tanrak

digital sampler-delay

tantek

R5	560K	
R6,9,24,98	1K	4 off
R7,20,21,32,33,56,22	22K	7 off
R8,34,50	150K	3 off
R10	110	
R11,12,14	1K8	3 off
R13	3K9	
R15,28	4K7	2 off
R23	2K2	
R25	8K2 1% Metal film	
R26,30,31	3K0 1% Metal film	3 off
R27	2K2 1% Metal film	
R29	3K3	
R35	27K	
R36,60	5K6	2 off
R37	180K	
R38,39	82	2 off
R40,57	56K	2 off
R41	39K	
R42	33K	
R43	120K	
R44,45 ,65	47K	3 off
R46	47	
R47,51,58,62	270	4 off
R48	15K	
R53,54	220K	2 off
R59	1K2	
R64	100K	
VR1,4,7,11	10K vertical preset	4 off
VR2	10K multiturn preset	
VR10	100K lin PC pot	
VR8	470K log PC pot	
VR9	47K vertical preset	
VR13	10K lin PC pot	
VR14	47K log PC pot	

Capacitors

C1,2,10,11,17,24,26,28,29	100nF polyester	9 off
C3,9,21	1000pF polystyrene	3 off
C6,7,14,19	470nF polyester	4 off
C8,59,60	4.7nF polyester	3 off
C12,31	10nF polyester	2 off
C13,15,58	33pF ceramic	3 off
C16,25,27	47uF 25v radial electrolytic	3 off
C18	2.2uF 63v electrolytic	
C20	47nF polyester	
C22	100pF ceramic	
C23	10uF 25v electrolytic	

Semiconductors

D1-8,10-14	1N4148	13 off
D9	5mm red LED	
D15	3mm green LED	
TR1	BC182	
TR2,3	BC212	2 off
IC1,9,16,18	LF353	4 off
IC2	CA3096	
IC3,8	LM311	2 off
IC6	DG308	
IC7	LF398	
IC10	4066	
IC11	AM6072	
IC17	MF4-50	
IC19	LM13600	
IC20	LM324	
IC21	MF6-50	
IC22	79L05	

Miscellaneous

JK1-5	¼" PC jack socket	5 off
SW1,2	PC push switch SPDT	2 off
	Buss connector	
	Track pin	100 off
	8-way DIL socket	8 off
	14-way DIL socket	3 off
	16-way DIL socket	3 off
	18-way DIL socket	
	Pot PC bracket	4 off
	PCB (with printed overlay)	

Logic Board

Resistors-1/3W 5% CF

R16	150K	
R17,18	10K	2 off
R19,94	2K2	2 off
R66	47K	
R67,69,70,72,74,82,83,90	22K	8 off
R71,73	1M	2 off
R75,76,77,78	270	4 off
R79,85	4K7	2 off
R80	68K	
R81,87	18K	2 off
R84,55	100K	2 off
R86	33K	
R88	10M	
R89	390	
R91	2R7	
VR3	100K lin PC pot	
VR5,6	47K log PC pot	2 off
VR12	2M2 log PC pot	

Capacitors

C30,36,37,38,42,47	470pF ceramic	6 off
C33	10nF polyester	
C32,34,35,48,50,51,5	100nF polyester	7 off
C39,40,43,44	100pF ceramic	4 off
C41,45,46,4	33pF ceramic	4 off
C49	47uF 25V radial elecrolytic	

Semiconductors

D15	3mm green LED	
D17	Tricolour LED	
TR4-7	BC182	4 off
IC4	4046	
IC12	MC14559	
IC13	74HC244	
IC15	74HC86	
IC23,24,25,28,14	4013	5 off
IC26,27	4528	2 off
IC29,31	74HC00	2 off
IC30,32,33,34	74HC02	4 off
IC35	74HC04	
IC36,5	74HC4024	2 off
IC37	4520	
IC38,39	4516	2 off
IC40	40103	
IC41	7805	
X1	6MHz ceramic resonator	

Miscellaneous

SW3,4	PC push switch SPDT	2 off
	Buss connector	
	Track pin	100 off
	14-way DIL socket	15 off
	16-way DIL socket	8 off
	20-way DIL socket	
	Pot PC bracket	4 off
	34-way male connector	
	PCB (with printed overlay)	

Memory Board

Resistors – ⅓W 5% CF

R92,93	1K	2 off
R95	10K	

Capacitors

C52	100nF polyester	
C53,54,55,56	100pF ceramic	4 off
C57	47μF 25v radial elect.	

Semiconductors

IC42,43	74HC244	2 off
IC44,45	74HC74	2 off
IC46	74HC00	
IC47,48	TMS4464	2 off
IC49	74HC02	

Miscellaneous

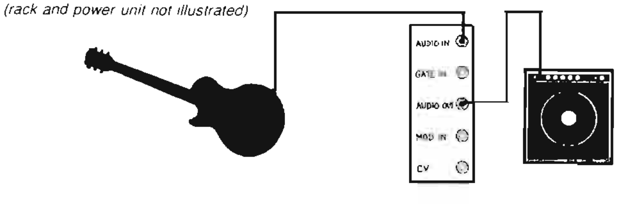
	34-way female connector	
	14-way DIL socket	4 off
	18-way DIL socket	2 off
	20-way DIL socket	2 off
	PCB	

digital sampler-delay

This could be the digital audio device you have been waiting for. It is both a delay and a sampler for serious creativity. This double-width TANRAK module has every facility you could wish for, and an uncompromised specification too.

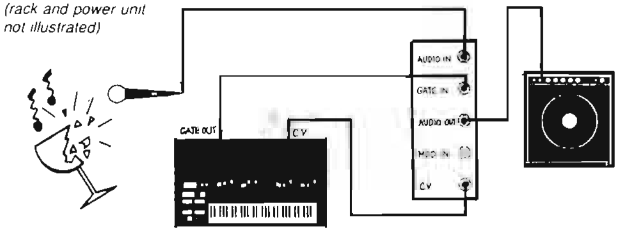
_ BUT IS IT FLEXIBLE?

(rack and power unit not illustrated)

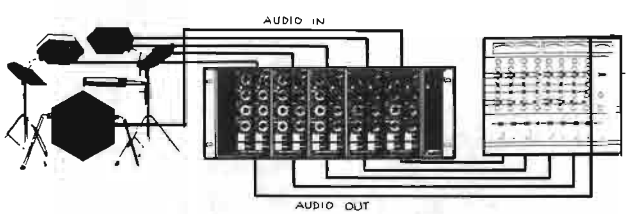


It's a Digital Delay Line creating delays of up to 8 seconds

(rack and power unit not illustrated)

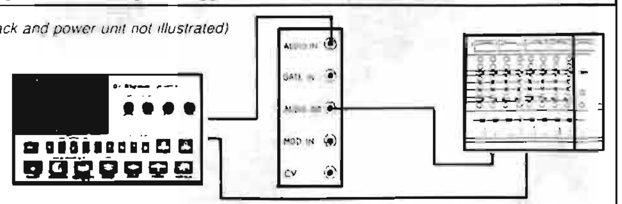


It's a keyboard controlled sound sampler with full editing



It's a digital drum voice module with velocity sensitive dynamics – use up to 5 in a single TANRAK along with trigger pads for a full percussion kit

(rack and power unit not illustrated)



It's a drum machine voice module with variable tuning – triggered from an instrument output of a drum machine for real drum sounds, 'orchestral stabs' or vocal 'scratching' etc.

DELAY OR SAMPLE MODE is selected by a single button to alter the personality of the device entirely. In the delay mode, the LENGTH control sets the delay time, the TRIM control being used to further 'stretch' the delay time for very long echo effects.

