

CHRONICS

# HOBBY PROJECTS



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# RADIO-ELECTRONICS HOBBY PROJECTS



**TAB BOOKS**

Blue Ridge Summit, Pa. 17214

## Preface

The projects collected in this book encompass five broad categories. There are projects for the hi-fi stereo buff, the musician or musically inclined, the auto enthusiast, gadgets designed for use around the home, and eight test and measuring devices.

These projects, selected from Radio-Electronics magazine, were chosen to appeal to a wide range of interests and for varying levels of technical ability and construction dexterity. Some are simple enough to put together in an evening or two; others are more complex and infinitely more challenging.

In each case, there is a complete parts list and, where appropriate, detailed construction procedures, augmented by schematics and detailed drawings. Where special components are used, a source of supply is given. Should you encounter difficulty in getting any project to operate properly, be sure to re-read the text and double-check each step to make sure "gremlins" have not crept into your work!

The Editors



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## PART 1

# Audio, Stereo & Hi-Fi

## IC MULTIPLEX DETECTOR

by KEN BUEGEL

INTEGRATED CIRCUIT DEVELOPMENTS often seem to be aimed at exotic applications which are of little interest to the general electronic public. But Motorola's MC1304 and MC1305 are different: on one chip is a complete, advanced-design multiplex decoder. The MC1305, although identical in cost to the MC1304, provides for a separation adjustment and was used in this project.

The internal circuitry is formidable: 10 diodes, 31 transistors and 29 resistors. Even the block diagram for the 1305 (Fig. 1) reveals a level of complexity not often used with discrete-component multiplex adapters.

The IC uses the proven balanced time-switching technique with its inherent SCA rejection without filtering, and also provides a driver for a stereo indicator. Two separate additional inputs provide for stereo-mono switching and audio muting. And as if this weren't enough, a series of diodes with emitter followers serve as temperature-compensated voltage regulators for the circuits.

Performance of the chip is phenomenal in comparison to the very conservative Motorola specifications. A separation of 45 dB at 1 kHz is typical; three units constructed revealed a separation of 55 to 57 dB at 1 kHz, 44dB at 100 Hz., and 37 to 49 dB at 10 kHz. These figures are on a par with the very best multiplex adapters of conventional design. SCA rejection exceeds 55 dB and no inter-

fering "birdies" have ever been audible with the device.

Channel balance is within 0.5 dB and total harmonic distortion is less than 0.5% at the recommended composite-signal input level.

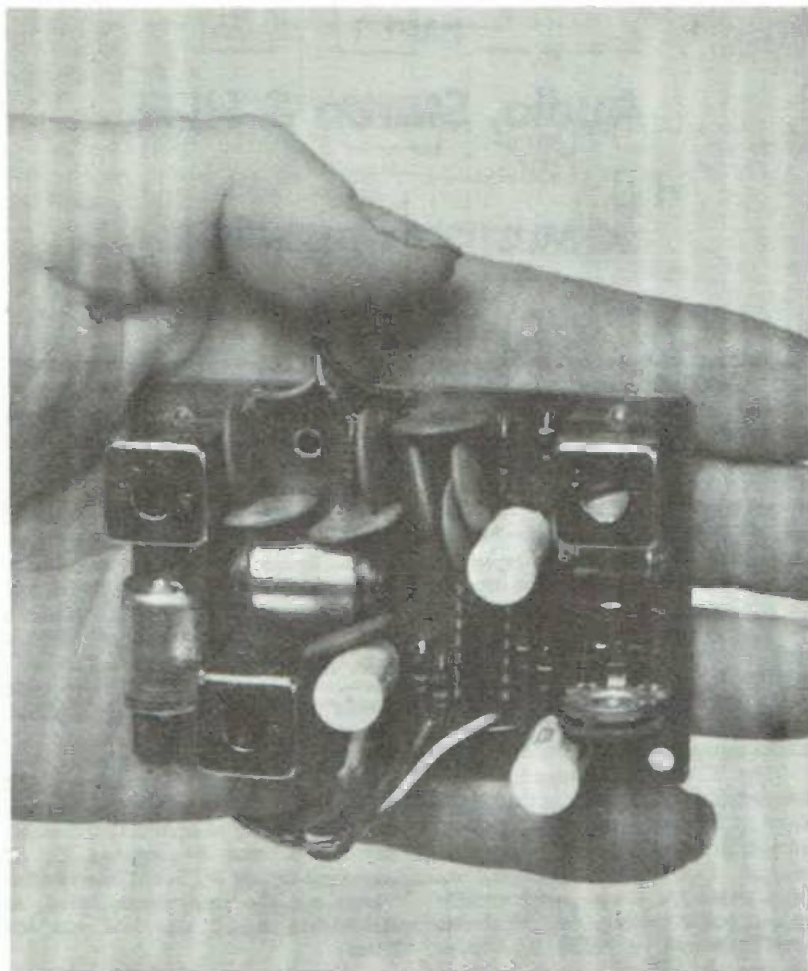
The only drawback found in the design is the 19-kHz and 38-kHz rejection figures. Although a typical rejection is 20 dB at 38 kHz, this is not enough to prevent possible "birdies" when beating against a tape recorder bias oscillator. Therefore a 'Twin-T' filter has been added (Fig. 2) to each output to increase rejection another 20 dB.

### Construction & alignment

You can build this project for under \$21. The actual-size PC pattern for the decoder is included for do-it-yourselfers, and a complete kit of parts and drilled PC board is available. Please use the components listed, as the circuit board was laid out for them.

Construction of the project is simple, as is the alignment procedure. A 2.2 x 3.2-inch PC board provides enough component space without crowding. Recommended coils for this circuit are available only in a PC mounting style.

The best technique is to insert the IC and then add other parts outwards from the IC. *Don't* bend the leads on R4, IC1, or the transformers. After all parts are properly soldered, add the required external wires. If you do not plan to use audio muting or mono-



stereo switching, wires are not connected to pins 4 and 5 of IC1.

Although a multiplex generator is the easiest alignment method, this unit can be aligned with a broadcast signal. Input level should be about 0.75 volt p-p to achieve maximum channel separation.

Each output should be connected to a 22K load to provide the proper terminating impedance to the Twin-T filters.

Connect an oscilloscope or ac vtvm to the junction of pin 1 and C4. Peak L1 and L2 for maximum 19 kHz as seen on the scope. This waveform should be about 1.6 volts p-p with a +15-volt supply. Move the scope probe to the junction of C5 and pin 4 of L3 and peak L3 for a maximum 38-kHz trace. This should be about 22 volts p-p. Caution: if you do not use a low-capacitance probe, the circuit may be slightly detuned when the

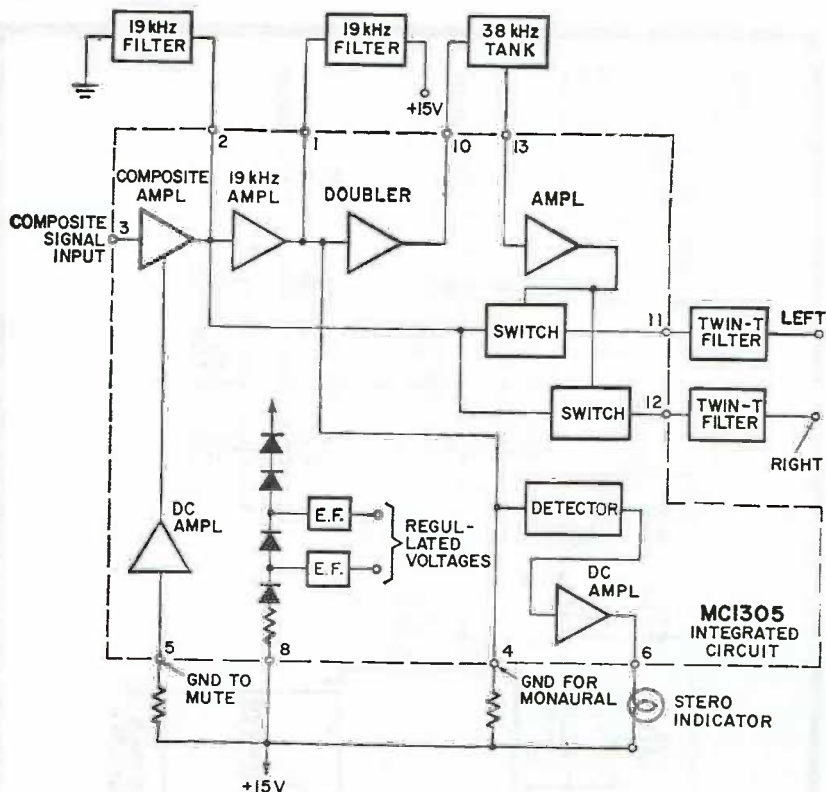


Fig. 1—Complex functions performed by 31-transistor MC-1305 (inside dotted line). Connections to pins 4 and 5 are optional.

probe is removed. This detuning will be slight and is corrected in the next step.

Connect the scope or vtvm to the right output and set up the generator for a left-only output. Set the wiper of R4 to midposition. Carefully peak L1, L2, and L3 for a minimum output on the right channel. Then set R4 for a minimum output.

Now set the generator for a right-only output and read the output level on the right channel. Then set the generator to a left-only output. The difference in readings is the channel separation. It will not be as high as the figures given earlier since the residual reading also includes 19-kHz

and 38-kHz components. An elaborate filter can remove these components for true separation readings, but the separation will not be increased.

### FM station alignment

If you do not have access to a multiplex generator, connect the input to your tuner output. First peak L1, L2 and L3 for maximum output waveforms as described earlier. Connect the scope vertical input to the left output and the horizontal input to the right output. Tune to a monaural station and set the scope gains until the trace is a straight line at a 45° angle (Fig. 3-a).

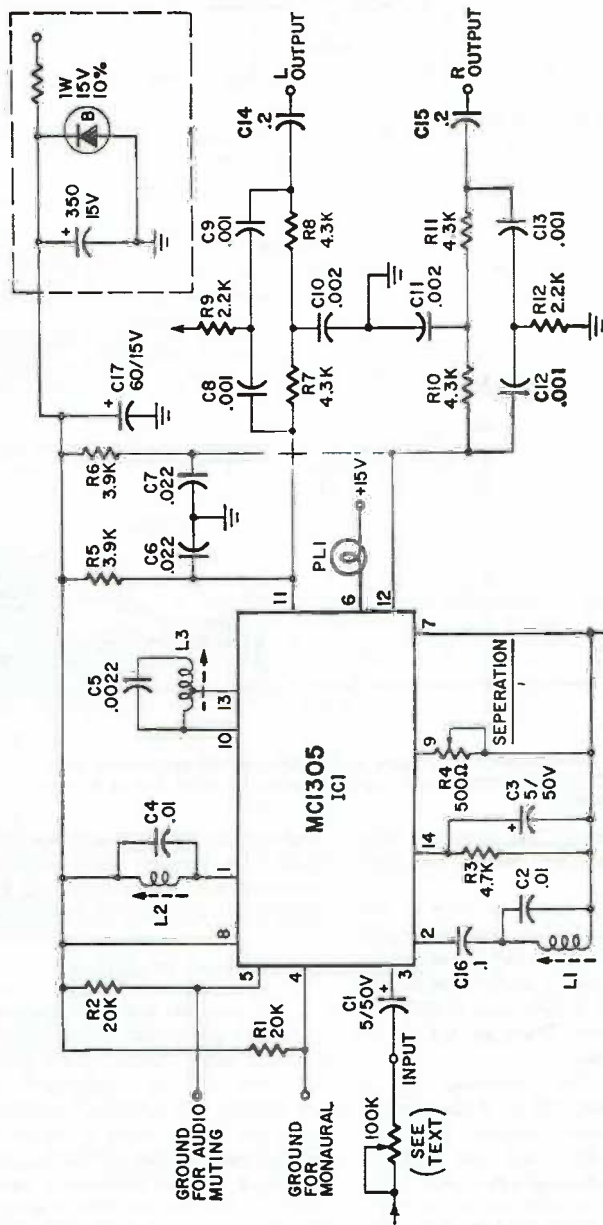


Fig. 2—Schematic showing components mounted externally to the 1305. Use a 12ES (12V, 40 mA) lamp for the lamp PL1.



## PARTS LIST

All resistors  $\frac{1}{4}W$ , 5%

R1, R2—20,000 ohms

R3—4700 ohms

R4—500-ohm,  $\frac{1}{4}W$  trimmer (CTS type X201R50113)

R5, R6—3900 ohms

R7, R8, R10, R11—4300 ohms

R9, R12—2200 ohms

### Capacitors

C1, C3—5- $\mu F$ , 50V electrolytic (Mallory MTV 5CB50)

C2, C4—.01- $\mu F$  polystyrene (Mallory SX110)

C5—.0022- $\mu F$  polystyrene (Mallory SX222)

C6, C7—.022- $\mu F$ , 100V Mylar (Mallory PVC1122)

C8, C9, C12, C13—.001- $\mu F$  ceramic (Centralab CE102)

C10, C11—.002- $\mu F$  ceramic (Centralab CF202)

C14, C15—.02- $\mu F$ , 10V ceramic (Centralab UK-10-204)

C16—.01- $\mu F$ , 10V ceramic (Centralab UK-10-104)

C17—60- $\mu F$ , 15V electrolytic (Mallory MTV 60CB15)

### Other parts

L1, L2—J. W. Miller type 1361

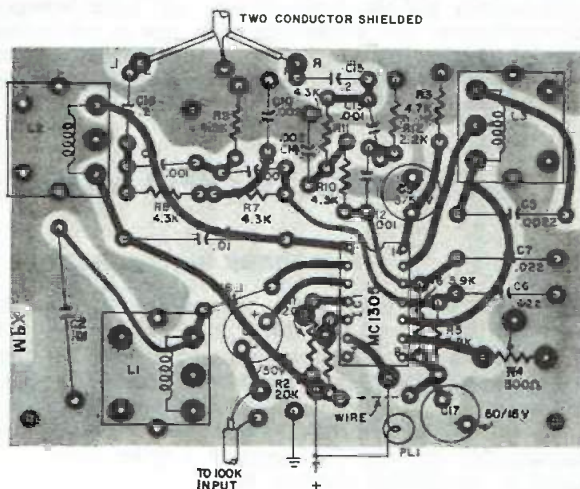
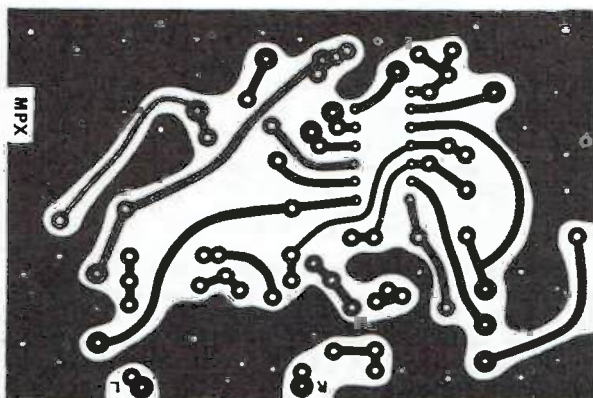
L3—J. W. Miller type 1362

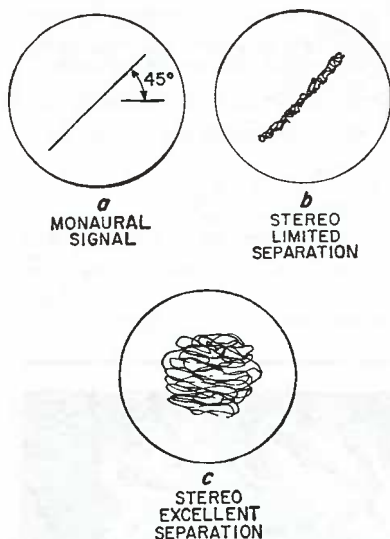
IC1—Motorola MC1305P

The following parts may be ordered from Transitek Co., P.O. Box 98205, Des Moines, Wash. 98016. All prices include postage. IC1, \$7.20. L1, L2, L3 (set of three), \$5.40. PC board MPX, \$2.95. Complete kit of all listed parts and drilled PC board, \$21.00.

Use the same-size printed circuit pattern above to make your decoder. Components specified will fit on the board.

Component-side drawing on the right shows where to mount the parts. Some of the wiring to the unit is optional.





**Figs. 3-a-c** show scope patterns with vertical scope input to left output of the decoder and horizontal input connected to the decoder right output.

Tune to a stereo broadcast and you will probably see something like Fig. 3-b. This indicates limited separation. Now, while watching the scope face, slowly tune L1, L2, L3, and R4 until you get a trace most like Fig. 3-c. If you can connect your amplifier to the adapter and also hear the output—preferably in headphones—so much the better.

Some stations have stereo programs which feature highly directional

microphone pickup, this type of program material is the easiest to use for alignment. When the output looks like Fig. 3-c, you must identify the channels. It is possible to tune the unit so that the output labeled L is actually the right channel. Careful tests do not show any difference in separation or other specifications, however, so if you wind up with the channels interchanged, simply reverse them when you plug them into your preamp.

Most tube tuners will have more than 0.75 volt p-p output. This adapter will have decreased separation and increased distortion at higher input levels. The input impedance is around 20K so a 100K pot inserted in series with the input may be adjusted until the input is correct.

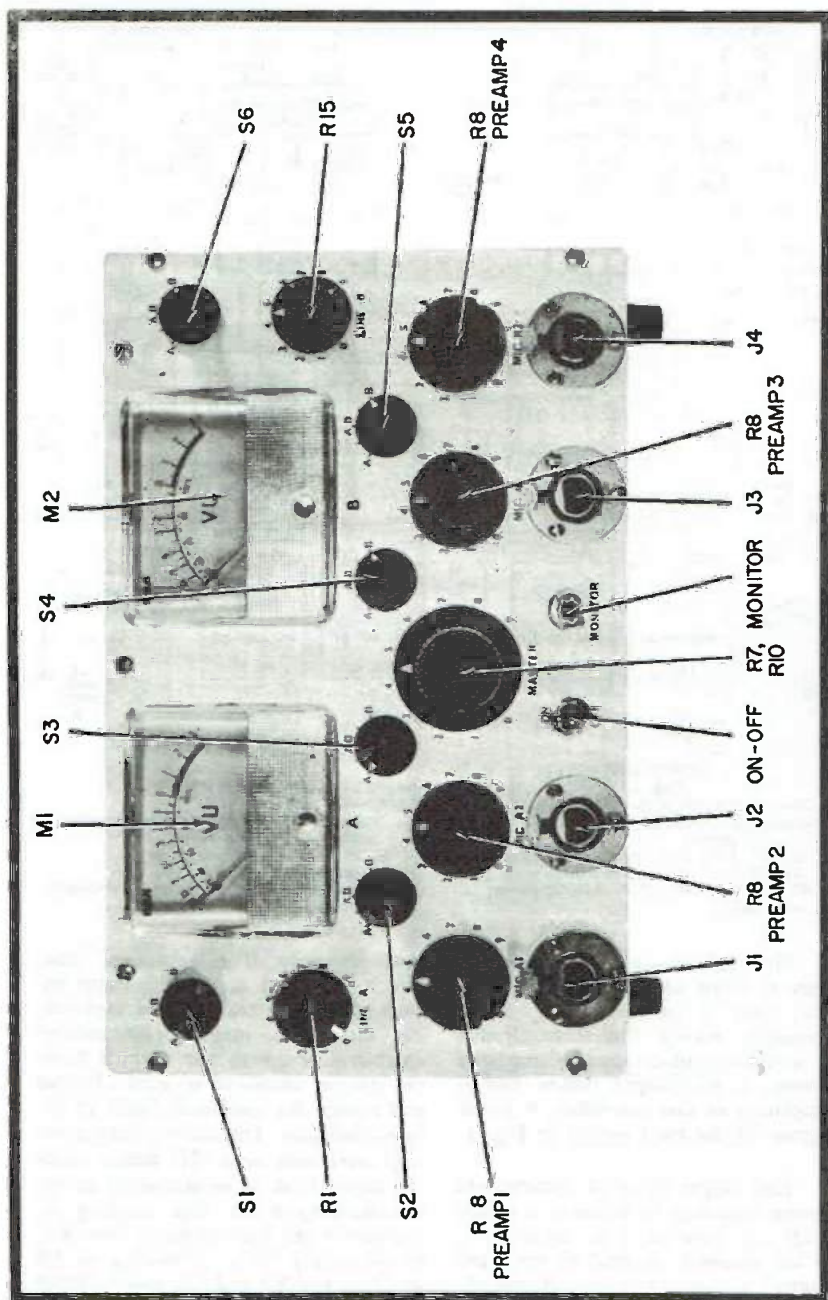
An interesting feature of this IC is its 8–22-volt supply specification. If the adapter is aligned at 15 volts and the supply voltage decreased, separation stays almost unchanged. This is not true if the adapter is aligned at a lower voltage which is then increased. In no event should the supply exceed +22 volts. Operation at +15 volts is highly recommended, since no performance characteristic was improved at higher voltages. As the supply voltage is not critical, a relatively inexpensive Zener diode with capacitor filtering will provide very stable operation.

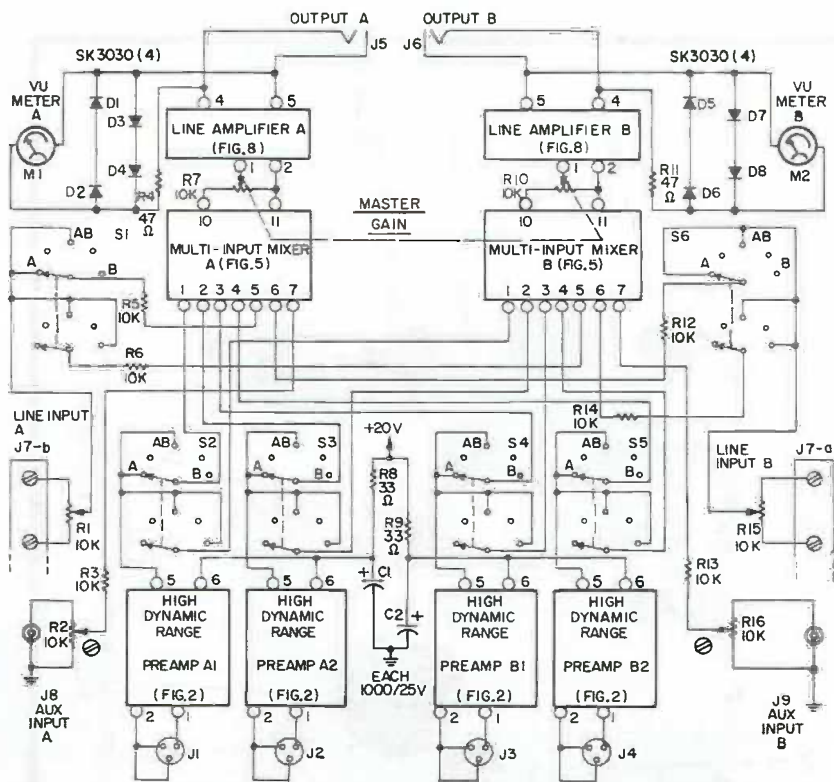
## 6-CHANNEL STEREO MIXER PREAMP

by **GEORGE D. HANCHETT**

THIS STEREO PREAMPLIFIER AND mixer is particularly interesting to those who want to make high-quality tape recordings. The preamp has four microphone and two line inputs that can be switched to left, right, or both channels. In addition, two auxiliary inputs are provided, one for each

channel. The auxiliary inputs cannot be switched. All inputs that can be switched are controlled from the front panel. The two auxiliary inputs are controlled from the rear of the unit. Each output channel has its own VU meter.





#### PARTS LIST

(Fig. 1)

C1, C2—1000  $\mu$ F, 25 volts, electrolytic  
D1 through D8—SK3030 (RCA)

M1, M2—VU meter

R1, R2, R7, R10, R15, R16—potentiometer,  
10,000 ohms (R7, R10 is a dual linear pot)

R3, R5, R6, R12, R13, R14—10,000 ohms,  $\frac{1}{2}$   
watt, 10%

R4, R11—47 ohms,  $\frac{1}{2}$  watt, 10%

R8, R9—33 ohms,  $\frac{1}{2}$  watt, 10%  
S1 through S6—rotary switch, 2 poles, 5  
positions, shorting type

J1, J2, J3, J4—microphone jacks

J5, J6—phone jack

J7—4-lug terminal strip (screw terminals)

J8, J9—phono jack

The stereo preamplifier and mixer is made up of three basic circuits and a minimum of interconnecting wiring. The three circuits are a high-dynamic-range microphone preamp, a multi-input mixer, and a headphone or line amplifier. A block diagram of the total unit is in Fig. 1.

The output of each microphone preamp (see Fig. 1) is fed to a switch which can connect it to channel A, the left channel, channel B, the right channel, or both channels (A and B)

simultaneously. The output of these switches as well as the line input for each channel is fed into the multi-input mixers. A master gain control combines or gangs the outputs from the mixers installed in each channel and passes the combined signal to the line amplifiers. The diode limiting circuit used with each VU meter keeps the meter from being damaged during the charging of the large coupling capacitors in the line-amplifier. Two R-C power-supply filters consisting of R8 and C1, and R9 and C2 assure circuit

stability. Each filter services two microphone preamplifiers.

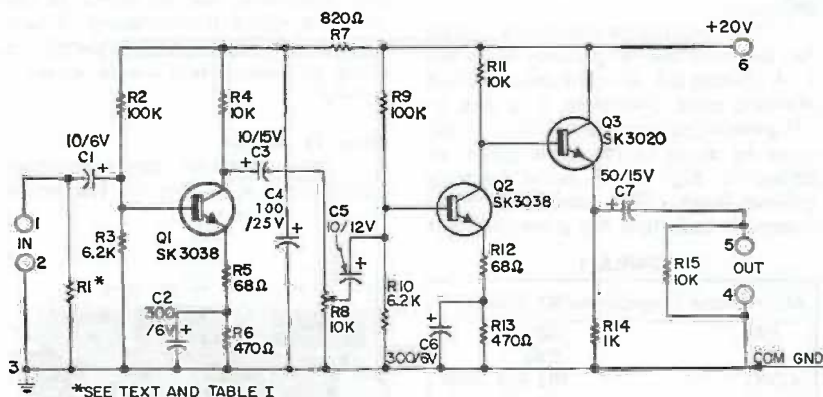
The stereo preamplifier and mixer is made up of a number of circuits as described above. The description of each of the three circuits includes circuit boards and component placement diagrams. The individual circuit boards and the interconnecting wiring required for the stereo preamp and mixer may be assembled as desired by the builder to form the kind of custom unit he needs.

