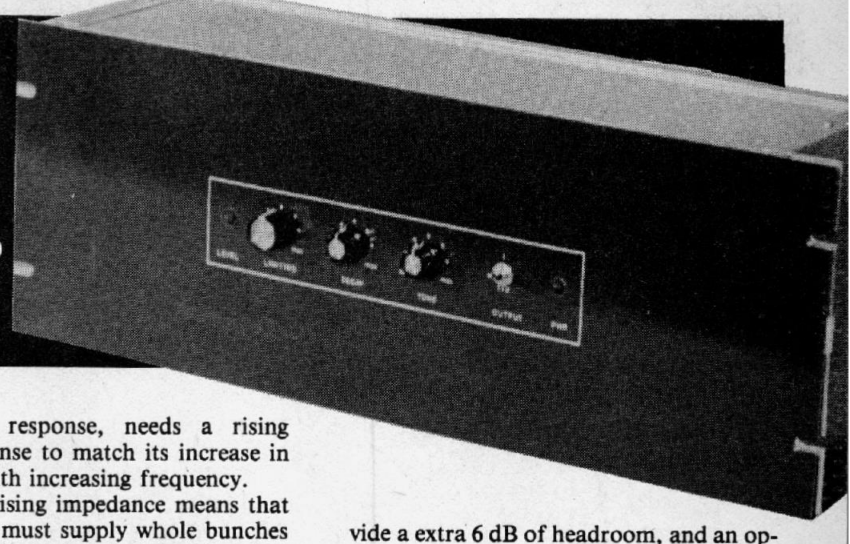
Reverb Amplifier

PROJECT



A spring line for professional audio? Don't sneer - those inexpensive springs have remarkable potential when used with the right electronics. By Bill Markwick.

IF YOU'VE listened to the average guitar reverberation, you're probably creasing over giggling with the idea of using the familiar Gibbs or Accutronics springs for home or studio recordings. They bring to mind that hollow boingy twang, like yelling down a pipeline. But wait: it isn't so much the fault of the line as the electronics driving it. Here are the major drawbacks.

1. Some driver amps are low impedance voltage sources such as op amps or emitter followers; this approach neglects the fact that the spring's input coil looks like a pure inductance, and, for good treble response, needs a rising voltage response to match its increase in impedance with increasing frequency.

2. This rising impedance means that the amplifier must supply whole bunches of voltage at the high end; if it isn't a constant voltage source, the output climbs to follow the line impedance, and if it is, the operator is tempted to crank the input level to make it sound brighter. Most amplifiers run out of headroom very quickly.

3. Under the pressures of recording in a studio, the operator can't always monitor the signal level carefully, and too much signal is usually the cause of the springy sound we all know and hate.

Here's an amplifier that goes a long way towards curing the reverb spring blues. It features a constant-current type of drive to allow for the spring's rising impedance, a bridge configuration to provide a extra 6 dB of headroom, and an optional limiter circuit to prevent overload distortion in case you nod out during the session.

To begin at the beginning: the input signal can come from either balanced or unbalanced sources; IC1 and the three-conductor input jack form a balanced amplifier for rejection of any noise induced on the line, and if you're using an unbalanced source with regular two-conductor phone plugs, it just acts as a 47K buffer amp.

IC2 is the drive circuit for the line. It's a voltage-controlled current-source, which means that the current through the line will be constant for a constant input; the circuit automatically follows varia-

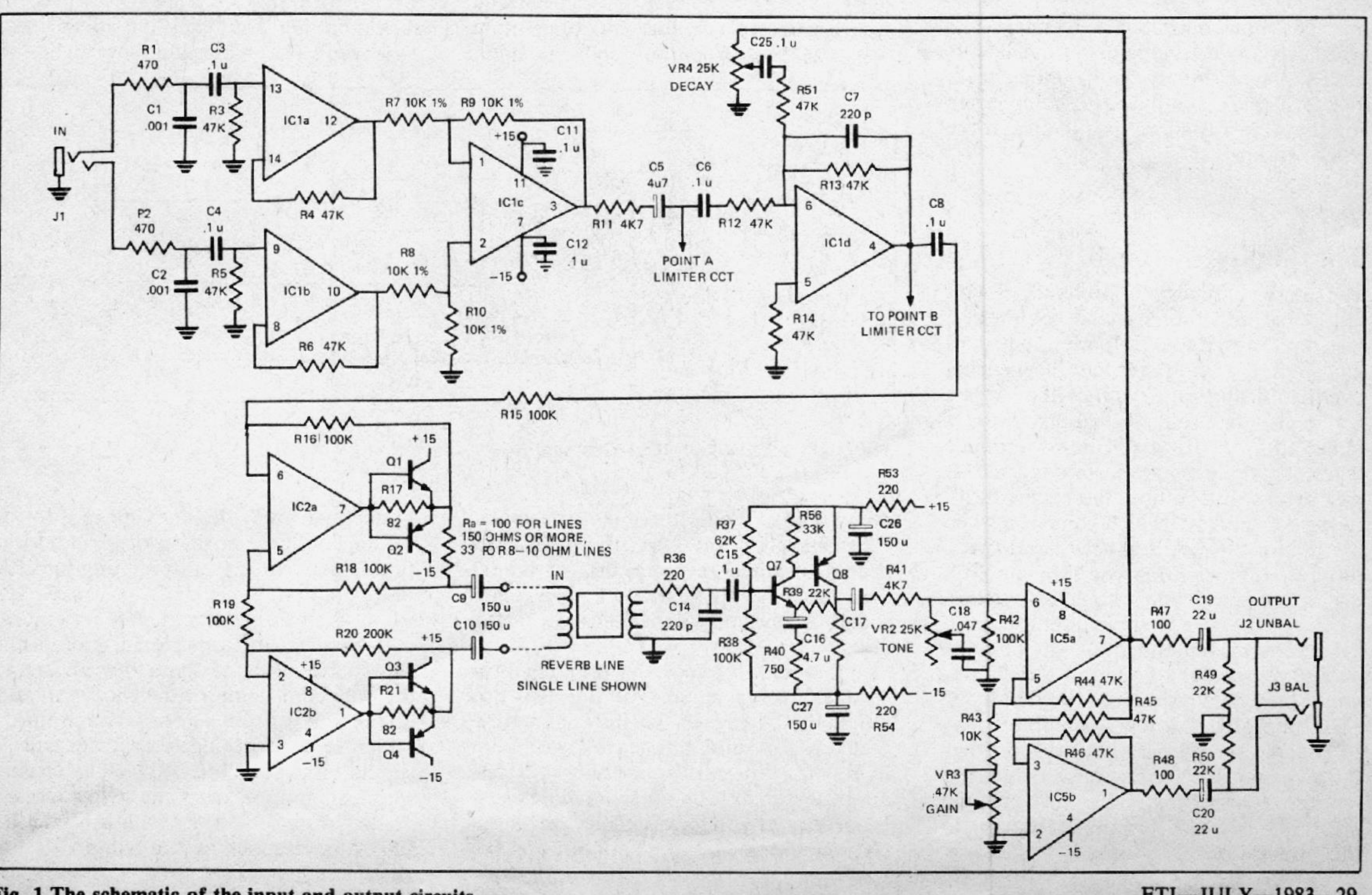


Fig. 1 The schematic of the input and output circuits.

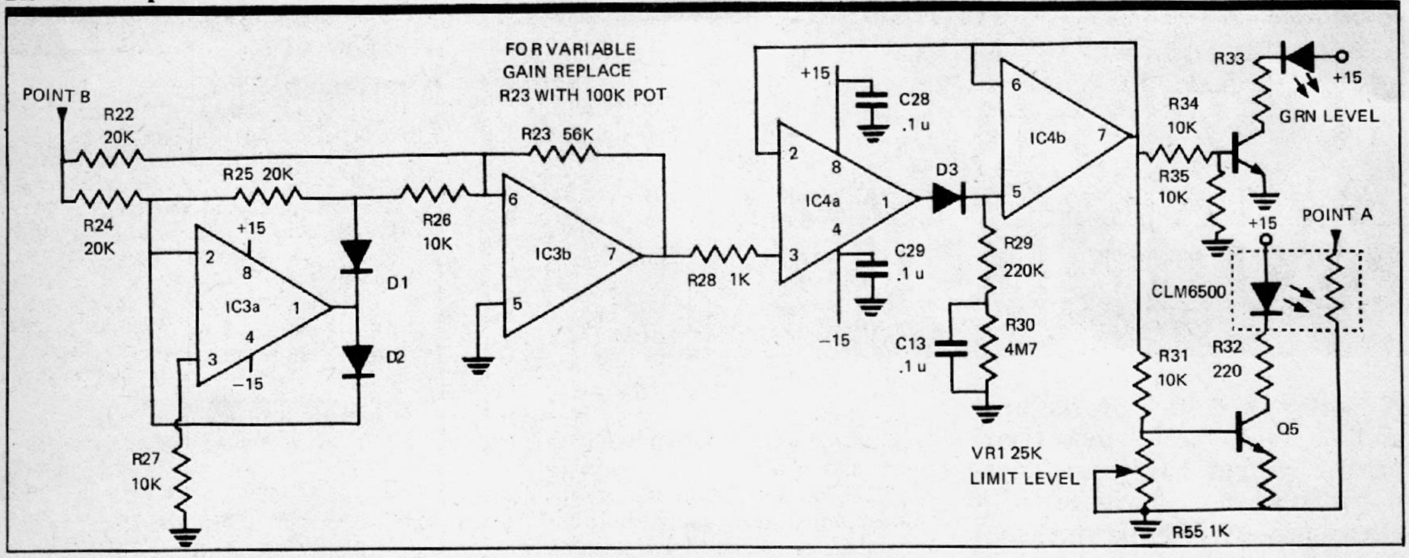


Fig. 2 The schematic of the limiter circuit.

tions in line impedance. IC2b is out-of-phase with IC2a; this effectively doubles the signal swing across the line - it's equivalent to using a 60 V supply.

The loss through the line itself is about 40 dB. This loss is restored by the output amplifier Q7, Q8 and IC5. The sound of the line tends to be rather bright because of the excellent HF response of the current source drive, and a tone control is added after Q8 to allow you to set the treble to suit; cutting the treble also improves the signal-to-noise ratio considerably, without seriously muffling the sound.

The final output is a balanced-line type, and a two-conductor output jack is added in case your system is unbalanced. The level can be adjusted to compensate for the wide variations you'll find in the reverb springs.

The Limiter

The limiter circuitry is optional; if you don't feel into the cost and complexity, just omit the circuitry. However, it does have the advantage of preventing overload distortion if you set the levels a bit high, or someone leans into a microphone. It also helps reduce "flutter" on percussive sounds such as kick drums. In addition, there's the facility for a green LED which comes on when the level is adequate; a drive level that's too low will mean that you'll be amplifying noise, hum, and people's footsteps when you hoist up the reverb return pot to get acceptable volume.

If you decide to build in the limiter, you'll have to find the Clairex CLM6500 LED/photocell. I know it's a bore trying to hunt up special parts, but this one has the advantage of not causing any increase in noise or distortion levels; it's worth the effort. The CLM6000 should also work in this application.

The Line

Now as to the line itself: the real piece de resistance in the way of spring lines is the Accutronics 99 system. It consists of two lines for a total of six springs; the delay and decay times are chosen for a realistically full reverb sound, and they aren't quite as plagued with mechanical whistles and pops when the signal goes through. You can bug your local music store about getting you one of these systems; most places that repair guitar amps should have access to Accutronics parts. These are special order lines, and here's what to ask for: 240 ohm input coils, 2575 ohm output coils, medium decay time, and floating connectors. When installing the lines, the input coils should be wired in series, and so should the outputs. This seems to give a richer sound, as one coil's impedance affects the other.

If you can't face the ordering process, any large guitar-type line will do, although the higher the impedance, the better. Many guitar lines have 8 or 10 ohm lines, and the driver circuitry in the amplifier may be restricted in its headroom. If you have a choice, the input coils should be 200 to 800 ohms, and the output coils 600 to 3000 ohms.

Mounting

The reverb lines and the amplifier will fit nicely into a standard 19 inch rackmount box, as shown in the photograph. This reverb amp had an amazingly wonderful idea incorporated into it: the two sets of output coils from the Type 99 each had their own amplifier; this gave a "stereo" output from a mono input. The trouble was, nobody could hear difference as the switch was changed back and forth from stereo to mono. Such is life.

There is one small hassle involved in rackmounting, and that is the reverb line's obsession with picking up every bit of hum it can find; this may mean a lot of fiddling with the power transformer location to prevent it from radiating into the line's coils. If all else fails, you may have to put the transformer in a separate mini-box. Remember, too, that the audio jacks should be insulated from the chassis — the only connection to the chassis should be the power supply's zero-volts point.

