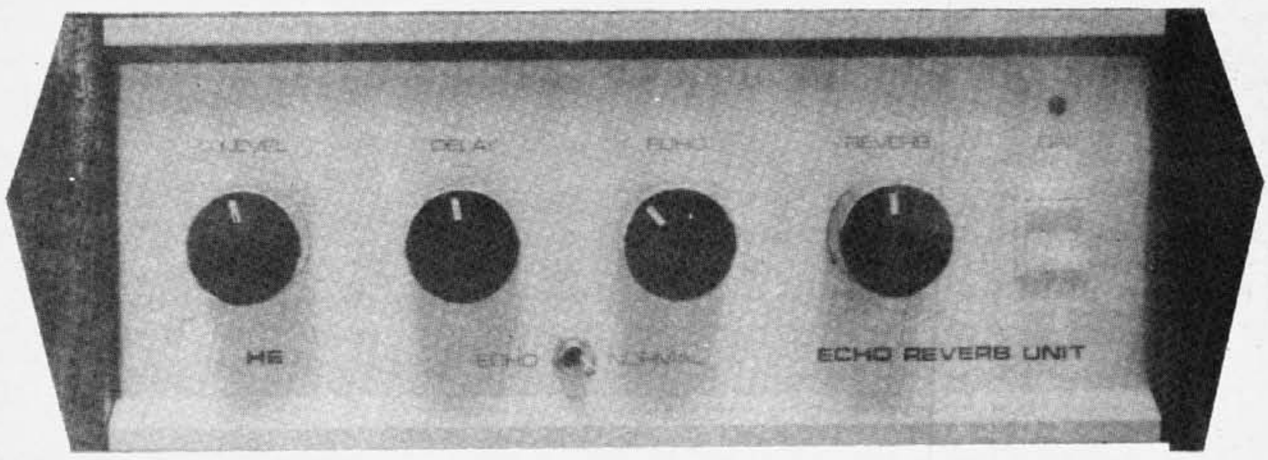
Echo-Reverb Unit



A superb project for creative musicians and audiophiles.

AN ECHO-REVERB unit is an accessory that can be added to virtually any existing audio or electronic music system and used to impart new life to existing sounds. In any audio system, the unit is simply interposed between the output of the preamp and the input of the main amplifier, so that the audio signals have to pass through the echo-reverb unit on their way to the main amplifier.

Some modestly-priced echo-reverb units use a crude mechanical spring-line to create the time-delayed echo-reverb and provide only a single, fixed delay time. Commercial (all electronic) echo-reverb units can be rather expensive, but provide fully-variable delay times. Many use clocked CCD (charge-coupled device) analogue ICs to implement the delays (see **How It Works**). The best commercial units use digital techniques to implement the delays, but these are extremely expensive!

We have used the analogue CCD technique to implement the time delays and the Echo-Reverb gives a performance that is at least as good as some commercial units using CCDs. The important thing about our unit, however, is that the design uses some clever techniques to reduce the hardware costs of its circuitry, while at the same time enhancing the overall performance and facilities.

Echo-Reverb

As the audio signals enter the echo-reverb unit they split into two paths which are re-united again near the output of the unit via a built-in-low-distortion audio mixer. One of these paths is a direct link from the input of the echo-reverb to the input of the mixer stage; the other path is via a variable signal-delay network. By varying the signal delay and then the mixture of direct and delayed signals, a variety of interesting effects can be obtained. Here are a few ideas to try:

(1). With equal levels of direct and delayed signals, and a few milliseconds of delay, a 'double-tracking' or 'mini-chorus' effect is obtained. This makes a single input sound like a pair of independent but time-synchronised outputs. Thus, a single violin can be made to sound like a duet and a duet is made to sound like a quartet.

(2) With a reduced level of delayed signal in the mix, and with a delay time of tens of milliseconds, a simple echo effect is obtained. The audio sounds as if it were being played in a softly furnished room where there is a single hard wall or reflective surface, facing the sound source. The apparent size of this room is directly proportional to the milliseconds delay time of the echo unit, and is fully variable up to 50 feet (50 mS delay).

A standard features of most echo units (including ours) is a Reverb facility. This allows a fraction of the output signal from the delay line to be fed back and added to the delay line *input,* so that you end up getting echoes of the echoes of echoes. By using only small amounts of feedback (often called 'Recirculation' or 'Regeneration' on commercial units), you get 'soft' reverb or, by adding lots of feedback you get 'hard' reverb. A variety of impressive effects can be obtained from the reverb facility, as follows:

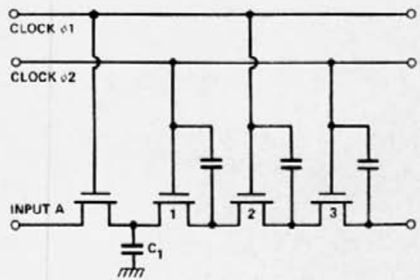
(3) When equal levels of mixing are used with maximum (50 mS) delay and maximum feedback, the sounds seem as if they are being played in a large hard-faced cave or chamber. The apparent dimensions of this 'chamber' can be varied via the delay-time control, while the apparent 'hardness' of the chamber can be varied by altering either the mixing or reverb level controls. Thus, the apparent sounds can be varied from those of a hard cave, to a small church, or down to a large but softly furnished lounge.

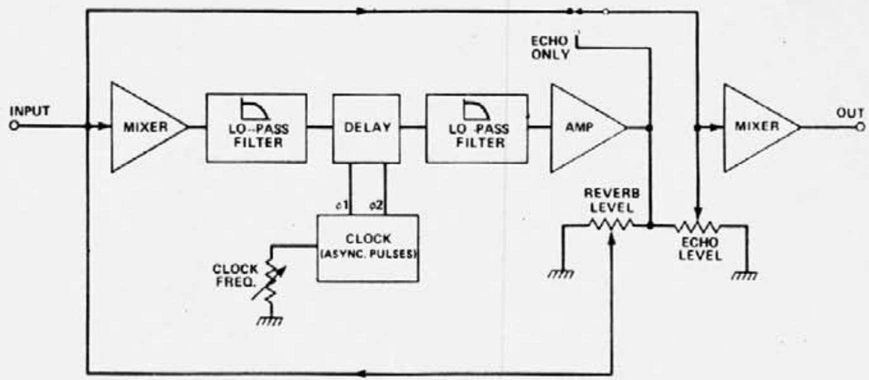
(4) When equal levels of mixing are used with short (a few mS) delays and a

HOW IT WORKS

THE STAGES for producing the various reverb effects from the Echo-Reverb, are shown in the block diagram. The main signal path comes from the ouput of the first mixer and through a 7 kHz low-pass filter. This filter is necessary to limit the audio band width to less than half the clock frequency, because the audio input is being *sampled* at the clock frequency (variable, to change the length of delay), and it is a fundamental principle of sampling that the sampling frequency must be at least twice the maximum input frequency. The filtered signal then passes through the delay line to a second low pass filter (at 15 kHz), which removes any clock signal residuals. This second filter includes a buffer amplifier to give the unit an overall gain of one. The output then splits into two paths; one is sent back to the input, to provide the reverb effect, the other goes to a final mixer via a switch. In the other position, the final mix can be varied from 'straight through' to full reverberation.

The delay circuitry comprises two charge coupled devices (CCD's) with 1024 delay stages. The diagram **right** shows an example of the internal structure of a MOS CCD IC. The principle may be compared to a line fo firemen passing buckets of water from one end to the other — hence the name 'buckets' are capacitors and the 'water' is an electric charge which is proportional to an instantaneous value of the input waveform — a sample. Each sample is stored briefly, then passed on to the next stage at the time of a clock pulse. Although each sample is stored for a very short time, at each stage, the time taken to 'clock' a sample from input to output can be as much as 50mS.





large amount of feedback, all audio signals sound as if they are being played inside a small-diameter hard-faced pipe or drum. The apparent dimensions of the 'pipe' are variable via the time-dealay controls and the apparent hardness of the 'pipe' is variable via the mixing or reverb controls.

The Circuit

The principle of the echo-reverb unit is described in How It Works. Audio signals enter the unit via RV1 and split into two paths which are re-united again, near the output, via a low-distortion audio mixer (IC7). One of these paths is virtually a direct link from the input of the unit (RV1 wiper) to one input of the IC7 mixer, via level control RV4. Thus, by varying the delay time and the setting of RV4, a range of different echo times and characteristics can be added to the original audio signals.

A fraction of the buffered output of the delay line can be tapped on via RV3 and fed back to the input of the delay line via the IC2 mixer stage. This produces echoes of echoes of echoes etc ('regeneration' or 'recirculation'), and is the standard characteristic of a reverb sound. The quality of the sound depends on the setting of RV3 (Reverb) and the delay time.

The delay line is formed by IC3 and IC4, a pair of series-connected TDA 1022 CCD (Charge-Coupled Device) "bucket-brigade" analogue ICs. They are clocked by a two-phase variable frequency oscillator formed by IC5, a 4046B phase-locked-loop chip. The TDA 1022s are 512-stage delay lines, so our circuit uses a total of 1024 CCD stages. The delay time available from these chips is:

$$D = \frac{P}{2} \times S$$

where P is the clock-cycle period and S is the total number of delay stages in the line. Our prototype is set up so that the clock periods are fully variable (via RV2) from a minimum of 2.5 uS (400 kHz) to a maximum of 60 uS (16.6 kHz), thus giving a delay range of 1.28 mS to 30.7 mS. In practice, however, the delay times can be extended to 50 mS by adjusting PR2 to give a maximum clock period of 97.6 uS (10.24 kHz) if some clock-signal breakthrough is acceptable on the output signal (see setting-up instructions, Max Delay Time).

When using CCD delay lines it is important that the clock frequency must be at least *double* the maximum audio signal frequency that will be used. The delay line output signal must be well filtered to cancel residual clock signals and the input to the delay line must be low-pass filtered, to avoid intermodulation problems by ensuring that the maximum input frequency is no higher than half the clock frequency. With these points in mind, the mixer IC2 with R7 and C7 are configured to give a 12 dB/octave slope, rolling off at 7 kHz at the front of the delay line. IC6 acts as a 12 dB/octave, 15 kHz low-pass filter at the output of the line.

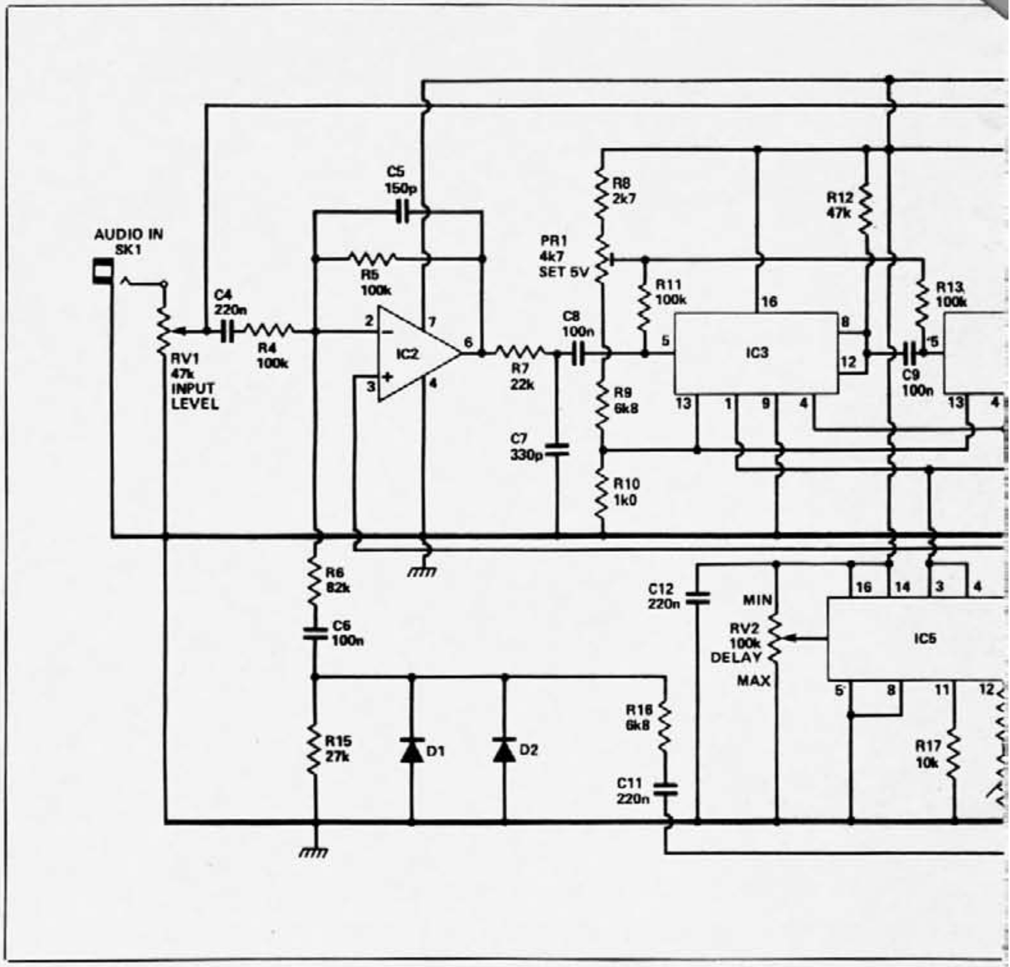


Figure 1. The complete Echo-Reverb circuit.

Final points to note about the circuit are that D1 — D2 — R15 — R16 are configured to give a degree of self-limiting on the reverb signals. This protects the delay line against destructive reverb overloads. The entire circuit is powered from a regulated mains-derived 15 volt supply via IC1 **(Figure 3 above).**

Construction

Most of the circuitry for this project is built on a single PCB, and construction should, therefore present very few problems. Before you start, however, a word of warning: the circuit includes a high frequency clock generator which tends to produce a fair amount of RFI (Radio Frequency Interference). Consequently, you should build it into a metal box and take lots of care over RF shielding.

Begin construction by fitting the seven wire links and the PCB-mounting mains transformer. Then proceed with the assembly of the remaining components, taking the usual care to observe component polarities, etc. Use sockets to mount the two delay-line chips (IC3 and IC4) and IC5; handle the chips with care, when fitting them into place.

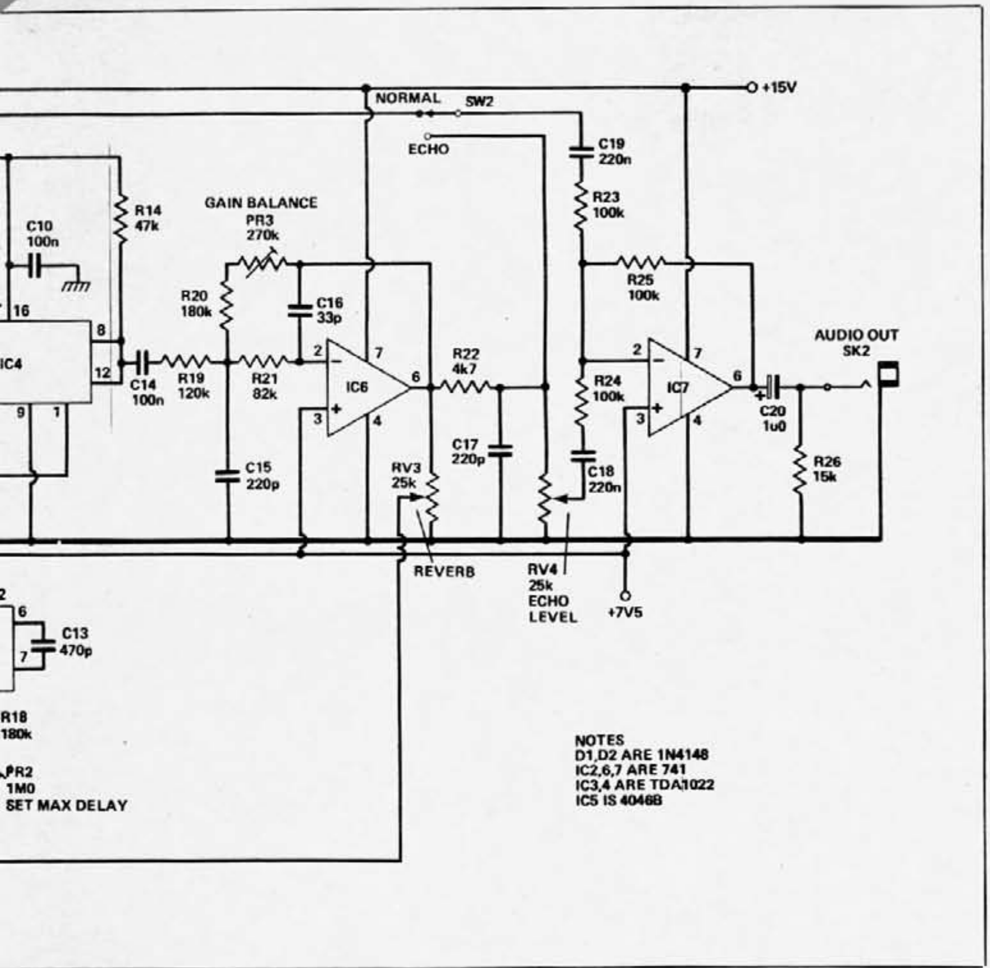
When the PCB is complete, temporarily wire the unit to all control pots, switches and sockets, then set-up the presets.

Setting Up Procedure

The Echo-Reverb unit contains three preset pots which must be correctly adjusted to make the unit fit for use; once these have been set correctly initially, they require no further adjustment. The pre-sets (PR1, PR3 and PR2) control the delay line biasing, the delay line loop gain and the maximum delay time, respectively. The settin up procedure is as follows:

Delay Line Biasing: With no input signal present, set all three pre-sets to zero, set SW2 to Echo Only, RV4 (Echo Level) to maximum and RV2 (Delay) to mid value. Connect a DC volt-meter between the + 15 V line (+ ve) and the wiper of PR1 (– ve). Then adjust PR1 for a reading of precisely 5 volts. Remove the meter. Now connect an audio (voice or music) signal to the input and check that it can be played through the unit without excessive audible distortion (i.e., the sound never becomes harsh).

Delay Line Loop Gain: With RV2 (Delay)

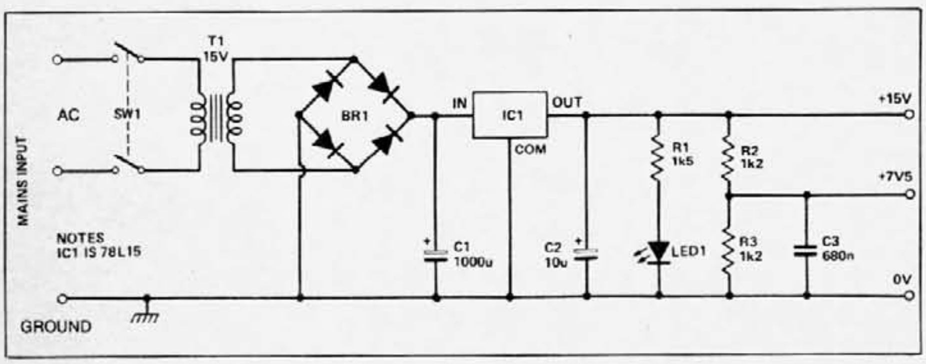


via RV2 and the reverb can be varied with RV3.

The setting-up procedure is now complete and the unit can be cased and made ready for use, as already described.

set to mid-range but with RV3 (Reverb) and RV4 (Echo Level) set to zero, connect a voice-range (350 Hz — 3k5 Hz) input signal of about 1 V peak-to-peak and monitor the output signal. Switch SW2 between the Normal and Echo Only positions, adjusting PR3 so that equal output levels are obtained in both positions (this test can be done with test gear or simply 'by ear', using a tape or disc signal source). When this adjustment is complete, set SW2 to the Echo Only position. Pass a music/voice signal through the system and use RV2 to check the Reverb sound is satisfactory.

Max Delay Time: Set SW2 to Normal, RV3 (Reverb) to zero, RV4 (Echo Level) to maximum (wiper at zero volts). Adjust PR2 while monitoring the output of the unit and note that high-pitched tone (whistle) is produced when PR2 is turned beyond a certain point. Now pass a voice signal through the unit; note that the echo effect is obtained, then trim PR2 to find a compromise setting at which a good delay (echo) is obtained with minimum acceptable intrusion from the 'whistle' sound. Finally, check that the delay can be varied over a wide range (roughly 2 mS to 50 mS)



Circuit diagram of the regulated power supply.

PARTS LIST

RESISTORS
(all ¼ W 5% carbon)

R1	1k5
R2, 3	1k2
R4, 5, 11, 13, 23, 24, 25	100k
R6, 21	82k
R7	22k
R8	2k7
R9 16	6k8
R10	1k0
R12, 14	47k
R15	27k
R17	10k
R18, 20	180k
R19	120k
R22	4k7
R26	15k

POTENTIOMETERS

RV1	47 linear carbon
RV2	100k linear carbon
RV3, 4	22k linear carbon
PR1	4k 7 miniature pre-set
PR2	1MO miniature pre-set
PR3	220k miniature pre-set

CAPACITORS

C1	1000u 40V electrolytic (axial)
C2	10u 35v tantalum
C3	680n polycarbonate
C4, 11, 12, 18, 19	220n polycarbonate
C5	150p ceramic
C6, 9, 14	100n
C7	330p polystyrene
C8, 10	100n polyester C280
C13	470p ceramic
C15, 17	220p ceramic
C16	33p ceramic
C20	1uO 35V tantalum

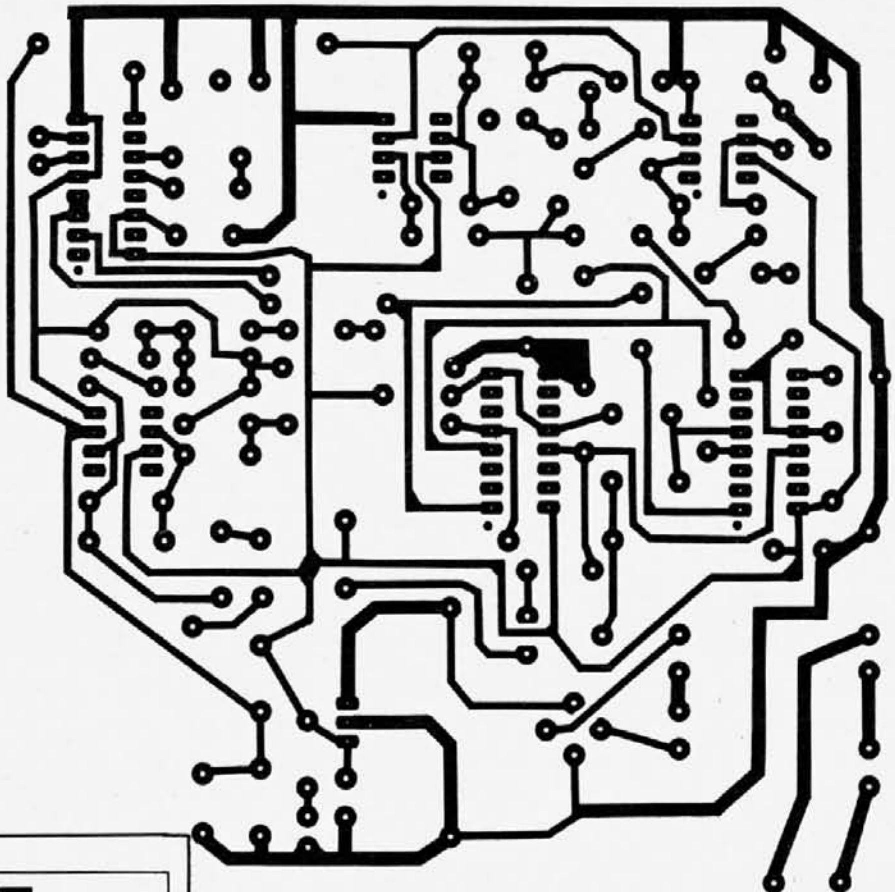
SEMICONDUCTORS

IC1	78L 15 voltage regulator
IC2, 6, 7	741 op-amp
IC3, 4	TDA1022 (Signetics) or DAC1022 (National) or TMS1022 (T1)
IC5	4046B CMOS phase locked loop
BR1	50V 1A bridge rectifier
D1, 2	1N4148 signal diode

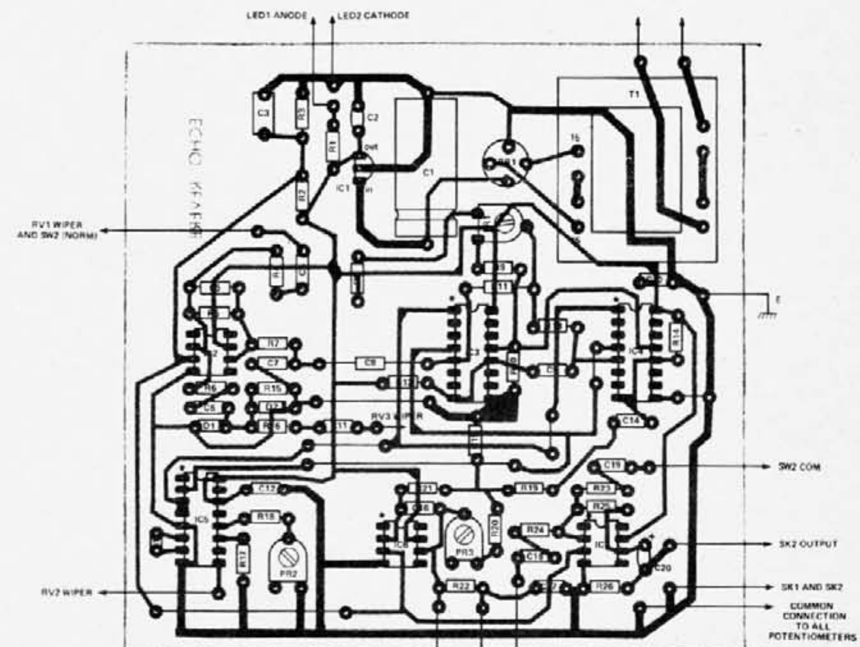
MISCELLANEOUS

T1	15 volt, 200 or 300 MA transformer
SW1	DPDT miniature rocker switch
SW2	SPDT miniature toggle switch
Sk1, 2	Phono Sockets

Case, PCB, bolts, knobs etc.



