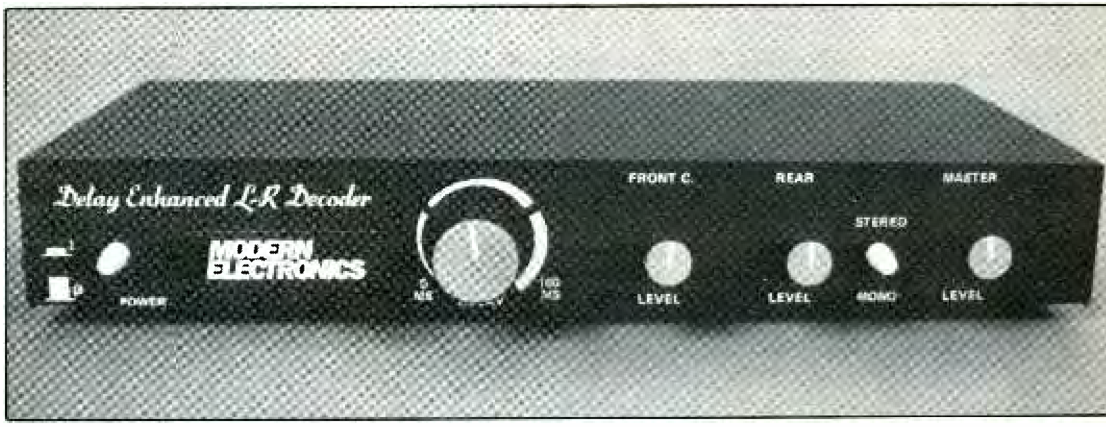


"Surround-Sound" Enhancer

Adding this accessory to your hi-fi system recovers ambience information from recordings and adds delay to provide realistic sound

By John H. Roberts

Ambience—so-called "room sound"—is the missing ingredient that makes even the best stereo system sound flat and lifeless when compared to live-performance sounds. Over the years, various techniques have been devised and employed in consumer products to simulate or recreate the ambience of the live performance. To some degree, all have been successful. Until now, however, few have offered the advantages of the Delay Enhanced L – R Decoder described here.

What makes this Decoder project a superior performer is that it uses two of the time-honored techniques that have met with relatively large success. It offers both time delay and L – R matrix (ambience-recovery) capabilities in the same accessory. Combining the two techniques results in a system that works better than either alone.

Performing L – R matrix recovery before adding time delay cuts the expense of using two complete channels, with no deterioration of the ambience information. Adding delay to the L – R matrix corrects that system's localization problems.

Some Background

Artificial reverberation, generated by either mechanical or electronic delays, offer some improvement over the unprocessed sound signals normally delivered to the speakers of a hi-fi system. However, even the most elaborate delay system requires adjustments to make the simulated reverberation match different recordings, the result often sounding unnatural. Discrete and matrixed four-channel recording had the capability of reproducing ambience but was not properly utilized and, hence, fell out of favor.

A certain amount of ambience is automatically captured whenever a microphone is located more than a few feet away from any sound source. Therefore, most recordings already contain significant amounts of ambience just waiting to be unlocked. To unlock it, you need a special signal processor.

Two popular techniques to extract this ambience information from conventional recordings are *time delay* and *L – R matrix.* Pure time delay, not to be confused with delay generated artificial reverberation, was discovered by E. Roerbaek Madsen, who was searching for a way to dramatically improve audibility in conventional recordings.

Some consumer hi-fi delay devices

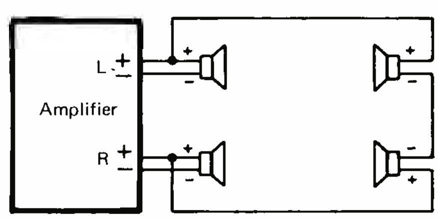


Fig. 1. The Dynaquad™ L – R ambience-recovery scheme places the rear speakers in parallel with the speakers used in the front.

"Uses two time-honored sound-enhancement techniques."