Also, the regulator IC's 6 and 7 should either be mounted on the chassis with insulating washers and nylon screws, or they should be fitted with a tiny heat-sink; a square inch or two will do.

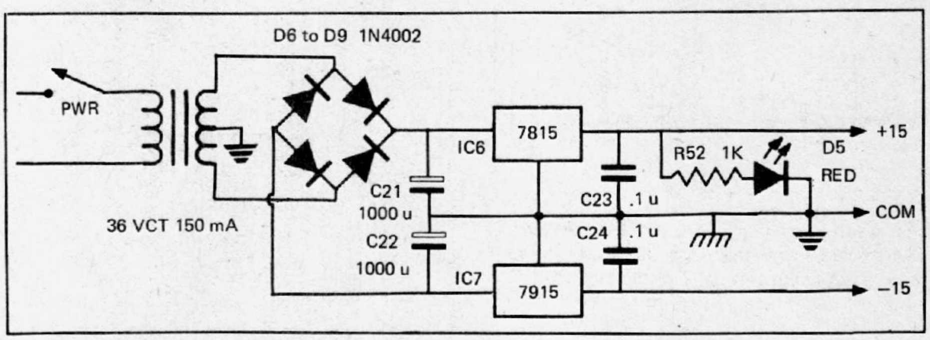


Fig. 3 The schematic of the power supply.

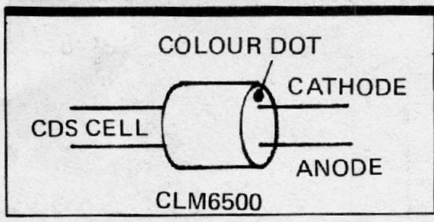


Fig. 4 The CLM6500 case outline.

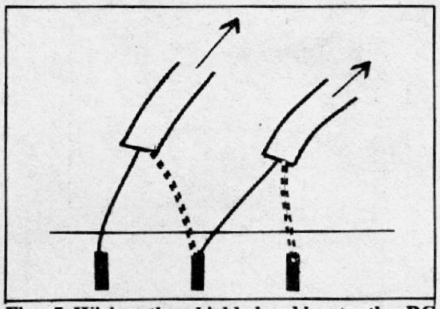


Fig. 5 Wiring the shielded cables to the PC board when two lines are used; inputs and outputs are wired in the same way.

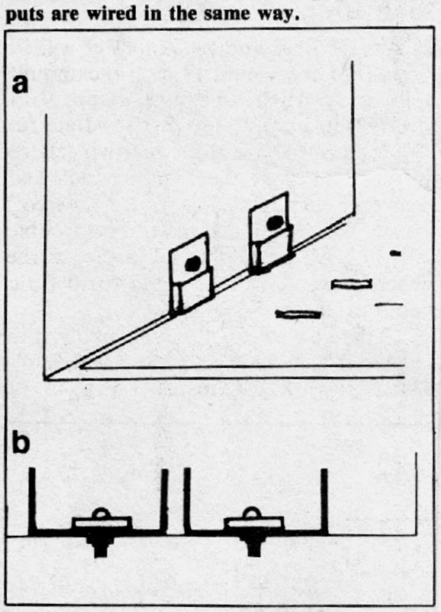


Fig. 6 In (a) the regulators are mounted to the chassis with insulating kits; (b) shows the top view when using two small heatsinks.

Calibration

You can't calibrate the level with a sine wave! There are so many resonances in any reverb system that the output sine amplitude can vary 20 dB within a few Hertz. The easiest way is to send it program material at a level consistently near 0 VU, and adjust the output trim pot to get the same level out. If you're a fanatic about level-setting, you'll need a white noise source such as an FM tuner set off-station. This will give a good relative indication of input/output levels.

Operation

Operation is straightforward. The green LED should flash every now and then,

PARTS LIST

Resistors, ¼ W 5%

R1,2	470
R3,4,5,6,44,45,46,51	47K
R11,41	4K7
R15,16,18,19,38,42	100K
R17,21	82
R20	200K
R22,24,25	20K
R23	56K
R26,27,31,34,35,43	10K
R28,52,55	1K
R29	220K
R30	4M7
R32,36,53,54	220
R33	1K2
R37	62K
R39,49,50	22K
R40	750
R47,48	100
R56	33K

Ra is 100 ohms for lines 150 ohms or greater, 33 ohms for 8 to 10 ohm lines.

¼ W, 1%

R7,8,9,10	10K
VR1,2,4	25K potentiometer
VR3	47K trim pot such as Philips 411 series.

Capacitors

C1,2	.001 *u* ceramic
C3,4,6,8,11,12,13,15,23,24,25,28,29	.1 *u* film
C5,16,17	4*u*7 electro.
C7,14	220 p
C9,10,26,27	150 *u*, 16V electro
C18	.047 film
C19,20	22 *u* 16V electro
C22,25	1000 *u*, 35V electro

Semiconductors

IC1	4136 low noise quad op amp
IC2,3,4,5	TL 072 dual FET op amp
IC6	15 V regulator
IC7	-15 V regulator
IC8	Clairex CLM6500 opto-isolator
Q1,3,5,6,7	2N3904 NPN transistor or equiv.
Q2,4,8	2N3905 PNP transistor or equiv.
D1,2,3	1N4148 silicon diode or similar
D4	any green LED
D5	any red LED
D6,7,8,9	1N4002 silcon power diode or equiv.

Hardware

J1,3	three-conductor phone jack
J2	two-conductor phone jack

Insulators for above, if necessary
Mounting kits or heatsinks for regulators

Miscellaneous

Rackmount box or similar, 36 V 150 mA transformer such as Hammond 166E36, line cord, power switch, knobs, RCA phono cords for connecting reverb lines, PC terminals.

HOW IT WORKS

THE FIRST three sections of IC1 form a true differential amplifier. IC1a and IC1b are unity-gain buffers, and feed the balanced signal to difference amplifier IC1c. The common mode rejection of this amplifier, which is set largely by the matching of R7 to R10, drops out any noise which has been induced on the incoming line; the CMR should be about 40 dB.

IC1d functions as both a mixer and a buffer; it mixes the output signal back into the reverb line, with R51 preventing oscillations, and also buffers the signal after the limiting resistors R11/IC8.

IC2 is a constant-current source. The current is set by Ra, and the feedback loops set by R16 and R18 maintain the output current at the proper level regardless of load impedance changes. Q1 and Q2 increase the output current capability; if it rises beyond a few milliamps, the voltage drop across R17 will turn on one of the transistors. IC2b is an inverting amplifier; the resultant out-of-phase signal across the load means an increase in maximum output of 6 dB, or twice.

The recovery amplifier is a straightforward compound connection with a gain of about 30 dB. Low frequency rolloff is provided by C15, (and also by C6 in the drive amp) and prevents the line from producing distortion on large bass amplitudes.

After the tone control, a balanced output is provided by IC5, with an unbalanced output available at J2. This output is adjustable for up to 15 dB gain unbalanced, or 21 dB balanced.

IC3 is a full wave rectifier, and sends the rectified signal to peak detector IC4. This IC charges C13 to the peak level of the input signal, and this charge is buffered by IC4b and sent to control transistors Q5 and Q6. If Q5 turns on, the LED in the opto-isolator is turned on, and reduces the resistance of the CDS photocell. This shunts the input signal at R11. The minimum resistance of the photocell is about 270 ohms, and results in a control range of 25 dB when combined with R11.

say, once every one or two seconds. If it stays on consistently, the level is set too high. Incidentally, you can delete this LED entirely if you prefer watching a VU meter on your mixer.

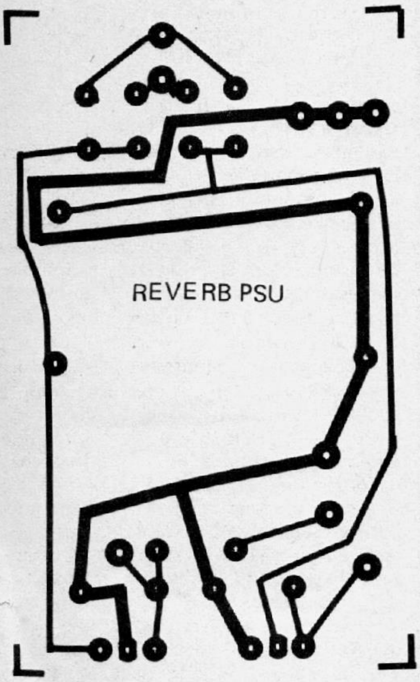
The limiter control takes a bit of experimenting to find the point where limiting begins; an audio source with loud peaks is good for this. You can then leave the control in this position forever, if you like.

The tone control is just set to suit your idea of good treble. It certainly isn't meant to be as good as returning the reverb back through an equaliser, but it'll do in a pinch.

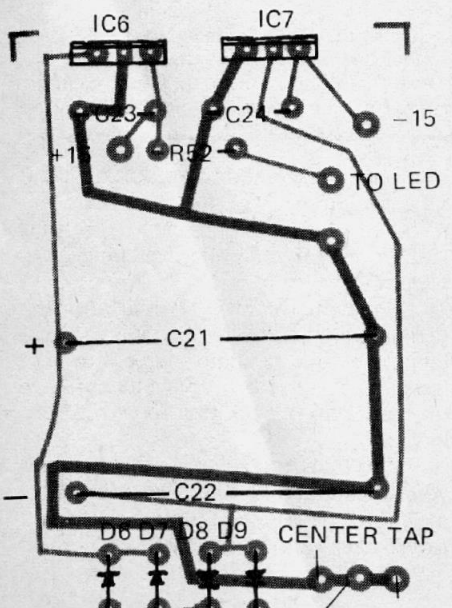
How Does it Sound?

Well, it won't sound like an EMT plate or an AKG spring. There are limits to the El Cheapo route, you know. On the other hand, it sounds remarkably better than the average spring line that you find in small mixers and instrument amps. The only objection usually raised relates to the little bit of ping and flutter you'll hear on very loud percussive drum sounds. This seems to be inherent in the line, and I've never heard any limiters or envelope followers that would take it out. The objections can be reduced if not stilled by minimizing the input level as best you can.

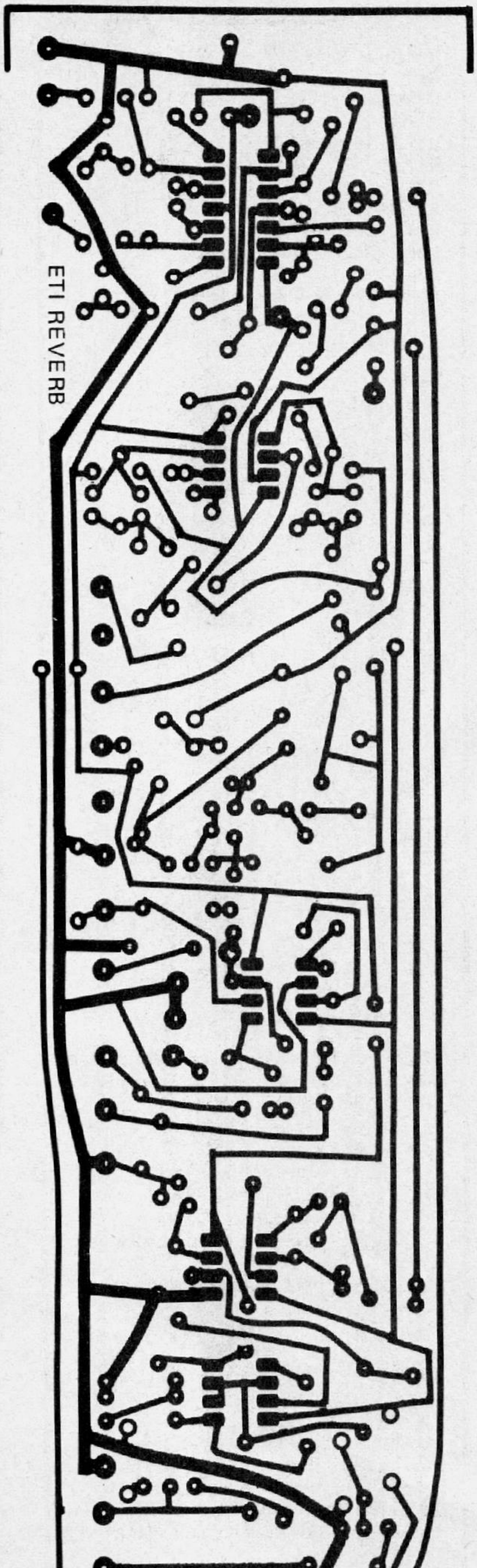
Happy reverbing!