### CIRCUIT BOARDS

A complete set of 8 circuit boards needed to build this unit are available for \$10. Order 1070B-1. A set of 4 boards for building a mono version are \$6. Order 1070B-2. Boards are G-10 glass-epoxy, undrilled. Photo negative for making your own boards containing all board patterns is \$1.50. Order from Photolum Corp., 118 E. 28 St., New York, N. Y.

The high-dynamic-range microphone preamplifier, intended to be used with low-impedance dynamic microphones, will handle loud passages of music and close talking without adverse effect on the output. The amplifier has a gain of 1500 to 2000 and can provide a maximum undistorted output voltage of 2 volts rms to a load impedance of 500 ohms or more. The maximum undistorted input is 400 mV rms. The frequency response is flat from 20 Hz to 30 kHz.

The circuit for the high-dynamic-range microphone preamp are in Fig. 2. The preamplifier consists of two stages of current-stabilized amplifiers separated by a gain control and an R-C filter, consisting of R7 and C5, that prevents motorboating. Resistors R5 and R12, placed in the emitter circuits of transistors Q1 and Q2, improve the frequency response of the preamplifier by providing some



### PARTS LIST

(Fig. 2)

C1, C5—10  $\mu$ F, 6 volts, electrolytic  
C2, C6—300  $\mu$ F, 6 volts, electrolytic  
C3—10  $\mu$ F, 15 volts, electrolytic  
C4—100  $\mu$ F, 25 volts, electrolytic  
C7—50  $\mu$ F, 15 volts, electrolytic  
Q1, Q2—SK3038 (RCA)  
Q3—SK3020 (RCA)

All Resistors  $\frac{1}{2}$ -watt 10%

R1—see table I and text  
R2, R9—100,000 ohms  
R3, R10—6200 ohms  
R4, R11, R15—10,000 ohms  
R5, R12—68 ohms  
R6, R13—470 ohms  
R7—820 ohms  
R8—potentiometer, 10,000 ohms, audio taper  
R14—1000 ohms



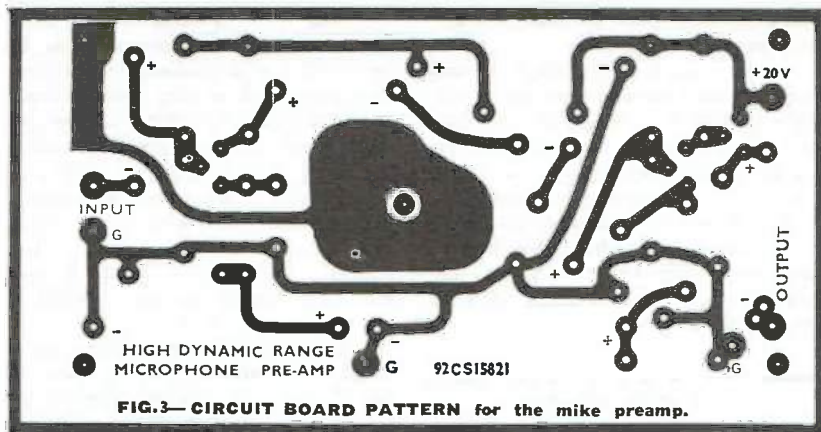


FIG. 3—CIRCUIT BOARD PATTERN for the mike preamp.

emitter degeneration. The output of the preamplifier is shunted with resistor, R15, to make the circuit compatible with the zero-point switching capability used in the master preamp. The output impedance of the preamp is low. The table shows the value of R1 to use with microphones of various impedances.

The printed-circuit-board layout for the microphone preamp is in Fig. 3. A photograph of a completed board showing parts placement is in Fig. 4. All ground connections in this circuit must be made to the same point, as shown in Fig. 3, to avoid forming ground loops. This common-ground feature is built into the printed-circuit

TABLE I

Microphone Impedance	R1 (ohms)
200	220
500	560
4,000	R1 not used

R1 connected across microphone input jack.

board and must be followed if some method of circuit construction other than the printed board is used. In addition, all preamplifier connections to external circuits should be made to the same ground point.

### Multi-input mixer

The multi-input mixer is designed to mix the inputs from up to seven sources, usually microphones, for input to an amplifier, recorder, or other piece of audio equipment. The mixer has a gain of approximately unity and, therefore, has no effect on the system in which it is installed. If more than seven inputs are required, as many as three mixers can be wired in parallel.

### How it works

The circuit for the multi-input audio mixer is in Fig. 5. The resist-

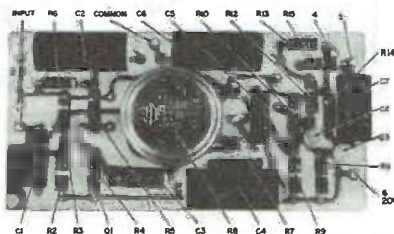
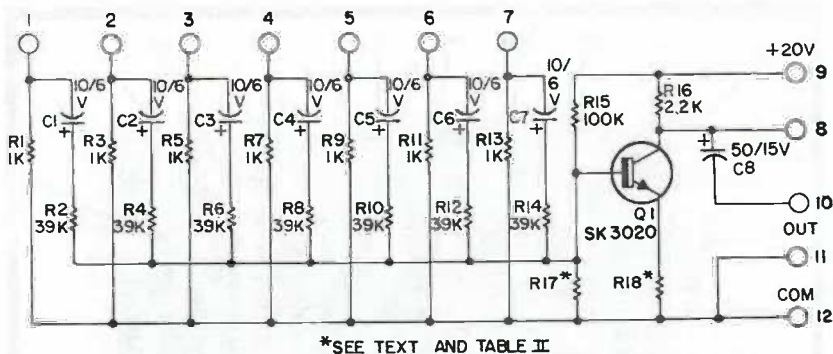


FIG. 4—PHOTO OF ASSEMBLED BOARD showing parts placement details.





#### PARTS LIST (Fig. 5)

C1 through C7—10  $\mu$ F, 6 volts, electrolytic  
 C8—50  $\mu$ F, 15 volts, electrolytic  
 Q1—SK3020 (RCA)  
 R1, R3, R5, R7, R9, R11, R13—1000 ohms,  $\frac{1}{2}$  watt, 10%

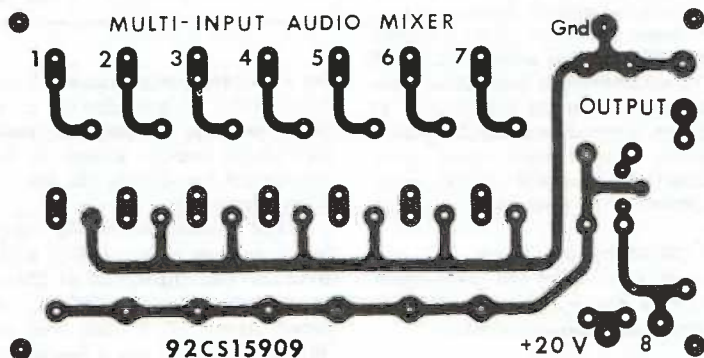
R2, R4, R6, R8, R10, R12, R14—39,000 ohms,  $\frac{1}{2}$  watt, 10%  
 R15—100,000 ohms,  $\frac{1}{2}$  watt, 10%  
 R16—2200 ohms,  $\frac{1}{2}$  watt, 10%  
 R17, R18—see table II and text

**FIG. 5—CIRCUIT** of multi-input audio mixer. Two are needed to build the unit described.

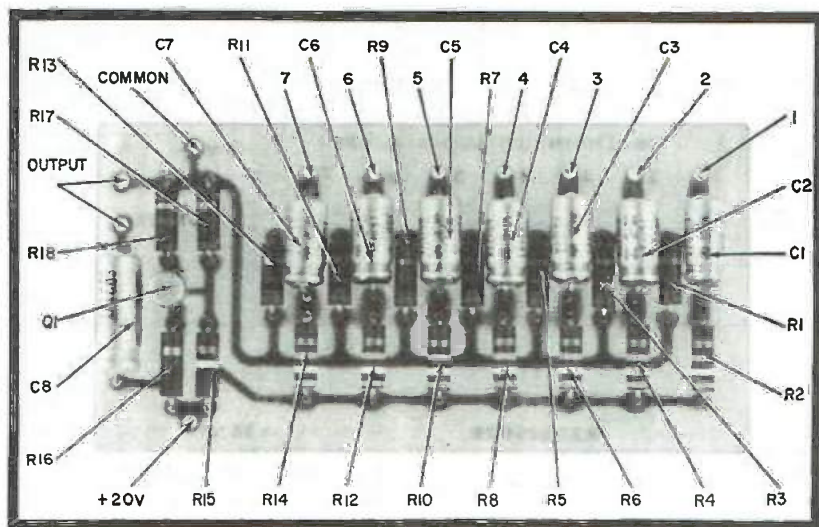
ance network shown at the left not only provides the mixing function but also to make possible zero-point switching of the inputs.

In the zero-point switching, as used in this unit, the capacitors at the

output of the microphone preamps as well as the input capacitor of the mixer are kept charged. This is done by connecting a resistor across the output and input. Thus there is no disturbance, no cracks or pops, when



**FIG. 6—CIRCUIT BOARD PATTERN** for the audio mixer.



**FIG. 7—PHOTO OF MIXER** shows parts placement on the circuit board. Refer to Fig. 1 for interconnection details of the complete preamp.

inputs are switched in or out. The amplifier portion of the circuit, shown at the right in the schematic, is current stabilized by the emitter resistor. This resistor is not bypassed, and provides a greater degree of degeneration and reducing the overall gain of the mixer to unity.

Some adjustment of resistor values is required if less than seven inputs are used. Table II shows these resistor values for from 2 to 7 inputs. When three or more mixers are paralleled to accommodate more than seven inputs, not only must the outputs of the mixers be paralleled but the ground points on each circuit board must be connected. The gain of the mixer thus connected is somewhat less than unity.

Component placement and circuit board pattern for the multi-input mixer are in Figs 6 and 7 along with a photograph of a completed board.

#### Headphone or line amplifier

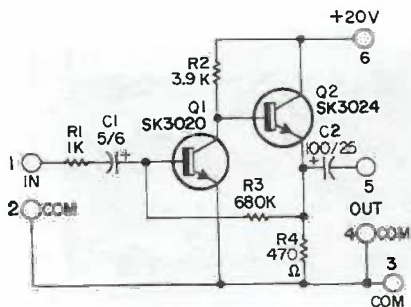
The headphone or line amplifier is very useful when the power ampli-

**TABLE II**  
**RESISTANCE VALUES**

No. of inputs	R17	R18
2	8.2k	120 ohms
3	7.5k	110 ohms
4	6.8k	91 ohms
5	6.8k	82 ohms
6	6.2k	75 ohms
7	6.2k	68 ohms

fier is located some distance from the microphone. If preceded by a microphone preamp, the amplifier makes a very useful remote pickup. It is also very useful for driving the line inputs of tape recorders.

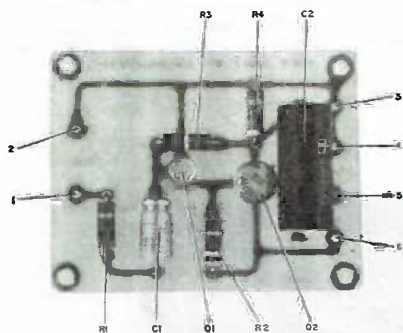
The headphone or line amplifier has a voltage gain of 100 and can drive any line impedance of 250 ohms or more. It has a maximum undistorted output of 3 volts rms into a 500-ohm line and has a frequency response flat from 20 Hz to more than 25,000 Hz. The input impedance is 1,800 ohms.



**PARTS LIST**  
(Fig. 8)

- C1—5  $\mu$ F, 6 volts, electrolytic
- C2—100  $\mu$ F, 25 volts, electrolytic
- Q1—SK3020 (RCA)
- Q2—SK3024 (RCA)
- R1—1000 ohms,  $\frac{1}{2}$  watt, 10%
- R2—3900 ohms,  $\frac{1}{2}$  watt, 10%
- R3—680,000 ohms,  $\frac{1}{2}$  watt, 10%
- R4—470 ohms  $\frac{1}{2}$  watt, 10%

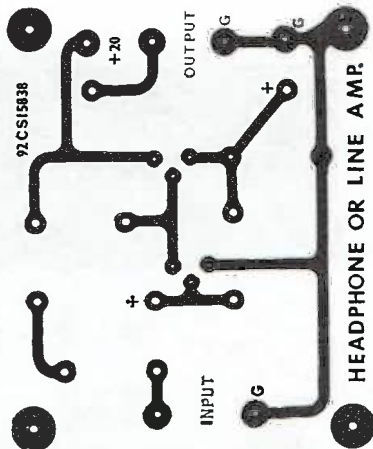
**FIG. 8—LINE AMPLIFIER** circuit uses two transistors.



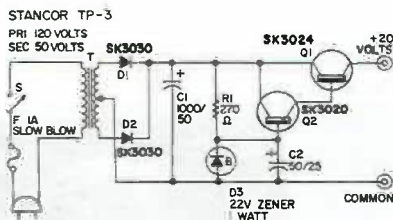
**FIG. 9—CIRCUIT BOARD** pattern for the line amplifier is shown actual size.

### Amplifier operation

The circuit for the headphone or line amplifier is in Fig. 8. The interconnection of the transistors in the amplifier makes the operating condition of the amplifier self-adjusting, i.e.; the amplifier can maintain itself in a stable operating state in spite of variations in power-supply voltage and ambient temperature. Stability is insured by feedback through R3. If Q1's

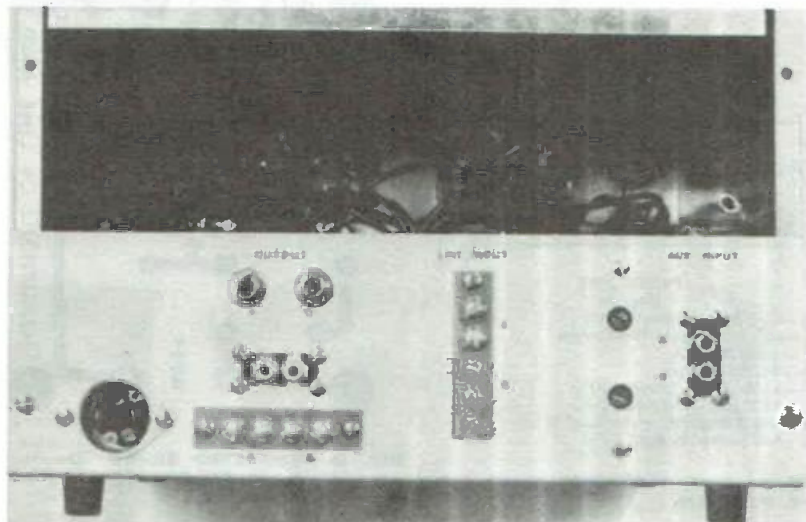


**FIG. 10 COMPLETED BOARD** showing parts placement. See Fig. 1 for circuit board interconnections.

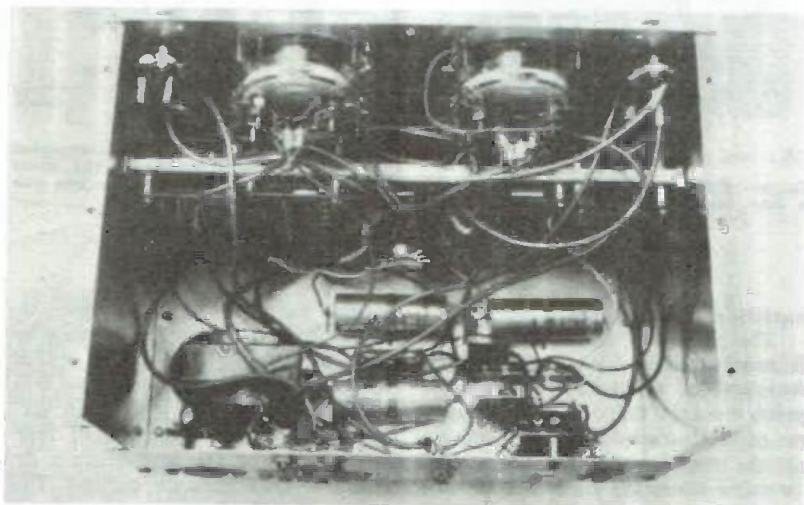


**FIG. 11—POWER SUPPLY** is regulated circuit. It is built right on the main chassis of the mixer preamp as you can see in the photo

emitter current should increase, the base voltage of Q2 would decrease because of the additional voltage drop in R3. However, the decreased base voltage of Q2 results in a drop in Q2's emitter current, a reduction of feedback voltage to Q1, and hence a decrease in Q1's collector current. This decreased collector current causes an increase in Q2's base voltage that compensates for the original decrease



**FROM THE REAR OF THE MIXER-PREAMP** you can see the auxiliary input and line input jacks. Three sets of outputs were connected in parallel to provide connectors for all likely applications.



**LOOKING IN FROM THE TOP** you can see the power-supply components in the lower right of the photo. The two VU meters are at the top. Most of the jumble of wiring consists of leads connecting outputs to the circuit boards.

and the amplifier is stabilized. The interconnection of transistors just described also makes possible the low output impedance of the amplifier.

The printed circuit board pattern for this circuit, and a photo of a completed board, are in Figs. 9 and 10.

### **Power supply is last**

A simple power supply completes

the unit. This supply is shown in Fig. 11. It is assembled directly on the chassis of the unit and is not built onto a circuit board. Once you have completed the power supply, selected the desired number of input, mixer and output boards you can proceed to assemble your custom mixer-preamp. We are sure you will enjoy it.

## **STEREO EXPANDER-COMPRESSOR**

by **W. E. McCORMICK**

IF YOU PREFER THE FULL, DYNAMIC music range of original performances to the electronically compressed material from tuner, tape or disc, the stereo volume expander-compressor described here will enable your hi-fi system to deliver it.

Or, if you want a limited, preset level of music or TV audio, the unit will provide that too. In addition, the volume expander-compressor will act as a limiter with other reproducing equipment.

It operates with any hi-fi system, and can be driven by almost any signal source, including carbon microphones. It has two operating modes: expansion and compression. Specifications meet those of commercial units (see chart), and the compressor-expander can be constructed for about \$20.00.

Amplitude swings produced during live musical sessions are often too great to be broadcast or recorded, and must be reduced in level for the media used to convey them. When these dynamics are linearly restored, realism is greatly enhanced.

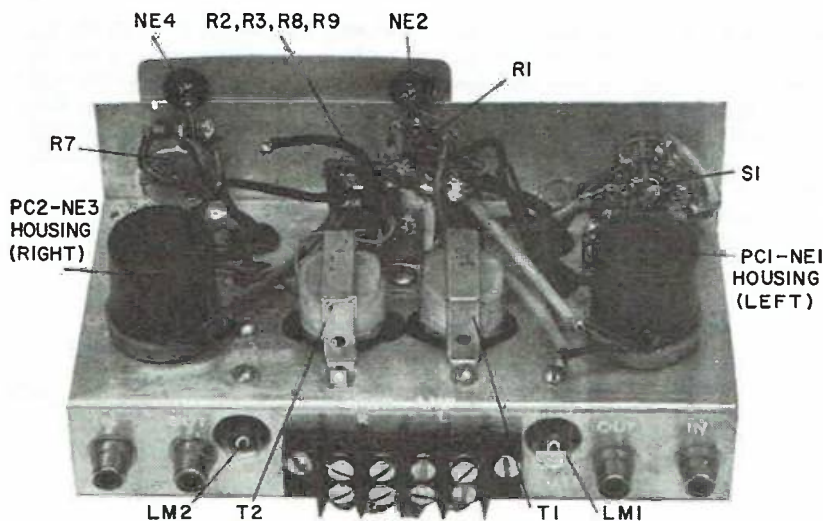
Various devices for replacing this dynamic dimension have been marketed.

### **SPECIFICATIONS**

**Expansion.** To 8dB per channel  
**Compression.** To 20 dB per channel.  
**Distortion.** None  
**Input impedance.** 1000-100,000 ohms with RCA photo-cell. About 250-100,000 ohms with Clairex photo-cells.  
**Output impedance.** 47,000-470,000 ohms average.  
**Use with amplifier.** 4-8 or 16-ohm output impedance.  
**Drive voltage required**—0.7 volt to initiate dynamic action  
**Frequency response.**  $\pm 1$  dB throughout audio range.  
**Inputs** . . . 2 Stereo version  
**Outputs** . . . 2  
**Rise time\*** . . . 10 msec. with RCA cells; 12-15 msec. with Clairex cells.  
**Insertion loss** . . . 2 dB on compression. 6 dB average on expansion.  
\*Measured from instant of applied illumination until cell current reaches 63% of its total value. This is a function of distance between lamp and cell and illumination intensity. Since the maintaining neon lamp voltage is less than the firing voltage, a sufficiently long decay time is automatically produced. This also tends to produce a light bias on the cells, which keeps them ready for firing.

After deciding to add this feature to my hi-fi system, I considered several methods. Some were merely gimmicks: resistors, varistors, even incandescent lamps were placed across the amplifier output, where their load-





*Compander parts arrangement. Note left-right symmetry. Identical parts arrangement was used for each stereo channel. It makes assembly considerably easier and neater.*

ing effect varied inversely, but seldom linearly, with the current through them.

Other units with built-in "take charge" circuits use the variable conduction of back-biased diodes to make the output of a cathode follower increase faster than its input. These operate between fixed points over which there is no convenient control. They also require alterations in critical signal circuits, which many audio-philists do not like.

The principle employed in some commercial expanders had an appealing simplicity. Signal-driven, they use neon lamps to activate photoconductive cells and require no other power supply. Their major component, however, had been especially designed for a specific product.

### **Special components needed**

With a circuit configuration in mind, however, what was demanded of the components could be assessed. Some of the qualities needed were out

of the ordinary, but hardly unique. Needed were:

- A transformer with practically flat frequency response over the audio range.
- A high-impedance input winding to permit connecting it across an output transformer's secondary without appreciable loading.
- A turns ratio high enough to drive a neon lamp from an audio signal voltage of 1 volt or so.
- A neon lamp with a firing voltage and ionization time the same in total darkness as in ambient light.
- A light output that would remain proportional to the voltage applied and not behave erratically after a few overloads.
- A photoconductive cell with proper spectral response. Great sensitivity. Fast rise time. Proper light-to-dark conductance ratio. Appropriate resistance extremes and extreme linearity at very low light levels.

Such parts were rounded up, and Fig. 1 shows a stereo version of the



# Parts List

C1, C2, C3, C4—0.47  $\mu$ F, 200V  
 R1, R7—500-ohm, 5-watt, wire-wound potentiometer (Mallory VW500 or equal)  
 R2, R3, R8, R9—330,000 ohm, 1-watt, 10% resistor  
 R4, R5, R10, R11—47,000-ohm, 1/2-watt, 10%  
 R6, R12—100,000 ohm, 1/2-watt, 10% resistor  
 NE1, NE2, NE3, NE4—Neon lamp, Signalite NE2V or ASA No. A2B. Allied Radio part #K002 122.  
 PC1, PC2—Photo cells, Clairex CL504L or RCA 4425  
 T1, T2—Transformer, primary impedance 500,000 ohms, secondary impedance 50 ohms (Argonne AR-142, Lafayette Radio see text)

LM1, LM2—fuse, No. 44 lamps, 6.8V, 0.25 amp  
 S1—4 pole, 3-position switch (or Lafayette part No. 99H6170, 9-pole, 3-position)  
 Misc—Terminal strip (Jones 4-screw, barrier type or equal), resistor board (for 4, 1-watt resistors), solder lug terminal strips (two with 2 insulated and 1 gnd. lug, one with 4 insulated and 1 gnd. lug), phono jacks (4 RCA audio type and plugs as needed), chassis (Bud CB-1628, 1 1/4" h x 6 1/8" w x 3" d), cabinet (Spectrum Products, No. 488, Bud SC-2130 or CU-465), metal panel (approx. 3" x 6"), plastic or metal light housings, panel jewels and threaded bushing set, lamp-cell housings, knobs, grommets, wire, etc.

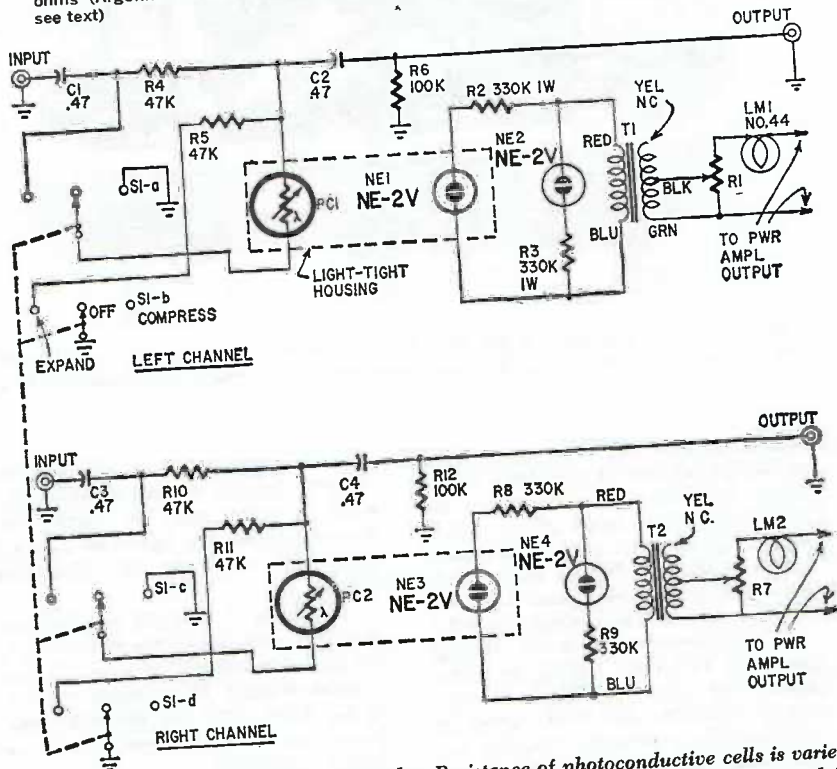


Fig. 1—Stereo version of the compandor. Resistance of photoconductive cells is varied by stepped up signal from amplifier that fires neon lamps. Cell resistance is used in voltage dividing network to produce desired effect. R1 and R7 are 500 ohms, 5 watts.

