The final display will therefore have the amplitude of the signal across the screen (X), and the spectrum up and down (Y) with high frequencies at the top. Here is the whole precedure:

(1) With scope in normal mode (time base running and inputs to Y) carefully check Osc. 1 sine and Osc. 10 triangle for best possible shape. In the symmetry position Osc. 10 will have a high speed spike at the negative end of its excursion, but this will not affect the result. Osc. 1 sine must be as pure as possible because it is going to sweep the filter(s), and Osc. 10 will provide the time base.

(2) Switch off scope time base and make X input switch. This is fitted on the terminals at the bottom right of the scope, and should always be to 'off' when using scope normally, to avoid patch crosstalk from Y inputs. Find the two spots, wind Y_2 off the screen and place Y_1 spot in the centre of graticule (don't have a stationary spot very bright or you can damage the screen). If the graticule is not very clear find the illumination dimmer below the screen. Set Y_1 gain to 0.5V/cm., and input push button to DC.

Envelope Shapers 1 - 3

These three identical devices have an amplitude range of 80dB. Voltage control of time parameters is ideally exponential to within 10% of dependent parameter over a range of 1000:1, with gradual departure outside this range.

Like the Filters, these are multi-purpose devices whose correct operation merits study, because they have more uses than might at first appear. The main controls are concerned with time, and the principle is that of a A.D.S.R. Generator (which can either be free-running or triggered in one of three ways) controlling an amplitude modulator which in turn controls the level of the audio signal.

Envelope Shapers as Free-Running Audio Oscillators

In most applications of a shaper a sub-audio envelope frequency is selected, but it is possible to use the trapezoid generators as audio oscillators at the fast end of the time ranges, and various rather subtle timbre controls can be applied; there is also the possibility of using the device as an audio frequency amplitude modulator. The upper frequency limit, however, is around 250Hz, or the general area of Middle-C. The audio output is taken from either Envelope output to the Output Amplifier voltage inputs on the Control Board. If you wish to apply treatments such as filtering, the signal can be picked up and patched into any signal devices required.

Put all time controls (Delay, Attack, Sustain, Release, Decay) to minimum and turn the appropriate level output to about - or +4 (polarity is immaterial in this application). Select FREE RUN on the mode switch. Signal Level is not effective because we are not using the amplitude modulator. You should now obtain a tone at the output channel selected. Altering any of the time controls will lower the frequency and change the shape (and timbre). Move all the time controls up slightly so that the note is an octave or more lower than the maximum (say 100Hz). By manipulating the time controls in an equal and opposite fashion it is possible to change timbre while the pitch remains steady, and frequency modulation or any other automatic change can be applied (say from Oscs. 10-12) at the control inputs (C3-17) where the trigger (not applicable here) and all time parameters of each shaper can be controlled. A wide range of interesting and controllable low pitched sounds is available, based on versions of the trapezoid shape.

If a second tone is now put through the amplitude modulator - i.e. the normal signal inputs and outputs of the shapers there will be modulation products - i.e. the sums and differences of the two tones and all their harmonics, as well as some of the new input (listening now to the Signal and not the Trapezoid output). If we call the trapezoid tone A and the new input B, the result will be AB + B. If A is a symmetrical trapezoid its series will be A, 3A, 5A, 7A etc. Let us suppose that B is lower than A and sinusoidal. The result will be B, A-B, A+B, 3A-B, 3A+B, etc.

These are just a few examples of a range of uses for the envelope shaper not often exploited. Amplitude modulation can be alternatively applied by using the Output Amplifiers (q.v.) as modulators, and pure multiplication (i.e. modulation products only) by means of the Ring Modulators (q.v.). In Section 3 we look again at the Trapezoid outputs, as Control rather than Signal sources.

Envelope shapers as Signal Treatments

The maximum and minimum lengths of the four time sections making up the trapezium vary slightly with each example, but the range is approximately from 1mS for each section (or 4mS per cycle = 250Hz) up to 10S or more, particularly Decay which normally reaches well over 20S. Because the time range is over 10,000:1 a small percentage variation in component values can easily double the maximum times, but I give an average length. Envelopes can be at least 30S long, and with voltage control can be extended to 50S or more. Once you have noted settings for your own machine, you can rely on a consistent performance for a given setting.

Connect a tone to one of the signal inputs and patch the output to a monitor. Turn up Signal Level, set mode switch to Free Run and time controls about halfway. The red light should flash on and off and the signal should be heard to fade in and out. Still on Free Run, experiment with the time controls. You can produce a variety of shapes by changing these settings. For example a percussive sound will need quite short Attack and Decay times and a longer Delay time. Delay, in the Free Run mode, determines the length of the quiescent quarter cycle of the trapezium. For example a long Delay setting in conjunction with short settings on the other three will produce short 'blips' of sound with long silences between.

In the other three modes a trigger is required in order to 'key' the envelope shaper, or no sound will be heard because the trapezium will remain indefinitely in the Off part of its cycle, awaiting a trigger. It is important to understand the action of Delay, which is the time the shaper remains quiescent after a continuous trigger known as a gate is sent. But there may also be an Off period, when the shaper has finished its programmed cycle and is awaiting a new trigger. Off and Delay must be added to give the total quiescent period. In many practical cases, Delay will be set at or near zero.

A trigger can be applied manually by pressing the red button to the right of each mode switch, or electrically in a variety of ways. If the keyboard is connected a pin connecting the envelope output to the appropriate gate input will trigger the shaper whenever any key is depressed. Still on the Control Board, triggers may come from the sequencer, from the Random Generator, or from one of the oscillators. Provision is also made for trigger inputs on the Signal Board, at S12, 13, 14. So very comprehensive facilities for a variety of trigger sources have been made, and you will find all of them useful at different times. But for test purposes either the keyboard (q.v. Section 3) or the manual buttons are probably the most convenient. We will now look at the three trigger modes in order:

Gated

In this mode the length of the actual trigger voltage is important, because the shaper will remain in the Sustain state until the voltage is removed. This has the advantage of giving manual control of note shape when the keyboard is used, since there is direct response to the degree of legato used. This does not apply to automatic triggers such as that from the Random Generator, whose length is always the same. 'Gated' should not be used when a very short sound is required, because it is often impossible to send a short enough trigger to produce a true staccato. An important point to note about this mode is that if the trigger voltage is removed before the shaper has reached the Sustain part of its cycle it will immediately begin decaying and never reach its maximum amplitude. So if Delay and Attack are set long it is quite possible for no sound to be heard, since the shaper will never open the modulator enough to pass a signal. On the other hand this facility can be used to produce manually controlled amplitude undulations which vary according to the touch used.

When automatic, short keys are used, the Delay setting should be very short, and Attack short enough to ensure that the signal appears before the trigger is removed. In most such cases, Single Cycle is the preferable mode.

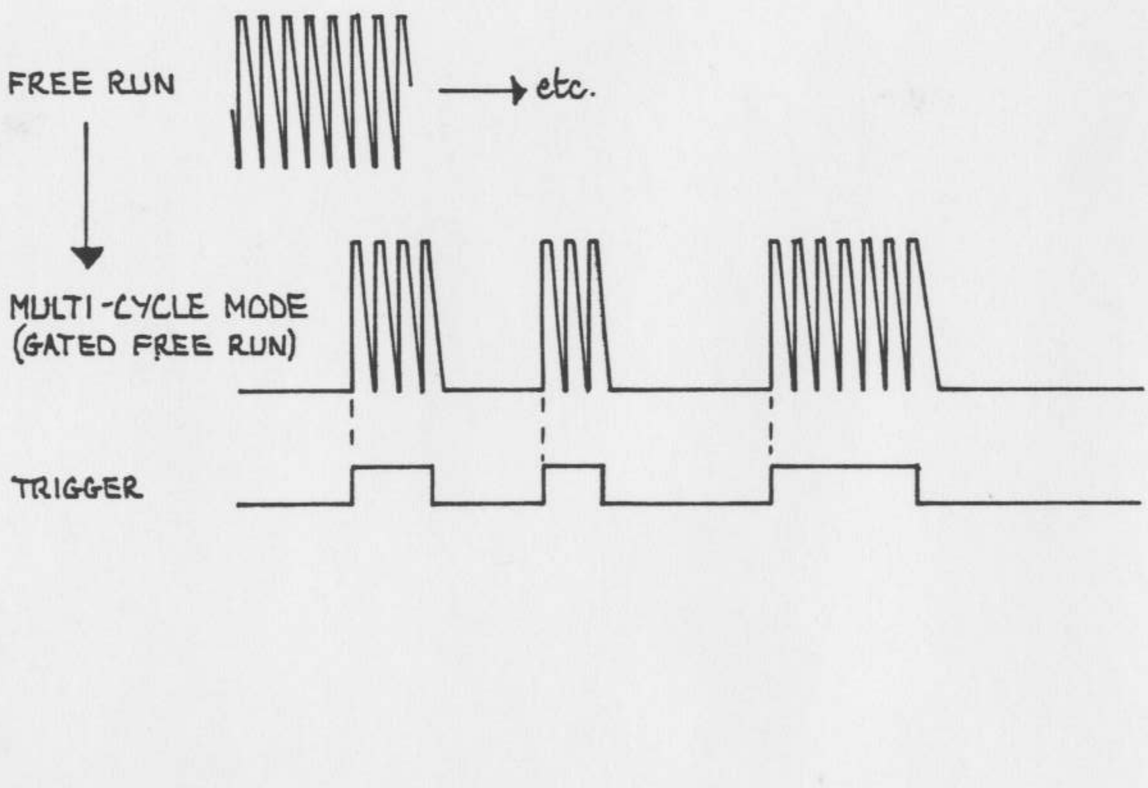
Triggered

In this mode the trigger length is immaterial, provided that it is longer than the delay time because the arrival of a trigger pulse will initiate the whole cycle, whether or not the trigger voltage remains. If a long Delay and Attack has been set, you will have to wait for perhaps 20" for the sound to appear, but it will eventually come through even though the trigger may have been removed long before (the time parameters will be as in Free Run, but for one cycle only). More often, however, the requirement is for very short rather than very long sounds - shorter than you could manually produce using gated. Very short sounds can be set and played from the keyboard even though the note itself is held down. To play another note it is necessary to remove the trigger voltage momentarily and apply it again. This means playing in a detached style, one note being released before the next is struck, whereas in Gated a full legato can be used. Practice and experience will show which to choose for a particular occasion, or mode can be switched at any time during a passage.

Multi-Cycle

An alternative label for this mode is "Gated Free Run". The trigger voltage sets the shaper going in its Free Run mode as long as it is present, and typically the settings are arranged to produce rapid repetitions when in Free Run, so that bursts of short sounds emerge when a trigger is sent. To set up, switch first to Free Run and arrange any desired repetitive envelope. Turn mode switch to Multi-Cycle and send varying lengths of trigger. If the shaper is set fast, the result will be modulated tone whose pitch can be varied by applying different control voltages to the time control inputs on the Control Board. Fig. 11(c) shows a typical multi-cycle set-up.

FIGURE 11 (C)



Control Inputs

The time control inputs can be used together to shorten or lengthen all the envelope times simultaneously, and in Section 6 we describe the use of the Clock Rate Knob voltage for this purpose. As an experiment pin a joystick to all four time inputs of a shaper. With the joystick central, set a Free Run of moderate speed and a clearly defined internal time scale - for example Delay ½", Attack 1", On 2", Decay 4". Moving the joystick will speed or slow the envelope cycle, but the proportions of the times will remain the same. You will find the control inputs extremely versatile in making the envelope vary according to numerous controlled or random influences. One of the limitations of most small synthesizers is that a whole passage has to be played with a preset and identical envelope, although the Synthi VCS3 and AKS do have a Decay control. But with S.100 great subtlety of envelope control is possible, including the combination of several envelopes into a compound function.

Compound Envelopes

There are several ways of making a more complex curve than is possible with a single A.D.S.R. and in later sections we deal with A.D.S.R. outputs controlling output amplifiers (another envelope shaper configuration) and with slew limiters as envelope modifiers. A more direct approach is to parallel the same signal through two or more envelope shapers set differently, and apply either the same trigger to all or a delayed trigger to some of them. This naturally reduces the number of signals that can be shaped, but this is a matter of the way the music is planned. Shapers can also be cascaded, but this is a harder configuration to manage because when any shaper is in an Off or low level condition it will render the others ineffective - i.e. it is very easy to achieve a non-result. It is important to make sure that maximum amplitudes - when all shapers are On - are not excessive, bearing in mind that gains are being added.

By the use of suitable automatic time controls a virtually infinite variety of envelope profiles can be generated, and the difficulty lies not so much in the capability of the machine as in the ability of the user to design the right control programme.

SIGNAL TREATMENTS

Ring Modulators
Eight-Octave Filter Bank
Reverberation Units
Output Amplifiers

These twelve devices are all purely treatments modules, without source possibilities. The Ring Modulators and Eight-Octave Filter Bank are not voltage controlled - there is no relevant parameter to control in the modulators, and the filter is designed as a formant manipulator whose setting will normally remain constant in a given application. As we know, the eight Filter/Oscillators take care of dynamic filtering when this is required. The reverberation units and output amplifiers are all voltage controllable, the former in respect of direct/reverb. ratio, and the latter in gain.

Ring Modulators 1 - 3

Maximum input for undistorted output - 8V p-p to each input.
Breakthrough with 8V p-p on one input only - 5mV p-p (-60dB).