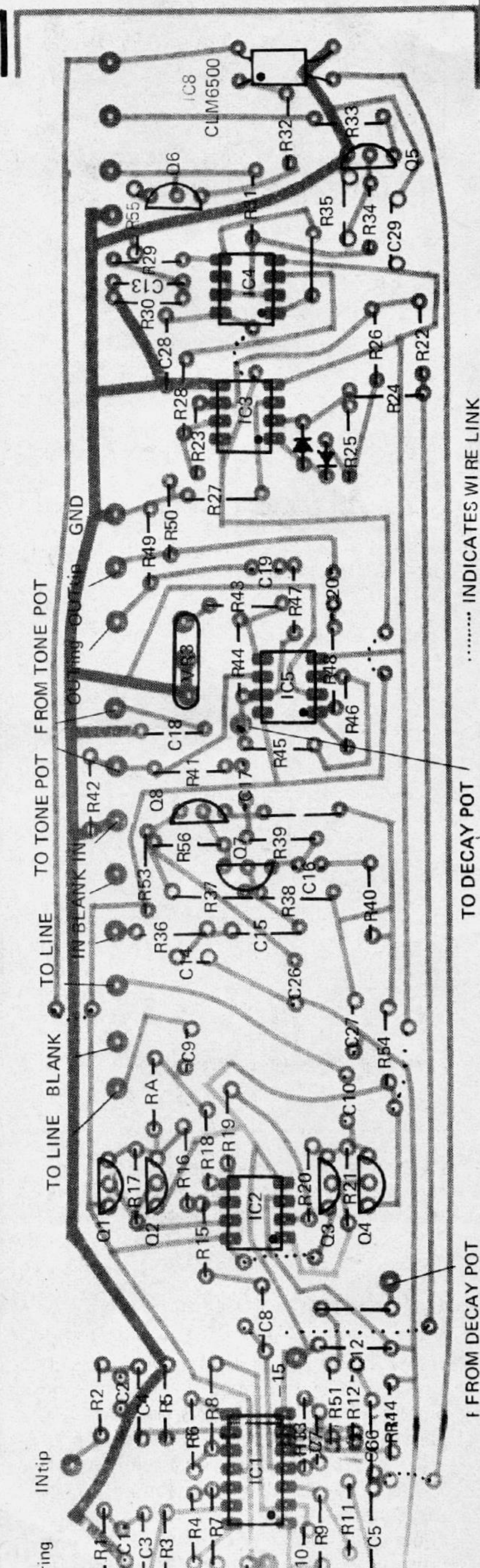
The printed circuit for the power supply.



10 The power supply component location.



The printed circuit for the amplifier.



The amplifier component location.

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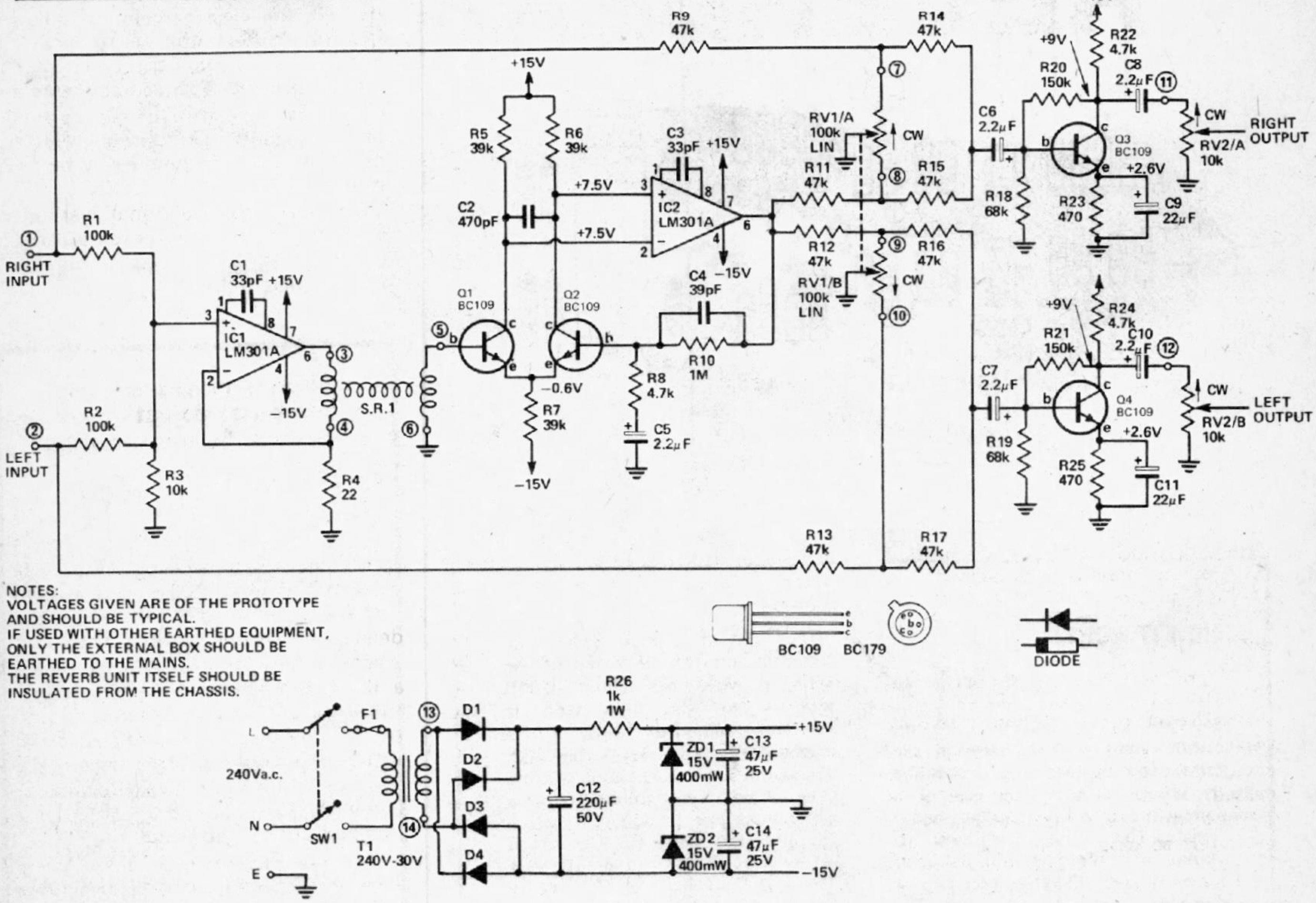


NOTES:
VOLTAGES GIVEN ARE OF THE PROTOTYPE AND SHOULD BE TYPICAL.
IF USED WITH OTHER EARTHED EQUIPMENT, ONLY THE EXTERNAL BOX SHOULD BE EARTHED TO THE MAINS.
THE REVERB UNIT ITSELF SHOULD BE INSULATED FROM THE CHASSIS.

Fig. 1. Circuit diagram of the spring reverberation unit.

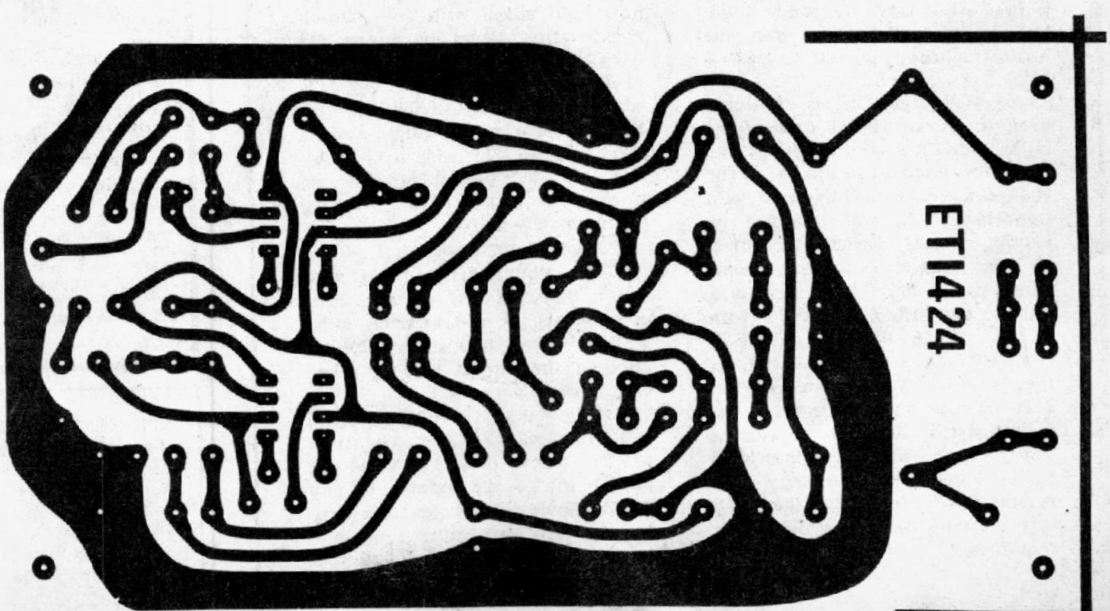


Fig. 2. Full size printed circuit board layout.

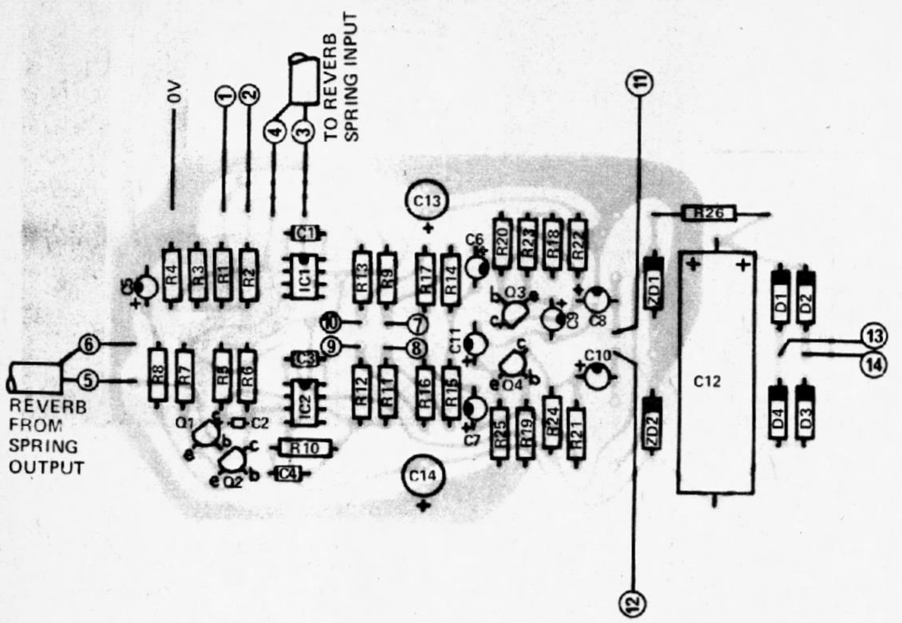


Fig. 3. Component overlay.

This enables the unit to be used as a flexible system component, but, if desired, the electronics may easily be incorporated within an existing system-box if room permits.

The majority of the components are mounted upon one single printed-circuit board, although matrix or veroboard can quite easily be used if preferred.

Whichever constructional method is used, it is essential to check polarized components, for correct orientation, before soldering.

ETI 424 PARTS LIST

R4	Resistor	22	½W	5%
R23,25	"	470	"	"
R26	"	1 k	1W	"
R8,22,24	"	4.7 k	½W	"
R3,	"	10 k	"	"
R5,6,7	"	39 k	"	"
R9,11,12,13	"	47 k	"	"
R14,15,16,17	"	47 k	"	"
R18,19	"	68 k	"	"
R1,2,	"	100 k	"	"
R20,21	"	150 k	"	"
R10	"	1 M	"	"

RV1	Potentiometer	100k dual lin rotary
RV2	"	10 k dual log rotary
C1,3	capacitor	33pF ceramic
C4	"	39pF ceramic
C2	"	470pF ceramic
C5,6,7,8,10	"	2.2μF 10V electrolytic
C9,11	"	22μF 10V electrolytic
C12	"	220μF 50V electrolytic
C13,14	"	47μF 25V electrolytic

Note: all electolytics except C12 are pc mounting.

D1-D4	diodes	1N4001 or equivalent
ZD1-ZD202	Zener diodes	BZX 79 C15 or any 15V 400mW type
Q1-Q4	transistor	BC109 or equivalent
IC1,2	operational amplifier	LM301A

PC board ETI 424
SW1 switch 2 pole on-off 240V rated
F1 fuse and fuse holder 500 ma chassis mounting
Spring reverb unit

T1 transformer 250V/30V 500mA

3 core flex and plug
2-way phono sockets — 2 off
12mm long spacers 4 off
chassis to Fig. 7
metal cover to Fig. 8
front panel to Fig. 6
rubber grommet for power cord and insulating reverb unit.
insulated phono socket for reverb.