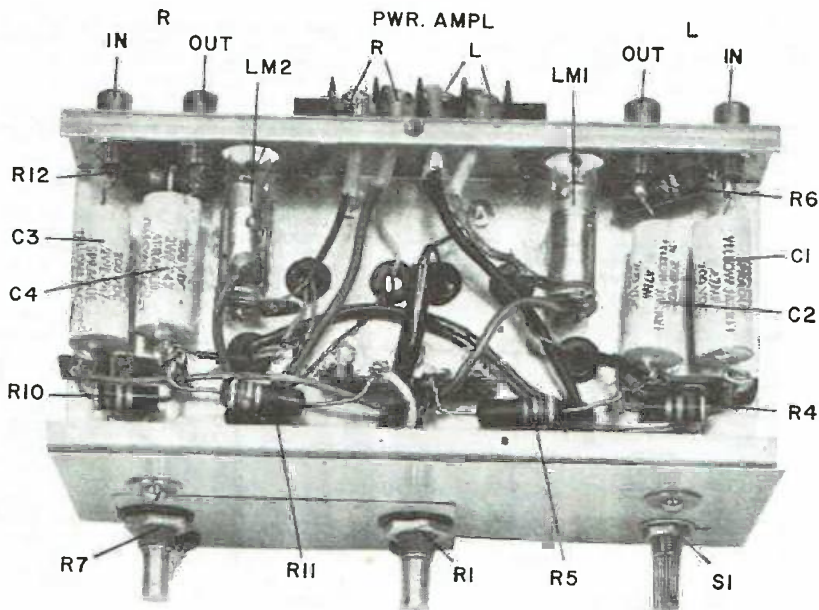
expander-compressor. If a monophonic unit is desired, build only one channel.

## How it works

Briefly, here's how the expander-

compressor works. Since operation of both channels is identical, only one is described.

A small signal voltage across potentiometer R1 from the power amplifier is stepped up by transformer T1,



*Underchassis closeup shows where the rest of the compandor components are located. The circuit is completely passive and, therefore, has no power supply of any sort.*

passed through current-limiting resistor R2 and applied to neon lamp NE-1. Lamp LM1 is a fuse to protect the transformer and lamp NE-1. NE-2, located on the front panel, is a remote indicator showing NE-1's response. At some pot setting, NE-1 and NE-2 will begin to flicker. The intensity of NE-1 will be proportional to the voltage across it. This light, falling on photoconductive cadmium selenide (or cadmium sulphide) cell PC-1, causes its conductance (resistance) to vary with the illumination applied. Resistance decreases as light intensity increases.

Cell PC-1 now acts as a resistor that varies linearly with amplitude variations of the power amplifier.

Using the changing cell resistance in the signal-voltage dividing network consisting of R4, R5, R6 and blocking capacitors C1 and C2, the switching arrangement behaves as follows:

Position 1 (EXP) places the light-variable section of the dividing network in series with the signal. On louder passages the signal has less resistance to overcome and expansion occurs.

Position 2, the off position, allows the signal, minus the insertion loss of the expander-compressor, to pass straight through. By selecting your listening level with the expander-compressor in its off position, compensation is automatic.

Position 3, COMP places the dividing network across the input signal. On louder passages more drop takes place across series resistor R4 and compression occurs.

Parts placement is not critical. There is no interference between channels with the compact arrangement shown.

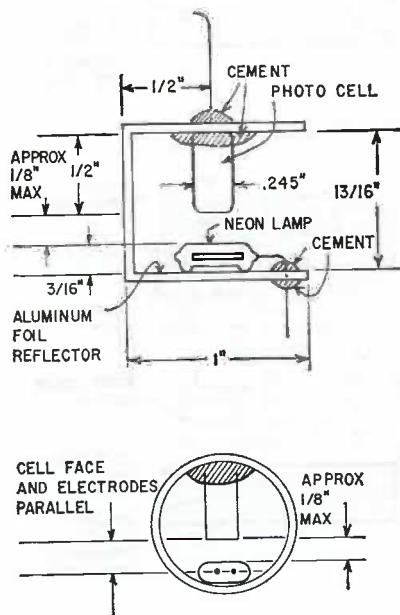
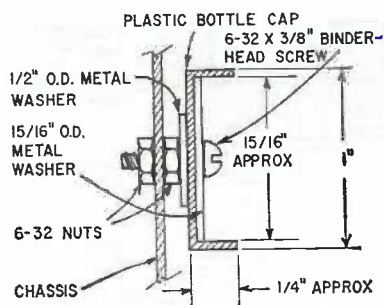


Fig. 2—Lamp-cell housings are plastic. Secure lamps and cells with clear cement.

The function selector switch, the two operating-level pots and the two remote-indicator neon lamps are mounted on the front panel. The lamps are held in grommets and jeweled bushings are used for dressup.

On the back of the chassis are four phono jacks (an input and output pair for each channel) and a terminal strip with four screw terminals (two for each drive-signal circuit). Other parts

are located as shown in the photographs. Grommets are used to protect leads passing through the chassis.

Resistors and capacitors are mounted on solder-lug terminal strips or supported by their leads. Current-limiting resistors R2, R3 and R8, R9 are mounted on a resistor board on top of the chassis near the lamp housings.

### Build the lamp-cell combinations

Each lamp-cell combination should be in a light-tight housing. The housing shown in Fig. 2 was made from a 1 5/16" ID plastic bottle. Metal 35mm film cans are also suitable. Saw about 1/8" off the bottom of each bottle, and paint the inside black. Cement the cells and neon lamps in place. Give the outside of the module a one-turn-plus wrap of black construction paper. Slit the ends of the paper for the neon lamp leads, and slit the center for those of the cell. Next, apply cement to the wrap, which is pressed over the cell and lamp leads. Then plug the assembly into its cap, which contains a metal stiffening washer, and bolt the assembly upside down on the chassis.

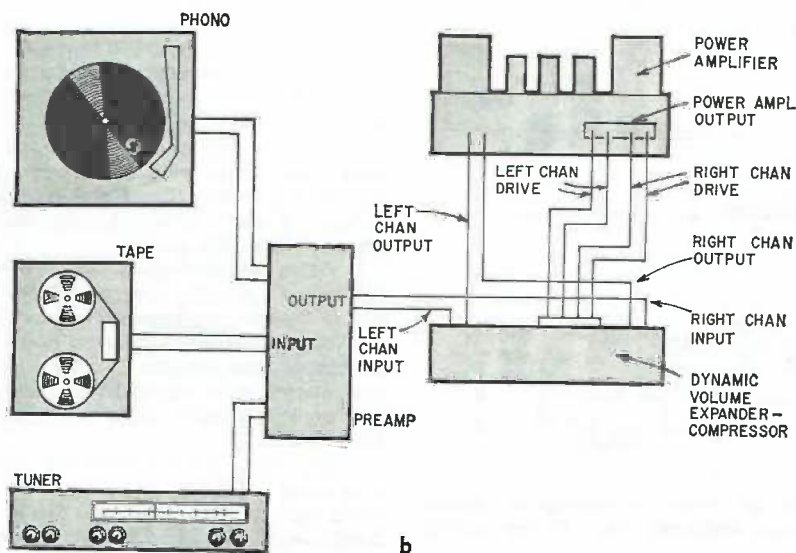
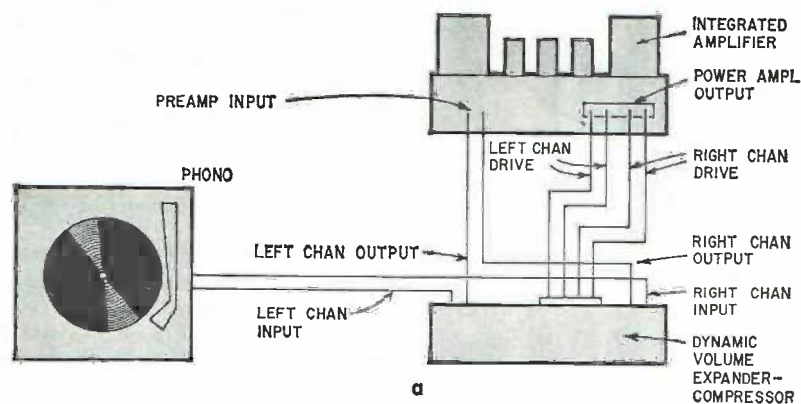
Pieces of aluminum foil about 7/16" long x 3/16" wide, cemented *bright side in* beneath the neon drive lamps, will increase the unit's dynamic range, sometimes considerably. Cement all components with a clear-drying adhesive such as Elmer's Glue or White Glue.

The neon lamp should be not more than 1/8" from, and perpendicular to, the face of the cell being driven. Lamp anodes should be parallel to the face of the cell.

Alternate lamp-cell housings can be devised, but must be lightproof. The flash of a room lamp or the flicker of fluorescents will affect the sensitive cells. Light from one channel striking the cell of the other will disrupt the entire system.

### How to use it

When used with an integrated system (preamp and power amplifier



**Fig. 3—Hookup for integrated system (a), program material is fed directly to inputs. Drive signals come from speaker terminals and compandor output is fed to preamp. Component system (b) compandor is inserted between preamp and power amplifier.**

on one chassis), the expander-compressor is inserted between the program source and the preamp, as shown in Fig. 3-a. Be sure to use the right preamp input, since compensation for a record player, for instance, is differ-

ent from that for tape. If the unit is unstable with an integrated system, shield the top of the chassis with perforated metal and add a metal bottom plate.

With component systems, place the expander-compressor between the preamp output and the power amplifier input (Fig. 3-b).

In this location, the unit will act on program material from the preamp, allowing you to switch from one program source to another without changing cables.

Signal voltage to drive the unit is obtained from the output of the power amplifier, preferably from a 16-ohm tap. If this impedance is not available, use the 8- or 4-ohm tap. Never connect the unit across a 70.7-volt or other constant-voltage line. A small signal swing drives it through its entire dynamic range.

Regardless of which terminals the drive circuit is connected to, the speaker system always sees its matching impedance.

The volume level at which the unit will go into action depends largely on speaker sensitivity. If this is more sound than you customarily use when loud passages are played, the circuit shown in Fig. 4 will compensate for above-average transducer efficiency, usually permitting full expansion and compression at normal listening levels.

To connect the unit to high output impedance amplifiers of any kind, place a resistor in series with one side of the line to the drive circuit. Use a value about 50 times that of the output impedance.

Input and output cables for use with integrated systems can be standard shielded audio cable. A length of

3' is recommended although longer ones can be used if a capacitance of about 100pf is maintained.

With component systems, your existing audio cables may be used. Cathode follower preamps usually permit the use of much longer cables. Lamp cord of any reasonable length can be used between the power amplifier output and the drive terminals.

The expander-compressor can perform several limiting jobs. It can be used when making tape recordings to prevent overload distortion right in the concert hall, if you wish.

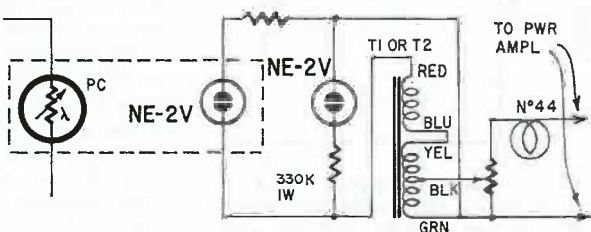
You can hear soft passages of program material played in high-ambient noise locations. By compressing the loud passages and raising the volume level of all the material to where the loud passages were, the soft passages will come through as only the loud ones previously did.

#### Even works with TV

Used with a TV receiver, it keeps those "important messages from the sponsor" at a level commensurate with their importance. Connect from the high side of the de-emphasis capacitor to ground, and across the volume control. Remove the low side of the de-emphasis capacitor from ground; a 3' cable will substitute its capacitance closely enough.

Here's another bonus. In the compression mode, try playing those noisy records you were about to throw away. Start with the operating level controls fully clockwise.

Fig. 4—Transducer too efficient? This circuit takes care of that problem should it arise in your unit.





What happened to those clicks and crackles? Being spike pulses, they were cut off. The level controls can now be adjusted to the point where only loud clicks are cut off. Too much compression will make even a splendid performance sound singsong.

Now get out your finest tapes or records and play them with dynamic expansion. Set the amount of expansion desired for the *loudest parts* of the

material. As the operating level controls are advanced, the remote indicator lamps will begin to flicker, indicating when the compandor takes hold.

The great dimensional changes you now hear left the recording studio as tiny pips on the signal, some of them barely got out at all. But now, even at low listening levels, the full dynamics of music are present.

## FM STEREO TUNER

by KENNETH E. BUEGEL

FM TUNERS FOR STEREO RECEPTION have steadily improved in quality. Today's audiophile can select a solid-state unit with specifications which were never achieved before.

All tube designs, all the older transistor designs, and the present FET designs use an agc technique known as reverse bias in which a negative bias is applied to either a grid or base to reduce the output current (and gain) of the device. While reverse bias

works well with devices such as tubes and certain FET's, which can operate at zero or negative bias, it fails when used with ordinary transistors which must always have some forward bias to maintain operation.

Strictly speaking, the forward bias on the transistor was only reduced until the gain decreased at some very low collector current. Unfortunately, this meant that the base-emitter junction could be a very good rectifier for the same strong signal that was producing the bias. Thus this biasing technique was a failure in operation and the setting was ripe for the introduction of FET's.

The use of a different type of transistor, however, solves the overload problem. This transistor is known as the "forward agc" type—one in which gain reduction is caused by an *increase* in the forward bias applied to the transistor as shown by the curve of Fig. 1. Note that at collector currents of only 2-3 mA the power gain of the device is maximum, but when the current has increased to a higher value the power gain may even become negative, with less output than input.

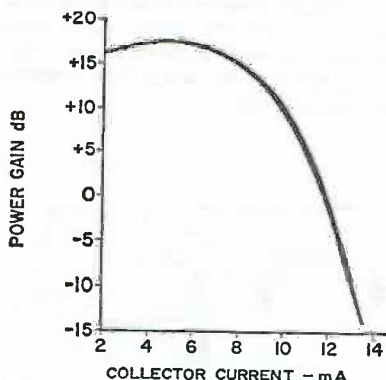


Fig. 1—Curve of forward-agc transistor. Low collector currents yield high gain.



Of course, since the base-emitter junction is heavily biased, a strong signal cannot be rectified at this point. Thus, in the FM-1 tuner described here, we can apply a signal of 0.2V rms to the antenna terminals without degrading performance.

Although forward-bias transistors can successfully compete with FET's under extremely strong signal conditions, they deliver superior performance when handling very weak signals. We get this kind of performance by close impedance matching between the coupled circuits, which implies maximum power transfer. With an antenna impedance of 300 ohms it is simply not possible to transform this upwards to the higher values required for field-effect transistors. Circuit configurations capable of this transformation such as transmission lines and tuned cavities are bulky and of prohibitive size.

The transistors used in this design present a much lower input impedance and consequently more useful power is transferred. The result is a tuner with high sensitivity and an

extremely important side benefit—no shielding is required in the rf section. Whereas a stray coupling capacitance as low as 0.1 pF (16,000 ohms @ 100 MHz) can couple a considerable amount of power between two high-impedance circuits, the same stray capacitance effects little power transfer between impedances nearer 1200 ohms.

Remember also that any electrostatic shield placed over an inductance must be much larger than the coil diameter to prevent drastic reduction of Q values and inductance since the shield acts as a single shorted turn coupled to the inductor.

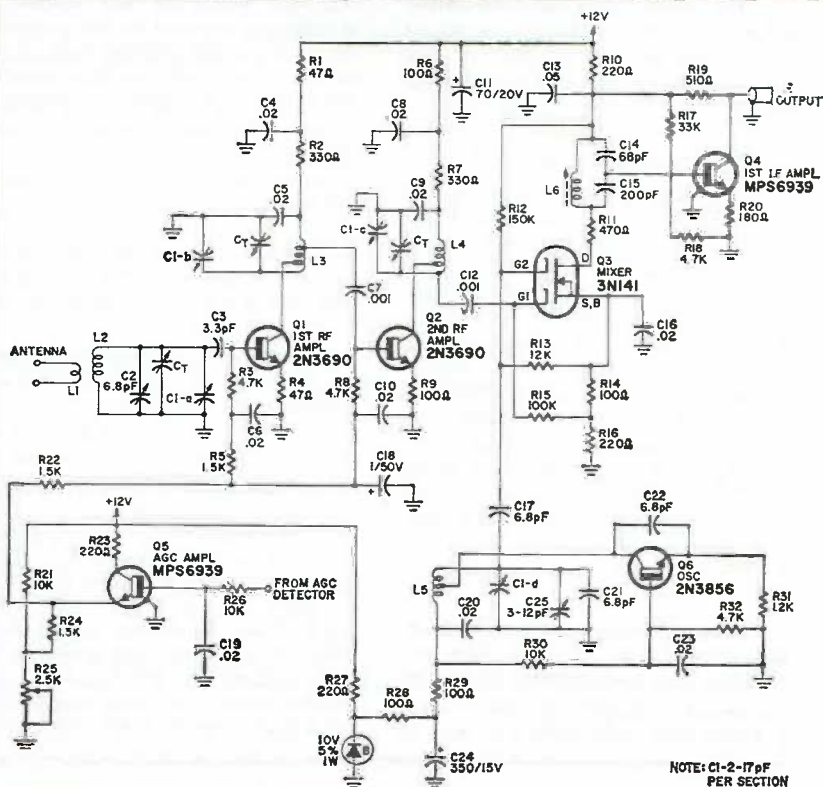
The crystal filter impedances in the FM-1 are closely matched to reduce phase non-linearity and preserve good stereo separation.

On weak signal reception, forward bias to Q1 and Q2 (Fig. 2) is supplied through R22 and R24 from the junction of R21 and R25. Adjusting R25 sets the bias for highest sensitivity. As signal strength increases, the base voltage on Q5 becomes more positive until, at a higher signal level, Q5 is providing bias to Q1 and Q2.

## FM-1 Specifications

Using the detailed alignment procedure that appears on these pages, the reader should be able to achieve or improve the following figures.

20 db Quieting Sensitivity .....	1 $\mu$ V	Multiplex Separation .....	.45 db mid-frequency
30 db Quieting Sensitivity .....	1.4 $\mu$ V		35 dB @ 50 Hz
50 db Quieting Sensitivity .....	3.9 $\mu$ V		30 dB @ 10 kHz
Capture Ratio .....	1 dB	Output Level .....	adjustable up to 1Vrms
AM Suppression .....	60 dB	Output Impedance .....	approx. 20,000 ohms
Harmonic Distortion .....	0.5% (measured at detector output)	Distortion .....	1%, 1000 Hz, 75 kHz deviation
Hum and Noise .....	less than -70dB	SCA Suppression .....	48 dB
Spurious Response .....	less than -90dB	Hum and Noise .....	45 dB
Image Rejection .....	less than -90dB		-60 dB (below reference IV output with 19 and 38 kHz components notched out)
Half i.f. Response .....	less than -80dB		
Adjacent Channel Selectivity .....	-70 dB		
Overload Sensitivity (for 1% distortion) .....	0.2V rms		



The resistance in the collector circuits of Q1 and Q2 is low to prevent saturation at these higher collector currents. The base connections to L2 and L3 provide maximum power transfer from the tuned circuits. Q3 is a dual-gate FET mixer with the rf signal applied to gate 1 and the local oscillator signal to gate 2.

Transistor Q6 is arranged to provide a low distortion oscillator signal to the mixer. An additional Zener regulator prevents frequency variation while the R28-C24 combination removes any Zener noise. The base of Q4 is matched to the mixer output by a tap point formed by the capacitive division of C14 and C15.

Transistor Q4 provides the 500-ohm output impedance for the first crystal filter. Each filter requires about 5 pF at the IN terminals to give the specified bandpass. For F101 (Fig. 3) this capacitance is the length of shielded cable connected to Q4. For the second filter it is the output capacitance of IC101.

Transistor Q101 amplifies the signal and applies it to Q103 and IC101. The gain of Q103 and Q104 is just enough so that the i.f. voltage rectified by D101-D102 will provide further bias to the transistors only when the signal is well past the point of maximum noise reduction, but long before the signal level reaches the overload level. The design gain of the

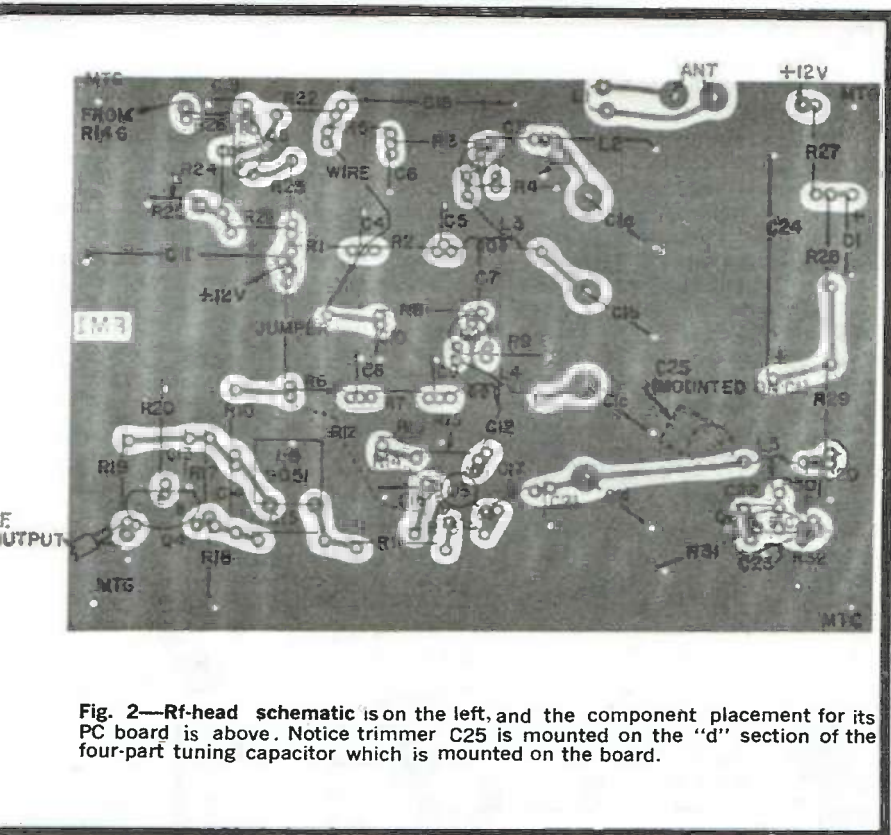


Fig. 2—Rf-head schematic is on the left, and the component placement for its PC board is above. Notice trimmer C25 is mounted on the "d" section of the four-part tuning capacitor which is mounted on the board.

Q103 and Q104 stages is centered within this range.

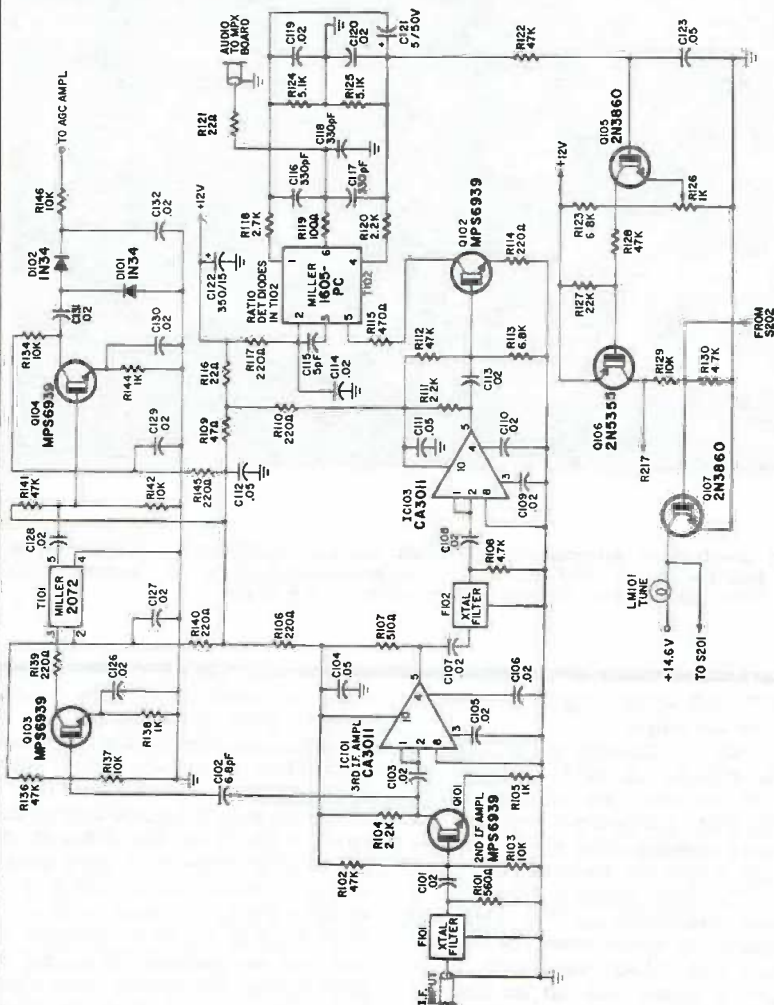
All i.f. transistors are of a special type with the emitter lead placed between the base and collector leads. This lead arrangement prevents unwanted coupling (and phase shift) between output and input of each stage.

