WARNING

The Synthi AKS is an extremely delicate instrument. It has an enormous amount of circuitry in it for its size, and it is essential that, despite its portability it is not thrown around, as it may receive damage to some of the very delicate circuits. The machine is naturally guaranteed, but it is important that, in order to comply with this guarantee, that it is not tampered with in any way. Some of the screws are auto-destructive, so that if they are removed, they cannot be replaced. Therefore, great care must be taken in dis-assembling the Synthi.

THE
SYNTHI
EDUCATIONAL
HANDBOOK

by
Peter Grogono

APRIL 1972

ELECTRONIC MUSIC STUDIOS OF AMERICAN INC
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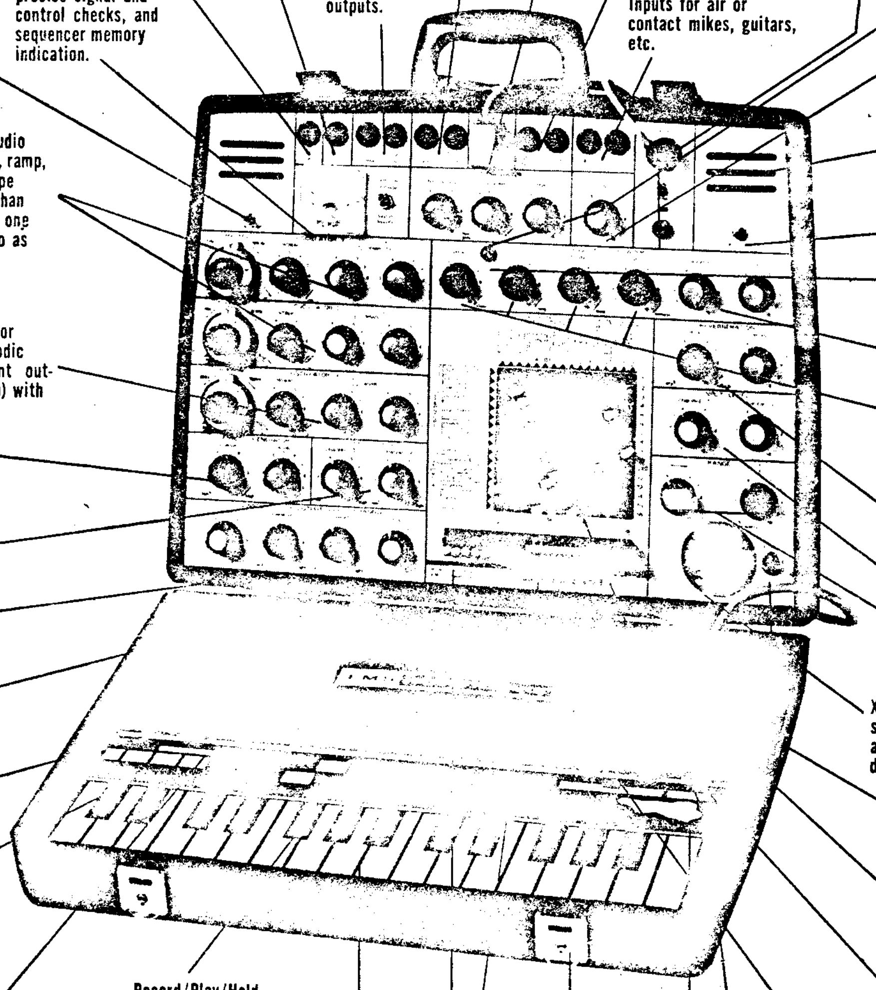


"Scope" output for waveform monitoring.

Stereo headphones jack. High level outputs.

Facilities socket for signal and power connection to keyboard or EMS modules (e.g. pitch to voltage converter, random voltage generator).

AC power socket with voltage range switch and indicator neon.

DC outputs for voltage controlling external equipment.

Line level inputs for signals or control voltages — e.g. tape recorders, foot pedal, etc.

AC/DC meter for precise signal and control checks, and sequencer memory indication.

Line level signal outputs.

Inputs for air or contact mikes, guitars, etc.

Left-hand monitor speaker with muting switch.

Two voltage controlled audio oscillators with rectangle, ramp, and sine outputs and shape controls, covering more than the entire audio range in one sweep from over 15KHz to as low as 1Hz.

Voltage controlled slow oscillator (0·05-500Hz) for voltage control and periodic effects. Two independent outputs (rectangle and ramp) with variable shape control.

White noise generator with colour and amplitude control.

Hi-Lo filters for wide colour control on each output.

Stereo outputs with panning controls.

Half hinges on lid to enable easy removal.

Entire keyboard lifts out for easy playing.

Automatic half tone through one octave transposition touch pads.

Clock rate control for sequencer (indicated on meter).

Record/Play/Hold select pads.

2½ octave plastic-coated tuneable touch keyboard with dynamic control. Total versatility with no moving parts.

Stowage for patch pins.

Prestopatch for instant patching of a pre-programmed sound (can be made to order).

Lockable safety catches.

Random voltage key.

256 hole pin panel. For patching any combination of signals and voltages. With pins.

Independent timing controls for real time and sequenced voltage.

Hi-Q Multifunction Filter/ Oscillator with variable response and voltage controlled frequency, giving sharp resonant filtering to steep cut low pass or pure sine output.

Transformerless high rejection IC ring modulator. For very dramatic tone transformation or octave splitting.

Right-hand monitor speaker.

Three position switch — speaker on/speaker mute/ trigger from external instrument.

Envelope shaper recycle indicator lamp.

Trapezoid output. DC analogue of envelope shaper for controlling other circuits.

Envelope shaper for automatic or manual time control of signals (attack, on, decay and off times all separately adjustable).

Reverberation with voltage controlled direct/echo ratio.

Input level controls.

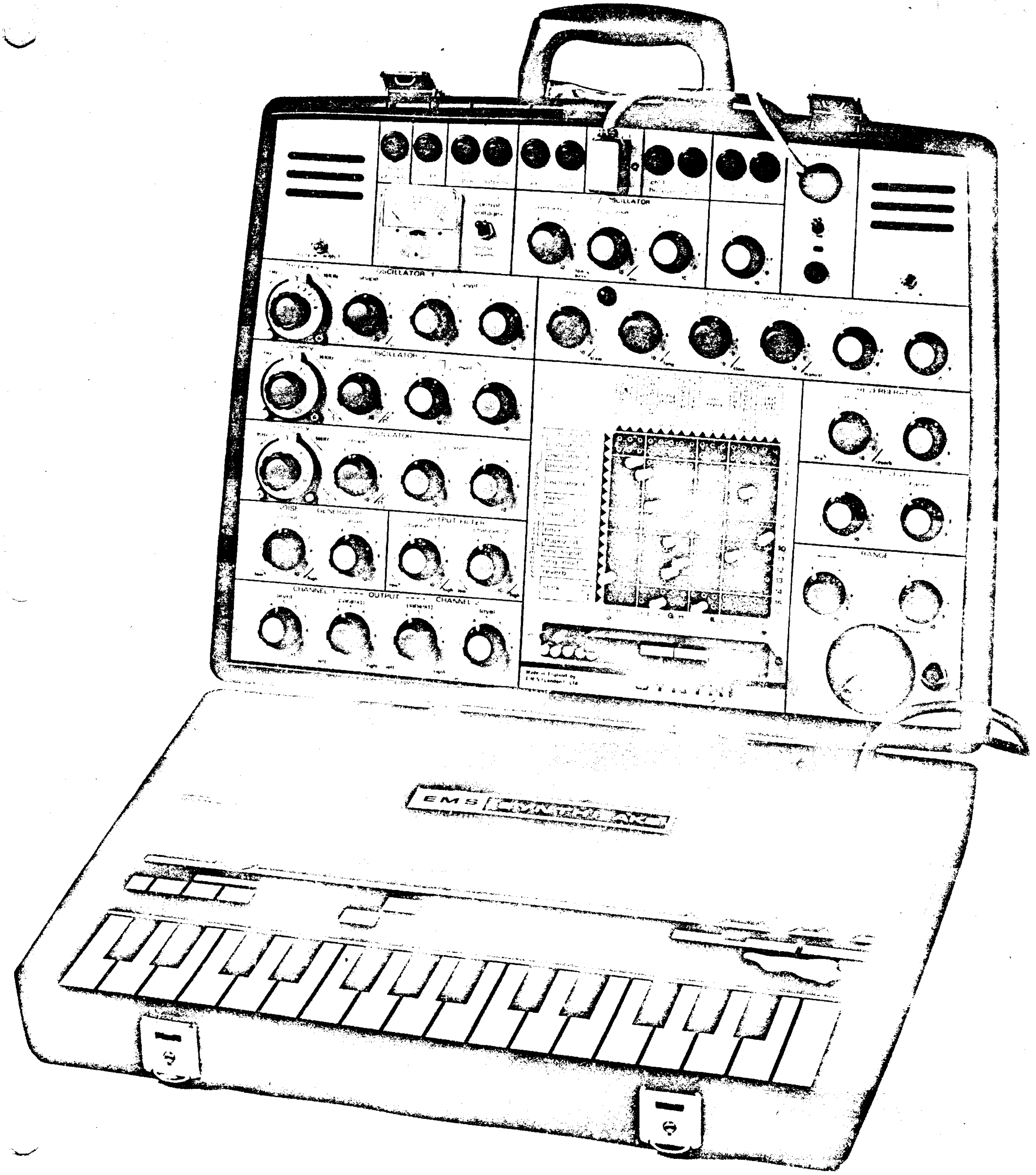
Joystick voltage range controls and sequencer on/off switch.

X-Y Joystick for simultaneous control of any two or more devices.

Attack button for precise manual initiation of envelope cycle. Conveniently placed for one hand operation with joystick.

Digital sequencer and clock logic circuitry. Up to 256 events controlled and stored by a complex of integrated circuits.

Envelope shaper trigger mode selector for real time or sequencer.


OSCILLATOR 1
OSCILLATOR 2
OSCILLATOR
NOISE GENERATOR
OUTPUT FILTER
CHANNEL 1 ---- OUTPUT ---- CHANNEL 2
REVERBERATION
RANGE
EMS SYNTHI AKS
Made in England by
EMS (London) Ltd

THE SYNTHI EDUCATIONAL HANDBOOK

Wee also have Sound-houses, where wee practise and demonstrate all Sounds, and their Generation. Wee have harmonies which you have not, of Quarter Sounds, and lesser Slides of Sounds. Diverse Instruments of Musick likewise to you unknowne, some sweeter than any you have; Together with Bells and Rings that are dainty and sweet. Wee represent Small Sounds as well as Great and Deepe; Likeweise Great Sounds, Extenuate and Sharpe; Wee make diverse Tremblings and Warblings of Sounds, which in their Originalle are Entire. Wee represent and imitate all Articulate Sounds and Letters and the Voices and Notes of Beasts and Birds. Wee have certain Helps, which sett to the Eare doe further the Hearing greatly. Wee also have Strange and Artificial Echos's, Reflecting the Voice many times, and as it were Tossing it; And some that give back the Voice lowder than it cam, some Shriller, some Deeper; Yea some rendering the Voice, Differing in the letters or Articulate Sound, from that they receyve, Wee have also means to convey Sounds in Trunks and Pipes, in strange Lines, and Distances...

Roger Bacon "The New Atlantis" 1624

SYNTHI EDUCATIONAL HANDBOOK

CONTENTS

List of Experiments

List of Diagrams

INTRODUCTION

Primarily, this book is intended to show you how to use your SYNTHI AKS. Although it is not a comprehensive text-book, it may also tell you something about the way in which sounds are made or altered.

The SYNTHI AKS is a *Synthesizer*, which means that it is used to build up or *synthesize*, sounds from basic tones. It differs from other musical instruments in that it has no characteristic sound of its own, and in order to use it you have to know something about the nature of sounds so you can build your own creatively. For this reason, we do not apologise for writing a book which is educational in tone. We hope that this book will be used in schools which have a SYNTHI AKS, and we also hope it will be used by others, in every field, to familiarise themselves with what a sound is before they become confused by a multitude of electronic devices.

In the first chapter, USING YOUR SYNTHI, we explain how the SYNTHI is set up and what the controls do. Unless you have experience of electronic equipment, you will probably not understand much of this chapter, but you will find later on when you have learnt a bit more about the SYNTHI, that it is a useful chapter for reference.

In the other chapters, there are sections of explanation and experiments. (The experiments are written in *italics* for clarity). The experiments are not intended to be complete: rather they are starting points for the much more detailed experiments which we expect you to do. We want you to think of the SYNTHI as a kind of constructional set, with which you can build your own sounds. The "patches" we provide are not complex, because we want to illustrate how the SYNTHI works, but when you have understood an idea or technique, you should try to incorporate it into your own patches.

Many of the experiments in this book are more interesting if you can connect your SYNTHI to an Oscilloscope. An oscilloscope is an electronic instrument which shows the voltage in a circuit visually, on a small screen rather like a television screen. We do not mention Oscilloscopes much because most SYNTHI owners do not possess one; if you do have access to an Oscilloscope you will find it very helpful in understanding how the SYNTHI works.

CHAPTER I: USING YOUR SYNTHI

This chapter describes the use of the SYNTHI, and also describes the devices it contains. It should be read through first of all (it may seem rather abstract initially) and then should be used for reference later, when you are doing the experiments for the SYNTHI. Before doing any of the experiments, make sure you have read the next section, Connecting the SYNTHI.

Unlike most of the electronic devices we are accustomed to, which perform one function only (radio and television receivers for example), or a few functions (a radiogram can play records and receive radio programmes), the SYNTHI has very many functions. The SYNTHI contains a number of separate circuits which are connected by the user; the knobs on the front control the circuits and the patchboard (the black square in the lower centre) connects them. If the SYNTHI looks complicated to you, remember that it uses only simple devices, and that complex sounds arise from the use of several devices together.

Connecting the SYNTHI

Power Supply

Before it can be used at all, the SYNTHI must be connected to the mains. European models are adjusted for 200-250VAC, and North American models for 100-130VAC. If for any reason the setting is incorrect it should be altered, using the slide-switch on the front panel before connecting the SYNTHI. The SYNTHI uses a transformer designed for 50-60Hz AC *and must never be connected to a DC supply.*

Connect the SYNTHI to the mains supply with the lead provided. The earthing (ground) connection (green/yellow) is not essential and may be left unconnected if two-pin sockets only are available. Turn the POWER switch on and check that the mains indicator light glows red.

Signal and Control Connection

Other electronic equipment is connected to the SYNTHI by the jacks at the top of the panel. The SYNTHI can be used as an entirely self-contained instrument, but its resources can be exploited more fully when other equipment is used with it. In particular, the built-in loudspeakers are very small and, although they are adequate for monitoring purposes, an external amplifier and loudspeaker system should be used for anything but personal experiments.

Jack plugs are always connected as follows:
Earth (ground) goes to the body or shaft (when screened cable is used, this will be the outer screening).
The "live" connection goes to the top of the plug.
The headphone jack must be used with a stereo plug, which has an additional ring on it for the second "live" lead.

It is best to solder all wires, but if you cannot solder obtain some jack plugs (Igranic type) with screw terminals.

The functions and specifications of the Jacks on the SYNTHI are explained here:

Stereo Phones:	Stereo (three wire) jack giving a two channel signal suitable for powering a pair of stereo headphones.
Level:	*10V p-p max into 50 ohms*
Scope:	A mono Output (two wire) connected to the Meter Circuit. It is suitable for an oscilloscope display, and is sometimes connected to the SYNTHI inputs as part of a patch.
Level:	*Device dependent, 5V p-p max*

Signal Outputs: These outputs are suitable for a power-amplifier or stereo tape-recorder.

Level: *2V p-p max into 600 ohms*

Control Outputs: These outputs should be used when the SYNTHI is required to provide a DC control voltage to another synthesizer or voltage controlled circuit. They are connected to the Output Channel rows on the Patchboard.

Level: *±5VDC max into 10Kohms*

Keyboard: This socket is designed to connect the SYNTHI to a SYNTHI 'K' or 'KS' keyboard and other SYNTHI modules and should not be used for any other purpose.

Signal or Control Input: Jacks connected to the Input Amplifiers of the SYNTHI. An external source with line-output, such as a pre-amplifier or tape-recorder, should be connected here.

Sensitivity: *2.5VAC p-p*
±2.5VDC into 50Kohms for voltage-control

Microphone Input: These inputs are suitable for most forms of low-level signal, as produced by air-microphones, contact microphones and guitars.

Sensitivity: *5mVAC into 600 ohms*

The Devices

The SYNTHI has a number of devices, each with its own controls on the front panel. The brief specifications below are intended for reference, and do not describe the uses of the device.

Control refers to the function of the device, if any, which can be voltage-controlled. *Matrix* gives the identification by digits and letters of the rows and columns on the Patchboard allocated to the device.

Oscillator 1: Audio frequency oscillator with sine and ramp waveforms. Manual controls for frequency, shape of sine output (mainly even harmonics may be added), level of sine output, level of ramp output. The outputs are mixed at the Patchboard.

Range: *1Hz - 10KHz on dial*
Output: *sine 2V p-p*
ramp 2V p-p
Control: *frequency 0.32 V/octave*
Matrix: *3, I*

Oscillator 2: Audio frequency oscillator with rectangular and triangular waveforms. Manual control for frequency, shape, levels. Outputs are mixed at the patchboard.