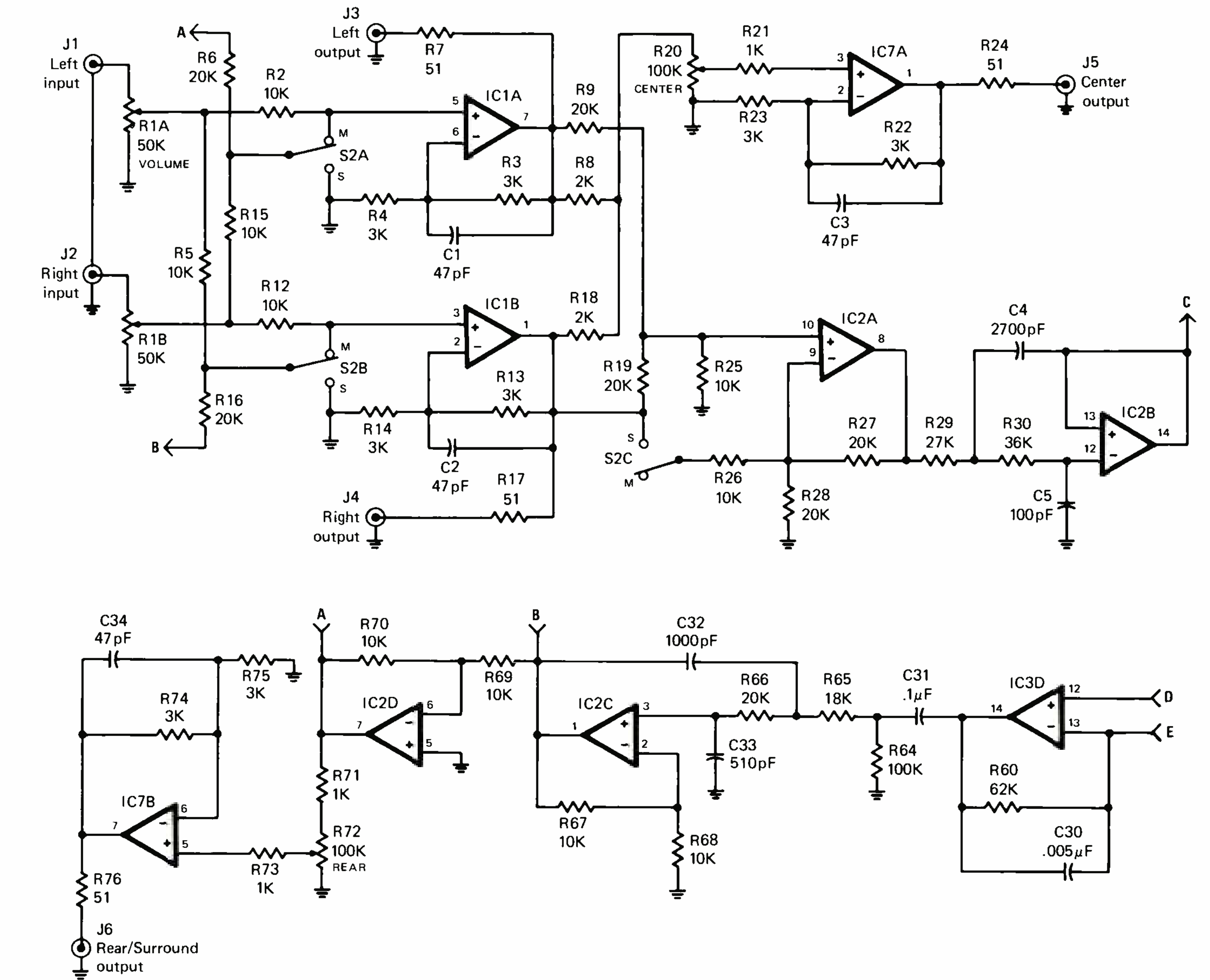
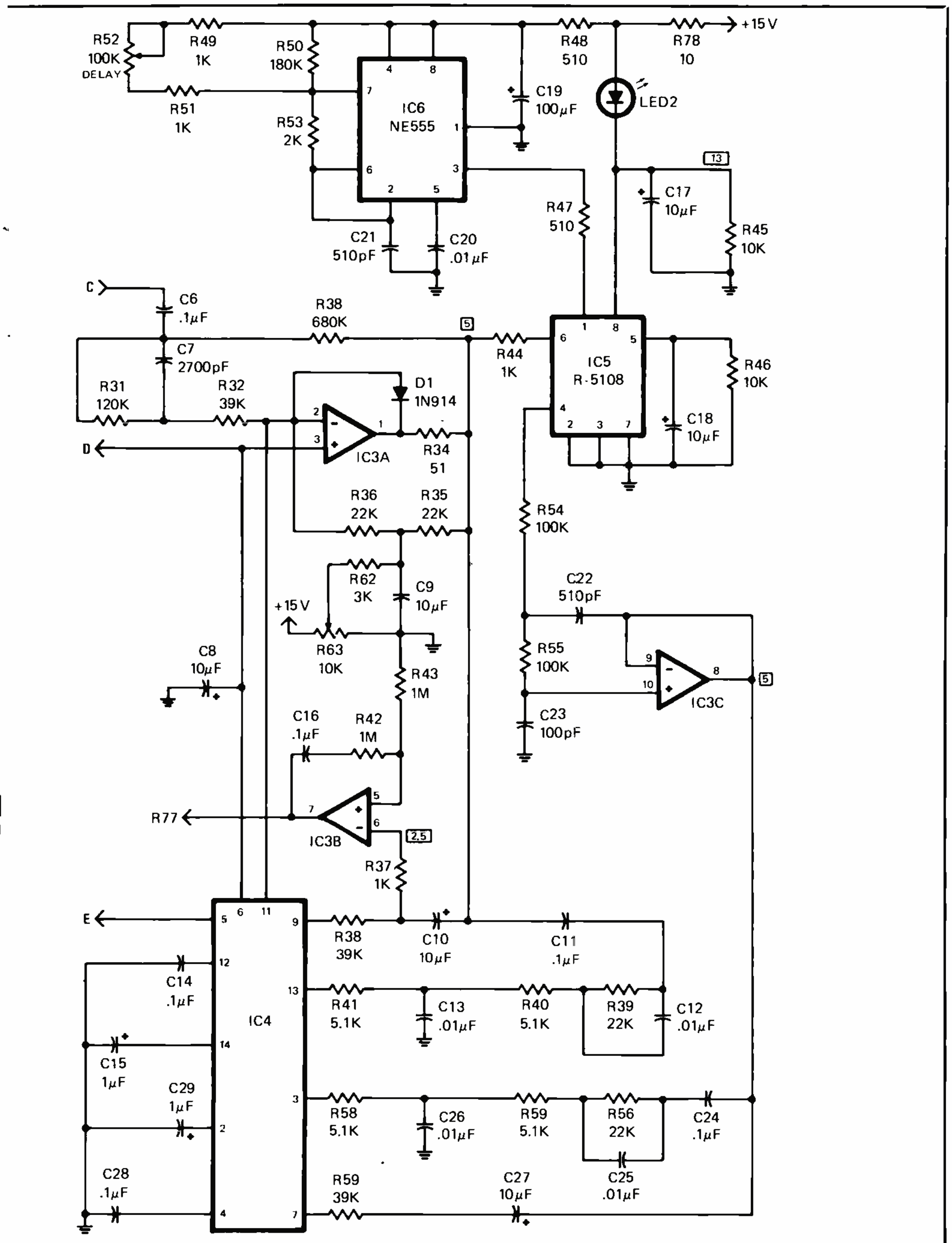


