Studio synthesis

Brian Hodgson

To fully realise a synthesiser's capabilities, it is a distinct advantage to appreciate what happens 'backstage'.

ELECTROPHON MUSIC LIMITED is a small studio in Soho, London, specialising in the composition and realisation of electronic music for ballet, film, television and advertising. The standard studio facilities are fairly unexceptional, all available money having been channelled into the development of our synthesiser.
It was decided that Electrophon would not purchase a large complex synthesiser like the Moog, ARP or *Synthi 100*. This decision was taken for two reasons: firstly, I felt that a modular approach would make it possible to keep abreast of the rapidly changing technology; and secondly, I wished to keep close contact with equipment designers so that we would end up with a range of modules that did what we wanted, and not what a design engineer thought we would want. In this we have been moderately successful in spite of the fact that money supply usually lags a year or so behind the birth of any idea.

All of the non-standard equipment was designed to my specification, formerly by Rodgers Studio Equipment and latterly by Ken Gale, one of the most truly creative engineers I have ever encountered.

Every synthesiser can be broken down into four main parts: *generation*, *modification*, *interconnection* and *control*. I shall attempt to stick to those headings in describing our studio synthesiser.

Generation

A bank of voltage-controlled oscillators provides both audio and control signals in three waveforms—sine, triangular and square. The oscillators are extremely stable and can track over five octaves around the manually set centre frequency. The manual controls are as follows:

Coarse frequency. This is switchable in octave steps from 27.5 to 3250 Hz in the audio mode, and from 1 cycle per 16 seconds (0.0625 Hz) to 4 Hz in the control mode. Of course, voltage control extends this range considerably.

Fine frequency. A 10-turn potentiometer allows fine control for lining up the centre frequency. Tuning and tracking are checked against a switchable frequency standard using a scope to provide Lissajous figures.

Function switch. This selects waveform and 'audio' or 'control' mode.

Even harmonic. A potentiometer allows progressive addition of even harmonics to the sine and triangular waves, and alters the mark/space ratio of the square wave.

Output gain. A centre-zero potentiometer controls output level of both audio and control signals, and allows easy inversion of control voltage outputs.

Voltage control is available for frequency and squarewave mark/space ratio. A fixed level pulse output provides a signal for synchronising other oscillators.

Of all our modules this is the least satisfactory; a new design is being worked on to provide better tracking over a wide range, and voltage control of all wave forms to give us the opportunity of creating even more complex timbral changes.