These three identical devices have only one manual control each - output level. As indicated above, the input level should be restricted to 8V p-p for best results, but no harm will occur if higher levels are put in, and in practice unacceptable distortion or breakthrough will show itself clearly. Being a multiplier, two inputs are required, since anything x 0 = 0.

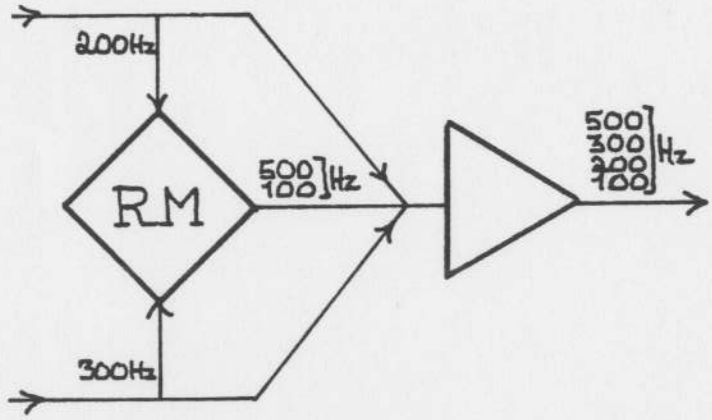
The traditional ring modulator is so called because it is based on a 'ring' of diodes between two centre-tapped transformers, but the advanced design used in S.100 is tranformerless and uses balanced integrated circuits; if run within the design limits it is virtually a perfect multiplier - i.e. if driven by two sines the output will be only the sums and differences of the input frequencies, the inputs themselves being suppressed. The product referred to is not that of the two frequencies but of the instantaneous amplitudes. If you wonder how a product can result in sums and differences, the expression is as follows:
$\sin 2\pi f_a t$ and $\sin 2\pi f_b t$ describe the two inputs, where t is time and f_a and f_b the two frequencies. Multiplying, $\sin 2\pi f_a t \times \sin 2\pi f_b t = \frac{1}{2}(\cos 2\pi(f_a - f_b)t - \cos 2\pi(f_a + f_b)t$. This expression contains only two frequencies - (f_a+f_b) and (f_a-f_b).

There are several distinct uses for a ring modulator, which is one of the most useful treatments in the studio when properly used, and we give typical examples of each main function.

1. Ring Modulator with two inputs, one audio and one audio or sub-audio

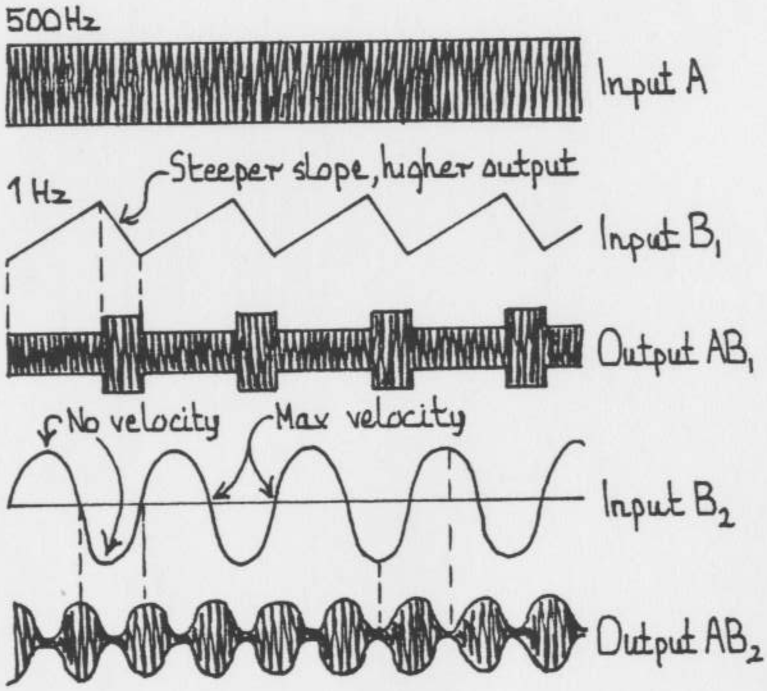
If the inputs are both sines, the relationship is easy to calculate, because only the products (A+B and A-B or B-A, whichever produces a positive number) will result. For example a perfect fifth, say 200Hz and 300Hz, will produce 500 Hz and 100Hz. Try patching this so that you can hear the originals as well as the products. The resulting chord will be a fundamental (100Hz), plus its second, third and fifth harmonic. Fig. 13 shows the block version of this patch, which you can translate into a practical patch on the Signal Board. Now this is a deliberately simple relationship using sine waves, but as soon as the fifth, or any other simple interval, is mistuned, the products become aperiodic to the originals because they are linear shifts in a logarithmic context. So quite small changes in the frequency of either oscillator can cause dramatic changes of timbre.

FIGURE 13



Try now keeping one oscillator steady at, say, 500Hz, and moving the tuning of the other, starting at a sub-audio frequency and gradually moving up to unison with the fixed oscillator. Re-patch so that you listen only to the products and not the direct output from the oscillators. At very low frequencies (of the second input) the modulator will pass the 500Hz in bursts at the frequency of the second input. In fact the output is proportioned to the rate of change of voltage on the second input, so a square wave, for example, will give stabs of sound when it changes direction but nothing in the steady state during each half cycle.
Fig. 14 shows the effect of a ramp and a sine at these low frequencies.

FIGURE 14



Now move the slower oscillator towards the frequency of the other (500Hz). Note that as soon as the second input is in the audio range (around 25Hz) you hear two notes moving apart from their starting point at 500Hz, varying in concordance, of course. When the unison is reached, the difference frequency is zero and the sum is the octave of both oscillators.

FIGURE 15

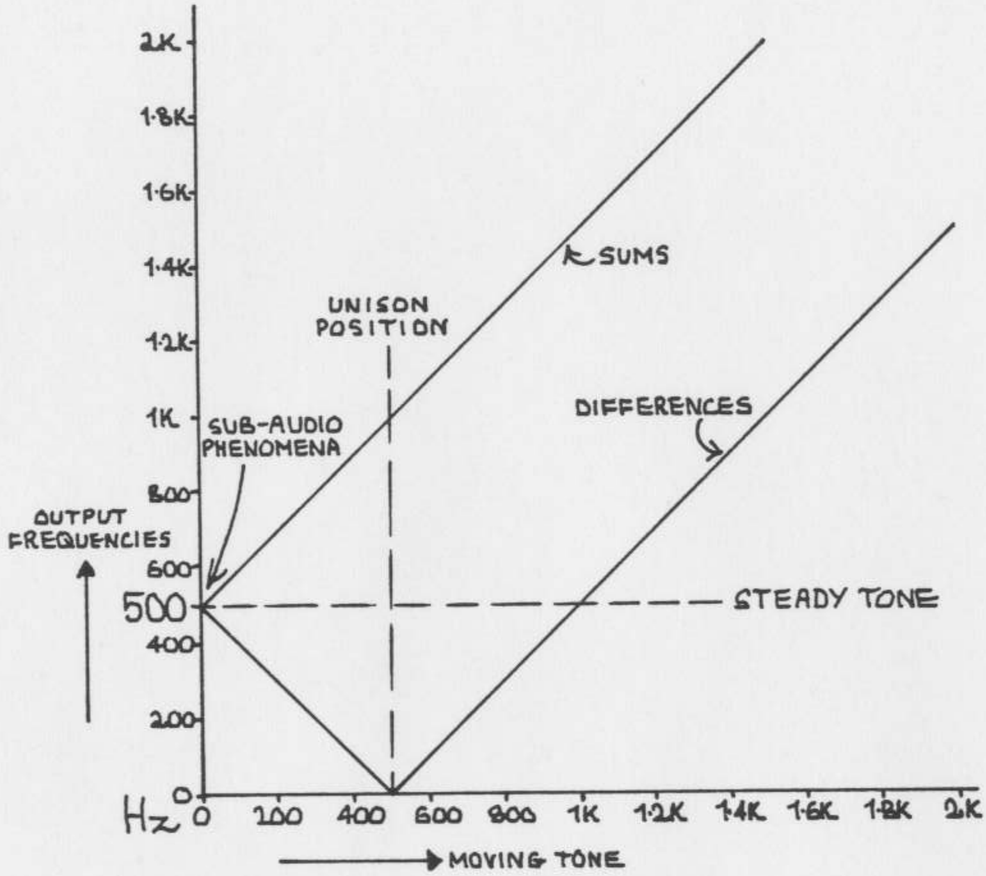
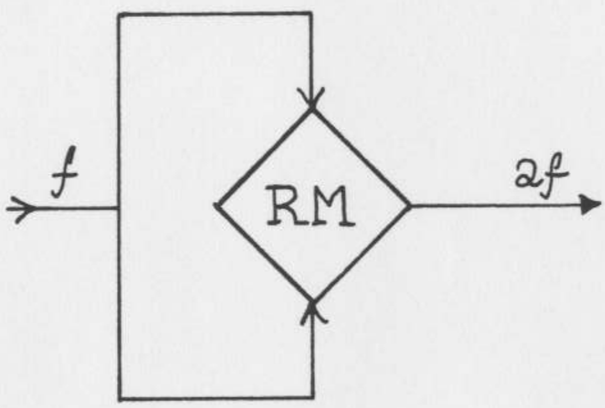


Fig. 15 shows how after this point both frequencies move up at a fixed distance equal to twice the frequency of the steady oscillator. The effect of this linear shift of frequency is to throw up all kinds of aperiodic overtones, becoming again consonant at the points of simple ratio. The overtone structure is even more complex if one of the inputs is not sinuoidal, because the output will also contain the sums and differences of all the harmonics present. This use of the modulator is particularly suitable for chime, bell and gong-like sounds, and you should try some of these sounds, using a suitable percussive envelope, with perhaps some filtering and reverberation. The most interesting sounds are usually to be found when the two oscillators are not too far apart (i.e.

in their contrary-motion area in the first octave), but a large range of effects is possible and can be accurately predicted if the simple rules above are followed. Naturally if both inputs are non-sinusoidal the products become very complicated, and when they are also dissonant the resulting sound can be very dense. The above experiment has also demonstrated a second use for the modulator:

2. Ring Modulator as Frequency Doubler (One Input)

FIGURE 16

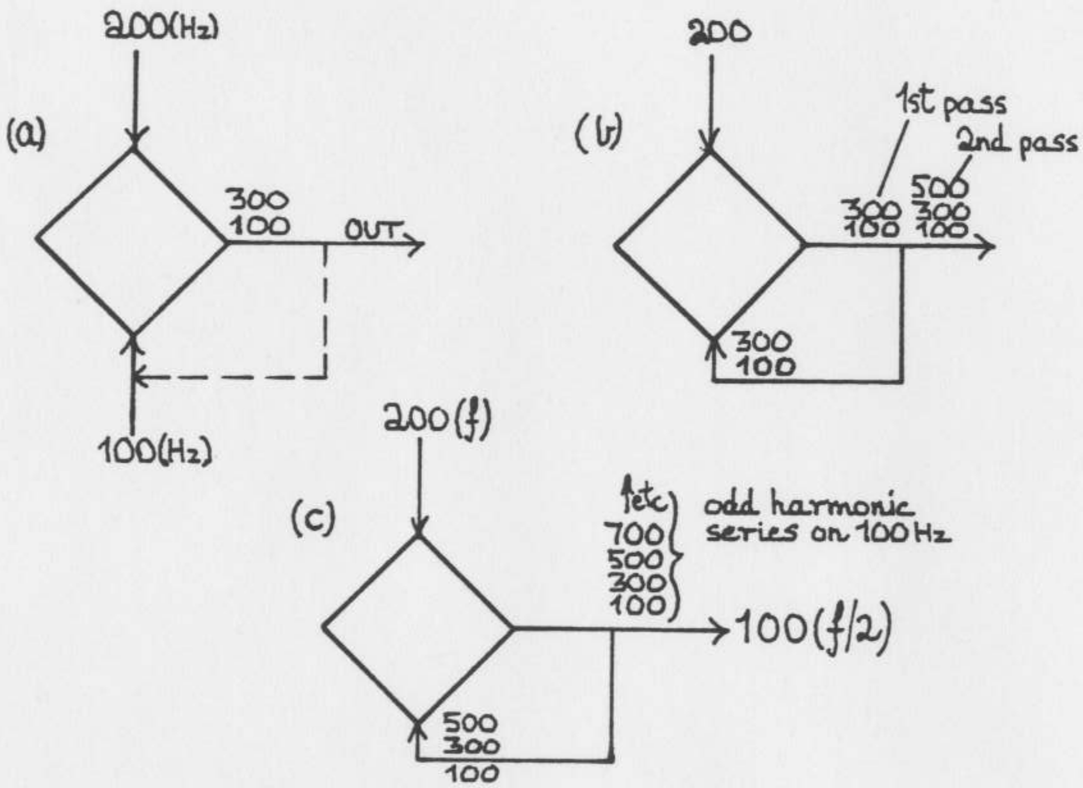


This one input configuration of the modulator gives a perfectly precise octave, though some of the fundamental will also be present unless the input is a very pure sine. If we put the same oscillator to both inputs we automatically obtain the special condition which we found as a transient state above - i.e. Difference Frequency = 0, Sum Frequency = 2 x Input Frequency. If the original is also patched to the output, perfect octaves must occur. It is also possible to cascade two or all three modulators to produce a number of octaves. Fig. 16 shows the way in which the modulator is connected. This mode can be used to create organ-like tones, regarding each doubling stage as a shorter rank of pipes - 4ft., 2ft. and so on. By using a synchronising input to lock two oscillators at another interval, such as a fifth, 'mixture stops' can also be generated.