Fig. 2. Entire schematic diagram of the Delay Enhanced L – R Decoder, minus its built-in power supply. Arrowed letters in each part match up with the same arrowed letters in other parts (A goes to A, B to B, etc.). Numbers shown in small boxes are voltages that can be measured with a good circuit, provided here in case you have to troubleshoot

of the late 1970s were based on the Madsen principle. This approach reproduced a delayed version of the front signal through additional speakers located off to the sides or rear of the main front speakers in the listening room. It is interesting to note that this technique also works with mono recordings.

L – R matrix, perhaps better known as Dynaquad™, was a passive system that simply connected a pair of rear speakers differentially across the "hot" or + terminals of the left and right output channels of an amplifier (see Fig. 1). This system operated under the principle that sounds coming directly from an orchestra arrive at both microphones almost simultaneously, while room-reflected sounds arrive from various odd angles with relatively large time differences between the two microphones.

By subracting the output of one microphone from the other, the almost identical sounds would cancel out, leaving predominantly ambience. This system improved the ambience situation quite a bit, but it



suffered from poor front imaging and localization. That is, sounds occasionally appeared in the back that were not supposed to be there.

A note of caution: If you wish to experiment with the passive L – R matrix, keep in mind that the rear speakers will be in parallel with those in the front and may present an unacceptably low impedance to your power amplifier. Additionally, peaks occurring in only one channel may upset the protection circuitry of the other, which must then sink the full current being sourced, even if it is sitting at 0 volt.

In the Delay Enhanced L – R Decoder presented here, the mechanism that allows the time delay to cure the passive L – R matrix's front localization problem is the same mechanism that caused it in the first place, namely the Hass effect (see "Haas Effect" box).

In the passive L – R system, sounds from the rear speaker will reach you before the sounds from the front speakers because they are closer. Because of the Hass effect, your brain will attempt to lock onto these rear sounds, causing you to hear false localization.

Delaying the sounds being fed to the rear speakers by 20 to 30 milliseconds will ensure that the sounds from the front speakers will always arrive at the listener's position before the sounds from the rear do. In fact, when properly adjusted, this project should make it so that you never actually hear the rear speakers as discrete sound sources. What you will get, then, is ambience that simply "surrounds" you as you listen.