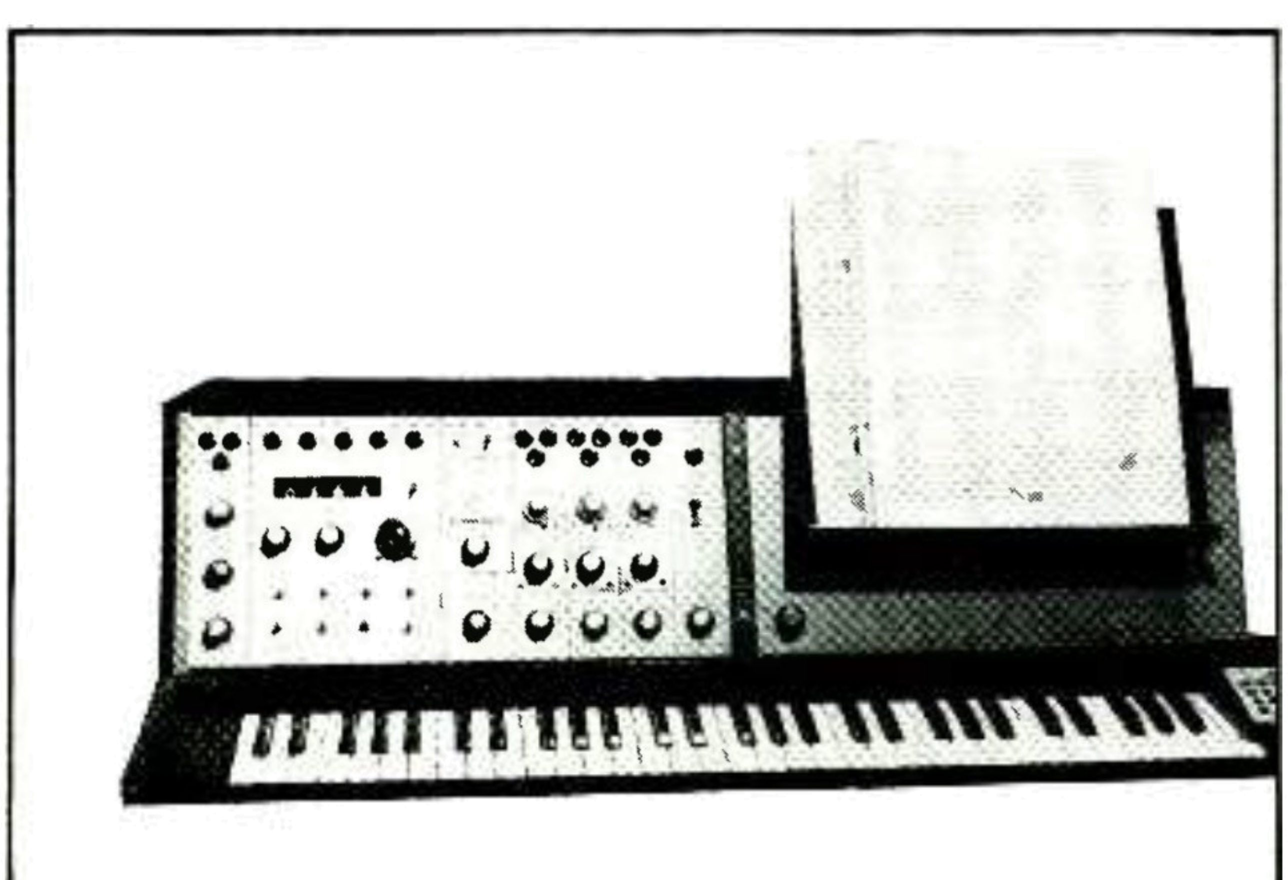
Single pulses can be derived from envelope-shaper trigger outputs, and white noise is provided from a *Synthi A* that we also use for control voltage generation.