SPRING LINE: a sensitive spring line unit is needed — the LM301A cannot drive the common 16Ω type. An input impedance of 150Ω or more is required. Elvins Electronic Musical Instruments of 40 Dalston Lane, Hackney, London E8, have an 150Ω unit, the E150, selling at £7.50 + 25% VAT inc. P & P.

HOW IT WORKS

The reverberation spring is an electro-mechanical device for delaying and producing echo on audio signals – it operates in the following manner. A relay-like transducer vibrates one end of a spring in response to an input audio signal. The spring continues to vibrate after the excitation has been removed and thereby produces a decaying 'echo' as well as delaying the propagation of the signal to the transducer at the other end.

The mechanical system naturally has many resonances and the frequency response therefore cannot be flat over a small frequency range, but is substantially flat over the broad frequency range of 50 Hz to 4 kHz.

Integrated circuit IC1 is connected so as to provide current drive to the input transducer of the spring. The transducer is inductive and hence, the voltage across it will increase with frequency. However, since the current remains constant, the power in the transducer also remains constant. The stereo input is summed into R3 by resistors R1 and R2 (with a loss of 20 dB) to provide a composite signal at pin 3 of IC1. As the amplifier always tries to keep pin 2 at the same potential as pin 3, the voltage across R4, and the current through it, is therefore proportional to the input voltage. As very little current flows into pin 2 of the IC, all this current flows through the transducer.

The output signal from the transducer at the other end of the spring is very small (about −50dB referred to the input and is therefore amplified back to a reasonable level by Q1, Q2 and IC2. Transistors Q1 and Q2 are low noise types and are arranged as a differential pair to add gain before the inherently noisy IC. The gain is set by (R10+R8)/R8 to about 46 dB. The low frequency cutoff is set by C5 and R8, and the high frequency cutoff by R10 and C4. Note that these last figures refer only to the receiving transducer amplifier and not to the whole system.

The direct inputs, left and right, are now both mixed with the common reverberation signal in mixers Q3 (right) and Q4 (left). The proportion of direct and reverberation signals is adjustable by means of depth control RV1. The gain of the output stage is set by R20, R21 and the bias by R18, 19, the overall gain of the complete system being approximately unity.

If single channel operation only is required, simply delete the second mixer transistor and its associated components. If reverberation only, without the mixing facility, is required the output may be taken direct from pin 6 of IC2.

In the event that a volume control is not required resistors may be fitted to the board (holes provided on board) to set the volume to any desired level. These resistors may have any value between 10 k and 1M.

The unit should be wired, as shown in Fig. 1, taking care to keep all 240 volt ac wiring well clear of the electronics and especially clear of the receive end of the reverberation spring. The metal case itself should be earthed even though the electronics itself is not earthed.

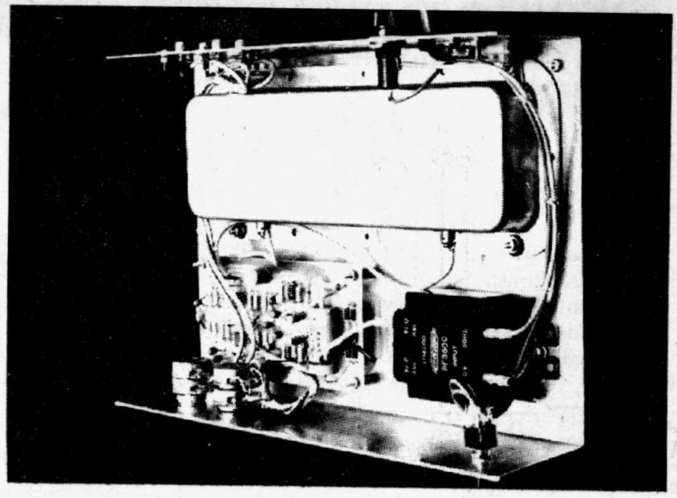


Fig. 4. Method of mounting the hardware and printed circuit board into the chassis is illustrated in this internal view.

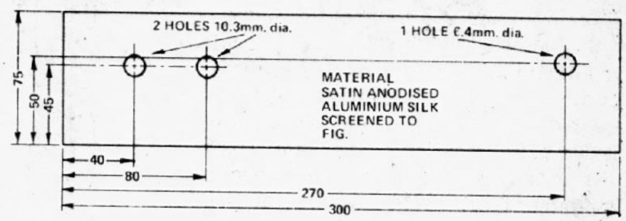


Fig.5 Front panel drilling details.

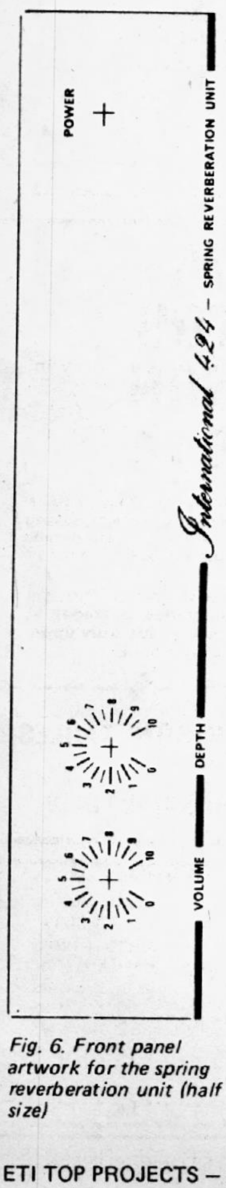


Fig. 6. Front panel artwork for the spring reverberation unit (half size)

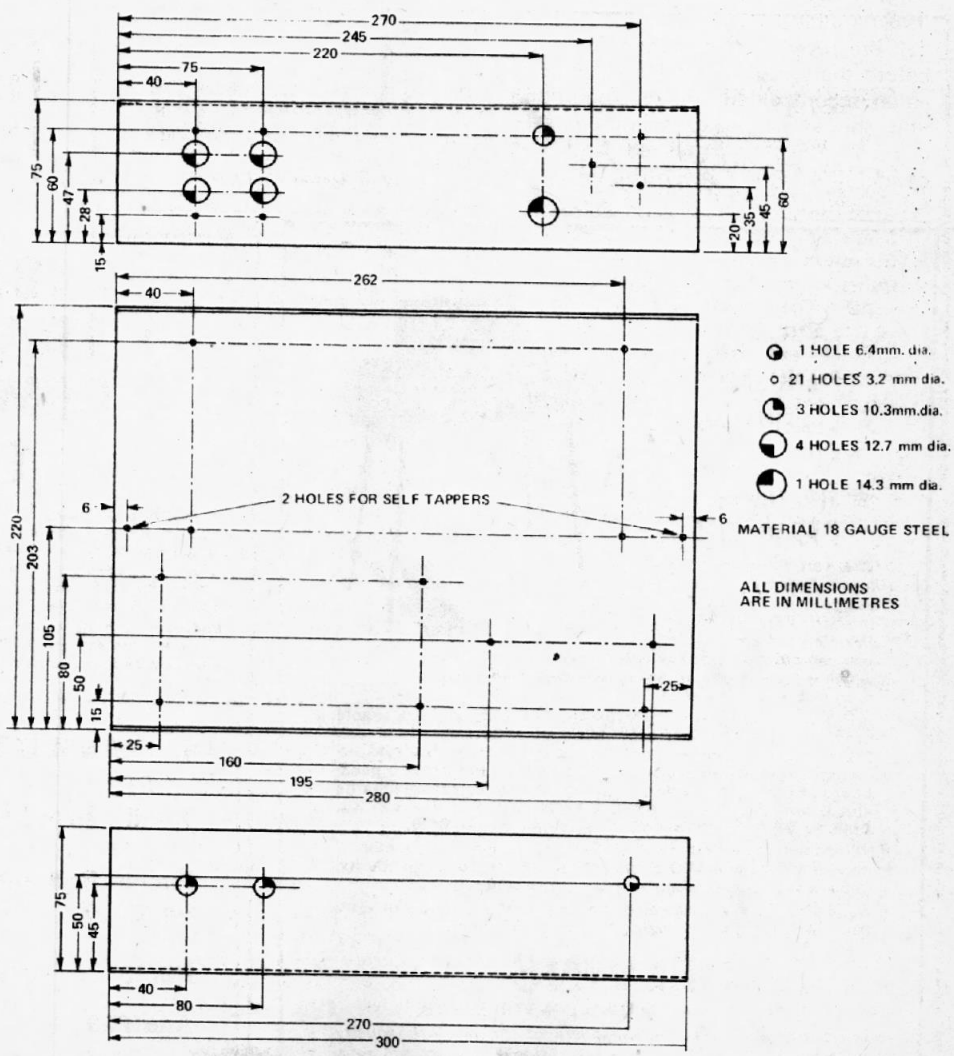


Fig. 7. Dimensions and drilling details of the chassis.

SPRING REVERBERATION UNIT

SETTING UP

As the reverberation spring is a mechanical device, vibration will produce unwanted outputs. Hence it is an inherently noisy device and should be used at a point in the system where the signal level is high.

Two typical points at which the unit may be inserted in the system are:—

1. Between the preamplifier and the main amplifier.

2. After the disc preamplifier, or high level input and the preamplifier.

If inserted between pre and main-amplifiers, i.e. after the volume control, turn the reverb volume control to maximum and adjust the preamplifier volume control such that the main amplifier is just below clipping level. The reverb volume control can then be used to set the level required.

If the reverberation unit is inserted before the system volume control, the volume control on the reverberation unit should be set to maximum (or deleted altogether if desired) and the preamplifier volume control used to set the required level. ●

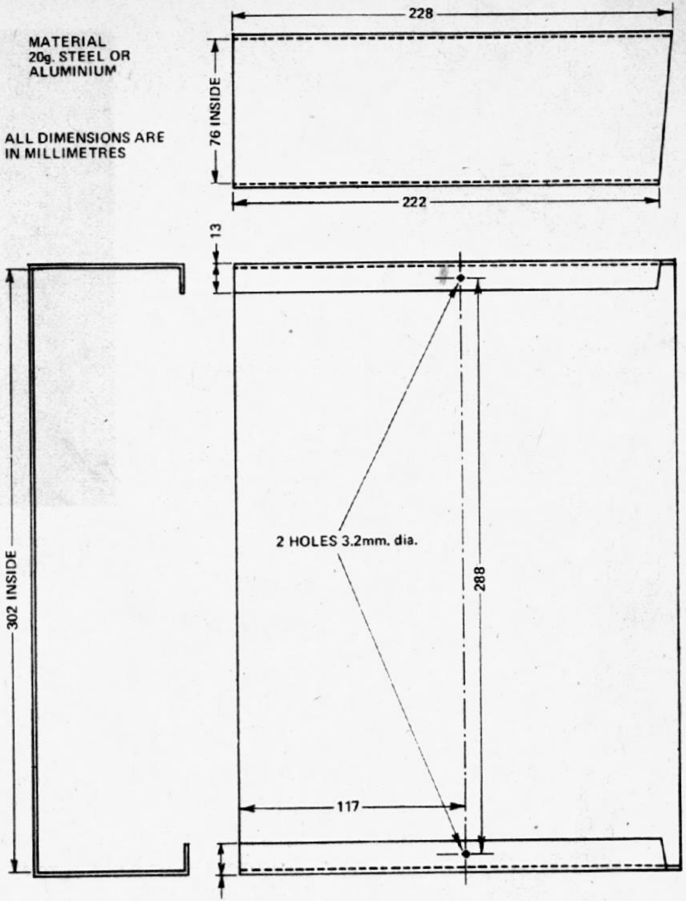
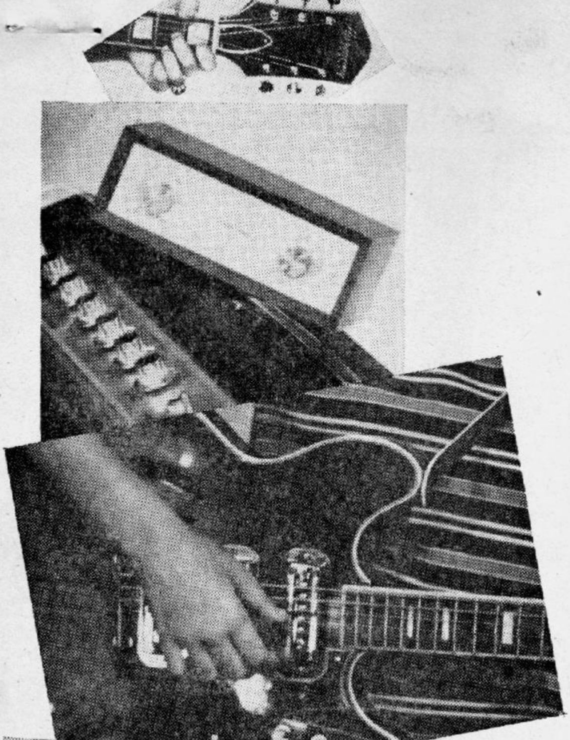


Fig. 8. Detail of the cover.