The L – R matrix plus delay works equally well with a wide range of program sources, including many that were not originally recorded using stereo microphones.

About The Circuit

The schematic diagram for the ambience-recovery system is shown in three parts, in Fig. 2A, B and C. The system is designed to work with a wide range of both mono and stereo program sources. The output of the ambience/surround channel, available at *J6* in Fig. 1C, offers a full 12-kHz bandwidth. (Listening tests have revealed that the 6-kHz bandwidth used in past designs was inadequate for best ambience extraction from CD and other high-quality recordings.)

In addition to the normal L – R + delay, or surround, mode, a stereo synthesizer is built in to enhance playback of monophonic program sources. Jacks *J3* and *J4* in Fig. 2A provide the outputs for both normal stereo and synthesized stereo, the latter when a mono source is connected to input jacks *J1* and *J2* and switch *S2*

PARTS LIST

Solid-State Devices
D1, D2—Not used
D3 thru D6—1N4002 rectifier diode
D7—1N914 signal diode
IC1, IC7—NE5532 or TL072 dual op amp
IC2, IC3—TL074CN quad op amp
IC4—NE572 dual compander
IC5—R-5108 2048-stage ASR (Reticon)
IC6—NE555 timer
IC8—78M15 +15-volt regulator
IC9—79M15 −15-volt regulator
LED1, LED2—Light-emitting diode

Capacitors
C1, C2, C3, C34—47-pF, 10%, disc
C4, C7—2700-pF, 5%, polystyrene
C5, C23—100-pF, 5%, polystyrene
C6, C11, C14, C16, C24, C28, C31—0.1-μF, 5%, polystyrene
C8, C9, C10, C17, C18, C27, C38, C39—10-μF, 35-volt electrolytic
C12, C13, C25, C26—0.01-μF, 5%, polyester
C15, C29—1-μF, 25-volt electrolytic
C19—100-μF, 16-volt electrolytic
C20—0.01-μF disc
C21, C22, C33—510-pF, 5%, polyester
C30—0.005-μF, 5%, polyester
C32—1000-pF, 5%, polystyrene
C35, C36—1000-μF, 35-volt electrolytic
C37, C40 thru C49—0.1-μF disc
C50—0.022-μF, 600-volt disc

Resistors (all ¼-watt, 5%):
R2, R5, R12, R15, R25, R26, R45, R46, R67 thru R70—10,000 ohms, carbon-film
R3, R4, R13, R14, R22, R23, R62, R74, R75, R77—3000 ohms
R6, R9, R16, R19, R27, R28, R46, R66—20,000 ohms
R7, R17, R24, R34, R76—51 ohms
R8, R18, R53—2000 ohms
R10, R11, R78—10 ohms
R21, R37, R44, R49, R51, R71, R73—1000 ohms
R29—27,000 ohms
R30—36,000 ohms
R31—120,000 ohms
R32, R35, R36, R39, R56—22,000 ohms
R33—680,000 ohms
R38, R59—39,000 ohms
R40, R41, R57, R58—5100 ohms
R42, R43—1 megohm
R47, R48—510 ohms
R50—180,000 ohms
R54, R55, R64—100,000 ohms
R60—62,000 ohms
R65—18,000 ohms
R1—Dual 50,000-ohm, linear-taper potentiometer
R20, R52, R72—100,000-ohm, linear-taper potentiometer
R63—10,000-ohm trimmer potentiometer

Other Components:
F1—¼-ampere pigtail fuse
J1 thru J6—Phono jack
S1—Dpdt push-push pc-mount switch
S2—4pdt push-push pc-mount switch
T1—28-volt, center-tapped transformer

Miscellaneous:
Printed circuit board; sockets for ICs; suitable enclosure; line cord; strain relief; control knobs; panel lens for LED1; machine hardware; hookup wire; etc.

Note: The following items are available from Phoenix Systems, Inc., PO Box 628, Manchester, CT 06040 (tel. 203-643-4484): No. P-250-DL complete kit of parts for $179.00; No. P-250-B etched and drilled pc board for $19.00; No. P-250-T 28-volt, c.t. pc-mount transformer for $7.00; No. R-5108 Reticon 2048-stage ASR IC for $30.00; NE5532N dual op amp for $2.25; TL074CN quad bi-FET op amp for $2.50; NE572N dual compander for $3.25; 78M15 regulator for $1.50; 79M15 regulator for $2.50; No. P-2X50KB dual 50,000-ohm, linear-taper potentiometer for $2.50; No. P-100KB 100,000-ohm, linear-taper potentiometer for $1.00; No. S-1 dpdt pc-mount switch for $1.00; No. S-2 4pdt pc-mount switch for $1.50. Add $1.00 S&H for orders of less than $10.00, $2.00 on COD orders. Connecticut residents, please add 7.5% sales tax.

is set to MONO. Also, an L + R, front-center fill, output is provided at *J5* in Fig. 2A, for use in small movie screen applications and in hi-fi setups as a mono feed to a subwoofer.