The ratio detector, T102, has a linear bandwidth of 500 kHz and contains the diodes within the housing. Because of T102's bandwidth, alignment is simple, and can be done almost as well with off-the-air signals as with a signal generator.

The tune indicator lights when the signal is tuned correctly. The voltage at R125 will be about 1.1 V positive without a signal and rises to

nearly 1.3 volts when even a weak signal is tuned in. R126 provides an adjustment for this voltage discrimination. When the voltage at R125 rises in the presence of a signal, Q105 is turned on, in turn saturating Q106 and Q107. LM101 in the collector circuit of Q107 lights to indicate correct tuning. A S202 contact (Fig. 4) is wired to the turn-on bias for Q107. If S202 is set to the stereo position this bias will be clamped to ground by Q207 unless the 19-kHz pilot signal has resulted in turning on the stereo indicator, LM201. Thus if the MUTE switch is on, and S202 is set to the stereo position, only stereo signals will be heard.

Since Q107 saturates at this time,



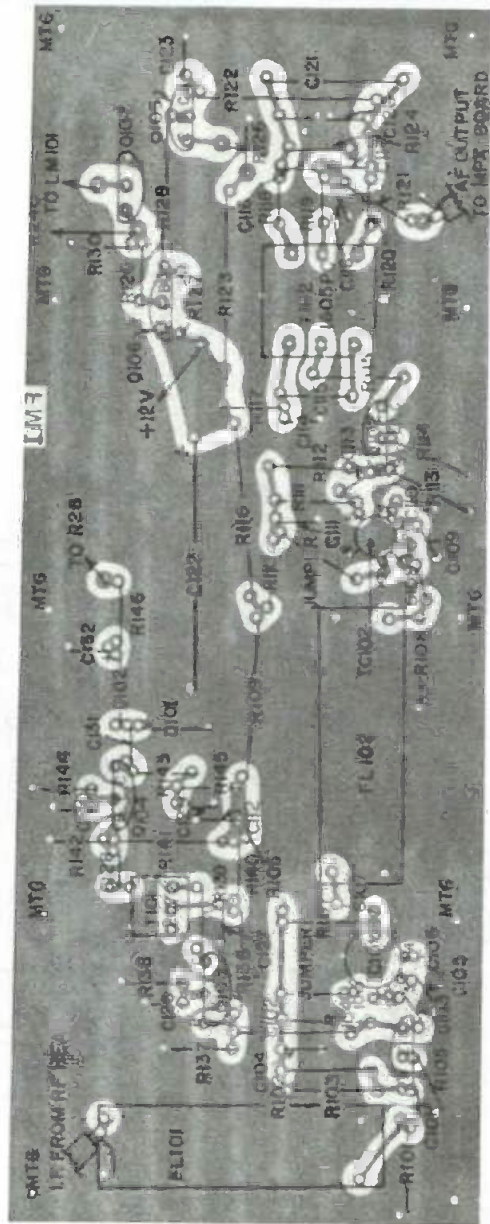


Fig. 3—l.f. strip for the FM-1 uses two crystal filters and two IC's. Diagram below shows where to place parts on the PC board.









a contact on S201 may return source resistors of Q210 and Q211 to ground, allowing the audio to appear at the output jacks.

The collector voltage at Q106 also saturates Q208 in the stereo indicator circuit, allowing an indication only when stereo is tuned in.

The detected composite signal feeds the high-impedance input of Q201. T201 and C203 present a high impedance in the emitter circuit of Q201. The output voltage appearing across R201 is low in 19-kHz components. T202 and C202 are tuned to 67-kHz to eliminate SCA interference at the stereo outputs. The 19-kHz pilot signal appearing across T201 is amplified by Q203. T203 and T204 are tuned to the 19-kHz signal. The input to emitter follower Q204 is a very pure 19-kHz signal which provides superior lock-in of the oscillator even on extremely weak signals. Since the voltage at the emitter of Q204 is a low impedance it is used to drive the pilot detectors, D201 and D202, as well as synchronizing Q209.

The rectified pilot signal saturates Q205, which in turn provides a turn-on bias to Q206, the stereo lamp driver. The emitter of Q206 is in series with Q208. Therefore, the indicator will not light unless Q208 is also turned on. Q208 is driven by the voltage appearing at the collector of Q106, and this voltage will bias Q208 only when a signal is received. The stereo indicator lights only when a stereo signal is received. There are no false indications between stations.

Locked oscillator Q209 can only operate when pin 6 of T205 is returned to ground by the saturation of Q206. T206 in the collector circuit is tuned to 38kHz and drives the balanced detectors D203-D205 and D204-D206. Balanced detection reduces the 38-kHz components which can appear across the de-emphasis networks, R224-R225-C219 and R226-R227-C220.

The gates of output amplifiers Q210 and Q211 are returned to ground potential. Source resistors R230 and R231 are switched to ground to turn on and to +12 volts to mute. In muting, the audio output is reduced over 50 db. This switching voltage is taken from the collector of Q107, the TUNE lamp driver. The output of each channel may be adjusted to any voltage up to 1V rms by the output controls in the drain circuits of Q210 and Q211.

Resistor R204 selects the amount of composite signal, minus 19-kHz and 67-kHz components, applied to the switched detectors. This composite output contains the L + R signal at 50-Hz to 15-kHz, as well as the L - R sidebands at 23-kHz to 53-kHz. These sidebands may be attenuated by multi-path reception as well as phase distortion in the i.f. strip. The detector transformer used in the FM-1 has a linear bandwidth of 500-kHz to minimize this distortion. However the tuner alone is not able to compensate for all possible causes of L - R sideband attenuation. Therefore C204 should be installed on the following basis: use the .02- $\mu$ F unit if this multiplex board is *not* used with the remainder of the FM-1, or if all your useable signals originate more than 30 air miles from your receiving location. Use the .05- $\mu$ F capacitor when the multiplex is used with the FM-1 and the stations are both local and distant.

The power supply provides +14.6V for LM101 and LM201 and regulated 12V for all other circuits. Q212 functions as a capacitance multiplier; the +14.6V has less than 5 mV p-p ripple. D210 provides a regulated output from Q213. If there is excessive Zener noise you can add a simple R-C network to eliminate it.

Diode D209 serves only one function; to prevent destruction of Q213 in case of a wiring error or accidental short during testing. With-

out D209 any short would cause the charge in C232 to be placed directly across the base-emitter junction of Q213. The resulting current flow would destroy this transistor. With D209 in place any short drains C232 of its charge at the cost of only a momentarily increased dissipation in Q213.

Three circuit boards are used in the tuner. These boards are the rf head, i.f. strip, and multiplex and power supply boards. Each board is completed according to the sequence shown. When all boards are wired, they are temporarily mounted in the chassis for alignment. After alignment is complete, the dial panels are mounted and the unit is ready for final wiring and assembly.

#### Alignment procedure

Set dial to 88 MHz. Loosen the dial drum set screws and slide the drum off the shaft. Using small pieces of masking tape, tape the dial cable in place so it cannot become tangled.

#### I.F. STRIP

1. Set generator for a sweep width of about 300 kHz, centered about the 10.7 MHz. If possible, use post marker adding to set two markers; one at 10.6 and the other at 10.8 MHz. Set the generator output level to about 200  $\mu$ V. Connect the output cable ground to the edge of the i.f. strip near the input to F101. Clip the "hot" lead to the insulated cable jacket leading to F101; DO NOT tie the hot lead directly to the input pin of F101.

2. Set the bottom (primary) slug of T102 for a symmetrical response with the top slug tuned off resonance. The scope vertical input should be connected to the output lead from R121, and the horizontal input should be fed from the sweep voltage in the generator. Follow the generator instructions for connections and phasing procedures.

3. Next adjust the top slug (secondary) of T102 for best linearity

between the 10.6 and 10.8 MHz markers. Do not retune the bottom slug.

4. Connect a dc vtvm set on the lowest range between ground and the top end of R119. Slowly adjust the top slug of T102 for exactly zero volts. This should be a very small part of a turn.

5. Connect the vtvm to the output lead from R146 and increase the generator output until you can read some voltage above zero. Tune T101 for a maximum reading. Reduce the generator output and repeak T101.

6. Place a small ceramic-disc capacitor in series with the output lead of the generator and connect the capacitor to L4. Leave the vtvm on R146. Increase the generator output and tune L6 for a maximum reading. Reduce the output and repeak L6. This completes the i.f. alignment.

#### RF HEAD

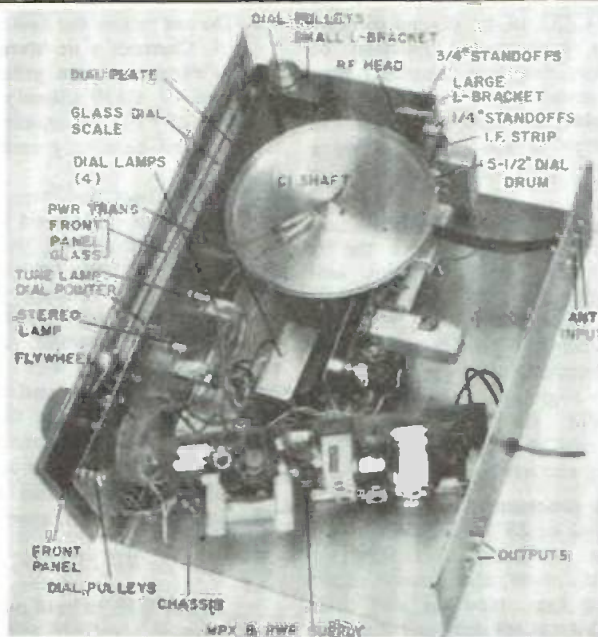
1. Set R25 to mid-position; connect a small amplifier and speaker to the output lead from R121. Set the generator output to 88 MHz with enough deviation so that an audio signal can be heard from the speaker.

2. With the plates of C1 fully meshed adjust L2, L3, L4, and L6 for maximum quieting of the received signal. Reduce the generator output as needed so that the signal is always slightly noisy. Use a small nylon or fiberglass tool to move the coil turns.

3. Set the generator output to 107.8 MHz and set C1 so its plates are at minimum capacitance. Peak each trimmer capacitor on C1 for maximum quieting, as well as C25.

4. Repeat steps 2 and 3.

5. Set generator output frequency to a quiet spot near the upper end of the dial, around 105 to 106 MHz. Tune the receiver to this frequency and repeak the trimmer on L4 for best quieting. "Rock" C1 about this point while adjusting this trimmer since there is some interaction between peaking this trimmer and the local oscillator frequency. If you do



Chassis layout of tuner sections used by the author. A large L-bracket supports i.f. and rf boards. C1 is mounted on rf board, concealed by dial drum here.

not rock C1 you may only succeed in "fine-tuning" the oscillator frequency and sensitivity will be reduced from optimum.

6. Repeat steps 2, 3, and 5.

7. Tune C1 to a quiet spot near the bottom of the dial and adjust R25 for maximum quieting on a very weak input signal. At this time you may wish to check the sensitivity. It should be 20 dB quieting at less than 1  $\mu$ V input. L1 should be spaced about 1/8 inch below L2. If sensitivity is too low, carefully push L1 closer to L2 and repeak the antenna trimmer capacitor at 107.8 MHz and adjust the turns on L2 at 88 MHz. Ten sets of transistors were tried in the prototype unit and sensitivity varied between 0.95 to 0.55  $\mu$ V for 20 dB quieting.

## MULTIPLEXER

You will need an oscilloscope and a multiplex generator to align this section. An ac vtvm is also helpful when making separation measurements.

1. Connect the scope vertical input to the wiper of R204 and inject a 67-kHz input at the input to the multiplex board at C201. Tune the slug in T202 for a minimum output as seen on the scope.

2. Connect the scope to the Q202 end of C202. Inject a 19-kHz pilot signal at a 25-mV rms level and tune T201 for maximum trace height.

3. Connect the scope to the collector end of C206 and set S202 to the MONO position. Tune T203 and



T204 for maximum trace height. Reduce the input level of the generator and carefully repeak the tuning of transformers T201, T203 and T204.

4. Connect the scope to either channel A or B output and inject a 19-kHz signal at 25 mV rms. Tune oscillator transformer T205 for zero beat as observed on the scope. S202 must be in the STEREO position.

5. Connect the scope to either end of C214 and tune T206 for maximum 38-kHz trace height.

6. Set up the generator to deliver a 0.25V rms L-only or R-only composite signal to the input. Connect the scope vertical input to channel A and the horizontal input to channel B. Slowly tune T205 until the scope is displaying either a vertical line or a horizontal line. One of the settings of T205 will be very unstable; use the setting which is the most stable and connect the scope so that an L-only input gives a vertical line.

7. Repeak T201 for the most vertical line and adjust R204 for the best vertical line. Switching to an R-only input should result in a horizontal line on the scope.

8. Inject an L plus R signal, with pilot, and set R232 and R233 for equal outputs from each channel.

### SPECIAL TUNER PARTS

The following special parts are needed for the construction of the tuner. All are available from Transitek, P.O. Box 98205, Des Moines, Washington.

Tuning capacitor C1	\$7.50
Miller coil No. 9051	\$2.04
Miller coil No. 2072	\$2.58
Miller coil No. 1354PC	\$2.61
Miller coil No. 1355PC	\$2.61
Miller coil No. 1361	\$1.80
Fairchild 2N3690	\$1.25
Crystal filter	\$13.50
Rf printed circuit	\$5.90
If printed circuit	\$7.20
Mx and power supply printed circuit	\$9.20

## COMPLETE TUNER PARTS LIST

### PARTS LIST RF HEAD

All resistors are 1/4 W, 5% unless noted as 1/2 W.

R1, R4—47 ohms  
R2, R7—330 ohms  
R3, R8—4700 ohms  
R5, R22, R24—1500 ohms  
R6, R9, R14, R29—100 ohms  
R10, R16, R23—220 ohms  
R13—12,000 ohms  
R15—100,000 ohms  
R21, R26, R30—10,000 ohms  
R25—2500 ohms, 1/4-watt trimmer  
R31—1200 ohms  
R32—4700 ohms  
R11—470 ohms, 1/2 watt  
R12—150,000 ohms, 1/2 watt  
R17—33,000 ohms, 1/2 watt  
R18—4,700 ohms, 1/2 watt  
R19—510 ohms, 1/2 watt  
R20—180 ohms, 1/2 watt  
R27—220 ohms, 1/2 watt  
R28—100 ohms, 1/2 watt

### Capacitors

C1—4 gang FM tuning capacitor, (2-17 pF per section. Trimmer range 0.5 to 12 pF. Only three trimmers are used. Local oscillator section uses separate trimmer.  
C2, C17, C21, C22—6.8 pF, NPO ceramic  
C3—3.3 pF, NPO ceramic  
C4, C5, C6, C8, C9, C10, C16, C19, C20, C23—.02  $\mu$ F, 25V cer.  
C7, C12—.001  $\mu$ F, ceramic  
C11—70  $\mu$ F, 20V electrolytic (Mallory MTA-70E20 or equiv.)  
C13—.05  $\mu$ F, 25V ceramic  
C14—68 pF polystyrene  
C15—200 pF polystyrene  
C18—1  $\mu$ F, 50V electrolytic (Mallory MTA 1D50 or equiv.)  
C24—350  $\mu$ F, 15V electrolytic (Mallory MTA 350 F 15 or equiv.)  
C25—3-12 pF ceramic trimmer (Centralab 822FZ or equiv.)  
C<sub>T</sub>—trimmers on C1

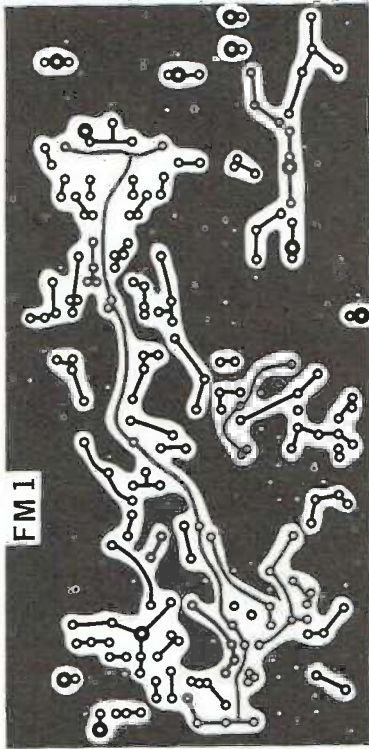
### Semiconductors, coils

Q1, Q2—2N3690 (Fairchild)  
Q3—3N141  
Q4, Q5—MPS 6939  
Q6—2N3856  
D1—10V Zener diode, 1W, 5%  
L1—2T, #26 enamel, next to ground end of L2  
L2—4T, #18, 1/4" dia, 7/16" long  
L3—4T, #18, 1/4" dia, 7/16" long, tap at 1T and 3T  
L4—same as L3 but tap only at 3T  
L5—2T, #18, 5/16" dia, 1/4" long, tap at 3/4 T from C1d  
L6—J.W. Miller 9051

### IF Strip (101 to 199 series)

C101, C103, C105, C106, C107, C108, C109, C110, C113, C114, C119, C120, C126, C127, C128, C129, C130, C131, C132—.02  $\mu$ F, 25V ceramic





Here are the printed circuit patterns used for the FM-1. They are shown one-half actual size and must be enlarged photographically. The multiplex/power supply is above, the rf head below (left) and the i.f. strip is below.



C102—6.8 pF, NPO ceramic  
 C104, C111, C112, C123—.05  $\mu$ F, 25V ceramic  
 C115—5 pF, polystyrene  
 C116, C117, C118—330 pF ceramic  
 C121—5- $\mu$ F 50V electrolytic (Mallory 5D50 or equiv.)  
 C122—350- $\mu$ F 15V electrolytic (Mallory 350-F15 or equiv.)  
 C124, C125—omit due to design change

#### Resistors all $\frac{1}{4}$ watt 5% unless noted

R101—560 ohms,  $\frac{1}{2}$  watt  
 R102, R112—47,000 ohms,  $\frac{1}{2}$  watt  
 R103—10,000 ohms,  $\frac{1}{2}$  watt  
 R104, R111, R120—2200 ohms,  $\frac{1}{2}$  watt  
 R105, R138, R144—1000 ohms  
 R106, R110, R117, R139, R140, R145—220 ohms  
 R107—510 ohms,  $\frac{1}{2}$  watt  
 R108—4700 ohms,  $\frac{1}{2}$  watt  
 R109, R121—22 ohms,  $\frac{1}{2}$  watt  
 R113—6800 ohms,  $\frac{1}{2}$  watt  
 R114—220 ohms,  $\frac{1}{2}$  watt  
 R115—470 ohms,  $\frac{1}{2}$  watt  
 R116—47 ohms,  $\frac{1}{2}$  watt  
 R118—2700 ohms,  $\frac{1}{2}$  watt  
 R119—100 ohms,  $\frac{1}{2}$  watt  
 R122—47,000 ohms  
 R123—6800 ohms  
 R124, R125—5100 ohms  
 R126—1000 ohms,  $\frac{1}{4}$ -watt trimmer  
 R127—22,000 ohms  
 R128, R136, R141—47,000 ohms  
 R129, R137, R142, R143, R146—10,000 ohms  
 R130—4700 ohms  
 F101, F102—XTAL FILTER, \$13.75 ea.

#### Specifications for F101 and F102:

Center frequency—10.7 MHz  
 Bandwidth—1.5db, 200 kHz min., 3db, 240 kHz min., 20db, 500 kHz max., 50db, 1100 kHz max.  
 Ultimate attenuation—50dB minimum  
 Maximum insertion loss—5dB  
 Phase linearity—Within plus to minus three degrees of best straight line over 200 kHz minimum bandwidth  
 Input to filter—500 ohms plus 4.2 pF  
 Load on filter—500 ohms plus 7.5 pF  
 Size—51mm long, 18mm wide, 33mm high  
 Operating temperature—45 to 110 degrees F.  
 T101—J. W. Miller 2072  
 T102—J. W. Miller 1605 PC  
 Q101, Q102, Q103, Q104—MPS 6939  
 Q105, Q107—2N3860  
 Q106—2N5355  
 D101, D102—1N34  
 LM101—12-14V, 25-40 mA

#### Multiplex and Power Supply (201-299 series)

C201, C209, C224, C225—1 $\mu$ F/50V, electrolytic (Mallory MTA 1D50 or equiv.)

C202—560 pF polystyrene  
 C203, C206, C207, C211, C212—.01- $\mu$ F polystyrene  
 C204—.02  $\mu$ F or .05- $\mu$ F, disc ceramic; see text  
 C205—.001- $\mu$ F polystyrene  
 C208—.01  $\mu$ F disc ceramic  
 C210—15- $\mu$ F, 35V, electrolytic (Mallory MTA 15E35 or equiv.)  
 C213—5- $\mu$ F 50V, electrolytic (Mallory MTA 5D50 or equiv.)  
 C214—.0015  $\mu$ F, polystyrene  
 C215, C216, C217, C218—.002- $\mu$ F temp. stable ceramic  
 C219, C220—.001- $\mu$ F temp. stable ceramic  
 C221, C222—.05  $\mu$ F, disc ceramic  
 C223—10- $\mu$ F, 35V, electrolytic (Mallory MTA 10D35 or equiv.)  
 C226—500- $\mu$ F, 25V, electrolytic (Mallory TC-2505B or equiv.)  
 C227, C228—500- $\mu$ F, 15V, electrolytic (Mallory MTV 500DN15 or equiv.)

#### All resistors $\frac{1}{4}$ -W, 5% unless otherwise noted

R201, R206, R212, R213, R214, R215, R216, R217—10,000 ohms  
 R202, R207—2 megohms  
 R203—3300 ohms  
 R204—1000 ohms,  $\frac{1}{4}$  watt trimmer  
 R205—1000 ohms  
 R208, R218—6800 ohms  
 R209—2,200 ohms  
 R210, R234, R235—100,000 ohms  
 R211, R230, R231—4700 ohms  
 R219—560,000 ohms  
 R220, R221, R222, R223—39,000 ohms  
 R224, R225, R226, R227—150,000 ohms  
 R228, R229—1 megohm  
 R232, R233—25,000 ohms,  $\frac{1}{4}$ -watt trimmer  
 R236—220 ohms,  $\frac{1}{2}$ -watt  
 R237—100 ohms,  $\frac{1}{2}$ -watt  
 R238—56 ohms,  $\frac{1}{2}$ -watt  
 R239—470 ohms  
 T201, T203, T204—J.W. Miller 1361  
 T202—J.W. Miller 1362  
 T205—J.W. Miller 1354-PC  
 T206—J.W. Miller 1355-PC  
 T207—117V pri., 24V CT secondary, 1A power xfmr.  
 D201, D202, D203, D204, D205, D206—1N-4154  
 D207, D208, D209—40267  
 D210—13V Zener diode, 1W, 5%  
 Q201—2N3391  
 Q202, Q203, Q204, Q206, Q207, Q208, Q209—2N3860  
 Q205—2N5355  
 Q212—40408  
 Q213—40407  
 S201, S202—rocker switches  
 LM201—12-14V, 25-40 mA pilot bulb

# BOOKSHELF SPEAKER SYSTEM

by ALEXANDER N. RETSOFF

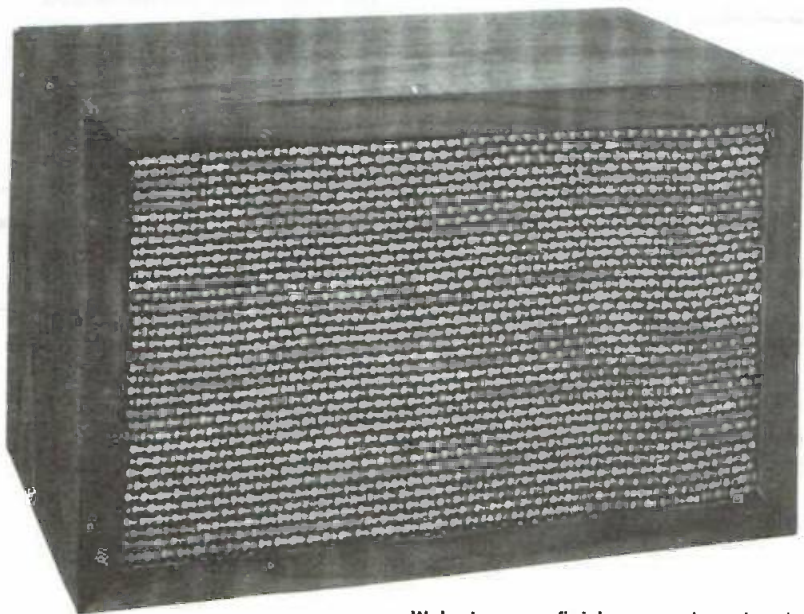
WHAT I WANTED WAS A COMPACT speaker system with a wide, smooth frequency response, low distortion and sufficient power-handling capability to fill at least a small room with realistic sound.

The requirement for smooth response and low distortion in a small cabinet calls for an acoustic suspension system. Although horn-loaded speakers provide excellent response and low distortion, the laws of physics being what they are, it is impossible to have a "small" horn-loaded

speaker that extends to 50–80 Hz. That being the case, the choice was between a bass-reflex system and an infinite baffle.

## Bass-reflex speakers

The bass-reflex approach includes all systems in which the sound emanating from the rear of the speaker is allowed into the room. Such a system may go under the name of "bass-reflex," "tuned-port," "ducted-port," "distributed-port," "Helmholtz resonator" or some other title. In all cases, a



Walnut-veneer finish can make a handsome extension speaker for any hi-fi.

hole somewhere in the cabinet allows the rear sound wave from the speaker to enter into the room. These systems boast relatively high efficiency and extended low-frequency response.

In general, every dynamic loudspeaker exhibits a primary resonance at the lower end of its frequency response. The frequency at which the resonance occurs depends upon the mass of the moving system, i.e. the cone and voice coil, and the compliance of the suspension.

For a speaker in free space, the suspension is made up of the flexible spider, which supports the voice coil end of the cone, and the rim support at the large end of the cone. The system is analogous to a weight on the end of a spring. Once the weight is set in motion it bobs up and down at a rate or frequency that depends on the mass of the weight and the compliance of the spring. By increasing the size of the weight and/or picking a softer spring, frequency can be lowered. The converse is also true.