3. Ring Modulator as Frequency Halver (One Input).

A further one input configuration is shown in Fig. 17, and the explanation for this is not quite so obvious. It produces the octave below the input, but the output is a modified square wave, not unlike clarinet quality. It is often used in live performance applications, where an amplified instrument sounds in the loudspeaker an octave lower than its playing pitch.

FIGURE 17

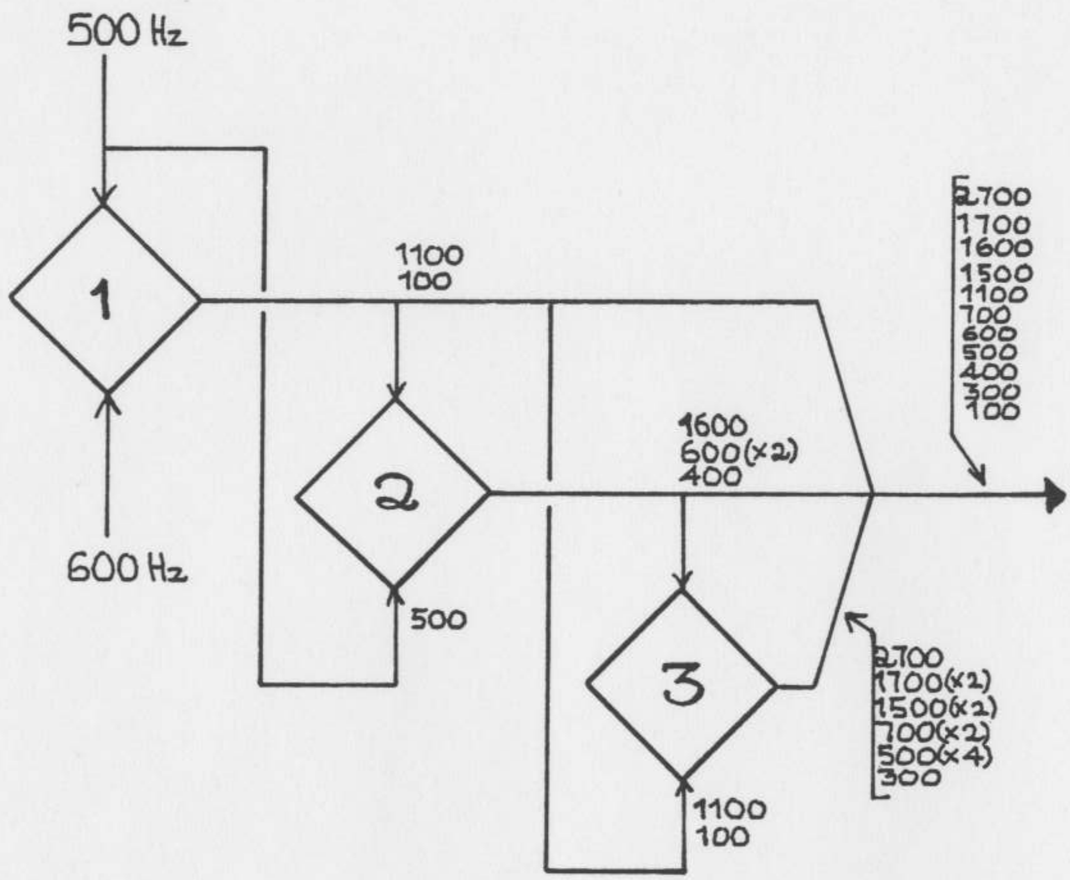


Imagine a ring modulator connected as in Fig. 17(a). The two inputs (100, 200) will produce outputs of 100 and 300. If we suppose that with this signal present we could instantaneously switch the output to the second input, removing the original 100Hz input, the result would be as in Fig. 17(b). Next time round, like (c). You can see that we have an odd harmonic series on 100Hz, or an octave below 200Hz, and the ear will hear the note as 100Hz even though this frequency is quite weak. But, you may say, the imaginary switch of inputs is impossible, and if we simply connect as in (b) how does it ever get going, since a one-input ring modulator produces no output? The answer is that in the first few instants the circuit is completed by breakthrough, and in fact the level of input is fairly critical (and must not be too low) if this configuration is to be successful. You might further object that the obvious breakthrough would be 200Hz, since this is the input frequency, but in fact 100Hz (+300, 500 etc) is the only frequency that will satisfy the conditions to maintain the phenomenon.

The modulators can also be used as switches, because the removal of one input will inhibit the second input as well, and various types of gate can be made which make it in effect another envelope shaper. Experiment with slow oscillators, clock pulses or a joystick as the second input - this last can produce a kind of bowing effect, since the output is proportional to the speed of movement of the stick (this will have to be cross-patched from Control to Signal Board - see below under Output Amplifiers).

As a final example, I show in Fig. 18 the result of taking only two sines, a major third apart, and triple modulating as shown, using all three modulators. You can easily calculate the products for any frequency and waveform input, provided you know the harmonic content of the waveforms used.

FIGURE 18



Right-Octave Filter Bank

Max. Input Level - 8V p-p. Max. Overall Gain - 10dB±1.5dB. Filter Frequencies - 63, 125, 250, 500, 1000, 2000, 4000, 8000Hz. Filter Tuning - Centre Frequencies ± 10%. Overall Frequency Response (all controls at max.) 50Hz-12KHz. Filter Slope - 12dB/Octave. Chan.-chan. cross-talk - -60dB.

The Eight-Octave Filter Bank is a group of eight individual filters arranged in parallel. The unit enables very subtle variations of colouring to be made. It can be placed anywhere in a signal line, but often before the Output Amplifiers. One of the problems of operating a filter in a circuit where levels have been set is that since its action is essentially subtractive, the signal will emerge smaller in proportion to the severity of the filtering. 10dB of gain is therefore built into the circuit to help compensate for these losses, but it must be realised that if a high proportion of the signal is filtered out, as it sometimes will be, there is bound to be overall attenuation which will need making up at a later stage in the chain.

This filter is very simple to use, with one input and one output. It is not voltage controlled, because its purpose is to provide characteristic colourings. The cut-off is sharp enough to permit sections of the spectrum to be virtually removed, making profound changes in the character of a sound. If all knobs are at maximum the signal will emerge unchanged except for some extra level. But unlike a low-Q octave filter, where setting all knobs to the same position would produce an attenuated but still essentially flat response, the resonantly tuned 8-octave filter shows a comblike characteristic at intermediate settings. With all knobs at 0 only a very slight breakthrough of any signal will be heard.

A rich input (harmonically or polyphonically) will yield the most dramatic results. Sine tone, on the other hand, does not benefit at all from filtering since there is no structure to filter. There is also a slight but noticeable resonance effect at higher settings, which can help 'presence' and liveliness.

An additional use for a selective octave filter is as a "signal improver". Hum and other unwanted fixed frequency phenomena can often be removed completely without serious distortion of the general signal spectrum. This can be very helpful if you are trying to process a poor quality input signal.

Voltage Controlled Reverberation Unit

This spring line reverberator has two springs each with delays of 35mS and 40mS. Maximum reverberation time is 2.4 seconds. They are useful not only for the normal purpose of adding reverberation to any signal, but as a further timbre modifier, particularly on continuous sounds where the normal action of the springs is not so obvious as with discontinuous inputs.

Because of the multiple delays it is possible to make a multi-source sound (timbre, chord or unison) more homogeneous. On the other hand any device of this kind is bound to cause some loss of clarity and sharpness, just as a naturally echoey place will do, as well as throwing up resonances due to standing wave patterns in the springs.

The Control Inputs (C1 and 2), provide complete control over the ratio of direct to reverberated signal, and very unusual effects can be obtained by isolated reverberated sounds (or sudden removal of reverberation) during a passage. For example the keyboard could be patched so that the higher the note (or the heavier the touch) the less reverberant the result. As with all the devices we are describing, it is worth spending an hour or two experimenting with every configuration you can think of.

Naturally if the effect of reverberation is to be heard, the unit should be patched last in the chain before the output amplifiers, but used as a timbre modifier it might well be used earlier. The two units can be cascaded or paralleled if required, for extra 'remote' reverberant sounds, or for complex differential control effects - e.g. an "echo-pan" in which the reverberation rather than the signal moves from one channel to another.

Voltage Controlled Echo Unit

This device uses an electronic Delay which unlike the reverb spring, gives only one, clear, echo. This echo may be fed back however to give multiple echoesup to the point where an interesting self-oscillation sets in. The amount of delay the unit provides is adjustable up to a max. of ½ second, and both the mix and delay time may be voltage controlled. The latter control, when fed from a very slow oscillator gives a 'chorus' effect.

Voltage Controlled Multi-Function Output Amplifiers 1 - 8

Fig. 1 in Section 1 showed how the direct and pan output sockets are related. The eight Output Amplifiers are all voltage controllable, as well as having manual slide controls. Control slope is logarithmic at 10 dBV. The control inputs are typically used for dynamic control of the output from a keyboard or other device, when the amplifiers are used in their function of providing the final link in the signal chain. As mentioned in Section 1, the choice of whether to connect to a pan output or an individual output depends to some extent on the peripheral equipment in the studio - e.g. what type of external mixer and patch is fitted.

As well as manual and VC gain, an output filter is provided, giving a continuous transition from first order low pass to first order high pass with a central level response position. It is very important to remember to return these filters to the central position if they have been set differently for any reason because they will affect any output passing through that channel and may impair the effect of earlier treatments. But they are very useful as supplementary non-voltage controlled filters and often used in conjunction with the eight octave filter bank.

The meters above the sequencer controls (top panel) are related to the channels over which they are mounted, and show output level when switched to Signal.

As well as the signal output function (for which only occasionally would all eight amplifiers be used), these amplifiers have an important internal function. They are direct coupled, which means they can carry DC controls as well as AC signals, and the OFF switch fitted above each fader disconnects the pan control and enables the channels to be used internally without producing unwanted sounds in any outputs that may be patched externally.

Typical uses are for cross-patched signals and controls, and for this purpose all eight outputs are duplicated on both boards. These are equivalent to the individual output sockets. On the Signal Board all eight inputs are available, and some of them will of course be used to take signals out of the studio. Others, however, can be used to cross-patch signals or controls to the other board (with extra gain available if required).

When the signals (or controls, more often) are slow, switch the output meter to Control, which makes them centre zero DC meters (reading bi-polar controls) rather than left-hand zero AC meters.

The control inputs can also be used with audio frequencies to give audio frequency amplitude modulation (reminder - a ring modulator gives AB only, an amplitude modulator gives AB+A or AB+B depending on which is the signal and which the control).

The slider pots used are not controlling the signal but the control voltage, and will therefore not go noisy even when worn.

SECTION 3

SECTION 3

ANALOGUE CONTROL DEVICES AND IN-BUILT PERIPHERALS

Control Devices (Sources and Treatments)

- Filter/Oscillators (2)
- Envelope Shapers (2)
- Sub-audio Oscillators (2)
- Double Keyboard
- Joysticks
- Random Generator
- Slew Limiters

Filter/Oscillators (2)

The high pass Filter/Oscillators (5-8) have no very effective use as control devices, but there are occasions when the four low pass filters are useful. In oscillatory mode, the low end of the frequency range (extending down to about 5Hz) can supply a vibrato (f.m.) or tremolo (a.m.) input. The appropriate control output must be cross-patched by using an output amplifier, and applied on the Control Board either to an audio oscillator (f.m.) or an output amplifier carrying a signal (a.m.).

A more interesting use involves applying a random vibrato or tremolo by filtering noise at a very low frequency. Patch a Noise Generator to a low pass filter and set the response about halfway between zero and the oscillation point, also setting the colour control on the noise generator to low pass (counter-clockwise). By using a similar patch to that above, a variable speed tremolo or vibrato gives a quite different effect from constant speed modulation. By adjusting the levels of both noise and filter outputs and the response and frequency settings of the filter various random undulations can be caused which cannot be made in quite the same way by, for example, the Random Voltage Generator (q.v.). But except for these rather specialised uses it is rare to find the filters as control sources or treatments.

The six outputs can be used for a large number of control functions, but are particularly suitable for those not requiring a trigger, since these are free-running oscillators. F.m. or a.m. of a signal, dynamic control of a filter or envelope, changes in slew rate, automatic staircases, etc.etc.

If two or all three of the oscillators are used on one control input, a variety of very long term changes can be put in, of a quasi-random nature, and these can be further randomised by using the Random Voltage Generator (q.v. below in this Section) to control the frequency of the slow oscillators. Even without additional randomising very long repetition cycles can be obtained - for example if the three oscillators are tuned in a frequency relationship of 27, 29 and 31 secs/cycle, exact repetition of a given function will not occur for well over 6 hours! This is an extreme case, of course, but ostinati of a fluidly changing nature can be very expressive. It is also possible to use these free-running controls in conjunction with trigger-locked controls such as the Trapezoids, and a cross-influence can be set up by using a slow oscillator to trigger a shaper (via C3, 8 or 13).