For convenience of setup and use, the circuit includes a master volume control (*R1* in Fig. 2A) and separate level controls for the front-center (*R20* in Fig. 2A) and surround (*R72* in Fig. 2C) outputs. Both outputs are capable of boost and cut relative to the master volume control. Hence, it is a simple matter to correct for differences in sensitivity between the front and rear speaker systems. Once relative gains are set, the level controls track the master volume control for routine system level changes.

This circuit uses the newest ASR integrated circuit, the R-5108 from Reticon, to extract ambience signals from stereo sources. This device, shown as *IC5* in Fig. 2B, has 2048 stages of delay, which is twice as long as the popular SAD-1024 and half as long as the SAD-4096. This new chip has the biphase clock drivers and output sample-and-hold circuit built into a smaller chip that is housed inside a compact 8-pin DIP package.

Input and output filters, tuned for −3 dB at 12 kHz, condition the audio in the surround channel to avoid sampling rate aliasing and to smooth out the output waveform. An NE572N, *IC4* in Fig. 2B, companding noise-reduction chip is used

Specifications

Input impedance	10k ohms or greater
Output impedance	50 ohms
Maximum output	
into hi-fi load	8 volts
into 600 ohms	7.5 volts
Gain (each output)	+6 dB to full off
Delay time	5 to 30 ms, adjustable
THD + N	
direct	<0.01%, 20 Hz to 20 kHz
delay	0.5% nominal, 100 Hz to 10 kHz
Noise (IHF A)	
direct	< −100 dBV
delay	−91 dBV
Frequency response	
direct	dc to 20 kHz +0/−0.25 dB
delay	20 Hz to 12 kHz ±3 dB

"Simulates stereo sound from mono sources."

around the delay chip for noise-free performance with even the most dynamic sources available. The NE572N is an improved version of the popular NE570.

NE555 timer *IC6* in Fig. 2B generates the timebase for the system. The frequency of this time base controls how long an audio sample takes to work its way through the ASR. Potentiometer *R52* provides the means for adjusting the clock frequency to vary the system delay time. High-slew-rate op amps are used throughout the system to deliver maximum audio fidelity.

The power supply for the system is shown in Fig. 3. Note that this full-wave bridge circuit provides full regulation of both the +15- and −15-volt buses. The schematic also shows the pins to which the buses connect on the ICs.

Construction

Owing to its complexity, it is highly recommended that you assemble the delay system's circuit on a printed-circuit board. You can fabricate your own board, using the etching-and-drilling guide given in Fig. 4. Alternatively, you can purchase a ready-for-installation board from the source given in the Parts List. Whichever way you go, you will note from the components-placement guide in Fig. 5 and the photo of the interior of the project in Fig. 6 that all components except the various input and output jacks mount directly on the circuit board.

There is nothing critical about assembly, except that you must carefully observe the polarities and orientations of the integrated circuits, diodes, light-emitting diodes, and electrolytic capacitors before soldering them into place. Sockets are recommended for all ICs, though you can, if you wish, install these devices directly on the pc board and solder their pins to the copper pads.

Approach assembly logically. Start component installation with the lowest-profile devices first and work your way up to power transformer *T1*. That is, install first the jumper wires (indicated by the *J*s in Fig. 5) using bare solid hookup wire, except between pin 1 of *IC2* and the junction between *R16* and *R69* and between pin 14 of *IC3* and *C31*, both of which must be *insulated* solid hookup wire.

The Haas Effect

Haas, an early researcher into psychoacoustics, characterized how we perceive and localize sounds. He determined that, to avoid being confused by the echoes caused by reflections when trying to localize the direction from which a sound is coming, the brain ignores all but the first sound it "hears" for a small fraction of a second. All reflections arriving during this time period, called the Haas Fusion Region, are fused into the first sound, thus increasing its apparent loudness.

As a result of this "fusion," you perceive one louder sound coming from the direction of its first arrival. Reflections and echoes arriving after fusion, delayed by 20 to 30 milliseconds, are once agains perceived as separate sounds. Their density and rate of decay contain information that your brain uses to gauge the nature of the acoustic space you are occupying.

Next, install the resistors and diodes, followed by the IC sockets (if you have decided to use them) or the ICs themselves, trimmer potentiometer *R63*, and the low-profile capacitors.