The printed circuit board.



The component location diagram. Transformer T1 may have to be located off the PCB and wires connected to the points labelled "15".

ANALOG REVERB FOR YOUR HI-FI

CARL SAWTELL*

While nothing may sound quite as good as music performed live in a concert hall, this system will add that "concert hall" feeling to your home sound system.

THE USE OF DELAY LINES IN AUDIO REPRODUCTION is an increasingly popular way to add a sense of realism to recorded music. By simulating the reverberation characteristics of a large room or hall electronically, and feeding that information to loudspeakers, the sensation of a large listening area is created. State-of-the-art systems using digital storage and microprocessors are capable of producing a complex, realistic simulation of large concert halls in a typical living room. Simpler systems, using either mechanical or electronic (digital or analog) delay schemes can produce a significant improvement in the audible performance of a music system, particularly if they can avoid the artificial quality that is associated with electronic reverberation.

Ideally, it would be desirable to simulate the natural reverberance of a concert hall. When a sound is produced on stage, a small fraction of the sound reaches a listener in the audience directly. This direct *(first arrival)* sound determines the direction and pitch of the source. Shortly thereafter, echos *(reflections)* reach the listener by the shortest path from a wall. More reflections occur as the sound bounces off other walls, the ceiling, the floor, and objects in the hall. Because there are, in essence, an infinite number of paths for the reflections to follow, the reverberation is not a series of individual echoes, but a continuous flow of sound. It builds up in a short period of time (typically a few milliseconds) and may take several seconds to die away. The reverberation time of a hall is defined as the time required for the sound level to decrease by 60 dB.

Electronic reverberation

Unfortunately, using either digital or analog delay lines, this sort of reverberation is difficult to simulate. The typical scheme for producing reverberation electronically is shown in Fig. 1. When a signal is applied to the input, it is delayed before it appears at the output. The delayed signal is fed back to the input after being reduced in level, so that it is delayed again. That feedback arrangement allows the reverberation quality to be changed, either by increasing the delay time or by changing the amount of delayed signal fed back to the input.

A system of this type does, however, have several drawbacks. First, the

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R-E TESTS IT

LEN FELDMAN
CONTRIBUTING HI-FI EDITOR

THE ANALOG REVERBERATION SYSTEM IS AN audio add-on unit that simulates the ambience and acoustic environment of large listening spaces such as concert halls, night clubs, auditoriums, and even cathedrals. An audio delay is introduced by using charge-coupled devices (CCD's), commonly referred to as bucket-brigade systems, instead of the A/D and D/A converters and digital signal-storage used by some other time delay/reverberation units. In use, program signals are taken from the main stereo system (using a TAPE OUT jack-pair) and connected to the two inputs on the reverberation unit. Some delayed and, if desired, reverberated, signals are fed from the output of the reverberation system to a secondary amplifier and, in turn, to one or more speakers positioned behind or to the sides of the listener. A single speaker, without an amplifier, can also be used.

As is true of all time-delay units of this type, the longer the time delay introduced, the narrower the bandwidth or pass-band of the time-delay system. The time delay available on this reverberation system varied from ap-

Continued

proximately 5 to 50 milliseconds. Those delay times are illustrated in the oscilloscope traces shown in Figs. 1 and 2. The upper trace in Fig. 1 is a tone burst. That tone burst was used as the input signal to the analog reverb system. With its DELAY control at minimum (fully *clockwise*) the output signal (the lower trace in Fig. 1) was displaced by approximately 5 milliseconds (the sweep rate is 5 ms per division in both Figs. 1 and 2). Figure 2 shows the maximum time-displacement between the input (upper trace) and output (lower trace)—about 50 milliseconds.

Two frequency-response curves are shown in Fig. 3. The upper curve, which has a rolloff of 3 dB (at the output of the device) at 3.5 kHz, shows the response obtained with the *minimum* time delay; the lower curve, in which response is already down by some 13.5 dB at the same 3.5 kHz test frequency, shows the response obtained with the DELAY control set to maximum.

While those response curves may appear to be anything but "high fidelity", you must understand that reflected sounds (which the delayed sounds are intended to simulate) also have their high frequencies highly attenuated. Highs are more easily absorbed by walls, floors, ceilings, and other surfaces, while mid-frequencies and lows tend to bounce back with little, or no loss. Thus, the tendency of the analog reverberation system to increasingly attenuate high frequencies as the delay time is increased is desirable, and is not an unwanted side-effect of this, or any other time delay/reverb system.

We measured the total harmonic-distortion of the reverberation system for a mid-frequency (1 kHz) and for a relatively low frequency, with 1 volt applied to the inputs. With delay time set to minimum, the total harmonic distortion was 1.6% at 1.0 kHz and 3.1% at 100 Hz. Turning the delay control to its opposite extreme, we measured a total harmonic distortion of 1.3% at 1 kHz and 1.55% at 100 Hz. Again, while these levels may seem a bit on the high side to audio buffs, it must be remembered that the total contribution of sound energy by the delayed channel is but a fraction of the total sound reaching the listeners' ears. That's because in an ideal setup of this kind, the listener adjusts the rear-channel (delayed) sound so that he or she is not consciously aware that there is a separate source at the rear of the listening room. Thus, if the contribution of the delayed channel is even just 3 dB lower than that of each of the primary channels, the total harmonic distortion added by the rear channel is only one-third as great as the numbers would imply.

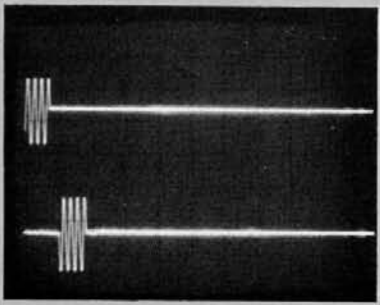


FIG. 1

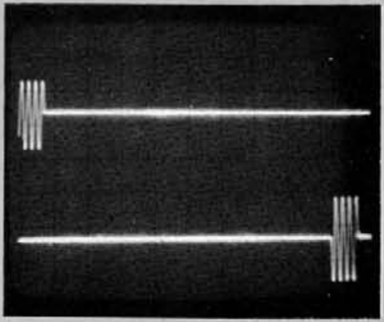


FIG. 2

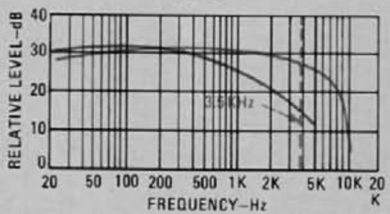


FIG. 3

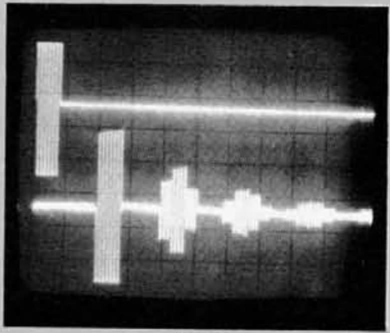


FIG. 4

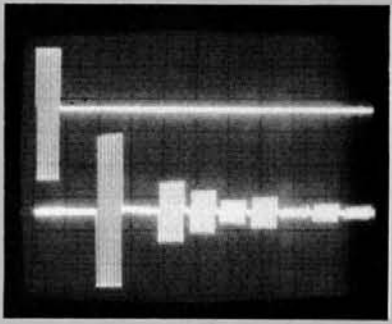


FIG. 5

Figure 4 shows what happens when the reverberation control is advanced, while the basic time delay is kept at its minimum and the FEEDBACK DELAY control is set to one extreme. Note the appearance of additional, delayed signals of decreasing intensity. Those extra signals have a decay characteristic similar to what would be found in a large hall with its own natural reverberant decay-time. Additional reverberation effects can be obtained by altering the setting of the FEEDBACK DELAY control, as can be seen in Fig. 5. In that figure the FEEDBACK DELAY control was set to its opposite extreme.

Listening tests

In addition to the measurements and observations just described, we hooked up the analog reverberation system to our own sound system and to an extra amplifier (in the "mono" mode) and pair of speakers. We played a variety of musical material through this system, alternately switching in and switching out the reverberation unit. The unit, once properly adjusted for the type of program material (and that is very important), added a sense of space to our modestly proportioned listening room. We found that the reverberation control should be used in moderation. If used to excess, it gave a false quality—almost a ringing or oscillatory characteristic—to the music. That was not the case with the DELAY control however. When that control was varied, the apparent size of the listening room simply seemed to change. **R-E**

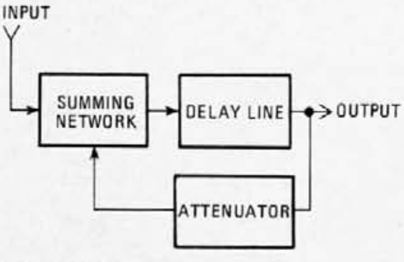


FIG. 1—ELEMENTS OF A REVERBERATION SYSTEM. Part of the output-signal is attenuated and fed back to the input to the delay line to generate a slow decay.

echoes are not random. If the delay time is 10 ms, for example, the second echo appears after 20 ms, the third at 30 ms, and so forth, as shown in Fig. 2. If the delay time is long, those echoes may be heard individually and a "flutter" echo is produced. A natural echo contains many separate random echoes that (in a well designed hall) cannot be heard individually.

A second problem with this simple reverberation system is that it produces a "comb-filter" effect. If we again assume a 10-ms delay, then the input

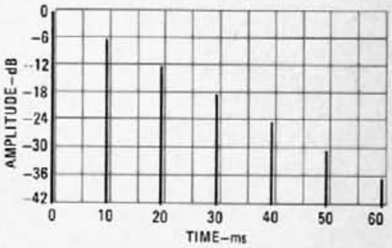


FIG. 2—THE RESPONSE OF A SIMPLE REVERBERATOR to a short pulse. If the delay time is long, the echoes can be heard individually producing a flutter effect.

and the delayed signal are 180 degrees out of phase at 50 Hz. (The period of a 50-Hz signal is 20 ms.) When added together, the sum is the *difference* of the two, and the result is a decrease in output level. At 100 Hz, the input and output are in phase (360-degree phase shift), and add together. Under those conditions, dips in the response would also occur at 150 Hz, 250 Hz, 350 Hz, and so forth. Likewise, peaks in the response would occur at every 100-Hz interval. Figure 3 illustrates that response. Over a 10-kHz range, there would be 100 of those peaks and dips. And as the amount of feedback is increased, the height of the peaks and the depth of the dips also increases.

These problems are actually quite similar to those encountered in a room of poor acoustical design. A large tiled shower is a good example. The hard, reflective walls and the boxiness of the room's shape will create the same sort of flutter echoes. The room's dimensions will also set up "standing waves"

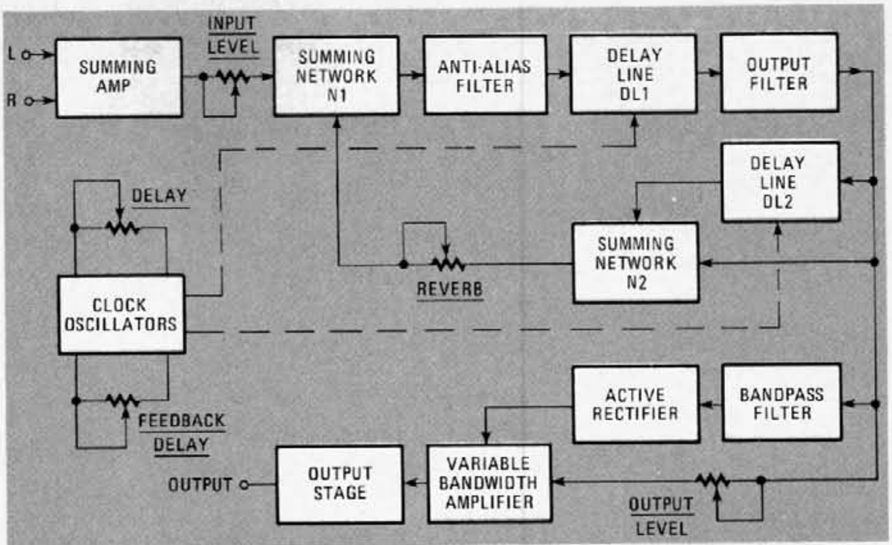


FIG. 6—THE SIGNAL-PROCESSING PATH of the analog reverberation system shows how the original signal is recirculated at ever-decreasing levels.

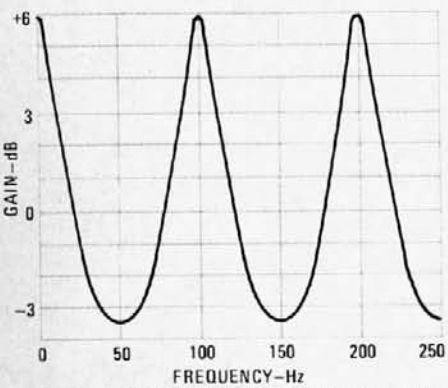


FIG. 3—A COMB-FILTER EFFECT takes place when the input and delayed signals are 180° out of phase, causing a decrease in the overall output-level.

that will simulate the "comb-filter" effect described earlier.

Fortunately, the physical solutions to the problem of poor room-acoustics can be carried over into the design of an artificial-reverberation device. Instead of changing the dimensions of the room to give more echoes of different lengths, we can provide delay lines of different lengths in the device. Instead of breaking up the standing waves with objects or acoustical treatment, we can inject delays into the electronic feedback to break up the pattern. A block diagram of the simple system to do that is shown in Fig. 4.

The multiple-feedback reverberation technique is still not a close simulation of actual concert hall reverberation, with its complex combination of delays. This technique does, however eliminate or reduce the most objectionable artifical aspects of electronic reverberation. If delay line DL1 is 10 ms as before, and delay line DL2 is 3 ms, then the resultant echoes will be as shown in Fig. 5. Note that not only do more echoes appear, but that the number of echoes increases with time, just as it would in a natural environment.

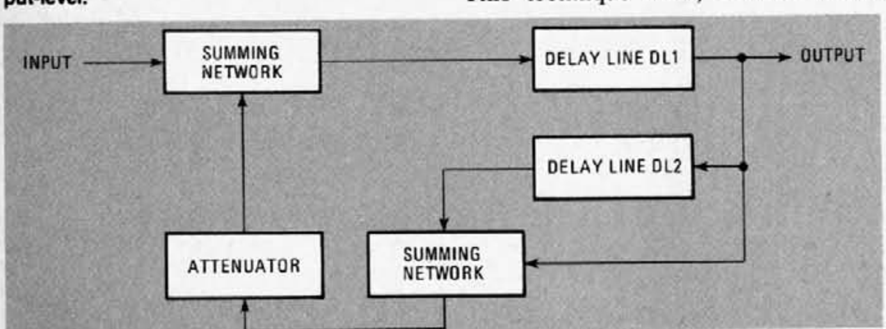


FIG. 4—A MULTIPLE-FEEDBACK SYSTEM uses more than one delay line to break up the pattern of regular echoes.

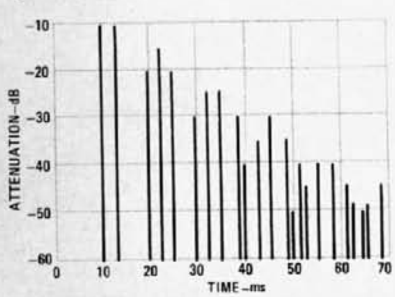


FIG. 5—THE NUMBER OF ECHOES increases with time in a multiple-feedback system, just as it would in an actual concert hall.

The frequency response of a multiple-feedback reverberation device is complex. The peaks and dips remain, but are irregular. The large peaks occur only if the delayed input, the output, and the delayed output are all in phase, which makes the number of large peaks decrease; however, that situation is unlikely to occur. The number of deep dips in the response tends to be reduced similarly. There are still many ripples present, but they are not deep.

A block diagram of the complete reverberation system is shown in Fig. 6. The stereo signal from a receiver or preamp is converted into a monaural signal by a summing amplifier. The INPUT-LEVEL control allows the signal level to be adjusted for the optimum signal-to-noise ratio. The signal is then filtered through a five-pole "anti-aliasing" filter to minimize intermodulation distortion. *Aliasing* is a phenomenon that occurs in sampling systems. Our delay lines use bucket-brigade IC's. Essentially, those IC's are sampling devices with the sampling rate being determined by the clocking frequency of the IC's. If the input signal being sampled contains components that are higher in frequency than can be handled by the sampling rate, aliasing occurs. Then, the high-frequency components are "read" as *low-frequency* components and appear at the output of the sampling device along with the low-frequency components of the input signal. The low-frequency components mix together and intermodulation distortion results. To prevent aliasing from occurring, the input signal is filtered before it reaches the sampling device by either a low pass or bandpass filter to eliminate the signal components that are too high in frequency for the sampling rate to handle. A filter of that sort is commonly referred to as an *anti-aliasing filter*.

The main delay line in Fig. 6 is DL1. Its output is filtered by a seven-pole active filter to eliminate switching waveforms, ultrasonic signals, and to reduce the likelihood of creating beat frequencies from the high-frequency signals present in the system. The de-

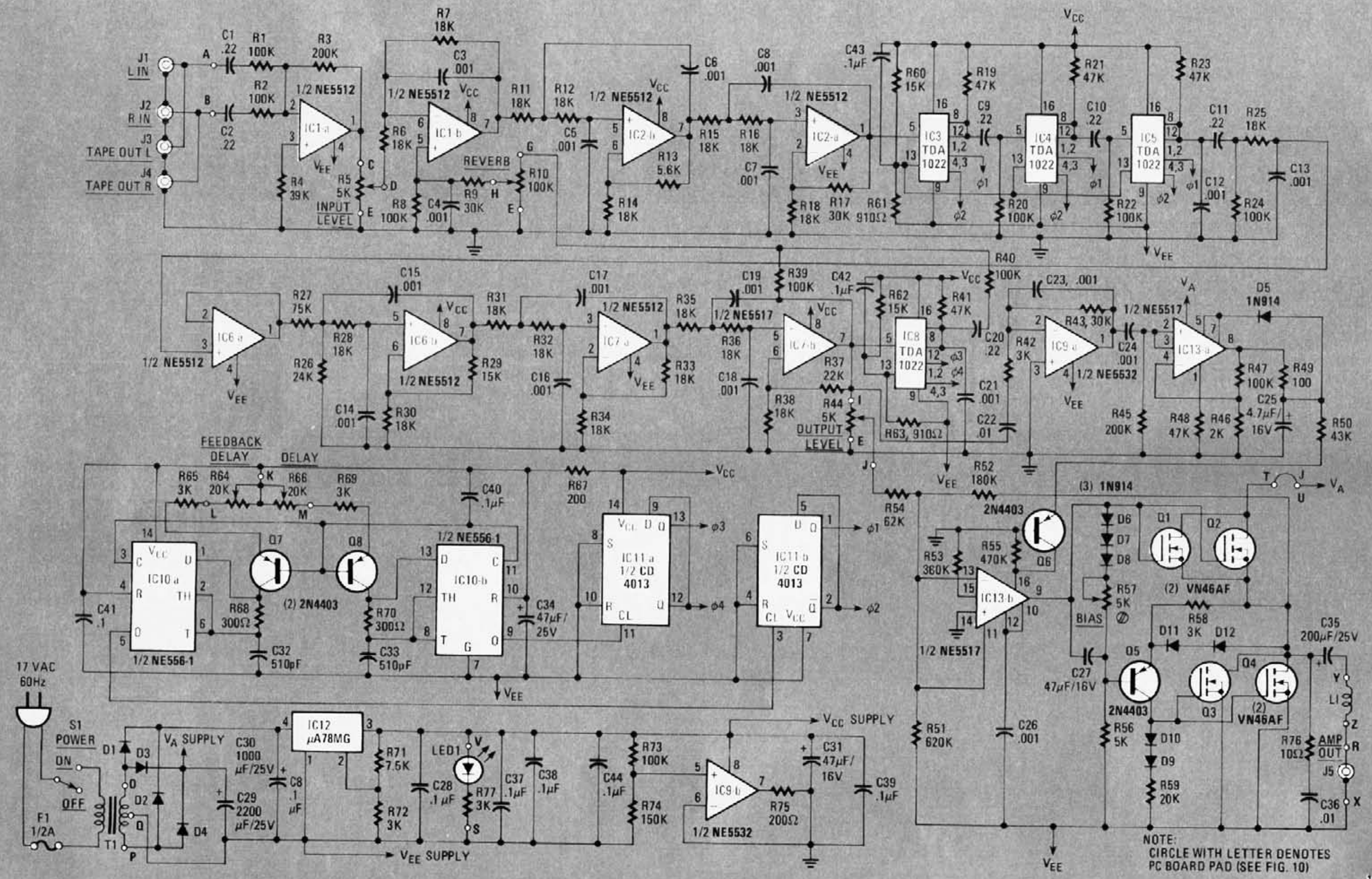


FIG. 7—SCHEMATIC DIAGRAM of the analog reverberation system. The delay required by the system is provided by IC3, IC4, IC5, and IC8—four TDA1022 "bucket-brigade" IC's (how they work is discussed in the text).