Range: *1Hz - 10KHz on dial*
Output: *square 2V p-p*
triangle 1-5V p-p (symmetrical position)
ramp 3V p-p (extreme position)
Control: *frequency 0.32 V/octave*
Matrix: *4, J*

Oscillator 3: Audio and subsonic oscillator with rectangular and triangular waveforms. Manual controls for frequency, shape and levels. This oscillator may be run at very low frequencies for control applications.

Range: *0.02Hz - 500Hz on dial*
square 4V p-p
triangle 3V p-p (symmetrical position)
ramp 6V p-p (extreme position)
Controls: *frequency 0.26 V/octave*
Matrix: *5, 6, K*

Noise Generator: White and coloured noise source. Manual control of level and colour.

Output: *3V p-p*

Filter/Oscillator: A multi-purpose filter, resonator and oscillator, with adjustable bandwidth (Q) and frequency. Manual control of frequency, mode and output level.

Range: *5Hz - 10KHz on dial*
Cut-off Rate: *18db/octave max*
Q: *20 max*
Control: *frequency 0.2 V/octave*
Matrix: *10, H, N*

Ring-Modulator: An advanced design with high input rejection. Manual control of output level.

Input Level: *1.5V p-p max*
Output: *6V p-p max*
Rejection: *-50db at 1.5V input*
Matrix: *13, E, F*

Envelope Shaper: A programmed variable-gain amplifier for envelope control. Manual controls for attack, on, decay and off times, signal output level and control output ("trapezoid") level. There is also a recycle ("attack") button by the Joystick.

Timing: *Attack 2ms - 1s*
On 0 - 2.5s
Decay 3ms - 15s
Off 10ms - 5s + "off" position to inhibit recycling.

Signal Output: *5V p-p max*

Control Output: *±3VDC max*

Control: *Decay time 0.4 V/octave*

Matrix: *11, 12, D, L*

Reverberation Unit: A double spring device with delays of 25ms and 30ms. Due to the small cabinet of the SYNTHI, acoustic feedback can occur when the internal loudspeakers are used and this should be avoided as far as possible. Manual controls for mix and level.

Max Reverb Time: *2s*

Output: *5V p-p max*

Control: *Reverberation time -2V for 0%, +2V for 100% mix*

Matrix: *14, G, M*

Joystick: Two manual control voltages are provided by moving the joystick up or down or from side to side. Manual range controls are provided.

Output: *2 x ±1.5VDC*

Matrix: *15, 16*

Meter:	Multi-purpose meter for monitoring, checking and cueing. Switch for selecting SIGNAL or CONTROL modes.
Input:	*±1VDC in Control Voltage mode* *4V p-p in Signal Level mode*
Input Amplifiers:	Signals and controls may be input to the SYNTHI at line or microphone level. Manual control of sensitivity.
Sensitivity:	*2 x 2.5VAC signal* *2 x ±2.5VDC control* *2 x 5mVAC microphone*
Matrix:	*8, 9*
Output Amplifiers:	Two output channels with manual control of level, filtering and panning. Internal monitoring speakers with muting switch.
Outputs:	*2 x 2V p-p into 600 ohms signal panned* *2 x 10V p-p into 50 ohms signal unpanned* *2 x ±5VDC into 10Kohms unpanned*
Control:	*level 15db/V average*
Matrix:	*1, 2, A, C, O, P*
Signal Trigger:	A signal sent to Channel 2 can be used to trigger the Envelope Shaper if the Channel 2 muting switch is set to the mid-position, "Trigger".
Keyboard:	30 Touch Contacts providing pitch and Dynamic Output. Sequencer with 256 event storage, and touch controls for Record, Play, instant Transposition and random note generation.
Matrix:	*8, 9, 16*

Patching

The devices in the SYNTHI are not connected internally. Before the SYNTHI will make any sound at all it must be "programmed" by putting pins into the Patchboard. Each pin connects the output of one device to the input of another. Suppose you want to connect the output of the Noise Generator to the input of the Filter, for example. Look down the labels at the left hand side of the Patchboard and you will see that row 7 is labelled "noise". Now look along the top of the Patchboard to "filter" at the top of the 8th column: putting a pin at the intersection of the 7th row and 8th column will connect the two devices as required. It helps to imagine the signal travelling from left to right along the 7th row until it meets the pin, and then travelling up to the top of the Patchboard along the 8th column.

Later in this book, patches will be described by a letter A - P and a number 1 - 16. These refer to the column and row respectively and for convenience they are inscribed on the right and bottom sides of the Patchboard. The pin used in the example above (Noise to Filter) was H7.

The 30-way socket in the spare pin repository at the bottom of the panel is connected to each row and each column of the Patchboard. It is used for "Prestopatches" which set up any combination of pins immediately.

CHAPTER II: SIMPLE PROPERTIES OF SOUNDS

We are very accustomed to hearing sounds. Animals, birds, people, machinery and natural events make sounds. People in particular make complex sounds to communicate with each other, and go to great lengths to make instruments which produce the organised sounds called music. Anything that makes a sound we will call a *source* of sound.

What do sources of sound have in common? The answer is simply that they *move*. There are many different kinds of movement and they give rise to many different kinds of sound. Some movements are obvious - a rock falling, or waves on the beach. But many sounds are caused by tiny vibrations which are not visible to the unaided eye, such as the vibrations of a tiny bell when it is ringing.

The reason that we hear sounds is that the movement of the sound sources makes the surrounding air move, and it is the air movement that our ears detect. Inside each ear there is a small mechanism which converts vibrations of the air into vibrations of a nervous membrane and then into electrical vibrations which are interpreted by the brain.

During the last hundred years, men have developed their own artificial ways of turning sound vibrations into electrical vibrations and *vice versa*. It was by doing this that they were able to send sounds over long distances by using telephones and radios, and to store them in permanent form on phonograph records and recording tapes. A *microphone* is a device which turns sounds into electrical signals, and a *loudspeaker* turns electrical signals back into sounds.

Loudness

Loudness is probably the most obvious quality of a sound. The vibrations in the air cause changes of pressure, and it is these changes which move our eardrums. A loud sound (for instance, a rifle shot as heard by the marksman) produces more than a hundred million times as much pressure on the eardrum than a quiet sound which you can only just hear. It is astonishing that the ear can accommodate this enormous range of sound levels: imagine a weighing machine that could weigh fleas as accurately as elephants.

We could use a unit of pressure to measure loudness, but there are two good reasons for not doing so. One is the inconvenience of using numbers which vary over such a wide range (100,000,000:1), and the other is that it does not agree with our own idea of loudness. Most people, if asked, would not guess than an orchestra playing loudly produces a thousand times as much sound as the same orchestra playing softly, and a million times as much sound as a quiet conversation.

Using a logarithmic scale, which means that we measure *ratios* of sound power, we find that the numbers are much more reasonable. The ratio unit we use is named after Alexander Graham Bell: the Bel is a large unit, so in practice we use one-tenth of a Bel, which is called a decibel and is abbreviated db. Since the logarithm to the base 10 of 2 is 0.301, when we *double* the sound power we add 0.301 Bels, or approximately 3db. Similarly, taking away 3db (often called a *3db loss*) means *halving* the power. You can easily see that gains or losses of 20db or 30db represent large multiples of small fractions of the original signal. We use 0db as a reference point, sometimes giving it a definite value and sometimes merely using it as a comparison. For instance, we might say that *mezzo-forte* (*mf*) in music is a "normal" dynamic, and call it 0db. Then *mp* might be -3db and *f* +3db. In the Table, Fig.II-1, we have defined 0db to be the level of the quietest sound that can be heard, so that a sound at -3db (or any negative value) would be inaudible, but we could have started anywhere without affecting the accuracy of the scale.

Quietest sound	*0 db*
Gentle rustle of leaves	*10 db*
Whisper at 4 feet	*20 db*
Quiet bedroom	*35 db*
Conversation at 12 feet	*50 db*
Busy office with typewriters	*65 db*
Alarm clock at 3 feet	*80 db*
Heavy diesel lorry at 20 feet	*90 db*
Very noisy factory	*100 db*
Jet aircraft taking off at 75 feet	*140 db*
Moon rocket taking off at 1000 feet	*200 db*

Fig.II-1: Table of sound levels

Experiment 1

Turn the SYNTHI on and connect it (if possible) to power amplifiers. Connect Oscillator 1 to Output Channel 1 by putting a pin at A3. (If you don't understand this, re-read the section on "patching" in Chapter I.) Set the FREQUENCY control of Oscillator 1 to 7, the SHAPE control to 5, and the LEVEL controls to 8 (sine) and 0 (ramp). Set OUTPUT LEVEL 1 to 7 (or to a convenient listening level) and check that OUTPUT FILTER 1 is set to 5 (centrally).

By altering the LEVEL control, observe that the sound can be made inaudible or (if sufficient amplification is available) painfully loud. By altering the FREQUENCY control note that the level at which the sound becomes inaudible depends on the frequency.

As well as being loud or soft, sounds can be long or short. Some sounds are continuous - their loudness is the same all the time. Other sounds start and stop; they are called *transient* sounds, and the exact way in which they start and stop is very important in music, which is made up almost entirely of transient sounds.

Experiment 2

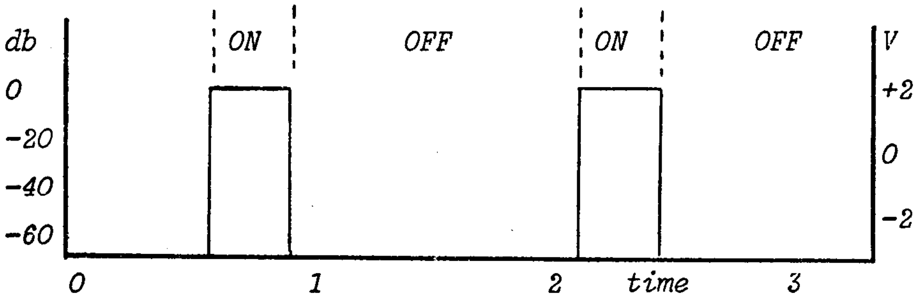
Connect Oscillator 1 through the Envelope Shaper to Output Channel 1. The input to the Envelope Shaper is column D and its output ("env signal") is row 12, so the pins required are at D3 and A12. Make sure that the Oscillator and Output Controls are set to give an audible signal, and turn the Envelope Shaper SIGNAL OUTPUT control to 8.

Set the timing controls of the Envelope Shaper (labelled ATTACK, ON, DECAY, OFF) according to the table, Fig. II-2, and listen to the different envelopes. Graphs of the signal level against time are shown in Fig. II-3. In case (a) the signal *reaches its maximum level instantaneously, stays there during the ON time, and falls instantaneously to zero where it remains during the OFF time. In (b), the signal dies away slowly (decays), and in (c) it rises slowly (attacks) as well. (c) has no ON time because the note starts decaying as soon as the attack is completed (ON time would be shown as a flat top on the wave form). In (d) the OFF time has been set to be infinite, but the Envelope Shaper can be reactivated by pressing the ATTACK button which starts a new cycle. The red light on the front panel lights up during the ATTACK and ON parts of the cycle. Experiment with the timing controls yourself, trying perhaps to imitate various instruments. Alter the frequency of the Oscillator and listen to the different aural effect of the same envelope at different pitches.*

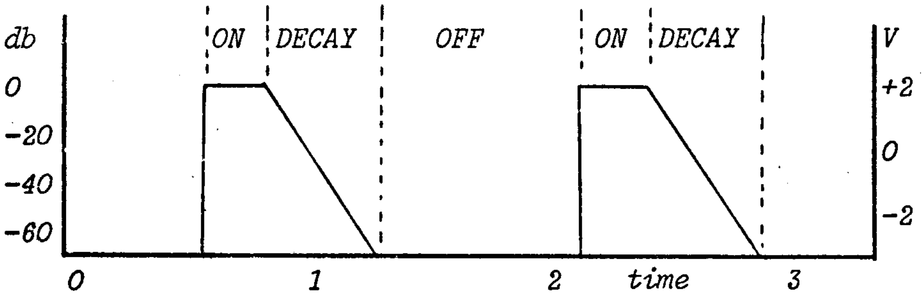
Add a pin at B11 and set the Meter switch to CONTROL VOLTAGE. The output at row 11 is not a signal but a control voltage (of which more later) - and in this case it is the voltage used to control the signal output level of the Envelope Shaper. The Meter should move in time with the Attack/Decay cycle - if it doesn't, turn up the TRAPEZOID LEVEL control. Fig. II-3 shows the way in which the loudness of the sound coming from the Envelope Shaper changes with time, and it also represents the voltage coming from the Trapezoid, which explains its name.

Setting:	*(a)*	*(b)*	*(c)*	*(d)*
ATTACK:	0	0	4	4
ON:	3	3	2	2
DECAY:	0	7	7	7
OFF:	5	0	0	10

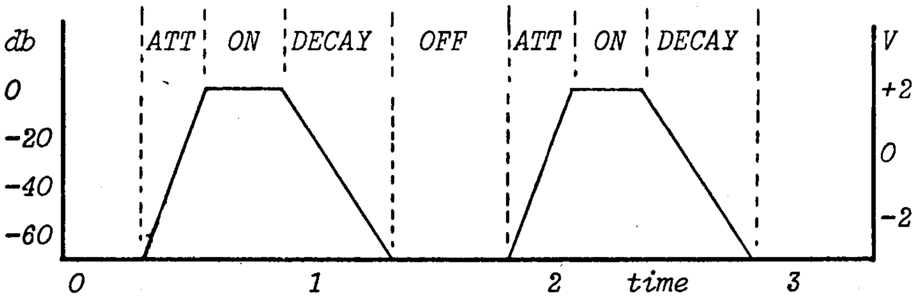
Fig. II-2: Envelope Shaper timing control settings for Experiment 2



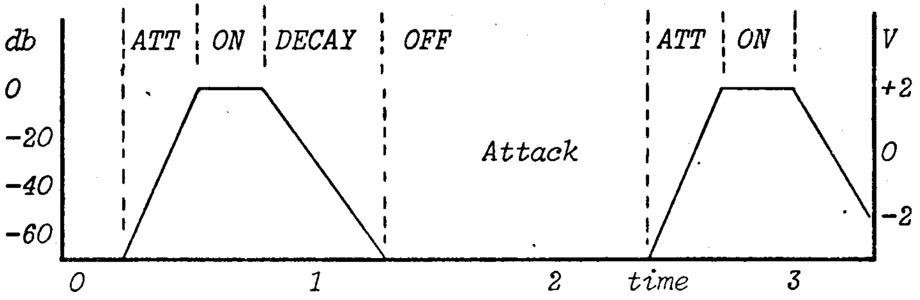
(a) Instantaneous Attack and Decay



(b) Instantaneous Attack, Slow Decay



(c) Fast Attack and Slow Decay



(d) Infinite OFF time, cycle restarted by Attack Button

Fig. II-3: The Envelope Shaper

The graphs show time horizontally and output signal level vertically. The left hand scale is db (0 db is maximum loudness) and the right hand scale is the trapezoid voltage. Fig. II-2 shows the control settings required to produce the envelopes shown.

Frequency

The second obvious quality of a sound is its *frequency*. This is both more complex and in many ways more important than loudness. More complex because most sounds have more than one frequency, and more important because we identify sounds more by their frequency structure than by their loudness. Frequency is the *rate of vibration* of the sound source, measured in (after Heinrich Hertz, the radio pioneer), which means *cycles per second,* and is usually abbreviated to *Hz.* You do not have to say *Hertz per second* any more than you say *knots per hour* when talking about a ship's speed. The "per second" and "per hour" are part of the definition.

We hear different frequencies rather as we hear different loudnesses, with a logarithmic law. When the frequency of a sound is doubled, we hear it an octave higher. 800Hz is an octave higher than 400Hz and two octaves higher than 200Hz. We can hear sounds over almost ten octaves, from 20Hz to 20000Hz, and the ear is most sensitive at 4000Hz. As we get older, our ability to hear very high sounds diminishes and only young people can hear sounds above 20000Hz.

Experiment 3

Use the same experimental set-up as in Experiment 1. Set the SHAPE control to 5.

Adjust the RAMP LEVEL control to zero, and the SINE LEVEL control for a fairly loud signal, and move the FREQUENCY control. At very low frequencies the sound will disappear altogether (or it may turn into a series of clicks which are heard as separate sounds when they are slower than about 20 per second) and at very high frequencies it becomes very thin and then disappears. (It may be impossible to make the frequency high enough using the dial alone. If this is so use the Joystick to extend the frequency range as follows: put a pin *at I16 and set the VERTICAL RANGE control to 10, then move the Joystick up and down to control the frequency.)*

Most sounds contain not one frequency but several, and we talk about frequency *components*.

Experiment 4

Use the same experiment set-up as Experiment 2.

With the SHAPE control at 5, the sound from Oscillator 1 is almost "pure". Turn the SHAPE control either way and note that the sound gets "buzzier" in quality. By moving the FREQUENCY control try to discover whether your range of hearing is altered by the change in tone. Move the pin to B2 and listen to the different qualities of "square" and "ramp" waveforms, adjusting the SHAPE control and both LEVEL controls of Oscillator 2.