A FULL 15KHz BANDWIDTH for delays or samples up to 1.4 seconds long is the normal figure: although the delay sample time can be extended using the TRIM control to 8 seconds at reduced bandwidth. The trim control affects both delay time and bandwidth parameters simultaneously and automatically, with no switches hidden around the back!

A DYNAMIC RANGE OF 87dB at the input is achieved by means of a built-in limiter which further extends the already ample 72dB dynamic range of the digital converter.

EASY SAMPLING; just hit the SAMPLE button and make the source sound – playback is then ready automatically. It could hardly be simpler!

AUTO OR MANUAL SAMPLE RECORD TRIGGERING can be selected so that recording commences either at the command of an external trigger source such as a synthesiser: at the depression of the TRIG button, or when the input signal level exceeds a preset threshold (screwdriver adjustable from the front).

FULL SAMPLE EDITING facilities are available for trimming both ends of a sample using the START and LENGTH controls. And, since the editing controls also determine the memory position for recording a sample, several samples can be stored in different parts of the memory for playing individually, or in a chain.

SINGLE OR LOOP PLAY of samples can be selected at the touch of a button. Non-looping samples will play once between the edit points each time the unit is gated (triggered). If a second trigger arrives before the sample has finished playing, playback will commence again at the START point. When looping, each new gate signal causes the 'zero memory' attack period to be played so as to retain the natural attack of the sampled sound. Looping then continues around the edited points. No waiting around for the sound to arrive either, since the response time is just 50 millionths of a second!

WIDE RANGE CV PITCH CONTROL input allows an analogue synthesiser to 'play' samples both above and below the sampled pitch over a 6 octave range. Tuning can be adjusted by means of the 'TRIM' control, which is also used to set the sample record time between 1.4 and 8 seconds.

'VELOCITY SENSITIVE' DYNAMICS allow percussion samples to be played dynamically by using an electronic drum pad (Trigger pad).

SAMPLE OVERDUBBING is available at the touch of a button, allowing complex overlays or chords to be built up.

THE DECAY CONTROL can be used either to 'sweeten' the end of an abruptly finishing sample, or to impress envelope characteristics on a looping sample since looping continues during decay. Decay is initiated at the end of the Gate period. With little decay selected, the sample can be made to terminate immediately on key release.

REAL-TIME CONTROLS put YOU in charge of all the operating parameters of the device.

'BURIED TECHNOLOGY' keeps the panel controls simple, easy to use and powerful. Innocent, the unit may look from the front: but inside, absolutely bristling with the latest devices.

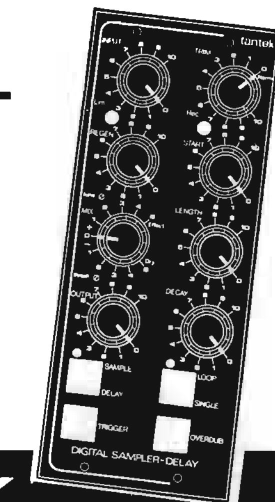
BUILT-IN EXPANDABILITY ensures that technological progress will not march relentlessly past TANTEK's Digital Sampler-Delay. An expansion socket allows the memory size to be increased using an expansion module.

MODULATION of a sampled sound for vibrato effects, or in the delay mode for modulated time domain effects is made possible by the inclusion of a modulation input which can be driven by a modulation oscillator for instance. This allows high quality chorus or animated ADT to be produced.

SOCKETS for connection with the outside world are INPUT, OUTPUT, GATE IN (positive going) CV IN (1v octave) and MODULATION IN.

SPECIFICATION	
DRY SIGNAL	
Frequency response (–3dB)	12Hz to 35KHz
Output noise (unity gain)	–98dBm(A)
TREATED SIGNAL	
Frequency response (Trim clockwise) (–3dB)	15Hz to 15KHz
Delay/sample time (Trim clockwise)	15mS to 1.4 sec
Delay/sample time (Trim anticlockwise)	90mS to 8 sec
CV keyboard range	6 octaves
Input dynamic range	87dB
Converter dynamic range	72dB
Headroom above limiting	20dB
Response time (gate to sound)	0.05mSec
Current consumption	90mA

The TANRAK Digital Sampler-Delay from tantek puts powerful creativity in YOUR hands.



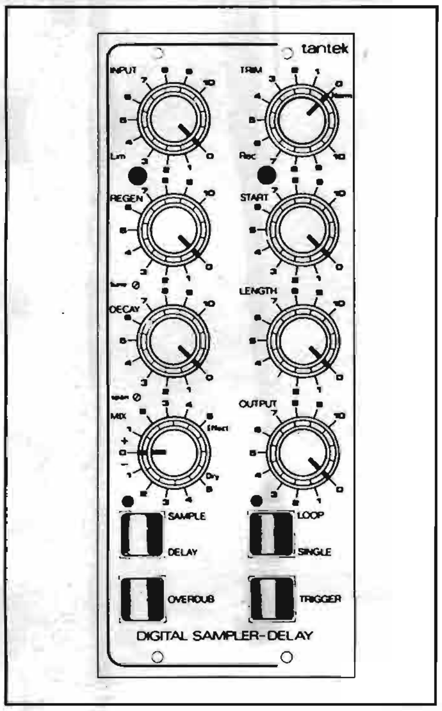
tantek

DON'T COMPROMISE – CHOOSE TANRAK

TANTEK
Business & Technology Centre,
Bessemer Drive, Stevenage, Herts. SG1 2DX

Digital Sampler/ Delay 1

The Modular Effects Rack takes a significant step forward this month with it's first digital module; the Digital Sampler/Delay. Paul Williams presents this high spec device which incorporates full sample editing, CV pitch control, 'velocity sensitive' dynamics and a bandwidth/delay product up to 15kHz and 1.4 seconds.



It is not without good reason that the Modular Effects Rack has been lacking a digital module for so long. Having evolved it over a long period since the sampler has 'come of age' gives it the advantage of incorporating all the facilities the modern musician looks for today, together with an uncompromised specification. An affordable price has only recently been made possible as new silicon devices have become available.

The fact that both samplers and DDLs are based around fairly large memories into which digitised audio is stored and manipulated makes it almost illogical to produce one of these devices without incorporating the facilities for the other. This module is thus a dual personality digital unit which can function as either a delay line or a sampler at the touch of a button. In the delay mode, full bandwidth delays of up to 1.4 seconds can be produced, and this figure is stretchable (at the expense of bandwidth) up to 8 seconds.

In the sample mode, a sample is recorded into the memory where it's 'frozen'. The sample can then be played back either at the command of an external trigger pulse, by pushing the 'Trig' switch, or continuously by setting the 'Loop' switch. The pitch of the sample can be varied using the 'Trim' control, or played via the CV input by an analogue synthesiser. The start and finish of the sample can be edited using the 'Start' and 'Length' controls so that any pre-sample silence or after-sample noise can be eliminated. since the editing controls also effect the recording start and finish points, several samples can be stored for playing separately, or in a chain.

A 'Decay' control helps to tidy up any samples that end abruptly, as well as allowing looped samples to be envelope shaped. Complex overlays can be built up using the 'Overdub' switch for 'bouncing' samples. Of great advantage when playing percussive samples is the velocity sensitivity, which when used with an electronic drum trigger pad, makes the amplitude of the regurgitated sound proportional to the violence with which the pad is assaulted!

The sound quality? Well, a companding Digital-to-Analogue Converter (DAC) with pre- and de-emphasis, and a built-in limiter ensure a healthy dynamic range which, together with the 15kHz bandwidth add up to a sound quality significantly better than most cassette players.

To allow for the inevitable future progress in silicon memory, the plug-in memory card occupies an expansion socket, which would allow either a larger memory card to be plugged in, or connection to an expansion module, depending on user demand and future advances in technology.