UNIVERSAL REVERB

Guitar kicks with sound on the rebound

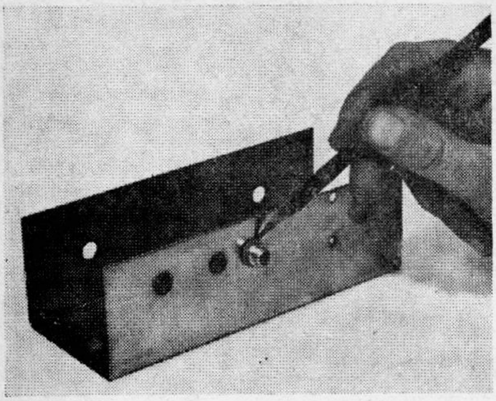
By Herb Freidman
W2ZLF/KBI9457

Hottest item going these days for the in-crowd guitar pickers is Olson Electronics' RA-844, a twenty buck add-on reverberation device that can make a twenty-dollar amplifier (or even your hi-fi amp) sound like a $200 guitar amplifier. You simply plug the electric guitar to the RA-844's input jack, connect the reverb amp to the guitar amp (or the hi-fi), and you can make the guitar sound like it's at the bottom of a five-story cavern.

Or, if you don't go for overpowering echo, you can add just a smidgen of reverb to make the overall sound very bright (like they do down at the local radio station.)

(Continued overleaf)

REVERB



After the reverb unit has been disassembled, the keying jack hole is drilled and the jack installed. Note that the jack is insulated from chassis with fiber washers.

You can even use the RA-844 reverb amp with a dynamic mike to put a little pizzazz on a vocalist.

Few Bucks, Big Buy. Sound like a great buy for twenty bucks? You're right, it is; that's why it's so big with the in-crowd. Only problem is that the reverb amp cannot be keyed in and out while playing. If you want to change back and forth from reverb to "dead strings," you have to stop playing and shut down the depth (reverb) control. But if you're willing to go for a few extra dollars and about an hour's work, you can add a switch (keying) jack to the reverb amp so it can be keyed in and out with a foot switch as you play.

The foot switch modification for the reverb amp is shown in the schematic. The components to be added are shown in the dotted line. The reverb amp components show only the parts value. To understand what the modification does, let's take a quick run through the circuit of the basic reverb amplifier itself.

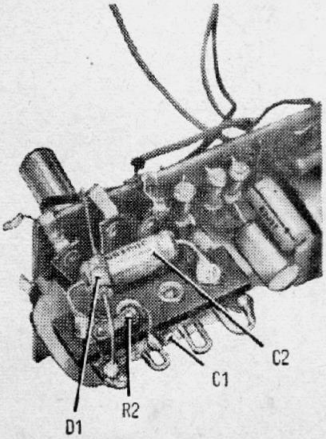
The guitar pickup feeds into the input jack—its level controlled by the 10K volume control—and is amplified by transistor Q1. The unmodified (no reverb) signal is tapped off Q1's collector through a capacitor and the 470K resistor, and is again amplified by Q4. (The loss through the 470K resistor compensates for Q4's gain.) The guitar's signal is then fed to the output jack—which is connected to the guitar amplifier's input jack.

Springy Sound. Now go back to Q1's collector. Note that the guitar's signal is also fed through the transformer and is amplified by the push-pull amplifier (Q2 and Q3) and is then fed to the reverb unit. The heart of the reverb unit is a spring that literally *bounces* the signal back and forth, just like the echoes in a canyon (when you holler hello-o-o-o).

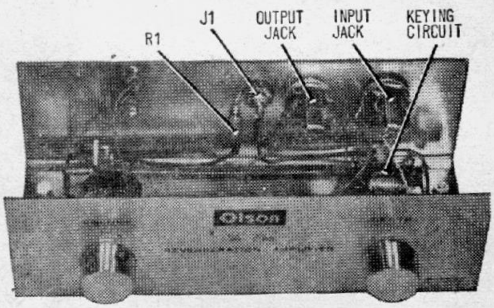
The output from the reverb unit—which now consists of "echoes," or reverberation—is fed through the *Depth Control* into Q4, where it mixes with the direct guitar signal.

When the *Depth Control* is closed, only the direct guitar signal passes through Q4 and there is no reverberation. As the depth control is advanced and more reverb signal is mixed with the direct sound, the total effect at the output jack varies from no reverb, to slight "liveness," to cavernous reverberation.

The Keying Circuit. If one attempted to

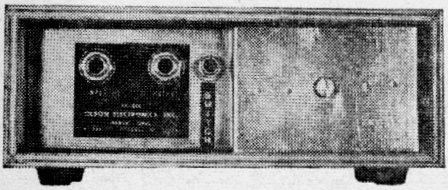


All components of the keying circuit mount on a 4-lug terminal strip which is soldered to the back of the depth control.



Completed modification of the Olson reverberation amplifier. R1 is soldered directly to the keying control jack.

line along the edge of the input and output jack's trim plate and the edge of the battery holder. Drill a 9/16-in. hole exactly midway between the two lines (there is virtually no extra clearance so make certain the hole is centered before you drill). Install a single-mounting-nut type phono jack in the hole.



Rear view of completed modification showing keying control jack where foot switch is plugged in.

Be sure it is insulated from the chassis with shoulder washers.

Thoroughly clean the back of the depth control with some kind of contact cleaner, or radio-cement solvent, and solder a small terminal strip to the back of the depth control as shown in the photographs. The control cover will not take solder if it is not thoroughly clean. If possible, use a miniature terminal strip as supplied in the Allied Radio Terminal Strip Kit.

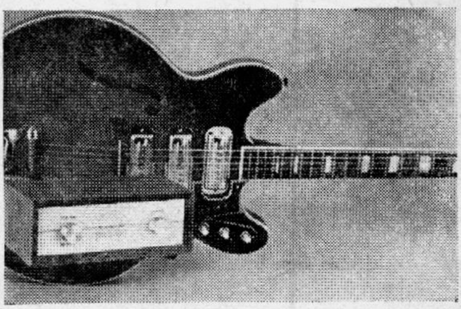
Install C1, C2, R2 and D1 on the strip as shown in the photographs. We used very small 300-uF capacitors to keep things neat; these capacitors as specified in the parts list are somewhat expensive. You can, if you want to cut costs, substitute any cheap capacitor as long as the voltage rating is three volts or more. In addition, C1 and C2 can be reduced to 100 uF, though the keying thump will be somewhat louder than with the bigger capacitors.

Foot switch is disassembled so cable and plug can be attached. Be sure to connect cable to the right two contacts.

Finally, connect a 10-in. wire to the circuit-side power terminal on the volume control and re-assemble the reverb amp.

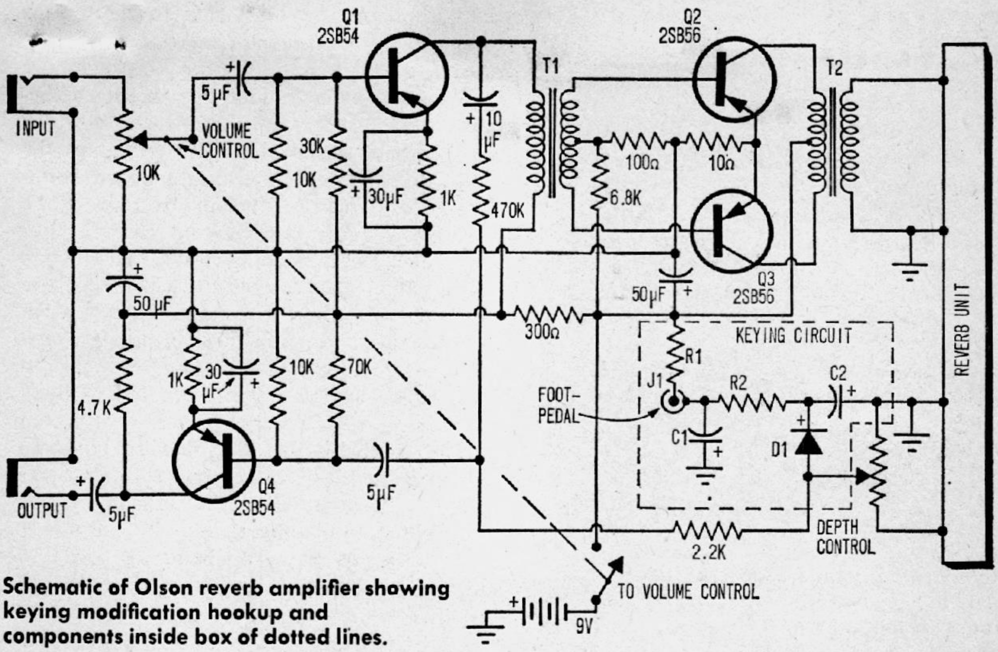
Cut the 10-in. lead just long enough to reach jack J1 and connect R1 between the jack (either terminal) and the lead; insulate the R1/wire joint with tape or spaghetti. Connect the remaining J1 terminal to the C1-R2 junction. Make certain the leads to J1 do not interfere with the reverb unit's spring.

Switching Feet. We suggest the foot switch listed as it's inexpensive, though just about any switch will work. Disassemble the foot switch and connect a length of ordinary lamp cord (or any two conductor cord) to the two switching terminals. The switch has s.p.d.t. terminals, so make certain you select the right two—check for the right ones by having a close look or with an ohmmeter. Now, to finish up the job, connect a phono plug to the free end of the lamp cord being careful not to melt the cord insulation when soldering and causing a short circuit.



With the modification finished and the reverb amp plugged into guitar and amplifier, you're ready to make with the great wild sounds of the seventies.

Connect the switch to J1, the guitar to the input jack, and the guitar amp to the output jack. Key the switch so the reverb effect is off and adjust the guitar, reverb volume, and amplifier volume controls for desired sound level; then key the echo effect in and adjust the depth control for the desired reverb effect. As you play, you can key the reverb effect in-and-out as desired. Try St. Louis Blues or Kansas City with reverb. Man, it's the greatest! ■



Schematic of Olson reverb amplifier showing keying modification hookup and components inside box of dotted lines.

key the reverberation in and out by disabling the power supply to the class B amplifier, the sudden change in load on Q1's collector transformer will cause a drastic change in level of almost 20 dB at the output jack, so this technique is out. If a foot switch was used to short-circuit the reverb signal across the *Depth Control,* not only would there be the possibility of severe hum and noise pickup in the control leads, but there would be severe clicks and pops as the effect was switched in and out.

To avoid clicks and pops, our modification uses delayed diode keying of the reverb signal that shorts (or restores) the reverb signal from the *Depth Control's* wiper contact to ground. With the delayed reverb signal taking approximately one second to key in and out, there is virtually no noise when keying the reverb effect—at most a slight sound well under the music level when the reverb is keyed out. We could entirely eliminate the slight thump but the circuit would get unnecessarily complex; when you're playing a gig no one will hear the thump anyway.

When jack J1 is open, the reverb amp functions normally, since diode D1 is non-conducting (D1 is very slightly back biased by the leakage through Q4's base-input capacitor). When J1 is closed, the battery voltage is applied to the C1-R1-C2-D1 circuit through R1. At the instant J1 is closed by a foot switch, C1, being uncharged, drops

PARTS LIST

- **1—Reverb Amp—Olson RA-844, available from Olson Electronics Corp., $19.95**
- **C1, C2—300-uF, 3-VDC miniature electrolytic capacitor**
- **D1—1N34A diode (don't substitute)** YEA SURE
- **J1—Phono jack**
- **R1—20,000-ohms, ¼-watt resistor**
- **R2—2200-ohms, ¼-watt resistor**
- **1—Foot-switch (Linemaster T51S or equiv.)**
- **1—Phono plug**
- **1—Terminal strip, 4-lug**
- **Misc.—Wire, solder, etc.**

the voltage at the C1-R2 junction and lets it build up slowly, thereby forward biasing diode D1. D1 conducts, shorting the reverb signal to ground through C2.

When J1 is opened, interrupting the battery voltage, C2 discharges through D1 slowly (1 second) giving a slow fade-in of the reverb effect. Because of D1's natural "break-over" voltage, the echo effect is never 100-percent disabled. There is a slight residual effect that adds a smidgen of liveness; you won't know it's there unless you're a golden-eared pro. (Again, it would unnecessarily complicate the installation to get rid of the reverb effect entirely.)