The "buzzier" quality of the sound in this experiment is due to the presence of extra frequencies. The "pure" sound is a vibration at a certain frequency, f say, and the SHAPE control introduces other frequencies at $2f$, $3f$, $4f$, etc. Provided that the new frequencies are simple multiples (twice, three times, four times, etc.) of the original frequency, we hear only a single note with a characteristic tone. The initial frequency (f) is called the *fundamental* and the other frequencies are called *harmonics* or *overtones*. The "tone" of the note is technically called its *timbre*. In the experiment above, you should have found that you could hear the Oscillator at lower frequency settings when the SHAPE control was moved, since although the fundamental became inaudible (less than 20Hz) the overtones could still be heard.

We will now describe two ways in which a sound can be represented on a graph. Firstly, we can draw the sound as it appears on an oscilloscope screen. What we are doing here is

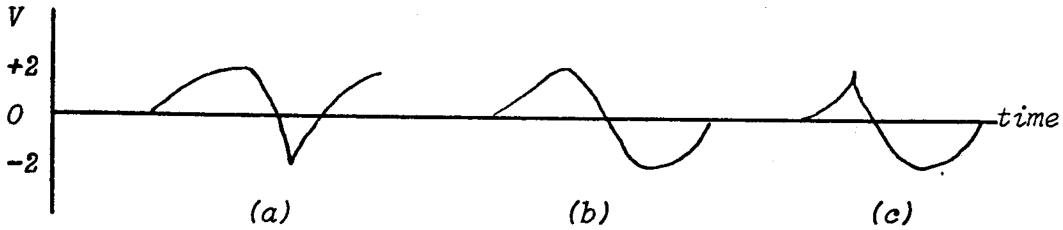


Fig. II-4: Effect of SHAPE Control on Sine Wave

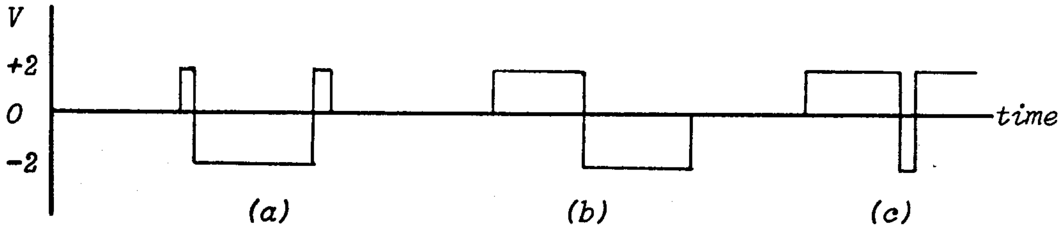


Fig. II-5: Effect of SHAPE Control on Square Wave

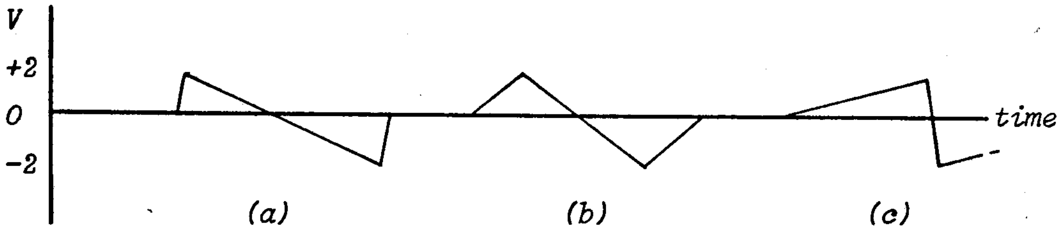
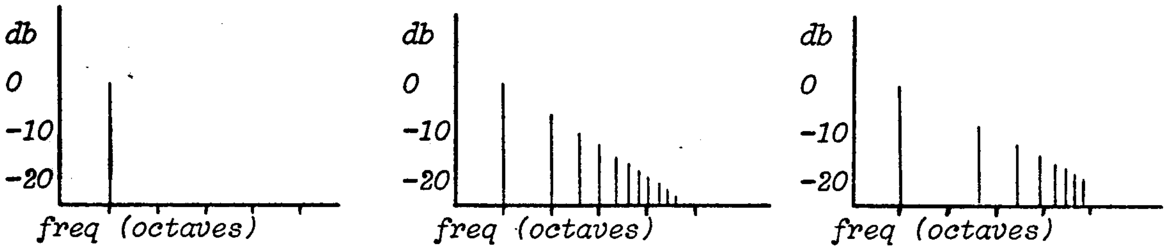


Fig. II-6: Effect of SHAPE Control on Triangular Wave

In Figs. II-4, II-5 and II-6, (a) corresponds to a SHAPE setting of 1, (b) to a SHAPE setting of 5, and (c) to a SHAPE setting of 9.



(a) sine wave

(b) ramp wave

(c) square wave

Fig. II-7: Spectra of three waveforms

plotting a variable which changes with time: it does not matter much whether we plot air-pressure, movement of the ear-drum, position of the diaphragm of a microphone or the voltage which the microphone produces, because they all look the same. The voltage is most convenient, because the oscilloscope is an instrument designed especially to plot a changing voltage. Figs. II-4 to II-6 show the waveforms that we have heard so far as they would appear on an oscilloscope screen. These drawings clarify the meaning of the little squiggles that you can see on the front panel of the SYNTHI by the Oscillator controls, and also explain the names "square" and "ramp" which we have given to the waveforms. By listening to the sounds and looking at the graphs you will learn to associate a sound with its waveform: note that the waveforms with the sharpest corners have the hardest sounds. If you have access to an oscilloscope, connect it to the SYNTHI output or SCOPE jack and experiment with the SHAPE controls while watching and listening.

The other way in which a sound is represented on a graph is slightly more difficult to understand, but it is often more useful. Horizontally we use a *frequency* scale and vertically we use an *amplitude* scale. Since this kind of graph has no time axis, it does not represent the way a sound changes, but only an instantaneous situation. For reasons which you will now understand, both scales in this graph are logarithmic: in fact, it is convenient to use octaves for the frequency scale and decibels for the amplitude scale. The resulting graph is called the *spectrum* of the sound, and you will see that the spectrum of a sound is analogous to the spectrum of light, since we see different frequencies of light as different colours.

Now we will look at the spectra of some of the sounds we have been listening to; see Fig. II-7. (a) shows the spectrum of a pure sound which has exactly one frequency component, represented in the graph as a single line at that frequency (we have chosen 160Hz). (b) shows the spectrum of a ramp waveform at the same fundamental frequency (160Hz) but now there are overtones at the frequencies of the harmonics (320Hz, 480Hz, etc). The harmonics are equally spaced in frequency (each being 160Hz higher than the last) but appear unequal on the logarithmic scale of the spectrum. They also sound unequal musically: the intervals between harmonics are an octave, a fifth, a fourth, a third and so on, and this justifies our use of a logarithmic scale. (c) shows the waveform of a square wave which has odd harmonics only (3x160=480Hz, 5x160=800Hz, 7x160=1120Hz, etc). This is a property of all symmetrical waveforms.

There is an instrument which displays the spectrum of a sound directly, rather as an oscilloscope displays the waveform, but it is complicated and expensive. However, the SYNTHI can be used to demonstrate the frequency structure of a sound acoustically. A *filter* is a device which allows some things to pass through it but blocks others. An oil filter is a wire mesh which allows oil to pass through it but stops dirt, and a food strainer allows water and dirt to pass through it but stops vegetables. In sound experiments, a filter is used to select or reject certain frequency components of the sound. Acoustic filters can be made, but nowadays it is more convenient to filter an electrical signal and listen to the result with a loudspeaker.

Experiment 5

Connect the Filter to Output Channel 2 with a pin at C10. The Filter should be set to give the highest possible selectivity without resonating as follows: turn the RESPONSE control to about 5 and the LEVEL control to 10. Turn the FREQUENCY control all the way from 1 to 10; if there is no sound at all, turn the RESPONSE up a bit and sweep the FREQUENCY again. Do this until the Filter resonates (makes a pure sound of its own: there is no input signal yet) over a small frequency range, and then turn the RESPONSE down a bit. The Filter is now in bandpass mode, *which is to say that it will only let through a narrow band of frequencies. The frequency at the centre of the band is controlled by the FREQUENCY control.*

Now put various sounds into the Filter and sweep the FREQUENCY control from 0 to 10 to find their frequency components. Use as low a setting for the source device as possible because the Filter is most selective when its input signal is small. A pin at H3 enables you to analyse a sine-wave (which should have no harmonics with the SHAPE control at 5) or ramp, and moving it to H4 enables you to analyse a square wave or ramp, or combination of the two. Although the audible results are not quite as clear as the graphs, the harmonic structure should be apparent. You can continue the experiment with external sources, using the Input Amplifiers and a pin at H8.

Much of the original research into the frequency components of sounds was carried out in the nineteenth century by Helmholtz. He did not have the advantages of modern electronic equipment, and he made acoustic filters which serve the same purpose as the SYNTHI Filter in the experiment above, using the actual sounds rather than their electrical equivalents. His filters, now called *Helmholtz resonators*, were designed for maximum selectivity, but almost any container with one or two holes in it functions as a filter as you can verify by speaking into a tin can or along a tube. Almost all musical instruments use resonators - in fact, without them, they would be almost inaudible. The body of a violin is a resonator, as is the gourd on an Indian sitar. The tympanum (kettledrum) is a resonator. Unlike an acoustic resonator, which works at one frequency only, the SYNTHI Filter's frequency can be adjusted over the whole audible range.

An important advantage of the sound spectrum graph over the waveform graph is its ability to show the characteristics of noise. "Noise" is a colloquial term used for any sound which is not deliberately organised for a purpose, as speech and music are. But it also has a technical sense which we will define here: *noise* is a sound composed of all frequencies. We also make a distinction between *white noise* which contains all frequencies equally, and *coloured noise* which contains more sound at some frequencies than others. As with "spectrum", the words are chosen by analogy with light: a mixture of light of all frequencies (colours) is white. Noise is caused in nature by many random events, and is different from pitched sounds which are produced by steady vibrations of strings, membranes etc. The sea, wind, and distant traffic are examples of noise sources. The waveform of noise, viewed on an oscilloscope, varies constantly, and is not at all like the steady pattern of a note. The most characteristic feature of noise is its spectrum since this shows the average energy levels at different frequencies.

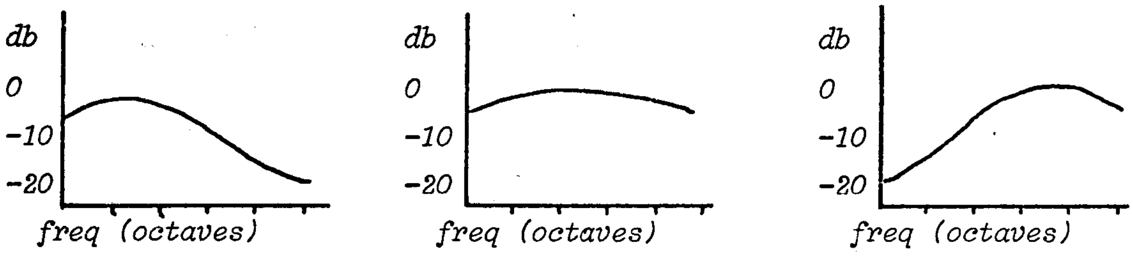
Experiment 6

Connect the Noise Generator of the SYNTHI to Output Channel 1 with a pin at A7.

Adjust the Noise LEVEL control so you can hear the noise. With the COLOUR control central, the noise is white (make sure you have OUTPUT FILTER Channel 1 set to 5); turning the COLOUR control to the left emphasises low frequencies, and turning it to the right emphasises high frequencies. See Fig. II-8.
Note: the Noise Generator requires 10-30 seconds to warm up after the SYNTHI has been switched on.

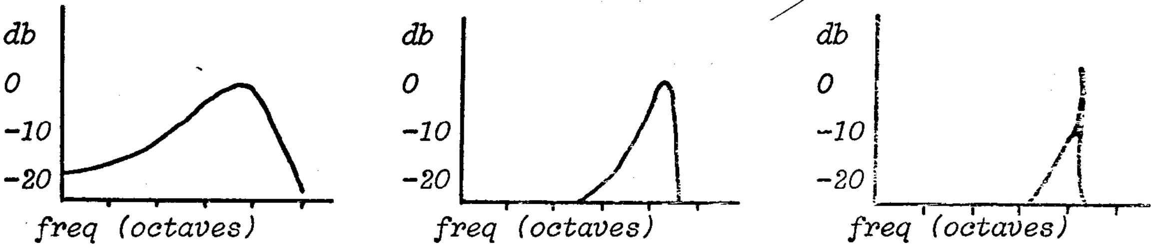
Experiment 7

Adjust the Filter to give a narrow passband as in Experiment 4. Connect the Noise Generator through the Filter to Output Channel 1 (H7, A10).



(a) COLOUR at 0 (b) COLOUR at 5 (c) COLOUR at 10

Fig. II-8: Effect of COLOUR Control on Noise Generator



(a) RESPONSE at 0 (b) RESPONSE at 6 (c) RESPONSE at 10

Fig. II-9: Effect of RESPONSE Control on Filter/Oscillator

A "band" of noise can now be selected by moving the FREQUENCY control of the Filter. The width of the band can be altered by moving the RESPONSE control, but remember that above a certain setting of RESPONSE, the Filter will resonate, generating an additional pure note in the middle of the band of noise. See Fig. II-9

CHAPTER III: TECHNIQUES OF ELECTRONIC MUSIC

In this Chapter we will look at the origins and history of Electronic Music, and the technical developments that lead to the SYNTHI, and then show how these principles are incorporated into the SYNTHI.

Electronic Music is almost as old as electronics; within a few years of the invention of the radio valve, Dr Theremin was experimenting with an instrument played by a performer who moved his hands around two aerials, and M. Martenot was building early versions of a keyboard instrument called the "Ondes Martenot". Both instruments are built in modified form and used today. The invention which had the most profound effect on electronic music was the tape-recorder. Although phonograph records had been in existence for a long time, and some of the pioneers of electronic music had made considerable use of them, a more flexible medium was needed. With tape, it became possible to edit, mix, superimpose and perform other important processes very much more easily than with discs.

The problem now became one of *control*: a well-equipped studio of the nineteen-fifties (and there were very few) might have an assortment of oscillators and filters, but very limited techniques for "performing" on them. A composition would be realised very laboriously by recording the required sounds, often one note at a time, and editing the resulting tapes until the composition was complete.

The next step was to make more devices to control the existing ones. The flexibility of a modern studio is determined less by the sound-producing devices themselves than by the controlling equipment. The SYNTHI uses the powerful technique of *voltage control,* and also an entirely new and sophisticated method of *sequencing,* or playing a number of notes in succession without intervention from the player. The most advanced modern studios use a digital computer, which may be regarded in this application as a very advanced sequencer.

Voltage Control

So far, with one exception, the signals which we have used in the SYNTHI have been analogous to sounds; that is, the signal could be taken to an Output Amplifier at any stage in the process and be heard. The exception was at the end of Experiment 2 when we watched the Meter move in time with the Attack/Decay cycle: with the pin at B11 we can read the voltage, but if we move it to A11, connecting Trapezoid to Output Channel 1, there is no sound. This is not because the signal is of a different kind, but because its frequency is too low to be heard. Although such signals are no use for making sounds, they may be very useful for controlling other devices, and they are called *control voltages* for this reason.

Experiment 8

Connect Oscillator 1 to the Filter (H3), the Filter to the Envelope Shaper (D10), and the Envelope Shaper to Output Channel 1 (A12). Use the Meter to display the Trapezoid voltage (B11) and adjust the Envelope Shaper timing controls for a medium cycle about 2 seconds long. Turn the TRAPEZOID LEVEL control up so that the Meter sweeps across most of the scale (the Meter switch should be set to CONTROL VOLTAGE). Now use the Trapezoid voltage to affect three different variables of the sound:

1 *Use the Trapezoid to control the frequency of the Oscillator with a pin at I11. The pitch swoops in time with the Attack/Decay cycle. The FREQUENCY control of the Oscillator still has an effect, but now it only alters the average frequency.*

2 *The Trapezoid will control the filtering if the pin is moved to N11. Once again, it is the frequency of filtering which is controlled and this can be very useful in instrumental simulation.*

3 *The Trapezoid can be used to control the Output level. Since the Envelope Shaper is already controlling the Output level of Channel 1, put a pin at C10, which takes the Filter output to Channel 2, bypassing the Envelope Shaper. Put another pin at P11, and Channel 2 level will fluctuate because the Trapezoid is now controlling its level. Note that the two channels are "out of phase" - when one is loud, the other is soft, and* vice versa. *In fact, with suitable adjustment of the timing controls, it is possible to make the sound appear to move from one side to the other. It may be necessary to set the TRAPEZOID LEVEL control to 8 or 9 to achieve this effect. See Fig.III-1.*

In some electronic music systems, great care is taken to distinguish signals and controls. In the SYNTHI there is no such distinction, and the result is a considerable economy in wiring and number of devices. Any voltage may be used either to produce a sound or to control a device. All we can say is that voltages with very low (often subsonic) frequencies will more often be used for control, and voltages of higher (audible) frequencies will normally be used as sounds.