DIGITAL DELAY VS. BUCKET BRIGADES

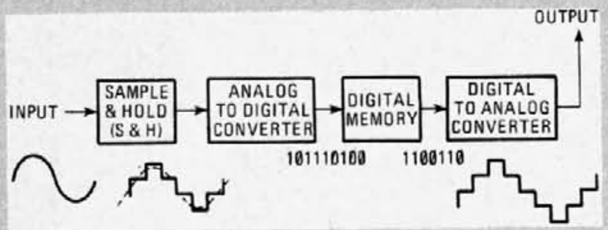


FIG. 1

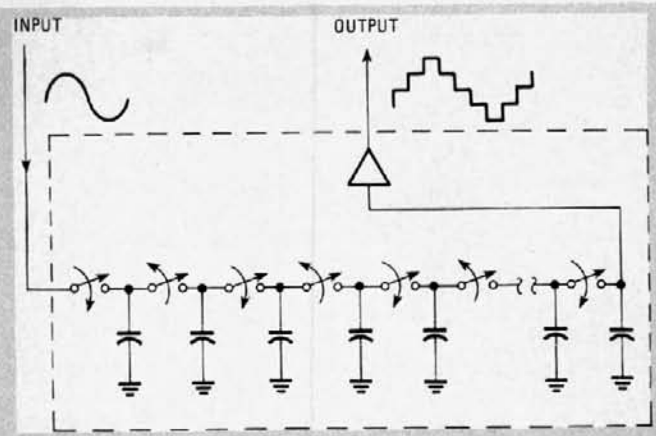


FIG. 2

TWO ELECTRONIC APPROACHES ARE CURrently used to delay analog signals. The one that has received the most attention lately is digital delay, shown in Fig. 1. In that system, the signal is measured at regular intervals *(sampled)* and the sampled voltage is converted to a number *(quantized)*. The sampling operation is done by a sample-and-hold circuit that uses a capacitor to store a voltage representing the instantaneous signal-level. The voltage stored by the capacitor is temporarily held constant, making it possible for an analog-to-digital converter to derive a digital value for it. (Not all A/D systems require sample-and-hold circuits, but those commonly used for audio produce significant errors if the input voltage changes during the conversion process.) A number representing the signal can be stored in digital memory and, after the desired delay-time, be reconverted to an analog voltage. The output of the D/A converter may also contain a sample-and-hold circuit to store the output voltage during the next conversion.

The second method is the bucket-brigade delay line (BBD), a single-IC delay system that is, in effect, an analog shift register (see Fig. 2). Like the digital delay-line, the bucket brigade is a sampled system, but no digitizing is involved. Manufactured as a long string of MOSFET switches and capacitors, the BBD acts like a long string of sample-and-hold devices. At the beginning of a clock cycle, the input signal is stored by the first capacitor. During the second half of the clock cycle, that voltage is transferred to a second capacitor, and the input capacitor is ready to store a new voltage. During the next clock cycle, the original input signal is transferred from the second to the fourth capacitor, and so on. After 256 clock pulses, the original input-voltage appears at the 512th storage capacitor, which is the output of the TDA1022. The length of time it takes to transfer a signal (the delay) depends on the frequency of the clock used. **R-E**

layed signal is fed back to the input of the anti-aliasing filter through summing network N2. It is also fed to a second delay line to provide the second feedback path. The level of the delayed signals is controlled by an attenuator (REVERB control) and then combined with the input signal by summing network N1.

The output-filter signal, which consists of the input-filter signal plus all of the delayed signals, is first attenuated by the OUTPUT-LEVEL control and then fed to a variable-bandwidth filter. The control signal for this filter is derived by a bandpass filter and active rectifier. Those three blocks make up a noise-reduction circuit that minimizes the level of audible hiss in the output signal.

The final block in the signal-processing chain is an amplifier. This amplifier provides an output of approximately one watt to an 8-ohm speaker. The amplifier is also quite capable of driving a 4-ohm speaker. Although this output is minute when compared to that of the main amplifier in a hi-fi system, it is more than adequate when the reverberation system is properly set up.

In practice, the output from the reverberation unit drives a third speaker at the rear of the listening room. The output-level control is adjusted so that the listener is not consciously aware of the third speaker. In fact, the third speaker should be barely audible over the two main speakers.

How the circuit works

Figure 7 shows a complete schematic of the reverbation system. The input is taken from the tape-monitor outputs of your hi-fi system. Because the input signal will typically come from the tape output of a preamp or receiver, the inclusion of a second set of jacks (J3 and J4) allows for the connection of a tape deck. The input impedance of the reverberation unit is 100 kilohms and it should not load down your hi-fi's tape-monitor circuit significantly.

Capacitors C1 and C2 couple the input signal into summing amplifier IC1-a. The gain of this stage is 6 dB and can be modified by changing the value of R3. You may want to alter the gain if the unit is to be used in applications with particularly low-level signals, such as those from a microphone or electric guitar, or with high-level ones like the output of a power amplifier. Since the INPUT-LEVEL control, R5, follows this first stage of amplification, it is important that the signal applied to it not be too large, or overload will result. (As designed, the inputs will safely handle a 2-volt input level, more than sufficient for line-level inputs from a receiver.)

Op-amp IC1-b serves two purposes. First, it provides unity-gain inversion of the signal from the input-level control with capacitor C3 limiting the bandwidth of the signal to 9 kHz. Its second purpose is to sum the input signal with the delayed (feedback) signal. The level of this feedback signal is controlled by the REVERB control, R10. The R-C network connecting R10 to the positive input of IC1-b serves to reduce the feedback at high frequencies. This simulates the natural tendency for acoustically-reflective materials to absorb high-frequency sound, giving more reverberation at low frequencies.

Op-amps IC2-a and IC2-b form an active low-pass filter with a cutoff frequency of 9 kHz. Together with the filtering action of summing amplifier IC1-b, they form a five-pole anti-aliasing filter that rolls off the input signal at a rate of 30-dB per octave. This filter reduces noise and distortion in the system by reducing the potential for intermodulation distortion (aliasing) in the delay lines. The slew rate of the input signal is also reduced by this filter. Too high a slew rate causes distortion in the op-amps. (This is not to imply that this would be a significant problem. The NE5512 op-amp specified can produce full output at 20 kHz without reaching its slew-rate limit.)

Philips TDA1022 bucket-brigade delay IC's are used for the main delay line. These are called out as IC3, IC4, and IC5 in Fig. 7. Although identical to other commercially available bucket-brigade IC's in most respects, the TDA1022 is unusual in that it uses p-channel MOSFET's. The three delay-IC's are driven by a common clock and are cascaded to give three times the delay of a single IC. Capacitive coupling between stages minimizes the effects of DC offsets in the delay line.

The output of the third bucket-brigade device, IC5, is filtered by another 9-kHz R-C filter and fed into op-amp IC6-a, which is connected as a voltage follower. This is followed by three more active-filter stages consisting of IC6-b, IC7-a, and IC7-b. This active filter provides a 36-dB-per-octave rolloff above 9 kHz and is designed so that the complete system—including the input and output filters—has a flat response below the cutoff frequency. This means that neither the input filter nor the output filter has a flat response, but the minor ripples in the responses tend to cancel and give a flat frequency-response below 9 kHz followed by a 72-dB-per-octave rolloff.

The output of the filter is fed to a fourth bucket-brigade device, IC8, which was shown as DL2 in Fig. 6. Both the filter-output and the output of IC8 are fed to the REVERB control and combined with the input signal by summing amplifier IC1-b. The output of the filter is also fed to the OUTPUT LEVEL control, which is the primary volume control for the system.

The signal from the OUTPUT LEVEL control is fed to IC13-b. Two major functions are provided by op-amp IC13-b. First, it supplies drive for the output-amplifier stages. Although certainly not a high-power amplifier, its output is adequate for most purposes, and it eliminates the expense of having to add a power amplifier to the system. This IC also acts as the variable-bandwidth filter in a noise-reduction circuit. By making the bandwidth vary as a function of the signal level, the noise in the output signal is reduced.

A transconductance amplifier, the NE5517, was selected for IC13. This IC contains two independent transconductance-amplifiers in one package. By varying the current applied to pins 1 and 16, the gains of the two sections can be controlled independently.

To minimize audible side-effects of the noise-reduction circuit, the bandwidth of IC13-b is made a function of the high-frequency content of the input signal. The high-frequency content is sensed by a bandpass filter composed of IC9-a and its feedback network. The input to this filter is taken from a point ahead of the OUTPUT LEVEL control so that it is not dependent on the volume setting. The output from IC9-a feeds IC13-a, which is configured to operate as an active rectifier. The gain of IC13-a is set to 51 by bias resistor R48. After the signal is rectified by IC13-a, a DC potential exists on C25 that is a measure of the high-frequency level, and determines the bandwidth of the dynamic filter.

The variable-bandwidth output filter works by varying the gain of IC13-b. Compensation for the amplifier is provided by capacitor C26, which sets the gain-bandwidth product. If the gain of the amplifier is varied by changing the bias current applied to pin 16, the bandwidth is also varied. To accomplish this, the control voltage from C25 is fed to pin 16 of IC13-b through a network consisting of R50, R55, and Q6 which develops a current proportional to the rectifier output. This causes the bandwidth of the amplifier to be proportional to the high-frequency level detected by the rectifier circuit.

The power amplifier's output stage uses four VMOS power transistors (Q1-Q4) in a push-pull configuration. Transistors Q1 and Q2 act as a source follower to drive the load directly. Transistors Q3 and Q4 are driven by Q5, a small-signal PNP-type, which senses the gate drive-voltage on Q1 and Q2 through diode-string D6-D8. When the gate drive-voltage is low (Q1 and Q2 beginning to turn off), Q5 provides drive for the gates of Q3 and Q4, which supply drive current for the negative portion of the cycle. Although Q1 and Q2 will turn off completely on large negative swings with a low-impedance (speaker) load, for most purposes the amplifier operates class A. The operating current is set by trim pot R57. In the case of high current-loads, D11 and D12 prevent the gates of Q1 and Q2 from being pulled negative with respect to the source, an undesirable condition for the VMOS transistor.

The amplifier is coupled to the speaker through capacitor C25. The R-L-C network at the amplifier's output decouples the loudspeaker from the feedback loop to improve stability. Although this amplifier has a limited power-output (due primarily to power-supply limitations), it can safely be used with 4-ohm loads. Transistors Q1 and Q2, and Q3 and Q4, are connected in parallel for this purpose. No current-hogging or thermal instability results from connecting VMOS devices in parallel (as

continued on page 104

PARTS LIST

Resistors ¼ watt, 5%, unless otherwise noted

R1, R2, R8, R20, R22, R24, R39, R40, R47, R73, R74—100,000 ohms
R3, R45—200,000 ohms
R4—39,000 ohms
R5, R44—5000 ohms, potentiometer, audio taper
R6, R7, R11, R12, R14-R16, R18, R25, R31-R36, R38—18,000 ohms
R9, R17, R30, R43—30,000 ohms
R10—100,000 ohms, potentiometer, linear taper
R13—5600 ohms
R19, R21, R23, R41, R48—47,000 ohms
R26—24,000 ohms
R27—75,000 ohms
R28—27,000 ohms
R29, R60, R62—15,000 ohms
R37—22,000 ohms
R42, R58, R65, R69, R72, R77—3000 ohms
R46—2000 ohms
R49—100 ohms
R50—43,000 ohms
R51—620,000 ohms
R52—180,000 ohms
R53—360,000 ohms
R54—62,000 ohms
R55—470,000 ohms
R56—5000 ohms
R57—5000 ohms, trimmer potentiometer
R59—20,000 ohms
R61, R63—910 ohms
R64, R66—20,000 ohms, potentiometer, linear taper
R67, R75—200 ohms
R68, R70—300 ohms
R71—7500 ohms
R76—10 ohms

Capacitors

C1, C2, C9-C11, C20—.22 μF, 100 VDC, Mylar
C3-C8, C12-C19, C21, C23, C24, C26—.001 μF, polystyrene
C22—.01 μF, polystyrene
C25, C27, C31, C34—4.7 μF, 16 VDC, electrolytic
C28, C37-C44—.1 μF, ceramic disc
C29, C35—2,200 μF, 25 VDC, electrolytic
C30—1000 μF, 25 VDC, electrolytic
C32, C33—510 pF, ceramic disc
C36—.01 μF, 400 VDC, electrolytic

Semiconductors

D1-D4—1N4002, 100 PIV, 1 amp
D5-D12—1N914
LED1—jumbo red LED
Q1-Q4—VN46QF VMOS transistor (Siliconix)
Q5-Q8—2N4403 PNP transistor
IC1, IC2, IC6, IC7—NE5512 low-noise dual op-amp (Signetics)
IC3-IC5, IC8—TDA1022, 512-stage bucket-brigade device (Philips)
IC9—NE5532 low-noise dual op-amp (Signetics)
IC10—NE556-1 dual timer (Signetics)
IC11—CD4013 dual D flip-flop (RCA)
IC12—μA78MG adjustable voltage regulator (Fairchild)
IC13—NE5517 TCA (Signetics)
L1—10 turns of No. 22 wire wound around C35
T1—36 VCT, 300 mA

Miscellaneous: PC board (double-sided with plated-through holes), case, hardware, etc.

NOTE: The following are available from Advanced Analog Systems, Inc., 790 Lucerne Dr., Sunnyvale, CA 94086 (Tel. 408-730-9786): ARS-911—complete kit including case, $149.95; PC-911—PC board only, $24.00; IC-911—IC1-IC13 and Q1-Q8 only $49.95. Visa and Mastercard welcome. California residents please add sales tax. Prices include shipping (within continental U.S. only).

ANALOG REVERB

continued from page 48

might be the case if bipolar transistors were used), and this configuration serves to split the power dissipation between the transistors and increases the transconductance (gain).

The power supplies for this unit are somewhat unusual. A separate supply is used for the power amplifier. This 25-volt supply (V_A) is formed by D3, D4, and C29 in a simple full-wave rectified unregulated supply. Since this voltage may drop considerably when driving a speaker at high levels, separate rectifiers and filter capacitors are provided for the main supply (V_{CC}). This supply is regulated by IC12 to +18 volts.

It was found that using split supplies for the op-amps introduced excessive noise—any variation in the positive supply appeared at the outputs of the op-amps along with the signal. To eliminate this problem, an artificial "AC ground" was created that closely tracks V_{CC}. That artificial AC-ground serves two purposes. First, it is used as the ground reference for the input signal. Second, by connecting it to the "unused" input of the op-amps, it cancels the variations originally induced by V_{CC}.

The AC ground is generated by a resistor divider-network, R73 and R74, and applied to op-amp IC9-b. The ground is coupled closely to V_{CC} by capacitor C31. Integrated circuit IC9 is a 5532 dual op-amp, used because of its high drive-current and its high slew-rate; it reduces voltage variations much better than a voltage-regulator IC.

You should note that there is a possibility for confusing the input and output grounds. The input should be grounded to "AC ground." This provides minimum noise and is most consistent with the rest of the circuit. If the output is used to drive another amplifier or other electronic device, care must be taken because AC ground is the output of an op-amp, and only about 30 mA can be drawn from this supply.

Because of this situation, direct connection to a speaker is made to the negative supply, V_{EE}, which is 10 volts below AC ground. *The loudspeaker ground must remain isolated from the AC ground or the supply will short out!* (This will cause no damage, but *will* result in silence. The 5532 is fully protected against such abuse.)

The last circuit we will discuss is the one for clocking the bucket brigades. The NE556-1 (IC10) is a dual 555 timer configured in the astable (oscillator) mode. The clock signal for the main delay-line (IC3, IC4, and IC5) is produced by IC10-a and the clock signal for the feedback delay-line (IC8) is produced by IC10-b. The frequencies are determined by the values of capacitor C32 for IC10-a and C33 for IC10-b, and by the current available to charge them.

Instead of using resistors to develop charge-current, transistors Q7 and Q8 provide the currents directly. These currents are adjusted by potentiometers R64 (FEEDBACK DELAY) and R66 (DELAY). The outputs of the 556 are applied to IC11, a dual-D flip flop. The flip-flop converts the pulse outputs from the 556 to square waves of one-half the original frequency. The availability of both Q and Q outputs provides the out-of-phase square wave clock-signals needed for the bucket-brigade devices.

Now that you know how the Analog Reverberation Unit works, you probably want to know how to build one. We'll show you when we continue this article next month. **R-E**

ANALOG REVERB FOR YOUR HI-FI

CARL SAWTELL*

Duplicate the acoustics of a large concert hall with this accessory for your stereo system. The construction details are presented here.

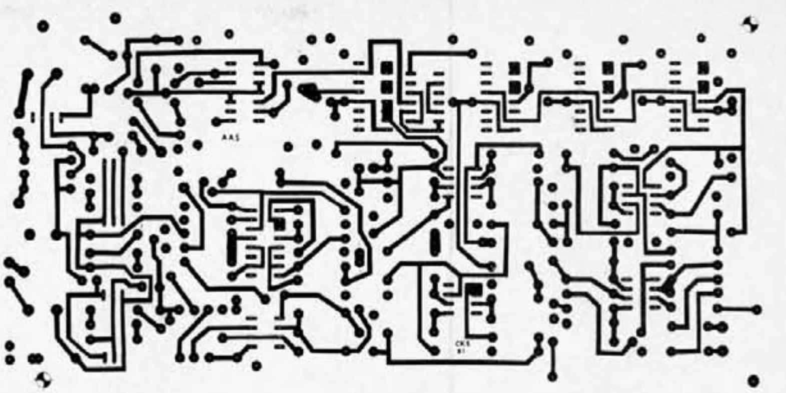
[redacted]rt 2 IN SEPTEMBER WE described an analog reverberation unit that adds realism to recorded music. Have that issue handy as we continue by showing you how to build your own.

Construction

Foil patterns for the double-sided PC board are shown in Figs. 8 and 9. Note that the component-side of the board is laid out so that, when trimmed, it will be divided into two electrically-isolated sections. Almost all of the reverberation unit's components are mounted on the PC board (refer to Fig. 10). The board is double-sided, so unless it is plated-through, care must be taken to connect the two sides using jumpers where the foil patterns on both sides of the board coincide. Generally, components (including integrated circuits) that connect to the ground plane should be inserted first and soldered to minimize static-electricity problems—especially if you are not using sockets. Do not install LED 1—you'll need it to check out the unit. Note that connections to the off-board components (front panel controls, input jacks, etc.) are made to the pads labeled "A" through "U," corresponding to similarly labeled points in the schematic.

The front-panel controls should be connected to the board before they are mounted mechanically. The AC ground (point "E") is common to the INPUT LEVEL, OUTPUT LEVEL, and REVERB controls; wiring one end of each of those pots together and then wiring to point "E" on the PC board is the simplest way to make that connection. The wires from the controls to the PC board should be about 8 inches long.

Care must be taken to isolate the output jack from the chassis (AC ground). The easiest way to do that is to put an insulating layer of electrical tape or Mylar film over the chassis hole (from the inside), and cut a similar, but slightly smaller, hole through the tape or film. If that is done, the jack will not come in electrical contact with the chassis when it is installed.



8–1/2 INCHES

FIG 8—FOIL PATTERN for the bottom side of the PC board. Note that the PC board is double sided (see Fig. 9).

*Signetics Corp., Sunnyvale, CA

The jumper from "T" to "U" *must be left disconnected* during checkout to reduce the chance of damaging the output stage. Also, R57, which sets the bias current in the output stage, should be set at its minimum value (i.e., D8 shorted to the base of Q5). The checkout procedure is as follows:

1. Using an ohmmeter, make sure that all sections of the ground plane (on the component side of the board), *except for the one that runs to the right-hand edge of the board,* are connected together. (That section is V_{EE}; the others are AC ground).
2. Plug the unit in, and turn it on. Using a voltmeter, read the voltages between the transformer's center tap (V_{EE}) and the cathode of D1 and the cathode of D3. In both cases the voltage should be 25-volts DC.
3. Referring to voltmeter readings to V_{EE}, check the positive voltage on pin 8 of each op-amp (IC1, IC2, IC6, IC7, and IC9). It should be about 18 volts. The voltage at the ground plane on the left-hand side of the board should be about 10 volts.
4. With the LEVEL and OUTPUT LEVEL controls set at their minimum values, and the DELAY and FEEDBACK DELAY controls set at 50% of full scale, the output (pins 1 and 7) of all op-amps should be about 10 volts.
5. Check the clock pulses at the outputs of the CMOS D flip-flops (IC11-a and IC11-b). The DC reading should be about 9 volts at those points; the AC reading should be 5-10 volts, depending on the type of meter used.
6. To adjust the bias current in the output stage, temporarily connect LED 1 from point "T" to V_{CC}. Current flowing through Q1 and Q2 (also Q3 and Q4) will now flow through the LED. (If you wish, an ammeter can be connected between point "T" and V_{CC} and used in place of the LED for this checkout procedure.) Adjusting R3 will cause Q1 and Q2 to conduct and LED1 to glow. Since distortion is reduced with increasing current level, it is desirable to keep the bias current reasonably high; however, if the current is too high, reliability will be reduced. Carefully touch the power FET's (Q1-Q4) and adjust R57 until LED1 glows brightly but the FET's do not get hot. You should read about 15 mA if you're using an ammeter. After the bias current has been set, LED1 should be installed between R77 and point V and a jumper placed between points "T" and "U."

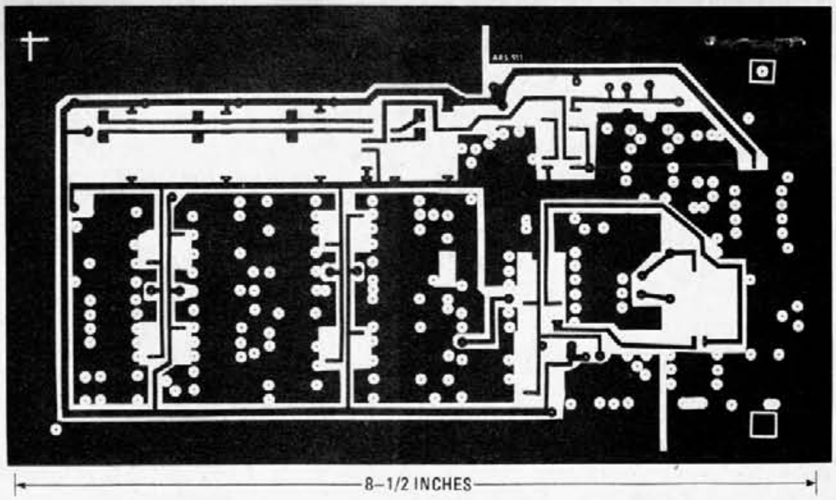


FIG. 9—COMPONENT (TOP) SIDE of the double-sided PC board. This pattern is NOT a negative—there is no foil around many of the holes so the component leads can pass through the board without contacting the ground plane.

Once you've completed the checkout procedure and are sure that everything is correct, the PC board, potentiometers, transformer, and jacks should be mounted securely in the case. The PC board should be mounted on standoffs. The case used, and its layout, are not critical as long as everything fits comfortably (the case shown in Fig. 11 is included with the kit available from the supplier listed in the Parts List).