Oscs. 10-12, like the others, have synchronising inputs, but unless they are used in the audio part of their range (see Sect.2), the method of synchronising is different in practice and quite difficult to set up at very low frequencies. Put a Joystick control to all three of the slow oscillators and set it in the middle. Take the ramp outputs to a loudspeaker via one of output amplifiers 1-4, and starting with Oscs. 10 and 11 only, tune manually to give about 0.5Hz from both. Turn up the square output of Osc.10 and cross-patch it (another of output amps. 1-4) so that you can reach the synchronising inputs on the Signal Board. If you have tuned closely, you will have two sets of clicks which gradually change phase with each other. Arrange for Osc. 11 to be a fraction slower than Osc. 10 if anything - since the synch. input raises the frequency slightly. Wait until the clicks are coming together naturally, and insert the synchronising pin (Osc. 10 to Osc. 11) just before the clicks. Assuming an adequate output from Osc. 10 square to synchronise, they should now lock, and the clicks will remain together. By the same procedure lock Osc. 12 to the other two (it will also be possible at half or twice the frequency). If you now move the joystick down, the oscillators should remain locked at a much lower frequency. Fig.20 illustrates the sequence, and also shows how a complex combined waveform can remain in synch. with itself.

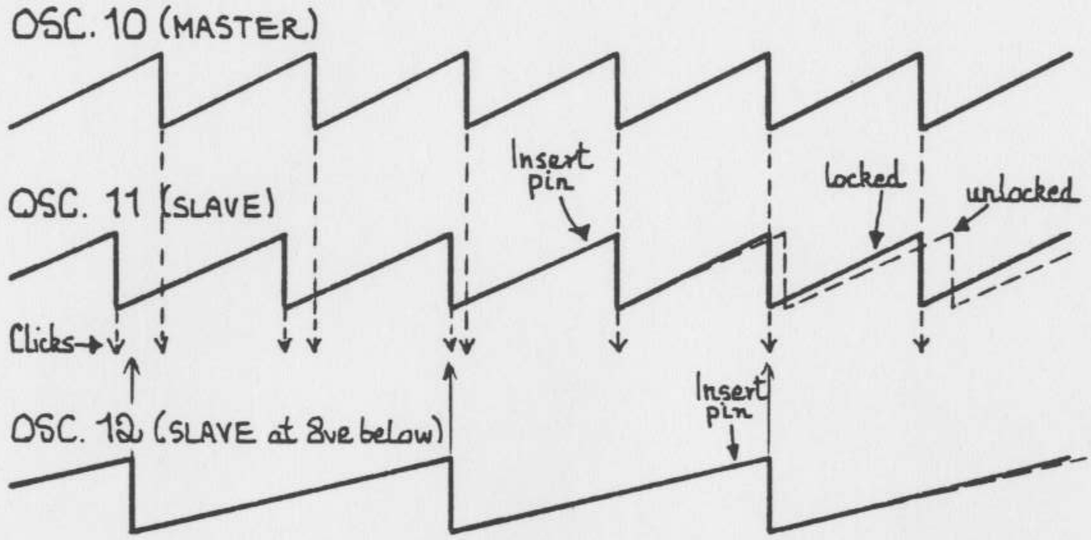
Dynamically Proportional Two Manual Keyboard

WARNING! Do not plug in or unplug the keyboard without switching off the Synthi 100.

This consists of 2 x 5 octaves, each having three outputs: (1) a 'pitch' proportional output - i.e. precisely equal voltage divisions from each step of the keyboard, (2) a dynamically proportional 'velocity' output which responds to the speed of depression of a key; (3) a 'key' or trigger, which sends a voltage whenever a key is depressed. This voltage remains as long as the key is held down.

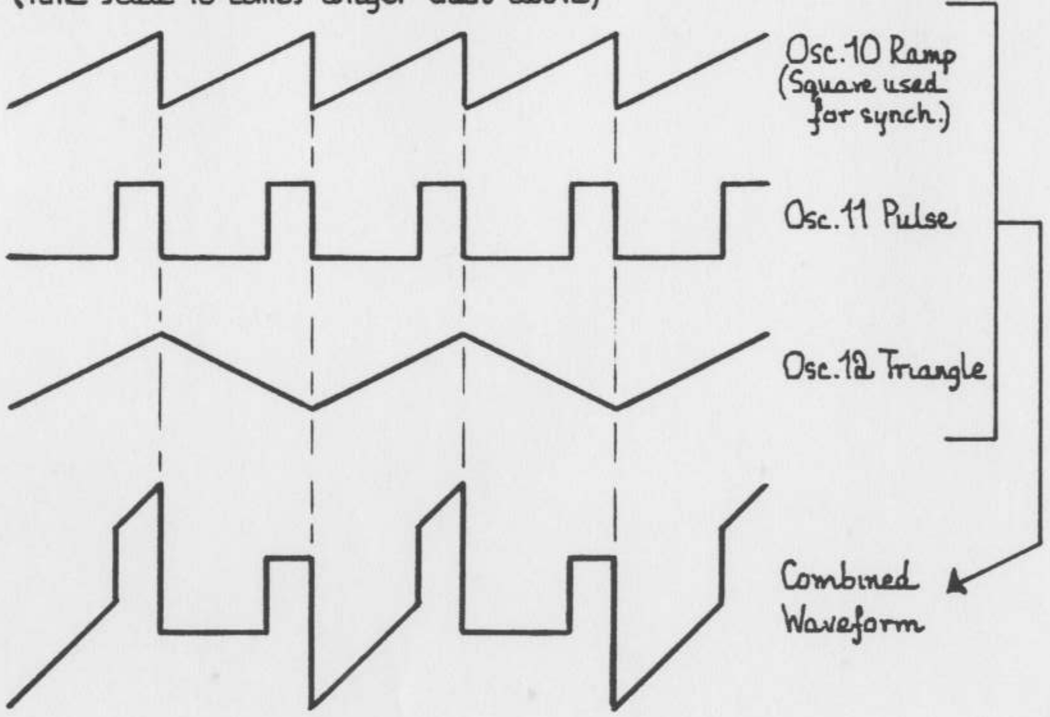
There are two controls on the Keyboard itself, which merely requires plugging to the socket provided, towards the right hand end of the rear connection panel. The lead can be taken through the tunnel and brought out from the back of the S.100 if the keyboard is to be mounted fairly close, or from the front if the keyboard is to be used some way from the main instrument. It is not a difficult matter to make an extension lead if necessary so that the keyboard can be operated more remotely. A good practical position, particularly if the sequencer is being used in conjunction with the keyboard, is at right angles at the right hand end of the console, making a working corner with the keyboards and sequencer controls all to hand.

FIGURE 20



Final waveform, lowering pitch with joystick and selecting alternative shapes:

(Time scale 10 times longer than above)



The keyboard range controls can be found on the right hand vertical panel, clearly marked with function. The pitch outputs have slow-motion controls, and give a IV/octave output when set at '9' on the major scale. Upside down pitch is not normally wanted, of course, but it is important to realise that the keyboard may not be used for pitch control at all, and there will be occasions when an inverted control is essential. To accomplish this, an inverting/non inverting buffer has been included. This has a bi polar gain control, an offset control which simply adds A negative or positive D.C. offset to the input signal as required.

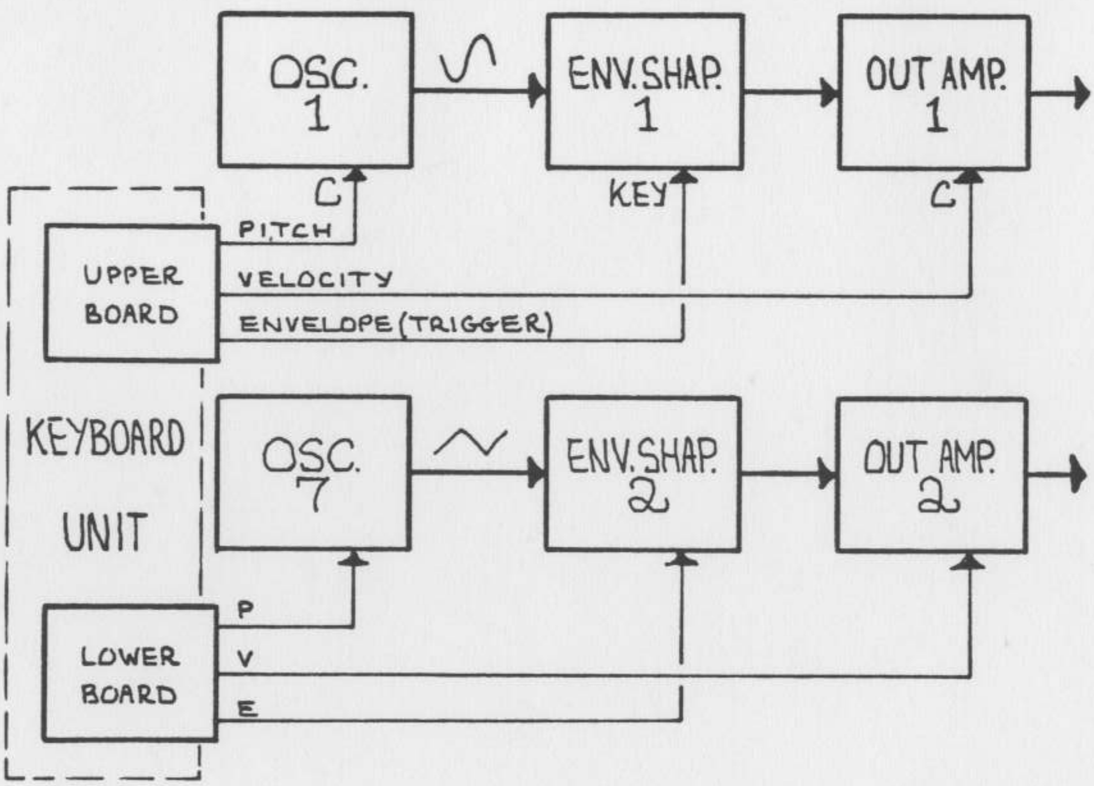
The velocity output has a possibility of having no effect (knob at or near 5), or a progressively large velocity - proportional output which can be normal or inverted (the 'louder' the 'softer'). This too may seem an unlikely requirement, but although the velocity often is used for actual dynamic control, it may very well be used for filters, reverberation mix or some other influence possibly needing an inverted control. Both 'pitch' and 'velocity' have memories in the sense that when the finger is removed the two outputs will remain at the levels last selected until this instruction is cancelled by striking a new key.

The trigger or 'envelope' output is also variable and invertible, but for most internal purposes it can be left set at +5, which means that it is in a normally zero state, going positive whenever a key is pressed. This output has no 'memory', and would indeed be very hard to use if it had, because the delivery of staccato by finger touch would be impossible.

The rear keyboard switches we will explain below, because it is easier to understand its use when a patch is plugged. Make up a simple keyboard patch as in Fig. 21.

I have illustrated this as a block diagram only, because it is useful to practise translating your own block diagrams into practical S.100 patches. Find the keyboard outputs and distinguish carefully between the control and the signal part of the patch. Always use 100k pins (white) to patch the keyboard to the control inputs, or the spread possibility is reduced. Suggestions: Do the signal patch first, and set the Envelope Shapers in Free Run. Make sure you are getting some sort of tone through to the output from each oscillator. The sine and triangle will give very 'plain' sounds, but distinguishable from each other. You will need only four pins for this side. Now patch the controls (six pins). Assuming that we want two ordinary scales, in tune with each other, try each keyboard for pitch, and note whether you are getting less or more than an octave when you play an "octave". For tuning purposes the 'pivot' note is middle F-sharp. Since we have a bipolar control voltage the middle note of the keyboard should move very little if at all.

FIGURE 21



Using the pitch control knob, tune each keyboard to give a true octave up and down from the middle F-sharp. Then check the C's right through and adjust for best tuning. The setting on the pitch control should be around 9 on the dial. Now go to the two oscillator tuning knobs and tune for unison at middle F-sharp (or an octave if you prefer) - again checking other unisons right through the keyboard.

For the moment put velocity outputs at zero (5 on the dial). If you have left the Shapers on Free Run the notes will have been coming and going during the tuning procedure. Put them to Triggered, and the Key Release switch on the keyboard down to Key Release or New Pitch. Play some notes and adjust envelope Shapers for the note shape you prefer; also adjust the six relevant level controls (two each of oscillator output level, envelope shaper output level and output channel level) to give a reasonable mezzo-forte without distortion. Rule: Don'toverload the next stage - i.e. set output channel to give reasonable speaker level, shaper probably to maximum, oscillator levels not too high for shaper input. Now increase the velocity output control until 'normal' playing dynamics give you normal results- i.e. a forte touch a forte sound, and so on. Don't forget that the envelope trigger should be at +5.

You now have a very simple keyboard patch with which it should be possible to play ordinary music in two parts. Try changing the range, the envelopes and the envelope mode switch (already discussed in Sect. 2). Add filters, add more dependencies and controls - in fact exercise imagination in building up from the simple patch to a more complex and interesting one.

To return to the Key Release switch - for detached notes, particularly when the Envelope Shaper is set to triggered, the up position of the switch - Retrigger Key Release - should be selected, but the keyboard key must be released before another one (or the same one) is played or a new key will not be sent. This means that in triggered either Fig. 22 (a) or (b) will result in the first note only triggering the Shaper - and if the Shaper is set short, only this note will be heard.

FIGURE 22

In the down position - Key Release or New Pitch - the notes can be detached but need not be when the pitch is ascending, which makes possible much more legato playing. Looking at Fig. 22 again, the Shaper will retrigger if (a) is played, but not if (b) is played. The same rule applies if you play a chord. The voltages given are those of the upper note, the others being ignored.

FIGURE 23.