If you look at the top of the Patchboard, you will see that the inputs are divided into two sections *Signal* Inputs and *Control* Inputs. So far we have used the Signal Inputs mostly, and we will now look at the Control Inputs in more detail; we will refer to a Control Input by the letter in the corresponding at the bottom of the Patchboard, I, J, etc. The Control Inputs do not provide any new functions, but they do enable the manual controls to be altered automatically. They do not override the Manual Controls, but add their own effect to them.

I,J,K: Voltages in these columns control the frequencies of the three Oscillators. The range of voltage control is greater than that provided by the knob. Oscillators 1 and 2 have the same sensitivity, so they can be set to a given interval, say a fifth apart, and that interval will be maintained as the frequencies are changed by the same voltage. There are no "forbidden" combinations on the SYNTHI - an Oscillator can control its own frequency, or the Oscillators can be connected in a ring, controlling one another.

L: This column controls the decay time of the Envelope Shaper; +2V lengthens the longest decay time (knob set to 10) by about 50% to 25 seconds.

M: This column controls the proportion of reverberated signal mixed with direct signal when the Reverberation Unit is used. It can be switched on and off rapidly, permitting unusual effects.

N: This column controls the frequency of the Filter. As with the Oscillators, the effective frequency range of the Filter is increased by voltage control.

O,P: These two columns control the signal level of Output Channels 1 and 2. When they are in use it will often be necessary to adjust the OUTPUT LEVEL controls to balance the channels correctly.

Experiment 9

In this experiment we give a simple example of the use of each voltage control input. It is a useful exercise to build more elaborate patches from them, trying to discover more interesting sounds. We do not give control settings in detail, but leave it to you to discover useful combinations of controls.

(a) Patch: A12, D3, I4. Oscillator 2 is used to control the frequency of Oscillator 1. A good vibrato is obtained with Oscillator 2 controls at 3, 5, 0, 1.

SYNTHI Dopesheet
E.M.S.(London)Ltd.

Name: EXPERIMENT 9 (c)
Sheet:
Patch:
Start:
End:
Notes:

Description:

External Connections:

C	
S	

FILTER/OSC

RINGMOD

9

OSCILLATOR 1

7	5	9	9

ENVELOPE SHAPER

OSCILLATOR 2

OSCILLATOR 3

5	5	5	

NOISE GENERATOR

OUTPUT FILTER

5	

CHANNEL 1 - - OUTPUT - - CHANNEL 2

6			

REVERBERATION

INPUT LEVEL

RANGE

7	7

External Patch:

		SIGNALS								CONTROLS								
		1out	meter	2out	envelope	r.m. A	r.m. B	reverb	filter	osc freq 1	osc freq 2	osc freq 3	decay	reverb	filter	out lev 1	out lev 2	
out chan	1																	1
out chan	2																	2
osc 1						●												3
osc 2																		4
osc 3	⊓⊔						●			●								5
osc 3	/\/																	6
noise																		7
inputs	1																	8
inputs	2																	9
filter																		10
trapezoid																		11
env signal																		12
ring mod		●																13
reverb																		14
stick	↔									●								15
stick	↕											●						16
		A	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P	

(b) *Patch: A12, D4, J7, J11. Oscillator 2 is controlled by Noise and Trapezoid. The Noise LEVEL control can be adjusted to give an almost pure tone or a wide range of coloured noise.*

(c) *Patch: A13, E5, F3, I5, I15, K16. This is a more complicated patch than we have used before, and it is illustrated on a Dopesheet. Oscillator 3 is controlled by the Joystick, and is ring-modulated with Oscillator 1 to produce "plucked" sounds. The other Joystick output controls the frequency of Oscillator 1 (I15) so both the repetition rate and the pitch of the notes can be altered in "performance". The Joystick gives good manual control of any two parameters of the sound, and a use can be found for it in practically every patch.*

(d) *Patch: A12, D7, L6. Set the Envelope Shaper controls to 0, 5, 0, 0, 10 and the Noise COLOUR to 5 and LEVEL to 10. Oscillator 3 (initially at 5, 6, 0, 6) controls the Decay time and hence the cycle duration of the Envelope Shaper.*

(e) *Patch: A14, E5, F3, G13, M5*
Oscillator 1: 5, 5, 8, 0
Oscillator 3: 5, 5, 5, 0
Ring-modulator: 10
Reverberation: 7, 10
Correctly set up, this patch can produce a sound resembling a short roll on a pitched drum. Without the pin at M5, which is controlling the reverberation mix, it is merely a blur. This patch illustrates the value of controlled reverberation.

(f) *The patch is shown on the Dopesheet. It illustrates another way of making a dull sound from an unmodified Oscillator into an interesting musical timbre, in this case by controlling the Filter frequency with Oscillator 3 and the Trapezoid.*

(g) Patch: A3, C12, D3, O11. This patch moves the sound (in this example it is Oscillator 1, but it could be any sound) from one channel to the other. It depends on the fact that the Trapezoid output is inverted, so that the Channel controlled by the Trapezoid is loud when the output of the Envelope Shaper is loud, and vice versa.

Modulation

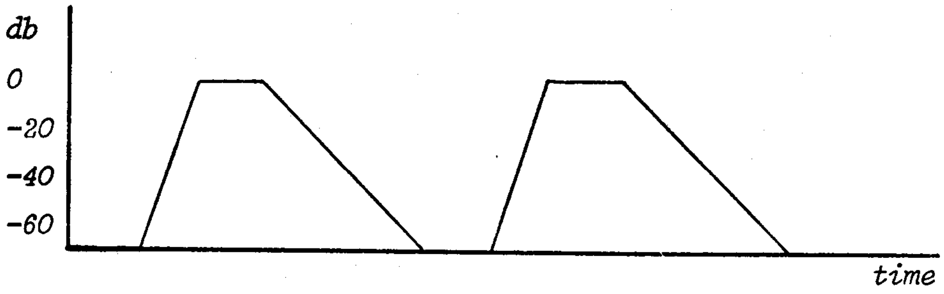
When a signal is modified in some way by another signal, we say that the first is *modulated* by the second. The most common forms of modulation in conventional music are small fluctuations in the amplitude or frequency of a note. Both kinds of fluctuation generally have a frequency of 5 - 10Hz and a proportion of 2 - 5%. It is convenient to distinguish between amplitude modulation by calling the first *tremolo* and the second *vibrato*. Unfortunately, this convention is not always adhered to: for example, violinists talk about *vibrato* (frequency modulation produced by altering the length of the string with one finger), *fingered tremolo* (same but with two fingers), and *bowed tremolo* (amplitude modulation produced by bowing). Experiment 9(a) showed how vibrato is obtained with the SYNTHI, using Oscillator 2 to modulate Oscillator 1. This can be made into a demonstration of tremolo by moving the pin at I4 to O4, so that Oscillator 2 is controlling the level of Output Channel 1.

The mathematics of amplitude modulation is not hard if you know some trigonometry. Suppose we have a pure sine tone, A, with amplitude a and frequency f, so that

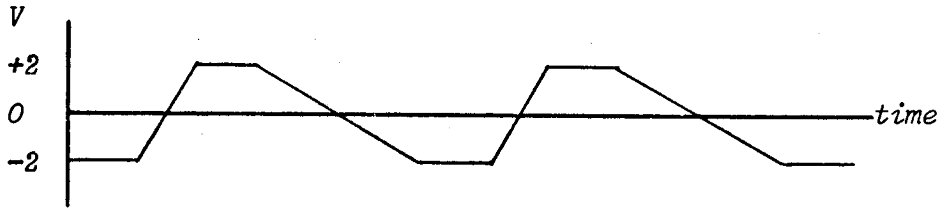
$$A = a.\sin(2\pi f t)$$

Then suppose that the amplitude is disturbed by a small signal

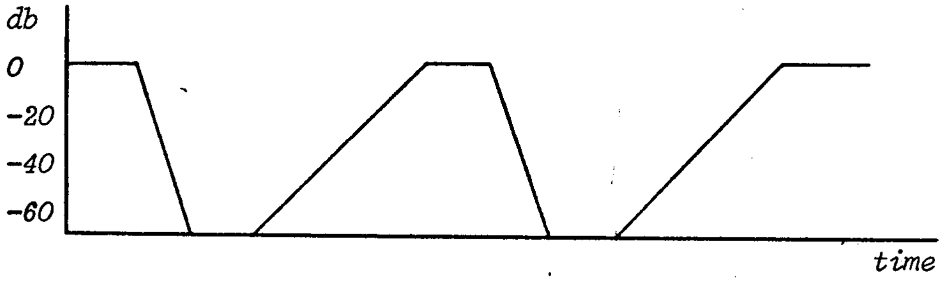
$$B = b.\sin(2\pi g t)$$



(a) Envelope Shape Output Level

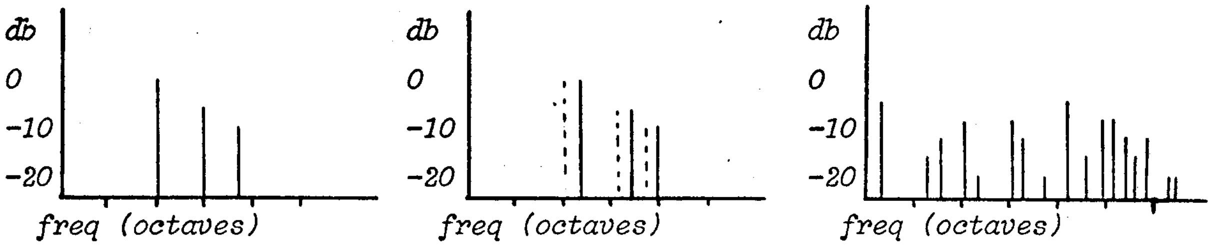


(b) Envelope Shaper Trapezoid Voltage



(c) Output Level of Amplifier controlled by Trapezoid

Fig. III-1: Panning with the Envelope Shaper and Trapezoid Voltage



(a) Spectrum of C
(261Hz)

(b) Spectrum of E
(330Hz)

(c) Spectrum of
ring-modulation
of C and E

Fig. III-2: Spectra of C, E and their Ring Modulation products so that the apmlitude modulated signal, A^1, is given by

$$\begin{aligned} A^1 &= \{a + b.\sin(2\pi gt)\}\ .\sin(2\pi ft) \\ &= a.\sin(2\pi ft) + b.\sin(2\pi gt)\ .\sin(2\pi ft) \\ &= a.\sin(2\pi ft) + \frac{1}{2}b.\cos\{2\pi(f-g)t\} + \frac{1}{2}b.\cos\{2\pi(f+g)t\} \end{aligned}$$

In most cases of amplitude modulation, b is less than a and g is less than f. If this is so, we can analyse the modulated signal A^1 in the following way:

$a.\sin(2\pi ft)$ is the original signal
$\frac{1}{2}b.\cos\{2\pi(f-g)t\}$ is a smaller signal ($b<a$) at a slightly lower frequency
$\frac{1}{2}b.\cos\{2\pi(f+g)t\}$ is another smaller signal with a slightly higher frequency $f+g$.

Thus we have introduced two signals with new frequencies, above and below the original. These are called *sidebands*.

The algebra for frequency modulation is more complicated and the sidebands produced cannot be found by elementary trigonometry. The expression for the signal A frequency modulated by B is

$$A^1 = a.\sin[2\pi\{f+b.\sin(2\pi gt)\}t]$$

Ring-modulation is a third kind of modulation which is often used in electronic music. The signals fed to a ring-modulator are multiplied, and the result is much richer in overtones than either of the input signals. To see why this is, consider two pure tones A and B, with amplitudes a and b, frequencies f and g, so

$$A = a.\sin(2\pi ft)$$

$$B = b.\sin(2\pi gt)$$

Then the ring-modulated signal, C, is given by

$$\begin{aligned} C &= A.B \\ &= ab.\sin(2\pi ft).\sin(2\pi gt) \\ &= \frac{1}{2}ab.\cos\{2\pi(f-g)t\} + \frac{1}{2}ab.\cos\{2\pi(f+g)t\} \end{aligned}$$

That is, the result is a sound with two frequency components of equal amplitudes and frequencies which are the sum *f+g* and difference *f-g* of the original frequencies. If we put in a more complex sounds with more than one frequency component, all the frequency components of each sound will interact in this way and many new frequencies will be produced. Fig.III-2(a) shows the spectra of two sounds each with two overtones, the fundamentals being at middle C and at E a third above. Fig.III-2(c) shows the spectrum of the result of ring-modulating these two sounds, and it has eighteen different frequency components.

The result of ring-modulation is sometimes harsh because many of the new frequencies are not related simply to each other. However, instruments such as bells and chimes have resonances which also are not related in a musical way, and the ring-modulator can be used to simulate them, although it is usually still necessary to filter out some of the new sounds which it introduces.

Experiment 10

The patch and control settings for this experiment are shown in the Dopesheet. The Keyboard is used to provide a pitch voltage at Input Channel 1, and Input Amplifier 1 LEVEL should be adjusted accordingly. The tuning of Oscillators 1 and 2 is critical since very small changes in their relative pitches will produce sounds of widely varying timbre. The sound can be made to give deep bell sounds or lighter chime sounds by adjusting the Filter FREQUENCY.

SYNTHI Dopesheet
E.M.S.(London) Ltd.

Name: EXPERIMENT 10
Sheet:
Patch:
Start:
End:
Notes:

Description:

External Connections:

C ☐
S ☐

FILTER/OSC
7 5 9

RING MOD
7

OSCILLATOR 1
8 5 6 0

ENVELOPE SHAPER
0 2 8 0 ○ 9

OSCILLATOR 2
7 5 5 5

OSCILLATOR 3
7 5 1 4

NOISE GENERATOR
○ ○

OUTPUT FILTER
5 ○

CHANNEL 1 -- OUTPUT -- CHANNEL 2
6 ○ ○ 6

REVERBERATION
5 9

INPUT LEVEL
7 ○

RANGE
○ ○

External Patch:

		SIGNALS								CONTROLS								
		1out	meter	2out	envelope	r.m. A	r.m. B	reverb	filter	osc freq 1	osc freq 2	osc freq 3	decay	reverb	filter	out lev 1	out lev 2	
out chan	1																	1
	2																	2
osc 1					●	●												3
osc 2							●											4
osc 3	⊓⊔										●							5
	∿														●			6
noise																		7
inputs	1									●								8
	2																	9
filter					●													10
trapezoid																		11
env signal								●										12
ring mod									●									13
reverb		●		●														14
stick	↔																	15
	↕																	16
		A	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P	

Reverberation

When a source in a large room or hall stops producing sound, the sound we hear does not cease abruptly, but continues to echo round the room for a short time. This is called *reverberation*. In assessing the reverberation time of a particular room, we make a loud abrupt sound, and measure the time taken for it to decay to one-millionth of its original intensity. The time depends on frequency, so the measuring device must use a burst of white noise (sometimes a pistol shot) or a sound with many frequency components.

A classroom has a reverberation time of at most 1 second: a longer time would make speech hard to understand. A small hall for chamber music has a reverberation time of 0.9 to 1.7 seconds, and a large hall for orchestral music may be from 1.5 to 2.5 seconds. Cathedrals and large churches have long reverberation times - as much as 8 seconds - and are suitable only for choral and organ music written for them.

Electronic music is in some ways like instrumental music recorded in a room with no reverberation at all. It may be interesting musically but it is dull and lifeless to listen to. Having constructed electronic analogues of acoustic oscillators and filters and so on, to complete the picture we ought to simulate reverberation. But there is a difficulty because reverberation involves long delays, which are difficult to produce with purely electronic techniques. There are various ways of getting round this problem. The best is to use a soundproof room with a loudspeaker and a microphone inside it; the loudspeaker makes noises, and the microphone detects the noises directly, and also the reverberation from the walls of the room. A good alternative is to use a large steel plate with *transducers* on it; one transducer is similar to a loudspeaker but it makes the plate move rather than the air around it, and the other corresponds to the microphone but it is designed to pick up vibrations of the plate. The principle is that echoes from the edges of the plate are similar in nature to echoes in a room.