Setup

The reverberation system is connected to your stereo system using the stereo's tape-monitor output. The output of the delay unit can be connected to a high-efficiency speaker. That speaker is generally placed at the rear of the listening room. Set up that way, the analog reverberation system can simulate the sound reflected from the rear of a concert hall.

Use of the controls on the front panel is reasonably straightforward. The INPUT LEVEL control adjusts the sensitivity of the unit for maximum dynamic range. With the OUTPUT LEVEL control set so that the output level is low (to avoid overloading the amplifier), the INPUT LEVEL control is set so that the level is as high as possible without overloading on loud passages. Initially, the REVERB control should be kept at its minimum position. The DELAY control is adjusted for the desired (first-arrival) delay; this is best done with your system playing at a low level so that both outputs can be heard at the same time. The FEEDBACK DELAY control is not likely to have a dramatic effect on the sound quality. While that control's presence in the circuit breaks up the "standing-wave" effect, its precise setting is unimportant. Adjust the FEEDBACK DELAY control for minimum noise. (The presence of two clock-signals causes a limited amount of intermodulation, heard as whistles and tweets. They are eliminated by adjusting the FEEDBACK DELAY control).

The degree of reverberation is adjusted with the REVERB control. There is a definite threshold where audible reverberation begins. Beyond that point, the reverberation becomes both more pronounced and more artificial; the system will actually oscillate if the REVERB control is turned up too high. Even before oscillation occurs, there is an increase in the peak signal-level that may force you to turn down the INPUT LEVEL control.

The most difficult adjustment to make is setting the OUTPUT LEVEL control. There is a strong temptation to make the delayed signal too loud. Bear in mind that the more subtle the effect of the reverb, the more impressive it will be! That seeming contradiction is something one usually learns the hard way; perhaps this advice will help.

Speaker selection

The choice of a rear speaker is important if the system is to work properly. The delayed channel does not have as wide a bandwidth as the front channel, so a wide, flat, powerful high-frequency driver is unnecessary. With about one watt of power available, efficiency is far more important than power-handling capability.

Looking at some of the "mini-speakers" that are currently on the market can help us understand the reverberation system's speaker requirements. Those small, acoustic-suspension, two-way systems have two notable features: most have excellent high-frequency response, and all are inefficient. Their lack of efficiency prevents them from playing loudly, but their output is more than adequate for most purposes. The high-frequency response is, if anything, a point against that type of unit. The high frequencies do not help the reverberation system

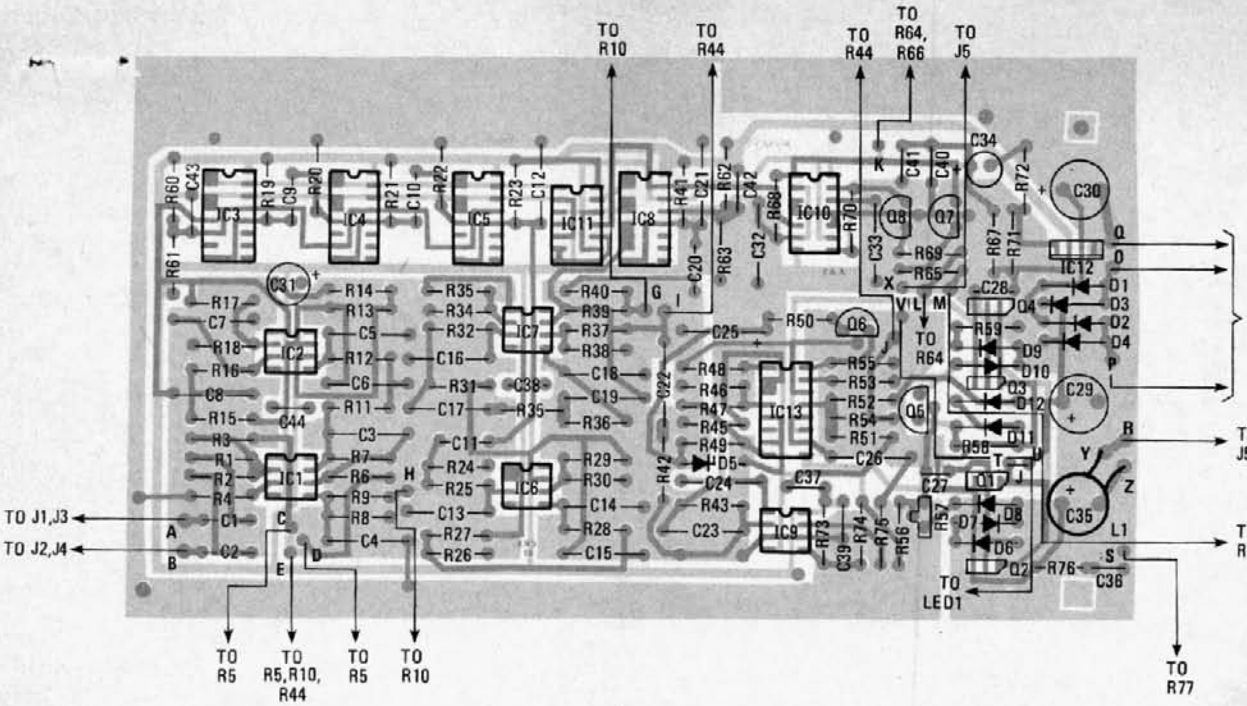


FIG. 10—DOUBLE-SIDED BOARD should be trimmed so that foil on component side is split into two areas to separate areas of differing potential. Also, be sure transistors Q1-Q4 are positioned with beveled edges at lower right.

PARTS LIST

Resistors ¼ watt, 5%, unless otherwise noted

R1, R2, R8, R20, R22, R24, R39, R40, R47, R73, R74—100,000 ohms
R3, R45—200,000 ohms
R4—39,000 ohms
R5, R44—5000 ohms, potentiometer, audio taper
R6, R7, R11, R12, R14-R16, R18, R25, R31-R36, R38—18,000 ohms
R9, R17, R30, R43—30,000 ohms
R10—100,000 ohms, potentiometer, linear taper
R13—5600 ohms
R19, R21, R23, R41, R48—47,000 ohms
R26—24,000 ohms
R27—75,000 ohms
R28—27,000 ohms
R29, R60, R62—15,000 ohms
R37—22,000 ohms
R42, R58, R65, R69, R72, R77—3000 ohms
R46—2000 ohms
R49—100 ohms
R50—43,000 ohms
R51—620,000 ohms
R52—180,000 ohms
R53—360,000 ohms
R54—62,000 ohms
R55—470,000 ohms
R56—5000 ohms
R57—5000 ohms, trimmer potentiometer
R59—20,000 ohms
R61, R63—910 ohms
R64, R66—20,000 ohms, potentiometer, linear taper
R67, R75—200 ohms
R68, R70—300 ohms
R71—7500 ohms
R76—10 ohms

Capacitors

C1, C2, C9-C11, C20—.22 μF, 100 VDC, Mylar
C3-C8, C12-C19, C21, C23, C24, C26—.001 μF, polystyrene
C22—.01 μF, polystyrene
C25, C27, C31, C34—4.7 μF, 16 VDC, electrolytic
C28, C37-C44—.1 μF, ceramic disc
C29, C35—2,200 μF, 25 VDC, electrolytic
C30—1000 μF, 25 VDC, electrolytic
C32, C33—510 pF, ceramic disc
C36—.01 μF, 400 VDC, electrolytic

Semiconductors

D1-D4—1N4002, 100 PIV, 1 amp
D5-D12—1N914
LED1—jumbo red LED
Q1-Q4—VN46QF VMOS transistor (Siliconix)
Q5-Q8—2N4403 PNP transistor
IC1, IC2, IC6, IC7—NE5512 low-noise dual op-amp (Signetics)
IC3-IC5, IC8—TDA1022, 512-stage bucket-brigade device (Philips)
IC9—NE5532 low-noise dual op-amp (Signetics)
IC10—NE556-1 dual timer (Signetics)
IC11—CD4013 dual D flip-flop (RCA)
IC12—μA78MG adjustable voltage regulator (Fairchild)
IC13—NE5517 TCA (Signetics)
L1—10 turns of No. 22 wire wound around C35
T1—36 VCT, 300 mA

Miscellaneous: PC board (double-sided with plated-through holes), case, hardware, etc.

NOTE: The following are available from Advanced Analog Systems, Inc., 790 Lucerne Dr., Sunnyvale, CA 94086 (Tel. 408-730-9786): ARS-911—complete kit including case, $149.95; PC-911—PC board only, $24.00; IC-911—IC1-IC13 and Q1-Q8 only $49.95. Visa and Mastercard welcome. California residents please add sales tax. Prices include shipping (within continental U.S. only).

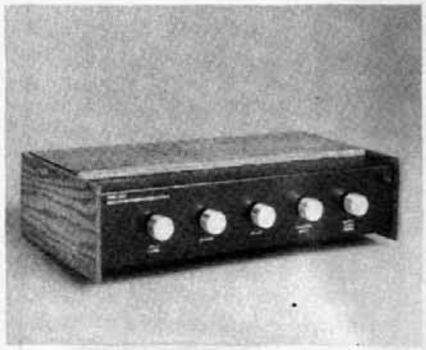


FIG. 11—THE COMPLETED REVERB unit. Enclosure shown is included with kit available from supplier listed in Parts List.

recreate the feeling of a large hall, but instead make any system-noise or distortion much more obvious. We found that disconnecting the tweeter and operating the woofer over the full range gave impressive performance.

Generally, a single full-range speaker is adequate for the reverberation system. Better still, an array of speakers will help improve the "spaciousness" of the reverberation. As long as there are no gross frequency-response irregularities, the characteristics of most speakers are generally no worse than the frequency-response variations found in actual concert halls. Those variations are caused by the resonances of the reflecting walls and ceilings in the hall, and the frequency-dependent sound-absorption properties of those walls and ceilings.

One major problem that you may have initially is amplifier-overload. It's rather obvious that you won't get rock-

continued on page 95

ANALOG REVERB

continued from page 47

concert levels from the rear speaker with just one watt of power. However, the level of the reverberation should be 10 to 20 dB lower than the level of the front channel. That corresponds to a difference of 5 to 50 watts. Furthermore, the distortion in the system that's caused by the rear (delayed) channel appears to be 10 to 20 dB lower than actually measured because the music from the louder front-channels serve to mask that distortion.

The reverberation effect is not obvious as the reverberation or output levels are gradually increased. It's only when the reverberation decreases or disappears that you really notice it. The effect should be subliminal—you should not be able to hear the reverberation unless you really listen for it, but your mind will always know it is there. There will be a "fullness," without an increase in volume, that is deceiving. You'll often find that you are listening to your stereo system at a lower volume level than before simply because the music no longer needs to be loud just to full a room with sound. **R-E**

analogue reverberation unit

Until comparatively recently the only audio delay units that were within the budget of most home constructors were of the spring line type, which suffer from a number of disadvantages such as fixed delay time, uneven and limited frequency response, and susceptibility to mechanical vibration. Recently, however, completely electronic delays have become a feasible proposition, with the result that high-quality reverberation and other audio effects are now within economic reach of the amateur. A design for a digital reverberation unit has already been published in Elektor. The circuit published here represents an alternative approach using analogue techniques.

As explained in the article on the digital reverberation unit (Elektor 37, May 1978) a digital delay line is an elegant method of producing reverberation and other time-related audio effects. In a nutshell, the analogue input signal is converted to a digital code using an A/D converter. This code is then fed through a shift register of the desired length to produce the delay, and the analogue signal is reconstituted at the output by a D/A converter. This method has a number of advantages. Firstly, since it is a digital signal that is being passed through the shift register, the signal that comes out will be identical to that which goes in irrespective of the length of the shift register. Any noise and distortion in the retrieved analogue signal are due only to deficiencies in the A/D and D/A conversion processes.

Secondly, once the initial investment in A/D and D/A converters has been made, the digital delay line can be extended to any length, simply by the addition of inexpensive digital shift registers. These two factors make the digital delay line an ideal choice for long delays such as those required for echo effects.

An alternative approach to a digital delay line is an analogue delay line using analogue shift registers (bucket brigade memories) such as those used in the Phasing and Vibrato Unit (Elektor No. 20, December 1976). These accept an analogue signal directly and transfer it from input to output as a sequence of charge packets, of which more later. Analogue shift registers are an attractive proposition for short delay times, since the cost of a 1024-bit analogue shift register (between £12 and £18) is less than the cost of an equivalent digital shift register plus A/D and D/A converter. Furthermore, the analogue shift register does not suffer from 'quantisation noise' which is inherent in the A/D conversion process.

The analogue shift register is thus ideal for producing effects such as phasing, flanging and vibrato and for the modest reverberation times required for enhancement of room ambience. However, the analogue shift register is not such an attractive proposition for longer delay times, since noise and distortion increase as the analogue shift register is made longer.

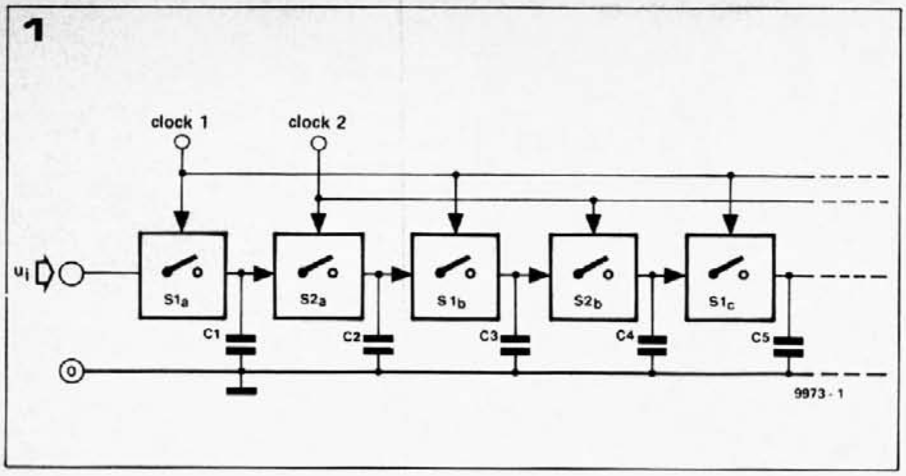
Analogue shift registers

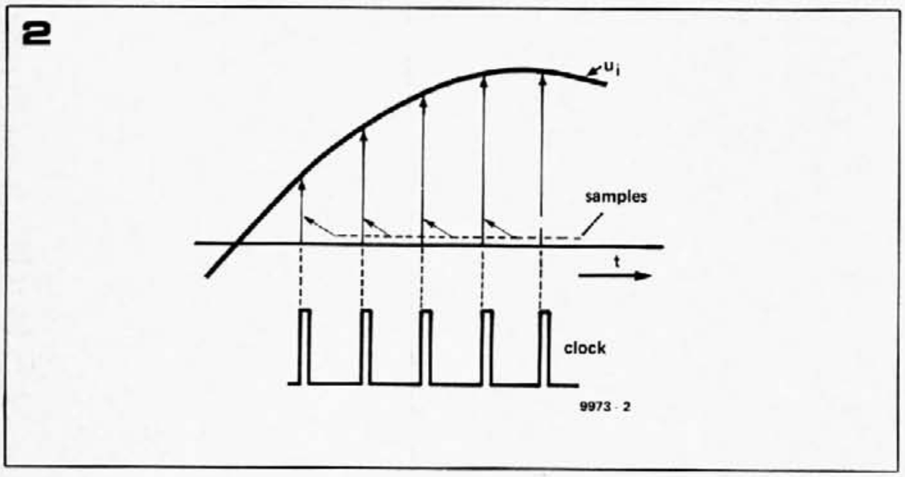
Analogue shift registers are commonly referred to as 'bucket-brigade memories', since their operation is analogous to that of a chain of men passing buckets of water from hand to hand, the 'buckets' being capacitors and the 'water' being electric charge.

The basic principle of an analogue shift register is illustrated in figure 1. It consists of a number of capacitors and (electronic) switches. The switches are opened and closed alternately by a two phase clock generator, i.e. an oscillator which generates two squarewaves in antiphase. When S1a, b, c etc. are closed then S2a, b, c, etc. are open, and vice versa. The input signal is applied to S1a. When this switch is closed then

Specification

Signal-to-noise ratio at maximum output level:	> 60 dB
Bandwidth of reverberation signal:	2.5 kHz, 5 kHz or 15 kHz (see text)
Maximum delay time:	200 ms, 100 ms or 33 ms (see text)
Monitor output bandwidth:	25 Hz to 100 kHz.
Input sensitivity:	variable; most sensitive setting gives maximum output for 30 mV RMS (100 mV peak-to-peak) input.
Maximum output:	2.5 V peak-to-peak
External clock input:	15 V p-p, 5 kHz to 500 kHz.
Supply voltage:	+15 V/75 mA, −15 V/25 mA





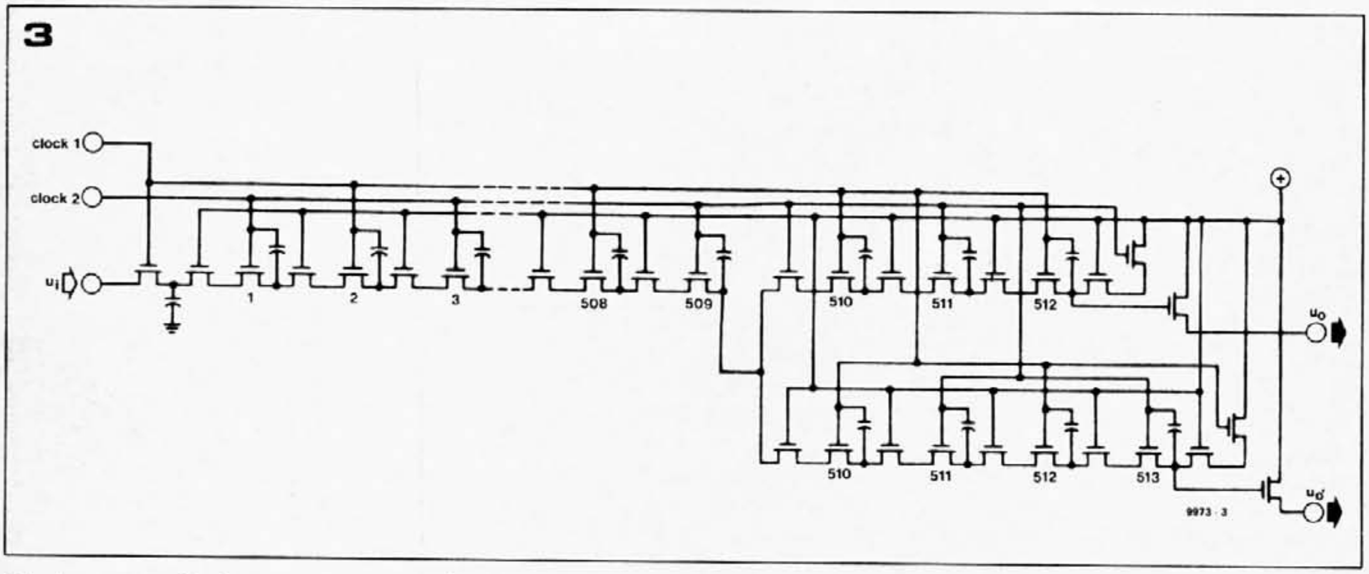


Figure 1. Illustrating the principle of the bucket-brigade memory.

Figure 2. The analogue input signal is sampled during each clock pulse.

Figure 3. Internal circuit of an analogue shift register, the Reticon SAD 1024.

C1 charges to the instantaneous value of the input signal, i.e. the input signal is sampled.
When S1 opens and S2 closes then some of the charge on C1 is transferred to C2 via S2a. When S1 again closes C1 takes a new sample of the signal, whilst C2 transfers some charge to C3 via S1b and so on.
In this way a number of samples are taken at various points along the input waveform as shown in figure 2, and these are transferred through the shift register as a sequence of charge packets. The actual operation of an analogue shift register is somewhat more complex than this simple explanation would suggest, but the basic principle involved is that described above. In a practical shift register IC the switches are MOSFETS and the capacitors are also fabricated on the chip. An abridged internal circuit of an analogue shift register is given in figure 3.
The output signal of the shift register will appear as a series of pulses synchronous with the clock signal, whose envelope follows that of the original input signal. The original signal can be recovered quite simply by lowpass filtering to remove the clock frequency component. The sampling theorem tells us that the clock frequency must be twice the maximum signal frequency. In fact, it is fairly obvious that the clock frequency must be greater than the maximum signal frequency, otherwise it will be impossible to filter it out. Furthermore, if the limitations imposed by the sampling theorem are not observed, an objectionable effect known as 'foldover distortion' can occur. This is caused by the signal and clock frequencies interacting to produce spurious products within the audio spectrum, which can occur even if the clock frequency is above the audio range and therefore inaudible.
The delay time produced by a bucket-brigade memory is dependent upon two

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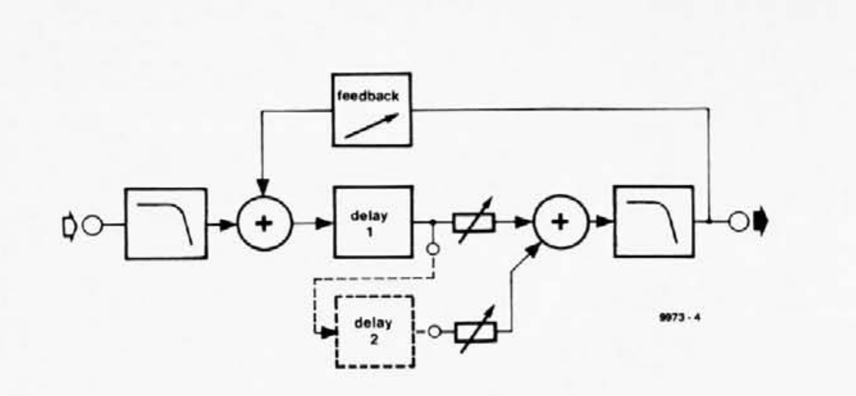
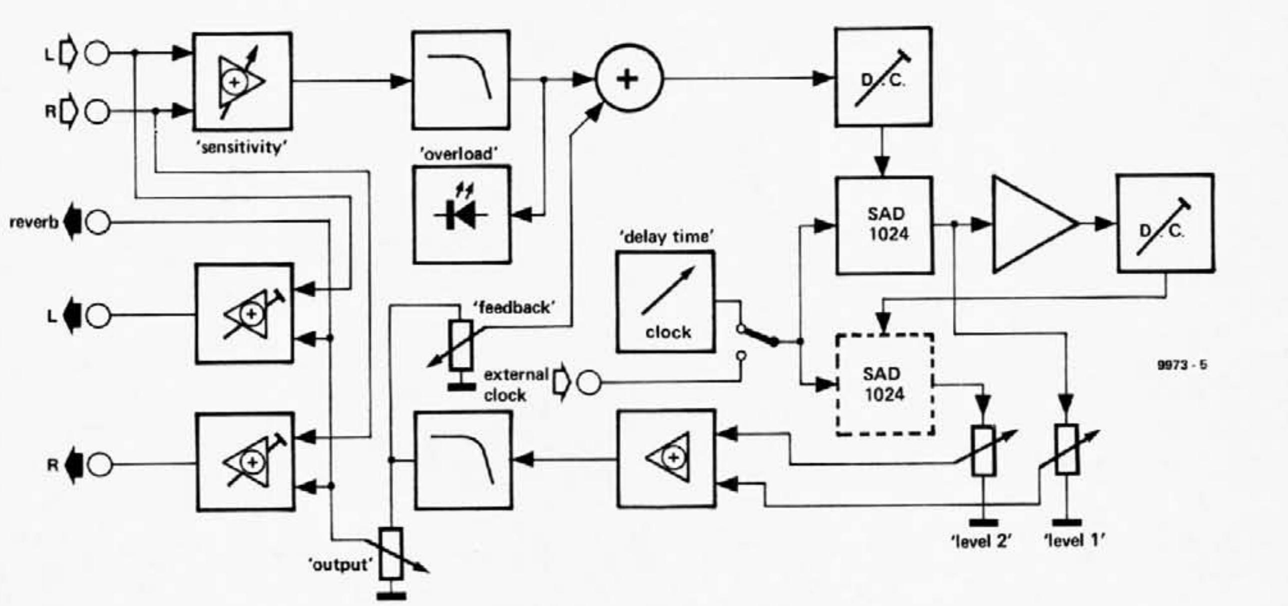


Figure 4. Principle of a reverberation unit. The input signal is delayed be feeding it through the bucket-brigade memory and a portion of the delayed signal is fed back to the input. Lowpass filters at the input and output limit the bandwidth of the signal to avoid foldover distortion and remove clock frequency components.