Before installing *LED1*, trim its leads to 1½" long, taking care to remember which lead is which (it is best

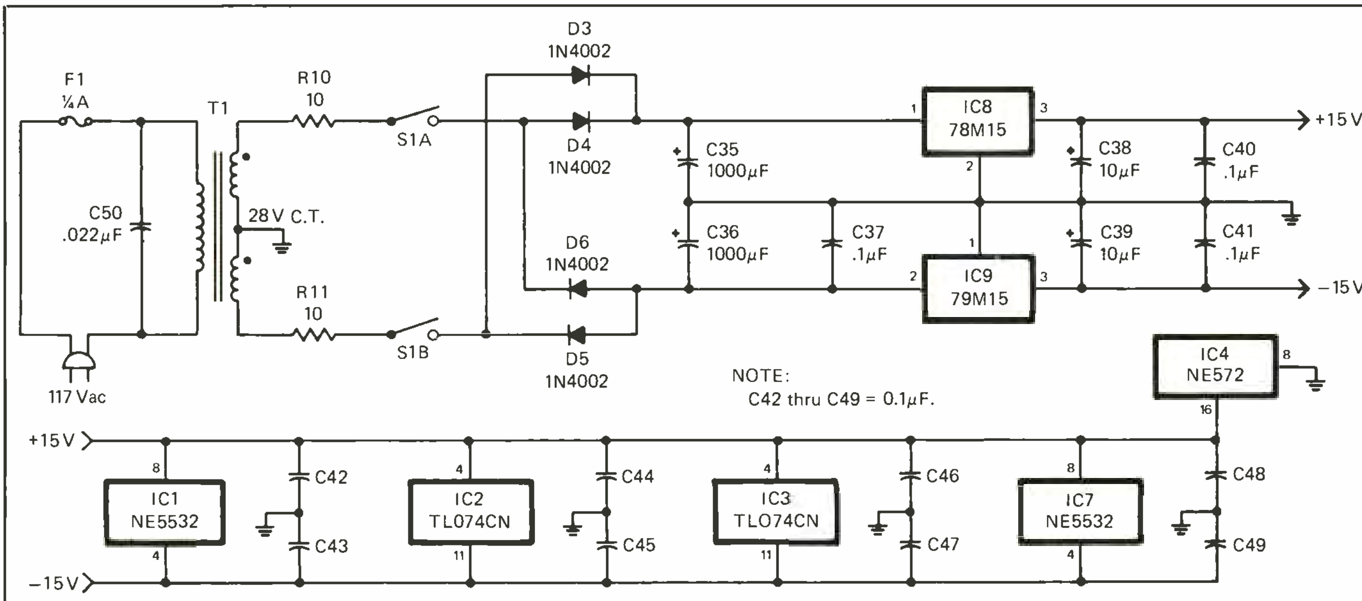


Fig. 3. This is the power supply, showing full regulation of the +15- and −15-volt lines. The lower portion of the diagram shows power connections to ICs.

Dolby Sound Movies

Movie sound tracks are usually recorded in discrete left, right, center (front), and surround (rear) channels. Special effects, like earthquake sounds, can bring the count up to six discrete channels. When these movies are mixed down to the ordinary two-channel stereo track format, the center channel is matrixed onto the left and right channels in-phase, while the surround channel is matrixed onto the left and right channels out-of-phase.

When this stereo mix-down is played back through a stereo system, the mostly dialog center channel is projected from the left and right speakers at equal volume and appears localized between the two. The surround signals also come out of the left and right speakers. But because they are out-of-phase, they appear to take on a diffuse quality with wider apparent separation. If this stereo mix-down is played back through a monophonic system, the surround signal cancels out.

Theater installations use reduced bandwidth, (−3 dB at 6 kHz), Dolby B noise reduction, and logic separation enhancement on the surround channel. This is done to accommodate the significant number of movie-goers who must sit directly beneath one of the surround speakers and would not otherwise receive an acceptable balance of front to surround signals.

to somehow mark the cathode lead for easy identification). After installing and soldering the lead to the copper pads on the board, bend the leads first back away from the lip of the board and then forward, about half way along their lengths. When you are finished, the LED should be about 3/8″ above and its body parallel with the board's surface. Be careful not to flex the LED's leads too much or they will break away from the device's body or the board.

The largest components should be mounted last on the board. These include electrolytic capacitors *C35* and *C36* and transformer *T1* in the power-supply section and controls *R1* (MASTER LEVEL), *R20* (FRONT LEVEL), *R52* (DELAY), and *R73* (REAR LEVEL). Temporarily set aside the pc board assembly.