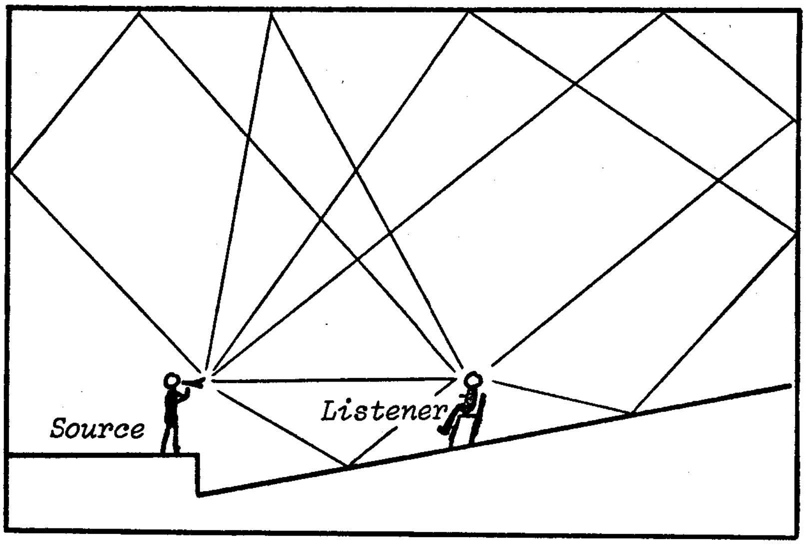


Fig. III-3: Reverberation in a Hall

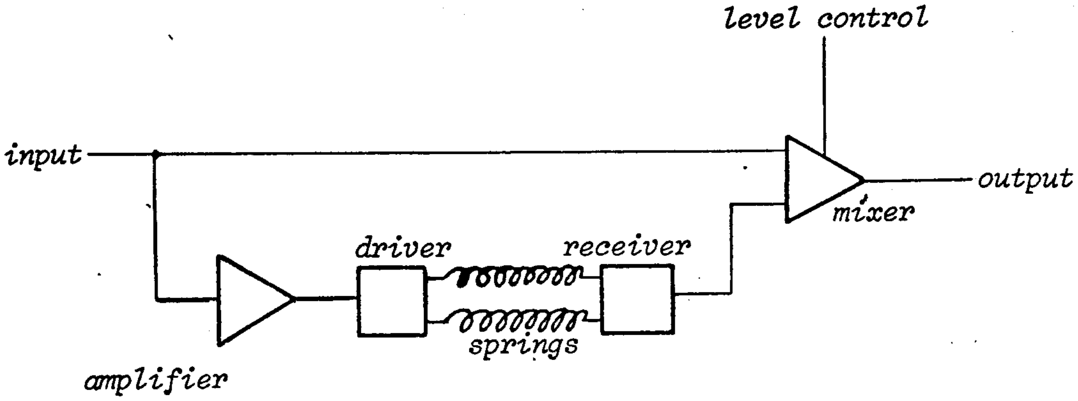


Fig. III-4: Spring Reverberation Unit

Unfortunately, we could not build either a room or a large steel plate into your SYNTHI, but we did manage to fit in two small springs, and these provide an effective imitation of reverberation in a small space. Once again, two transducers are used and signals are sent along the springs by the first and collected at the other end by the second. Two springs are used rather than one to simulate the complex "multiple echo" nature of live reverberation; the delays are slightly different so that the echoes do not reinforce one another. In the SYNTHI, the springs have delays of approximately 25 milliseconds and 30 milliseconds, the time taken for sound to travel 23 feet and 27 feet respectively.

In the SYNTHI, the amount of reverberation can be altered by the Reverberation MIX control. A large amount of reverberation (MIX at 7 - 9) is a very dramatic effect but is seldom required. It is often much more effective to give a dry sound a little "life" with reverberation MIX at 3 - 5. It is usually best to patch the Reverberation Unit last in a chain of devices, since placing it early in the chain can make the final sound muddy, but sometimes this effect can be exploited, as is shown in Experiment 11.

Note: *Unlike any of the other SYNTHI devices, the Reverberation Unit uses a mechanical link which is subject to external vibration. This has two effects: the first is that when reverberation is used, the SYNTHI is sensitive to mechanical shocks of vibration, which will be picked up by the springs and amplified. The other is that the sound of the Monitor loudspeakers may be picked up in the same way, resulting (at worst) in feedback and howling. It is therefore best to avoid high levels of reverberation when the internal loudspeakers are being used.*

SYNTHI Dopesheet
E.M.S.(London)Ltd.

Name: EXPERIMENT 11
Sheet:
Patch:
Start:
End:
Notes:

Description:

External Connections:

C	
S	

FILTER/OSC

7	6	9

RING MOD

OSCILLATOR 1

6			3

ENVELOPE SHAPER

10	2	9	0	3	9

OSCILLATOR 2

3	5		1

OSCILLATOR 3

NOISE GENERATOR

0	9

OUTPUT FILTER

5	

CHANNEL1 - - OUTPUT - - CHANNEL 2

6			

REVERBERATION

7	7

INPUT LEVEL

RANGE

External Patch:

		SIGNALS								CONTROLS								
		1out	meter	2out	envelope	r.m. A	r.m. B	reverb	filter	osc freq 1	osc freq 2	osc freq 3	decay	reverb	filter	out lev 1	out lev 2	
out chan	1																	1
	2																	2
osc 1								●										3
osc 2										●								4
osc 3	⊓⊔																	5
	∿																	6
noise										●								7
inputs	1																	8
	2																	9
filter					●													10
trapezoid										●					●			11
env signal		●																12
ring mod																		13
reverb									●									14
stick	↔																	15
	↕																	16
		A	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P	

Experiment 11

The patch and control settings are shown in the Dopesheet. The sound is rather uninteresting with no reverberation (MIX at 0) but can be given a haunting quality by increasing the MIX until the optimum level is found.

This patch is likely to cause howling if the experiment is performed using the internal loudspeakers.

Level Controls and Mixing

Every device on the SYNTHI has an output level control. At first, it may seem unnecessary to have, say, four different level controls in a chain of four devices, but in fact they are all important. In order to achieve the best possible sound from a SYNTHI, it is important that at each point in a chain of devices the level should be correctly adjusted. It is not possible to define a set of hard and fast rules for setting levels, but we will give some general guides.

Firstly, the level should remain as constant as possible through the chain of devices. If at any stage the level is high, the next device may distort. Levels can be checked by patching each device in turn to either an Output Channel or the Meter.

Some devices require special treatment. The Filter will produce more dramatic effects if the signal fed to it has a very low level, and the Ring Modulator will "break through" (in fact signals will go straight through to the output, usually distorted) if the signal is larger than about 1.5V p-p. The Reverberation Unit is capable of giving very good results, but it should not be overloaded, especially at high settings of the MIX control.

A very low level at any stage should be avoided, since the next stage will have to be set at high gain which may add to the noise level.

Mixing is the combination of two signals into one. In Oscillator 1 and 2 the two waveforms are mixed, so each oscillator has two level controls but only one output. Other devices can be mixed in the SYNTHI by the simple expedient of putting several pins in one column. Control voltages as well as signals can be mixed, and this facility is often used, for example, when adding vibrato. Clearly, when two signals are mixed both the absolute and relative levels are important.

Panning

It is obvious that with two loudspeakers a sound can be made to come from one or the other. It is not quite so obvious that the sound can be made to come from any point in between them, by sending a proportion of the sound to each channel. In the early days of stereo recording, which depends on this fact, there was a great deal of dispute about whether it was necessary to have phase differences between the loudspeakers as well. The answer is complicated, but roughly speaking, our ears make good use of any information they can get, and dividing the sound proportionally works sufficiently well for most purposes. *Panning* (a word probably borrowed from the film industry) means moving the sound from one side to the other. In the SYNTHI this can be done manually, using the PAN controls on the Output Channels, or automatically using the Envelope Shaper and Trapezoid (see Experiment 9(d)). The PAN controls, which enable two sounds to be moved independently between the loudspeakers, make the SYNTHI a true stereophonic machine, not simply a two channel machine.

A signal which is faded slightly and simultaneously given more reverberation seems to recede into the distance. Using this fact and simultaneously panning the signal gives effective three dimensional control.

Note: The PAN controls are only operative when external amplifiers are used. To clarify this point, refer to Fig.III-5, which is a functional diagram of the Output Channels.

Output Filters

The Output Filters are incorporated in the Output Amplifiers, and they are a kind of "tone control". They should be used for making final adjustments to the sound, and also for matching the SYNTHI to power amplifiers connected to it or to the acoustics of the room. These controls should be returned to the central position before setting up a new patch.

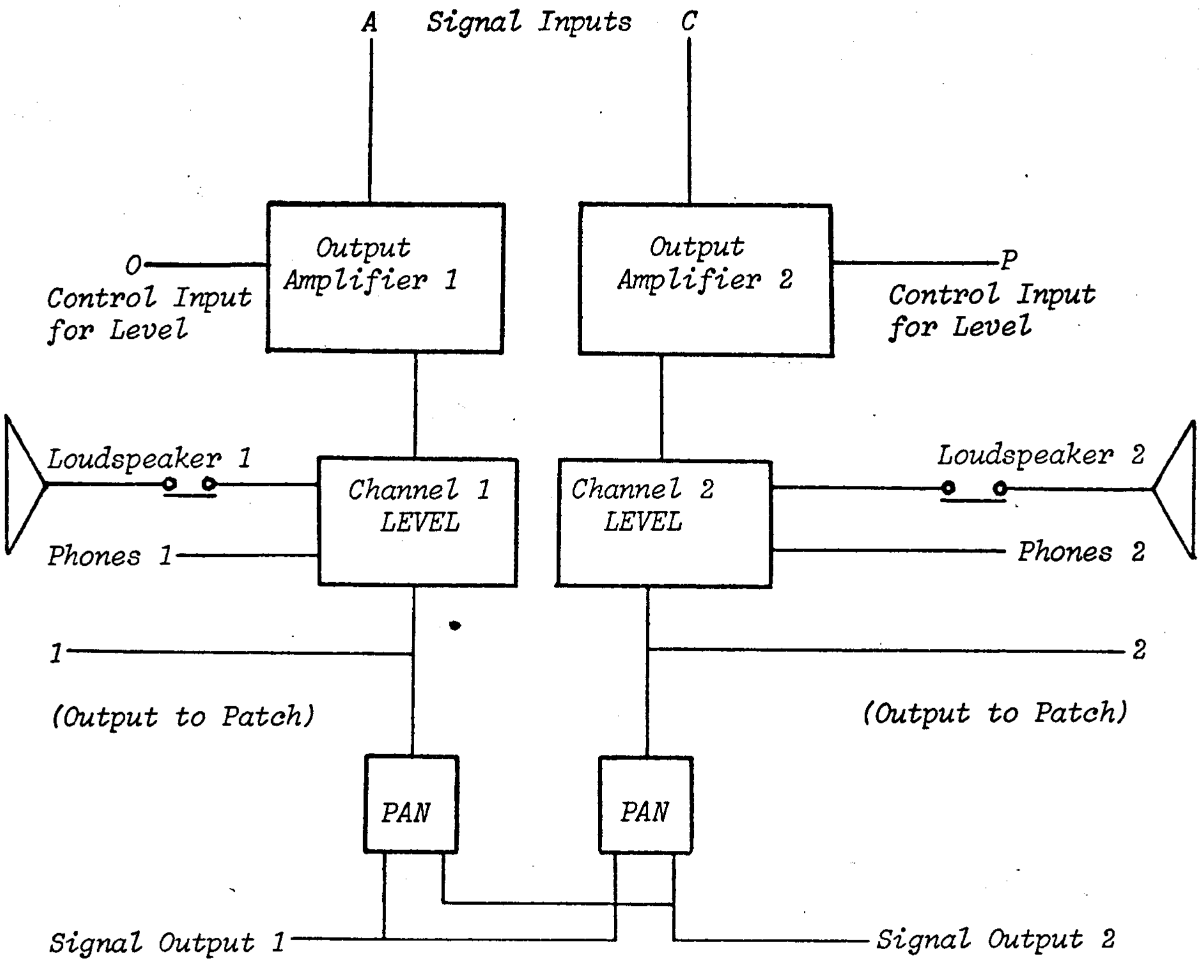


Fig. III-5: Block Diagram of Output Amplifiers

Letters (A,C,O,P) and numbers (1,2) are Patchboard connections.

CHAPTER IV: THE KEYBOARD AND SEQUENCER

<u>Note</u>: *This chapter describes the use of the electronic touch-keyboard and the digital sequencer which are part of the SYNTHI AKS. It does not apply to the EMS keyboards DKO and DK1, or to the EMS Sequencer 64.*

As we have seen already, it is often easier to make a piece of electronic equipment than it is to devise a way of controlling it. However complex a synthesizer we make, performances on it are ultimately limited by the player's ability to turn knobs. We can only extend the power of the synthesizer by making it easier to play.

The SYNTHI AKS uses a touch-keyboard engraved with a piano-like scale of black and white notes. Although it is possible to play tunes on this keyboard as if it was a one-finger organ, we should think of it more as an additional device for providing two *control voltages* which may be connected to any of the voltage controlled devices in the SYNTHI. As a special case, we may choose to control the pitch of an oscillator and the loudness of an amplifier in order to play tunes, but this is only one application.

The *Sequencer* is a special form of storage unit. When the Sequencer is *recording,* any key touched will cause a control voltage and a time to be stored. Later, when the *sequence* is *played back*, each voltage reappears at the appropriate time after the beginning of the sequence. The Sequencer is rather like a tape-recorder (which also has record and playback functions) but as it is completely electronic, the pitch and speed of a sequence can be changed independently, and there is no "rewind". The Sequencer can only be used to record notes from the touch keyboard; it will not record other instruments or voltages. The largest number of events that can be stored is 256.

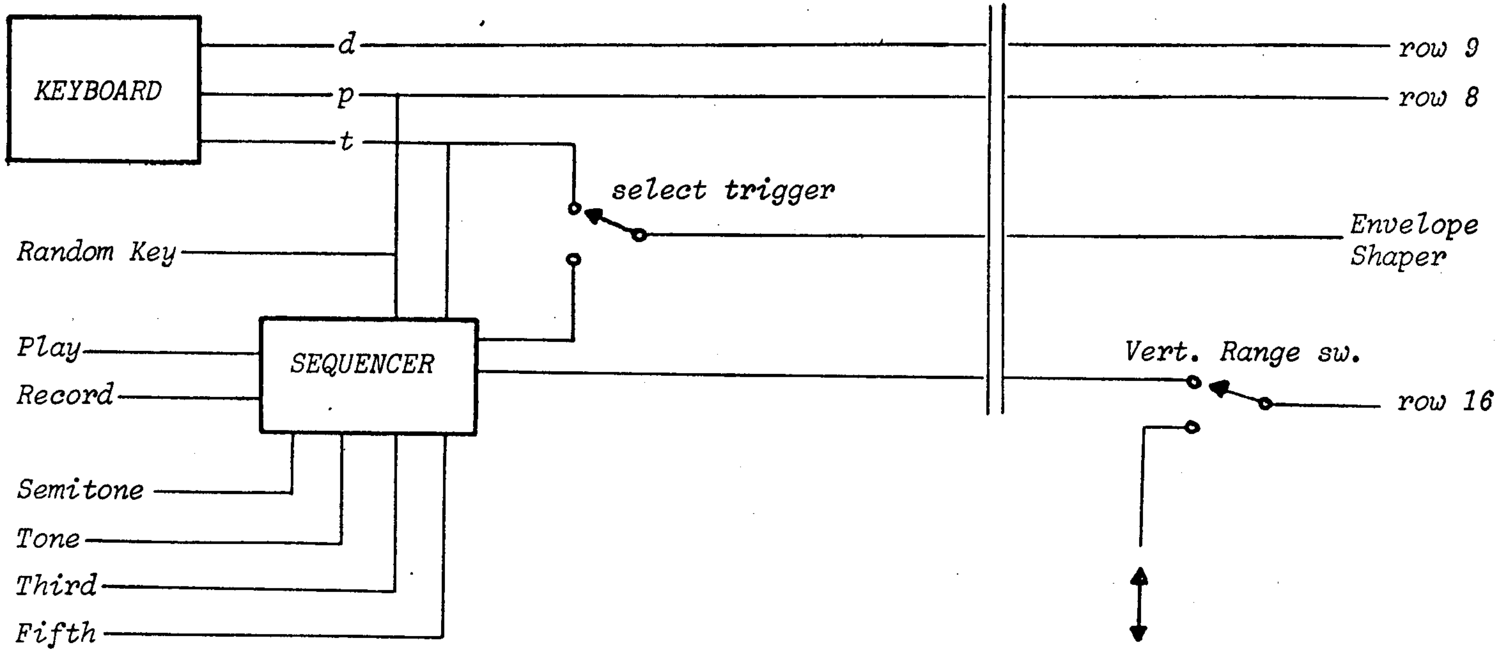


Fig. IV-1: Simplified Block Diagram of the SYNTHI AKS Keyboard and Sequencer

Functional Description

Fig.IV-1 is a simplified block diagram of the Keyboard and Sequencer. The dotted lines represent the lead which joins the Keyboard unit to the SYNTHI proper.

The Keyboard generates three voltages: *p* is a voltage which depends on which key is touched, and is called the *pitch voltage; d* depends on how hard it was struck and is called the *dynamic voltage*, and *t* is a *trigger voltage* to fire the Envelope Shaper. The pitch about may be adjusted by the REALTIME PITCH SPREAD control, and determines the interval (tone, semitone, quartertone, etc) between each contact. The dynamic voltage is an inverted trapezoid which is normally used to control an output amplifier. The trigger is switched so that the Envelope Shaper can be controlled either by the Sequencer or by the keys being touched; this switch is on the extreme right of the Keyboard controls. The RANDOM key plays a note at random when it is touched and is connected to the pitch voltage line.

When the Sequencer is recording, it receives pitch and trigger signals and passes them straight through so you can hear what you are recording. The SEQUENCER PITCH SPREAD control enables you to tune the Sequencer independently of the Keyboard. The Sequencer sends the direct or recorded pitch voltage to row 16 of the Patchboard so that sequences can accompany playing. (The vertical RANGE control must be switched to zero and only horizontal movements of the Joystick are effective when the Sequencer is used.) The contacts marked FIFTH, THIRD, TONE and SEMITONE raise the pitch voltage from the Sequencer when they are touched. The pitch change depends on the tuning, and it is only as stated when the tuning is chromatic (12 notes per octave); for other tuning, the FIFTH raises the pitch 7 notes, and the others 4, 2 and 1 notes respectively.