For those handy with a soldering iron, the kit is an excellent way of stretching your tight budget a little further, but it's not recommended for the complete novice. If you have no experience of constructing electronic kits, then this is probably not the one to start with. However, this need not stop you from owning a module though, since they are also readily available in an assembled form too.

Architecture

Figure 1 shows the overall structure of the device. Noteworthy are the tunable low pass filters which at the input provide anti-aliasing with the clock frequency, and reconstitution and clock removal at the output. By using switched capacitor filters, the cutoff frequency of both is maintained at a set multiple of the main clock. The bandwidth of the system is thus optimised for whatever clock frequency is in use. The aforementioned clock is actually a Voltage Controlled Oscillator (VCO) which, apart from manual Tune and Trim controls, can be

Figure 1

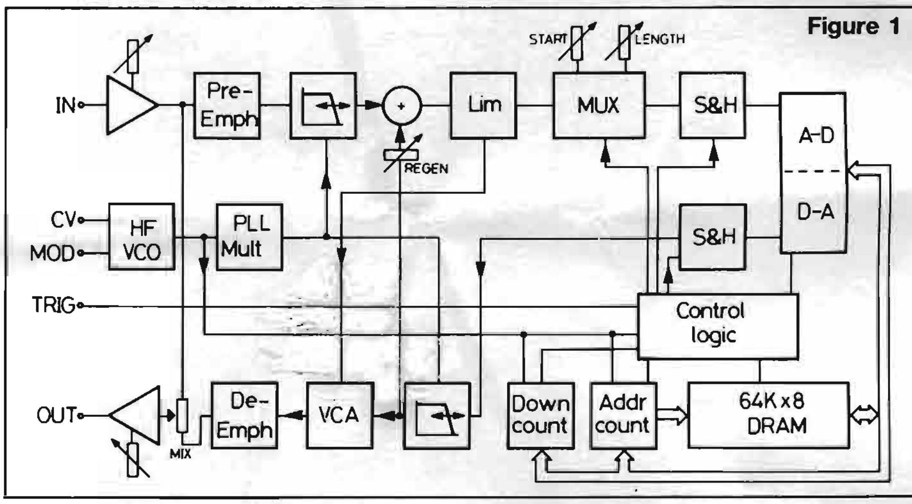
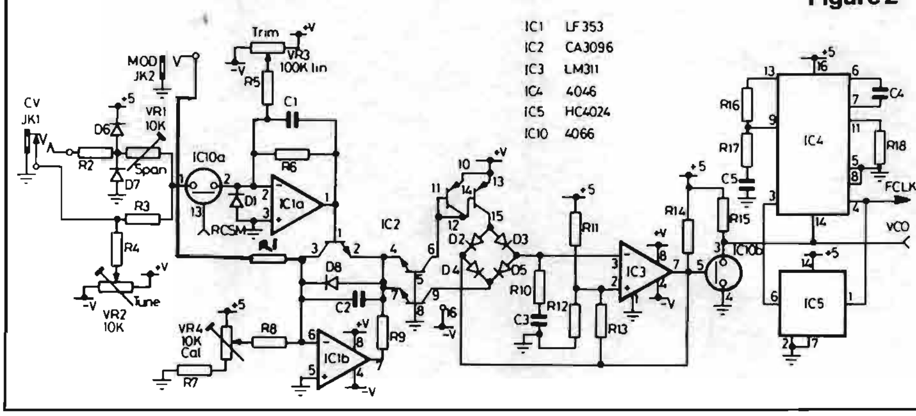


Figure 2



controlled by a voltage applied at the CV input. The filter clock frequency is maintained at sixteen times the VCO rate by means of a Phase Locked Loop (PLL) frequency multiplier.

Before the input signal can be converted into it's digital form, it must be broken up into 'samples' of instantaneous voltage (not to be confused with the more general term 'sampling' of musical sounds). Each sample is, for a short time, held constant in a Sample and Hold (S & H) amplifier. The Analogue-to-Digital (AD) converter sub-system then goes about it's task in producing a digital word corresponding to each sample.

The all-important memory has sufficient capacity for over 65,000 samples. The operational location, or address is determined by the address counter which is incremented by the VCO. Each sample is thus allocated a different address. In the delay mode, the existing contents of each new address are read and converted back to analogue by the Digital to Analogue (DA) converter. A further S & H is used to fill in the gaps between the read samples, after which the signal is passed via the reconstitution filter to the output. The Regen control allows some of the delayed signal to be re-cycled for echo repeat effects. In the sample mode, writing new data and subsequently reading is performed as two separate tasks. Once a memory full of data has been written, it will remain in the memory indefinitely while the power remains on. The trigger input controls the onset of sample playing, unless the loop mode is selected where playing is continuous. The Voltage Controlled Amplifier (VCA) allows the amplitude of the sample to be controlled by a trigger source applied to the audio input.

The editing controls both produce an analogue voltage corresponding to the editing point chosen which is given digital words during a special conversion cycle. The same converter is used for editing and audio by the use of an analogue multiplexer (MUX). The editing data is used to preset two counters. The address counter is preset with the start point address, and the down counter with the length data. Each run through the sample therefore, initiates at the start address, and terminates when the number set in the down counter has been exceeded. In the delay mode, the start address is always zero, leaving the length control to set the amount of memory being used, and thus the delay time.

An inbuilt limiter permanently safeguards any signal being recorded from becoming clipped in an overload situation. This applies both to the delay mode, and to the sample record mode. An indicator associated with the limiter thus indicates when the limiter is in operation, *not* when you have just ruined your irretrievable sample with clipping distortion.

VCO

Of paramount importance to the musicality of the device is the VCO circuit, which is shown in Figure 2. IC1a sums the various control and offset voltages applied to it's input. The voltages from the CV input socket and Tune preset are only allowed to effect the VCO when a sample is being played and a jack inserted in the CV socket, as determined by the CMOS switch IC10a. The summed voltage at pin 1 of IC1a is passed to the logarithmic amplifier formed around IC1b and IC2. This uses the well known logarithmic relationship between transistor Vbe and Ic in the devices Q2 & 3 in IC2. The summed voltage drives the emitters of Q2 & 3 negatively, with a temperature compensating offset produced by Q1. The current thus produced is drawn via D4 & 5 and, due to current mirroring in Q4 & 5, is also pushed into D2 & 3.

The oscillator itself is based around the fast comparator IC3 whose output at pin 7 'steers' current of the appropriate polarity from the diode ring into the integrating capacitor, C3. When the trigger threshold is reached, the comparator toggles whereapon R13 shifts the threshold level and the diode ring reverses the current polarity, charging C3 towards the other threshold and so on. Since IC3 is the only active device concerned with oscillation, operation extends up to about 1.5MHz, although normal CV control would be within the range 10kHz to 200kHz, centering around a nominal figure of 46.875kHz. IC10b merely buffers the VCO output so that external noise cannot effect stability.

IC4 is a complete monolithic Phase Locked Loop (PLL) system whose inbuilt VCO is controlled so as to keep the input on pin 3 at the same frequency as that on pin 4, the main VCO. Since IC5 divides the PLL VCO by 16, the output on IC4 pin 4 is 16 times the main VCO frequency. This new clock frequency is used to control the turnover frequency of the switched capacitor filters, of which more next month.