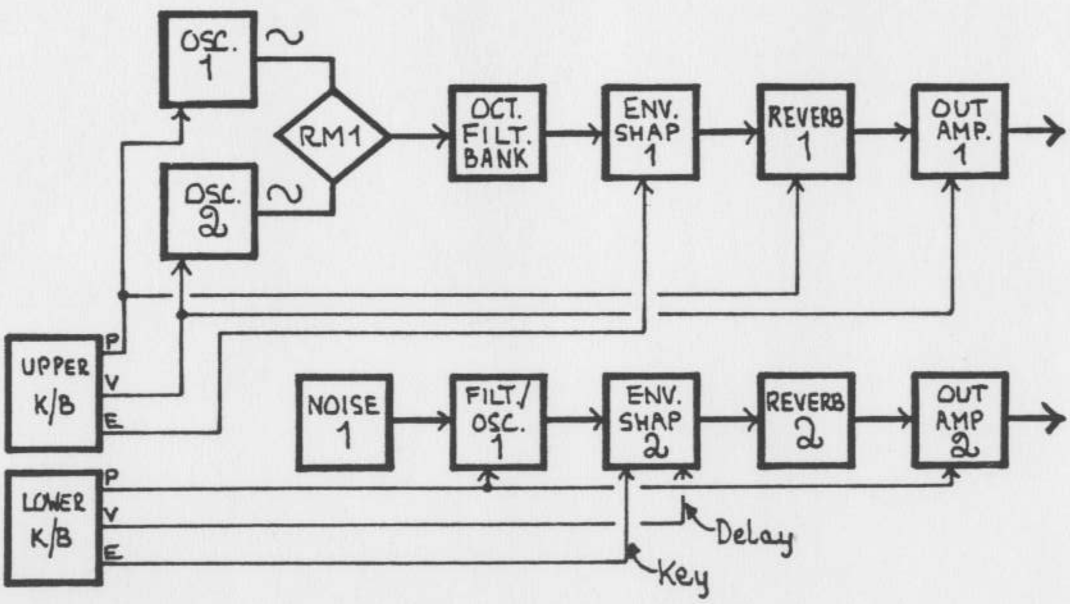


Fig. 23 shows a different and slightly more complex keyboard patch, demonstrating the use of the keyboard in other than straight melodic applications. This pitch will yield a large variety of sounds, depending on the oscillator frequencies selected and control settings, etc. There will also be big changes if some of the keyboard voltages are inverted. Try it first as we show it, then vary the controls - e.g. try putting Pitch and or Velocity to different envelope time parameters. Put in more filters. Bring in Oscs. 10-12 to give vibrati, etc. Use the Trapezium outputs on various control inputs. Basically you will get chime or gong-like sounds from the upper keyboard, and filtered noise in various forms from the lower. On the lower keyboard, try Multi-Cycle on the Envelope Shaper (the 'louder' the faster); also try tuning Filter response to nearly resonance or actually into oscillation.

Once the general principle is understood, the keyboards are very simple to operate, and they have an important function as inputs to the sequencer as well. Occasionally, particularly in dirty atomospheres, the contact wires may become dirty, and their adjustment is very delicate - the velocity output depends on measuring the time lag between the closure of two successive contacts. For cleaning, take the bottom off the keyboard unit and spray with switch cleaner while vigorously operating the keys. Do not soak, however, and never wipe or otherwise physically handle the contact wires.

Joysticks 1 and 2

Max. output ±8VDC x 2 from each Joystick. Each of the two sticks gives two steady DC voltages, one on the X and the other on the Y axis, and both voltages can be changed simultaneously by moving the stick on any angled course. We have already given various uses for them in the course of other descriptions, but in general their applications fall into two kinds - static or dynamic. In a static use the joystick voltage is applied as a permanent offset to another voltage, biassing a symmetrical bipolar or other DC voltage so that it moves bodily in a positive or negative direction. Or it can be used alone to put a standing DC into any control input.

When the stick is dead centre both voltages will be zero, and the two range controls determine how much change will occur with a given stick movement. For example you may have set Osc. 10 to give an amplitude modulation on an output channel, and the shape is right but the range of control is not - e.g. it might cut off completely before you wish and not be loud enough at the maximum end. Increasing or reducing the amplitude gives an undesirable effect at one end or the other, so what you need is to move the voltage swing bodily sideways. Put a joystick near the centre, plug one of its outputs to the same control input as Osc. 10, and move the stick to correct the dynamic range - you may find you also have to adjust the level of Osc. 10. If a very small movement of the stick results in dramatic changes it is hard to control, and you should reduce the range appropriately. This is a static use because once the right position has been found the stick is likely to be left alone until the operation in hand is completed. Another static use is when you wish to change a whole group of controls together, preserving their relative proportions. This could apply to a group of oscillators or filters. If patched to all the time parameters of an envelope shaper the joystick enables you to set up shapes using a fast envelope and slow all the time while preserving the chosen shape. Here again the joystick will probably be left alone when set, and in many such static applications you will only need one voltage at a time.

Dynamic uses include the changing of a large number of controls with predictable results and with only two hands. This is a typical live performance requirement, and by 'dynamic' we imply the movement of the stick in real performance time, so that its effects are heard as part of the music. When using both X and Y outputs it will be noted that the stick actually moves in a square, though the holder is round, and it is important to select the right range setting - e.g. the controls affected by X may need a much larger range than the Y controls, and you should also try different groupings on the two outputs when several different influences are being put on. Let us take two examples:

(1) The keyboards are both in use, with very different patches, and with certain automatic controls which are trigger-dependent (e.g. Trapezoids). But you want variable timbre and variable vibrato when using the upper keyboard, manually controllable and not trigger-dependent. Put (say) X to two Filters, Y to a slow oscillator. Different stick positions will now enable you to select any combination within the ranges set, using one hand for the keyboard and the other for the stick. Probably neither voltage range will need to be large in this case, but check extreme positions because it is useful in live performance to know what will be the effect in easy-to-find settings, particularly if you cannot look at the joystick. These settings are the four corners and the excursion along each side. Middle positions are harder to judge, but with practice you can use your fingers against the bowl as a gauge. The lower keyboard could have another set of dependencies on the other joystick.

(2) You have written (digitally recorded) a sequence in the memory and are thus free to sit back and listen, but you want to vary the effect of the sequence in certain ways. Manual controls are therefore put in from the joysticks, and since both hands are free you can choose a greater number of 'safe' corner and side positions. Another point is that as the joysticks are conveniently sited near the sequencer controls, you can alter clock speed and/or sequence direction without moving.

By rapid movements of the joystick it is possible to 'play' a portamento tune in the manner of a saw, or put in an effect like wah-wah (filter control at about 1-2Hz) manually. Cross-patched to one side of a ring modulator, an oscillator can be 'bowed', since the output will be proportional to the speed of movement of the stick (see Section 2, Fig. 14 and text). In fact there are many uses for this very simple device. I have found that so many excellent dynamic controls are already available in S.100 that I use joysticks more for setting up and static offsets then anything else, but each user will find his own favourite applications.

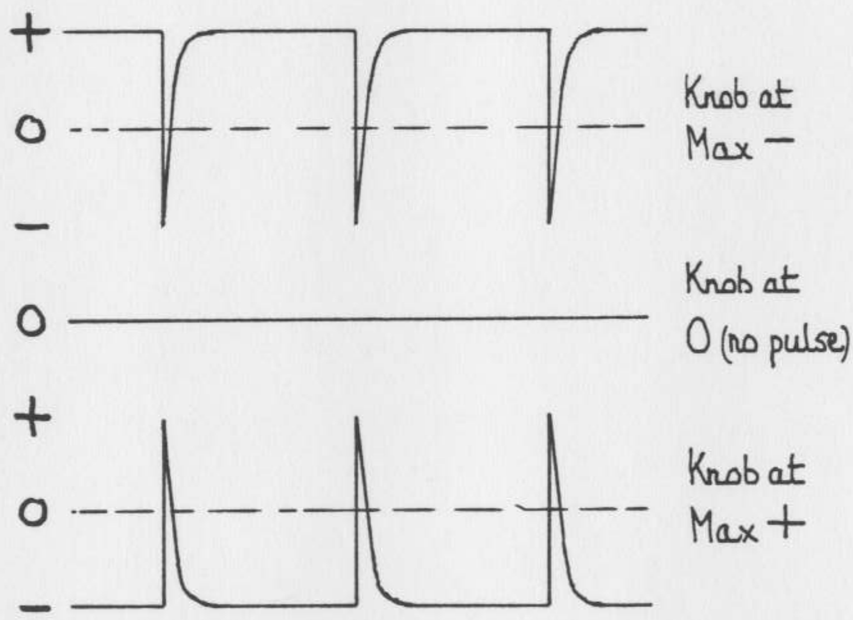
Random Voltage Generator

This consists of a free-running staircase source, capable of a wide variation of mean rate, with adjustable time and pitch variances. There are three outputs, a trigger and two voltages. Note that unlike the keyboard outputs the trigger comes above the other on the Board. Because the 'steps' of the staircase may be very long and very short intermixed, it is often meaningless to refer to a 'frequency', and for convenience we call each step an 'event'. The two main controllable parameters (the key is also controllable but not in a probabilistic way) are Time and Voltage.

Time: Look at the two lefthand controls, and set VARIANCE to zero. Set MEAN about halfway and plug the KEY output to an output amplifier and speaker. You should now hear a regular series of clicks. Moving the Mean control will alter the speed of the clicks, and the range is from approximately 0.2Hz (5" per event) to 20Hz (50mS per event) - at the upper end the clicks merge into a low buzz. With Mean again halfway, gradually increase Variance; the clicks become slightly and then very irregular, till at maximum variance you have long gaps, normal clicks and clusters of very short events. Actually it will seem on average slower, because you notice the long gaps more, but the mean rate is still the same - i.e. a sufficiently long sample will produce the same total. In fact the odds against any new event being the same length as the last vary from a certainty (knob at 0) to over 100:1 (knob at 10). We thus have a controllable degree of randomness or conversely predictability, and correctly used this can be a very powerful instrument. We can introduce small irregularities in an otherwise highly regular group of sounds, or we can set it to produce endless varieties of wildly random patterns. There is a limitation, however, in that the clock can only work within the design range of 0.2 to 20Hz, and the full variance will be obtained only if the Mean setting is near the central position. If you require predominantly slow or fast events you move the Mean control accordingly, but of course the variance may never reach the limit at the opposite end, thus reducing the odds.

The key output (righthand knob) delivers a pulse whenever an event occurs, and this pulse is short enough (less than 20mS) to make sure that even the shortest events generate a discrete trigger. The amplitude and polarity of the trigger is adjustable as with the other key outputs, and in most cases +4 or +5 will be the correct pulse to use; the wide choice of key available provides for occasions when the S.100 may be interfaced with voltage controlled equipment of other manufacture. Fig. 24 shows how the trigger pulse changes with different settings of its output control.

FIGURE 24



At intermediate settings steady state and pulse maxima close up towards 0

VOLTAGE: The random voltage part of the generator delivers a staircase (i.e. each event a constant voltage) which changes level whenever it receives a clock pulse, so that its randomness is independent of that of the clock. The two voltage outputs are independently controllable, but of course driven by the same clock, so they are time-locked. At zero, there will be no variance, and an output voltage of zero. As the control is turned up there will be excursions in both directions (statistically equal in both polarities but not alternate by any means), and the excursions vary in size up to the maximum of ± 2.5V. This large range of 5V will give huge random leaps if connected to an oscillator - indeed the staircase disappears at both ends into inaudible regions.