As with the pc board, you can fabricate your own low-profile enclosure or purchase it ready-to-use, including all machining and labeling, with the complete kit of parts from the source given in the Parts List. If you decide to make your own enclosure, make sure you drill the holes for the controls, switches, *LED1*, input and output jacks, and the line cord in the proper locations. Use the circuit-board assembly to take all measurements for this operation.

Once machined, the enclosure should be spray painted and, when the paint completely dries, labeled. If you use a dry-transfer lettering kit, apply two or three *light* coats of clear spray lacquer to the front and rear panels to protect the lettering. Be careful not to make the lacquer too thick or runny or the lettering will lift off and dissolve.

When the enclosure is ready, install the input and output jacks on the rear panel and wire together all ground lugs with bare solid hookup wire. Then pass the free end of the line cord through its hole and secure it in place with a plastic strain relief. (If you prefer, you can line the hole with a rubber grommet, pass the free end of the line cord through, and knot it about 7″ from the free end.) In any event, leave 6″ to 7″ of loose wire with which to work.

Retrieve the pc-board assembly and install and solder into place seven separate lengths of hookup wire, or use a seven-conductor, preferably color-coded, ribbon cable to the appropriate points on the circuit board. Make the wires long enough to reach their respective jacks, plus some slack, when the board is in its mounting location inside the enclosure.

Twist together the fine wires in first one and then the other free end of the line cord. Make sure that all fine wires are twisted into the bundles. Then lightly tin each bundle with solder. Slip these wires into the holes provided for them in the pc board, solder them to the copper pads, and trim away any excess.

Carefully align the shafts of the controls and the buttons on the switches with their respective holes in the front panel and slide the board into place. Start hex nuts onto the control shafts but leave them quite loose. Tilt the board upward from the rear.

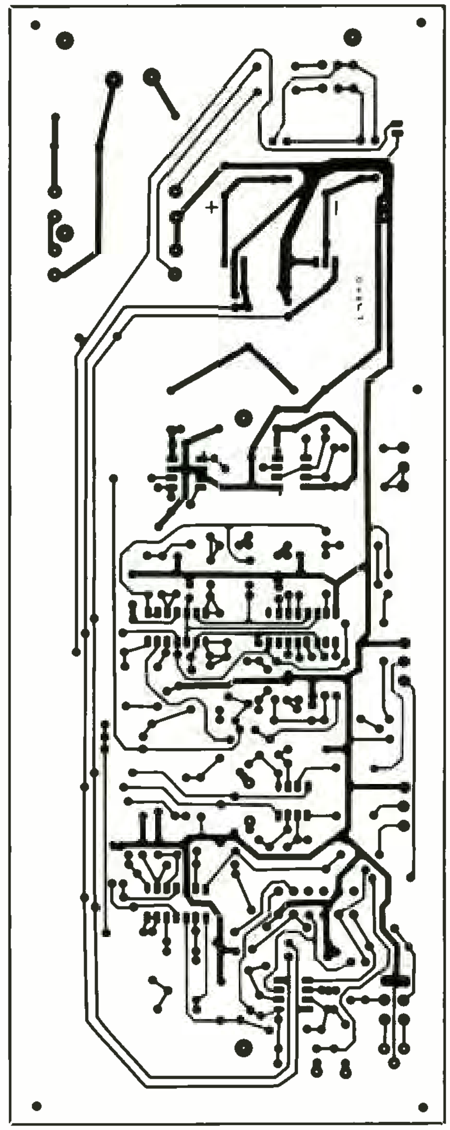


Fig. 4. This reduction of the actual-size etching-and-drilling guide for the pc board requires 2× blowup.