Modification

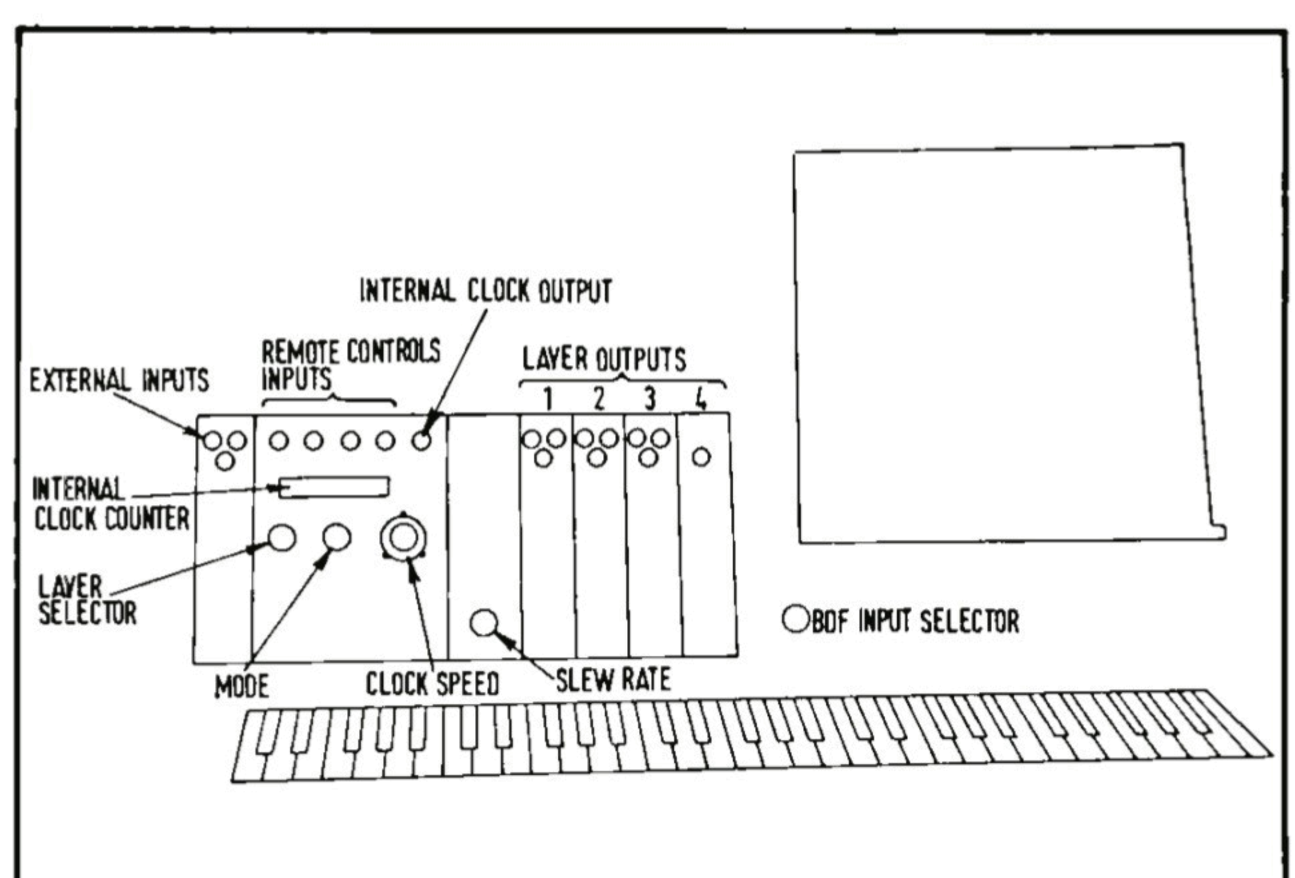
Filters. A bank of multimode filters provides all forms of filtering. Manual controls are: input gain or attenuation; mode, switchable between lowpass, highpass, band reject and bandpass; centre frequency from 30 to 20 kHz; lowpass 'Q', which alters the resonance of the filter up to the point of oscillation; bandwidth; and output level. A meter gives indication of overload within the filter.

Voltage control is available for centre frequency, bandwidth and lowpass 'Q'. This is a very successful module both for treatment of electronic sounds, where very complex changes can be produced within a single note, and in modifying pre-recorded tapes. The combination of slowly sweeping bands of sound against the original gives very complex but controllable types of phasing effects.

EMS Sequencer 256 keyboard and control.



Schematic of units in adjoining photograph.



Modulation. A multiplier gives modulation of both audio and control voltages; its ability to handle dc provides an excellent voltage-controlled amplifier for control voltages. By using envelope control voltages, one is then able to make controlled alteration to vibrato frequency and/or depth within each note. Conventional ring modulation is available from the *Synthi A.* Voltage-controlled amplifiers give level control of the signals, and a voltage-controlled panning amplifier provides stereo displacement.

Envelope Shaping. These modules are twin-channel devices with full adsr characteristic. Manual controls are featured for the time parameters: attack, decay, sustain, release, with delay controllable from 0-8s. A 4-position function switch controls: free run, by which on receiving a key voltage the shaper will attack, decay and hold in sustain until the voltage is removed, whereupon it will continue to cycle according to the settings of the time controls. (A further key voltage will hold the shaper in the sustain mode.); multicycle, which on receiving a key voltage will continue to cycle according to the time settings until the voltage is removed; one cycle, which will perform one cycle regardless of the duration of the key voltage, must be removed and re-applied to initiate a further cycle; and key on, whereby the key voltage initiates the attack and decay modes of the shaper, and will hold it in the sustain mode for as long as the voltage is present, after which the shaper will complete its cycle.