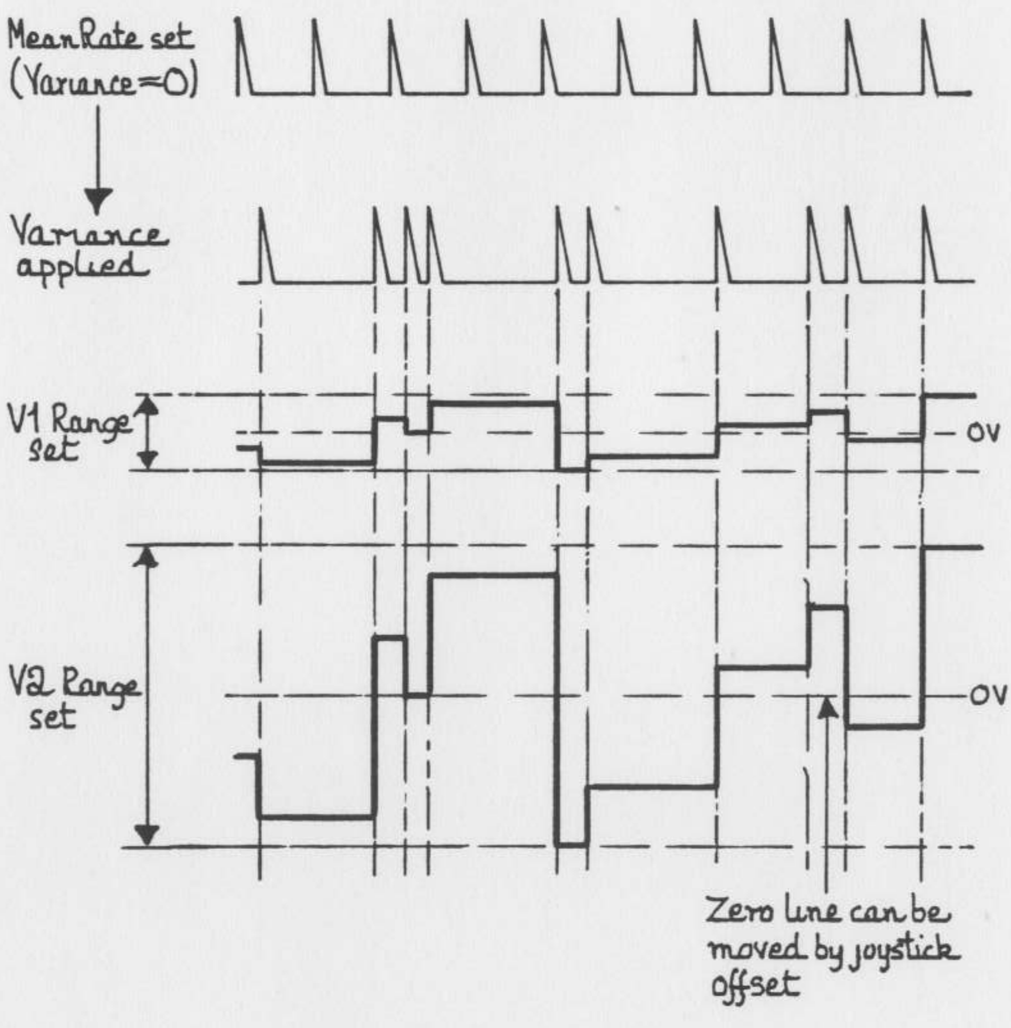
Since the voltage generator will produce <u>any</u> voltage within the range set, estimation of chances depends on the degree of resolution which is relevant, and no firm figure can be set. For example if the control input is audibly sensitive to a change of 0.01V the odds against a particular effect turning up will be ten times greater than if the user only notices a change of 0.1V. But having mentioned the large steps possible, it is also worth saying that a very useful function of this device is to produce tiny, micro-tonal variations in the character of a 'natural' vibrato. It can also be set, or course, to produce endless 'tunes', but because of its infinite variability it obviously cannot be set to play only notes in a particular key, for example. The distribution of both time and voltage chances is rectangular rather than Gaussian, giving an equal loading to the chances of any event occurring within the limits set.

To return to the key. Because it is short, you cannot treat it like the keyboard trigger output, which can be made any length you like. For this reason "Hold On" is not an advisable Envelope mode to use, because unless Delay and Attack are very short you will never hear the sound. So "Single Cycle" is the better mode to choose. You can also apply random voltages to the envelope time controls, but a suitably slow staircase must be used or the changes will not be effective.

As mentioned in the next Section, the RVG can be used as an input to the Sequencer. In this case a randomly generated sequence is 'frozen' in the memory and can be repeated at will. This effectively alters the philosophy, because whereas in normal use the RVG output goes by in ever changing patterns, by storage we have the ability to select any sequence permanently, and may apply many different uses to it at leisure.

Experiment for yourself, at all rates, variances and voltages, and with different devices. One 'trick' is to use the key (by cross-patching) as an input to a ring modulator, the other input being an oscillator controlled by the random voltage. The effect is a kind of quasi-pizzicato caused by the gating of the modulator by the trigger pulse. Fig.25 summarises the action of the time and voltage controls; note that the voltages can easily be made unsymmetrical about OV by the means of a joystick offset.

FIGURE 25



Slew Limiters 1-3

The function of these three identical devices is to convert a sudden change of voltage into a gradual one - to produce, in pitch terms, a portamento or glissando. So far, the portamento devices we have considered are the slow oscillators, the trapezoids and the joysticks. Only the trapezoids are keyed, and for various reasons (such as already being linked to an envelope) they are not well placed to modify a staircase in the kind of versatile fashion we want. The slew limiters can be, in effect, under control of the sequencer.

'Slew Rate' is the slope of a change of voltage from one steady state to another. In an ideal square wave or staircase generator the transition from one step to another should take no time at all, but this is of course impossible. All the same, devices like the keyboards and the random generator can alter a pitch through several octaves without perceptible portamento. The object of the Slew Limiters is to produce a deliberate and controllable rate of change between one step and the next.

Only one control (Slew Rate) is necessary for each device, and these are situated under the Sequencer output controls on the righthand panel. They are placed here because they are often used in conjunction with the Sequencer (Sect.6), but there is in fact no hard wired link between the devices and the Sequencer. They are quite independent and you can if you like use Slew Limiter 1 with Seq.Layer 3 etc., and the uses we describe in this Section do not involve the Sequencer at all. The division of function is emphasised in the printed Dope Sheet, where the Sequencer controls are ruled off from the others by a heavy line.

Each Slew Limiter has two inputs and one output. All refer to control, but the voltage inputs and the outputs are for the control voltages whose slew is to be changed and slew control is for further control - of the slew rate. So we have another situation of the controller being controlled; and if this is not quite clear, it will become so when we experiment a little.

Pin a normal melodic keyboard patch, such as that in Fig. 21. Get it working, then change the pitch control pins. Move them across so that the keyboard pitch outputs go in to SL1 and 2 voltage inputs. Now connect SL outputs to Oscs.1 and 7. Set SL1 and 2 slew controls at 0 and play again. There should be no difference. Now play fairly wide intervals - octaves or more - and gradually advance the slew controls. The leap becomes a portamento and eventually a slow glide from one note to the next. The reason why I suggested wide intervals is that the larger the leap the more obvious the slew. It is in fact a slope which we can measure in volts per second, or if talking in pitch terms, octaves per second or its reciprocal. With the slew control at 10 you should have a slew rate of about 3S/octave. Leaving the manual controls at 10, pin the left joystick (both X and Y) to the slew control inputs. Set joystick ranges at maximum. By moving the joystick you can change the slew rates differentially, and you will get very slow rates of well over 1 min./octave at maximum.

Slew rates of this order are only rarely needed, but to further examine the action, set at about 4S or 5S/octave (check by playing an octave with a stop watch - it is also good aural training to judge when a slow glissando has reached the octave). Now play notes all over the keyboard at a faster rate than will allow the pitch to reach its prescribed destination. The voltage will change direction and <u>point towards</u> the new voltage even though it may never go there. The important thing to understand is that the SL is looking back to the <u>last</u> voltage, or at least to where it is at when you select a new 'target' for it. So if you guess at the note it has reached and play that note as the next selection it will merely stay where it is. Fig. 26 shows the 'tendency' action of the SL, making a simplified assumption that a staircase consisting only of octaves and multiples of octaves is being used. Note that as the Slew slope becomes flatter, approximation to the original staircase becomes less and less accurate. In the last example we assume a four-step control change possibility, ranging from $\frac{1}{2}$ to 4 octaves/second, and labelled 1-4 on the diagram.

The Slew Limiters can be used in very subtle ways, and we now give a larger patch in which various control possibilities are exploited. This will also give you practice in relating a block diagram to functional operation and to the practical patch.

The block diagram is analysed as follows: Lower keyboard pitch is sent to O.7 through SL1, whose slew rate in controlled by the velocity output of the keyboard. Since O.7 forms one half of a Ring Modulator input (RM1), the products will contain a touch-dependent portamento (gentle touch for slow). The effect of RM1 is also changed by the joystick controlling O.1. The rest of the signal chain is Filters 1 and 5 (making a bandpass filter), Envelope Shaper 1, Reverb.1, Octave Filter Bank, and Output Amplifier 1. The low pass section of the bandpass (F1) has a timbre slew (SL2) which is velocity dependent but also rate controlled by pitch, i.e. the range given by touch will vary with the pitch of the note played. F5 is controlled through SL3, whose slew rate is slowly changed under the influence of O.12 triangle, which also controls reverberation ratio. O.10 square provides the input of SL3, so it only has two 'destinations' but approaches them at different speeds according to the state of O.12, whose frequency is itself controlled by Trap.1 and is therefore envelope dependent. O.10's frequency can be changed by the joystick. O.11 is affecting the Decay and Attack times of ES1, and has its frequency controlled by keyboard pitch. Thus keyboard pitch (via O.11, ES1 and Trap.1) has an indirect effect on O.12. There is also a secondary control to F5 from Trap. 11, and a normal dynamic control from velocity to OA1.

Finally it is worth mentioning an audio possibility of the Slew Limiters. They can be used as low pass filters on square or pulse tones of low frequency, converting them into triangles with an interesting character. Connect a joystick to Osc.10 so as to give the highest range of frequencies, and set at around 150Hz. Take Osc.10 square output to a Slew Limiter, and the limiter output to one of outputs 1-4 to obtain an audible sound. Patch it also to scope Y_1 and examine the waveform.

Depending on frequency, even with slew control at 0 there will be some slew noticeable, and increasing the slew control will gradually convert the square into a triangle (via a trapezium). Continuing further will merely reduce the amplitude of the triangle, of course. Move the joystick to a very low note, and experiment with timbre control by means of the Slew Limiter control input.

FIGURE 26

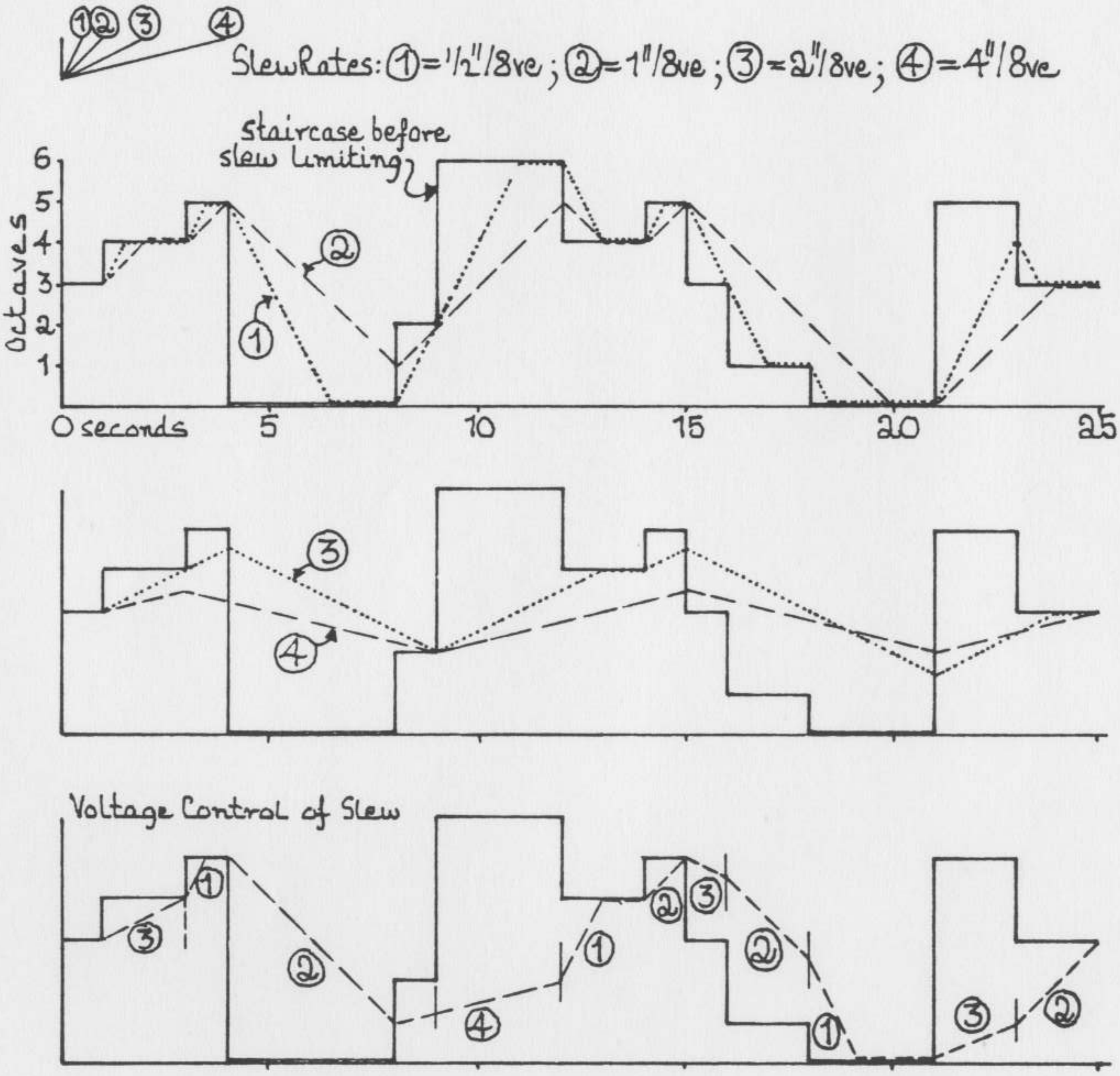
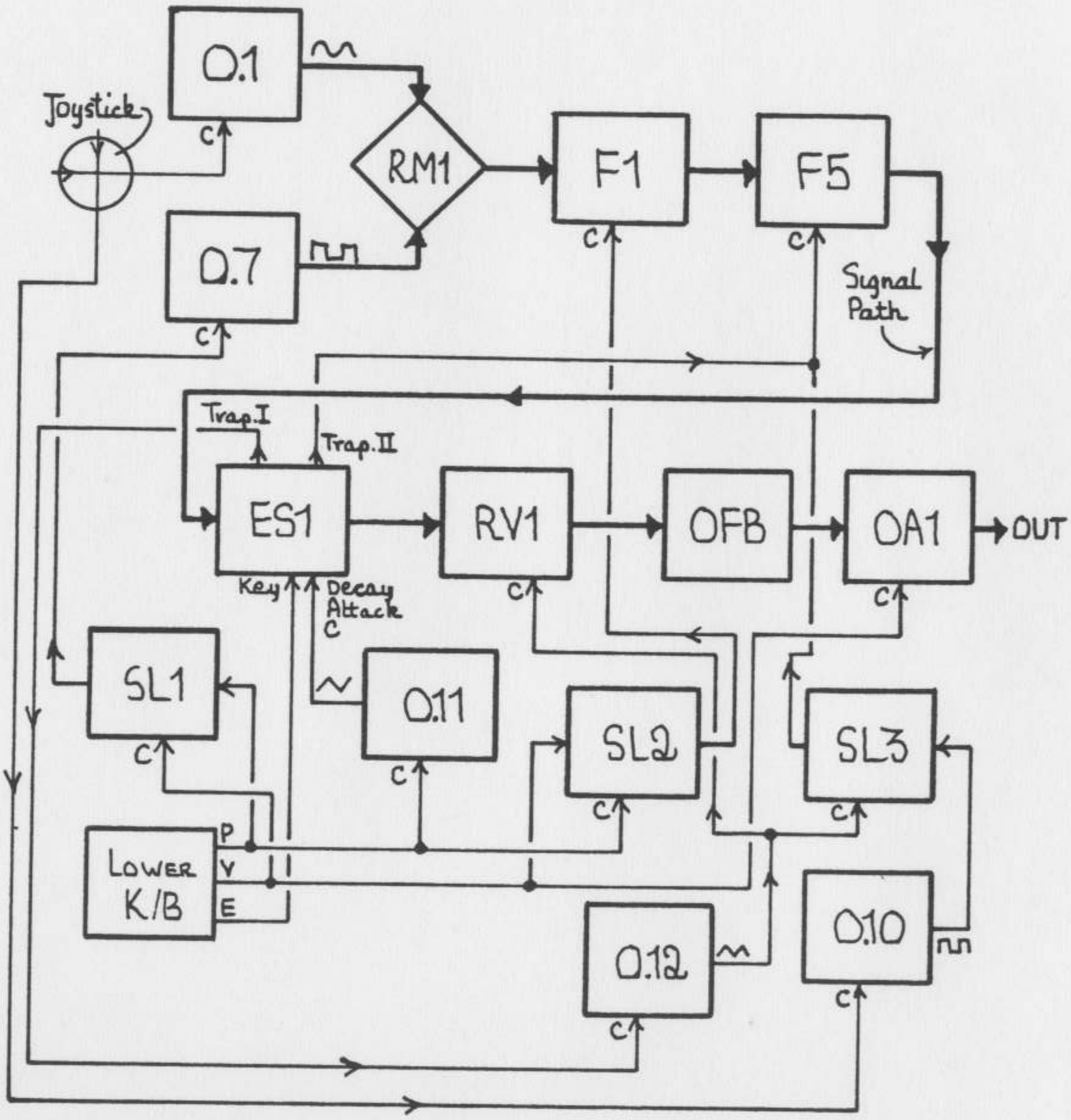