Installing The Keying Circuit. Completely remove the guts of the reverb unit as a single assembly by unscrewing the input and output jack mounting nuts, the volume and depth control mounting nuts, and the reverb unit's two mounting screws. Scribe a pencil

BUILD AN ELECTRONIC

Reverb-b-b Adapter

HOW HOW WOULD WOULD YOU YOU LIKE LIKE TO TO INCORPORATE INCORPORATE A A CONTROLLABLE CONTROLLABLE ECHO ECHO IN IN YOUR YOUR AUDIO AUDIO SYSTEM SYSTEM ? ?

BY DANIEL MEYER

THE ADDITION of electronically generated reverberation to any audio system adds a new dimension to the reproduction of music. By adjustment of the amplitude and decay time of reverb (really an echo), speech, guitar music, or even simple recorded sounds can be made to seem as though you were hearing them in a huge concert hall. When electronic reverb is used with electronic musical instruments, the artist can create a variety of new sounds—ranging from a simple echo to a playing-in-a-barrel effect*.

Most low-cost reverb units can be purchased over the counter at various electronics supply houses. However, you cannot simply connect one between the signal source and the amplifier and expect it to work. The reverb unit must have a driver and an output amplifier, in addition to a resistive mixing circuit needed to combine the straight-through and the reverb sound. The complete reverb adapter described in this article contains all of these electronic elements and is designed to be connected between a conventional preamplifier and power amplifier, either vacuum-tube or transistor types. It is particularly useful with POPULAR ELECTRONICS' "Brute-70" (February, 1967) or the "L'il Tiger" (December, 1967) power amplifiers.

Construction. Putting the reverb adapter (Fig. 1) together is simplified by using the printed-circuit board shown actual size in Fig. 2. Install the components on the PC board in accordance with Fig. 3. The usual PC board construction techniques should be observed—all parts should be mounted close to the board; use rosin-core solder, do not overheat when soldering, and do not form solder bridges across the foil sections. Clip all component leads close to the solder.

The adapter can be mounted on a small metal chassis as shown in Fig. 4. Four small standoffs (approximately 1/4") and associated hardware secure the PC board to the base of the chassis, potentiometer *R13* is mounted on the front panel,

**Don't confuse reverb and tremolo and Leslie effects. Reverb is an echo, tremolo an amplitude variation, and Leslie a warble—as though the sound were changing point source of direction.*

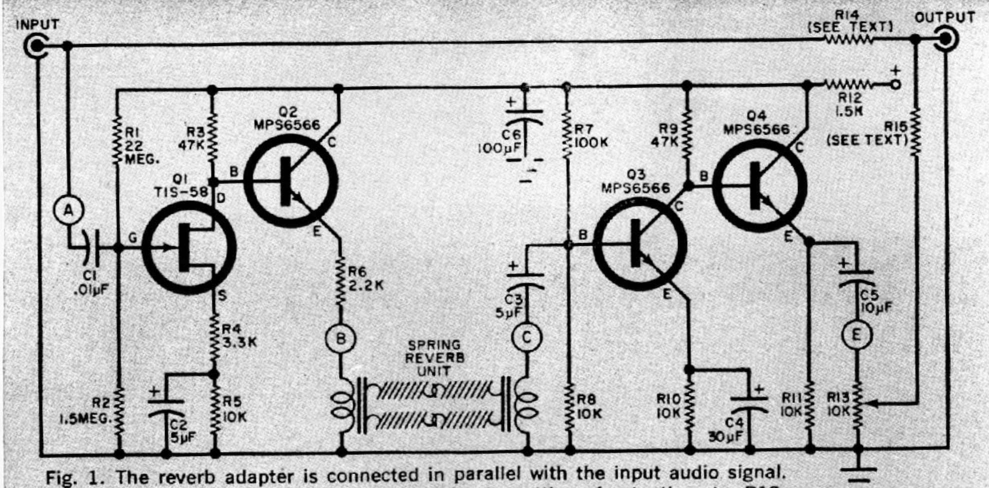


Fig. 1. The reverb adapter is connected in parallel with the input audio signal. Amount of echo to be added is determined by the setting of potentiometer R13.

PARTS LIST

C1—0.01-μF capacitor
C2, C3—5-μF, 15-volt capacitor
C4—30-μF, 6-volt capacitor
C5—10-μF, 25-volt capacitor
C6—100-μF, 50-volt capacitor
Q1—Texas Instruments TIS-58 field effect transistor
Q2, Q3, Q4—Motorola MPS-6566 transistor
R1—22 megohms
R2—1.5 megohms
R3, R9—47,000 ohms
R4—3300 ohms
R5, R8, R10, R11—10,000 ohms
R6—2200 ohms
R7—100,000 ohms
(All resistors ½ watt, ± 10%)
R12—1500 ohms—see text
R13—10,000-ohm linear potentiometer
R14, R15—See text
1—Spring reverberation unit (Gibbs IV-C, Hammond Organ)
*1—Printed circuit board**
Misc.—Phono jacks (4), single-hole mounting type; chassis—see text; wire, solder, spacers, bolts, nuts, etc.

*A kit of the circuit board, chassis, and electronic parts used in the driver amplifier is available from Southwest Technical Products Corp., 219 W. Rhapsody, San Antonio, Texas, 78216, for $8.75 postpaid (#CA-139); the IV-C reverberation unit for $10 plus 2 lb. postage; the circuit board alone for $2 postpaid.

HOW IT WORKS

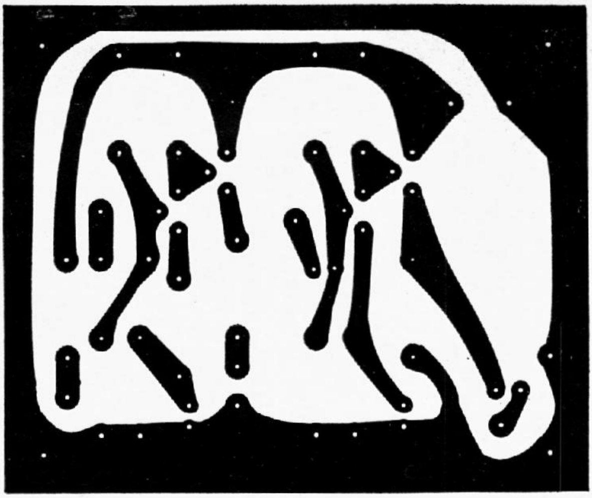
The heart of the reverb adapter is the spring reverberation unit: the electromechanical device that produces the delay and echo effects. Basically, it consists of a pair of contrawound springs (it could be only one spring) suspended between a pair of transducers. When the input transducer is supplied with an audio current, it causes the springs to twist in step. The twisting motion travels down the springs and excites the output transducer, generating an output voltage. Two simultaneous actions occur—there is a slight time delay of the signal in traversing the springs (approximately 25 milliseconds); and because of coupling inefficiencies, some of the signal "bounces" back and forth from transducer to transducer a couple of times, producing an "echo." As each mechanical reflection produces a weaker and weaker signal in the output transducer, multiple weakening of acoustic signals in a "live" room is simulated.

Because the typical loss in the spring reverb system is about 40 dB, an input amplifier (*Q1*) is used. This stage employs a FET to produce a high input impedance (about 1 megohm), which allows the reverb adapter to be used with almost any type of input equipment without loading problems. An emitter follower (*Q2*) matches the input amplifier to the approximately 2000-ohm input impedance of the spring unit.

The electrical output of the spring unit is coupled to amplifier *Q3*, which raises the signal level back to the same level as was applied at the adapter input. Emitter follower *Q4* isolates *Q3* from any loading effects introduced by the external audio power amplifier. Potentiometer *R13* acts as the "reverb level" control and is used to set the desired amount of reverberation. The input audio signal is directly coupled to the output via *R14*, while the reverb is introduced through *R15*.

while the four phono jacks are mounted along the rear apron. Connect short pieces of insulated wire between points A, B, C, and E of the PC board and their respective phono jacks (see Fig. 1).

The value of resistors *R14* and *R15* will depend on what type of audio system the reverb adapter is to be used with. With vacuum-tube equipment, these two resistors should be between 47,000 and 100,000 ohms, with the exact value determined by test. Start with 47,000-ohm units, and remember that some signal loss will be encountered through the use



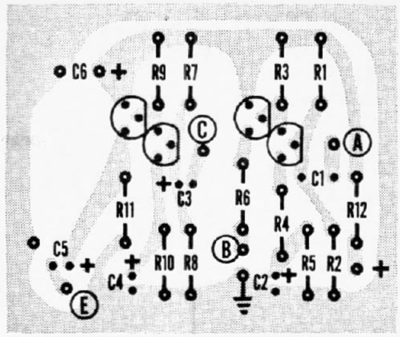


Fig. 2 (left). Actual-size printed board.

Fig. 3 (above). Component installation. Resistors R14 and R15, with potentiometer R13, are installed on the chassis.

of these two resistors. With transistor audio equipment, resistors *R14* and *R15* will be in the range from 1000 to 4700 ohms, again with the best value determined by experimentation. A good compromise is 2200 ohms.

On transistor amplifiers, such as the "Brute 70" and "L'il Tiger," the reverb adapter can be added without a loss in gain by utilizing the present input resistor as one of the mixing resistors (*R14*). Figure 5 shows how this is done. Simply, *R14* in the reverb adapter is omitted and the input resistor of the amplifier is used in its place. In this case, the value of *R15* should be about the same as the input resistor of the amplifier. The value would be about 82,000 ohms with the "Brute 70" and about 4700 ohms with the "L'il Tiger." The "L'il Tiger" was designed with an extra input jack just for this purpose.

Power for the reverb adapter can be obtained from the power supply of

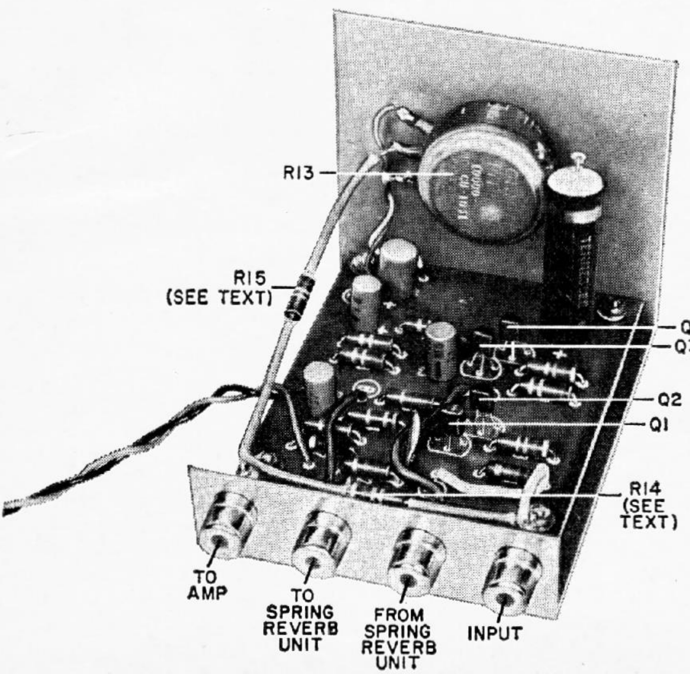


Fig. 4. The finished board can be mounted on a metal chassis with the external connections completed as shown here. Short spacers isolate board from chassis.

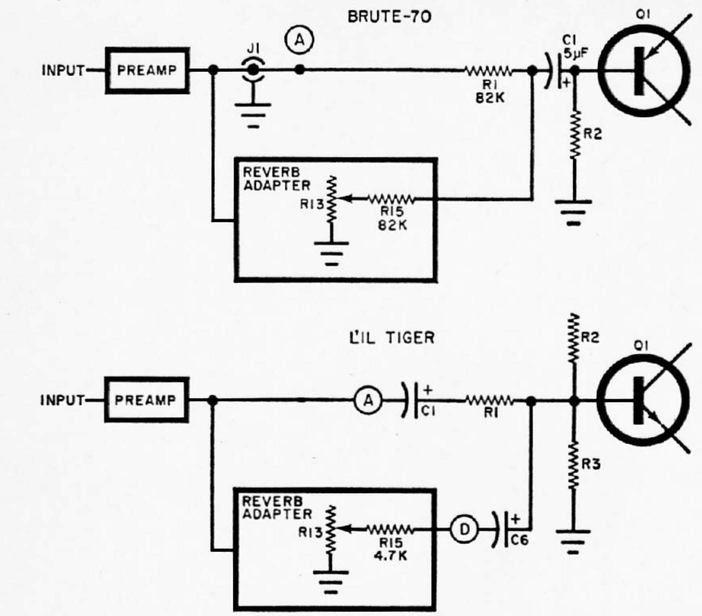


Fig. 5. Method of connecting the reverb adapter to either the "Brute-70" (February, 1967) or the "L'il Tiger" (December, 1967) power amplifiers. The reverb adapter can just as easily be used with other power amplifiers as described in text.

either the "Brute 70" or "L'il Tiger." The value of *R12* shown in Fig. 1 is correct for use with power supplies between 40 and 50 volts. For higher voltage sources, such as are found in vacuum-tube equipment, the value of *R12* will have to be increased to a value that delivers the approximately 30 volts required by the reverb adapter, as shown in Fig. 6.