The point is that a speaker's output drops off very rapidly below its resonant frequency. Also, at its resonant frequency the cone likes to move and does so very readily, resulting in a peak in the response curve. Just like the weight on the spring, once in mo-

tion at its resonant frequency, the speaker tends to keep moving even after the signal has disappeared. This "hangover" causes poor transient response and muddy sound.

The degree to which the peak appears in the response curve and the transient response is impaired is determined by the "Q" of the resonant system. This in turn is controlled by the damping or friction in the moving system. The chief causes of damping are the air loading on the speaker cone, the friction in the suspension system and the damping factor of the amplifier.

If damping is high, there is only a mild peak in the response curve, since the energy is rapidly dissipated in the friction; the cone quickly comes to rest after the excitation is removed. An underdamped system will exhibit a large peak and long "hangover." Obviously this is to be avoided.

What has all this to do with the choice of a speaker enclosure? Well, the speaker and its enclosure must be designed for one another. It is the combination of the two, the speaker *in* its enclosure, which must be tested for resonance, response, etc.

Now, a bass-reflex enclosure is a box with a hole in it, and a box with

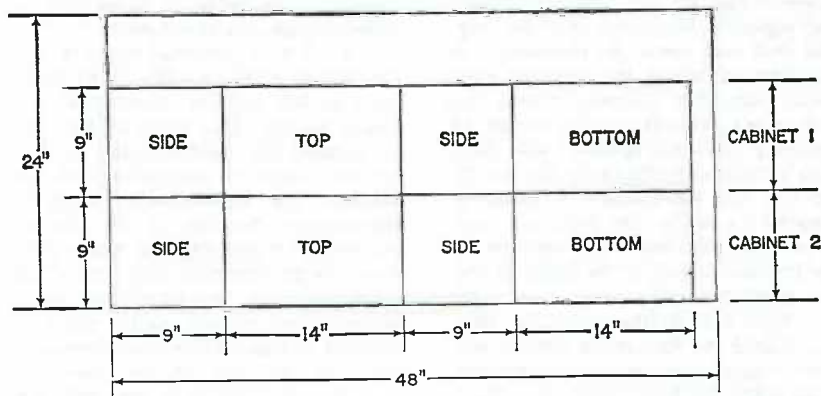


Fig. 1—Cutting diagram shows you how to cut two speakers from a single board.

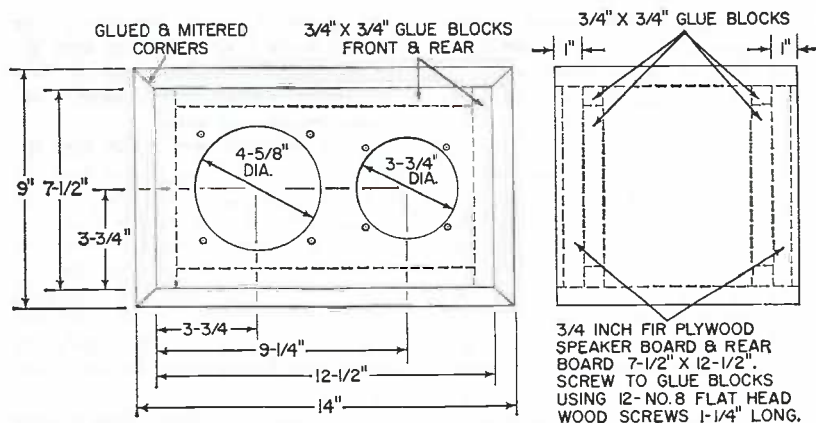


Fig. 2—Use good cabinet assembly techniques when constructing your speaker.

a hole in it is a resonant system itself. It is called a Helmholtz resonator. A soda bottle is a Helmholtz resonator if you blow across its mouth.

The design philosophy behind a bass-reflex enclosure is to have the box resonate at the same frequency as the speaker mounted in it. This can be accomplished by properly controlling the volume of the box and the port size and ducting.

When two resonant systems are coupled together like this an odd thing happens. Instead of getting twice as big a resonance as you might suspect, you get *two* resonant peaks, one higher in frequency than the original and one lower in frequency. A dip appears where the original resonance was. The spread between the two peaks depends on the degree of coupling between speaker and box. This in turn depends upon the size of the box and the amount of damping material in it. In this way, one can extend the low-frequency response of the speaker system to the lower of the two resonance peaks.

This sounds like a terrific idea, and indeed the bass-reflex system was widely used several years ago, and still is in many smaller enclosures. There are certain disadvantages to this sys-

tem: (1) increased distortion, especially near the resonant points, since the system is somewhat uncontrolled near resonance; (2) less than optimum transient response, leading to muddy sound; (3) irregular response in the bass region formed by the two resonances and the trough.

### Why an infinite baffle?

This leaves us the infinite baffle. The infinite-baffle enclosure includes any type which prevents the sound from the rear of the speaker from getting into the room. In its simplest form it is an infinitely large wall in which the speaker is mounted.

In a more practical form, it is a completely sealed box filled with fiberglass or felt mats to absorb the rear sound energy. The walls of the box are strong and rigid enough so they do not vibrate in sympathy with the speaker. The infinite baffle performs the primary function of the speaker enclosure: preventing the rear sound wave from blending with the front wave. Since the two waves are out of phase, they cancel each other if allowed to meet. (The sound emanating from the port of the bass-reflex enclosure is delayed by the enclosure design enough to emerge in phase



with the front wave in the bass reinforcement region. At higher frequencies all rear energy is absorbed by the damping material in the cabinet, and in this region the bass reflex acts as a sort of infinite baffle.)

Early infinite-baffle enclosures were made very large. Since the air trapped in the cabinet acts as a spring behind the speaker cone, system compliance is reduced and the resonant frequency of the speaker system is raised above that of the speaker alone. Since the output of the system drops below the resonant frequency, it appeared advantageous to use a very large cabinet for minimal increase in the resonant frequency.

The acoustic-suspension system is basically an infinite baffle in which the compliance of the trapped air is counted upon to provide some of the speaker suspension. Such a system uses a speaker with a very light (compliant) suspension that has a very low free-air resonance. It is put in a relatively small sealed enclosure. The trapped air decreases system compliance and substantially raises the resonant frequency by as much as an octave.

Thus one can see the necessity for starting with a very-low-resonant-frequency speaker. The acoustic suspension provided by the trapped air has one very distinct advantage: it lowers the harmonic distortion of the system. The main cause of nonlinear distortion in a speaker system is the nonlinearity in the suspension. The typical spider and rim-surround suspensions do not provide a linear restoring force at extreme cone excursions.

The "spring" provided by the trapped air, however, is extremely linear. In a well-designed acoustic-suspension system, the air cushion provides the majority of the restoring force. For example, if the resonant frequency of the system is one octave above (twice as high) the free-air resonance of the speaker alone, three-

quarters of the restoring force is provided by the air cushion and only one-quarter by the speaker suspension. Thus, the effect of the nonlinear suspension, that is the distortion, is reduced three times.

In addition, the air cushion is provided by making the enclosure small instead of large. Further, the response is relatively smooth—assuming good damping to an acceptably low frequency—if the original speaker is of the high-compliance type. We pay for all these advantages with a decrease in overall efficiency.

In choosing a speaker for use in an acoustic-suspension system one must look for a low resonant frequency. Next, you need a good, powerful magnet, which implies good damping and the capability of long speaker excursions. The latter, although they give rise to Doppler distortion, are unfortunately necessary to get a reasonable sound-pressure level from a small cone.

### How to build it

Probably many speakers will fill the bill. I chose to build the system around one available for \$8.95 from Lafayette Radio Electronics (catalogue No. 99E01554). It is a 5-inch unit with a 1½-lb magnet and resonant frequency of 40 Hz. It is rated at 16 watts (peak). For a tweeter I chose another Lafayette unit (catalogue No. 99E01562) at \$2.95.

The enclosure is 14 x 9 x 9 inches and is constructed from ¾-inch plywood. I chose to construct it from single-sided walnut-veneered plywood, available from several firms. Following the cutting diagram of Fig. 1, two speaker enclosures can be made from one 24 x 48-inch sheet of plywood.

Other veneers are also available. If you like oiled walnut, two or three coats of boiled linseed oil rubbed in with cheesecloth and the excess removed will give you a better finish than you can expect on commercial

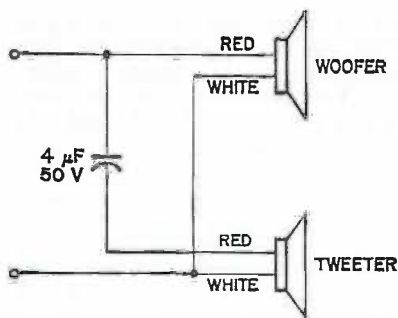


Fig. 3—Wiring diagram for speakers. Do not use tantalum electrolytic capacitors.

cabinets. Let each coat dry overnight and rub lightly with the grain using *fine* steel wool. The recessed front and back of the cabinet will not show and can be made from  $\frac{3}{4}$ -inch fir plywood available from any lumber yard. A two-conductor barrier strip is mounted on the recessed rear wall.

Fig. 2 shows how the cabinet is put together. Be sure to make it airtight. Seal all cracks with glue, filler or caulking. The inside of the cabinet should be loosely filled with Tufflex or a similar sound-absorbing material. Such materials are readily available from most parts dealers.

The wiring diagram is shown in Fig. 3. A 4- $\mu$ F, 50-or 100-volt capacitor is used as a highpass filter to the tweeter. This capacitor should be a paper or mylar type. Aluminum or other electrolytics should *not* be used.

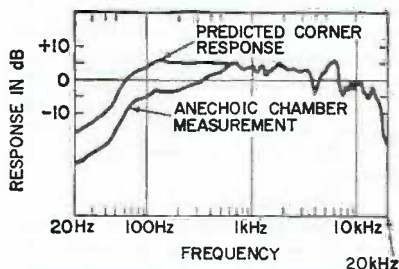


Fig. 4—Anechoic chamber curve is smoothed by placing unit in corner.

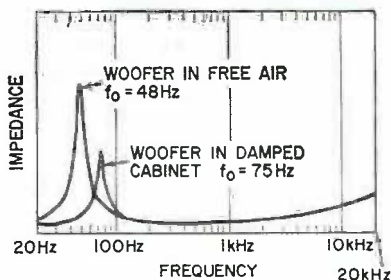


Fig. 5—Impedance curves for the woofer in free air and damped in the cabinet.

When wiring the speakers be sure to observe polarity. If you use the speakers recommended, you will find one terminal on each insulated with a red fiber washer and the other with a white one. Connect the whites together as the common return and the reds with the capacitor.

More exotic crossover networks could be used with this speaker system, but they are really unnecessary. The woofer used in this system handles itself so gracefully at the higher frequencies that there is no real reason to prevent it from reaching them.

When you're through, connect them to a good amplifier with at least 5 watts per channel capability into 8 ohms and listen. As is true for all speaker systems, bass response is affected by placement of the units in the listening room. As much as 9 dB boost in the low end can be achieved from corner placement.

Fig. 4 shows the response of the system, measured in an anechoic chamber, and the response expected in a corner placement, the recommended location for this system. Placed in a corner, a speaker sees a much reduced solid angle of radiation at low frequencies. Thus, its efficiency is greatly increased. This is especially important for small-size systems. You can experiment with various placements in your home. You will probably be quite surprised at the difference positions can make.

Impedance curves of the woofer in free air and in the damped cabinet are in Fig. 5. You can see that the resonant frequency rose from 48 Hz to 75 Hz, almost an octave. In addition, the height of the peak is substantially reduced. As explained, the acoustic response of a speaker system falls off at about 12 dB per octave below the resonant frequency. From Fig. 5, you can see the resonant frequency is at 75 Hz, and, sure enough, from Fig. 4, the response starts to fall off at just about that point.

The results of this little system will amaze you. The sound is sur-

prisingly clean and live. The instruments of the orchestra are well defined and project into the room. This is attributable to the smooth mid-range. Except for a dip and peak in the response at 4 kHz and 6 kHz, the response throughout the critical mid-range is extremely smooth and well balanced compared with *any* system. Couple this with adequate response, again without any sharp variations, out to the limit of audibility (16 kHz), and solid bass response down to 60 Hz or so, and you have a system to put many of its larger and more costly brothers to shame.

## 3-WAY ELECTRONIC CROSSOVER

by NORMAN H. CROWHURST

The basic amplifier circuit around which a 12-dB/octave filter can be built is in Fig. 1. It consists of a central voltage-gain stage and two emitter followers, to provide impedance isolation so performance is not affected by external circuits and their impedances.

The output emitter follower makes the output signal voltage developed by the voltage-gain stage available with a low source impedance, and the base circuit of the input emitter follower is fed by a 2:1 voltage divider. There is an additional 2:1 gain loss due to the 6 dB feedback used to sharpen the response to its correct shape.

So the voltage gain of the middle stage should be precisely 4 (12 dB). This will make the output signal voltage the same as the input, from each channel, within that channel's range.

Using a 12-volt supply and transistors with a current gain of about 100 (this is not critical in the circuit chosen) we calculate values. For the

output emitter follower, we want the emitter dc voltage to be 6 volts. This allows maximum swing with the 12-volt supply.

Picking 510 ohms as the emitter resistor (assuming a 500-ohm load as minimum external output termination), the dc load at the base will be about 100 times this, or 51K. Using a bias potentiometer of 5.1K and 5.6K will bring the emitter voltage very close to 6-volts on a 12-volt supply.

Now to calculate the base input resistance of the output emitter follower: This consists of 5.1K and 5.6K, along with the reflected impedance of about 51K (with no external load connected) all in parallel. This combination figures to 2.5K. An external load of 500 ohms would reduce the reflected part to about 25K, reducing the combined input impedance to 2.25K, which is not a serious change.

To achieve an in-range gain of 4:1, we assume a collector resistor of 1K and calculate the collector load,

which is this 1K paralleled for signal purposes with 2.5K, to make 720 ohms. With this collector load, an emitter resistor of 180 ohms will control the stage voltage gain to 4:1.

To get 6-volts on the collector of this stage, its current must be 6 mA, so the emitter voltage will be 1.08-volts. The 180 ohms will reflect to the base circuit as 18K.

The next step was based on using identical capacitors for controlling turnover in each stage of coupling. For the high-pass filter, the reactance of the coupling capacitor between collector and output emitter follower needs to be 3.5K (2.5K plus 1K) at root-2 times turnover frequency.

The input emitter follower will provide negligible source impedance, so the base input impedance of the voltage-gain stage should be 3.5K also.

Trial and error led to the choice of 5.1K and 43K, along with the 18K reflected through the stage from the 180-ohm emitter resistor, which comes very close to the 3.5K desired.

This combination of resistors also controls the emitter voltage to 1.08 volts, and thus collector current to 6 mA and the collector voltage to 6 volts.

The input emitter follower also uses a 510-ohm emitter resistor. Its bias will normally be taken through the feedback resistor. Its input impedance, reflected from the emitter resistor, will be about 51K. Using two 3.6K resistors across the input signal voltage will divide the input voltage in half (before feedback is considered).

Feedback will reduce this signal voltage to one-fourth input and the 4:1 gain will bring the output voltage up to equal the original input signal voltage, as well as reversing its phase.

The source resistance for the input signal voltage of  $\frac{1}{2}$ , at the base, is the parallel combination of two 3.6K resistors, or 1.8K. The signal voltage will be dropped, when feedback is applied, to  $\frac{1}{4}$ , while the feedback resistor will connect back to a signal voltage of 1, reversed phase. (Fig. 2).

So the feedback resistor will have 1.25 times the original input voltage

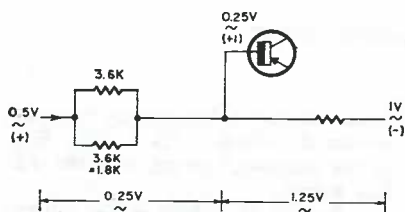
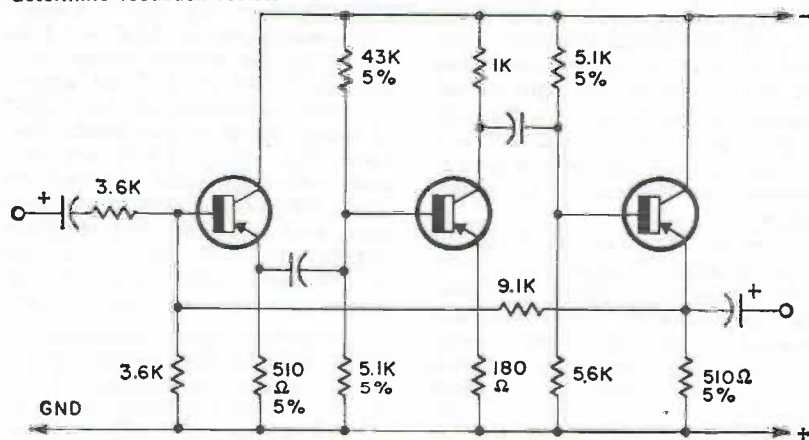


Fig. 1—Basic 12-dB/octave filter (below) uses two emitter followers. Fig. 2 (above)—Voltage relationships that determine feedback resistor value.



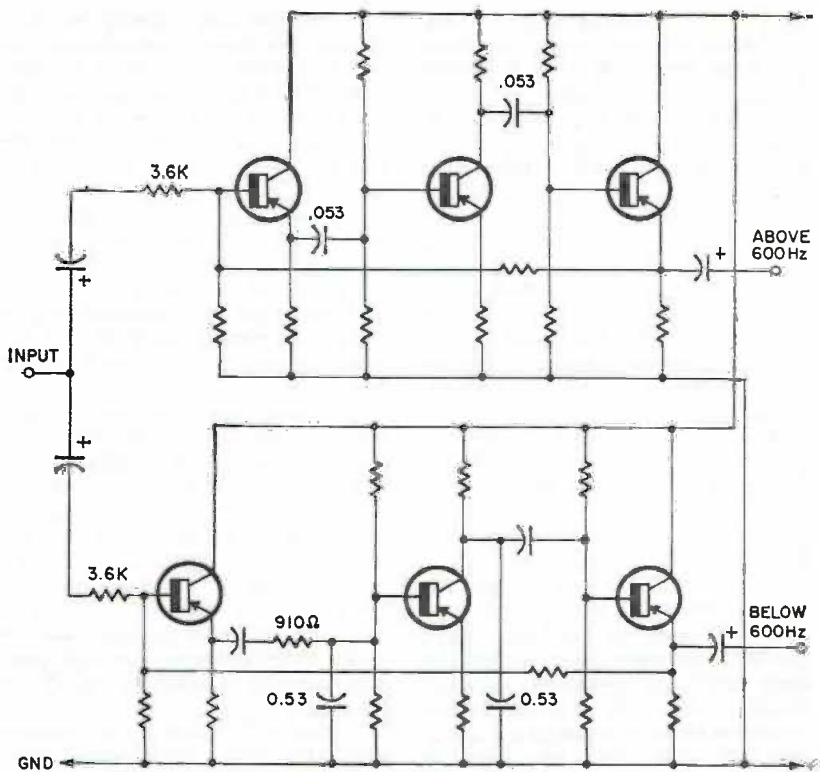


Fig. 3—High- and low-pass circuits alone make a useful 2-way crossover.

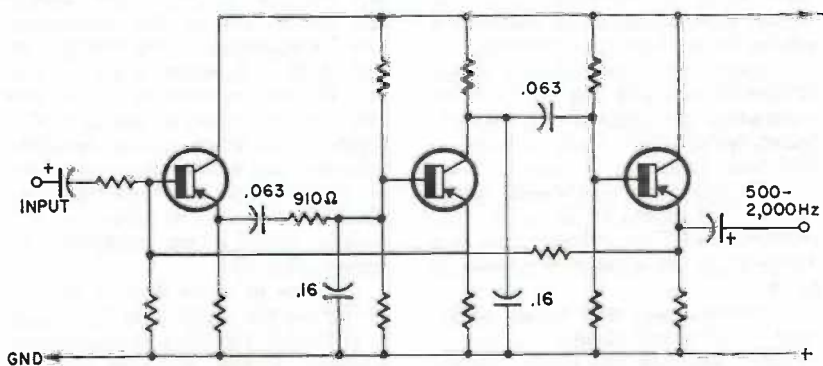


Fig. 4— Band pass circuit added for a 3-way crossover. See table for values.



across it, thus needing to be 5 times the impedance that "drops" the 0.25 part. Five times 1.8K is 9K, for which a 9.1K resistor will serve.

Using a blocking capacitor to keep dc out of the input circuit, the bias of the input emitter follower is taken from a 6-volt point (the output emitter) through 9.1K and 3.6K, yielding a working voltage of about 1.7 volts. As the signal level at this point is  $\frac{1}{4}$  the input and output signal level, this voltage provides adequate margin for handling the signal.

The impedance reflected through the input emitter follower will be 1.8K divided by 100, or 18 ohms. And through the output emitter follower, 720 ohms divided by 100, or 7.2 ohms. Thus output loading can have little effect on operation.

### Low-pass design

That about sets the picture for the high-pass configuration. For low-pass, we have the collector-circuit parallel impedance calculated at 720 ohms. This is the impedance the shunt capacitor works with. If we use a series resistor in the base circuit to make that also 720 ohms, the same value of shunt capacitor will serve. With the 3.5K in the base circuit, a 910-ohm resistor will bring the shunt combination at the base to 720 ohms. This resistor needs dc blocking, unless the section is bandpass, so the capacitor is needed for the high-pass function.

Thus the completed two-way crossover looks like Fig. 3. To make calculating the capacitors for turnover easier, we will figure them at whatever crossover frequency is chosen.

For high pass, they should be figured at root-2 times crossover, so their reactance will be root-2 times the series-circuit impedance at crossover, or 5K.

For low pass, they should be figured at crossover divided by root-2, so their reactance will be the parallel circuit impedance divided by root-2 at crossover, or 500 ohms,

This makes the figuring very convenient. We have tabulated some values of capacitors for selected crossover frequencies. If you want to make your crossover three-way or more, it is simple to combine the two functions into one circuit for any bandpass channels you use (Fig. 4).

If you plan on making several of these, in varying combinations, it may be worth planning an etched circuit (Fig. 5) that can be wired for low pass, high pass or bandpass. Fig. 6 shows a completed circuit for a three-way crossover, using these circuit boards.

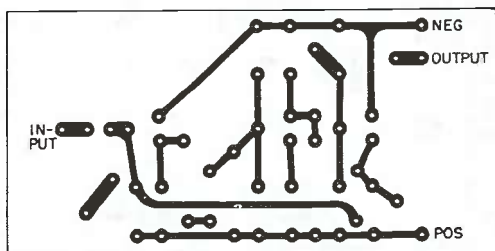
Finally, you may need to know how to check the performance. The first step is to open the feedback loop. This can most simply be done by lifting the 9.1K resistor from the output emitter and connecting a 12K resistor in series with it to the negative supply. Now the unit should show precisely 2:1 gain, and 90° phase shift with zero gain at frequencies that are root-2 times crossover for high pass, and crossover divided by root-2 for low pass.

This 90° phase shift pattern should look like a circle on the scope when horizontal is connected to input and vertical to output with both inputs set to the same sensitivity. This can be set by paralleling them on the input and adjusting to give a 45° straight line. Check this at the appropriate root-2 frequencies. If the result is incorrect, find out where it goes wrong.

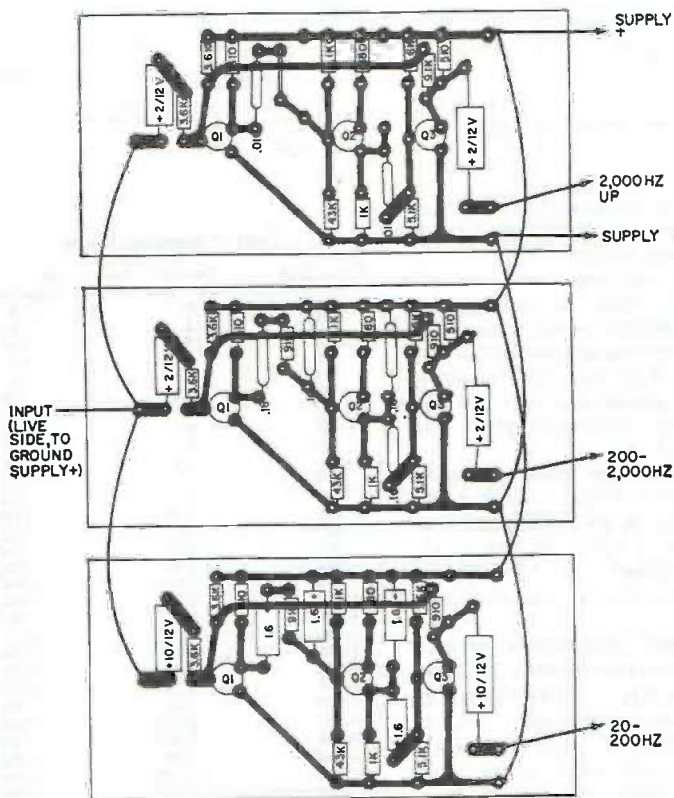
To find an error in a high-pass filter, or the high-pass function of a bandpass, use an electrolytic capacitor to bypass one of the turnover capacitors in turn; first one, then the other. Each of these should yield the 45° trace at root-2 times crossover frequency (Fig. 7).

If one of them doesn't, the capacitor has the wrong value and should be corrected. Finally, reconnecting the feedback should give the elliptical trace at crossover, instead of the circle at root-2 times frequency.