The Sequencer has a clock, and the duration of the sequence depends on the speed at which the clock is running. In order to achieve precise timing, you should run the clock at the fastest rate you can without the sequence being too short for your requirement. The duration is adjusted by the SEQUENCE LENGTH control on the left of the Keyboard, and the duration of the sequence is read from the Meter as described below. The SEQUENCE LENGTH control determines the *maximum possible length* of the sequence: you determine the *actual* length of the sequence by pressing the PLAY contact when you have finished recording.

The Sequencer provides another output which is connected to the Meter when the latter is switched to CONTROL VOLTAGE. This is a reading of the Sequencer clock; it is zero at the beginning of the sequence, and moves to the right, reaching maximum deflection at the end of the sequence. The Sequencer is always recycling, whether it is in record or play mode, and the Meter indicates its position in the cycle. The Meter should be used to adjust the sequence duration before recording as described in the next experiment. The Meter can still be used to measure control voltages with reasonable accuracy since a pin in the Meter column will override the Sequencer output.

Experiment 12

This experiment is in two parts: (a) describes a simple application of the Keyboard alone, and (b) shows how the Sequencer is used. For simplicity, both experiments involve playing tunes.

(a) *To use the Keyboard alone, patch D3, A12 (Oscillator 1, Envelope Shaper, Output). We are using the pitch voltage from the Keyboard to control Oscillator 1 frequency, so put a pin at I8. Since the pitch voltage comes through Input Amplifier 1, turn INPUT LEVEL 1 to*

7 and adjust the Oscillator to give a reasonable pitch and loudness. The Envelope Shaper knob on the Keyboard should be set to REALTIME. When you touch a contact, the ATTACK lamp will light, and if the Envelope Shaper is correctly set, you will hear a note.

To tune the Keyboard, touch contacts an octave apart alternately (not together - the Keyboard can only play one note at a time) and adjust the REALTIME PITCH SPREAD control until the two notes are an octave apart. Alter the Oscilloscope FREQUENCY and Envelope Shaper controls to obtain notes of different register and duration.

The dynamic voltage from the Keyboard is connected to row 9 (Input 2). An extra pin can be put at 09 for loudness control or at L9 for decay control. If you patch reverberation (remove A12 and substitute G12, A14) then you can control it from the Keyboard (M9). Play random notes as well: they can be interspersed in a melody if desired.

(b) *To use the Sequencer, patch D3, A12 again, and put the pitch voltage pin at I16. The ENVELOPE SHAPER control on the Keyboard should be turned to SEQUENCER (since this is where the trigger pulses will come from) and the vertical RANGE control should be set to zero (there is a switch at that position). Set the Meter mode switch to CONTROL VOLTAGE.*

The Meter will be moving slowly across the scale to the right, and resetting at the end of its travel. This represents the maximum length of your sequence, and you should adjust the SEQUENCE LENGTH control on the Keyboard so that the time in which the Meter moves across is a little longer than the tune which you wish to play. Record your tune as follows:

Touch RECORD. The Meter will move to zero, indicating that the sequence is ready to begin, and will wait until you start playing.

Play your tune.

Touch PLAY. This will make the Sequencer play the tune from the beginning and then repeat it indefinitely.

If you use the Sequencer to play a repeated pattern (such as an Alberti bass figure) it is important to touch PLAY quickly after the last note, or there will be a gap before the sequence starts again each time round.

Once stored, a sequence will play indefinitely. You can alter its pitch with the transposition contacts (FIFTH, THIRD, etc) or Oscillator 1 FREQUENCY, and its tempo with the SEQUENCE LENGTH control.

Some practice is required to use the Keyboard and Sequencer effectively. The Sequencer stores the beginning and end of each note, so the length of time for which you hold a note down should be used expressively. It is important to set the SEQUENCER LENGTH control correctly: incomplete storage or apparent timing errors may occur if it is too long or too short. A good length for recording a sequence is 25 seconds. After recording, a 25 second sequence can be varied from 2 seconds up to as long as 4 minutes. You should leave a small gap after the last note in the sequence, or you may record a "blip". When playing, the PLAY contact can be used to reset the sequence to the beginning, and a sequence can be cued by releasing the PLAY contact. Fortunately, it is very easy to re-record a sequence by touching the RECORD contact again, so start with a simple tune and try to obtain a perfect recording of it.

Tuning the Keyboard

When you touch a note of the Keyboard, you are actually generating a number. The number is in 5-bit binary form (in which form it is very easily stored by the Sequencer) and it is led to a *Digital to Analogue Converter* which turns it into a voltage. This voltage has one of 30 values, depending on which note you touched, and the range of voltage is altered by one of the SPREAD controls. The effect of this is that rather than tuning every note, or one particular note, we tune by *interval*. When the Keyboard is tuned conventionally, the voltage range is about 0.85V for the whole Keyboard, or 0.32V/octave which is correct for SYNTHI Oscillators 1 and 2. If we tuned it to 1.6V, we would have a 5 octave range and each note would be a whole tone apart; if we tuned it to 0.42V we would have a quarter tone scale covering a tenth. Once we have established a "spread" or range, for the Keyboard, we only have to tune one particular note to a particular frequency for the whole keyboard to be in tune.

The procedure for tuning the SYNTHI Keyboard is as follows: Tune the Sequencer by patching A3, I16 and adjusting the SEQUENCER PITCH SPREAD control for a perfect octave (or other interval if you can hear it better). Then choose one note, for example concert A = 440Hz, play it, and adjust Oscillator 1 to the desired pitch.

The procedure for the Keyboard is essentially the same but for one point. The Keyboard pitch voltage is taken through Input Amplifier 1 to row 8 of the Patchboard and so the Keyboard range is effected by the level of the Input Amplifier. It is simplest to preset the INPUT LEVEL to 7 and then not to alter it so long as the Keyboard is in use. The Keyboard can now be tuned with the KEYBOARD PITCH SPREAD control in the same way as the Sequencer.

The next experiment shows you how to accompany a sequence. The Sequencer is used to control Oscillator 1, and the Envelope Shaper is driven by the Sequencer trigger. When a sequence has been recorded, the Keyboard can be used to accompany it using Oscillator 2. The dynamic voltage from the Keyboard is similar to the control voltage produced by the Envelope Shaper but without time controls; here it is used to control the level of Output Channel 2.

Experiment 13

The Sequencer uses Output Channel 1. The signal path is connected with pins at D3, A12 and the pitch control uses one pin at I16. On the SYNTHI, adjust Oscillator 1, the Envelope Shaper, and Output Channel 1, and set the vertical RANGE control to zero. On the Keyboard, set the ENVELOPE SHAPER knob to SEQUENCER. Record a sequence as in Experiment 12 and then turn Channel 1 LEVEL control down while you set up the accompaniment patch.

Connect Oscillator 2 to Output Channel 2 (C4) and adjust the levels so that the sound is just inaudible at mid-range frequencies. Add a pin to J8 for pitch control and another at P9 for dynamic control. Turn INPUT LEVEL 2 up to 10 (the Input Amplifier is amplifying the dynamic control signal) and notes touched on the Keyboard should be audible. The only control of the envelope of each note in this mode is the relative levels of Input Amplifier 2 and Output Amplifier 2, but it is possible to obtain a range of sounds from a staccato "ping" to a legato "organ" sound. Now by turning up Output Channel 1 LEVEL again, you can accompany the sequence.

The timbre of either voice can be enriched by using other devices. You should set up the experiment as described and then improve the timbres with filtering, ring-modulation, reverberation, vibrato, etc.

Finally, we describe a few ways of using the Keyboard without using the Envelope Shaper or dynamic voltage.

<u>*Experiment 14*</u>

(a) *Patch A13, E3, F8, I8. This uses the ring-modulator to produce a short, staccato envelope. An interesting feature of this patch is that the loudness of each note depends on the interval between it and its predecessor.*

(b) *Move E3 to E7. This is a "drum" sound which can be used as part of a patch for playing tunes.*

(c) *The Keyboard and Sequencer can be used as a control for many SYNTHI patches. Try "playing" the Chimes described in Experiment 12, for example.*

CHAPTER V: THE SYNTHI AND OTHER DEVICES

Amplifiers

Although the SYNTHI has internal amplifiers and loudspeakers, these are only intended for monitoring, and in demonstrations or concerts the SYNTHI should be connected to a stereo power amplification system. The SIGNAL OUTPUTS on the SYNTHI provide signals which are suitable for most amplifiers. They should be connected to the AUX or RADIO sockets on a pre-amplifier or integrated amplifier, or they may be connected directly to the line-input of a power emplifier. DO NOT use tape-head or pick-up input sockets on the amplifier because they are intended for much smaller signals.

Amplifiers and loudspeakers are designed to produce speech and music. You should always bear in mind that the SYNTHI can produce sounds which do not occur in speech or conventional music, particularly large signals at very high and very low frequencies, which can damage loudspeakers. If the loudspeaker is distorting badly or ratting, you should turn the levels down or disconnect the SYNTHI immediately.

Fig. V-1 shows a SYNTHI connected to a stereo amplification system.

Tape Recorders

Although the SYNTHI is an excellent instrument for live performance, a tape-recorder is essential if elaborate electronic compositions are to be realised, and two tape-recorders are better still. The outputs of the tape-recorder can be connected to the HI LEVEL inputs of the SYNTHI, and the SIGNAL OUTPUTS of the SYNTHI to the AUX or RADIO sockets of the tape-recorder. Fig. V-2 shows a SYNTHI connected to

SYNTHI Dopesheet
E.M.S.(London) Ltd.

Name: EXPERIMENT 15 (b)
Sheet:
Patch:
Start:
End:
Notes:

Description:

External Connections:

C
S

FILTER/OSC			RING MOD
6	10	7	9

OSCILLATOR 1

7	5	8	0

ENVELOPE SHAPER

2	0	6	5		10

OSCILLATOR 2

OSCILLATOR 3

2	5		6

NOISE GENERATOR		OUTPUT FILTER	
		5	5

CHANNEL 1 - - OUTPUT - - CHANNEL 2

6			6

External Patch:

		SIGNALS								CONTROLS								
		1out	meter	2out	envelope	r.m. A	B	reverb	filter	osc freq 1	2	3	decay	reverb	filter	out lev 1	2	
out chan	1																	1
	2																	2
osc 1					●													3
osc 2																		4
osc 3	⊓⊔																	5
3	∿								●									6
noise																		7
inputs	1					●												8
	2																	9
filter							●											10
trapezoid																		11
env signal		●																12
ring mod		●		●														13
reverb																		14
stick	↔																	15
	↕																	16
		A	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P	

REVERBERATION

INPUT LEVEL

7	

RANGE

a stereo tape-recorder; later in this chapter we will look at ways of connecting more than one tape-recorder.

With a tape-recorder you can record several sounds in succession and then edit the tape to make a composition. By altering the speed of the tape-recorder between recording and playback you can change the speed and the pitch of sounds. If the tape-recorder has a separate playback head you can produce the interesting echo effects described in Experiment 15.

Experiment 15

Connect the AUX or RADIO input of the tape-recorder to the SYNTHI's SIGNAL OUTPUT, and the line-output of the tape-recorder (which must come from a separate playback head) to the HI LEVEL INPUT.

(a) Patch: A12, G8. Any sound presented on Input Channel 1 (for example, a note from the Keyboard) will be given a combination of reverberation and simple echo which is more effective than either separately.

(b) This is a fairly complicated set up and we show it in a Dopesheet. Each time the echo returns from the tape, it is ring-modulated by a sine-tone from the Filter. Simple primary sounds are best, and the Dopesheet shows short notes provided by Oscillator 1. This patch is capable of rather beautiful sounds if some care is taken to get the levels adjusted optimally. The settings on the Dopesheet were obtained from a tape-recorder which had been lined up to give the same level at its output as it received at its input.

Microphones and Instruments

As well as making its own sounds, the SYNTHI is able to make dramatic transformations of sounds fed to it. The MIC 500 ohm inputs are suitable for most air-microphones, contact-microphones and electronic instruments such as organs and guitars. Before connecting anything to the MIC sockets, turn the INPUT LEVEL controls to 0, then plug the device in and slowly turn up the INPUT LEVEL to a suitable setting. It is best to patch the Input Amplifier straight through to the Output Channel (pins at A8, C9) and make sure that a clear undistorted sound is obtainable by direct amplification before attempting treatments. Alternatively, send the signal to the Meter (pin at B8, or B9, Meter switch to SIGNAL LEVEL) and adjust the INPUT LEVEL controls for a maximum indication of 4 - 7. If the signal is to be taken to the Filter or Ring-Modulator it should be set to a fairly low level (3 on the Meter) for best results.

For each new sound, it is best to try out each SYNTHI treatment by itself first, and then to try more elaborate transformations using several devices. Experiment 16 describes how to try out the effect of individual devices, and Experiment 17 demonstrates more complex effects.

Experiment 16

A mono signal is present at Input Channel 1 (row 8) and its level has been set as described in the text above.

(a) *Patch: A10, H8. This filters the sound; the Filter may be controlled manually or automatically. For example, control it with Oscillator 3 (N5 or N6).*

(b) *Patch: A12, D8. The Envelope Shaper can fade the sound in various ways. Try each of the following with various settings of the Timing Controls.*

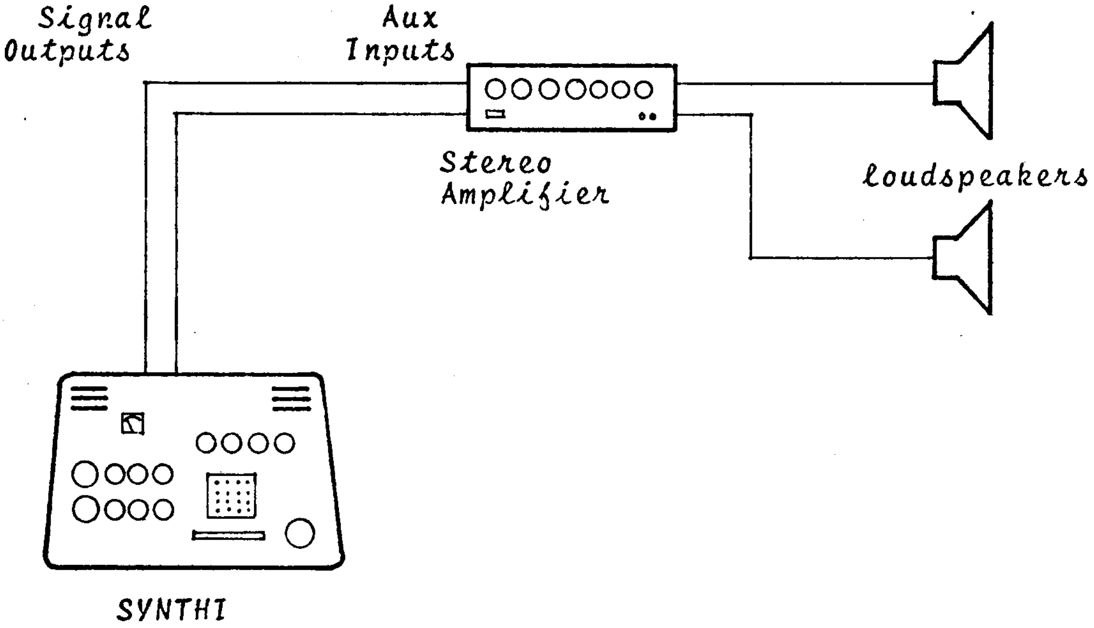


Fig. V-1: The SYNTHI connected to a Stereo Amplifier

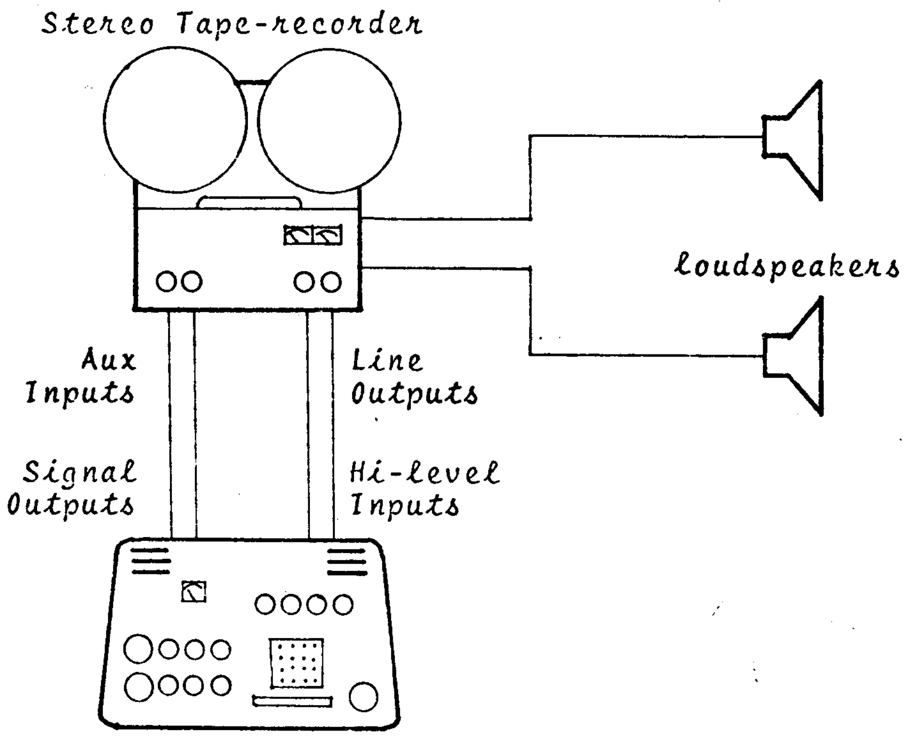


Fig.V-2: The SYNTHI connected to a Stereo Tape-recorder

Automatic: with OFF time at 7 or less the Envelope Shaper will recycle immediately.
Manual: with the OFF time at 10, a new sould will be produced whenever the ATTACK button by the Joystick is pressed.
Threshold Trigger: put a pin at C8, connecting the input signal to Output Channel 2, and put the right-hand loudspeaker switch to TRIGGER. The Envelope Shaper will recycle whenever the input signal reaches the triggering threshold. This can be used to give instruments an entirely different envelope without affecting their timbre.