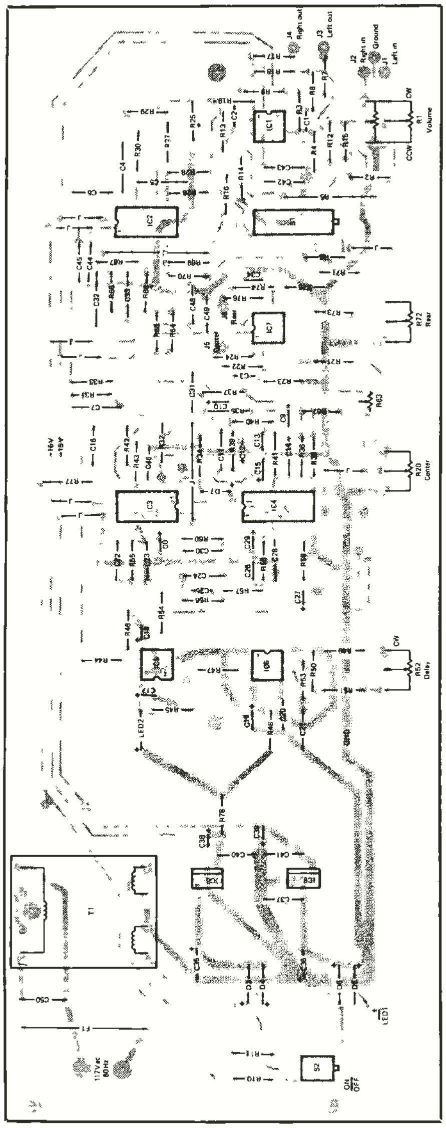
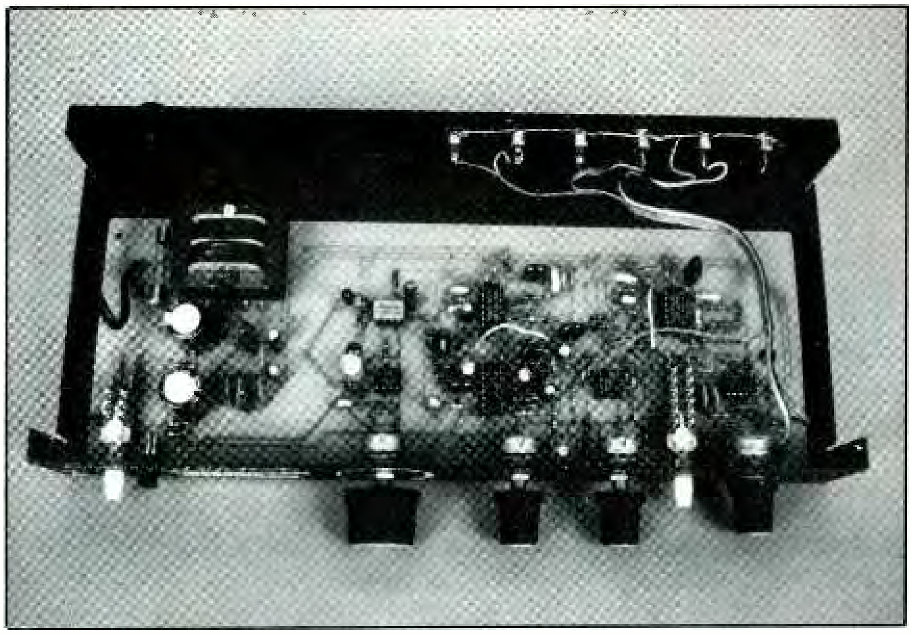


Fig. 5. When installing components on the pc board, be certain to orient them as shown.



All input and output jacks mount on the rear wall of the enclosure. The switches, controls and LED mount on the front panel. All other components mount on the pc board.

Using ¼" spacers and No. 4 machine hardware, mount the board to the floor of the enclosure. Before tightening any hardware, make certain that the buttons on *S1* and *S2* work without binding. This done, tighten the board mounting screws and the hex nuts on the controls. Then press a red panel lens into the remaining hole in the front panel and carefully push *LED1* into the lens. Install knobs on the front-panel control shafts.

Using Fig. 2 and Fig. 5 to guide you, connect the free ends of the wires coming from the printed-circuit board to the appropriate jacks on the rear panel.

Hookup and Use

The delay system is best connected between the outputs of a preamplifier and the inputs of a power amplifier. However, it can also be used effectively in a tape-monitor loop.

The rear speaker system and its driving amplifier need not be as powerful and wide ranging as the front speaker systems and amplifier. Typically, an amplifier for the rear source need not have more than 25% to 50% of the power of that used up front. The rear speaker need not be critical in performance, nor need it be matched to the speaker systems you use for the front, since the surround channel rolls off above 12 kHz and very low frequencies tend to be recorded in-phase and, thus, are suppressed in the surround channel.

To set delay time, begin with the DELAY control set to its midpoint position (straight up). This will be about the 30-ms position. You can optimize the delay time for your room by listening to a recording with impulse-type sounds, like record scratches (they are good for something, after all). When the delay time is set for too long a duration, you will hear discrete repeats or echoes. If it is set for too short a time, the image will shift to the rear speaker. When the setting is correct, the rear speaker will aurally disappear as an actual sound source.

In Closing

The project described above will extract ambience from any stereo program source and deliver spectacular "surround sound" effects from stereo-encoded movies. At least one of the new stereo TV programs, NBC's *Miami Vice*, uses surround sound; others are expected to follow. In the mean time, you can switch in the stereo synthesizer for the old-fashioned monophonic TV programs and still enjoy enhanced sound reproduction. ME