Figure 5. A more detailed block diagram of the analogue reverb unit.

Figure 6. Complete circuit of the reverb unit. Extensive use is made of FET op-amps.

5



factors, the number of stages in the memory and the clock frequency. Since the signal is shifted through two stages for each clock pulse it is apparent that the delay time is $t = \frac{n}{2 \cdot f_c}$.

where n is the number of stages and f_c is the clock frequency.

Since the clock frequency must be at least twice the maximum signal frequency, it follows that the maximum delay obtainable is $t = \frac{n}{4 \cdot f_{s(max)}}$.

In other words a compromise must be adopted between delay time and signal bandwidth. Increase one and the other must be decreased. In practice this means that the bandwidth of the reverb signal must be limited to somewhat less than the full audio bandwidth, if adequate delay times are to be obtained with reasonably short shift registers. This means band limiting the input signal using a lowpass filter at the input of the memory to prevent foldover distortion.

Reverberation unit

The basis of the reverberation unit is shown in figure 4. The input signal is fed through a lowpass filter and thence through the bucket-brigade memory. An attenuated portion of the delayed signal is fed back and summed with the input signal. Each time the delayed signal goes round the loop it is attenuated further and so gradually decays, thus giving rise to the characteristic reverberation effect. For longer delays a second memory may be added as an optional extra.

The SAD 1024

The analogue shift register chosen for the reverberation unit is the Reticon SAD 1024. This IC contains two, completely independent 512-stage bucket-brigade memories, which may be used separately or together. The compromise chosen between the delay time and maximum signal frequency was 100 ms and 2.5 kHz. With a 1024 stage memory and a bandwidth of 2.5 kHz it should theoretically be possible to achieve a delay of 102.4 ms at a clock frequency of 5 kHz. However, in practice the clock frequency must be set slightly higher than that demanded by the sampling theorem in order that it can be filtered out without attenuating the highest signal frequency. Even so, the output lowpass filter must have an exceedingly sharp cutoff and an astounding 48 dB/octave is used in this design.

A signal bandwidth of 2.5 kHz may seem rather small, but in fact it is quite adequate for a convincing reverb effect. For those who require a longer delay

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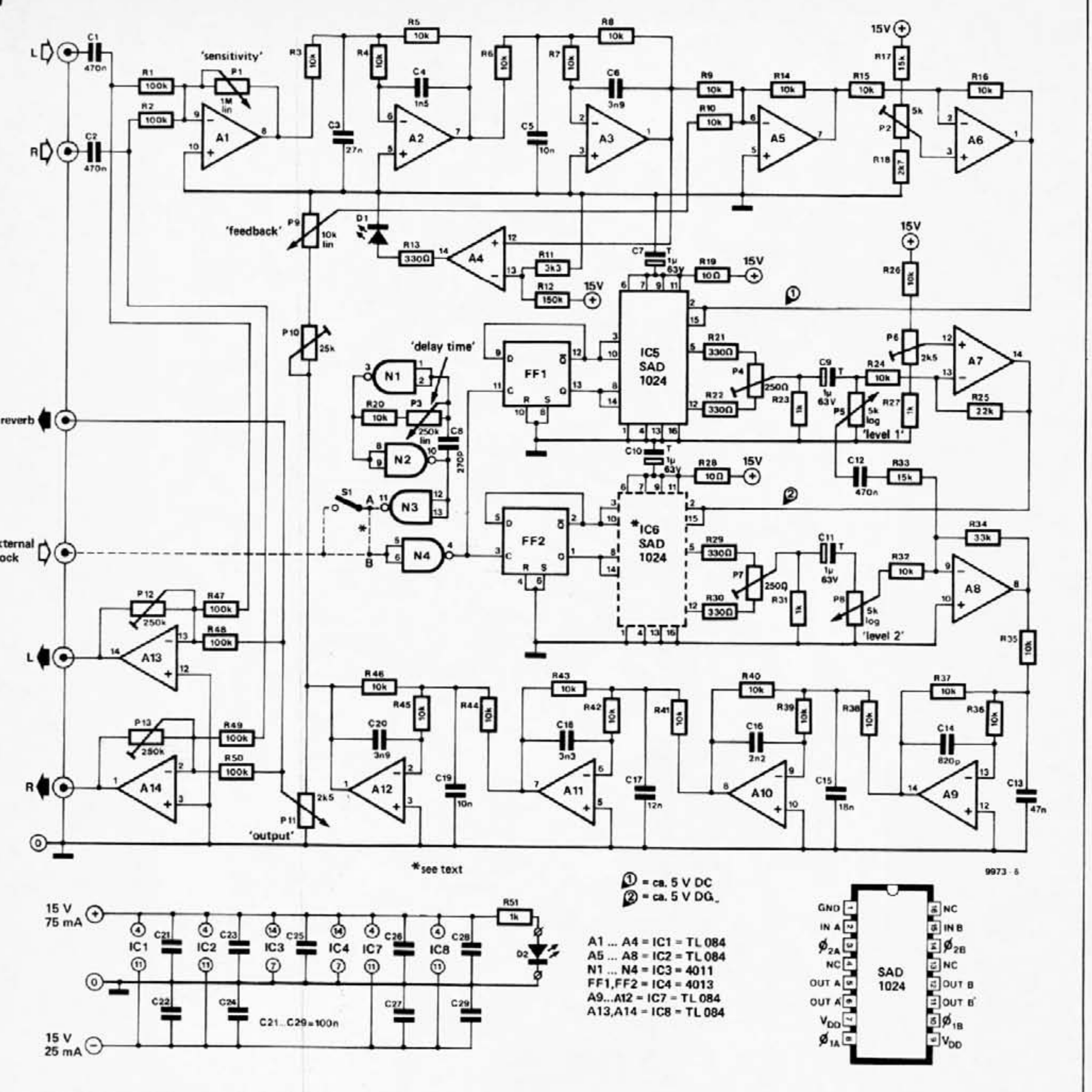

time or a wider bandwidth there is the option of adding a second SAD 1024 and/or raising the clock frequency.

Since the SAD 1024 has two sections of 512 stages the question arises of how to connect them to give a 1024 stage delay. It is, of course, possible to connect them in cascade, but this would not give the optimum signal-to-noise ratio and distortion, since passing through an additional 512 stages would further degrade the signal. The question of clock suppression also arises. With a clock frequency only slightly more than twice the maximum signal frequency it is not possible completely to filter out the clock component, even with a very steep cut filter. The solution to both these problems is to operate the two sections of the memory in 'parallel multiplex'. This means feeding the input signal to the parallel-connected inputs of both sections of the memory whilst clocking the two sections in antiphase, the result being that the signal is sampled twice per clock pulse, alternately by each shift register. The outputs of the two memory sections are then mixed, with the result that the clock frequency components, which are in antiphase, tend to cancel.

The clock cancellation effect can, of course, be achieved by summing the outputs of the last two stages of a single memory section, since these too are in antiphase. This was done in the Phasing and Vibrato Unit, which used a bucket-brigade memory with only one section.

It may appear that parallel multiplexing provides only a 512-stage delay. This is indeed the case; however, since the signal is sampled twice per clock pulse the sampling rate is actually twice the clock frequency. The clock frequency can therefore be lowered to 2.5 kHz whilst still achieving a 5 kHz sampling rate. This combination of a parallel-multiplexed 512-stage delay and 2.5 kHz clock of course gives the same delay as a 1024-stage memory (two cascaded 512-stage registers) and a 5 kHz clock.

Block diagram

A more detailed block diagram of the reverberation unit is given in figure 5, which is a stereo version of the reverberation unit. The left and right input signals are summed in a variable gain mixing amplifier and the resulting mono signal is fed to the input lowpass filter, which removes all frequencies above 2.5 kHz. The output signal from the filter is then fed to an offset circuit which sets the DC bias at the input of the SAD 1024. This is necessary as the SAD 1024 will only accept positive input signals, so the symmetrical AC

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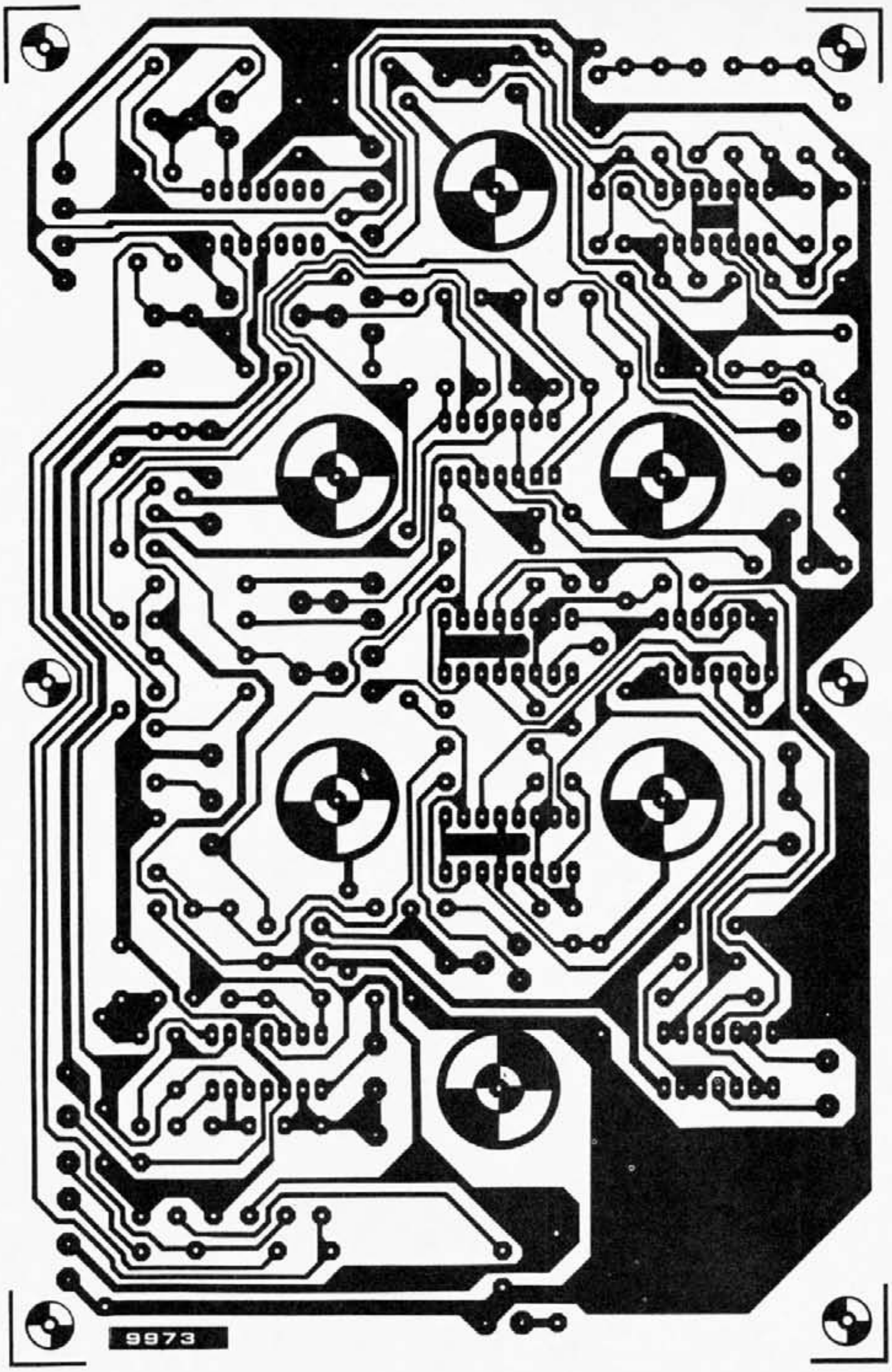
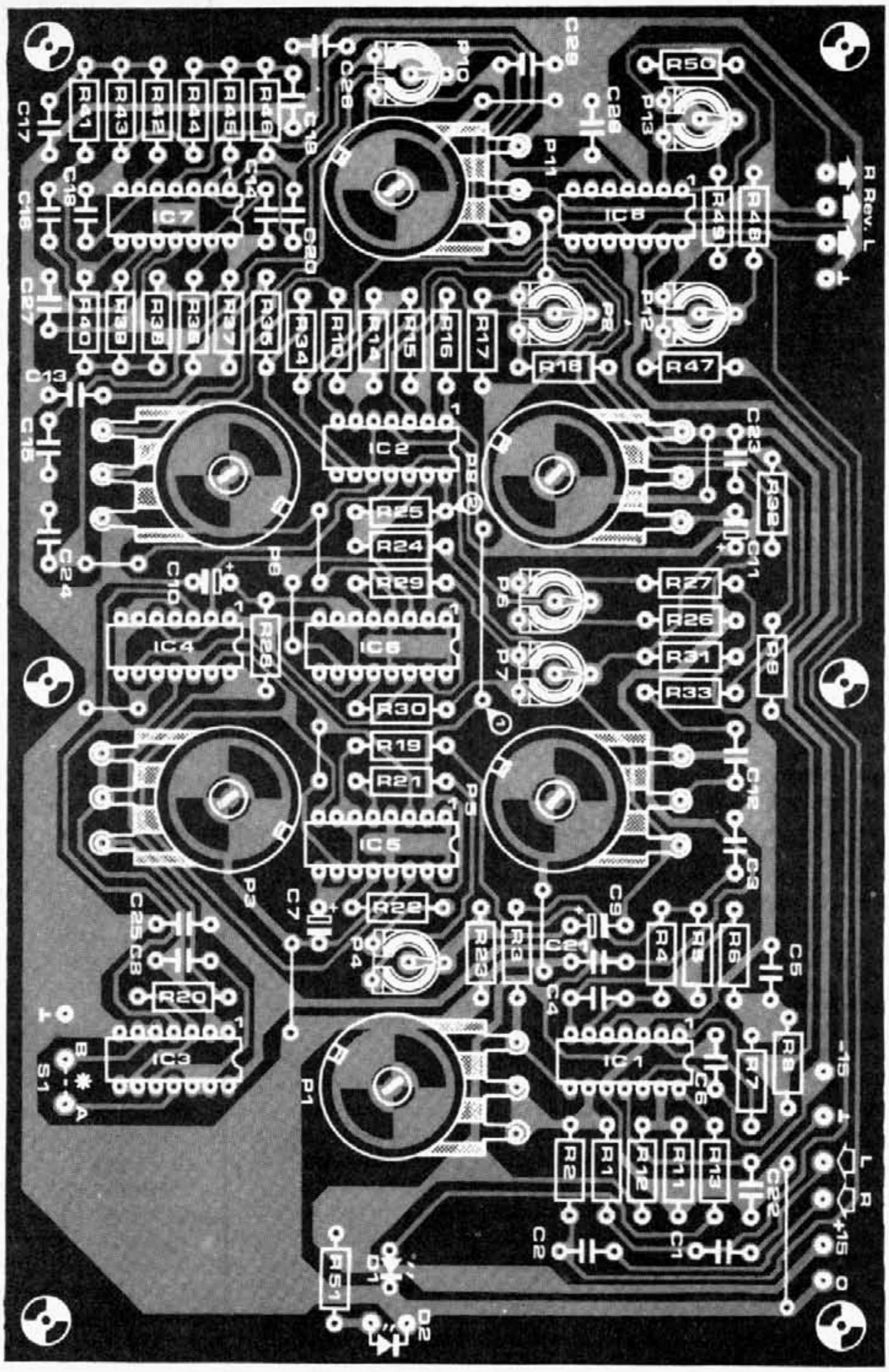


Figure 7. Printed circuit board and component layout for the reverberation unit (EPS 9973).

output of the lowpass filter must be offset by adding a positive DC bias.
The signal is then fed through the first SAD 1024. If a second SAD 1024 is to be used then the output of the first IC is fed through an amplifier to make up for the attenuation of the first IC. The outputs of both SAD 1024s are equipped with level controls.
The output of the two memories are fed to a mixer and thence to the output lowpass filter. Part of the output signal from this filter is fed back to the input of the first SAD 1024 via a feedback level control which determines the reverberation time. The remainder of the signal is mixed equally with both the original (undelayed) left and right signals, so that it appears as a mono image when fed to left and right loudspeakers. A separate output for the reverb signal only is provided. The output control varies the proportion of reverb signal in the output signal.
It may seem a little odd to add a mono reverb signal to a stereo signal, but in fact this simulates what happens in, say, a concert hall. Reverberation is the result of multiple reflections from the walls of the room and therefore conveys no directional information, i.e. it is mono. It appears more or less equally at both ears of the listener, superimposed on the direct left and right sounds.



parts list

Resistors:

R1,R2,R47 . . . R50 = 100 k
R3 . . . R10,R14,R15,R16,R20, R24,R26,R32, R35 . . . R46 = 10 k
R11,R21,R22,R29,R30 = 330 Ω
R12 = 150 k
R13 = 330 Ω
R17,R33 = 15 k
R18 = 2k7
R19,R28 = 10 Ω
R23,R27,R31,R51 = 1 k
R25 = 22 k
R34 = 33 k

P1 = 1 M linear potentiometer
P2 = 5 k (4k7) preset potentiometer
P3 = 250 k (220 k) linear potentiometer
P4,P7 = 250 Ω (220 Ω) preset potentiometer
P5,P8 = 5 k (4k7) logarithmic potentiometer
P6 = 2k5 (2k2) preset potentiometer
P9 = 10 k linear potentiometer
P10 = 25 k (22 k) preset potentiometer
P11 = 2k2 (2k2) logarithmic potentiometer
P12,P13 = 250 k (220 k) preset potentiometer

Capacitors:

C1,C2,C12 = 470 n
C3* = 27 n
C4* = 1n5
C5*,C19* = 10 n
C6*,C20* = 3n9
C7,C9,C10,C11 = 1 μ (tantalum)
C8* = 270 p
C13* = 47 n
C14* = 820 p
C15* = 18 n
C16* = 2n2
C17* = 12 n
C18* = 3n3
C21 . . . C29 = 100 n

Semiconductors:

IC1,IC2,IC7,IC8 = TL 1084
IC3 = 4011
IC4 = 4013
IC5,IC6 = SAD 1024
D1 = LED (red)
D2 = LED (green)

* see text and table 1

There is nothing to be gained by having completely separate reverb channels for left- and right signals.

Complete circuit

Figure 6 is the complete circuit of the reverberation unit. The input signals are summed by op-amp A1, whose gain can be varied by P1. The output signal is then fed to the input lowpass filter comprising A2 and A3, which consists of two, cascaded, second-order Butterworth filters, giving a total slope of 24 dB/octave above the cutoff frequency of 2.5 kHz. Since there is no clock frequency component to remove the slope of this filter is only half that of the output filter.
The output signal from A3 is summed with the feedback signal from the bucket-brigade memory by A5. The output of A3 is also fed to peak overload indicator A4. When the voltage on the non-inverting input of A4 exceeds that set on the inverting input by R11 and R12, the output of A4 will go positive and D1 will light.
The signal from the output of A5 is fed to A6, which is a unity-gain inverting amplifier with a variable DC offset at the non-inverting input. P2 is used to set the quiescent output voltage of A6 and hence the DC bias at the input of the first SAD 1024, IC5.
The output of IC5 is fed via P5, the 'level 1' control, to the input of A8, and thence to the output lowpass filter, A9 to A12. This consists of four, cascaded, second-order Butterworth filters and has a slope of 48 dB/octave.
The output of the filter is fed to P5, the reverb output control, which varies the proportion of reverberation in the final output signal. The reverb signal is mixed with the left and right direct signals in A13 and A14, the gain of these amplifiers being adjustable by P12 and P13 to suit the output level required by succeeding equipment.
If a second SAD 1024 (IC6) is included in the circuit then the output of IC5 is also fed to it via a second DC offset circuit, A7. This stage also has a gain of two to compensate for the attenuation introduced by IC5. The output of IC6 is fed, via the 'level 2' control P8, to the input of A8 and thence to the output filter.
The clock generator is an astable multivibrator based on two CMOS NAND gates N1 and N2. The clock output is buffered by the two remaining gates in the 4011 IC and is then fed to two flip-flops FF1 and FF2, whose Q and $\overline{Q}$ outputs provide the two-phase clock for IC5 and IC6. The actual clock frequency fed to IC5 and IC6 is half the clock generator frequency since FF1 and FF2 function as divide-by-two counters.

Construction

A printed circuit board and component layout for the circuit are given in figure 7. The six main control potentiometers are mounted on the p.c.b. to simplify wiring. The whole assembly can then be mounted on spacers behind a fascia panel through which the potentiometer spindles protrude.
If the component values given in the circuit diagram are used then the filters will have a turnover frequency of 2.5 kHz. If a higher turnover frequency is required for greater signal bandwidth then table 1 should be referred to, which gives values for 5 kHz and 15 kHz turnover frequencies.

Adjustment and use

The circuit contains six control potentiometers and seven presets. P12 and P13 simply set the gain of A13 and A14, and hence the output level of the reverb unit, to suit subsequent equipment.
The adjustment procedure for the remaining presets, and the operation of the controls, is as follows. P1 should be set so that D1 just lights on the loudest passages of the input signal. The optimum signal-to-noise ratio will then be obtained without overloading the circuit. P1 should not be used as a volume control, as overloading of the circuit or a poor s/n ratio may result.
The feedback control, P9, should be turned fully anticlockwise and the output control, P11, fully clockwise, after which the reverb output should be fed to an amplifier and loudspeaker so that it is clearly audible. The level 1 control, P5, should be turned fully clockwise and the level 2 control, P8, anticlockwise and the clock frequency should be lowered until it becomes audible. The balance control, P4, should then be adjusted until the clock noise has been reduced to a minimum. This should occur with P4 approximately in its mid-position.
The D.C. bias voltage of the shift register may now be adjusted. A signal should be fed in of sufficient amplitude to cause D1 just to light up and P2 should be adjusted until there is no audible distortion. Alternatively, if an oscilloscope is available, the input level may be increased until the output signal clips and P2 can then be adjusted so that the clipping is symmetrical.
If IC6 is included then the adjustment procedure for clock nulling and offset must be repeated for this IC, using P7 and P6 respectively. In this case the level 2 control, P8, should be fully clockwise, whilst the level 1 control, P5, should be anticlockwise.
Finally the feedback preset, P10, should be adjusted to give maximum decay time. This adjustment is carried out with P5, P8 if fitted, and P9, all turned fully clockwise. P10 is then adjusted so that the reverb signal decays gradually when the input signal ceases. If P10 is incorrectly set the system will be unstable and the reverb signal will swell to a cacophony of noise. This adjustment should be repeated at several settings of the delay time control, P3.
As already mentioned, the reverb unit has three outputs: left channel plus reverb, right channel plus reverb and reverb only. The unit can be used with an existing stereo system simply by using the tape socket(s) of the amplifier. The signal to the reverb unit is taken from the tape output and the signal from the reverb unit is taken to the tape input.
Alternatively, the left and right outputs of the reverb unit need not be used. Instead the reverb output may be fed to a separate amplifier and loudspeaker. This gives a more spacious effect to the sound.
It is important to note that the clock frequency will become audible if it is set too low. If the 5 kHz or 15 kHz bandwidth option is adopted then this will obviously occur at higher settings of P3. P3 may be fitted with a frequency-calibrated scale to reduce the chance of setting the frequency too low. Alternatively, its range may be reduced by connecting 'padding' resistors in parallel with P3. These should be chosen by experiment such that with P3 set to its maximum resistance the clock frequency is inaudible.