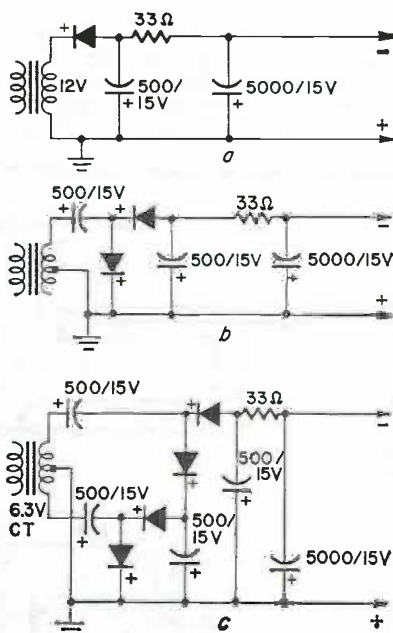
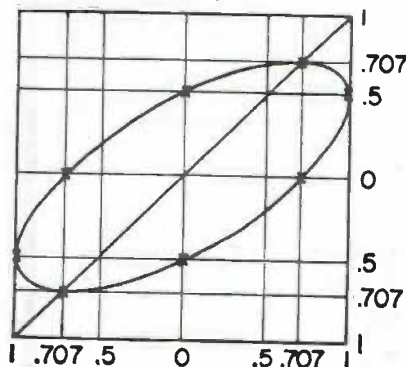
tions should yield the 45° trace at crossover divided by root-2. Again, if one of them doesn't, the capacitor remaining in circuit is off value and should be corrected.



**Fig. 5—Copper side of circuit pattern used for crossovers. Actual size of the PC board used is 3 x 6 inches. Draw or enlarge optically. Fig. 6 below shows the completed and wired 3-way crossover. Use table for crossover values.**



**Fig. 7—Scope pattern of the 45° phase-shift frequency, identifying 8 points on the ellipse. Here the 45° line is set at a mid-band frequency, and the ellipse should appear at the root-2 frequency. If the 45° line is set by paralleling vertical and horizontal, the ellipse will be twice as high for the same width. Fig. 8—Various power supply circuits: a is a 12V winding, one side grounded, b is 12V winding with center tap grounded (or 6V, one side grounded) and c is 6V with the center tap grounded.**



A little practice with this will find it quite simple to do, and the circuits obey the rules nicely.

Almost any pnp transistors will serve in this circuit. If you want to use npn transistors, merely reverse the supply polarity and the polarity of any electrolytics. With pnp, positive supply is also signal ground for input and output. With npn, negative supply is also signal ground.

The lowest low pass can incorporate a rumble filter, if desired, "at no extra cost," by using identical electrolytics with a value to give a 12-dB/octave rolloff at the desired frequency, say 20 Hz (as shown in Fig. 6).

The supply requirement for each unit of the crossover is about 21.5 mA. Thus a two-way crossover requires 43 mA; a three-way, 65 mA. This can easily be obtained from a filament transformer with small diode rectifier(s) and really large electrolytics for final smoothing (Fig. 8). The

**Table of Capacitor Values**

Crossover Frequency Hz	Capacitor Values (μF) Low Pass	High Pass
20	16	1.6
25	13	1.3
30	10.5	1.05
40	8	0.8
50	6.3	0.63
60	5.3	0.53
80	4	0.4
100	3.2	0.32
120	2.7	0.27
150	2.1	0.21
200	1.6	0.16
250	1.3	0.13
300	1.05	0.105
400	0.8	0.08
500	0.63	0.063
600	0.53	0.053
800	0.4	0.04
1,000	0.32	0.032
1,200	0.27	0.027
1,500	0.21	0.021
2,000	0.16	0.016
2,500	0.13	0.013
3,000	0.105	0.0105
4,000	0.08	0.008
5,000	0.063	0.0063
6,000	0.053	0.0053
8,000	0.04	0.004
10,000	0.032	0.0032
12,000	0.027	0.0027
15,000	0.021	0.0021
20,000	0.016	0.0016

configuration used depends on the filament supply. You can probably use something from the shelf for this, be-

cause nothing is at all critical, except that it needs adequate filtering to avoid hum.

## STEREO HEADPHONE CONTROL CENTER

By PETER E. SUTHEIM



STEREO HEADPHONES are great—they provide the ultimate in separation and you aren't distracted by room noise. Only trouble is they usually don't match your amplifier output. This stereo headphone control center takes care of the matching; moreover, it gives you full control of volume, bass, blend and balance. And you don't have to disconnect it to use the speakers.

The control center can be used with any stereo headset and any stereo amplifier—unless you have a very unusual headset or an amplifier.

### Controls

For controls, it has a three-position selector switch: You can choose speakers only, phones only, or both at the same time. (The third choice may seem a bit strange, but it is much appreciated where one listener is hard of hearing. He can wear the phones and enjoy the music at a suitable volume level. The others can listen at a com-

fortable volume through the speakers.) The switch is of the shorting (make-before-break) type to reduce switching clicks.

There is, of course, a volume control for the phones, ganged so that both channels are controlled simultaneously. A balance control compensates for uneven tracking between the two sections of the volume control, and for individual differences in hearing sensitivity between the listener's ears.

The blend control corrects for the sometimes excessive stereo separation in headphone listening. At its extreme counterclockwise position, the blend-control resistance is switched out of the circuit, allowing full separation. At the other extreme, the two channels are connected together, producing mono sound.

The bass control provides up to approximately 6 dB of boost at 40 Hz (referred to 1 kHz) to compensate for headsets whose air seal against the sides of the head is not good.

### Connections

Leaving aside the convenience this box provides—which may or may not interest you—it might seem that there's a simpler way of connecting phones to an amplifier. The most obvious would be simply to connect the phones across the speaker terminals with a switch in series to turn the speakers on or off. This works, but has a couple of very serious disadvantages. First, earphones require only between

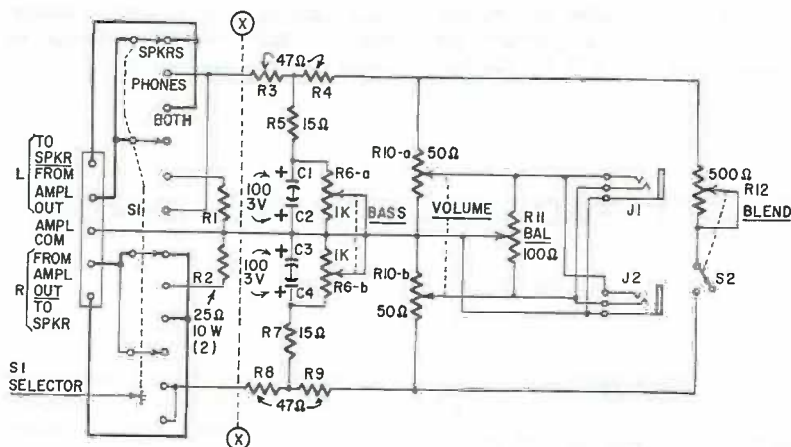


Fig. 1—Unique switching arrangement makes it possible to switch in speaker or phones only, or both. Amplifier is loaded, either by the speakers or R1 and R2.

#### PARTS LIST

C1, C2, C3, C4—100- $\mu$ F, 3-volt electrolytic capacitor  
 J1, J2—Three-contact phone jack  
 R1, R2—25-ohm, 10 watt wirewound resistor  
 R3, R4, R8, R9—47-ohm,  $\frac{1}{2}$ -watt resistor  
 R5, R7—15-ohm,  $\frac{1}{2}$ -watt resistor  
 R6—Dual 1000-ohm potentiometer, log taper (Centralab, F5-1000 & R5-1000)

R10—Dual 50-ohm wirewound potentiometer  
 R11—100-ohm wirewound potentiometer  
 R12—500-ohm potentiometer (with S2)  
 S1—4 pole, 3-position shorting-type (make-before-break) rotary switch  
 S2—S.p.s.t. switch (on R12)  
 Misc.—Plastic case with aluminum panel (Lafayette 99 H 6272 or similar); barrier-type terminal strip (5 terminals)

10 and 100 milliwatts for quite loud volume levels. This is in the neighborhood of one-thousandth of the power output capability of typical amplifiers.

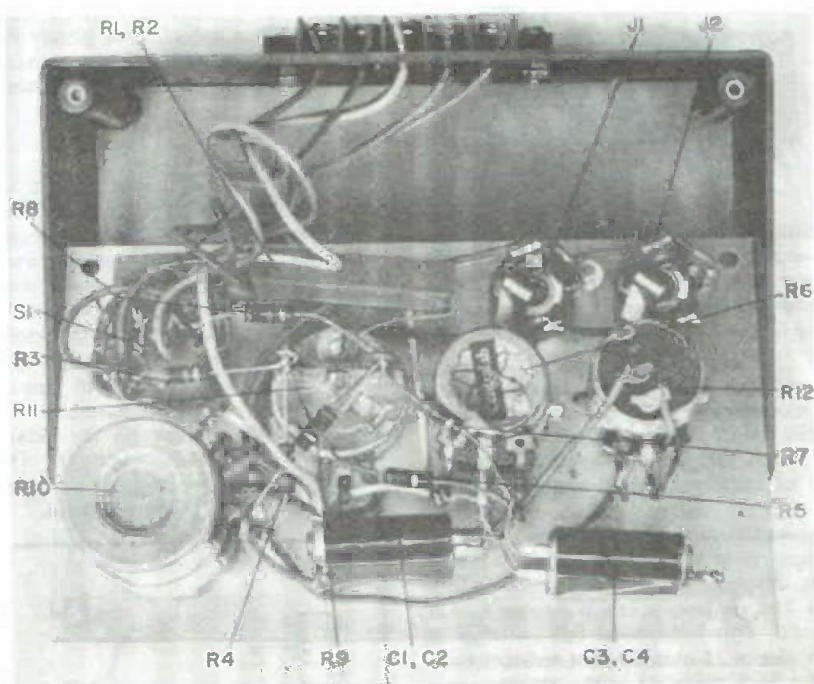
Not only is the full amplifier power unnecessary for phones, but it can destroy them in a fraction of a second by burning out the voice coils or rupturing the diaphragms. As a result, the volume control on the amplifier can be only just barely cracked open. There has to be some way of cutting the power fed to the phones.

Furthermore, the high sensitivity of the phones results in a good bit of noise along with the music. The normal amplifier hiss and hum, usually inaudible a few inches from a speaker system, become definitely audible in high-sensitivity headphones pressed close to your ears.

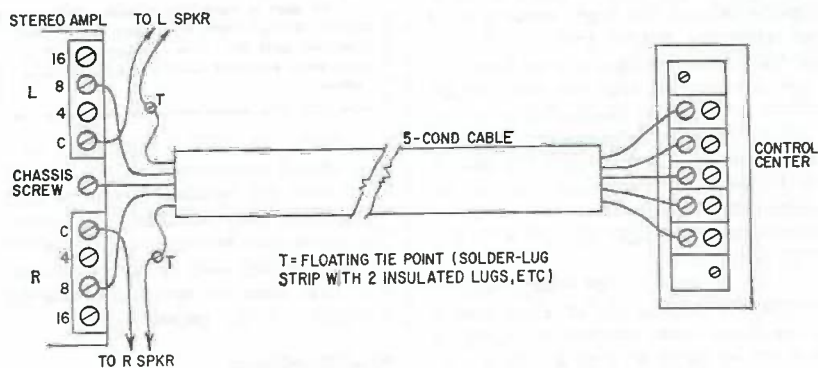
The usual way of solving both problems is simple and quite satisfactory: Stick a resistance in series with each "hot" earphone wire. The value most commonly used is around 300 ohms. It cuts down the power to the phones and, of course, cuts back the noise at the same time. It also reduces the damping factor to nearly zero. But, this seems to have little audible effect on phones, which have very small, low-mass cones or diaphragms with little inertia. They are usually pretty well damped by internal absorbents and by close coupling to the ear chamber.

But a load in the vicinity of 300 ohms is almost an open circuit as far as the amplifier output is concerned. It becomes necessary then to provide a dummy load for the amplifier when the speakers are switched out. The value of

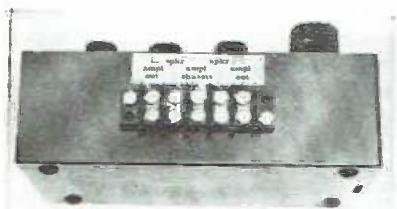




*The limiting factor in parts mounting is size of the controls on the front panel. Wire position is not important; the circuit is all low impedance and high level.*



*Fig. 2—To avoid undesirable power loss in the speaker wiring, don't use a cable longer than approximately 25 feet. Use impedance taps to match your speakers.*



*Shorted output circuits in a solid-state amplifier can be destructive—use a barrier type terminal strip. Note: speaker and amplifier output terminals are shown in reversed order—follow the schematic.*

the dummy load is not critical. It isn't necessary, or even desirable, to match the dummy to the amplifier output.

A considerable upward mismatch (for instance, 8-ohm output loaded by 25-ohm resistor) means much less power wasted in the dummy load as heat, since the amplifier will not usually develop nearly as much power into a 25-ohm load as it will into an 8-ohm load. This headphone box switches in a pair of 25-ohm 10-watt resistors (one for each channel) in the PHONES position of the selector only.

It might be worth while to digress for a moment to explain this matter of protecting amplifiers with a dummy load. In general, the dummy is most important with tube amplifiers, principally because the high inductance of an unloaded output transformer can develop peak voltages during loud signals or instability high enough to break down transformer insulation or cause arcs in tubes or tube sockets. Almost any value of load resistance less than 5 or 10 times the nominal output impedance of the amplifier will load the transformer enough to prevent this type of trouble.

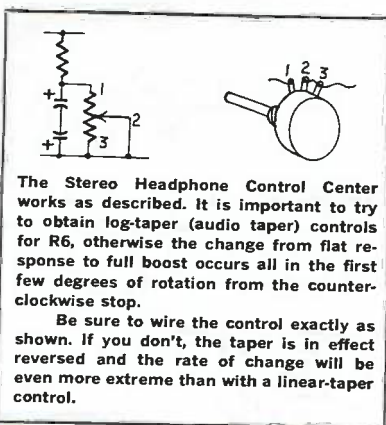
In transistor amplifiers, shorts across the output, rather than opens, most often cause trouble. However, it can do no harm to load the output of even a transistor amplifier in the same way. Dummy-loading may even save the output transistors if the amplifier

happens to be unstable with no load. Occasionally a defective or poorly designed amplifier will oscillate with no load and damage its output transistors.

## Construction

Few things could be simpler or less critical to wire than this switchbox. The low impedances and relatively high signal levels make it unnecessary to observe any precautions about wire length, shielding, routing, and so forth. Be sure, though, to use wire no thinner than No. 22 for the leads that will carry the full speaker current (heavy lines drawn on the schematic).

The photos show all the details for wiring the control center. The solder-lug strip mounting is soldered to the back of the balance control.



If the electrolytic capacitors have uninsulated metal cases, as the ones in this model did, insulate them with tape to prevent their touching the panel. The panel must be electrically common to the common side of the circuit unless you want to provide insulating bushings for the phone jacks.

## Modifications

Of course you can provide only one jack if you wish, or three or more.

Volume will diminish as you parallel more sets of phones. If you like, you can provide a separate volume control for each headset. You can, in fact, duplicate the portion of the circuit after the switching (to the right of the dashed line X-X on the schematic) so that each headset has full, separate control facilities.

If you like your headphone music extremely loud, you may have to forgo the bass-boost circuit (R5, R6 and R7 and C1, C2, C3 and C4). To provide 6 dB of bass boost, the circuit must introduce 6 dB of loss everywhere else in the spectrum, except at the low frequencies. In a passive equalizer, there is no way around this fact of life. So if you need loudness, disconnect or omit the bass boost circuit.

### Connecting the thing

Fig. 2 shows how the control center is to be connected to an amplifier. Note that the common terminal on the control center goes to the amplifier *chassis*, not to either of the speaker terminals often labeled "common" or

"C". The reason is that in many amplifiers the so-called "common" terminals are *not* common to each other or to the chassis, and cannot be connected together without disrupting some circuit function.

This connection is for headphone operation only. The normal connection to the speakers is in no way affected by the addition of the box if you follow the scheme given here. The speakers can be turned off or on by the selector switch in the control center, but this is done by interrupting the *high* (4-, 8- or 16-ohm) side of the wiring, not by disturbing the connection to the amplifier's low side.

Flat, five-conductor antenna-rotator cable is handy for making the connections between control center and amplifier because it can be slipped under a rug or run alongside of base-board molding.

The wire should be No. 22 or heavier. Avoid running more than about 50 feet to prevent a sizable portion of your expensive amplifier power to be dissipated in the wiring.

## FM STEREO ADAPTER

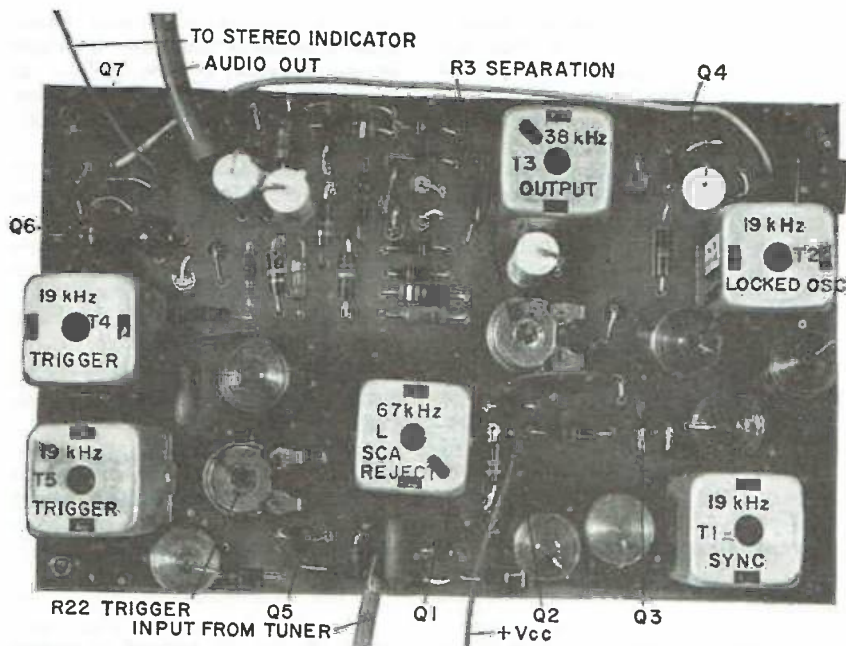
By KENNETH F. BUEGEL

THIS STEREO FM DEMODULATOR IS DESIGNED around recent silicon planar transistors and uses high-stability polystyrene capacitors. It is compatible with any detector output level between 0.3 and 5 volts peak to peak. Separation on a properly constructed unit will exceed 30 dB from 50 Hz to 14 KHz, and 40 dB from 100 Hz to 10 KHz.

Many stereo listeners are puzzled when a good stereo adapter added to a previously satisfactory tuner fails to produce adequate separation. Often the

trouble lies in unexpected areas. When you've been intelligent and careful in connecting the adapter to the tuner, and the antenna installation isn't faulty, what else could be wrong?

In almost every instance I have investigated, the problem was caused by restricted tuner bandwidth, which causes a distorted response in the 23-53-kHz subcarrier range. Many otherwise fine tuners may have this defect. It is possible to compensate for it.



This decoder incorporates just that kind of compensation. The schematic is in Fig. 1. Input stage Q1 has adjustable gain to match different tuner output levels. Because of heavy feedback the input impedance is high enough to allow direct connection to a vacuum-tube tuner without heavy loading. The value of R2 is selected during alignment.

L is the SCA (Subsidiary Communications Authorization) rejection filter, which, with the well-balanced output detectors, keeps SCA program matter inaudible. Emitter follower Q2 sends composite audio at low impedance to the center tap of T3, a 38-kHz doubler in the collector of Q4. The pilot signal developed across T1 is amplified by Q3 and synchronizes the 19-kHz locked oscillator, Q4.

Q5 selects and amplifies the 19-kHz pilot signal. R22 is the trigger adjustment, which sets the level of pilot signal required to switch the adapter to stereo reception. D5 and D6 charge C24 with rectified pilot signal until Q6, a dc amplifier, is forward-biased.

Q6 drives Q7 into saturation, which lights lamp PL and brings the lower end of T2 to ground potential. Under these conditions Q4 oscillates and T3 provides the switching signals to the output detectors. Diode pairs D1-D3 and D2-D4 will conduct alternately at the 38-kHz switching rate and channel left and right components of the signal applied to T3 centertap to correct outputs.

During monaural reception, when no pilot signal is present, Q7 will be cut off and its collector at supply potential. A slight reverse bias will prevent oscillation in Q4. There is now a path through D1, D2, R13, and R14; thus the monaural signal applied to the center tap of T3 appears at output A. A similar path exists for the same signal to appear at output B. The de-emphasis networks precede output jacks.

The decoder circuits draw about 8 mA from a 14-volt supply. PL's current is 20 mA from a slightly higher potential. If it is not convenient to draw this much current from an available source,

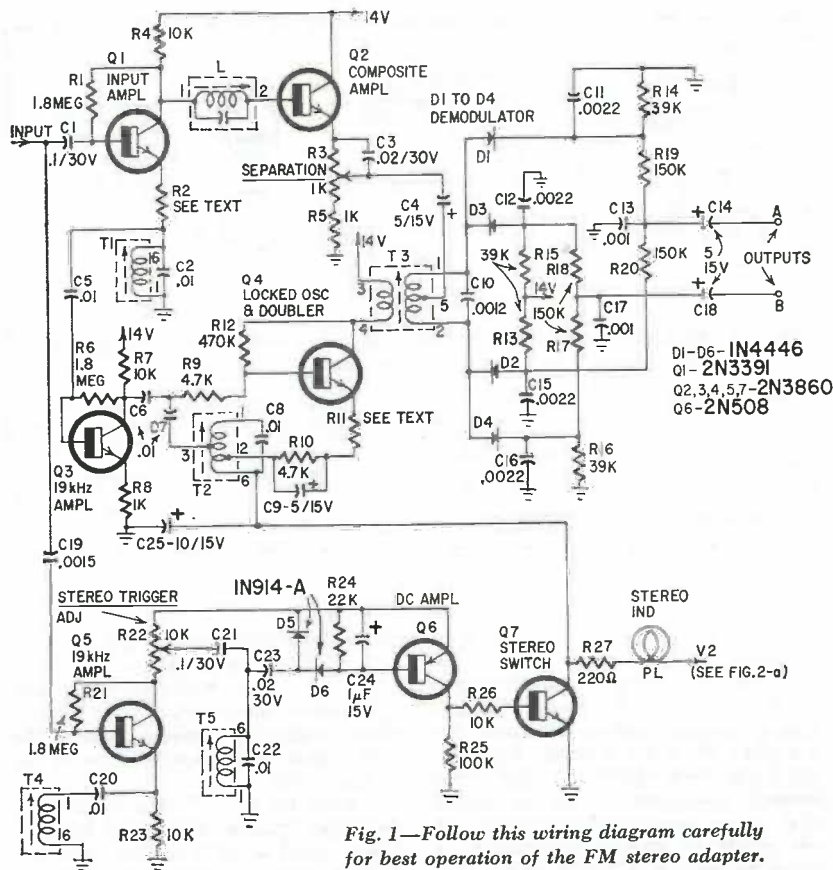


Fig. 1—Follow this wiring diagram carefully for best operation of the FM stereo adapter.

#### PARTS LIST

C1, C21—.01  $\mu$ F, 30V, ceramic  
C2, C5, C6, C7, C8, C20, C22—.01  $\mu$ F, polystyrene  
C3, C23—.02  $\mu$ F, 30V, ceramic  
C4, C9, C14, C18—5  $\mu$ F, 15 V, electrolytic  
C11, C12, C15, C16—.002  $\mu$ F (Sprague 192P or equivalent)  
C13, C17—.001  $\mu$ F (Sprague 192P or equivalent)  
C10—.0012  $\mu$ F, polystyrene  
C19—.0015,  $\mu$ F, disc ceramic

C24—1  $\mu$ F, 15V, electrolytic  
C25—10  $\mu$ F, 15V, electrolytic

Capacitors specified as polystyrene are Mallory type SX, listed on p. 293 of Allied's catalog 260 (1967 general catalog)

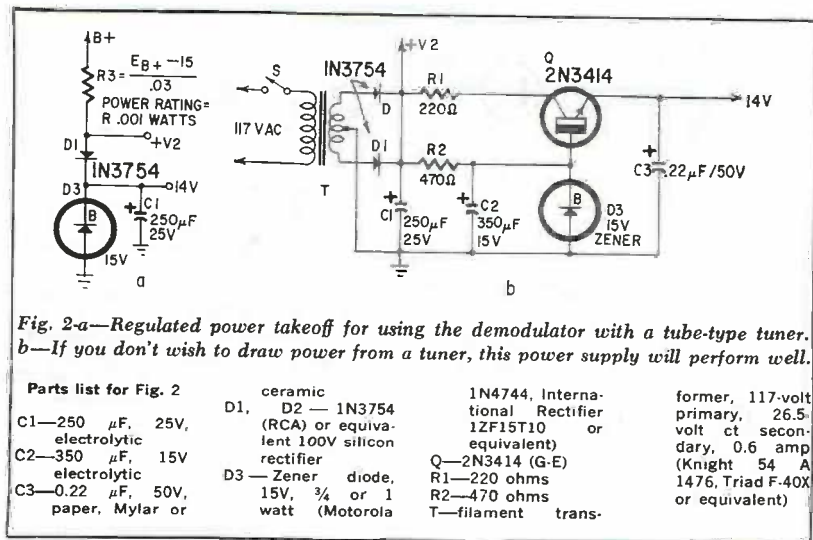
Ceramic capacitors can be Sprague's low-voltage type, although higher-voltage capacitors are of course equally usable.