When the shaper is keyed, a dc voltage at control voltage output-level one will rise during the attack mode to the set level, hold through the decay and sustain modes, and die away during the release mode. This voltage is invertible. Control voltage output-level two (also invertible) is modified by the control voltage function, which has four settings: compound, the voltage rising during the attack mode, falling during the decay mode, holding at a voltage proportional to the sustain level during that mode, and dying away during the release mode; accent, the voltage rising during the attack mode and dying away during the decay; delay, the voltage rising during the delay mode and collapsing in all other modes; and finally sustain, the voltage rising immediately the attack mode is completed and dying away during the sustain mode.

This multiplicity of control voltages means that each aspect of the sound, its timbre, vibrato, level, harmonic content, and even stereo positioning, can be altered during the period of a single note.

The sustain level control presets the level at which the sound will hold after decaying from its attack peak. The trigger output selector controls the trigger voltage pulse produced by each shaper, which can key any other envelope shaper. This voltage can be produced at the following points: end attack, begin sustain, end decay, end sustain, end release and end delay.

The second channel of the envelope shaper, known as signal 2 function, can be switched to follow the full cycle, or the attack and decay modes only. This facility is especially useful for making the 'head' of a sound more interesting, for example, adding filtered 'chiffs' of white noise or tuned interval 'pings'.

Voltage control is available over all time parameters and the shapers can be triggered directly, ie at the start of the attack mode or via the delay mode control. This manual trigger facilitates multiple attacks when using several shapers from a single key voltage. In combination with the trigger outputs, the delay trigger capability enables the shapers to be used in a chain as a simple sequencer.

Interconnection

We decided to employ the system used by EMS: that is a matrix board with connecting pins, since this avoids the messiness of dozens of patchcords. It also gives maximum flexibility of interconnection and, just as important, reconnection. (I tend to rewire the boards frequently as ideas change and new 'toys' arrive.) There are two 60 x 60 matrix boards—one for audio and the other for control voltages. Several lines interconnect the two, since the distinction between audio and control is often blurred. Unfortunately, we are fast outgrowing both boards so that certain inputs have had to be rerouted via selector switches. With such a large selection of interconnections possible, logging a sound for future use became a problem until a simple schematic of the synthesiser was devised which seems to function adequately. Human error is more of a problem, because here too 'Sod's Law' operates and it is always the vital control path that isn't filled in.

Control

Keyboard control is provided by the excellent EMS *Sequencer 256* keyboard. This can be used as a straight 5-octave keyboard or as a memory bank, storing up to 256 events in any distribution over three layers of memory.

Each event consists of a key voltage, which will operate the envelope shapers, and two other voltages. The first of these is usually programmed from the keyboard and used to define pitch, while the second voltage, which may be re-written independently of the first, may be programmed by a keyboard, joystick, foot pedal or any other voltage required. It can also be used to define pitch, level, filtering, slew-rate, time parameters, and even harmonic or stereo positioning, depending on the requirement of the composer.

Although the keyboard is monophonic, a chord of up to six notes can be produced by programming them into the memory bank very closely together—for example, two or three clock pulses apart—and retrieving them whilst running the clock at 200 Hz. This is achieved by using the 'stop at event end' key and the 'start forward' button, which is 'played' in sync with the rest of the piece. The internal clock allows the memory to be run at any speed, controlling either complex fast musical figures or a long complicated effects sequence (such as the half-hour programme entitled 'Journey through Space' we did for The Geological Museum last year).