(c) *Patch: A14, G8. This adds reverberation to the sound. The optimum setting of the MIX control will depend on the nature of the input signal.*

(d) *Patch: A13, E8. The Ring-modulator requires another input before it will do anything. To ring-modulate with a sine-wave, patch E10 and turn the Filter RESPONSE to 10 so the Filter resonates. Try the other Oscillators (E3, E4, E5 or E6), Noise (E7) and also ring-modulate the sound with itself (E8) and explain the result by referring to "Modulation" in Chapter III.*

SYNTHI Peripherals

There are three special purpose units manufactured by EMS for use with the SYNTHI or other electronic music equipment. These are described briefly below but come with their own detailed descriptions.

Pitch to Voltage Converter

This device is rather modestly named, since it will not only convert a pitch to a voltage, but will produce another voltage proportional to the amplitude of the signal (envelope following), and a trigger pulse (for the Envelope Shaper) as well. It is therefore possible to make the SYNTHI accompany a live instrument in tune (or in parallel harmony, an octave lower for example) and with correct rhythm, but with an entirely different timbre and envelope. Inverted voltage outputs are available to the SYNTHI can also accompany in contrary motion, or be loud when the live instrument is quiet.

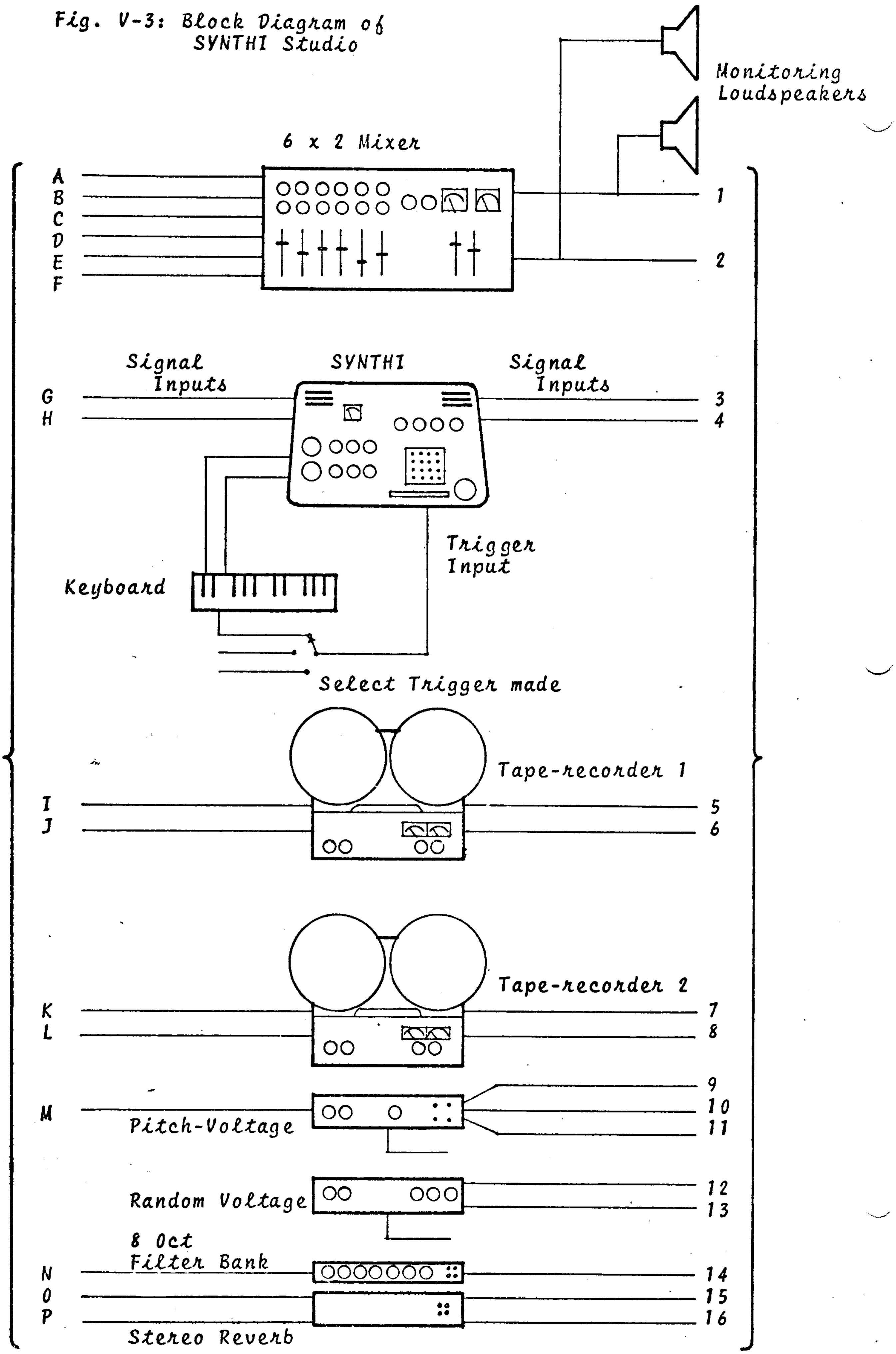
Random Voltage Generator

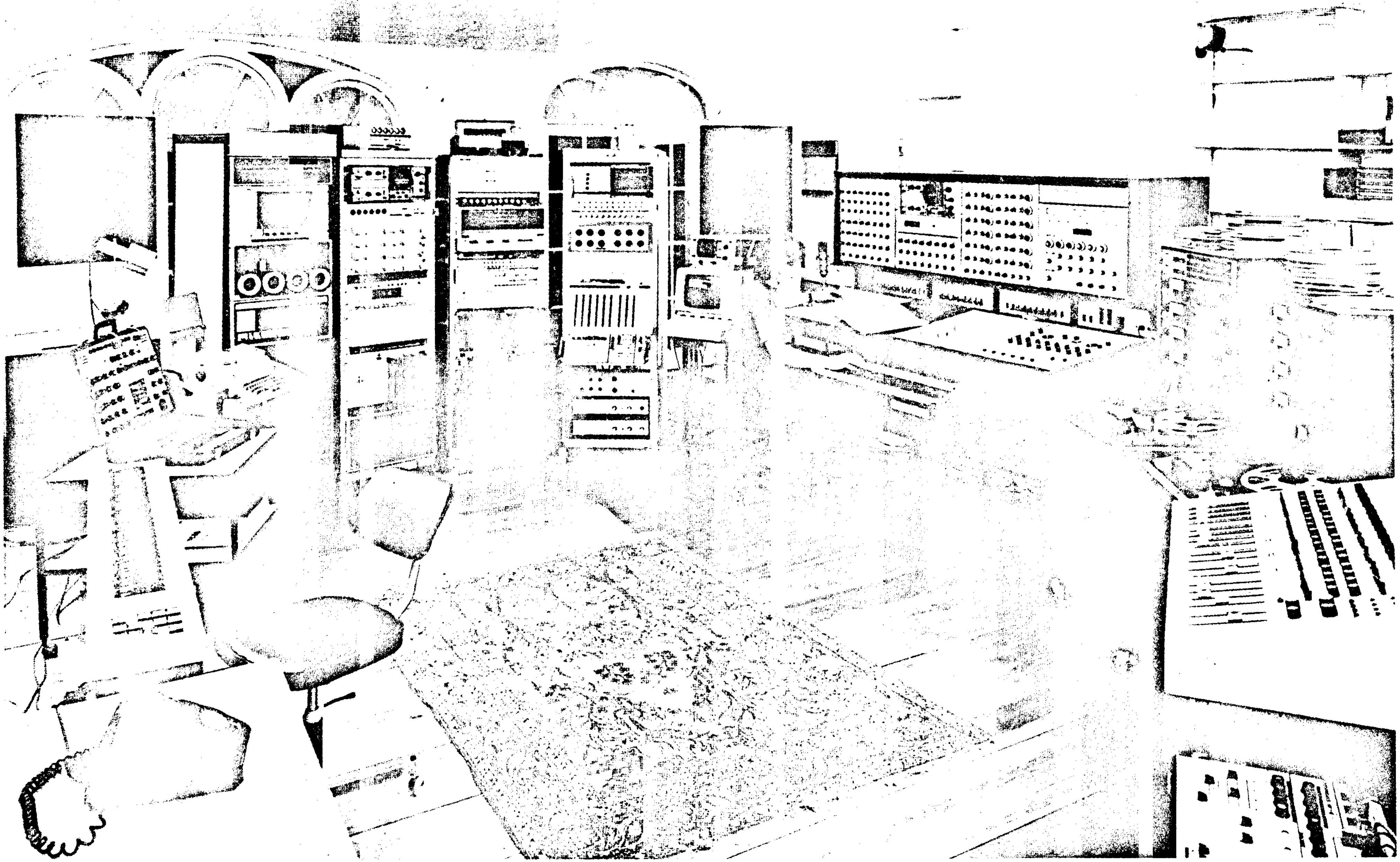
The Random Voltage Generator can be used for a number of effects apart from playing random tunes. One of the problems of electronic music is that the sounds are often too regular, or mechanical, and a small randomly fluctuating voltage, used to provide vibrato or tremolo, for example, can do a lot towards making an exciting "live" sound.

Eight Octave Filter Bank

This unit comprises eight filters tuned to fixed frequencies an octave apart. It enables very subtle modifications to be made to the timbre of the sound, and is useful for eliminating unwanted components, such as hum or noise, without degrading the music content of the signal. The individual output from each filter is available which greatly increases the flexibility of the Filter Bank as well as making it more useful educationally.

Fig. V-3: Block Diagram of SYNTHI Studio





A More Advanced Studio

You will have realised by now that the SYNTHI is a very versatile instrument. It is not an exaggeration to say that with a SYNTHI, a stereo amplifier, two tape-recorders, and some or all of the EMS peripherals, you would have a comprehensive modern electronic music studio. There are many people involved in the production of electronic sounds for concerts, advertising, films and theatre sound effects with fewer resources than this.

It is virtually impossible to say exactly how all these devices should be connected, because this will depend on the particular application. All studios of any complexity therefore have a *patchfield*, which usually consists of a number of jack sockets, with plug-leads to connect them. EMS recommend a *Patchboard,* as used in the SYNTHI, because they are more compact, more reliable, and easier to use. EMS will provide a 16x16 matrix, which is sufficient for many applications, or a 60x60 matrix (as used on the SYNTHI 100) for larger studios. A 60x60 matrix is equivalent to a patchfield with 3600 sockets. Fig.V-3 is a block diagram of a very flexible studio built around a 16x16 Patchboard using the following equipment:

SYNTHI AKS
6 - 2 Stereo Mixer
Stereo Power Amplifier
2 Monitor Loudspeakers
2 Stereo Tape Recorders
E.M.S. Pitch to Voltage Converter
E.M.S. Random Voltage Generator
E.M.S. Eight Octave Filter Bank
2 Reverberation Units

The audio signal connections of each device are taken to the patchboard. The EMS peripherals produce control voltages which can only usefully be taken to the SYNTHI. A four position switch selects the device to trigger the Envelope Shaper:

1 *Keyboard*
2 *Pitch to Voltage Converter*
3 *Random Voltage Generator*
4 *Attack Button*

Microphones can be connected to the mixer, the SYNTHI, or the tape-recorder. Two reverberation units are used in addition to the unit in the SYNTHI because reverberation is important in electronic music and external units, preferably using steel plates, will give superior results.

The photograph (facing page 62) shows the large electronic music studio used by EMS for developing their products. Both the SYNTHI AKS and the larger SYNTHI 100 are shown, as well as the computers used for digital control.

CHAPTER VI: SYNTHI PRESTOPATCHES

In this Chapter we describe the three Prestopatches provided with the SYNTHI: how they work, and how they are used.

In each of the Patchboard pins there is a resistor. When you put a pin into the Patchboard, you are joining two devices in the SYNTHI with this resistor. All the devices connected to the Patchboard are also connected to the 32-way socket by the spare pin store, and so a plug can be pre-wired with resistors forming a patch and plugged in. A plug made in this way is called a *Prestopatch*.

A Prestopatch only contains the pin positions, not the settings of all the controls. Any Prestopatch is therefore incomplete without a Dopesheet which shows how the controls should be set. The Dopesheet should also show which controls should be changed to give particular effects.

EMS will make up Prestopatches to your own specification. Send a Dopesheet with your patch clearly marked on it and we will provide a Prestopatch wired for your patch.

PRESTOPATCH 1: BATTLE

Naturally, the first thing to do when trying a Prestopatch for the first time is to plug it in, set the controls and listen to it. For the Prestopatch to be of any real value, however, you must understand how it works in order that you can alter the controls constructively rather than at random.

Experiment 17

Plug in the Battle Prestopatch and set up the controls as in the Dopesheet. Patterns of explosive sounds should emerge from both channels; the texture can be altered with Oscillator 3 SHAPE control and the Joystick.

Try and understand from the Dopesheet how the Prestopatch works: the principle defice chain is Noise, Filter (H7), Envelope Shaper (D10), Output 1 (A12), Reverberation (G12), Output 2 (C14). The Oscillators and Joystick are used to control Decay, Reverberation, Filtering, and Oscillator frequencies. Discover the function of each device by altering its controls.

You can always add pins to a Prestopatch, by inserting them in the Patchboard in the usual way. See if you can enhance the Battle Patch by introducing new controls or devices.

PRESTOPATCH 2: KEYBOARD

The Keyboard Prestopatch is used to obtain a variety of effects from the Keyboard and Sequencer; you will probably not use all the effects at the same time, and you will need to adjust the controls to get effects individually.

Experiment 18

Plug in the Keyboard Prestopatch and adjust the controls. The patch works in the following way: the Sequencer controls Oscillator 2 (J16) which is connected to the Envelope Shaper (D4), Filter (H12), Reverberation (G10) and Output Channel 1 (A14). Oscillator 3 controls the Filter to provide a tremolo

SYNTHI Dopesheet
E.M.S.(London) Ltd.

Name: BATTLE PRESTOPATCH
Sheet:
Patch:
Start:
End:
Notes:

Description:

* Adjust these controls yourself

External Connections:

C ☐
S ☐

FILTER/OSC: 9 4 9

RING MOD: ○

OSCILLATOR 1: 0 4 0 7

ENVELOPE SHAPER: 0 0 9 0 0 9

OSCILLATOR 2: 0 4 0 9

OSCILLATOR 3: 3 * 0 9

NOISE GENERATOR: 5 9

OUTPUT FILTER: 5 5

CHANNEL 1 - - OUTPUT - - CHANNEL 2: 5 ○ ○ 6

REVERBERATION: 7 10

INPUT LEVEL: ○ ○

RANGE: 8 8

External Patch:

		SIGNALS								CONTROLS								
		1out	meter	2out	envelope	r.m. A	r.m. B	reverb	filter	osc freq 1	osc freq 2	osc freq 3	decay	reverb	filter	out lev 1	out lev 2	
out chan	1																	1
out chan	2																	2
osc 1													●					3
osc 2														●				4
osc 3	⊓⊔																	5
osc 3	∿												●		●			6
noise									●									7
inputs	1																	8
inputs	2																	9
filter					●													10
trapezoid																		11
env signal		●						●										12
ring mod																		13
reverb				●														14
stick	↔										●	●						15
stick	↕									●					●			16
		A	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P	

effect (N5). The Sequencer also makes a drum sound using the Ring Modulator (E7, F16) which is mixed into the Reverberation Unit (G13). The Keyboard controls Oscillator 1 (C3) using the dynamic voltage to provide an envelope (P9). A vibrato is provided by Oscillator 3 (I6).

First, tune the Sequencer by turning Channel 2 LEVEL down and adjusting SEQUENCER PITCH SPREAD. Then turn Channel 1 LEVEL down and use REALTIME PITCH SPREAD to tune the Keyboard. Finally, play both at once and tune them together with Oscillator 1 and Oscillator 2 FREQUENCY controls.

Now record a sequence as described in Chapter IV and then accompany it.

Fine out which controls affect the sound in an interesting way; the following are the most important.

Oscillator 1 LEVELs - sound from Output Channel 2
Oscillator 2 LEVELs - sound from Output Channel 1
Oscillator 2 SHAPE - quasi-phasing effect
Oscillator 3 SQUARE - filtering
Oscillator 3 Triang - tremolo
Ring Modulator LEVEL - percussion
Trapezoid LEVEL - reverberation
Horizontal Joystick - decay

PRESTOPATCH 3: GUITAR

This Prestopatch is capable of providing a variety of effects from a single input. Although it is designed for use with an electric guitar, it can be used with any suitable input, including electronic organ or microphone.