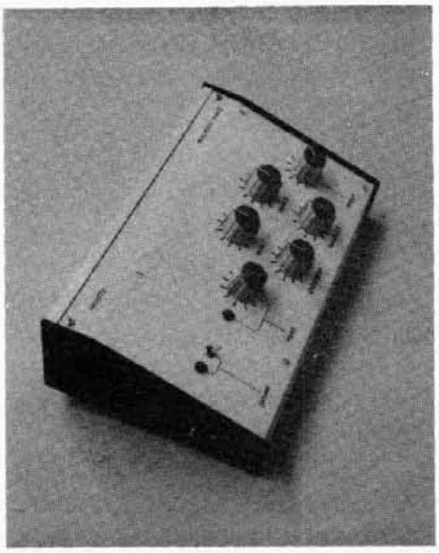


Table 1

turnover frequency (−3 dB)	5 kHz	15 kHz
C3	12 n	3n9
C4	820 p	270 p
C5	5n6	1n8
C6	1n8	680 p
C8	150 p	47 p
C13	27 n	8n2
C14	390 p	120 p
C15	8n2	2n7
C16	1n8	390 p
C17	5n6	1n8
C18	1n8	560 p
C19	4n7	1n5
C20	2n2	680 p

Digital Echo Unit

J. A. Murdie

The Digital Echo Unit described below may be constructed on standard Eurocard PCBs with 31 way connectors, and utilizes the cheap 2102 1K static RAM, of which from any amount from (say) 32-64K may be used to achieve a (continuously variable) delay of up to a second. The delay time is of course directly proportional to the amount of memory used. There are three PCB designs used: Fig. 1: Input/Clock board (1 off), Fig. 2: Output/Control board (1 off), Fig. 3: 8K Memory Board (max. 8 off).

Dealing with the input board first, it may be seen that the 555, 7476 and 7408 constitute a non-overlapping two phase clock whose outputs are 'Enable Read' (ER), and 'Enable Write' (EW). During the write phase a bit is taken from the digitized input and fed to the 'Data Write' (DW) line. The AD convertor used is the FX209 which was featured in the ETI June 1976 Data Sheet. The bits created are placed in the memory location addressed by the 12 bit counter ('Bit Address'). on this board and the 4 bit counter on the Output/Control board ('Block Address').

When the ER line goes high a bit is taken from the memory address pointed to by the counters with the 4 bit value produced by the Hexadecimal Priority encoder (Delay Switches) being added to the block address. Thus the 'distance' between the write and read 'heads' may be altered to place them any number of blocks apart, and thus create a choice of 16 basic delay lengths. The bit read is placed on the DR line and is then converted to an analog value by the DA convertor. Note that some of the output may be fed back to the input ('Regen') to create multiple echo effects.

After this sequence of a write and a read cycle the bit/block address is incremented by one so a succession of bits may be placed in memory by input, and read from the memory by the output. The rate at which this sequence occurs is controlled by the clock rate of the 555 astable, and thus this not only controls the delay time as do the delay switches, but also the quality of the sound reproduced as this independent on the number of samples taken per second in the digitizing process. The device may be set up to digitize the analog input at a maximum of 125 K bits/second — which is quite adequate for (say) an electric guitar which requires a bandwidth of some 10 KHz.

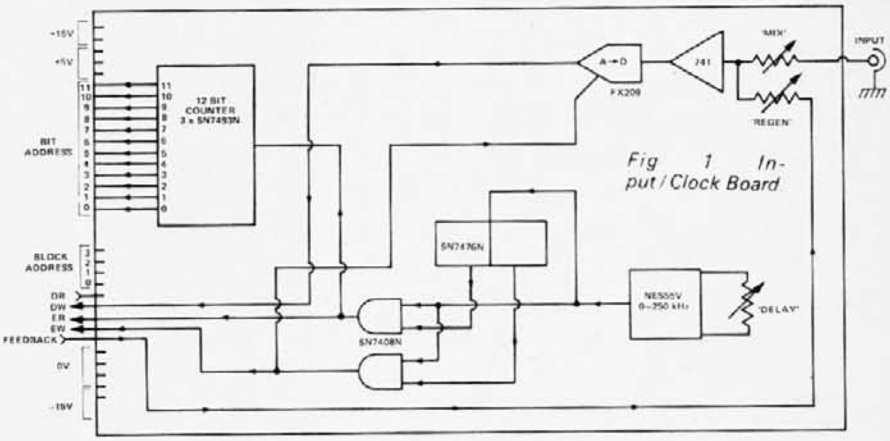


Fig 1 Input/Clock Board

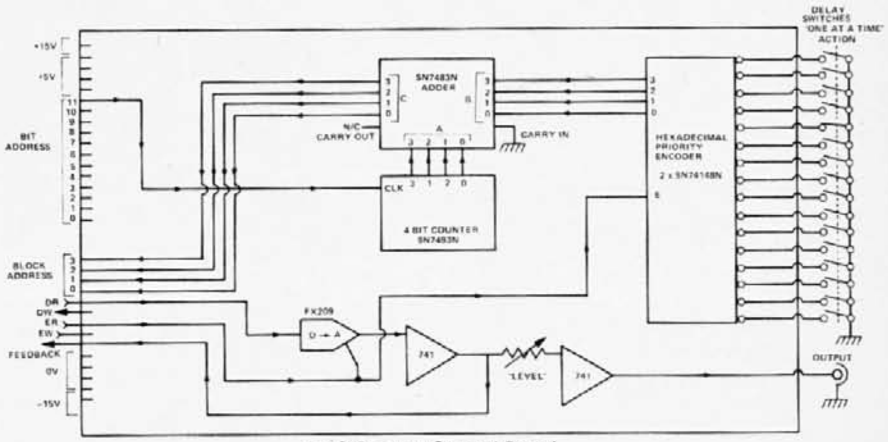


Fig 2 Output/Control Board.

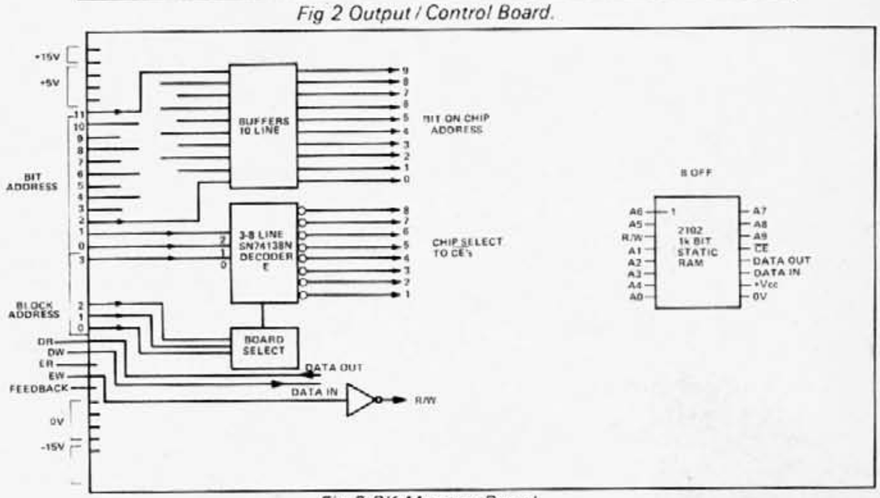


Fig 3 8K Memory Board.

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Popular Electronics
JUNE 1976

BY JOHN H. ROBERTS

THE "BUCKET BRIGADE" AUDIO DELAY LINE

Allows user to simulate larger listening room. Also used by recordists and musicians for special sound effects.

THE SOUND of recorded music being played is a listening experience that changes according to the room you are in. If the room is too "live" or too "dead", the sound appears to be unnatural. When the room has an ultra-modern decor and lots of glass window areas, the effect on the music is "bouncey." With heavy drapes, carpeting, and thickly padded furniture, plus a minimum of hard surfaces, the effect approaches that of an anechoic chamber—with very little sound reflection.

For the latter, you can either throw away your sofa pillows and pull down the drapes, or you can add a time-delay device to your audio system to create a more natural ambience. Since you may not care to redecorate, you can create an echo (audio signal time delay) and reverberation (later reflections) and achieve a livelier sound.

Until recently, the only means of obtaining an audio signal delay has been through the use of very expensive electronic equipment. Now there is a new type of IC—the "bucket brigade"—and you can build your own delay system for as little as $39 in mono and $59 in stereo. Connected between source and preamp or preamp and power amplifier (at the tape monitoring jacks possibly), it provides an adjustable, signal echo that can enhance the sound in most home listening rooms. With minor connection changes, it also can be used as a phasor/flanger, giving you a sound effect for tape recording purposes and electric-guitar playing used by the professionals.

The bucket-brigade IC is a MOS-type shift register that contains two 512-stage registers in a single 14-pin package. When an audio signal is applied to the input of the bucket brigade and a clock generator drives the IC, the signal is stepped along stage by stage until it comes out delayed a discrete interval in time. By adding this delayed signal to the original, reverberation is simulated.

In addition to providing real-time ambience, the bucket-brigade circuit can be used with a tape recorder to provide simulated stereo sound from mono sources, a means for "double voicing," and "phasor/flanging."

Technical Details. If you can delay an audio signal, you can create a number of useful sound effects. The most obvious is simulating echo, though delays provided by the bucket

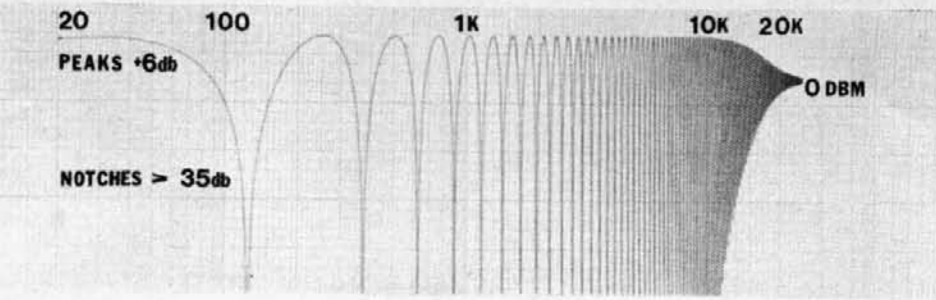


Fig. 1. Frequency between notches on a comb filter is adjusted by varying the clock frequency.

brigade are too short to be discerned as discrete echoes. Recirculating the delayed signal at reduced gain can approximate the natural decay of echoes in a reverberant room. By adding some gain during the recirculation of the delayed signal, you can create an unnatural "door-spring" effect on the music.

Delay an instrument or voice track by 30 or 40 ms and add the delayed signal back to the original signal, and you will make the output sound fuller and give it the effect of more than the original number of voices or instruments. This commonly used technique is known as "double voicing."

Another popular short-delay effect is a strange sound that results from a technique known as "phasing" or "reel flanging." The name is derived from its original implementation where a tape recorder was used to create the time delay and the friction of a well-placed hand on the outside edge of the tape-feed reel varied the delay to produce the acoustic effect. This effect can be created totally by electronic means by delaying the signal 0.5 to 5 ms while adding or subtracting the delayed signal from the original signal.

In the phasor/flanger mode, the frequency and its multiples whose wavelengths are equal to the time delay will be completely cancelled out while all other frequencies are reinforced. The result is a comb filter whose frequency between the notches is adjusted by varying the clock frequency (Fig. 1). In this manner, a tonal quality can be imparted to nontonal sound such as drums, cymbals, and even voices.

The phasor/flanger mode can be used to simulate stereophonic sound from a monophonic source. To do this, the phased output derived by adding the delayed signal goes to one channel, while the output derived by subtracting the delayed signal goes to the other. To the listener, the phasing effect cancels leaving a reasonable pseudo-stereo effect.

The basic block diagrams of the delay-line and phasor/flanger circuits are shown in Fig. 2. The hearts of the circuits, of course, are the bucket-brigade IC's, which can directly process analog signals. The circuits do not require costly analog-to-digital and digital-to-analog converters. When the clock pulse from the flip-flop is applied to the bucket-brigade IC, the dc voltage present at the input is shifted into the register. The discrete bits are transferred stage by stage with successive clock pulses until, after 256 pulses, they reach the end of the line and provide the output.

The output waveform is smoothed by a low-pass filter and duplicates whatever signal was present at the input but delayed in time by 256 times the period of the clock frequency. (Period is equal to the reciprocal of the frequency.) For example, if the clock frequency is 100,000 Hz, the delay would be $256 \times 1/100{,}000 = 2.56$ ms.

Since the audio signal at the input is being sampled at a rate determined by the clock frequency, a theoretical limit of half the clock frequency is the highest audio frequency that can be reliably passed. However, owing to practical limitations, a third of the clock frequency is a more reasonable design goal. Circuits can be cascaded to provide longer time delays at high clock rates, but the increase in noise in the series-connected circuits might outweigh the increase in bandwidth.

In the delay mode, the two shift registers are connected in series, which allows twice the clock frequency to be used. Therefore, twice the bandwidth of a single shift register can be programmed for the same time delay. Even in this double-bandwidth mode, the clock frequency required for a

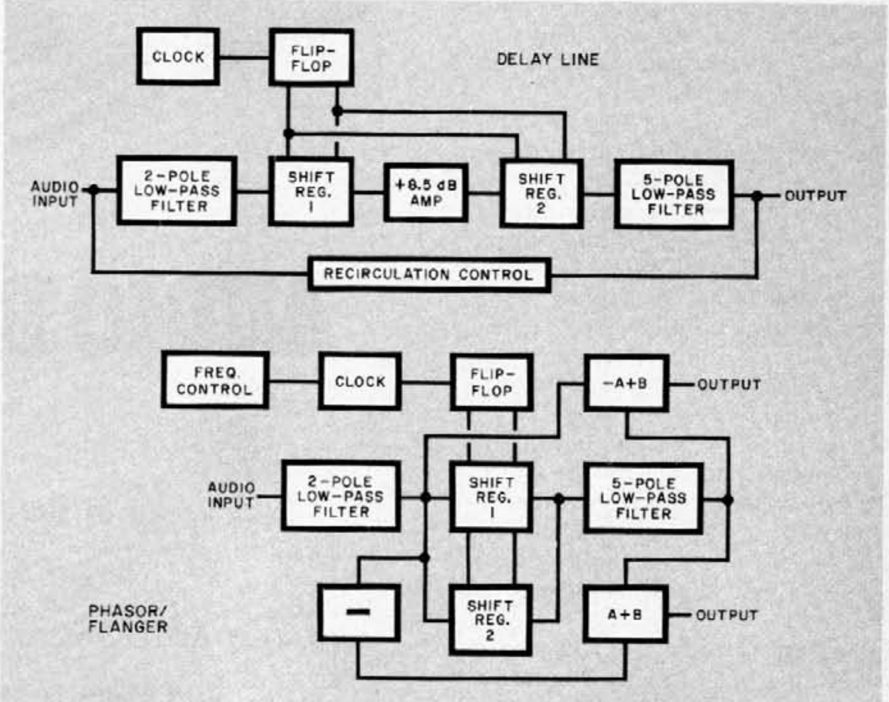


Fig. 2. Basic block diagrams of the delay line and the phasor/flanger circuits.

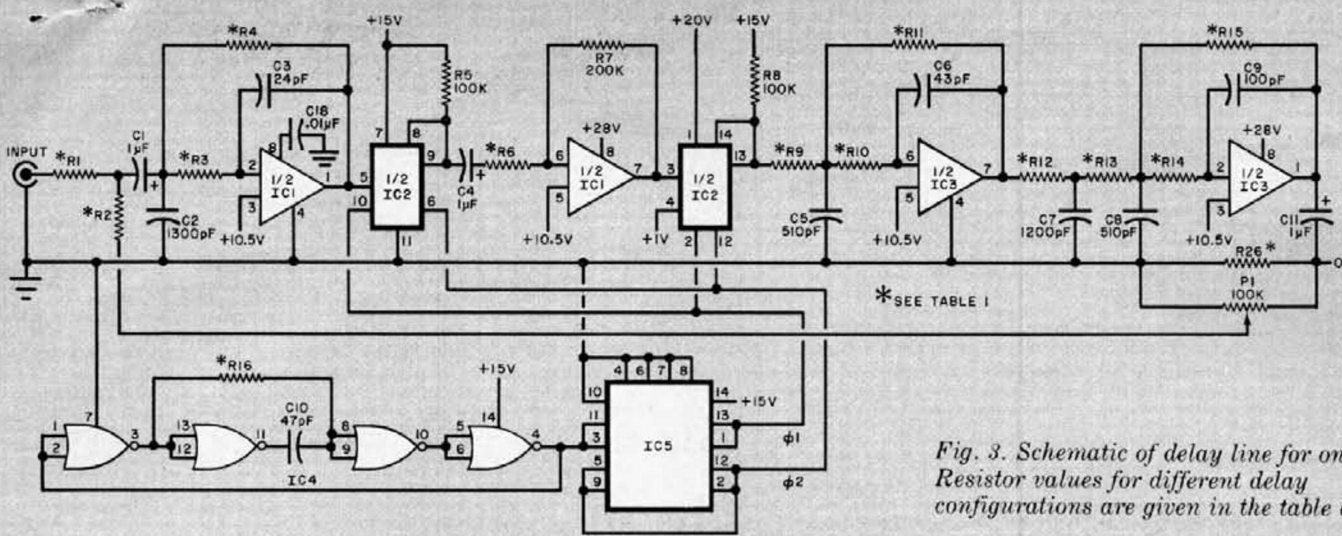


Fig. 3. Schematic of delay line for one channel. Resistor values for different delay configurations are given in the table below left.

TABLE OF FILTER RESITOR VALUES

(all values in kilohms)

	A	B	C	D
R1	100	200	300	390
R2	130	270	390	510
R3	36	75	110	150
R4	100	200	300	390
R6	75	75	75	75
R9	47	91	130	180
R10	43	82	130	160
R11	120	240	360	470
R12	10	20	30	39
R13	56	110	160	220
R14	33	68	100	130
R15	68	100	200	270
R16	110	240	360	470
R26	200	200	200	200

A = 10 ms or less, −3 dB at 15,000 Hz
B = 20 ms or less, −3 dB at 7500 Hz
C = 30 ms or less, −3 dB at 5000 Hz
D = 40 ms or less, −3 dB at 3800 Hz

PARTS LIST FOR FIG. 3

C1,C4,C11—1-µF, 25-volt electrolytic capacitor
The following are 5% polystyrene capacitors:
C2—1300 pF
C3—24 pF
C5,C8—510 pF
C6—43 pF
C7—1200 pF
C9—100 pF
C10—47 pF
C18—0.01-µF ceramic disc capacitor
IC1,IC3—1458 dual operational amplifier
IC2—MN3001 dual analog shift register (Matsushita)
IC4—4001 CMOS quad NOR gate
IC5—4013 CMOS dual D flip-flop
P1—100,000-ohm potentiometer
R1 through R4,R6,R9 through R16,R26—See Table
R5,R8—100,000-ohm, ¼-watt, 5% resistor
R7—200,000-ohm, ¼-watt, 5% resistor
Note—See Parts List for Fig. 5 for kit information.

40-ms delay limits the bandwidth to a maximum input signal frequency of 3750 Hz, which is adequate for voice but less than adequate for many musical instruments. In most applications where the delayed signal is added to the original signal, the reduction in bandwidth will be masked by the high-frequency signals present in the original. To compensate for normal signal attenuation, an 8.5-dB amplifier is used between the shift registers.

In the phasor/flanger mode, the maximum delay required is about 5 ms, which is short enough that a single shift register can be used without compromising the bandwidth. The second shift register is therefore connected in parallel with the first to improve the S/N ratio. The signals are added in-phase, while the noise adds and subtracts randomly.

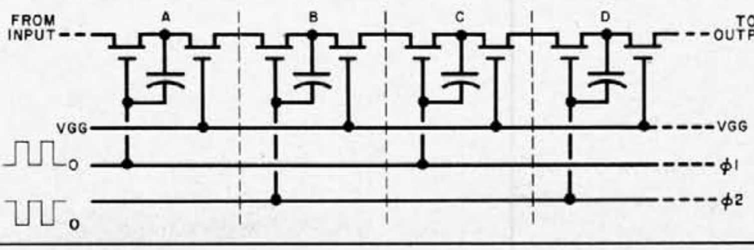
How It Works. The schematic diagrams of the delay-line and phasor/flanger configurations of the circuit

A BUCKET-BRIGADE SHIFT-REGISTER ANALOGY

The name "bucket brigade" conjures up images of a line of men passing along buckets of water to fight a fire. The bucket-brigade analog shift register operates in a similar manner, which is how it got its name. In the case of the shift register, however, the buckets are capacitors integrated right on the PMOS chip. There are more than 1000 such capacitors on each chip (one capacitor and two MOS transistors for each stage). What is being passed along are packets of electrical charge from stage to stage.

It is difficult to pour water both into and out of a bucket at the same time. So, too, it is difficult to simultaneously charge and discharge a capacitor. This problem is overcome in the shift register by utilizing two out-of-phase clocks. While the first clock is high, the "odd" buckets are dumped into the next consecutive "even" bucket. When the second clock is high, the even buckets are dumped into the next consecutive odd buckets. In this manner, individual charges are transferred along the line one stage at a time.

The drawing is a schematic representation of four typical stages of the MN3001 analog shift register. Each MN3001 IC contains two 512-stage shift registers. Note that stages A and C are connected to one clock, while stages B and D are connected to the other clock to provide the odd/even relationship.