FIGURE 27


O.1
Joystick
C
RM1
F1
F5
O.7
C
C
C
Signal Path
Trap.I
Trap.II
ES1
RV1
OFB
OA1
OUT
Key
Decay
Attack
C
C
C
SL1
O.11
SL2
SL3
C
C
C
C
Lower K/B
P
V
E
O.12
O.10
C
C

IN-BUILT PERIPHERALS

Treatment Sends and Returns
Oscilloscope
Digital Voltmeter (when fitted)
Frequency Meter
Store Unused Meter (when fitted)
Option

Treatment Sends and Returns

Reference has already been made (Sect. 1) to these four outputs and four inputs, and the extent to which use is made of them will depend on the other equipment available in the studio as a whole. Assuming that the eight plugs and sockets on the rear panel are connected to the main studio patch, any external device can be brought out on the S.100 patch. An example would be reverberation plates, which might be preferable in some instances to the inbuilt springs: extra filters or other treatments, or a complete external synthesiser such as a VCS3, some of whose facilities might be needed in a particular patch.

There is no special connection between Send 1 and Return 1, etc., and the 4 outgoing and 4 incoming ways can be used separately as additional inputs and outputs, assuming that signal levels are suitable for reception or onward transimission with amplification. Thus the Treatment Returns could be used as line level tape recorder inputs for signals already recorded which need processing in the S.100. Similarly the Treatment Sends can be used as outputs. On occasions you may send out from an output amplifier, but receive back the processed signal as a Treatment Return, or vice versa. Often the reverse is required, and a signal is sent in to the S.100 for perhaps one treatment, such as the octave filter bank, and returned out. If the device is not being used in an internal patch, there need be no disturbance to other operations in the studio.

In effect the addition of this facility means that the S.100 has 12, not 8, outputs and inputs.

Oscilloscope

We have already given in our examples many applications for the fitted double beam oscilloscope. This instrument is not made by us, but selected from the catalogues of proven manufacturers as being the most suitable. In nearly all cases the oscilloscope fitted to S.100 is by either Telequipment or Advance, and a separate manufacturer's handbook is supplied. If you are unfamiliar with the normal adjustments for an oscilloscope you should spend some time studying this handbook (which we suggest you punch and bind into this book) to familiarise yourself with the brightness, focus, shift, amplifier, synch and other controls.

The 'scope' inputs appear at four columns, S56-58 and C59. This last duplicates S56, giving an input to Y_1 on both Boards. Y_2 and X appear only on the Signal Board, but if X is driven from an oscillator for special time base purposes (as in Filt./Osc.1 - Sect.2) this can be done by cross-patching through an output amplifier in the usual way. Because of the very high input impedance of the scope, patching to it will have no appreciable effect on any functional operation being monitored, and the illuminated graticule can be used to measure voltage very accurately, the scale being determined by the gain setting of the appropriate Y amplifier. When reading rapid transients, such as square waves, it is advisable to use as low resistance a pin as possible (red 2K7) or some integration will take place.

The oscilloscope runs fairly warm and in cases where the S.100 has to be installed in conditions of poor ventilation, or in a hot climate, it is a good idea not to run it continuously, but only when being used for measurement. In any case the ventilation slots above the scope should always be kept clear of obstruction.

Digital Voltmeter

In some later models of S.100 a Digital Voltmeter is fitted, and mounted above the output meters on the righthand panel of the studio. This reads up to 9.999 volts negative or positive (a range of nearly 20V), and detection of polarity is automatic with visual indication. Its input is hardwired to scope Y_1, so anything patched to Y_1 (S56, C59) will also show on the DVM. It is particularly useful in reading the exact level of sequencer, keyboard or RVG voltage steps, for those occasions when a composer has prescribed a precise series of voltages. It will also read the half-cycle levels of square waves when they are slow enough to enable the instrument to follow. Another useful function is to detect the precise centre point of a joystick, which should of course be OV. The DVM is extremely sensitive, however, and if you can set a manual control like the joystick to within 0.002V of OV you are doing well.

Frequency Meter

As with the oscilloscope, various types and models have been fitted to the S.100, and we are constantly testing units for reliability and accuracy. To check the operation of the instrument fitted to your studio, consult the maker's handbook.

In all cases the frequency is detected by a counting process - that is a comparison between a known period generated internally and the number of incoming cycles detected during that period - normally 1 sec. The least significant digit is unreliable by one digit, since the clock may open the gate just before or just after the arrival of a zero crossing, and similarly close it when the period ends. If really important to know whether a frequency is, say, 571 or 57 Hz (and it very rarely is that important), the meter can be set to count for 10 secs., which means a much longer wait for the reading, but one more significant figure - though again the last figure is unreliable by 10% - i.e. you will now know that the frequency is 571.5 or 571.6.

This type of frequency meter is extremely accurate, and only fails to give clear results when the input is very rich, in which case it commonly reads the second harmonic, giving double the true reading. Some models have low pass filters to help in detecting the fundamental in such cases, but when checking an oscillator always try to send the f-meter a simple waveform. If the frequency is changing, either in staircase or glissando fashion, the meter will still read the number of zero crossings in one second, but represent the average frequency during that second.

For very low frequencies there are arrangements to read the reciprocal of the frequency (period measurement, normally in mS/cycle), though in some cases it will still read Hz and you will have to convert (it is easy to make a table for this).

Counting and clock modes may also be included, but this depends on the model, and the maker's notes should be consulted.

Store Unused Meter

When fitted, this is an edge-reading meter mounted to the right of the DVM, above the output meters. For the use of this device, refer to Section 4.

Option

Already referred to in Section 1, this consists of 2 x 8-way sockets on the back panel and an unwired 5K potentiometer mounted to the right of the Slew Limiter controls. No designation is given to this facility, and it may never be used in some installations. But is seemed a good idea to include the possibility of adding some extra devices, wiring them in and controlling at least one of them, without the necessity of drilling new holes etc. If necessary of course the potentiometer could be changed for one of another value. Occasionally when a special extra unit is supplied (e.g. Inverter Box - Sect. 5.) one of the Option sockets is used to bring out a power supply. These sockets have also been used for telephones, remote tape transport controls, talk-back circuits etc.

NOTES:

SECTION 4

S E C T I O N 4

Sequencer Operation

The sequencer operates in a similar manner to a six-track tape record-replay system, but with these differences:-

1) The sequencer will record steady-state D.C. levels. But not rapidly changing (i.e. audio) signals.
2) Record/replay may occur in either direction over a very wide speed range.
3) The recording is made of discreet events, rather than a continuous stream, although at high clocking rates the difference is not marked.
4) The recording medium is digital random access memory, and not magnetic tape.

The sequencer is designed to record any control voltage between ØV and 7V D.C. and store the amplitude of this voltage as it changes with time. The source of this voltage may be any device capable of generating it e.g. the keyboard, random voltage gen., dynamics, joystick, etc. Any two separate inputs may be recorded simultaneously on to any or all of the six layers, and all six outputs corresponding to layer content are available at all times, either when recording or replaying. In addition four digital, i.e. on/off signals may be recorded. The purpose of these being to record gate signals or control functions.

By suitable adjustment of controls, an individual event may be entered/changed or a continuously varying signal may be recorded.

The Controls

Master reset: This sets the event counter to location ØØØØ, and sets all outputs to zero.
Test Output: The display will read CAll. All outputs are set to maximum for calibration/test purposes. The sequencer will remain in this state until master reset is pressed.

Step Forward & Reverse: Assuming that there is memory available, the event counter will increment or decrement one step, and the data recorded at that location will be output. If any toggle switch is set to record, data will be simultaneously recorded at the selected layer, and the following event. If however there is no memory free, the display will read 'ofof' (Overflow), and only reset or a change of count direction will produce further result.

Reset Sequence: When pressed, will always reset the event counter to location ØØØØ. This may be done without stopping, so that a sequence may be caused to repeat indefinitely.

Stop: This command stops a running sequence, holding the event count at its present state. Count may resume by a run fwd/rev. command.

Run Forward/Reverse: When in these modes the sequencer runs in the selected direction at a speed determined by the clock input, unless stop, or master reset are invoked, or the clock stopped switch is down.
NOTE: Clock pulses from either the internal sequencer clock generator or an external oxcillator must be patched to the sequencer clock input on the left hand patch board, or nothing will occur. If an external oscillator is used, its frequency should be kept below 5ØC/S. The sequencer clock speed control knob will of course not affect the sequencer speed if an external oscillator is used.
NOTE that when any record/safe switch is down, an event will be recorded upon the layer(s) selected upon every event count up or down, or when the step forward/Reverse buttons are pressed. Data may therefore be on any or all layers, and overwritten if necessary, step by step, or in real-time. In all cases the outputs will immediately follow the new inputs.
NOTE also that the command buttons run F/R, stop, reset sequence, are also given as inputs on the patch board. This allows remote start/stop, and by using K4, the spare digital control track, the sequencer may be programmed to loop back and repeat. A positive voltage greater than 1 volt will activate the function. Inputs must not remain active when

not required. If in doubt, remove pins. The best source of key voltage is the gate outputs from the keyboards, with the envelope control knob set to give maximum positive output.

A Typical Arrangement

Patch the following: Seq. clock out to seq. clock in.
Pitch voltage from keyboard to layer A.
Keyboard gate pulse to K1.
Seq. layer A output to Osc. 1 control voltage I/P.
Osc. 1 Signal to Env. 1.
K1 output to env. 1 gate.
Ensure both keyboard and K1 give positive outputs.
Audibly monitor the output of env. 1 signal, and set it to gated.
Set sequencer clock speed to 8 on the major scale.
Put all rec/safe toggles to up (safe), except A+K1, which should be down.
Set pitch spread and layer A voltage controls to 9 on the major scale, and press master reset, followed by run-forward. The event counter should now increment fairly rapidly, and playing the keyboard should give some audible result. Almost certainly tuning will not be correct at this stage, and there may be some fluctuation in pitch when a note is held down. First adjust the pitch spread control for minimum fluctuation (check right over the keyboard range), then set the layer output control to give the correct musical intervals. (While doing this you will run out of memory, so reset sequence and continue).

This setting up is tiresome but unavoidable, since the sequencer must be designed to record continuous analogue changes, and not just discreet pitch intervals.

Having things working well, record a sequence from the keyboard, then press stop. Put up A+K1 toggle to protect your recording, press reset, and then run forward to hear it. Try reverse direction, stepping forward and back, and over-writing your recording to get familiar with the controls, then try adding other layers using not just the keyboard, but other sources to control filters, envelopes or whatever. In order to achieve a start from first key down, patch the run command from the keyboard gate pulse.