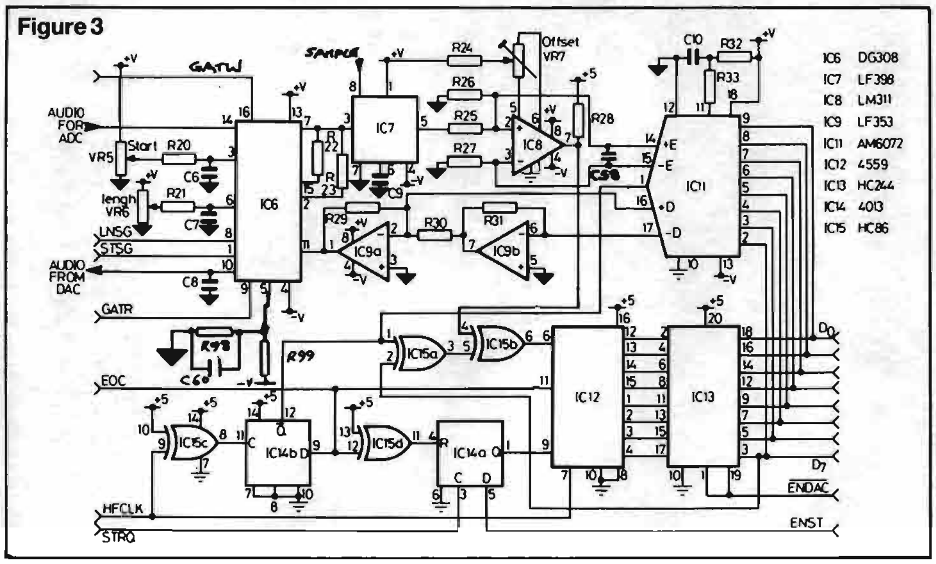
Converter

Figure 3 shows what is probably the most critical part of the system: the converter section. Since Digital-to-Analogue Converters are relatively expensive devices, only one is used here: IC11, which is multiplexed for audio D-A, and along with the Successive Approximation Register (SAR) IC12, for A-D conversion of audio, Start pot and Length pot voltages. The DAC used has non-linear converting characteristics which gives the effect of an inbuilt compander to enhance the dynamic range.

D-A conversion is a simple matter where data put on the data buss D0-D7 by the memory enters the DAC, causing it's current decode outputs on pins 16 and 17 to produce a corresponding voltage at the output of the current/voltage converter IC9. The voltage for each successive digital word is sampled onto, and held by C8 by one element of the analogue switch IC6. This signal is by now recognisable audio which is then passed to the audio section for reconstitution filtering.

A-D conversion is a rather more complicated affair. Firstly, the analogue switch, IC6 selects whether the conversion is for audio, Start pot or Length pot, gating the appropriate signal into the S & H amplifier, IC7. Once conversion is initiated, IC7 freezes the voltage presented to it at that instant, and keeps is constant throughout conversion. Under the control of IC14 & 15 and a crystal derived high frequency clock, the SAR IC12 builds up a digital word bit by bit. The data produced by the SAR is put on the data bus by the tri-state buffer IC13 and simultaneously collected by the DAC IC11. At each step, the comparator IC8 signals to the SAR whether the next bit needs to be high or low to get the analogue equivalent of the assembled word closer to the sampled voltage. When the digital word is fully assembled, the SAR notifies the control logic with the EOC signal so that the data is written into memory.

Figure 3



Digital Sampler/Delay 2

Having generally described this exciting new module for the Modular Effects Rack, last month, Paul Williams now discloses more of the intricacies of the design, and takes a preliminary look at construction.

The audio signal presented to the unit remains in it's analogue form within the confines of the signal conditioning circuit shown in Figure 4. The incoming signal is buffered with variable gain by IC16a and presented to IC16b via the pre-emphasis network C12, R35 and R36. The signal then passes via IC17: a fourth order low pass switched capacitor filter which removes any ultrasonic component in the signal which might otherwise lead to aliasing with the VCO frequency. IC18a then mixes some treated (delayed) signal to the, as yet 'dry' signal under the control of VR10 (Regen). It's the output of this stage which is fed to the converter electronics for analogue-to-digital conversion. The output level of this stage is watched over by IC20a and IC19c and d forming a full wave rectifier. Any excessively high levels cause TR2 to conduct, pushing current into the control pin of IC19a, an Operational Transconductance Amplifier (OTA). The increased transconductance of this device then increases the negative feedback around IC16b, reducing its gain and thus bringing the output of IC18a down to an acceptable level. This limiting action is indicated by the LED D9, driven by TR1.

Treated signal from the DAC is received by IC21a, a sixth order low pass switched capacitor filter. Both the filters are under control of the VCO multiplier via the FCLK signal so that the maximum bandwidth possible is available for any given delay or sample time. The OTA IC19b, along with IC20b forms a Current Controlled Amplifier (CCA) with a de-emphasis network C20, R59 and R60 to negate the effects of the pre-emphasis network, and at the same time reduce quantisation noise. The CCA is used to produce velocity sensitivity as we will see shortly. VR13 allows the mixture between the dry and treated signals to be varied, the composite signal being buffered by ICb with variable gain, and passed to the output.

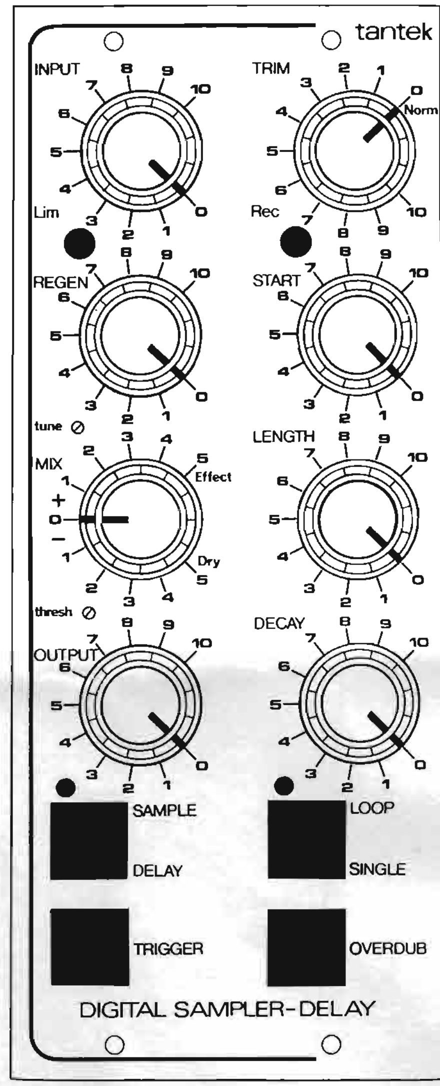
The output of IC16a is precision rectified by IC21b, producing a DC voltage on C18 which is proportional to the input signal amplitude. When the signal ceases, the control voltage leaks away at a rate determined by the Decay control, VR12. This control voltage is only allowed to pass to the voltage-to-current converter, IC20c and d and TR3 when the unit is playing a sample, as dictated by the analogue switch, IC10c. IC10d otherwise forces the V-C converter to produce maximum current (ie. no attenuation). IC21c acts as a comparator to detect when the input signal level is above a preset triggering level, variable by means of the Threshold preset: VR11. A plug inserted into the gate socket overrides the precision rectifier, giving control of triggering over to the gate socket.

Control Logic

The operation of the module is timed and controlled by the logic shown in Figure 5 (next month). The Sample/Delay mode is set by the flip-flop IC24a, this being toggled by SW2 and indicated by the LED D15. This flip-flop controls the manner in which many sequences take place within the logic. The other flip-flops under manual control are IC25a for looping and IC23a for overdubbing. the latter can only be set when the sample mode is selected. Timing is predominantly controlled by the VCO, although the 6MHz clock and associated divider IC36 provide the other timings necessary for conversion and memory control. The address counters, IC37, 38 and 39 select the current memory address, this being updated at every VCO cycle. At each new address the VCO fires the monostables IC26a and b. Since the memory devices used are dynamic, they have to be refreshed every couple of milliseconds in order to keep the data intack. So that refreshing and read/write cycles do not clash, IC26b produces a BUSY signal a couple of microseconds before reading to indicate to the memory control when refreshing must be suspended. The process is similar when a write operation takes place at the end of a conversion, when the EOC signal fires the monostables IC27a and b.