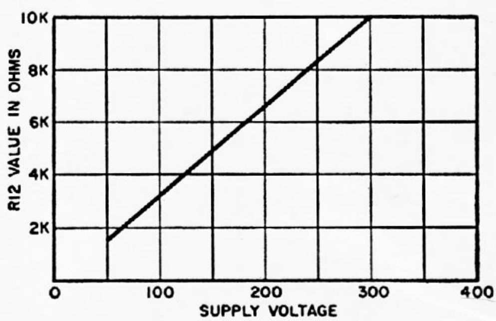
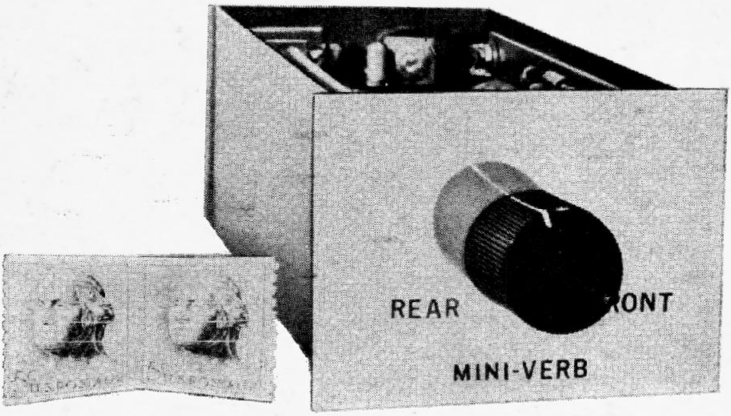


Fig. 6. To determine value of R12 at high voltages, draw a vertical line from the voltage to the reference line, then go horizontally to locate the value.

Installation and Use. On instrument amplifiers, the reverb adapter can be connected either between the instrument and its amplifier, or it can be inserted into the circuit between the preamplifier and the power amplifier stages. If you do not want to "go into" the amplifier, the first approach is the safest—but possibly not the best—as there is a possibility of hum pickup at these low-level stages.

The reverb spring unit is shock-mounted, but the long springs make it sensitive to any undue bouncing, which will produce a "boing"-like sound. Also, the magnetic pickups on the output end of the springs are sensitive to stray magnetic fields and will easily pick up any induction hum from an unshielded—or partially shielded—power transformer in the vicinity. Therefore, always mount the spring unit as far from power transformers as possible, and protect it from any mechanical shocks. In some cases, it may be necessary to wrap the reverb spring unit in fiberglass, or build a cover over it, to prevent acoustic feedback from a nearby speaker.

To obtain reverberation, rotate potentiometer *R13* for the desired amount. When full reverb is used, it may have a "barrel" effect on voices. Some compromise will have to be made, as the best "sound" with music generally causes more echo on voice than most people like. And remember that too much reverberation can be disturbing, unless you are trying for novelty rather than realism. —30—



BUILD THE "MINI-VER"

By DANIEL MEYER

IMPROVED AUTO REVERB CIRCUIT USES NEW MINIATURIZED UNIT; SUITABLE FOR ATTACHMENT TO STEREO HI-FI RIGS

EVERBERATION systems are offered as "accessory" equipment with many cars, included in many high fidelity systems or electronic organs, and even in public address amplifiers. Reverb systems give music a warmer tone and add a feeling of spaciousness by simulating the reverberation—or echo—effect of large concert halls. A car—or even the average living room—is not big enough to have an audible natural reverberation. So adding electronic reverberation makes the reproduction more pleasant and gives a feeling of concert-hall space.

The "Mini-Verb," an improved and updated version of the system described in POPULAR ELECTRONICS, Feb., 1966, was originally built for use in a car. It was miniaturized by using a smaller delay line than the one in the older system. However, it is also usable with your home hi-fi installation and can be hooked up to most stereo systems with little trouble. Quality and output power have been kept high in spite of the fact that the size has been brought down.

A standard high-fidelity solid-state circuit is used in the amplifier. The transformerless class-B output stage will deliver at least 3 watts into a 3.2-ohm speaker with less than 1% distortion. Silicon transistors are used throughout for maximum temperature stability.

The small delay line reverb unit (Gibbs Type VII) makes possible a compact system that can be installed almost anywhere. The case measures 2″ x 2¾″ x 6″ and includes the fader control and power switch. If you have room in your

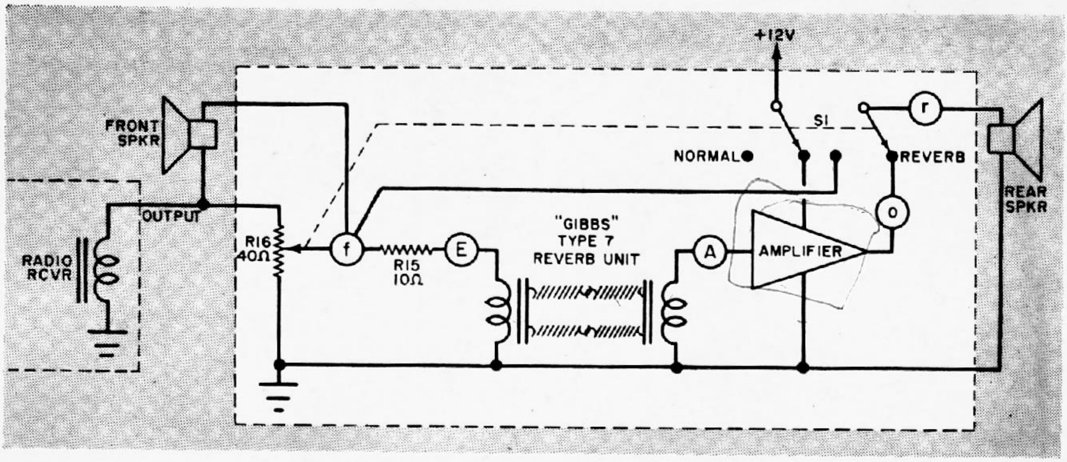


Fig. 1. In an automobile installation, reverberation is achieved by delaying and "reverb'ing" the sound from the rear seat speaker. The delay of sound fools the ear into believing the sound is in a large concert hall.

PARTS LIST

C1, C7—10-μF, 15-volt electrolytic capacitor
C2—30-μF, 6-volt electrolytic capacitor
C3, C4—5-μF, 15-volt electrolytic capacitor
C5—200-μF, 6-volt electrolytic capacitor
C6—500-μF, 25-volt electrolytic capacitor
C8—100-μF, 15-volt electrolytic capacitor
D1, D4—1N645 silicon diode, 600-mW
D2, D3—1N3754 single-ended diode (RCA)
F1—½-A Slo-Blo fuse
Q1, Q2, Q3—MPS-3708 transistor (Motorola)
Q4—MPS-3638 transistor (Motorola)
Q5, Q6—TIP-24 transistor (Texas Instruments)
R1, R3, R6, R10—4700-ohm, ½-watt resistor
R2, R7—22,000-ohm, ½-watt resistor
R4—1000-ohm miniature trimmer control (Mallory MTC-1 or similar)
R5—2200-ohm, ½-watt resistor
R8—25,000-ohm, ½-watt resistor
R9—100-ohm, ½-watt resistor
R11—1000-ohm, ½-watt resistor
R12, R13—220-ohm, ½-watt resistor
R14—470-ohm, ½-watt resistor
R15—10-ohm, ½-watt resistor
(All resistors 10% tolerance)
R16—40-ohm, 5-watt wire-wound potentiometer with slide actuator
Rb—220-ohm, ½-watt resistor—see text
S1—D.p.d.t. slide switch
1—Gibbs Type VII reverberation unit
Misc.—Case, printed circuit board, fuse holder, nuts, bolts, knob, wire, solder, etc.

A complete kit of parts including a special case is available from DEMCO, 219 W. Rhapsody, San Antonio, Texas 78216, for $16.74 (postpaid). Prices of individual components are available on request.

car to mount a speaker selector switch, you will probably have enough room for the reverberation system.

How It Works. The heart of any reverberation system is the audio delay line. It consists of two electromagnetic transducers and a pair of different-diameter springs coupling them. Audio frequency signals drive the input transducer, which twists the springs slightly. This mechanical motion travels down the springs and creates an electrical signal in the output transducer. Not all the mechanical energy is reconverted to an electrical signal—some energy continues to travel back and forth and gradually decays, resulting in both a delay and a decay of the original sound, as with natural echoes.

The audio signal is split between the speakers through a fader control (*R16*) and selector switch *S1* (Fig. 1). When the selector switch is in *normal* position, the same signal is applied to both front and rear speakers. The fader serves as a variable divider to balance or shift the sound output from each speaker as desired. When the fader control knob is pulled out, *S1* switches the power onto the reverb amplifier and connects the rear speaker to the amplifier's output.

The signal from the radio now drives the front speaker and the input transducer through *R15*. The output transducer of the reverb unit is connected to a high-gain amplifier (Fig. 2). The amplifier makes up the 40 to 45 dB loss in the delay line reverb unit. In this circuit, *Q1* drives voltage amplifier *Q2*, which is directly coupled to a pair of complementary driver transistors, *Q3* and *Q4*. The driver transistors drive the class-B output pair on alternate half cycles.

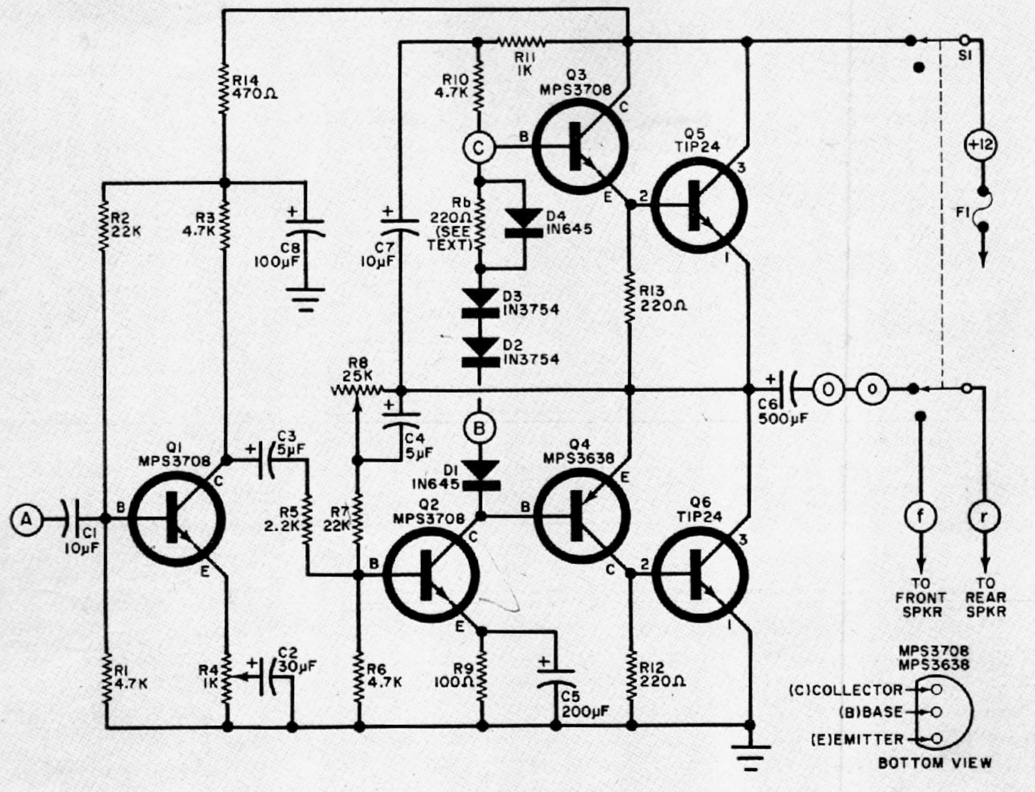


Fig. 2. Schematic diagram shows the simple solid-state high-gain amplifier required to compensate for the audio signal in the reverberation springs. The amplifier has a 3-watt output and is powered by 12 volts d.c.

The diodes between the base of *Q3* and the base of *Q4* provide a small forward voltage bias to prevent crossover distortion and also provide temperature compensation. Diodes *D2* and *D3* are in direct physical contact with the output stages, as shown in Fig. 3. Any heating which would increase output transistor idle current is quickly sensed by the diodes. The heat reduces the diode voltage drop, reducing the transistor forward bias and idle current.