D1-D6—IN4446 (G-E) (IN914A is also usable)  
Q1—2N3391 (G-E)

Q2, Q3, Q4, Q5, Q7—2N3860 (G-E)  
Q6—2N508 (G-E)  
PL—16-volt indicator lamp (15–20 mA) (Sylvania 16CSB—p. 316 of Allied catalog 260)  
R1, R6, R21—1.8 megohm  
R2, R11—see text  
R3—trimmer resistor, 1,000 ohms (Mallory type MTC-4)  
R4, R7, R23, R26—10K  
R5, R8—1,000 ohms  
R9, R10—4,700 ohms  
R12—470K  
R13, R14, R15, R16—39K, 5%  
R17, R18, R19, R20—150K, 5%

R22—trimmer resistor, 10K (Mallory type MTC-4)  
R24—22K  
R25—100K  
R27—220 ohms  
All resistors  $\frac{1}{2}$  watt, 10% unless otherwise specified  
T1, T2, T4, T5—19-kHz tank (J. W. Miller 1354-PC)  
T3—38-kHz output transformer (J. W. Miller 1355-PC)  
L—series bandpass filter (J. W. Miller 1352-PC)  
For construction on chassis instead of etched circuit board, order Miller units without PC suffix





the regulated supply shown in Fig. 2 will power the decoder.

When constructed on an etched circuit board, the complete decoder, including required bottom clearance, fits in a space 3 3/4 x 6 x 2 inches. Since the unit is extremely stable and won't need constant adjustment, it can be tucked away in an unused cabinet corner. If you decide on chassis construction, a recommended size is 7 by 9 inches. This provides room for the power supply, terminal strips, transistor sockets, etc. If you plan to build on a chassis instead of on a circuit board, be certain to order the transformers without the PC (printed circuit) suffix.

In mounting parts to the wiring board, first install the transformers, then R3, R22, all other resistors and capacitors (except R2 and R11), and finally the transistors and polystyrene capacitors.

The first step in alignment is to measure the multiplex output level of the tuner with an oscilloscope. Use the highest peak-to-peak reading as a reference. Use an input level, from an audio generator to the decoder, of 25% higher

than this reference level. (Use any convenient frequency in the audio range.) Choose a value for R2 between 3.9K and 12K (larger input, larger value) which allows undistorted reproduction of the input signal as seen on a scope at the emitter of Q2.

Next insert a 67-kHz signal (this should be accurate) and tune L for *minimum* signal at Q2's emitter. Insert a 19-kHz signal at one-fourth reference level and tune T1 for *maximum* voltage across its winding. Reduce the input signal and tune T4 and T5, with R22 at maximum, for maximum voltage across R24. During tuning reduce the input level until the voltage across R24 starts

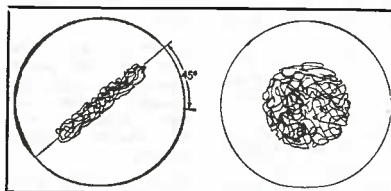


Fig. 3-a—Poor stereo separation; b—good separation; scope at adapter output.

to decrease; at this point the tuning effect is most pronounced.

Tack in a 470-ohm resistor for R11 and temporarily ground the collector of Q7. Tune T3 for maximum amplitude of the 38-kHz waveform at Q4's collector. Then tune T2 until any jitter in the waveform disappears, indicating synchronization with the input signal. Continue to retune T1, T2, and T3, while reducing the input level, until the oscillator is synchronized with a 19-kHz input signal at least 26 dB (one-twentieth) below reference.

Select a value for R11 between 200 and 1,000 ohms that provides maximum undistorted 38-kHz output. Too low a value will give lower-amplitude switching voltages and reduced separation. Adjacent cycles of the switching waveform will not have identical height (this is normal) but should be within 20%.

If you have access to a stereo multiplex generator, remove the ground at Q7's collector and inject a composite

signal at reference level. Set R22 to place Q7 in saturation and adjust R3 for maximum separation.

If such a generator is not available, you can run through the procedure just described with the decoder connected to a tuner receiving a stereo transmission. Connect a potentiometer (about 100K) to the tuner output to allow the input to the decoder to be reduced during alignment.

After preliminary adjustment remove the potentiometer from the circuit. Connect outputs A and B to the vertical and horizontal inputs of a scope. Tune to a station known to be transmitting a stereophonic broadcast. Adjust R22 until PL lights. The scope pattern will probably resemble Fig. 3-a. Adjust R3 until the scope display looks like Fig. 3-b. A slight touchup of T1 and resetting of R3 will result in best separation. If the separation seems poor, try another stereo station or broadcast.

## 200-WATT IC STEREO AMPLIFIER

ADVANCES IN HIGH-POWER THICK-film technology combined with a unique circuit have resulted in a hybrid IC power amplifier with universal applicability. The amplifier was designed for the industrial and commercial market where a high-power, audio-range, reliable and compact linear amplifier was needed.

Although not fabricated specifically for high-fidelity applications, the performance characteristics of the TA7625 hybrid power module (see Table I and Graphs I-III) suggest it is quite adequate for this purpose. The photo showing an encapsulated and unencapsulated module mounted

TABLE I

Data taken with split supply at  $\pm 37.5$  volts, 4-ohm load, 1-kHz signal frequency and feedback resistor = 820 ohms.

Zero signal dissipation = 1.725 watts\*

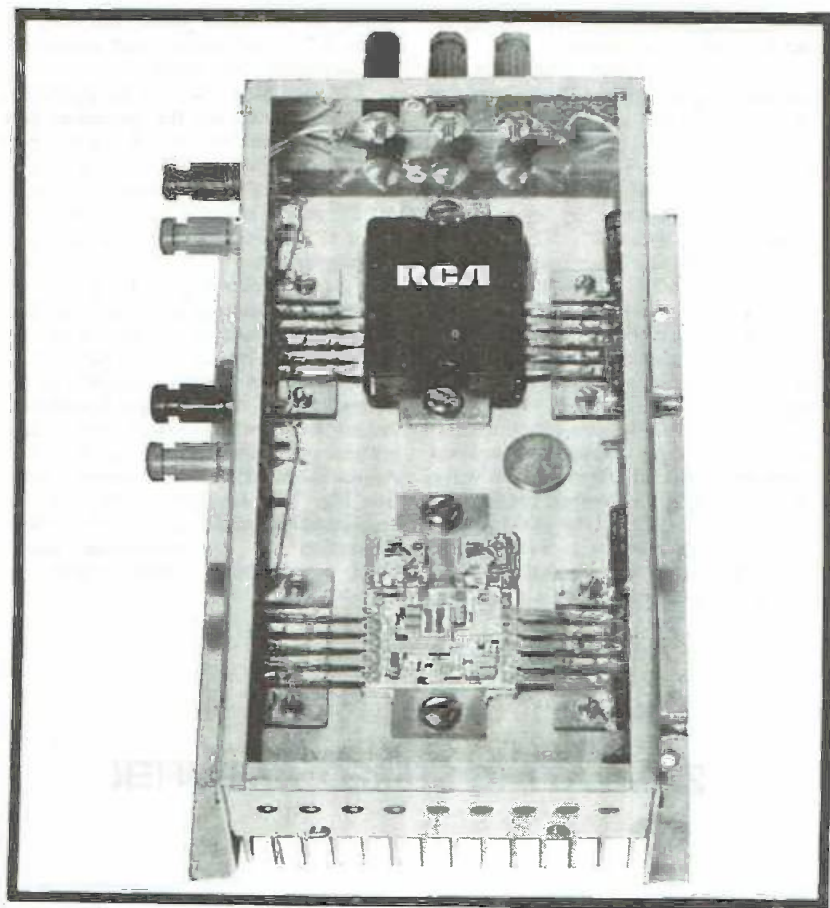
Input voltage = 1.00 @ 100 watts output

THD = 0.38% @ 100 watts output

Input voltage = 0.62 @ 35 watts output

THD = 0.14% @ 35 watts output

\* Of this, 585 mW is dissipation in resistor's  $R_1$ ,  $R_2$  and  $R_3$ , leaving a device dissipation of 1.14 watts.

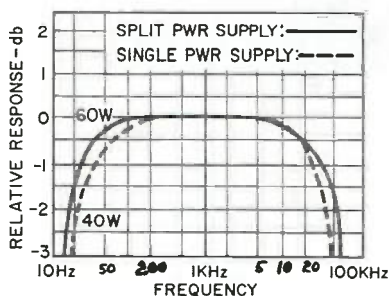
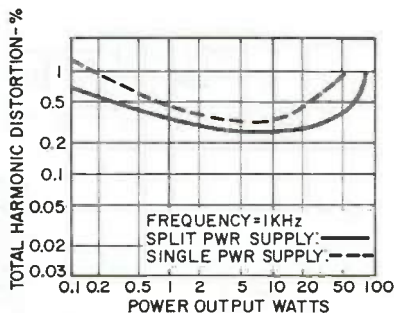
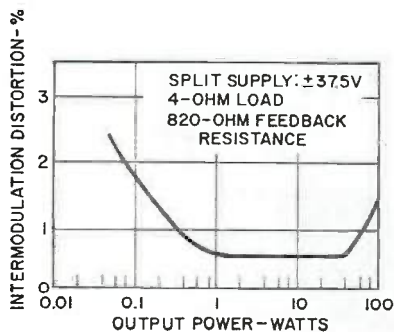


on a heat-sink type chassis indicates the extreme compactness possible for a 200-watt stereo amplifier. The power supply for the modules is mounted on a separate chassis, although single-chassis construction is possible if preferred. The power modules are available for \$80 each.

The protective circuitry of the modules insures they cannot be internally damaged by a sustained short at the output or faulty power-supply voltages. The design permits use of the module with capacitive loads (electrostatic speakers).

### Structure & circuits

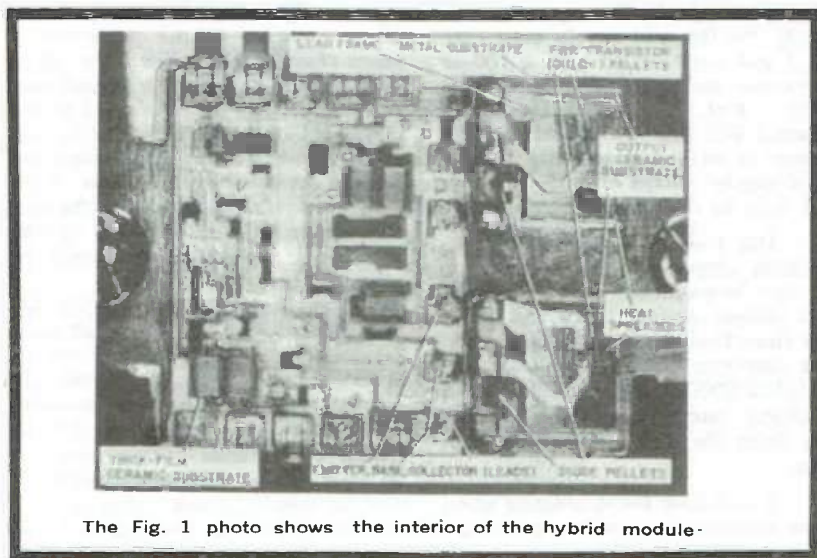
The internal structure of the hybrid amplifier is shown in Fig. 1. It has a rugged steel base plate on which both the thick-film circuit and the output assembly are mounted. This output-transistor assembly is an all lead-solder mount system with a lead frame for easy fabrication. A ceramic substrate is used under each pellet to isolate it from the base plate. The pellet is mounted on a heat spreader which reduces the heat flux density by the time it gets to the ceramic.



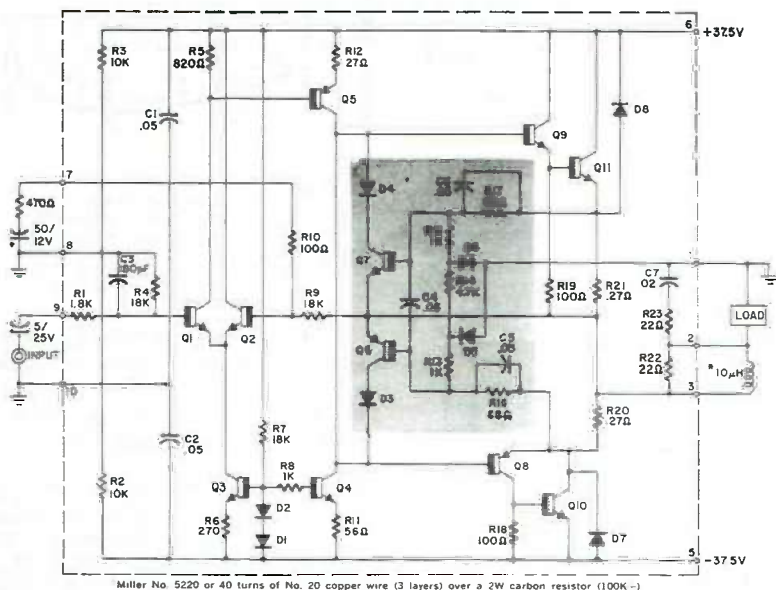
Graphs I-III above show performance curves for the IC.

The schematic for the circuit of the hybrid power amplifier is shown in Fig. 2. A differential bridge (Q1-Q2) is employed at the input of the amplifier to provide good control on the quiescent operating point zero-output voltage. Imbalances of the bridge circuit with supply variations are solved by using constant-current source Q3 in the emitter circuit. This approach is almost universally used in integrated circuits to function over a wide range of supply voltages.

The voltage at pin 8 dictates the output voltage at pin 3. A pair of



The Fig. 1 photo shows the interior of the hybrid module.



matched resistors (R2 and R3) control the output voltage to half the supply voltage. Pin 8 is grounded directly for the split supply case (Fig. 3-a) and ac-grounded with a 100- $\mu$ F capacitor for the single supply case (Fig. 3-b). The constant-current sources (Q3 and Q4) allow the amplifier to be biased properly from a total supply voltage of much less than 30 volts to the 75-volt maximum.

The true class-B quasi-complementary output/driver stage is driven by the bidirectional current source. The output devices (Q10 and Q11) are chips from the 2N3055 family and the pnp/npn drivers are from the 2N5322/2N5320 families. The bidirectional current source npn/pnp's are from the 2N2101/2N4036 families.

A technique for eliminating cross-over distortion in class-B output stages is to operate them from a high-imped-

ance source (a current source). The current transfer characteristics of a typical output stage do not exhibit the offset steps as do the transconductance characteristics. The high-impedance source consists of a second current source, Q4, direct-coupled to the class-A predriver, Q5. In this case, the class-A stage is a pnp device since the bridge devices are now npn type. This arrangement also allows the current source to be an npn type as well and can share the same reference diodes, D1 and D2.

This approach is a simple collector-to-collector connection of complementary devices providing an inherently high collector impedance source; together with the differential bridge, the system performs as a "controlled bidirectional current source." Its operation is completely independent of supply voltage variations.

Versatility was achieved by bringing the feedback terminal out (pin



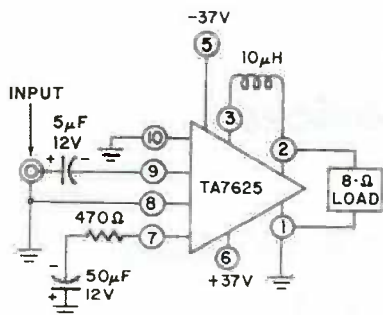


Fig. 3-a—This power supply configuration is a split supply (pin 8 grounded).

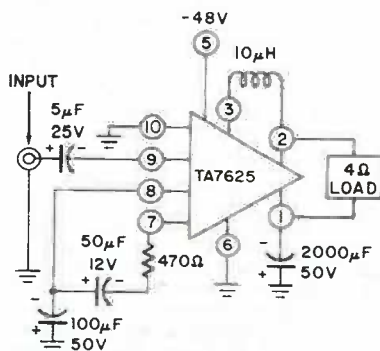
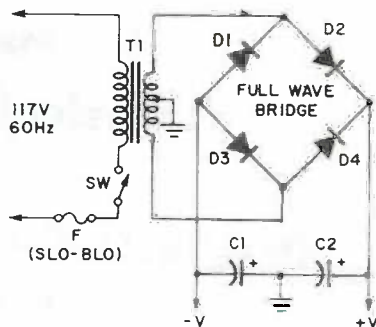


Fig. 3-b—The single supply arrangement with pin 8 ac grounded through 100  $\mu$ F.

7). Maximum gain is limited by the 100-ohm internal resistor, R10.

The load-line limiting circuit (shaded area) protects the amplifier from a sustained short at the output. This unique limiting circuit places restrictions on the allowable operating area of the drivers and outputs. It functions as a voltage and current sampling circuit to provide operating



Power supply circuit for stereo ampl.

### TA-7625 Power Supplies

$P_{out}$	C1-C2	D1-D4	F
120 watts (60W/chan.)	5000 $\mu$ F 50V	1N1202A	7A
40 watts (20W/chan.)	3500 $\mu$ F 50V	1N1614	2A

limitations on the output devices. The limits chosen do not inhibit normal operation, but any overload condition including a short circuit will remain within the safe area of operation of the device. Diodes D7 and D8 and the load-line limiting circuit protect against a faulty output transformer secondary that may reflect voltages to the amplifier output termination, exceeding the power supply voltages. The diodes clamp this reflected voltage to the power supply, while the load-line limiting circuit limits the drive.

Controlling the gain bandwidth product of the output stage in relation to the preceding stages and adding the 10- $\mu$ H inductor and R23-C7 for capacitive loads are other design factors for added versatility.

## PART 2

# Musical Devices

## DIGISYNTONE MUSIC SYNTHESIZER

by F. B. MAYNARD

THE DIGISYNTONE IS A RADICALLY NEW approach to a build-it-yourself music synthesizer. Basically, the Digisyntone system is a special-purpose IC digital computer. It reduces the musical scale to numbers, and generates tonal effects by digital operations with these numbers.

The Digisyntone system gives you a wide range of voices, whose quality can be controlled by mixing pure and accurate harmonics. Digisyntone is the only electronic musical instrument you can build that never needs tuning—it's automatically tuned when you build it.

The basic project uses 21 IC's, and costs from \$75 to more than \$100.

Although Digisyntone has a standard organ keyboard, it is monatonic. That is, it will play only one note at a time, and hence is strictly a solo instrument like the trumpet, clarinet, saxophone, etc. Pure harmonic mixing results in many tonal effects—the instrument's major musical capability.

### Numerical musical relations

There are several basic numerical relations in music. Octaves are related by a factor of 2. For any note, middle C, for example, there are other C's above  $2\times$ ,  $4\times$ ,  $8\times$ , etc. the middle C frequency, and some below  $\frac{1}{2}$ ,  $\frac{1}{4}$ , etc. the frequency. Hence, a simple way of generating octaves is multiplying or dividing exactly by 2.

The musical scale has 12 notes in each octave. The frequency relation here between any one note and its neighbor is a  $^{12}\sqrt{2}$  factor. That is, the frequency of C#, and this applies to any C# in the scale, is the frequency of any C  $^{12}\sqrt{2}$ . The numerical value of  $^{12}\sqrt{2}$  is irrational: like  $\pi$ , it never comes out even.

A close value is 1.0595. The series of numbers listed in Fig. 3 have this relationship. Other series of numbers could be used, but they would be no better, if as good as these. The derived intervals are not perfect, but are very close. The worst case error is less than 5% of a semitone.

A third numerical relation concerns harmonics. By definition, a second harmonic is  $2\times$  the frequency of any starting or fundamental frequency, or simply a frequency 1 octave above the fundamental. The fourth harmonic is  $4\times$  the fundamental, or 2 octaves above, and so on with the even harmonics.

A third harmonic is  $3\times$  the fundamental, and octaves of this give the sixth, twelfth, twenty-fourth, etc. The fifth harmonic is of course  $5\times$ , and this gives the octaves as  $10\times$ ,  $20\times$ ,  $40\times$ , etc. harmonics.

Much more could be written about the theory of musical tones but, for this project, the above is quite adequate.

### The basic digital system

Using electronic circuits, it is much easier to divide frequencies than to multiply them. With integrated circuits, a frequency can easily be divided by any positive whole number, and this is the basis for the digital music system to be described. Fig. 1 shows the basic system in functional block form.

The oscillator operates at a relatively high frequency. The precise frequency can be controlled with R1, and the frequency can be modulated with a vibrato signal through R3.

The oscillator drives a divisor counter which is in turn controlled by a

logic control or decoder and keyboard switches. The logic controller controls a feedback to the counter and the combination divides the oscillator frequency by any one of 12 whole numbers. These numbers, listed in Fig. 3, have ratios close to  $12\sqrt{2}$ .

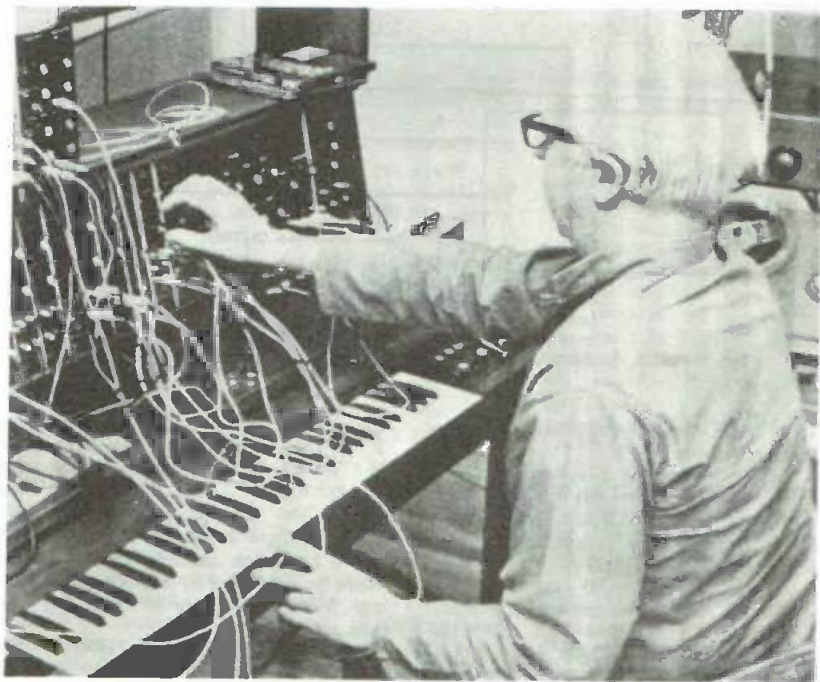
Hence the output from the logic control feedback circuit will present the factors of a musical scale which is independent of actual frequency. This output is now split into two paths. Path **PA** is divided by 2 twice, providing two octaves of the third harmonic. Path **PB** is divided by 3 and by 2 three times. This provides four octaves of the fundamental. (Note that since the path **PB** is divided by 3, the frequencies will be lower by this factor than from path **A**; i.e., **A** is a third harmonic of **B**.)

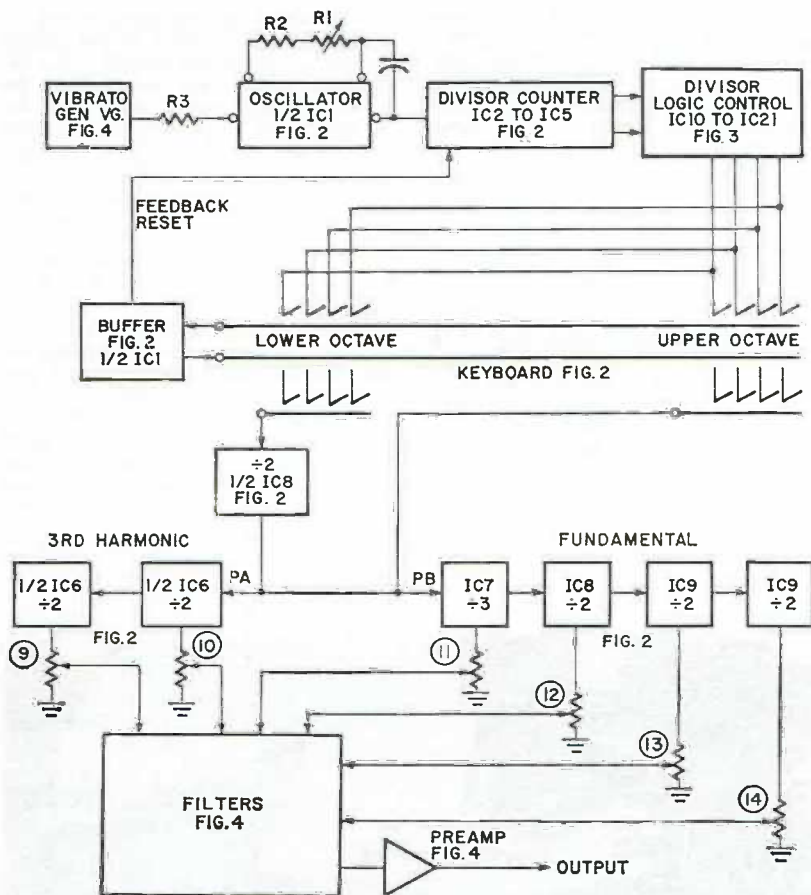
The six outputs are routed to attenuator controls and into a system of filters which change the output waveforms from square to sinusoidal. Four bypass switches are provided to pass the square waves for bright-tone effects.

Circuit details are shown in Fig. 2, 3 and 4. Figs. 2 and 3 consist almost entirely of integrated circuits. Two integrated circuits are called for: the MC724P quad 2-input gate, and the MC790P dual JK flip-flop. Also suitable is the HEP 570 quad 2-input gate, and the HEP 572 dual JK flip-flop.

In Fig. 2, IC1 is the oscillator, with C1 and R1 and R2. This simple RC oscillator can be set up to run at any frequency to about 4 MHz. In this case, its frequency range, variable with R1, is about 1.2-2.5 MHz. IC1 also provides a single-input, two-output buffer.

The oscillator output drives the input (pin 2 of IC2) of the divisor counter, an eight-stage binary counter (IC2 to IC5). There are two binaries in each IC. These are cascaded to count straight binary. Each binary has two outputs: Q outputs, pins 13 and 9 (for the two flip-flops in each package), and  $\bar{Q}$  (not Q), pins 14 and 8. These outputs, except for pin 9 in IC5, go to the decoder logic (IC10-IC21). These outputs are designated in Figs. 2 and 3 as 1,





**FIG. 1—BLOCK DIAGRAM** of the system. The three main sections, oscillator/counter, logic control and filters are detailed for wiring in Figs. 2, 3 & 4.

1, 2, 2, etc. from the flip-flops down the cascaded chain.

The decoder logic control is shown in Fig. 3. This shows the flip-flop inputs across the top of the figure, and horizontal lines indicating connections to the logic control IC inputs.