When the preselected length of memory comes to an end, IC40 produces an END signal which sets the HOLD flip-flop IC24b. During the Hold period, conversions take place to digitise the Start and Length pots. The first conversion during Hold is for the Length pot, during which time the LNSG flip-flop IC28a is set. The digital word generated from the conversion is used to preset the down counter, IC40. The STAFF flip-flop IC28b then initialises the Start pot digitisation, the result of which presets the counters IC38 and 39. After this, the Hold mode is terminated, allowing address counting to continue from the start address loaded into IC38 and 39. Since IC40 counts down along with the address counters, it produces an END signal when the equivalent of the length pot digitisation has been counted through.

In the delay mode, no start conversion is produced since address counting



always starts at zero. In the sample non-loop mode, the Hold state remains true until IC28a is triggered either by the TRIG signal becoming true with SAMOD true, or vice versa. The flip-flop IC25b causes sample playing to start at zero memory address when the unit is triggered and looping is selected. Subsequent loops start at the edited start point.

When the mode is toggled from Delay to Sample, The REC flip-flop, IC23b becomes set. Once IC28a becomes triggered, recording will start at the edited start point and continue for the selected length, when the END signal resets IC23b so that the recorded sample can be played. During recording, reading of the data already in memory is normally inhibited so that the setting of the Regen control does not effect the recording. Pressing SW4, the overdub switch sets IC23a along with IC23b for overdub recording. The only difference now is that the sample already in memory is played

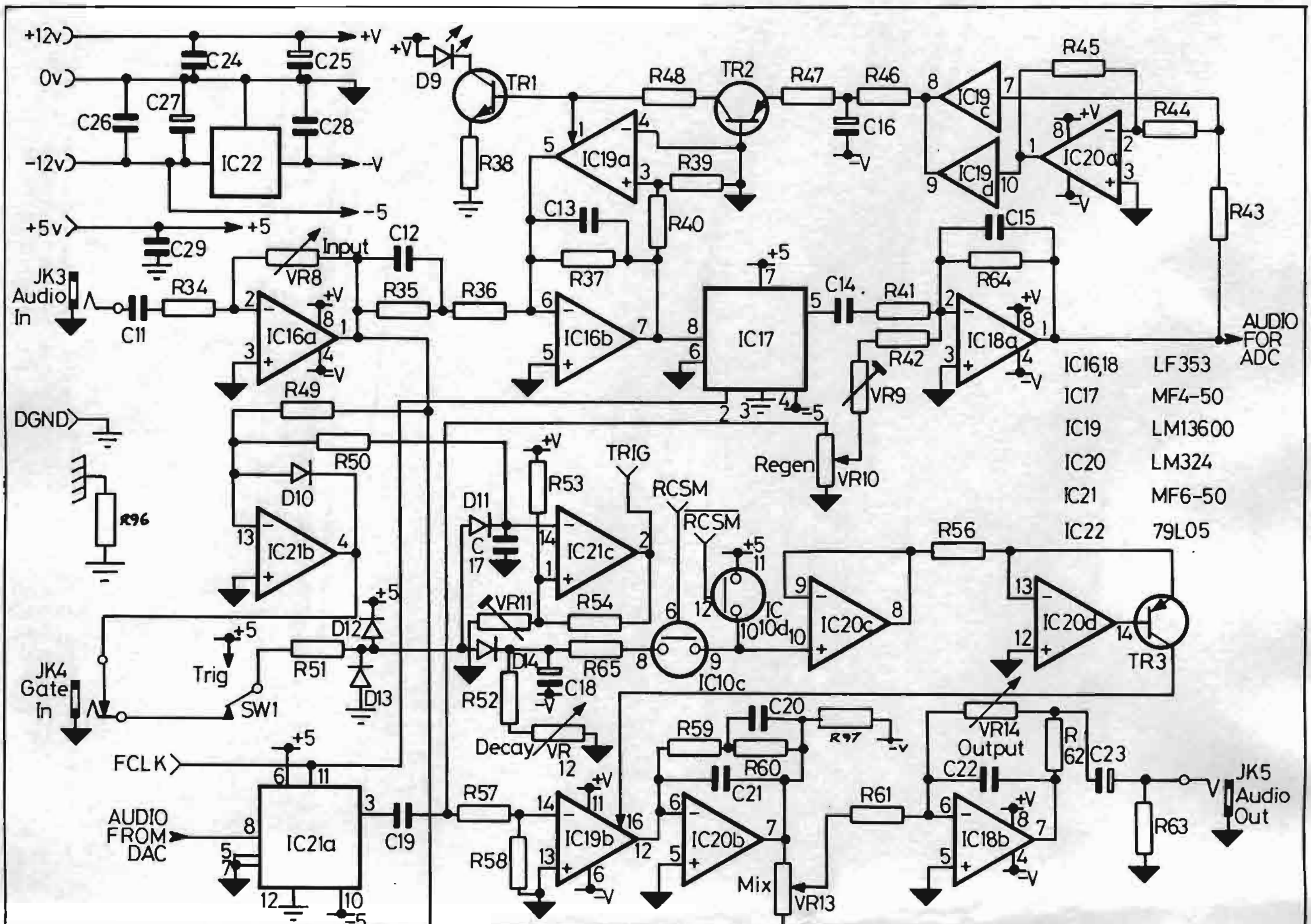


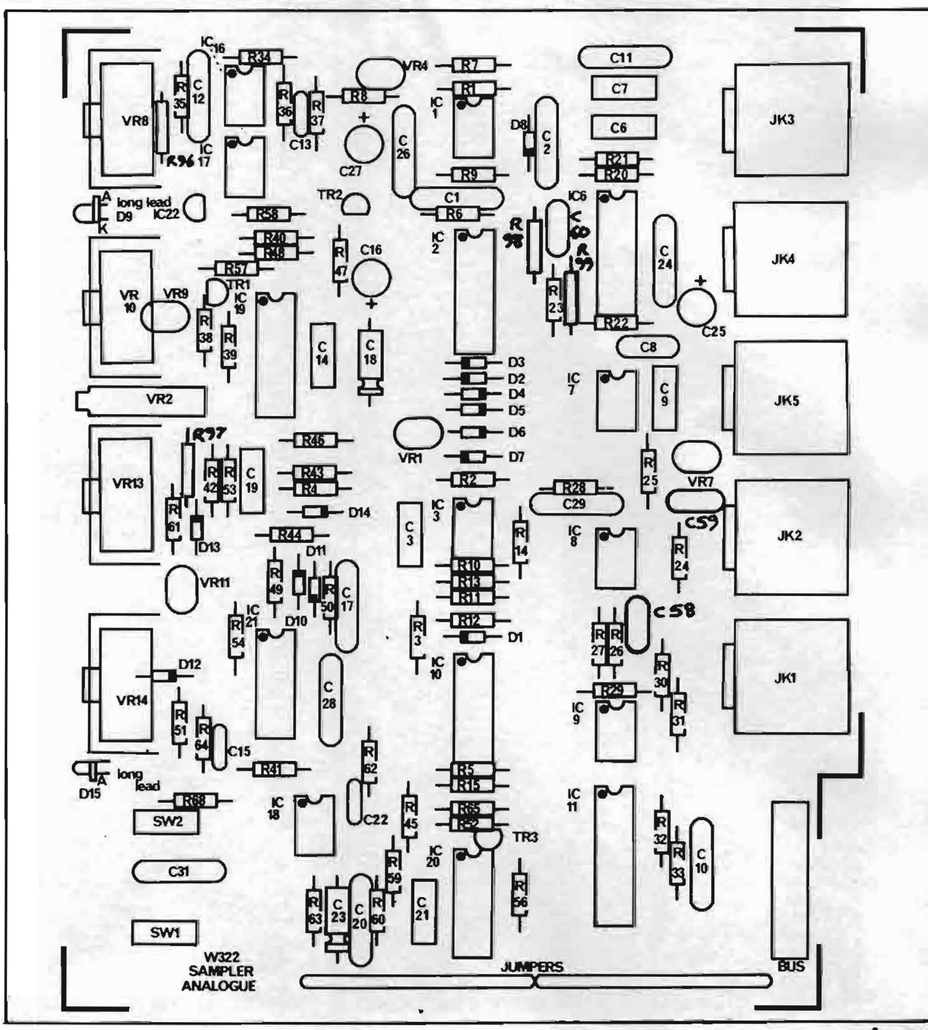
Figure 4. Signal Conditioning Circuit.

back simultaneously so that it may be mixed with the newly recorded sample using the Regen control.