SYNTHI Dopesheet
E.M.S.(London) Ltd.

Name: KEYBOARD PRESTOPATCH
Sheet:
Patch:
Start:
End:
Notes:

Description:

* Adjust these controls yourself

External Connections:

C
S

FILTER/OSC
7 7 6

RING MOD
*
Drum

OSCILLATOR 1
4 5 * *
Mix

ENVELOPE SHAPER
0 0 5 10 * 10
Reverb

OSCILLATOR 2
4 * * *
Phase Mix

OSCILLATOR 3
7 5 4 05
Filter Tremols

NOISE GENERATOR
5 7

OUTPUT FILTER
5 5

CHANNEL 1 - - OUTPUT - - CHANNEL 2
7 7

REVERBERATION
6 7

INPUT LEVEL
7 7

RANGE
7 0

External Patch:

		SIGNALS								CONTROLS								
		1 out	meter	2 out	envelope	r.m. A	r.m. B	reverb	filter	osc freq 1	osc freq 2	osc freq 3	decay	reverb	filter	out lev 1	out lev 2	
out chan	1																	1
	2																	2
osc 1				●														3
osc 2					●													4
osc 3	⊓⊔														●			5
	∿									●								6
noise						●												7
inputs	1									●								8
	2																●	9
filter								●										10
trapezoid														●				11
env signal									●									12
ring mod								●										13
reverb		●																14
stick	↔												●					15
	↕						●				●							16
		A	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P	

Experiment 19

It is best to establish levels first by connecting the guitar or other instrument to Input Channel 1 and patching it directly to the output (A8). For most instruments, the microphone input will be most suitable, but if the instrument overloads the microphone input (so the Input Amplifier Control has to be kept right down to avoid distortion) then the LINE input should be used.

Put in the Prestopatch and set up the controls. The effects possible with this patch include filtering, panning, reverberation, re-enveloping and octave-splitting, and the patch works as follows: the signal coming into the SYNTHI goes to the Ring Modulator (F8), and Filter (H8); the Filter goes directly to Output Channel (A10) and also to the Envelope Shaper (D10) and Reverberation Unit (G10). The Envelope Shaper and Reverberation Unit provide the outputs to the other channel (C12, C14), and the Ring Modulator is fed back into itself (A13, E13). Oscillator 1 LEVEL controls the panning together with Input Level 2 (the Prestopatch incorporates a connection from Oscillator 1 to Input Amplifier 2 which enables an inverted control voltage to be obtained). Oscillator 3 Square LEVEL controls reverberation, and Triang LEVEL controls filtering. The Trapezoid LEVEL controls also filtering, and the Envelope Shaper signal LEVEL controls the re-enveloped sound. The Joystick controls rate of panning horizontally and range of filtering vertically. The Ring Modulator provides an octave splitting effect, producing a note an octave lower than the one fed to it, and should be set up approximately as follows: Set Ring Modulator LEVEL at 8 and put the Bass control on the guitar to full.

This patch is capable of many dramatic effects. Experiment with it and note down setting which you find useful in performance. Here are the most useful controls:

SYNTHI Dopesheet
E.M.S.(London) Ltd.

Name: GUITAR PRESTOPATCH
Sheet:
Patch:
Start:
End:
Notes:

Description:
If this patch is set up with pins, an external patch lead joining SCOPE to HI LEVEL Input 2 will be required.

* Adjust these controls yourself

External Connections:

C ☐
S ☐

FILTER/OSC
6 5 8

RING MOD
7

OSCILLATOR 1
0 6 * 0
panning

ENVELOPE SHAPER
4 0 6 0 * *
filter ENV chop

OSCILLATOR 2
0 5 4 0

OSCILLATOR 3
6 5 * *
reverb filter

NOISE GENERATOR
○ ○

OUTPUT FILTER
5 5

CHANNEL 1 - - OUTPUT - - CHANNEL 2
6 ○ ○ 6

REVERBERATION
6 7

INPUT LEVEL
* 8
instrument

RANGE
9 8

External Patch:

		SIGNALS								CONTROLS								
		1out	meter	2out	envelope	r.m. A	r.m. B	reverb	filter	osc freq 1	osc freq 2	osc freq 3	decay	reverb	filter	out lev 1	out lev 2	
out chan	1																	1
out chan	2																	2
osc 1																		3
osc 2																		4
osc 3	⊓⊔																	5
osc 3	∿																	6
noise																		7
inputs	1																	8
inputs	2																	9
filter																		10
trapezoid																		11
env signal																		12
ring mod																		13
reverb																		14
stick	↔																	15
stick	↕																	16
		A	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P	

Oscillator 1 level - width of panning
Oscillator 2 level - variation in re-enveloping
Oscillator 3 - reverberation and filtering
Trapezoid level - filtering synched with re-enveloping
Ring Modulator level - octave splitting
Joystick horizontal - rate of panning
Joystick vertical - range of filtering

CHAPTER VII: CARE AND MAINTENANCE

The Synthi could hardly be easier to maintain, because the solid state circuitry is designed to run well within its capacity, even under conditions of electrical misuse, and there are no mechanical parts except the Joystick. But the following general points may be helpful:

JACK SOCKETS

These are of standard pattern, and extra jack plugs can easily be obtained. However there are some non-standard sizes on the market, and no plugs should be used if they are a very tight fit or on the contrary move too freely in the socket. They should push home with a firm click and have very little lateral play when in position.

JOYSTICK

The grease in the control slots may eventually dry out, particularly if the studio is kept in a warm place. To service, remove the bottom panel, and take off the red cover plate of the Joystick. Carefully clean the slots and ball joint, and re-lubricate with a little Vaseline of silicone grease. Do not allow any grease to touch the potentiometers and take care not to cross-thread the self-tapping screws when replacing the cover.

KNOBS

If knobs become loose, slacken off the set screw reset at eigher maximum or minimum position, and tighten firmly. The spindles are nylon, and it is normal for the screw to bite in slightly for a firm fixing. If knobs are lost, we can supply more of the correct pattern, or in an emergency any knob suitable for a 1/4" spindle can be used. If knobs are wrenched so that the whole potentiometer becomes loose, attend to it at once before an internal wire is broken. Take off the back, hold the potentionmeter from behind and tighten the nut firmly. If the wires are already displaced check that none are broken and resolder if necessary.

METER

If the pointer does not read zero when the studio is off, the zero can be adjusted with the small perspex screw at the bottom of the dial.

PANELS

Avoid any abrasive cleaner, and never use strong solvents like acetone, trichlorethylene or petrol (gasoline, benzine, essence). The best cleaner is methylated spirits (alcohol), but paraffin (kerosene) or turpentine can be used, though they tend to leave an oily deposit and often an unpleasant smell. Use wax pencil to mark the panels if you wish to do so, rather than lead, particularly hard lead, pensil. Do not use ball point, which may leave a permanent indentation in the finish, or fibre-tipped pens, which often contain an indelible stain.

STORAGE

Solid state devices dislike sustained heat. Hever leave the SYNTHI in a sealed car in summer sun, or in a similar situation. For long term storage choose a cool, dry position. If it is not used for a very long time (a matter of years) there may be trouble with electrolyptic capacitators when it is switched on again, and the best insurance against this is to run the studio for a few hours at least every few months.

DON'T connect unknown inputs with the input level controls wide open. Take particular care when connecting valve (tube) driven equipment to the SYNTHI.

DON'T overrun devices constantly for hours on end if the outputs are not connected, this can happen without the user knowing. The safest course is to remove pins when the studio is finished with for the time being.

DON'T connect unknown mains supplies without checking.

DON'T interconnect jacks at the back of the studio without thinking carefully first. Particularly take care if you connect the Keyboard Jones Socket, because of the danger of short-circuiting the supply rails.

DON'T grossly overrun the internal speakers, which may be damaged and give unreliable results thereafter. For any audience application, connect to external power amplifiers and concert speakers.

DON'T continue to use the studio if unexplained noises, heat or smells occur. Stop and investigate.

Hints and Suggestions

If you set up a patch and there is no sound, check the following:

1 The power supply lamp is lit.
2 The speaker muting switches are up.
3 Levels on all devices, and the Output levels are sufficiently high (typically 5-8 for Sources, 7-10 for Treatments, but there are no "rules"). Check input Amplifier levels if using external devices.
4 Oscillators are not running subsonically. (Below Frequency setting 3 for Oscillators 1 or 2, or 6 for Oscillator 3.)
5 Noise Generator needs 20-30 seconds to start up.
6 Envelope Shaper Off Time may be set too high for automatic recycling (try pressing the Attack Button).
7 Check the patch by swapping pins: the pins are reliable but occasionally dirt in the holes of the matrix board can prevent contact or damage a pin.
8 The Output Amplifiers can be cut off altogether by a large control voltage in rows O and P.

GLOSSARY

AC

Alternating Current. Current whose direction is continually changing, normally at a definite frequency (q.v.)

Amplitude

The maximum instantaneous value of a current (q.v.) or voltage (q.v.) during a half cycle of alternating current. High amplitude peaks can occur in agenerally low level signal.

Attenuate

Make smaller. The opposite of amplify. *An attenuator is usually a network of resistors (q.v.)*

Audio

e.g. a-frequency, a-output. Within the audible range. Electrical signals which would be audible if converted into sound. Compare video - the picture signals in TV.

Bandwidth

The width of a stated range or bank of frequencies, described by its lower and upper limits (e.g. a B of 100Hz-800Hz, or of three octaves upwards from 100Hz). May refer to a band which is being rejected from an otherwise flat response (q.v.), or an accepted band (as in the SYNTHI filter).

Beat

In its audio sense a phenomenon caused by two frequencies so close that the difference tone is obserfed as a throb or pulse - e.g. 250Hz and 252Hz would produce a beat of 2Hz or one every half second.

Capacitance (-tor, -tive) (C)

The amount of electrostatic storage available in an insulator (dielectric) separating two closely adjacent conducting surfaces. Measured in Farads (usually microFarads). A Capacitor is a device possessing Capacitance, and is Capacitive (general rule for properties and devices). A frequency sensitive component with many uses. The electrical equivalent of compliance.

Current (I)

Electrical flow, expressed in Amperes of milliAmperes. A current cannot flow without a potential gradient, or voltage, to impel it. The amount of flow depends on this voltage and on the resistance offered to it (see Resistance)

dB

Decibels, or tenths of Bels. A logarithmic ratio unit, used to express gain or loss of power or voltage, either in a single device or a complete equipment. Double the power = a gain of 3dB.

DC

Direct Current, or current which always flows in the same direction.

Differentiation

Degradation of waveform (q.v.) when the time constant (q.v.) circuit is much shorter than the cycle time of the waveform. Causes e.g. sharpening or spiking of a square waveform.

Earth

(see Ground)

Equalisation

(sometimes Compensation). Non-linear circuit to correct the response (q.v.) when a device or signal is also non-linear. A disc, for example, has a progressive bass cut in its recording characteristic, and this must be corrected by a bass-boosting equaliser on playback. There are certain agreed international tape equalising standards. See Linear.

Frequency

The number of times every second that an alternating current or sound repeats itself. The higher the frequency the higher the pitch. Expressed in Hertz (Hz) or Kilohertz (KHz).

Gain

see dB.

Ground

(in UK often Earth). The neutral return rail of most electronic circuits, used as a reference and regarded as 0 volts. The metal parts of the VCS3, the screens of cables etc, are all connected to Ground, which may or may not be actually connected to the Earth via the house wiring system.

Harmonic

(sometimes Partial or Overtone). The natural series resulting from any object vibrating at a definite main frequency (the Fundamental). The harmonics occur at 2, 3, 4 etc times the fundamental frequency. Their presence or absence and their relative amplitudes determine the timbre of a note, and certain waveforms have known harmonic contents, a useful fact for the composer.

Hz

see Frequency

Impedance (Z)

The total apparent resistance (q.v.) of a circuit at a given frequency. In fact the vector sum of the resistances and reactances (q.v.) in the circuit. Expressed in ohmns.

Inductance (-tor, -tive) (L)

The magnetic effect of a current. Acts oppositely from capacitance (q.v.) and when capacitive and inductive reactance (q.v.) are equal the result is resonance, resonant LC circuits being used in the design of some oscillators and filters. Devices like tape heads depend on inductance for their operation. Expressed in Henrys. The electrical equivalent of mass.

Level

Sometimes synonymous with amplitude (q.v.) but by no means always. In practical audio terms, higher level means louder. Sometimes used, however, to mean same as flat, e.g. a flat response or level response (q.v.) meaning the same at all frequencies.

Line

(mostly UK in this sense). In an audio system, the main signal output(s) from the mixing system, distributed to amplifiers, tape recorders etc. as required. It is normally at a standard agreed level and at 600 ohms impedance (q.v.).

Linear

Not curved, having a straight line characteristic, but not necessary level. A straight sided ramp waveform shows a linear *increment of voltage. An ideal flat response (q.v.) is linear, but most are non-linear.*

Modulation

One signal modified at the frequency of another, and vice versa, normally in one parameter (q.v.), such as frequency or amplitude, but sometimes more than one. The term is sometimes used to describe controlled panning from one speaker to another (spatial mod), and phase (q.v.) modulation can also occur. A perfect ring modulator is a pure multiplier, the result being the instantaneous product of the two input voltages. When the equation is worked out for two sinusoidal (q.v.) inputs, only two frequencies remain, the sum and the difference of the inputs.

Ohms

see Impedance, Reactance, Resistance

Oscillator

A circuit which can only ring or resonate at one frequency, with a regenerative amplifier to keep the oscillations continuously going. In a voltage controlled oscillator, the effective resonant frequency of the circuit is altered by applying a voltage to it.

Parameter

Any characteristic of a device whose alteration will affect the performance of that device. The most relevant parameters of e.g. an oscillator are the frequency, the waveform (q.v.) and the level, but a complete list would include everything else about it, such as circuit details, power supply, physical layout, which must all be known to describe the oscillator exactly.

p-p

(Peak-to-peak). The voltage found by measuring a waveform (q.v.) vertically from the highest positive peak to the lowest negative peak. Twice the peak voltage, and more than twice the r.m.s. (q.v.) voltage.

Phase

The time relationship of two alternating currents. Two waveforms that start their cycle simultaneously are in phase *(but will only remain so if the frequencies are the same). Exactly opposed waveforms are* 180 degrees out of phase, *and all lags or leads of one wave over the other are similarly measured as an angular difference.*

Potentiometer

Originally what it says - a device for measuring potential or voltage. Now normally any three-terminal variable resistor - i.e. the two ends of the resistive track plus the wiper which slides along it.

Reactance (X)

The effective resistive effect of a capacitor (q.v.) or inductor (q.v.). Not in fact the same as resistance, because frequency dependent and phase shifting, but measured in ohms as if it were resistance. See Impedance.

Resistance (tor, -tive) (R)

The electrical equivalent of friction. Expressed in ohms or Kilohms, but unlike capacitance and inductance not frequency dependent. The relationship between resistance, voltage, (q.v.) and current (q.v.) in a circuit is governed by Ohm's Law, which states that I (Current) = V (Voltage)/R (resistance). This simple formula (in its three forms) can be used to make many deductions from a specification, and Impedance (Z) may be substituted for R in the expression where applicable. Units are Amps, Volts and Ohms, so allowance must be made for e.g. milliAmps or Kilohms.

Response

The output level of a device compared with its input at all frequencies. If the output/input ratio remains constant the response is flat or level. A part of the spectrum may be specified, as high frequency response.

r.m.s.

Root Mean Square. A method of rating an AC voltage to indicate its power capabilities at a given current. 230 VAC r.m.s. will do the same amount of work as 230 VDC, although its actual voltage is almost never 230V (four times per cycle instantaneously). Arrived at by squaring samples of the instantaneous voltage and taking the square root of this. For example, mains at 230 VAC r.m.s. has a peak voltage of 325, or 1.414 x the r.m.s. value. Peak to peak voltage is twice this amount, or 650.

Sine

(Sinewave, Sinusoidal, Sinus). The shape of a waveform containing one frequency only, or simple harmonic motion in the case of mechanical movement. The graphical representation of the sine of an angle through 360 degrees. The output of an ideal alternator, an ideal oscillator or an ideal tuning fork. Any waveform, however complex, can in theory be reduced to a collection of sinewaves of different frequencies, amplitudes and phase.

Time Constant

When a capacitance (q.v.) is charged through a resistance (q.v.), the product of their values (CR) gives the time taken for the capacitor to reach 63.2% of its final charge. Important information in many circuits. See Differentiation.

Tolerance

The design limits of a device or circuit. A resistor with 10% tolerance may be as much as 10% higher or lower than its nominal value. The closer the tolerance specified for a component or system, the more costly it usually is.

Voltage

Electrical pressure. Before current can flow in a circuit a potential difference must exist, and the current flows in an attempt to equalise this difference. For the relationship between voltage, current and resistance, see Resistance.

Waveform

The shape of an alternating current, usually described as it looks when graphically represented or displayed on an oscilloscope. Thus ramp, square, sinusoidal waveform. Any waveform can be analysed to yield its harmonic (q.v.) content.