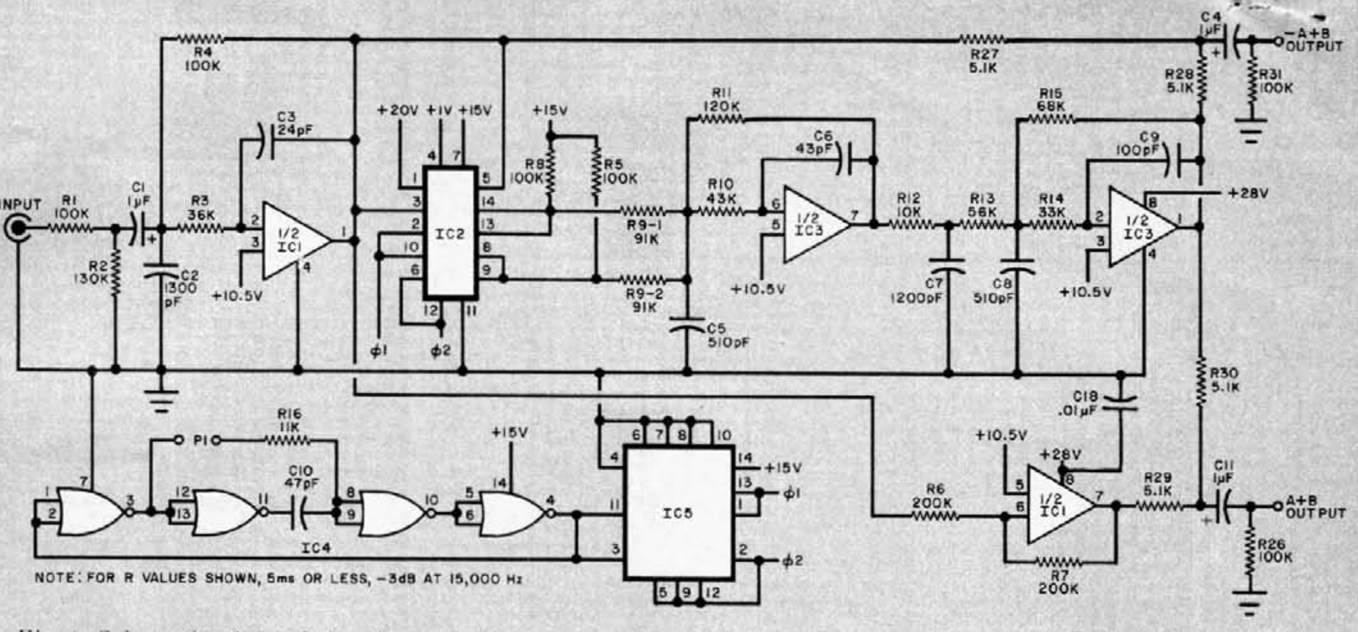


Fig. 4. Schematic of circuit for phasor/flanger.

PARTS LIST FOR FIG. 4

C1 through C11—Same as for Fig. 3
C18—0.01-µF ceramic disc capacitor
IC1 through IC5—Same as for Fig. 3
The following resistors are ¼ watt, 5% tolerance:
R1,R4,R5,R8,R26,R31—100,000 ohms
R2—130,000 ohms
R3—36,000 ohms
R6,R7—200,000 ohms
R9-1,R9-2—91,000 ohms
R10—43,000 ohms
R11—120,000 ohms
R12—10,000 ohms
R13—56,000 ohms
R14—33,000 ohms
R15—68,000 ohms
R16—11,000 ohms
R26—100,000 ohms
R27 through R30—5100 ohms
Note—See Parts List for Fig. 5 for kit information.

are shown in Fig. 3 and Fig. 4, respectively. In both cases, quad NOR gate *IC4* is wired as an astable multivibrator operating at twice the desired clock rate's frequency. The output of *IC4* goes to flip-flop *IC5*, which provides a pair of complementary (180° out of phase with each other) output clock pulses with 50% duty cycles. These pulses then "clock" the shift registers in *IC2*. Frequency determining resistor *R16* is fixed in the delay configuration, while resistance can be added via a pair of connectors to change the clock frequency in the phasor/flanger.

The audio input signal is conditioned by seven poles of low-pass filtering in which *IC3* and half of *IC1* are used. The filters provide a total of 42-dB/octave attenuation above the tuning frequency. For example, if the filter were tuned for 5000 Hz, a 10,000-Hz signal would be attenuated by more than 100:1.

When filters are designed with high-gain operational amplifiers (op amps), it is possible to have their outputs increase before rolling off at the rate of 6 dB/octave per pole. Such filters are termed "under damped." By carefully selecting the proper balance of under-damped and over-damped (RC) filter sections, it is possible to design a filter that is flat in the desired passband so that it is 3 dB down at the tuning frequency and has a roll-off rate of 6 dB times the number of poles. This is what has been done in the delay-line and phasor/flanger circuits.

Quite a bit of mathematical compu-

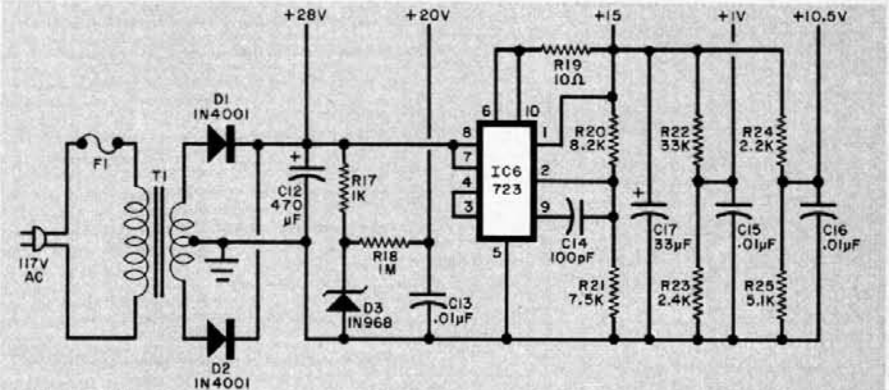


Fig. 5. Schematic of power-supply circuit. Parts List includes kit information for all circuits.

PARTS LIST FOR FIG. 5

C12—470-µF, 35-volt electrolytic capacitor
C13,C15,C16—0.01-µF disc capacitor
C14—100-pF disc capacitor
C17—33-µF, 25-volt electrolytic capacitor
D1,D2—1N4001 rectifier diode
D3—1N968 (20-volt) zener diode
F1—1/10-ampere fuse
IC6—723 precision voltage regulator
The following resistors are ¼ watt, 5% tolerance:
R17—1000 ohms
R18—1 megohm
R19—10 ohms
R20—8200 ohms
R21—7500 ohms
R22—33,000 ohms
R23—2400 ohms
R24—2200 ohms
R25—5100 ohms
T1—Power transformer with two 28-volt secondaries at 50 mA each
Misc.—Chassis; line cord; phono jacks (4); control knobs (2); rubber grommet; spacers; machine hardware; hookup wire; solder; etc.
Note: The following items are available from Phoenix Systems, P.O. Box 73, Saugatuck Sta., Westport, CT 06880: Complete kit of parts (delay line or phasor/flanger) No. P-1220-M (mono) for $39.00; complete kit of parts No. P-1220-S (stereo) for $59.00; etched and drilled pc board No. P-1220-B for $5.00; MN3001 analog shift register IC No. P-1220-C for $15.00. Add $1.00 for shipping and handling. Connecticut residents, please add sales tax.

CLAIMED SPECIFICATIONS

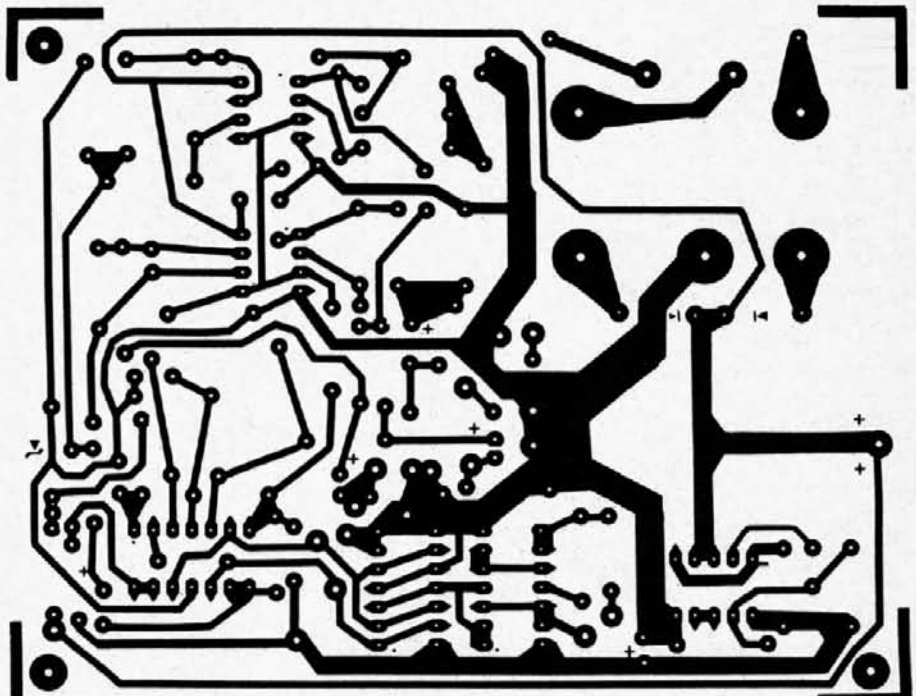
Delay Line:	
Frequency response	15 to 15,000 Hz (+2/−3 dB)
Distortion (THD)	Typically less than 1% (1000 Hz, 1 V rms)
Input impedance	Greater than 100,000 ohms
Clipping level	1.77 V rms (5 V p-p)
Signal-to-noise	Typically 50 dB below 0 dBm
Phasor/Flanger:	
Frequency response	15 to 15,000 Hz (+2/−3 dB)
Distortion (THD)	Typically less than 0.75% (1000 Hz, 1 V rms)
Input impedance	Greater than 100,000 ohms

tation is normally required to determine the values of the filter resistors to use. To simplify matters, you can select the appropriate resistor values from the Table of Filter Resistor Values. Use this Table for selecting resistor values for only the delay-line circuit. (The filter resistor values specified in Fig. 4 and its accompanying Parts List will provide an optimized 5-ms delay, with the output 3 dB down at 15,000 Hz for the phasor/flanger.)

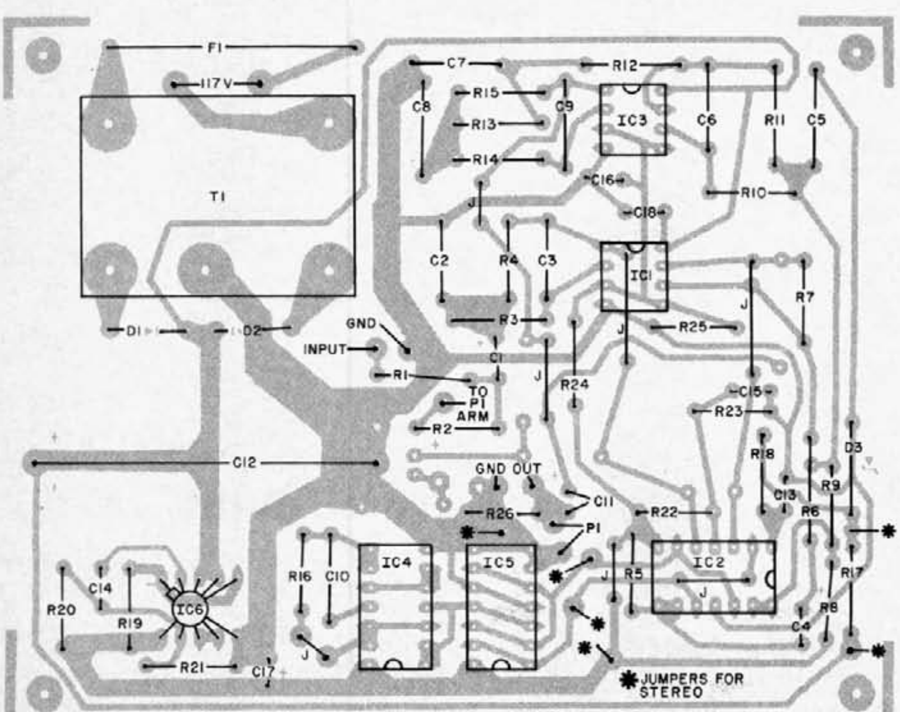
The power supply is shown in Fig. 5. It uses a voltage regulator, *IC6*, to generate the main 15-volt supply output. The shift register requires supplies of both +1 and +20 volts. The +20-volt line is obtained through the use of zener diode *D3*, while the +1-volt line is derived from the voltage divider consisting of *R22* and *R23*. Since the op amps are being operated from a single-ended supply, it is necessary to have the 10.5-volt supply line serve as the reference point in the circuit for these IC's.

Construction. The actual-size etching and drilling guide, the same for both circuit configurations but wired differently as required, is shown in Fig. 6A. The parts-placement guides for the delay-line and phasor/flanger configurations are shown in Figs. 6B and 6C, respectively.

Before installing any components on the board, mount and solder into place the wire jumpers. Then, wire the board as in Fig. 6B or Fig. 6C, depending on the desired mode of operation. Be careful to properly orient all semiconductor devices and electrolytic capacitors. Be sure to handle the MOS devices with care to prevent them from being damaged by static charges. You can mount the IC's directly on the board or use sockets. Use a low-power soldering iron (25 to 35



A



B

Fig. 6. Above (A) is etching and drilling guide for pc board. It can be used for either channel for delay-line circuit, or for the phasor/flanger. At left (B) is component layout for one channel of delay line. It includes the power supply. Component layouts for phasor/flanger and second channel of stereo delay line are on next page.

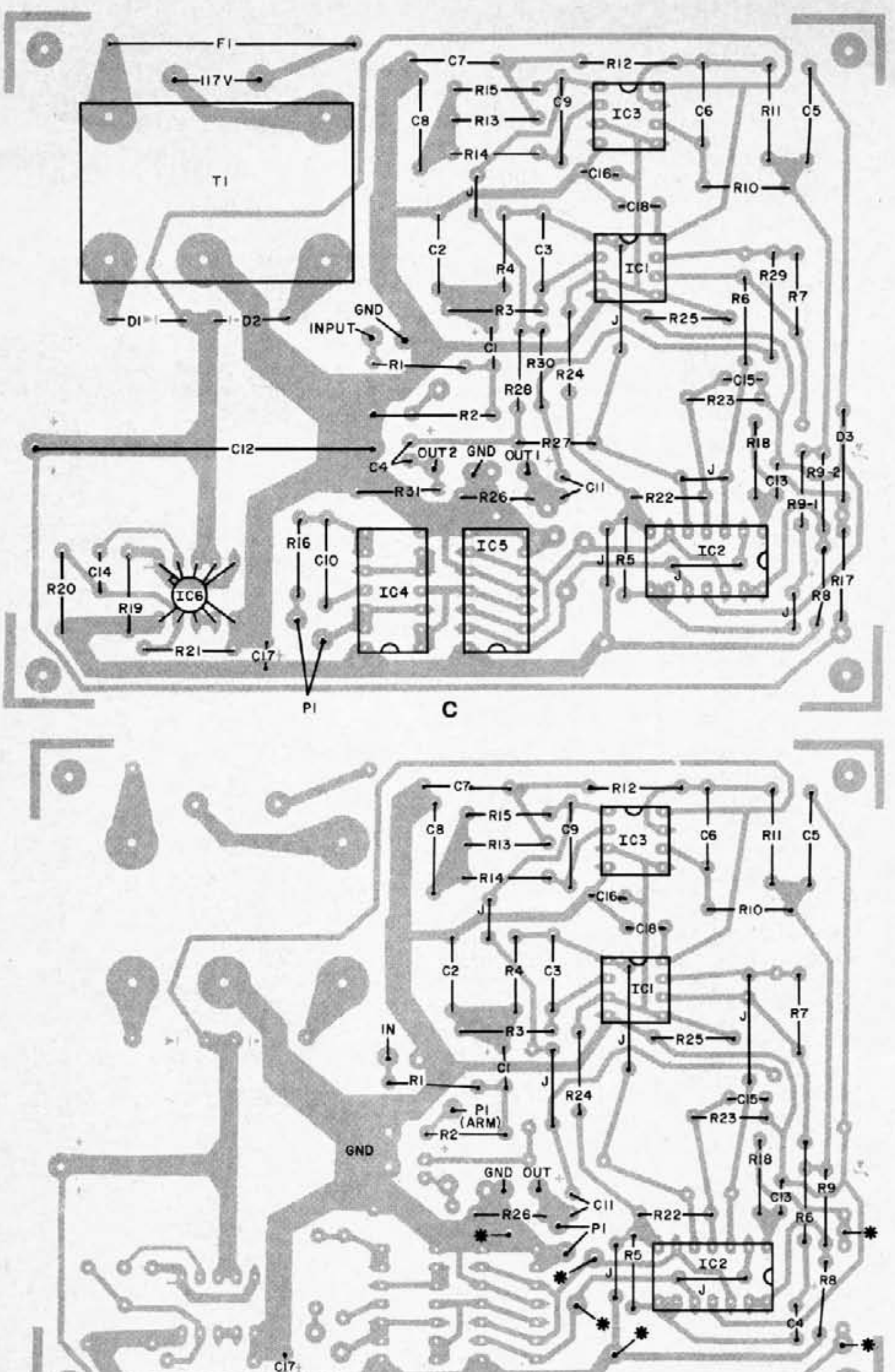


Fig. 6. Component layout at top is for phasor/flanger (C). Below (D) is for second channel of stereo system. It uses power supply in first channel.

watts) and fine solder, and watch out for solder bridges between the closely spaced pads on the board.

The wiring guide for the second pc board for a delay line for stereo is shown in Fig. 6D. Note that the power supply section is *not* repeated; you get power and clock pulses from the first board via wire interconnections.

Solder lengths of hookup wire to the pads that are to interconnect with the off-the-board pots and jacks. Then drill holes for the line cord, jacks, pots, and board mounting in a 5″ × 4″ × 3″ (12.7 × 10.1 × 7.6 cm) aluminum chassis box. Locate the line cord and jack holes on a wall directly opposite the wall through which the pot holes have been drilled.

Use machine hardware and spacers to mount the pc board assembly to the floor of the aluminum box. If you are assembling a stereo delay line mount the second board assembly over the first with short spacers and machine hardware after interconnecting the power-supply and clock-drive lines with hookup wire. (Be sure to make the interconnections before fastening the boards together.) Connect and solder the free ends of the hookup wires from the board(s) to the appropriate lugs in the jacks and pots. ◈

HANDS-ON EVALUATION

Both the time-delay and phasor/flanger configurations of this circuit should keep the home recordist occupied for hours, if not days. While the effects are not as apparent as those obtained with professional delay and flanging systems, this system does not cost the $4000 or so demanded for such top-of-the-line professional system.

The flanging effect is heard only while the potentiometer is in motion, at which time the variable comb filter sweeps across the audio bandwidth to create the "flanging" sound. At rest, the comb-filtered sound is noticeable, but it is not as apparent as one would expect from looking at the peaks and dips that occur at regular intervals on the frequency response curve.

Although you might not have occasion to use the flanger as a mono-to-stereo generator, don't overlook this operating mode for the enhancement of a single-output reverberation device. Reverberation is very diffuse by nature, and the flanger outputs, when panned left and right, are a noticeable improvement over a regular mono reverb return. When used in this application, the potentiometer remains at rest.

Use only one output when applying flanging to a recording. For an interesting Doppler effect, try combining the two outputs while rapidly revolving the pot. Better still, replace the standard pot with a free-spinning pot. (Connect the resistance element in series with *R16* and the wiper to either end of the element.)

On the delay line, the recirculation control must be used sparingly. A little goes a long way, and the "door spring" effect can easily get out of control. If you build both circuit configurations, you can experiment by wiring the flanger into the delay line's recirculation path. The slight additional delay in feedback creates even more echoes at the delay line's output. It also helps to keep the door spring from becoming a steady-state squeal.

—John Woram,
Woram Audio Associates

Solid State Reverb

Where have all the spring lines gone? Gone to lesser projects in other magazines, that's where. Meanwhile we present this cheap, simple, but high-quality unit using solid state technology. Design by Charles Blakey.

AT LAST - a reverberation unit which is not a pseudo echo effect and does not suffer from the defects of spring line devices. The unit described below will interface with virtually any preamplified signal and is ideal for direct use with most musical instruments or for incorporating in the 'echo-send' line of mixers. The design has been made possible by a new 3328-stage bucket brigade device having six tapped delays and capable of producing a useful reverberation time of about three seconds.

Sound emitted in an enclosed space will be subjected to both simple and multiple reflections from internal surfaces. Since these surfaces are at varying distances, the time for these reflections to occur and then decay by absorption will vary. The effect is a build-up of sound known as reverberation. When playing a musical instrument in the home, small studio or some other venue, the decay time can be very small coupled with a high absorption loss; the result is a weak sound when compared to recorded music or to live music played in a large hall.

Until now the only low-cost method of simulating acoustic reverberation has been the use of spring lines. These units, however, are prone to vibration, require a high power consumption for effective driving and are prone to producing distorted resonant peaks. Furthermore it is not possible to adjust the reverberation time and in many instances a short reverberation can be very effective. Another option has been available for some years, namely, the use of bucket brigade devices to electronically delay signals. While claims have been made for reverberation effects based on these products, a realistic unit would require at least three dual 512-stage BBDs, such as the Reticon SAD1024A. The cost and complexity of the latter approach puts it beyond the reach of the average constructor.

Beyond The Pail

The reverberation unit utilises the MN3011, which is the latest in a series of bucket brigade devices for audio applications to come from National Panasonic. They are all fabricated in PMOS and for a start you can forget most of what you may have read about the disadvantages of PMOS BBDs. It is a fact that they are somewhat limited in clocking speed (10 kHz to 100 kHz) and also have a limited bandwidth, typically 10 to 12 kHz. The latter, however, is not usually a limitation since the bandwidth is often restricted by the desire for long delay times. What makes the series ideal for audio applications is their low insertion loss, low distortion and excellent signal-to-noise ratio and for the MN3011 the specified values are 0 dB, 0.4% and 76 dB respectively.

The IC is unusual in that it has 12 pins but is the length of a normal 18-pin package; the functional block diagram and pinout for the MN3011 is shown in Fig. 1. As is normal with such devices it requires two power supplies, V_{DD} and V_{CC}; the former may be up to -18 V with respect to ground while V_{GG} should be 1 V higher than V_{DD}. Bucket brigade, or charge coupled, devices are analogue shift registers which operate by sampling

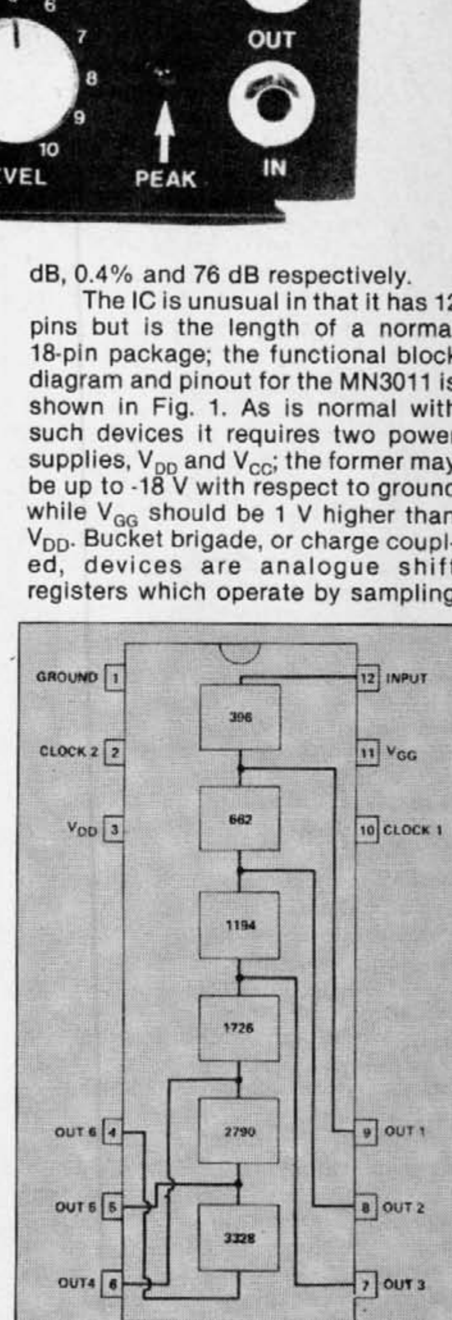
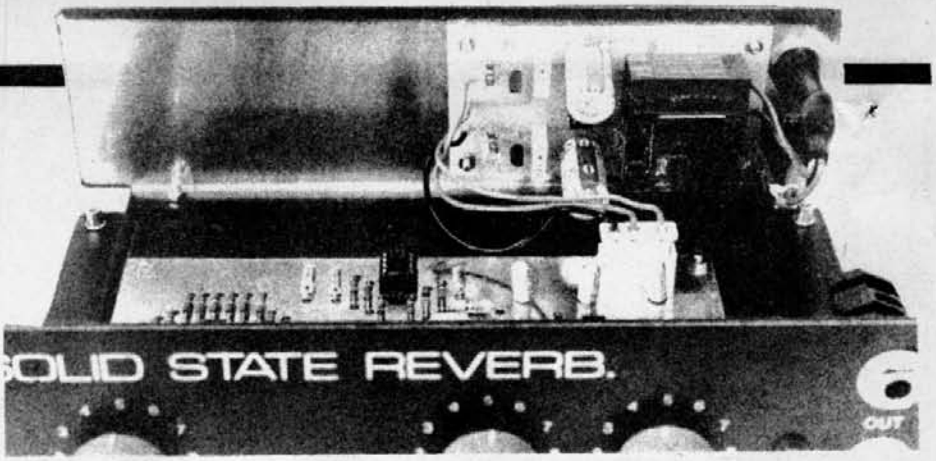


Fig. 1 Pinout and internal layout of the MN3011. The centre three pins on each side of this 18 pin package are absent.



the input signal at a rate determined by an external clock. The signal level at the time of sampling is stored on an internal capacitor; this charge is then clocked down a series of capacitors by means of internal switches. The transfer process is accomplished by a dual clock whose outputs are in antiphase and so are alternately opening and closing adjacent switches. It will be apparent that the slower the clock speed the longer the delay. Since the devices operate at high clocking speeds the input signals are faithfully reproduced at the output.