Memory

Since the latest 256K DRAMs are used, the entire 64K byte of storage is contained within just two chips. The devices used, IC47 and 48, as shown in Figure 6 (to be published next month) have internal refresh address counters, so they simply have to be 'reminded' to refresh a new location at least every 16 microseconds. The RFRQ signal is raised by the control logic every ten microseconds or so, causing the memory strobing flip-flops IC44 and 45 to produce the appropriate refresh strobing sequence on each RFRQ pulse. The strobe pulses are timed using the master 6MHz clock.

Just before the memory is required to perform a read or write operation, the BUSY signal from the control logic prevents any new refreshes being initiated, whilst allowing time for any current refresh to be completed. To reduce the number of pins on the memory devices, the 16 address lines have to be multiplexed onto the eight address pins by means of IC42&43. The RWP signal causes a memory strobe sequence to be produced which strobes firstly the row address via IC42, then the column address via IC43 into the memory. The memory will then either read data onto, or write data from the

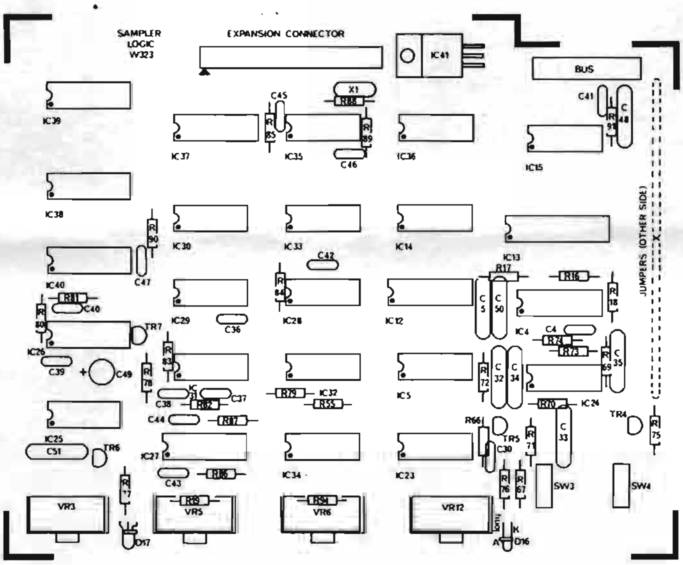


Digital Sampler/Delay 3

In this concluding article, Paul Williams rounds off construction, goes through the setting up procedure and describes how to get the best from this versatile module.

Having detailed the construction of the analogue board last month, it only remains to assemble the digital (logic) board, and the memory board.

Assembly of the these is eased by the use of double sided, plated-through PCBs. For those not familiar with plated-through PCBs, every hole is plated on it's bore so that a circuit is maintained from one side of the board to the other at every pad position. This means that track pins are eliminated, and soldering is not required on the component side of the PCBs.



Logic Board

Assembly of the logic board may therefore begin with the lowest profile components such as resistors, followed by IC sockets, then capacitors, transistors and the ceramic resonator, X1. Take care with the polarity of the electrolytic capacitor and the IC sockets, but leave the ICs themselves out until later. Insert according to the parts list, and the overlay printed on the PCB itself, a few at a time to prevent crowding, then solder and crop. Solder the buss connector and the 34-way male expansion connector in place, ensuring that they are pressed firmly down onto the PCB. The regulator, IC41 can now be positioned, forming its leads down so that when inserted into the PCB, the tag hole targets on the PCB hole. Using a short spacer over the regulator tag, fix in place using a self-tapping screw from the underside of the PCB. The regulator leads can now be soldered and cropped.

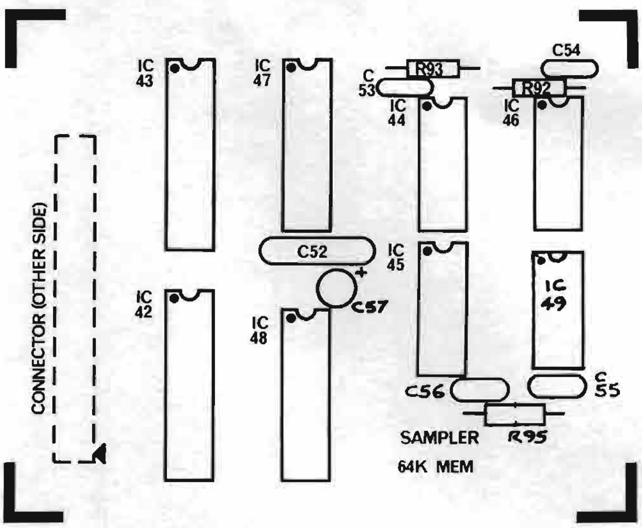
Trim each pot shaft to 8mm from the bush using a hacksaw, whilst holding the pot shaft in a vice, or just use a pair of cable cutters. Fit a PC bracket to each pot and locate into their respective PCB positions, but don't solder at this point. After determining the correct orientation of the LEDs, bend their leads down at right angles to suit the PCB holes, 4mm from the body of the large LED and 6mm from the body of the small one, and locate into the PCB without soldering. Place the shakeproof washers on the pots, then offer the front panel up, feeding the pot bushes and LED domes into the appropriate panel apertures on the right-hand (Trim, Start, Length & Decay control) side. The panel is then fixed in place by means of the pot nuts which should be fully tightened. The pots, brackets and LEDs can now be soldered having made sure that the pots are fully home, and that the panel is square to the PCB. Fit keytops nice and squarely to the two push switches and position them on the PCB. Temporarily tack one of the joints on each switch with solder and check the keytop positioning in the panel apertures. If it is not central, heat the tacked joint to re-position the switch. When all is well, solder all the switch joints.

Spend some time checking over the assembly very carefully since dry joints and solder splashes are all too common in such a dense assembly, even for the experienced constructor. When you're completely satisfied with the job, load the ICs into their sockets, being careful with orientation, and taking some anti-static precautions since most of the devices are static-sensitive. Keep the ICs in their conductive packaging until the last moment and, touching both the packaging and some of the PCB tracks, transfer the ICs from the packaging to the sockets, avoiding touching the pins if possible.

Now, taking the partly assembled and checked analogue board, load it's ICs into position, again taking static precautions as some of these are static-sensitive. Offer the analogue board up to the logic board so that the jumper cables can be fed into the logic board and soldered on the component side. Having done that, fold the analogue board down onto the work surface so that the pots and LEDs can be positioned on the analogue board as described above, folding the assembly back up and securing the pots to the panel before soldering. Fit the push switches as before, centralising them if necessary before soldering. An M3 screw then attaches the logic board to the free end of the long spacer on the analogue board.