The reverb system's gain is controlled by *R4* and the fader (*R16*)—*R4* for the coarse settings and the fader to make variations to suit the taste of the listener. The amplifier is efficient and draws only about 10 mA with no signal input. At full 3 watts output, the current is 0.4 to 0.5 ampere.

Construction. To make the system small but still easy to assemble, the amplifier is built on an etched board and the whole system is housed in a specially designed case. (See Fig. 4). If you follow the instructions, there should be no construction problems.

Begin by mounting the power transistors and diodes in the rear of the case. Use a 6-32 x ½″ screw, with a shoulder washer on the outside of the case and a mica spacer between the inside of the case and the transistor mounting flange. Be sure to coat both sides of the mica washer with silicone grease to insure good heat transfer.

The 1N3754 bias diodes are pushed into their clips and mounted with the same screw that holds the transistors. The diode leads are insulated from the case, so the clips can contact the transistor mounting tab. Turn the diodes so that the red cathode identification dots are opposite each other. Cut the lower leads and solder them together—cathode of one to anode of the other—as shown in Fig. 3. Mount the terminal strip next

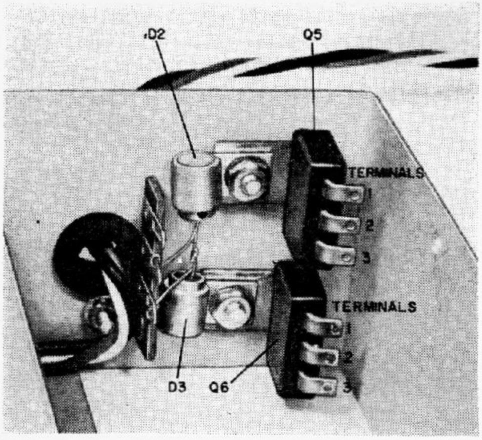


Fig. 3. The npn output transistors are of unusual construction and have not previously appeared in POPULAR ELECTRONICS projects. Each transistor is bolted to the chassis wall along with biasing diodes D2 and D3. You must follow this plan.

to the diodes and connect the remaining two leads to the center and lower lugs. Check for possible shorts between the transistors and case.

Now mount the fader control and reverb unit. Wire the leads for the front speaker, radio input, and ground to the fader control. Dress the wires behind the reverb unit and to the case bottom, install the grommet, and bring the wires out through the grommet.

Mount the various parts on the circuit board (Fig. 5) as indicated by the printed part numbers (see p. 46). Be sure the electrolytic capacitors and diode are mounted with correct polarity. Solder leads to *Q5* and *Q6* as shown in Fig. 3. Connect wires to points *+12V* and *r*, for the power and rear speaker, and to *B* and *C* for the bias diodes. Use a piece of lamp cord or equivalent to wire the rear speaker.

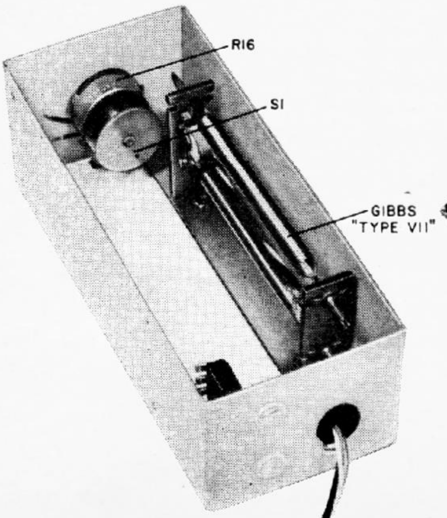
Connect the ground side of the rear speaker to the board's ground strip near the output transistor connection. Do *not* attempt to use the frame of the car for the ground lead to the rear speaker—this can result in noise and even circuit oscillation. Connect a short piece of hook-up wire to point *f*. Connect the green lead from the reverb unit's output (red coil) to point *A* and the green lead from the input to point *E*. Be sure all wires and connections are soldered.

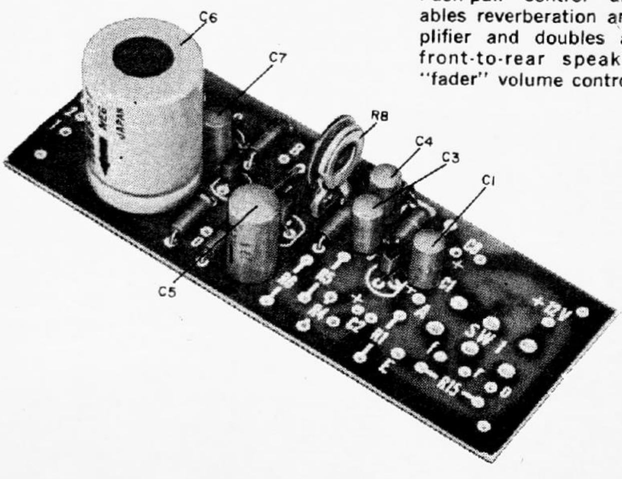
Mount the circuit board on the side of the case (Fig. 6) with *4-40* x 3/8" machine screws. Be sure the switch knob slides between the plates at the rear of the fader control. Use quarter-inch spacers between the case and the circuit board. Connect the wire from point *f* to the

Fig. 4 (below). Due to the intense interest in this project, the author in conjunction with POPULAR ELECTRONICS offers a special mounting case.

Fig. 5 (below). View of partially completed printed circuit board shows location of some of the components. The numbers alongside C6 pertain to connections to transistor Q6.

Fig. 6 (right). Photo of the completed unit. Push-pull control disables reverberation amplifier and doubles as front-to-rear speaker "fader" volume control.

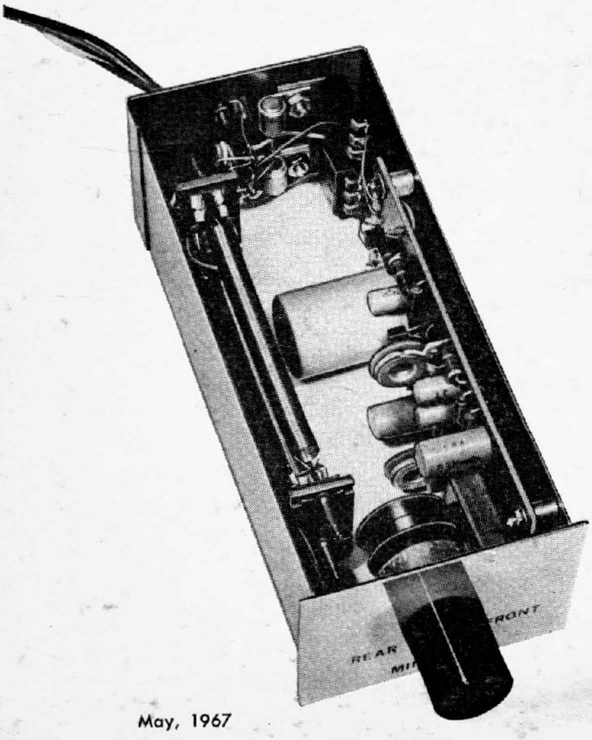




arm of the fader control. Mount a soldering lug under the mounting screw at the bottom front of the board, and connect it to the black wires from the reverb unit coils and to the ground side of the fader control, using a short piece of bare wire.

Connect the leads from the board to the power transistors. The numbers on the board and the transistors must match: 1 to 1, 2 to 2, etc. Connect the wire from point *C* to the upper (unused) lug on the terminal strip. Install *D4* and *Rb* on the terminal strip and, observing polarity, the other end to *D3*'s anode. Connect a wire from point *B* to the cathode side of *D2* on the terminal strip. Connect a lead at *+12V* and run it out through the grommet in the rear of the case to the fuse holder. (This in-line type holder can be picked up at an auto supply house, and is used with a half-ampere Slo-Blo fuse.) Label the leads—to protect the transistors.

Installation. The circuit is designed to work with an ungrounded front speaker. If one side of your speaker is grounded at the speaker frame rather than at the radio, simply clip the ground lead and splice on a piece of wire for connection to the reverb unit. Be sure the car has 4- or 8-ohm speakers. (There are some 40-ohm systems around which require a matching transformer from the radio to the reverb unit.) Also, be sure the speaker is not "hot." Some speakers have 12 volts on the leads.



Connect the +12-volt lead from the fuse holder to the radio, or connect it to the accessories terminal on the ignition switch. Connect the ground lead to an unpainted screw or to some other point that you are sure is a good ground on the car's frame. Connect the lead from point *f* on reverb input to the front speaker. Run the two rear speaker wires to the rear speaker.

Turn the radio on, with the reverb unit knob pushed in. The control should vary the volume of the front and rear speakers as it is turned, with near-zero volume on the front speaker at the extreme rear position, and vice versa. Now pull the fader knob out. You should have about the same volume as before with *R4* set for full gain (the resistor is partially bypassed by *C2*). Sound will probably be best with a bit less volume on the rear speaker when the system is in the reverb position. Set the fader control in the center position and adjust *R4* for the most pleasing rear speaker level.

The reverb unit cover can be fastened under the dash with sheet-metal screws, or mounted through the dash and held by the fader control bushing.

Testing. The system will work quite well without exact circuit adjustments. You can, however, get lower distortion and slightly greater output if you have the proper equipment to make a few tests.

Resistor *Rb* is specified as 220 ohms. This is slightly lower than the best value but safe in all cases. A slightly higher value may reduce crossover distortion. To check for the proper value, connect a milliammeter (VOM) in the +12-volt lead. Short the meter leads and turn on the amplifier. Unshort the meter leads and read the idle current. Now short *Rb* with a clip lead and watch the meter reading. If it drops between 5 and 10 mA, the value of *Rb* is okay. If the drop is less, increase the value of *Rb* to 270 ohms, and check again. The initial current reading should not be more than 15

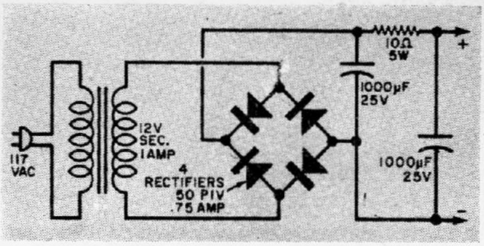


Fig. 7. This simple power supply will enable the builder to operate his Mini-Verb off the 117-volt a.c. lines. All components are easily purchased.

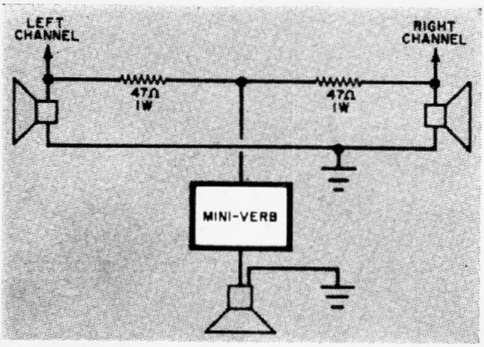


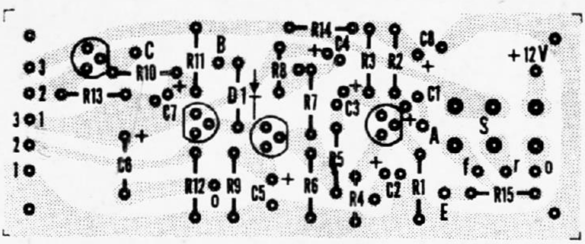
Fig. 8. Reverberation in your home calls for using the Mini-Verb as a third channel. Mixed signal from the right and left channels is derived as shown in this diagram. See text at right for more information.

mA, and *Rb* must not be increased past the value that gives a 10 mA increase in current.

You can adjust *R8* in either of two ways. If you have only a voltmeter, it can be set for a reading of +6 volts at the emitter (terminal 1) of *Q5*. If an oscilloscope and signal generator are available, drive the amplifier to full output (clipping level) with a 4-ohm load at about 1 kHz and adjust *R8* for symmetrical clipping of the observed waveform.

"Stationary" Applications. To use the "Mini-Verb" with your high fidelity system (or public address equipment), you will need a 12-volt power supply that can deliver 500 mA with good regulation and low hum. A typical circuit is shown in Fig. 7.

To use the "Mini-Verb" with your stereo system, follow the circuit of Fig. 8. (The resistors should be about 47 ohms, 1 watt; reduce the resistance if reverb volume is too low.) This circuit will give you a driving signal that contains information from both channels. Little separation remains in reverberation sound, so two channels are unnecessary in the reverb system. —30—



Actual-size printed circuit template—for those who like to make their own—is given below. Component location is shown on the unwired side of the board in the diagram at the left.