Any event can be altered and reprogrammed using electronic editing, and the sequencer can run in sync with a pre-recorded tape, by putting a pulse on one of the tracks. Pre-recorded pulses are of great use in the production of music for film, especially for complicated animation sequences. The pulse is usually set to a multiple of 24, or in the case of television 25 Hz, and the entire composition programmed to achieve sync to a fraction of a frame.

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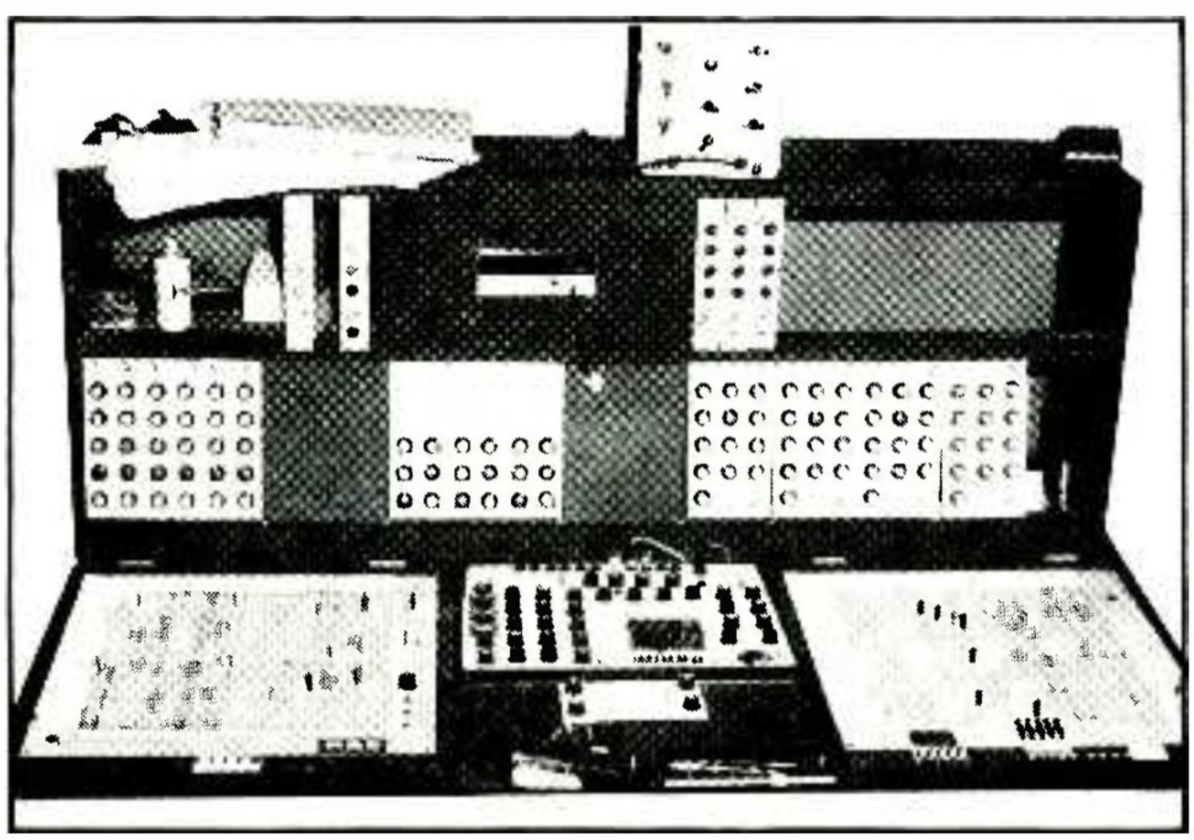
Control console at Electrophon.