Memory Board

Having completed the main boards, the tiny memory board will seem like childs play. Again the assembly procedure consists of soldering the resistors, then IC sockets and capacitors. This only leaves the 34-way female connector which is applied *to the underside of the PCB* and soldered on the component side. To keep things tidy, align the 'arrowed' Pin 1 of the connector with the PCB arrow. Again taking static



precautions, load the memory board ICs into position being careful with orientation.

After checking the assembly very carefully, offer the memory connector onto the expansion connector on the logic board, ensuring that the arrowed pins line-up, and secure with a self-tapping screw through the memory board into the free end of the short spacer on the logic board.

Finally, fit the knobs and caps so that the marker line of each covers the scale evenly, with equal 'dead-band' at each end.

Setting Up

No test equipment is needed to set the module up, just a sound source of some kind, a monitoring amplifier and a pair of ears!

Before applying power to the module, set the three lower top-access presets on the analogue board (all except the topmost one), and the front-access threshold preset centrally. The topmost preset should be set fully anti-clockwise. Remove the top dust cover from your sub-rack and tape a piece of card over the top of the power unit position to prevent electric shocks. Slide the module into a convenient position in the sub-rack and switch on the power. If the power unit indicator does not illuminate, or if it fades, this would indicate that there is a short-circuit fault and so the power should be switched off immediately. You then have to find a fault which thorough checking should have uncovered in the first place!

If all is well, set the Regen, Mix and Start controls fully anticlockwise and the Output, Trim, Length and Decay controls fully clockwise. Apply an instrument signal to the audio input (the top socket) and adjust the Input control so that the Lim indicator flashes on the loudest peaks. If the output socket (the third down) is now monitored, the signal should appear normally. Advancing the Mix control fully clockwise should reveal an identical, but delayed version of the input signal. Using a preset trimming tool, or a small screwdriver insulated over most of it's length with tape, slowly adjust the uppermost preset clockwise until the signal is heard to start to break-up. Back the preset off slightly to lose any hint of breaking up or crackling. The adjustment must be made slowly since the results are delayed by almost a second and a half.

Since the unit will have powered-up in the delay mode (with all LEDs extinguished), advancing the Regen control will cause echo repeats to be produced. The Length control will set the speed of repeating. With the Regen control fully clockwise adjust VR9, the preset just behind the Regen control so that the repeats sustain almost indefinitely, without building up into a 'mush'. This is best done at a moderately fast repeat rate.

Now set the Length control fully clockwise again, touch the Sample button and play your source instrument. The Sample LED will illuminate, and the Rec indicator will show yellow until the sound source triggers the unit, when red indication shows that a sample is being recorded. After a couple of seconds, the Rec indicator turns green, showing that the sample is ready to be played. The sample can then be heard either by playing the source instrument again, or by touching the Trigger button with no input signal present. Now touch the Loop button so that the associated LED lights. The sample will now repeat continuously, although it will fade unless the unit is kept triggered. The operation of the editing controls can now be proven. Select minimum Length so that the sample repeats rapidly, selecting a suitable Start position so that the repeats are distinct. Now adjust VR7, the preset just in front of the sockets so that the speed of repetition can be heard to alter. Find the band of adjustment within which the sample repeats the fastest, and leave VR7 set at a position at the centre of this band.

If an analogue synthesiser is available, the CV and Gate signals should be routed to the bottom, and second socket down respectively on the Sampler. The pitch of the sample should now be under control of the synthesiser. VR1 at the centre of the analogue board is used to calibrate the span of the CV control input so that octave changes do indeed produce one octave shifts in pitch. If the resultant change is less than one octave, adjust VR1 slightly clockwise and vice-versa. It may be necessary to trim the front access tune preset to scale the keyboard sensibly.

In Use

Once the ultimate position in the Sub-Rack has been established, the Self-adhesive socket label should be applied to the back of the Sub-Rack to cover the unused socket holes. It should be noted that since this module cannot be switched out directly (apart from by turning the Mix control anticlockwise), it doesn't form part of the rack's linking system. The link chain is however maintained around the module position, but without being processed by it.

The mode of the module is changed by touching the Sample/Delay button, above which is an indicator to show the mode selected. The unit will initially power-up in the delay mode.

The Input control is used to adjust the input signal level such that the Lim indicator glows on and off most of the time. If the indicator rarely lights, then the dynamic range of the unit is not being adequately utilised, but too much input drive will cause an almost continuous glow when the limiting action may become noticeable. This is not necessarily disastrous though since the limiter adequately protects against overload distortion, and the resulting compression may sometimes be desirable. The Output control allows for any necessary make-up gain or loss.

The modulation input socket can be used with a modulation oscillator module to achieve vibrato or, in the delay mode, animated ADT and chorus or modulated delay effects. In the sample mode, modulation is best used during sample playback, rather than during recording when the effect would be irreversable.

Delay Mode

When the Sample/Delay indicator is extinguished, the Delay mode is selected allowing ADT, echo and doubling effects to be produced. The Length control sets the delay time, the Trim control being used for fine adjustment. Bear in mind though, that full signal bandwidth is only achieved when the Trim control is fully clockwise. Sustain of echo repeats is determined by the Regen control. Almost infinite echo repeats should be easily attainable. The output can consist of all-dry signal, all-effect, or a mixture of the two as determined by the Mix control. If the effect is to be mixed at a desk, then an all-effect signal is obviously the order of the day.

The Decay and Start controls and the Overdub, Loop and Trigger switches have no effect in the Delay mode.

Sampling

The unit is armed for recording a sample as soon as the Sample mode is entered by touching the Sample/Delay button. This record standby mode is shown by the record status indicator turning yellow. As soon as the input signal amplitude is above a preset threshold level (or immediately on entering the sample mode if the signal is of sufficient amplitude), the record status indicator will turn red indicating that a recording is being made. The threshold level may be adjusted by means of the front panel accessible 'thresh' preset. Recording can also be initiated by applying a trigger pulse to the Gate input from a synthesiser for example; or if a dummy plug is inserted into the gate socket, the trigger button can be used, the dummy plug being necessary to prevent the input signal from auto-tripping.

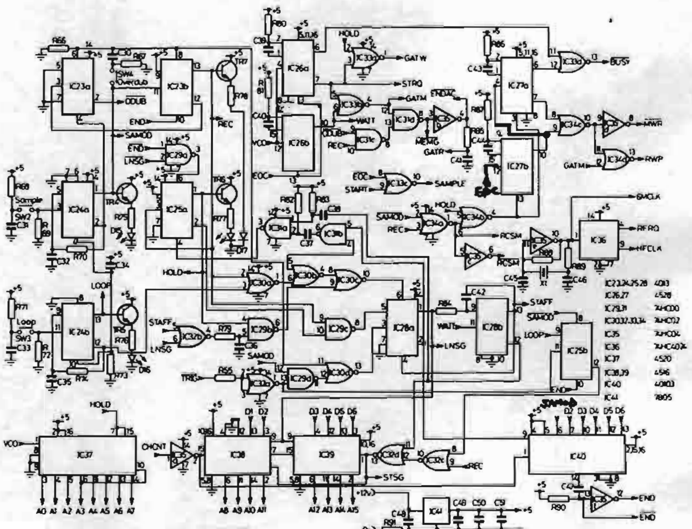


Figure 5. Sampler Logic.

When taking a sample, the Trim control would normally be set fully clockwise so as to achieve maximum bandwidth, although a lower setting could be used for very long samples. If the recording was unsuccessful, then touching the Sample/Delay button twice will once again arm the unit ready for recording. When a sample has been successfully recorded, the Sample/Delay button should be left well and truly alone otherwise the sample may be lost forever. So as to make all the memory available for sample storage, the Start control should normally be fully anticlockwise, and the Length control clockwise, which will result in a sampling time of between 1.4 and 8 seconds depending on the position of the trim control, although more on this later.