SECTION 5

SECTION 5

ANALOGUE SIGNAL-TO-CONTROL DEVICES, & USEFUL ADDITIONS TO STANDARD ESTABLISHMENT

Signal-to-Control Devices
  Envelope Followers
  Pitch-to-Voltage Converter

In the vast majority of cases 'voltage control' implies control of an audio device by a slowly changing or DC voltage. But there are occasions when the opposite is required - for an audio signal to result in a DC varying in proportion to one or more of its parameters. This particularly applies in the case of live performance instrumental inputs, because in the case of the internal devices the parametric information is already known - e.g. if we follow an internally generated audio envelope the result will certainly be the trapezium voltage, but we already have it in the form of a Trapezoid output, so it is usually pointless to generate it again. But if we play a flute (say) into a microphone and thence into the S.100 it may be of great value to be able to produce an envelope which is analogous to that of the flute whatever the player chooses to do. For this purpose two identical Envelope Followers are included, and for detecting and converting pitch to a voltage proportional to its frequency (about which the above remarks about internal devices also apply) there is a Pitch-to-Voltage Converter. Because these are Signal-to-Control devices, their inputs will be found on the Signal Board and outputs on the Control Board.

Envelope Followers 1 & 2

The Envelope Followers are simplicity itself to use, and each has one control on the left of the righthand vertical panel. If required to follow an external input, one of the input amplifier outputs is patched to an EF input, and the resulting control voltage used to influence any device on the Control Board. For example a live instrument might be played in and the output from the input amplifier taken through a filter network as well as the EF, and thence out to a loudspeaker. The EF control output (from which the original audio content is removed, the output being proportional to the mean level of audio) might be taken to a filter so that the timbre of the output is influenced by the dynamics of the performance. The Envelope Follower gives the composer a whole new range of live performance possibilities.

If you have no external input available, experiment by generating an envelope internally in the ordinary way (Osc. to Envelope Shaper). Take the Shaper output both to an output amplifier and to an EF input, and the output of this to control either the frequency of another oscillator or the gain of an output amplifier (but not the one carrying the first signal). You should now have two analogous signals, and the EF control knob will alter the level and polarity of the control output, so that inverted signals are possible. As with other bi-polar controls, the central position has no effect, but from 5 - 10 produces an increasing range in the same polarity as the input, up to approx. 1 volt/6dB of input change. From 5 - 0 gives a similar inverted range.

Experiment with the ability of the EF to follow different shapes and speeds of envelope. There is a limitation, because it is necessary in the design to arrive at a compromise between the necessity of filtering out audio ripple and the desirability of maintaining a fast response. A second order low pass filter is included in the output, and this limits response to a maximum of about 50Hz, enabling all but extremely rapid envelope fronts to be followed.

<u>Pitch-to-Voltage Converter</u>

There are considerable problems in designing a device which automatically detects and responds to the frequency of an incoming signal. In the first place a musical sound, particularly an instrumental one, is never a single frequency and may have a considerable aperiodic content as well as its own harmonic series. Or of course it may be a chord or note cluster. So which frequency will the converter choose as the 'right' one? The answer is to equip the PVC with an "adaptive filter" which seeks the lowest note. However it is best to choose relatively simple waveforms as input for reliable results. Rich or noisy sounds may produce unpredictable effects. In fact the circuit will find the fundamental and follow it providing it contains at least 10% of the total energy of the signal. If this condition is not met the device will either 'hunt' in an uncertain manner, or choose the strongest part of the spectrum. It is also necessary (see below) for the signal to be above a certain amplitude for detection to take place.

The other main problem is to detect fast enough, because the converter obvsiously cannot find the frequency until it has a sample to read. Most frequency measuring devices (such as the frequency meter in S.100) work by counting the number of zero crossings in unit time. But this process would take far too long to allow meaningful following of a changing frequency, so the PVC measures the <u>period</u> of a half cycle from one zero crossing to the next. It begins "guessing" as soon as the first half cycle has gone by, and improves this "guess" rapidly over the first few cycles. All the same, the ability of the PVC to follow rapid frequency changes is necessarily limited, and an instrumental or any other input should be as free as possible from vibrato and other f.m. phenomena.
Understanding the design problems should help you to select suitable inputs for it.

Like the Envelope Followers, the PVC will commonly be used with instrumental (or vocal) inputs - in fact often both devices are used together on the same input. Its single control knob is above those of the Envelope Followers. To test internally, arrange for an audio oscillator to have a slow to medium speed f.m. (say 0.25Hz) by patching a slow oscillator triangle to control it. Patch the audio oscillator output both to a channel and to the PVC input. Take PVC output to control another oscillator frequency, and monitor this also. The bi-polar control will give (as expected) no change to the second oscillator if set at about 5. At about 7 the follow will match the input octave for octave, and at 10 it will give approximately two octaves for one - i.e. it doubles the control range. Similar inverted positions are at about 3 and 0.

Experiment with more rapid transients (change triangle control to ramp progressively) and listen to the effect of the slew as the rate of change increases. At the sawtooth position the PVC cannot immediately adapt to the sudden change, and will slew towards the new frequency. A few milliseconds later it will "catch up" and continue following the ramp until the next sudden change. This slew can be used deliberately to change the control shape.

For reliable detection an adequate amplitude is necessary, and there is a critical level you can find by experiment below which the PVC cannot read a signal. At the threshold of this level it will dither. But because many instrumental signals, particularly percussive ones from struck or plucked instruments, die away at the end of a note, the PVC is equipped with a track and hold circuit which prevents spurious outputs and dither when such a signal is dying away - i.e. when it "loses" the input it continues delivering the last voltage until it receives a new signal above the detection threshold.

N.B. EMS manufacture the PVC as a separate module which also contains an envelope follower, a trigger detector and generator, and an internal oscillator, also a mixer which determines the relative levels of direct and converted signal in the output. This unit is extremely useful in live performance situations where it is inconvenient to move the S.100. A vast range of instrumental transformations can be obtained with a PVC and small synthesizers. It contains its own mains power unit.

ITEMS ADDITIONAL TO THOSE SUPPLIED AS STANDARD

In a machine as complex as S.100 there is always room for small additions and modifications, because each user will find slightly different uses for the machine, and different problems to solve. EMS are always prepared to consider a customer's proposal for a special unit or modification, and in any case we are always interested to hear of a new idea a S.100 owner may have. But many small additions can be made easily by the studio where the instrument is installed, and we give below a few examples of useful extensions to the standard facilities provided.

1. Jumper Leads

To increase the number of Board-to-Board patching possibilities, it is a good idea to make up some leads with a pin at each end. These can be taken directly from Board to Board. They are best made from springy wire which will self-coil to the right length. A resistor 22K-1M must be included. Only single core wire need be used, since Ground is hardwired internally. The connection is made to the tip of one pin and the shaft of the other, the other terminal being left open. Test for signal direction, and mark pins with upward or downward arrows to indicate output or input. Outputs are from any hole in the rows of either patchboard and inputs are any holes in any column of either patchboards.

2. Extra Measuring Instruments

As we have noted, later versions of S.100 are equipped with a Digital Voltmeter and a Store Unused Meter. These items can, however, be fitted to earlier models, and it is simple to patch in any meter with a suitable lead connected to pins. Always use as high resistance a meter as possible, and a DVM is ideal for control voltage readings, etc. When fitted internally, it is wired to Y_1 columns, but you need not necessarily do this. Extra oscilloscopes can also be patched in by the same method.

The Store Unused meter is more complicated to fit, and it is best to send for instructions if this modification is contemplated.

3. High Resistance Pins

Occasionally it is useful to be able to deliver different control input levels from the same control source, and this is most easily done by using different pin resistances. Because many control inputs only require a small voltage range, there is a need in such cases for pin resistances higher than 100K. For this purpose it is useful to have two or three pins at about 68K (green), and perhaps one at 1MΩ and one at 250K.

Either use open pins, without a top, or fit a resistor of the same size as the standard pins and mark the pin accordingly. Alternatively you can use an open pin to which you have soldered two leads and clips, so that any R value can be temporarily clipped in. You can experiment also with diode or capacitor pins on audio signals, which will modify waveforms by rectification or differentiation.

For all normal patches use the 100K (white pins supplied. 2K7 (red) pins are used only to connect the oscilloscope. Never use a shorting pin. Damage may otherwise occur.

4. Key 4 Remote Relays, etc.

Key 4 can be arranged to operate sequential or on/off switching, and various kinds of electronic and mechanical relay can be used, depending on the service required. If in doubt about a particular application ask the advice of our research and development department. If you are using Key 4 to, say, start and stop a tape recorder or signal a cue, you cannot also use it for Reset, of course, but if you are using only two layers Key 3 can be used for either purpose.

S.100 Dope Sheet No.________ of ________ Project No.________ Name________ Work________ Date________

LEFT BOARD SIGNAL

Columns: REVERB 1 2 | RING MODS 1 2 3 (A B A B A B) | ENVEL SHAPERS SIG 1 2 3, TRIG 1 2 3 | FILTER/OSC 1 2 3 4 5 6 7 8 | OFB | OSC. SYNCH. INS 1 2 3 4 5 6 7 8 9 10 11 12 | OUTPUT CHS 1 2 3 4 5 6 7 8 | EXT SEND 1 2 3 4 | ENV FOLL 1 2 | PVC | SEQ CLK, RES, SLEW, STOP, REF, SET, T | SCP Y1 Y2 X | GROUND

Column numbers 1–60

Rows:
INPUT AMPS 1–8 (61–68)
OUTPUT CHANNS 1–8 (69–76)
EXT. RETS 1–4 (77–80)
KEY 4 (81)
CLOCK (82)
NOISE 1, 2 (83, 84)
OSC1 (85, 86)
OSC2 (87, 88)
OSC3 (89, 90)

PIN CHECK : ● – ○ – ⊗ – Ⓢ – = TOT. PINS | PIN CHANGE NOTES

OSC4 (91, 92)
OSC5 (93, 94)
OSC6 (95, 96)
OSC7 (97, 98)
OSC8 (99, 100)
OSC9 (101, 102)
OFB (103)
LP FILTS 1–4 (104–107)
HP FILTS 5–8 (108–111)
ENV SHAPS 1–3 (112–114)
RING MODS 1–3 (115–117)
REVERB 1, 2 (118, 119)
GROUND (120)

PIN CODE: ● = SHORTING, ○ = 2K7, ⊗ = 10K, Ⓢ = SPECIAL, ●→● = BOARD-TO-BOARD JUMPER

RIGHT BOARD CONTROL

Columns: REV. MIX | ENV1 K D A O D | ENV 2 K D A O D | ENV3 K D A O D | FILT. CONT LP 1 2 3 4, HP 5 6 7 8 | OSC. CONTROL 1–12 | OUTPUT CHANS VOLT IN 1 2 3 4, LEVEL CONTROLS 1 2 3 4 5 6 7 8 | SLEW LIMTRS V IN 1 2 3, SLEW CONT 1 2 3 | SEQ V IN A B C D E F, KEY S | SCOPE Y1 | GROUND

Column numbers 1–60

Rows:
INPUT AMPS 1–8 (61–68)
OUTPUT CHANNS 1–8 (69–76)
OSC10 (77, 78)
OSC11 (79, 80)
OSC12 (81, 82)
RANDOM GEN K, V1, V2 (83–85)
ENV.FOLLS 1, 2 (86, 87)
SLEW LIMITERS 1–3 (88–90)

PIN CHECK : ● – ○ – ⊗ – Ⓢ – = TOT. PINS | PIN CHANGE NOTES

ENV1 (TRAP) I, II (91, 92)
ENV2 (TRAP) I, II (93, 94)
ENV3 (TRAP) I, II (95, 96)
SEQ.L1 VA, VB, K1 (97–99)
SEQ.L2 VC, VD, K2 (100–102)
SEQ.L3 VE, VF, K3 (103–105)
KEY 4 (106)
CLK.RATE KN (107)
UPPER K/BOARD P, V, E (108–110)
LOWER K/BOARD P, V, E (111–113)
LH J/STICK ↕ ↔ (114, 115)
RH J/STICK ↕ ↔ (116, 117)
PVC (118)
(119)
GROUND (120)

CONTROLS, LEFT TO RIGHT

LP FILTERS 1 2 3 4 | HP FILTERS 5 6 7 8

ENVELOPE SHAPERS 1, 2, 3

RM1 RM2 RM3 REVERB 1 REVERB 2

OCTAVE FILTER BANK

INPUT AMPLIFIERS 1 2 3 4 5 6 7 8

EXT. SENDS 1 2 3 4 | EXT. RETURNS 1 2 3 4

OSCS1 6: 1, 2, 3, 4, 5, 6 | OSCS7 12: 7, 8, 9, 10, 11, 12

NOISE I | NOISE 2 | RANDOM GENERATOR

DIG V/METER * + –

STORE UNUSED * 0 100 200 300

SEQ.EV.TIME

K/BOARD SEQUENCER: PVC, UP, LOW, LI, L2, L3, EFI, key4, EF2, new K, new K, new P, new P, SL I, SL2, SL3, OPT

J/STICK L | SEQ'R CONTROLS edit switches cl/rst | J/STICK R

ev.dist | clock

E R F R S

Sequencer Notes

METERS & OUTPUT CHANNELS

Input/output details

Peripherals, scope and f/meter

Operational Notes