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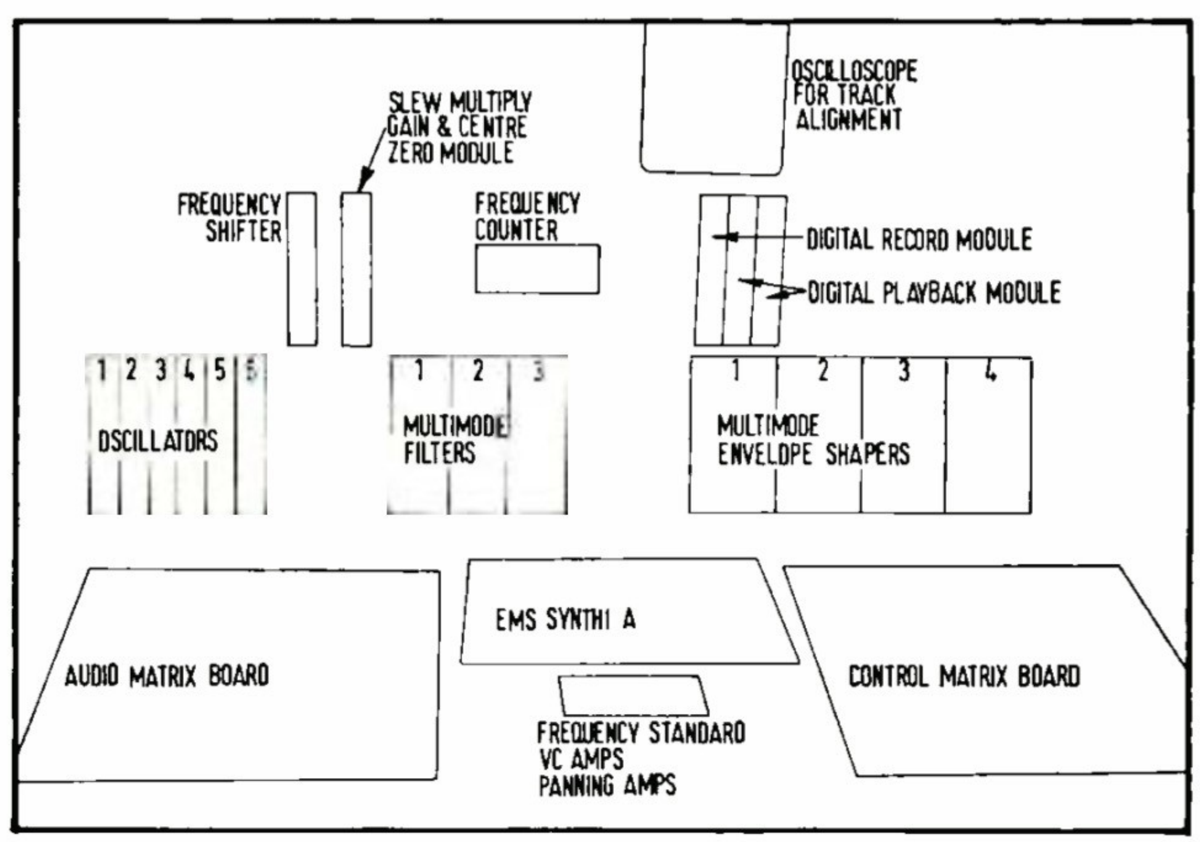

STUDIO SYNTHESIS

Indeed for a recent cinema commercial the picture was still being animated as we were producing the music, armed only with the story board and a frame count. Music and picture met for the first time on the morning of the dub, when to the editor's amazement and everyone else's relief the 39 sync points in 60 seconds matched absolutely.

The sequencer 'reset', 'stop' and 'start' pulses can also be pre-recorded and used to operate complicated pre-programmed sequences. The results are recorded either as a completed sound or a digital code, using Ken Gale's digital coding/decoding modules. At a later date the code can be used either to reprogram the memory bank, or to operate the synthesiser directly from tape. The advantage of storing the control voltages rather than the sound is that relative pitch, timbre and even speed can be altered later without destroying the original information, which may have taken hours to program or perform correctly.