Playing the Sample

After recording a sample, the record status indicator will be green, showing that a sample is ready to be played. If the input signal is still present and there is no jack in the Gate socket, then the sample may begin to play immediately. This can be stopped by turning down the Input control. The sample can then be played either by touching the Trigger button, by applying a trigger pulse to the gate input, or by applying an electronic drum pad signal to the audio input. The latter will also allow the sample to be played dynamically, in which case the Input control is adjusted so that hard 'whacks' on the pad cause the Lim indicator to flash. The 'thresh' preset plays an important role here since a good dynamic range on the played drum sound will depend on very sensitive triggering. The sensitivity will be limited by the pickup of extraneous sounds or vibrations, especially those from another nearby pad. A drum machine with separate instrument outputs could also be fed to the audio input to trigger percussive samples, orchestral 'stabs' or at least n-n-nineteen other sounds.

Samples would normally be played with the mix control fully clockwise so that any input signal is not passed to the output. However, it may be that some 'live' sound could be usefully mixed with the playing sample, such as when an analogue synthesiser is used to control the sample pitch. By feeding the synthesiser's output into the sampler input, the two could be mixed. This is also true when a drum pad or drum machine is used to trigger a percussion sound, where some of the sound from the pad or machine could be mixed with the sample.

The pitch of the sample can be tuned using the Trim control, or played by an analogue synthesiser via the CV input. The front accessible Tune preset is used to scale the synthesiser keyboard. The controllable pitch extends both above and below the sampled pitch, even if the Trim control is hard over to either extreme. If your synthesiser is not calibrated to precisely one volt per octave, then the sampler response can be trimmed to suit using the screwdriver adjustable preset which is half way down, and at the centre of the left-hand circuit board. The top dust cover will have to be removed for this, although care should be taken to cover the power unit end with card to prevent electric shocks. The adjustment is best made with a proper trimming tool, although a small screwdriver insulated along most of its length with tape will suffice.

A gate connection will usually also be used when using synthesiser control.

Editing and Looping

The Start control can be advanced to skip over any unwanted sounds which were recorded just after the trigger point. The Length control then determines when the sample terminates. The Decay control comes in very useful for tidying up the end of an abruptly finishing sample.

Since this module boasts a generous sample time, many sounds can be recorded in their entirity, eliminating the need for looping. Some sounds are difficult or impossible to loop with undetectable edit points, although driving the limiter hard during recording may assist here by flattening out the dynamics to a certain extent. The golden rule is: only loop if you really need to.

When Looping is selected using the Loop button (as shown by the indicator above the button), sample playing commences immediately after a recording has been made, although the looping signal will fade away unless the unit is triggered as previously discussed. Looping is also useful for setting up the editing points, even if the sound is not going to be looped

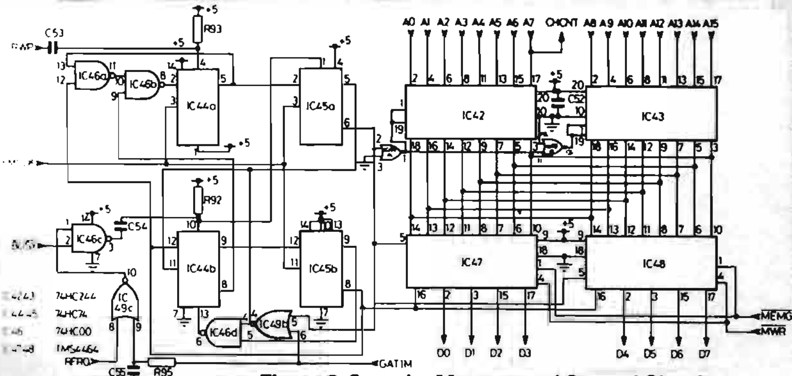


Figure 6. Sampler Memory and Control Circuit.

finally. If the Length control is turned anticlockwise, the Start control can be used to rapidly 'search' through the memory for suitable editing points. If the editing controls are set such that the length is greater than the memory left after the start position, then part of the sample at the beginning of the memory will be 'tacked' on to pad the play time out to the selected length.

A suitable start point for looping will usually be immediately after the initial attack period when the harmonics have started to settle down and the amplitude is reasonably stable. The loop length should be as long as possible to avoid 'burbling', although the limit will be determined by how quickly the sound changes its amplitude or harmonic content. Further adjustment of the editing controls may then be necessary to match the butting ends to eliminate clicks generated by steps. The Start control should be used for final trimming since it has slightly better resolution than the Length control.

Each time a looping sample is gated, a length of sample at zero memory will be played before the loop so that the initial attack of the recorded sound is preserved. When the synthesiser key is released, looping continues during decay, the rate of decay being controlled by the Decay control.

Overdubs

To avoid clicks being generated, the overdub button should only be pressed in the non-loop play standby mode (with the record status indicator green). Pressing the Overdub button will arm the unit for recording just as before, except that this time, the signal in memory can be preserved by advancing the Regen control. The mix between the new and old signals put into memory is controlled by Input and Regen respectively. Several overlays can be built-up in this manner. New recordings can be made to appear in a different key even if the recorded pitch is different by adjusting the Trim control. Alternatively, differently tuned instruments could be overdubbed at the correct pitch using the Trim control. Any plug in the CV input should be taken out when tuning overdubbed samples since the sampler does not respond to keyboard control during recording; only during playback.

Several samples can be stored simultaneously by using the editing controls to select different parts of memory. The position to be recorded into is set up on the editing controls whilst playing that position to see that it is vacant. Sounds could also be selectively deleted in this way. The Sample/Delay button must not be used since the memory will become corrupted. Instead the overdub button is used to set the record mode with the Regen control anti-clockwise. Different sounds could be chained together in this way for playing in a string or sequence.

Saving and Loading

To preserve a hard-won sample for future use, it can be saved on any analogue tape machine by playing the entire memory contents. The module should be in the non-loop mode with the Start control anti-clockwise and the Length control clockwise. Once the tape machine has been set to record, the Sampler can be triggered to blurt out its memory contents.

To subsequently load the sample back from tape, set the tape machine just before the sample, touch the sampler Sample button and start the tape playing. The auto record trip facility will then start the loading process when the sound begins. For samples which have a slow attack which adjustment of the threshold preset can still not adequately capture, you can either manually trigger loading just before the sound starts, using the tape counter as a guide or, if you can record in reverse by turning the tape over, record a short blip just after the reversed sound so that when playing the tape normally, the blip will appear just before the sample, and will thus trigger the sampler. The blip and any delay can then be edited out using the Start control.

Digital Sampler/Delay
Specification

Dry Signal	
Frequency response (−3dB)	12Hz to 35kHz
Output noise (unity gain)	−98dBm(A)
Treated Signal	
Frequency response (Trim clockwise)	15Hz to 15kHz (−3dB)
Delay/sample time (Trim clockwise)	15mS to 1.4 sec
Delay/sample time (Trim anticlockwise)	90mS to 8 sec
CV range	6 octaves
Input dynamic range	87dB
Converter dynamic range	72dB
Headroom above limiting	20dB

Digital Sampler/Delay Parts List

Main Assembly

Knob	8 off
Knob cap	8 off
Keytop	4 Off
Front panel (punched & printed)	
15-way jumper	2 off
Long spacer	
Short spacer	
M3 screw	2 off
Self tapping screw	2 off
Black M2,5 screw	4 off
Solder	
Backplane Label	

Analogue Board

Resistors – ⅓W 5% CF unless stated

R1,68	1M	2 off
R2	51K	
R3	330K	
R4, 96,52,61,63,97,99,49	10K	8 off