The most interesting feature of the MN3011 is that it has six tapped delays and Fig. 1 shows the number of stages for each tapping. The tappings are not evenly spaced since otherwise the reverberant sound would have a distinct flutter. If the device was being clocked at 10 kHz then the delays from outputs one to six would be 19.8, 33.1, 59.7, 86.3, 139.5 and 166.4 milliseconds respectively. If these delay times are mutiplied by 0.33 then one obtains the equivalent room path length for one trip, i.e. the longest delay is equal to a room length of 55 metres (181 feet). Reverberation time is usually measured as the time taken for the power to decay to one millionth of its initial level (60 dB down). For the present design the time was measured for the output level to fall to one hundredth of its initial level (-40 dB) and at the longest delay this was found to be about three seconds.

Blocks'n Clocks

The block diagram of the circuit for the reverberation unit is shown in Fig. 2. First there is the dual clock driver, which is another National Panasonic device, the MN3101. It has an oscillator, divider and wave form shaping and produces the dual clock pulses required by the MN3011. It reduces component count and is lower in cost than other alternatives, such as a 4007. A further advantage is that it also generates the required V_{GG} voltage.

The unit will operate satisfactorily with any input signal greater than 280 mV RMS and higher input signals are attenuated by the input potentiometer. The signal is also reduced by half an amplifier A1 and inputs higher than 140 mV to the first filter are indicated by a LED peak detector circuit. Although the MN3011 will accept signal levels up to 780 mV before the distortion value stated earlier is exceeded, it will become apparent that the effect of reverberation can lead to reinforcement of signals and consequently this has to be allowed for. The only preset in the circuit is used to apply a bias voltage to the signal. The precise value of this voltage is not very critical in the current design and the object is to keep the signal at a level where it will not be distorted or clipped within the BBD.

The main problem with BBDs is the inability to completely cancel out the clock pulses and these can form audible cross products with the input signal. In order to prevent this foldover distortion, the bandwidth of the input signal should be limited to between a half and a third of the clock frequency. Filter F1 in Fig. 2 is a lowpass filter with a cut-off frequency of 3.6 kHz. This may seem rather low but in fact it is equivalent to the upper reverberation limit of most spring lines and the BBD scores in respect of low frequency responses since springs usually give rise to 'booming' below 100 Hz. The limited bandwidth is compensated by mixing the original signal with the reverberated signal at the output stage. The filtered signal goes to the MN3011 and the six output stages are summed to give a composite signal with different delay times. The signal is again filtered with a lowpass filter with a cut-off frequency of 3.6 kHz, to remove residual clock glitches, prior to mixing with the original signal at the output amplifier, A2.

The most important feature, however, is that the signal from the longest delay is returned, slightly attenuated, to the input and subjected to further delays. This is the reverberation effect and with the times given earlier the sound will simulate the effect of the first reaching a surface 55 metres away (assuming slowest clocking rate) and then being reflected back as well as being reflected from other surfaces closer than the 55 metre surface. The whole process is repeated until the original delayed signal and its reflections die away. In the meantime new signals are being recycled and the overall effect is a build-up of sound — reverberation.

Construction

The construction is very straightforward but the following precautions should be observed. First, make sure you get the correct orientation of the ICs which are clearly shown on the component overlay. Second, the MN3011 is a CMOS device and with the advent of 'B' series devices we have all become rather careless as regards handling such ICs. For the MN3011, however, take the precaution of working on a grounded metal

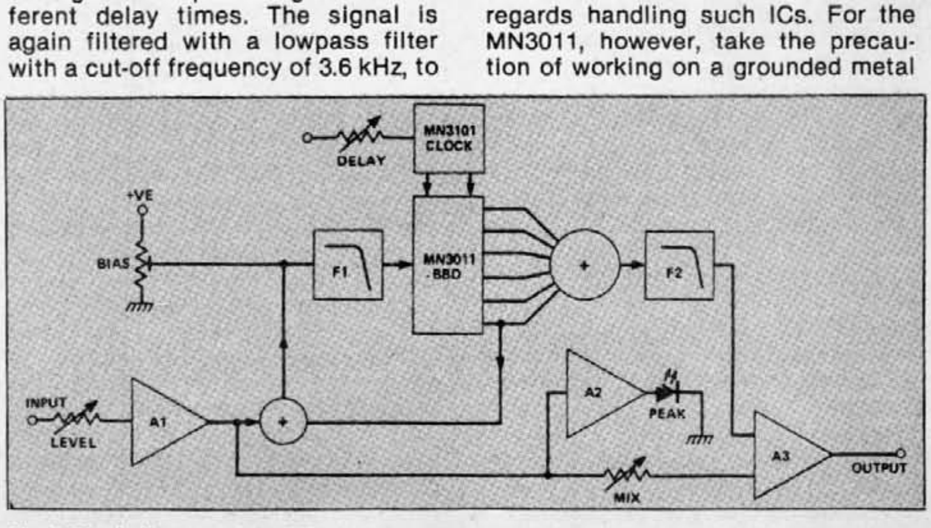


Fig. 2 Block diagram of the ETI Solid State Reverberation unit.

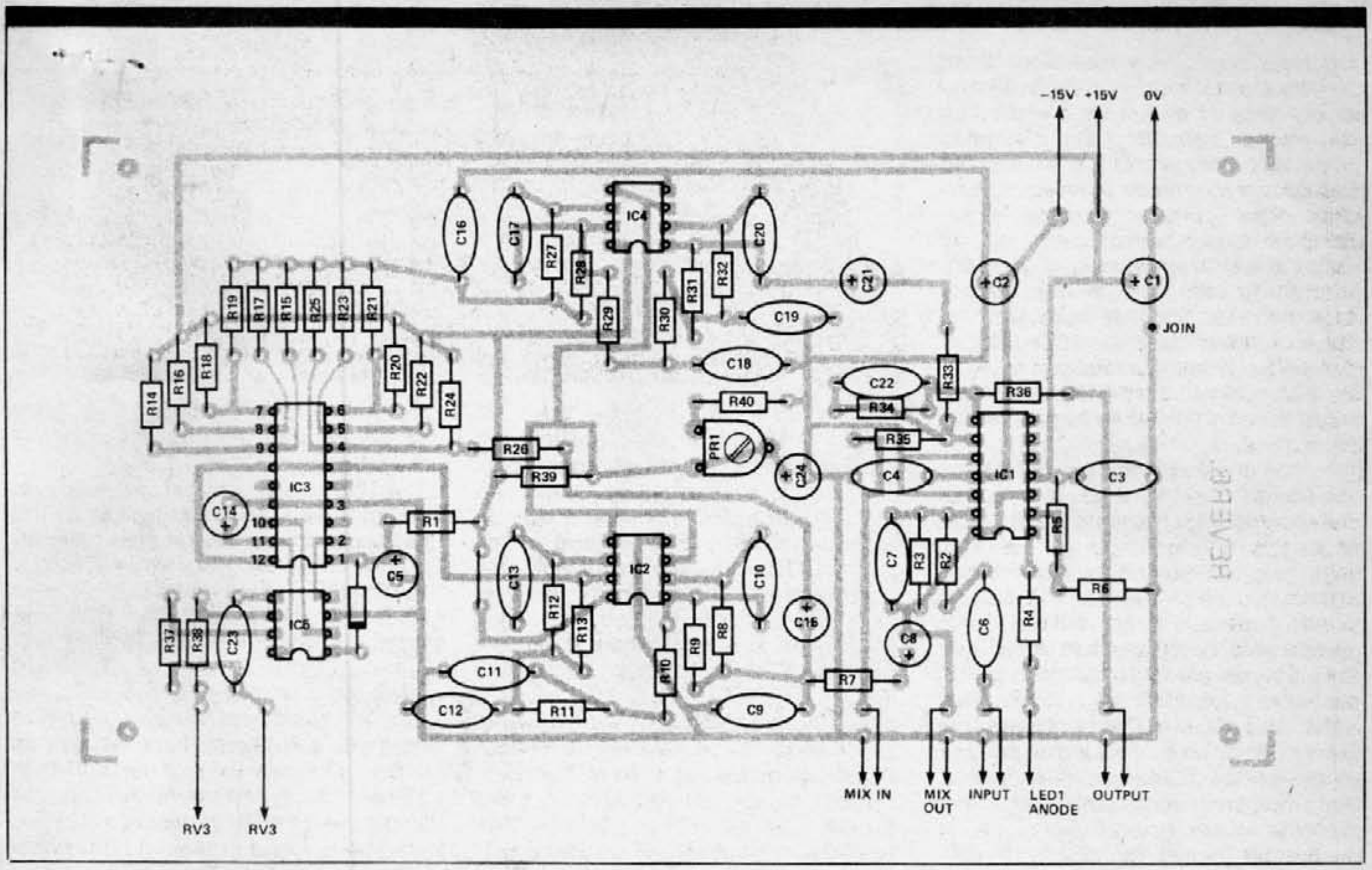


Fig. 3 Component overlay

PARTS LIST

Resistors (All ¼ W, 5% except where stated)

R1	10R ½W
R2,5,7,9 13,32,33,39	100k
R3,34	51k
R4	330R
R6	1k3
R8,12,27,31	33k
R10,29,37	47k
R11,30	56k
R14,16,18,20 22,24	56k 1%
R15	100k 1%
R17	110k 1%
R19	120k 1%
R21	130k 1%
R23	150k 1%
R25	160k 1%
R26	200k
R28	82k
R35	18k
R36	1k0
R38	36k
R40	68k

Potentiometers

RV1	100k logarithmic
RV2	10k logarithmic
RV3	470k linear
PR1	47k miniature horizontal preset

Capacitors

C1,2	10u 35V PCB electrolytic
C3,4	100n polyester
C5	22u 35V PCB electrolytic
C6	220n polyester
C7,10,13, 20,22	220p polystyrene
C8,14,15 21,24	3u3 63V PCB electrolytic
C8,11,12 18,19	2n7 polystyrene
C16	2n2 polystyrene
C17	270p polystyrene
C23	33p polystyrene

Semiconductors

IC1	TL074
IC2,4	LM358
IC3	MN3011
IC5	MN3101
D1	1N4148
LED1	5 mm red LED

Miscellaneous

SK1,2 mono jack sockets
PCB; IC sockets; case

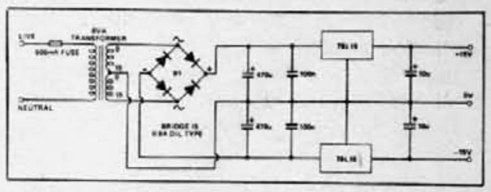


Fig. 4 Circuit diagram of a suitable PSU for this project.

POWER SUPPLY

PARTS LIST

Capacitors

C1,2	470u 35V PCB electrolytic
C3,4	100n polyester
C5,6	10u 35V PCB electrolytic

Semiconductors

IC1	78L15
IC2	79L15
BR1	0A9 DIL type

Miscellaneous

PCB; PCB-mounting transformer (15-0-15, 6 VA); 500 mA line fuse and chassis-mounting holder.

surface, such as a piece of aluminum foil, do not insert the IC with the power on and do not use a soldering iron on the PCB with the IC installed.

The PCB supplied with the kit has a ground plane to reduce interference from and to other electronic equipment as well as to reduce noise. This feature allows greater freedom in locating the unit, e.g. it does not have to be housed in a separate metal case. A ground plane comprises a metallized surface on the component side except for small areas around the holes for the components. Ensure that the component leads do not touch the ground plane — which is not difficult — and preferably solder the resistors and axial capacitors in place with a thin piece of card between the component and the board so that the former are not in physical contact with the ground plane. After soldering the card is removed. The latter step is not essential. The one wire link must be made with insulated wire. The ground plane has to be connected to the 0V line and some 15 mm from where the latter is connected to the PCB there is a hole marked 'join'. A piece of wire should be placed through this hole and soldered on both sides of the PCB.

HOW IT WORKS

The input signal is attenuated by RV1 and also by the inverting amplifier built around IC1a which has a gain of about 0.5. From IC1a the signal goes three ways. A comparator built around IC1b forms a peak detector to indicate optimum signal level, while RV2 and R35 allow mixing of the original signal with the reverberated signal in the inverting amplifier configured around IC1c. The component values in this section are such that equal proportions of the two signals may be mixed. Finally the signal also passes to two active filters constructed around IC2 which have a 12 dB/octave roll-off for each stage and a cut-off frequency of 3.6 kHz.

From the above filter stages the signal passes into the MN3011 and the six delay outputs are summed by the resistor network formed by R14 to R25. Note that the shorter the delay, the less the attenuation. From the longest delay (pin 4) the signal goes via R25 back to the input of the filter and thus provides recycling of the delayed signal in order to generate a true reverberation effect. The reverberated signal is filtered by two active filters constructed around IC4 and these have the same characteristics as the input filters. Between the active filter stages some passive filters have also been added to increase the roll-off; the loss in these filters is compensated by increasing the gain of the active filters.

The dual clock for the MN3011 is provided by IC5 and with the components shown, the clock frequency may be manually varied with RV3 over the range 10 kHz to 100 kHz, allowing maximum first pass delays from 16.64 to 166.4 milliseconds. Pin 8 of IC5 provides the V_{GG} voltage for the MN3011. Since both IC3 and IC5 are P-channel CMOS it would be normal to operate them from a -15V supply. Voltages are, however, relative and by connecting +15V to the ground pin and ground (0V) to the V_{DD} pin they will operate happily with positive signal inputs. R1 and C5 prevent clocking signals getting back into the power lines. The filters are also operated from a single +15V supply and this avoids any problems which may arise from excessive bipolar signals, ie they will be clipped at +15V or ground and not damage the BBD. The bias voltage required by the BBD and the filters is primarily to allow them to accept bipolar signals; this voltage is provided by the resistive divider using components R39, PR1 and R40 and is aplied to the non-inverting input of the filter op-amps.

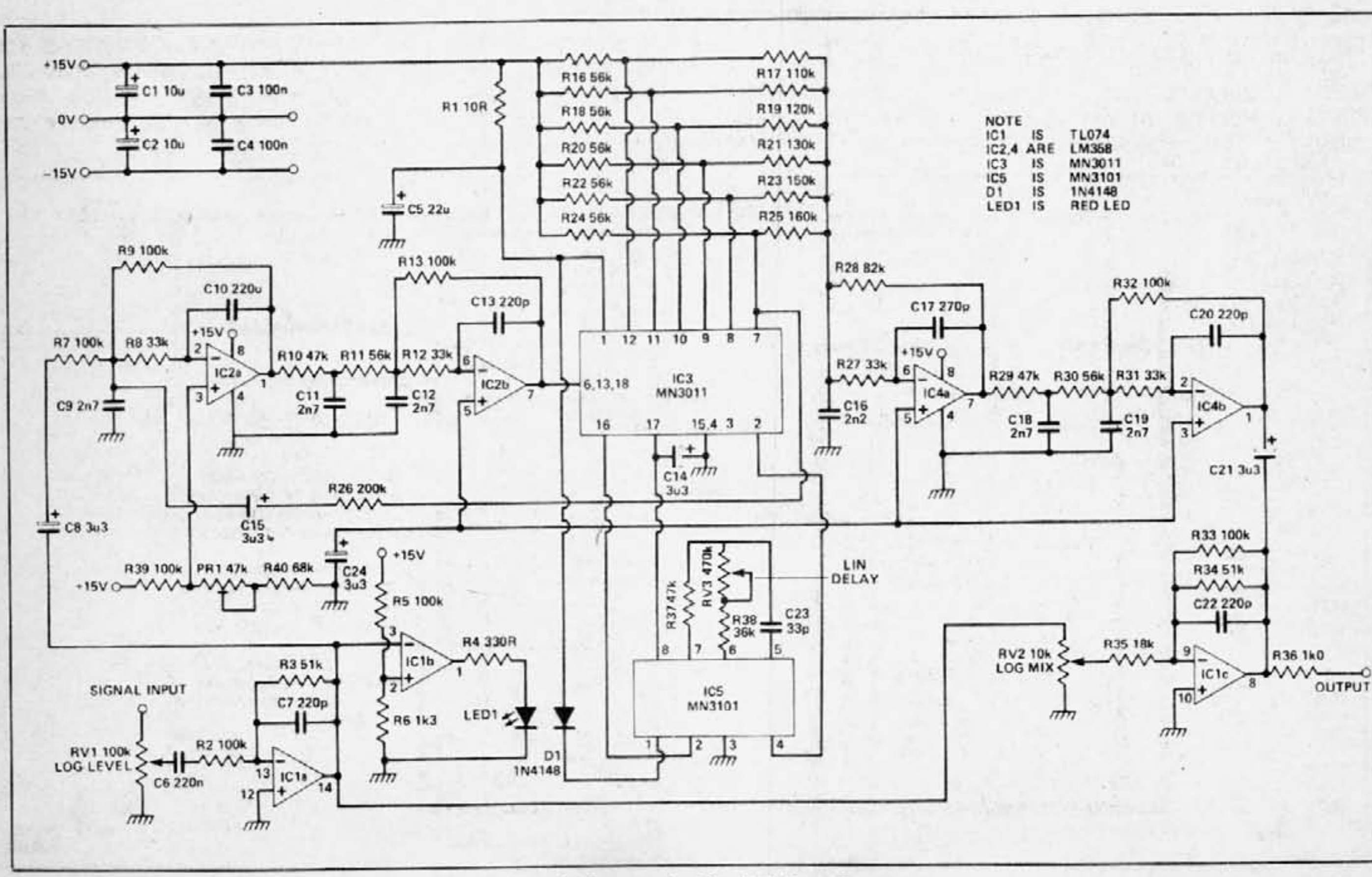


Fig. 5 Circuit diagram for the ETI Reverb.

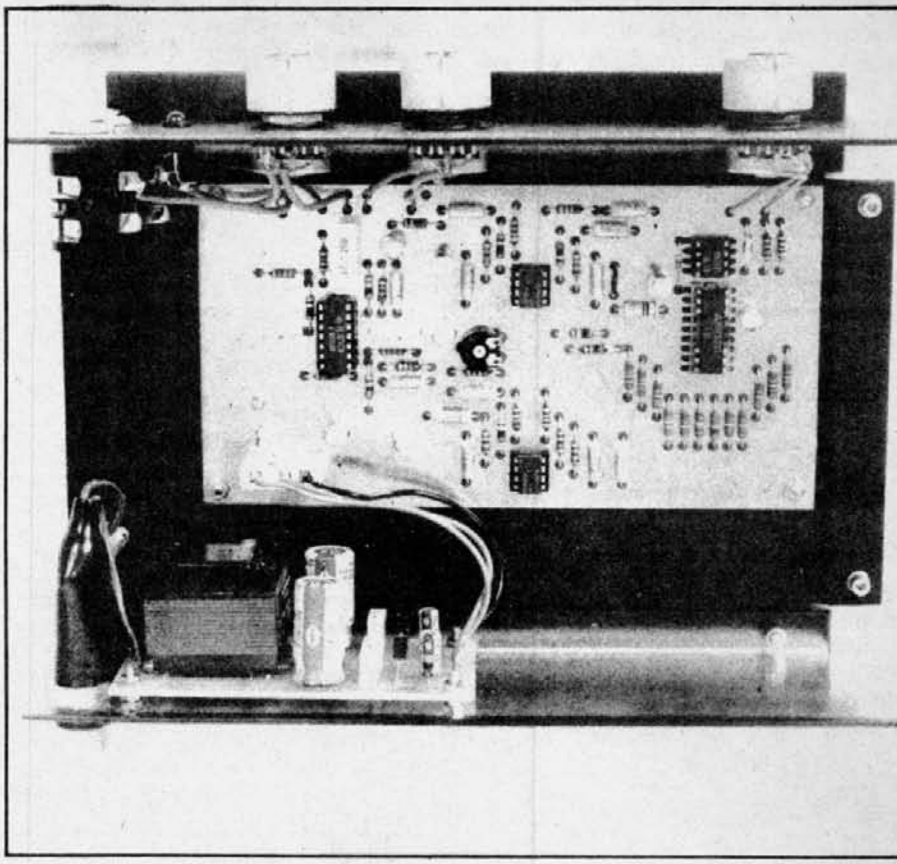
The PCB has been laid out such that the BBD and clock are as far away as practical from the signal input and output. This separation should be maintained if the unit is housed in a box and all wiring should be kept as short and as neat as practical, with the audio connections being made with miniature screened cable.

The unit requires a ± 15 V power supply and the current consumption is a miserly 13 mA at + 15 V and 9 mA on the -15 V line. If a separate power supply is required then a suitable PSU is shown in Fig. 4. A PCB-mounted transformer is preferred, and it should be mounted as far away from the BBD as practical.

Setting Up And Use

The only setting up required is adjustment of PR1. If a sinewave source is available then the latter may be used as the signal source and PR1 adjusted by ear, or with an oscilloscope, for minimum distortion. Alternatively measure the voltage at the junction of PR1 and R40 and adjust PR1 to give a reading of 6V2.

The unit has a signal-to-noise ratio of better than 60 dB but this requires that it is operated with the peak indicator LED just glowing or occasionally illuminating. The output level will vary from about 0V5 to 1V RMS, depending on the amount of mixing of the original signal, and these levels should ensure adequate response from most amplifiers, mixers, and so on. In other words, by keeping input signals at maximum level the amplifier setting will be such that during periods of no signal the residual noise will not be obtrusive. This is common practice with recorders, many of which have much lower signal-to-noise ratios.



Inside the reverb unit.