The digital recording/playback modules are a new idea of Ken Gale's. They will work from any synthesiser keyboard, and will operate any make of synthesiser. The recording module will accept two voltages and 'two keys' simultaneously but independently from the keyboard, and transform them into a single digital code. This code can be recorded on one track of any studio-quality tape machine. When the tape is played back the playback module separates the voltages and keys, which can then be used to operate the synthesiser directly or even reprogram the *Sequencer 256*. Thus a complicated program or performance can be stored for future use in a voltage form, which means that relative pitch, sound quality or any other parameter can still be altered without affecting the original information. The record modules, like the sequencer inputs, are set to accept voltage changes that are the equivalent of a semitone, but both the digital and sequencer playback modules can be decoded to produce any tuning interval, for example, $\frac{1}{4}$ or even $\frac{1}{8}$ tones.

The modules we have at present only deal with two voltages and keys, but an updated version will encode four voltages and four keys on a single track. For example, four players on four separate small synthesisers could play together and the information would be stored on one track. They could repeat the process on each track of a 24-track recorder until 96 separate musical lines have been stored, each one still capable of having its sound altered at any time. Then armed with a 24-track tape, 24 decoders and 96 synthesisers the whole piece could be performed 'live' (or perhaps 'undead' is a better word) in a concert hall. The sight of 96 synthesisers playing 'untouched by human hand' would provide an awesome, if somewhat expensive, spectacle. That would of course be taking things to extreme, but the technology is no problem at all.

Our equipment has been designed to give us maximum control over each sound produced, and this to our mind is the most important requirement. A musical instrument produces a sound that contains an incredible rate of change of information; to produce interesting synthesiser sounds we must be able to produce a similar rate of change, otherwise the ear becomes bored very quickly. If you listen to any of the very early synthesiser albums, the first track sounds fascinating, the second interesting, but the third rather dull—the hand is tempted to reach for a glass of that brand of beer which refreshes the parts of the ear that the sound cannot reach. Fortunately, all of us working with synthesisers realised the problem very quickly and work never stops on developing new ways of increasing control over the sound. Nowadays the production of a 'synthesiser album' requires an incredible amount of painstaking work on the structure of each sound.

We have just completed our fourth album for Polydor. It is called *New Atlantis*, and is based on the essay by Francis Bacon. The initial spur came from reading the following quotation:

'*Wee have also Sound-houses, wher 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 Great and Deepe; Likeweise Great Sounds, Extenuate and Sharpe; Wee make diverse Tremblings and Warblings of Sounds, which in their Originall are Entire. Wee represent and imitate all Articulate Sounds and Letters and the Voices and Notes of Beasts and Birds. Wee have certaine Helps, which sett to the Eare doe further the Hearing greatly. Wee have also diverse Strange and Artiviciall Eccho's, Reflecting the Voice many times, and as it were Tossing it: And some that give back the Voice Lowder than it came, some Shriller, and some Deeper; Yea some rendring the Voice, Differing in the Letters or Articulate Sound, from that they receyve, Wee have also meanes to convey Sounds in Trunks and Pipes, in strange Lines, and Distances.*'

Strange lines and distances . . . which must be the first description ever of an electronic music recording studio—but this was written in 1624! We then read the whole of this astonishingly prophetic essay and became fascinated. Bacon foresaw aircraft, submarines, lasers, genetic engineering and many other fantasies that have since become fact, although in his fabled society the scientists produced these wonders to further civilisation, not to destroy it.

Each track on the album is inspired by a passage in the book, my partner John Lewis composing the music to reflect the spirit rather than the literal meaning of the passages. The album took about six months to plan and record using a 16-track Studer and our new digital modules. Before even a note of music could be recorded, a pulse track was laid down so that we could run programmed lines in sync, or have access to a running memory for adjustment to the sound after the 'performance take' had been chosen. This was not just a question of recording a static pulse, as in some passages complex tempo changes were required. For example, in the passage entitled *Salomons House*, a pre-programmed phase is repeated at many times its normal speed and gradually slows down until it reaches the correct tempo for the rest of the track. In another piece, *Echoes III*, the tempo slowly increases during the second section, levelling out at a much faster rate for the third section. In these cases the external clock rates had to be carefully calculated in terms of pitch and slew rate so that these changes would occur over the correct period, starting and stopping at exactly the right point in the music. This information was programmed into the memory bank and used to control the sync-pulse oscillator. Once the pulse was recorded, metronome tracks could be programmed that would change as and where necessary, and used or ignored according to the musical requirements. Only then could work start on recording the music. Techniques varied according to the character of each passage and to describe them all would take an eternity, but several of them may prove of interest.

As far as possible 'live performance' takes were made. This improves the flow of the final result; programming each section gives an incredible degree of accuracy but has a rather inhuman feel to it. As a result this technique was only used for sections that required a high degree of exactness, or were physically impossible to perform. The digital modules proved of great value here because John Lewis was able to perform a line and then decide whether his performance was good enough before we settled down to work on the right sound. Often several lines were recorded digitally and played back together so that the interaction of the sounds could be judged, up to four lines being possible without two decoders.

After the sounds had been chosen they were then recorded conventionally, either as a mono mix or a stereo pair, and the digital tracks erased to be used again. In the chorale section of *Salomons House* John played a monophonic line, controlling pitch from the keyboard and 'swell' from a foot pedal. The voltages were recorded digitally, and a sound quality chosen and recorded. Simple retuning of the oscillator centre frequencies allowed us to record parallel reinforcements of the original line without John having to repeat his performance. When the accompaniment lines were required, the original digital information was used to control only the 'swell', with John playing in the new pitch information. In this way the 9-line passage was built up in a remarkably short time using the accuracy of the digital techniques and the 'live' quality of John's performance.

Where complicated links were required, they were first programmed into the sequencer and then added to the tape using the sync pulse and pre-recorded start pulses. The system of recording voltage pulses digitally first was also used in *The Pool*, where the 6-note phrase of the accompaniment moves in stereo patterns. In this case three digital lines covered six stereo positions and six changing pitches, the final result being recorded as a stereo pair, releasing the tracks for re-recording. The bass line was a mixture of several sounds in different stereo positioning, each with a different character in terms of attack, quality, change, etc, but produced by the same

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single line of digital information. Once the synthesiser lines were almost complete we laid down the percussion. Having been in constant contact during the period of electronic recording, Tony McVey (our guest percussionist) offered advice, suggestions and countless comforting cups of coffee. We don't like using synthesiser percussion except as an added effect, because we all consider that the sound and feel of a live percussionist adds so much to the excitement and impetus of the music. It was also felt that the synthesiser should be used to extend the range of available sound rather than reproduce conventional instruments.

What of the future? John Lewis has always wanted to give live synthesiser performances of his music, but both of us feel that to make a proper impact it would really have to be truly live, and not just a couple of lines played over the outputs of a bank of tape machines. The main problem up to now has been that synthesisers tend to be monophonic, or at best duophonic. Moog have recently produced the *Poly-Moog*, and we have been working for some time with Ken Gale to produce our own polyphonic keyboard. This would use an entirely different system but be capable of interfacing with any range of synthesiser equipment. The basic keyboard will be 2-manual with a 10-note polyphonic capability, but with an inbuilt capacity for '15-note polyphonic' on each keyboard—for duets or people who use their elbows! Selector switches will allow access to pre-programmable sound in any order or combination. We are now building the prototype, and hope to have it ready in time for exhaustive tests before a live commitment we have in Florida next January.

If any other of my colleagues using synthesisers want to contact me I would be delighted to hear their news, reviews and headaches. Synthesisers have come a long way since Robert Moog's original papers on voltage control in 1958, but we are still only on the threshold—so much work has still to be done. If we try to communicate more, maybe we can help each other along the way. ■

agony

... and if I ever find the guy who chopped up my Steinway ...