This decoder logic system consists of 12 MC724P gates connected as eight-input NOR gates. One of these IC's is detailed in Fig. 3. Pins 3, 5, 8, 11 and 14 are connected to a common-output cir-

cuit with external load resistors R5 to R16. The load resistors go to a 2.3V B+ source obtained by decoupling from the main 3.3V source through R17 and C3. When the final circuit tests are made, if there are any conditions under which operation appears fuzzy, they can often be corrected by adjusting this voltage. Making R17 smaller raises the voltage, and vice versa.

The input pins are 1,2,6,7,9,10,12 and 13. These are all the same, and any

input can be connected to any flip-flop output. These connections are shown as dots on the 12 horizontal lines which represent the inputs and outputs of the 12 IC gates, IC10-IC21. The IC outputs are labeled to notes of the scale, B, A#, A, G#, etc., indicating the key switches to which they are connected. The column of divisor numbers indicates the set of numbers which closely approximate the ratios of  $12\sqrt{2}$ . The logic connections (inputs to the gates) are derived from the equivalent binary numbers shown to the right in Fig. 3. For convenience, these binary numbers are shown backward from conventional—the least significant digits are to the left.

These NOR gates function as follows. IC10, for example, divides by the number 131, and connections are made such that when there are 1's in the binary numbers, the  $\bar{Q}$ , or  $\bar{1}$ ,  $\bar{2}$ , etc. flip-flop outputs are connected to gate inputs; where there are 0's, the Q outputs are connected. During its cycle of 256 counts, when and only when the counter has accumulated 131 counts, all the flip-flop outputs to IC10 will be 0's, and this is the only time when IC10 can produce an output.

The logic control system is shown in Fig. 2 as the lower dashed-line box, with flip-flop inputs  $\bar{1}$ , 1, 2,  $\bar{2}$ , etc. and the gate outputs.

Gate outputs B, A#, A, etc. are switched by key-switch contacts on keyboard KS1. This has two key-switch contact sets (that is, two make contacts) and buses on each key. The upper contacts are wired with all B keys, all A# keys, etc. on common circuits, making 12 key-switch inputs from the 12 gate outputs.

When a key is pressed, the output is transferred to bus A which goes to IC1 input pin 12. Part of IC1 serves as the oscillator already described, as well as the buffer function. The buffer has two outputs. The reset output (pin 8) goes to pins 10 and 12 of counter binaries IC2-IC5. This completes the feedback loop, causing a reset of the counter to zero at the number of counts designated in Fig. 3 for any key pressed.

The trigger output from the buffer (pins 9, 10, 14 of IC1) goes to the second set of key switches on the keyboard. This set is split into buses B and C, each 1 octave long. When a key in the upper

octave is pressed, the trigger from IC1 is transferred to bus B. This connects to pin 2 input of IC6 and pins 2 and 6 inputs of IC7. The input frequency for these will be the same as the trigger frequency, or the oscillator frequency divided by the divisor number for that key.

When a lower octave key is pressed, the trigger goes to bus C, and to the pin 2 input of IC8. The pin 13 output of the first flip-flop in IC8 is coupled through C2 to the same inputs connected to bus A. This operation divides the trigger frequency by 2, lowering it by 1 octave whenever a lower octave note is played. If the manual or keyboard used has 3 octaves of keys, a third trigger bus under the lower octave routes the signal to a divide-by-4 (two cascaded binaries) or one MC790 connected in cascade, and the output of this is connected through a 0.0015- $\mu$ F capacitor to the bus B input line. This lowers the trigger frequency 2 octaves. This connection (not shown in Fig. 2) is sketched in Fig. 5.

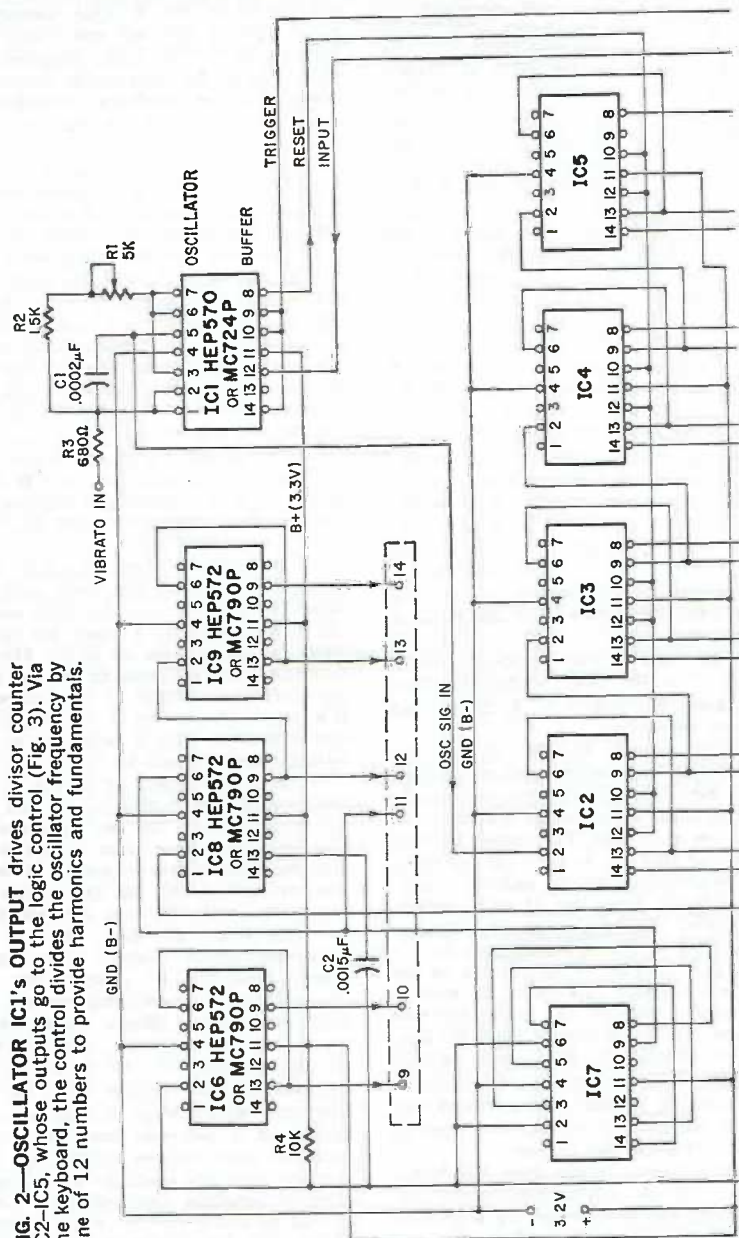
IC6 divides its input frequency by 2 and again by 2, providing two third-harmonic outputs, 9 and 10. IC7 divides this same output by 3 (note the special feedback connections on IC7). The pin 9 output from IC7 goes to the pin 6 input of IC8 and output 11. The output of IC8 pin 9 divides by 2 on output 12, and IC9 divides by 2 twice more, providing outputs 13 and 14.

These six outputs go to Fig. 4. Attenuator pots R18-R23 insure that any of these outputs, in any combination, can be gated from zero to maximum into the filter system of Fig. 4. Note 1K pots are called for, but any values between 1K and 10K are suitable here.

The filters are four low-pass RC networks which convert the square waves from the IC circuits into sine waves. These are desirable for the best harmonic mixing effects. Some square waves are also desirable, and two each of the fundamentals and harmonics are bypassed through switches S2 to S5. The filter and square-wave outputs are collected on a common line into a pre-amplifier with volume control Q2, and patched into any medium- or high-gain external amplifier through C21, which should be a 200-volt or more paper or Mylar unit for safety in patching into vacuum-tube amplifiers.



**FIG. 2—OSCILLATOR IC1's OUTPUT** drives divisor counter IC2-IC5, whose outputs go to the logic control (Fig. 3). Via the keyboard, the control divides the oscillator frequency by one of 12 numbers to provide harmonics and fundamentals.





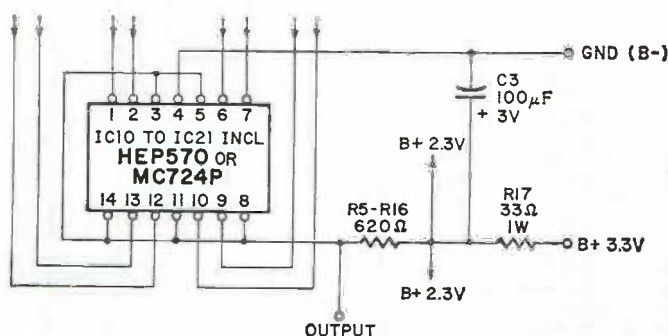
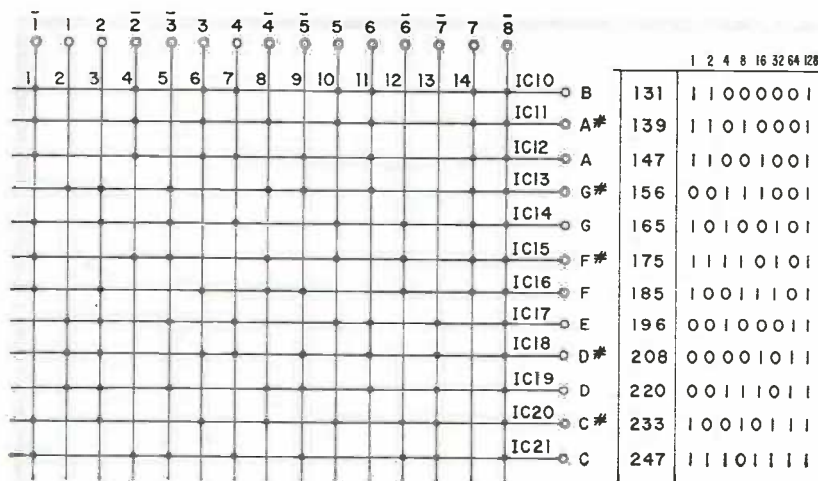


FIG. 3—DECODER LOGIC section uses 12 IC's connected as eight-input NOR gates as shown in the lower drawing. Dots on the horizontal lines indicate where the NOR gate inputs should be connected to the divisor counter (IC's 2-5). Outputs to the switches are the scale notes (vertical column).

Transistor Q1 provides a 6-Hz twin-T oscillator for vibrato. This output goes to resistor R4 on the main oscillator, IC1. In the event the vibrato is too deep, R4 can be made larger, and vice versa. The vibrato is stopped by opening switch S1.

### Construction

Despite its complexity the Digisyn-tone is surprisingly easy to build, thanks to the integrated circuits. Make sure all

connections are correctly and securely made, and that no shorts occur from stray bits of solder and incorrectly oriented connecting wires.

There are several good ways of mounting IC's including the use of sockets, which, however, are expensive. The best way is probably on etched circuit boards. The mounting used for the prototype is on pattern G Vector board with T-28 push-in terminals. The IC's are mounted upside down and the power

and in and out leads are made either directly from the terminals or to extended leads to terminals. Fig. 5 shows enlarged views of typical mountings for both the MC724P gates and the MC790P flip-flops. A suggested switch and control arrangement is shown in Fig. 6.

The circuit requires a supply voltage of 3.3 to 3.7 volts at about 180 mA. A suitable power supply to furnish this regulated voltage is simple to construct and is shown in Fig. 7.

The mounting of the assembled circuits, keyboard and controls is not criti-

cal. It is probably advisable to mount the oscillator, or at least control pot R1, in a shield, since it has a frequency in the radio broadcast range. Also, keeping the interconnecting leads as short as possible will reduce tendencies for cross-talk.

### Using the Digisytone

The several controls which govern the musical capability are described here. Refer to Fig. 6 for a pictorial layout of these controls. The instrument pitch is under PITCH control R1. Since the system automatically delivers a cor-

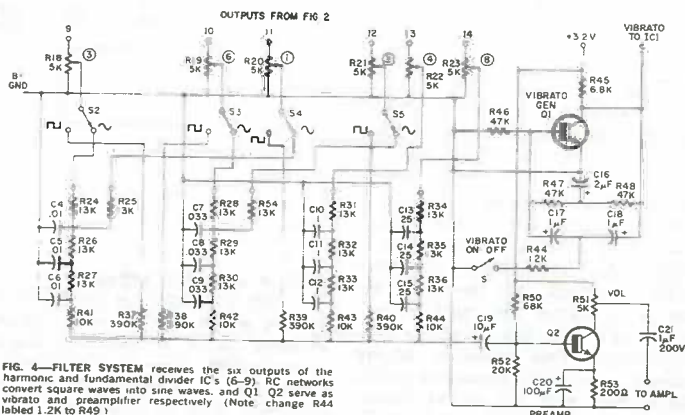


FIG. 4—FILTER SYSTEM receives the six outputs of the harmonic and fundamental divider IC's (6-9). RC networks convert square waves into sine waves, and Q1, Q2 serve as vibrato and preamplifier respectively (Note: change R44 labeled 1.2K to R49.)

### PARTS LIST

#### Capacitors

- C1—200 pF
- C2—.0015  $\mu$ F
- C3—100- $\mu$ F, 3V electrolytic
- C4—C6—.01  $\mu$ F
- C7—C9—.033  $\mu$ F
- C10—C12—.1  $\mu$ F
- C13—C15—.025  $\mu$ F
- C16—2  $\mu$ F, 10V
- C17, C18—1  $\mu$ F, 10V
- C19, C20—10  $\mu$ F, 10V
- C21—1  $\mu$ F, 200V
- C22—2,000- $\mu$ F, 6V electrolytic
- C23—5,000- $\mu$ F, 6V electrolytic

All resistors  $\frac{1}{2}$ W, 10% unless noted

R1, R18—R234, R51—5,000-ohm linear potentiometers

- R2—1,500 ohms
- R3—680 ohms
- R4, R41—R44—10,000 ohms
- R5—R16—620 ohms
- R17—33 ohms, 1W
- R24—R36—13,000 ohms
- R37—R40—390,000 ohms
- R45—6,800 ohms
- R46—R48—47,000 ohms

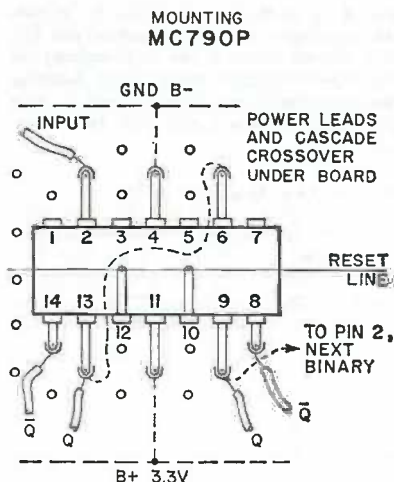
- R49—1,200 ohms
- R50—68,000 ohms
- R52—20,000 ohms
- R53—200 ohms
- R54—13,000 ohms
- R55—5 ohms, 10W
- R56—100 ohms, 2W
- R57—120 ohms, 2W

#### Semiconductors

- D1—3.6V, 1W Zener diode (HEP 102 or equiv)
- Q1, Q2—MPS2926 or HEP50 transistor
- Q3—npn power transistor (HEP 247 or equiv)
- RECT 1—full-wave bridge rectifier (HEP 175 or equiv)
- IC1, IC10—IC21—Quad 2-input gates (MC724P or HEP570)
- IC2—IC9—Dual JK flip-flop (MC790P or HEP-572)

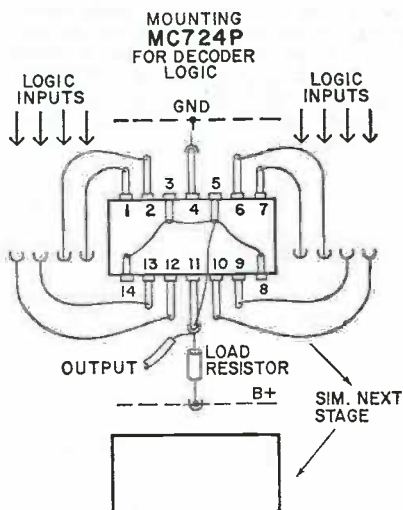
#### Other parts

- T1—6.3V, 2A filament transformer
- S1—S5—spst switch
- S1—spst toggle or slide switch
- S2 through S5—spdt toggle or slide switch
- Keyboard (see text)
- Pattern G vector board
- T-28 push-in terminals
- Miscellaneous wire and hardware



**FIG. 5—IC MOUNTING** technique will insure a neat and functional layout.

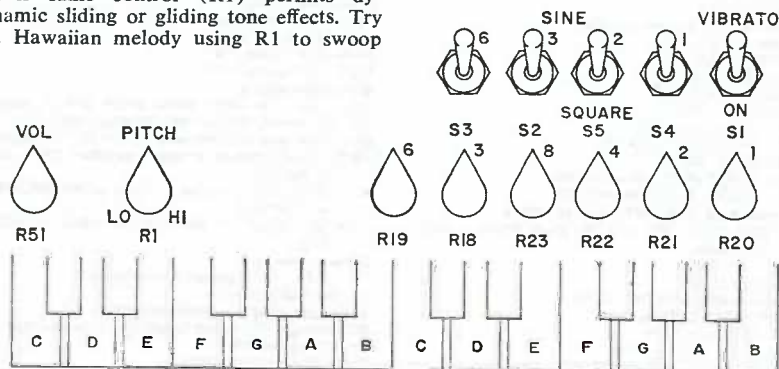
rect musical scale independent of pitch, the pitch can be adjusted to tune to any conditions. A point can be found where the middle A key, for example, will tune to 440 Hz, in which case the entire instrument is in normal tune to the standard scale. R1 can also tune to any other frequency. This provides, among other things, a key-shift feature in which the player may play on the keyboard, for example, in a simple key such as C, but obtain sound in any other key. This capability should be a welcome feature to the many "favorite key" musicians. This same control (R1) permits dynamic sliding or gliding tone effects. Try a Hawaiian melody using R1 to swoop



the tone upward or downward in real Island style.

### Other special effects

The VOL knob is simply a volume control which can be manipulated as a dynamic swell or loudness control for special effects. The four harmonic mixer controls (1, 2, 4 and 8) provide the fundamental, and 3, 6 controls the third-harmonic system. When off, to the left, these controls do not permit any signal admittance to the output system. In the



**FIG. 6—KEYBOARD** and control arrangement suitable for the Digisynth.



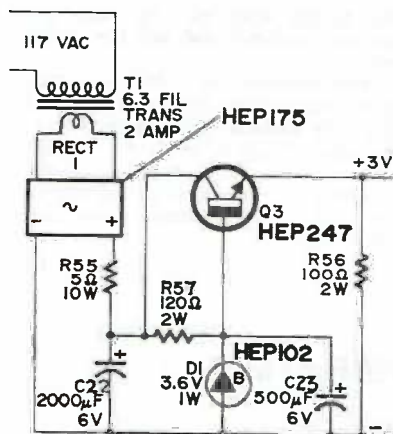


FIG. 7—POWER SUPPLY for the synthesizer provides well-regulated 3 volts.

full clockwise position, all the signal is injected. These controls may be adjusted of course to any level in between. Many

combinations are possible with these six controls, and most of them will sound distinctive.

The harmonic mixing will be most effective on the sine filters (S2–S5 up). Other tonal effects can be obtained with square waves (S2–S5 down). The vibrato is turned on or off with S1.

It has been said that the instrument is monatonic; i.e., only one note is played at one time. Note that if two keys are played at once, except for octaves, nothing sounds. Inherent in the digital tone system, this characteristic may be used to provide interesting chopped-tone effects by playing on a key and rapidly keying another to chop the tone.

### A note on keyboards

A suitable keyboard for the Digi-syntone is the one thing a builder cannot readily buy over the counter. There are a number of possibilities for salvaged keyboards from a piano, toy pi-

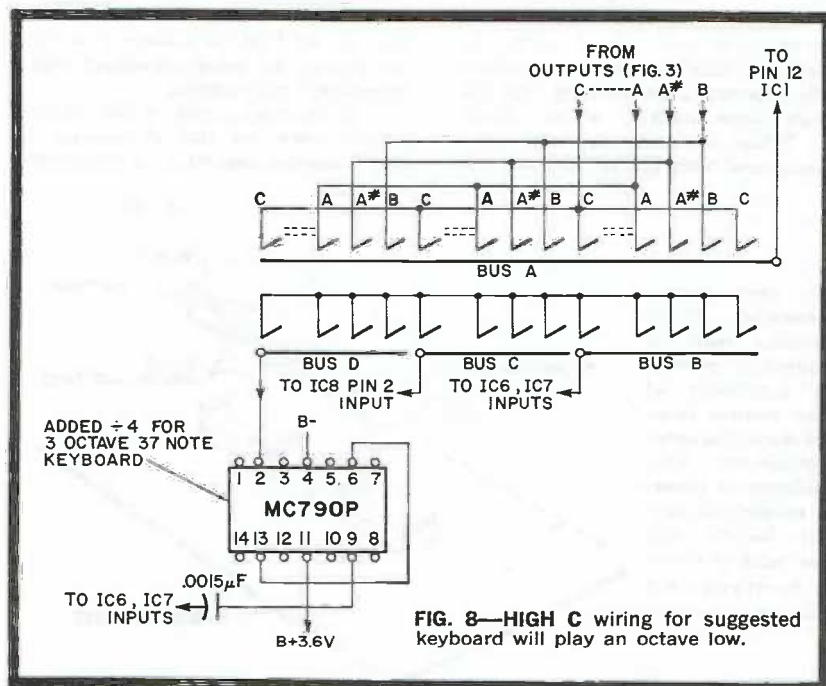


FIG. 8—HIGH C wiring for suggested keyboard will play an octave low.

ano, accordion or reed organ. Most any of these can be fitted with key switches to provide a suitable playing medium.

The best bet, however, is a commercial organ keyboard. These can be obtained from 3 to 5 octaves, with dual key switches and buses, for approximately one dollar per note, from Pratt Read & Co., Inc. Ivoryton, Conn. 06442. You should specify "3 octave, 37

note, manual with two key buses per note, no resistors," and ask for quotations.

This keyboard has a high C, which will be dead in the system shown. But this can be doubled or connected in parallel with the next lower C key. It will sound an octave low, but this is a fairly common way of handling an end note in some organs. This is shown in the diagram of Fig. 8.

## GUITAR VIBRATO

by JACK JAKUES

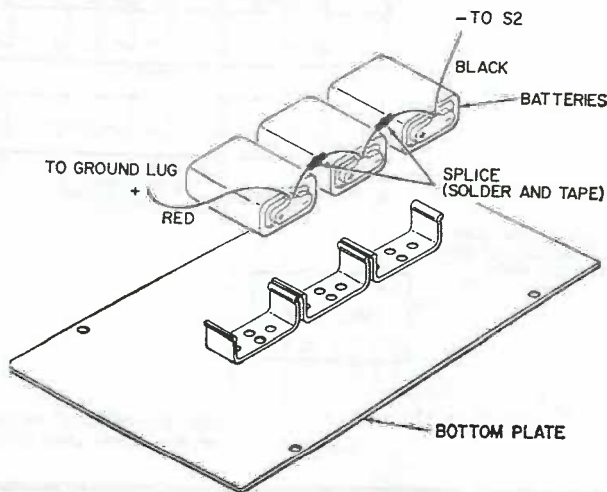
ADD THIS VIBRATO TO THE GUITAR and you'll have given that amplifier new life. In simple terms the vibrato makes the sound pulsate—come and go at a controlled rate. This pulsating rhythm adds real life to otherwise "flat" or "dead" music. Also, with a little educated knob twisting, you can create some unusual sound effects.

While vibratos are most commonly used with guitars and accordi-

ans, the sound of many other musical instruments can be greatly improved. For example, the reed type electric chord organ can be greatly enhanced by adding a vibrato. For instruments that do not lend themselves to a special pickup, try using a standard high-impedance microphone.

If you take a look at the vibrato circuit you'll see that it consists of one transistor, one FET, 2 potentiom-

*The three series-connected 9-volt batteries must be secured so there is no possibility of their coming loose and damaging other components. Too, replacement should be simple and easy. Use battery clips and holders bolted to the bottom cover plate as shown.*



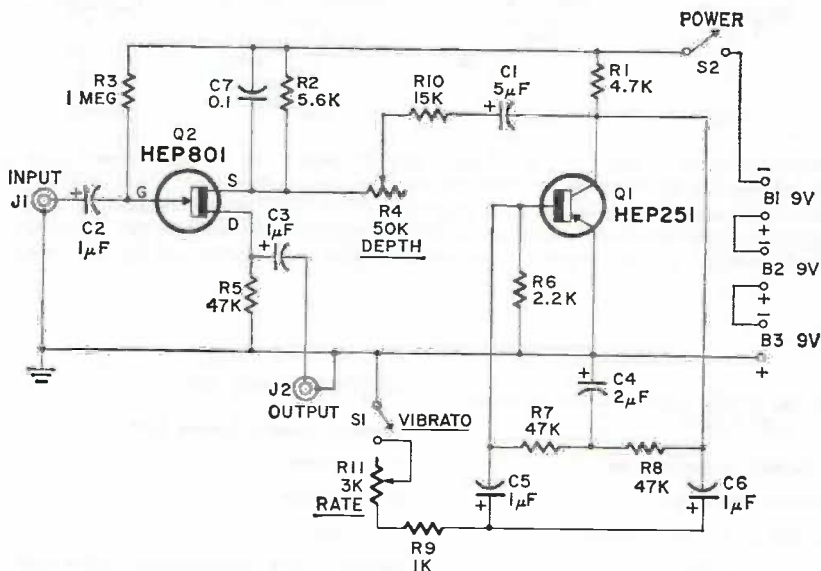
eters, a battery power supply, input and output jacks, and the various resistors and capacitors needed to complete the circuit. The unit delivers an overall gain of 6 to 8 dB, and its high-impedance output is ample for driving most power amplifiers to full output.

In the circuit transistor Q1 is connected as a twin "T" oscillator that produces a sine-wave of approximately 6 Hz. This signal is applied to the source (S) of FET Q2, where it amplitude modulates any incoming signal applied to the gate (G) of Q2 through input jack J1. The amplitude of this modulation can be varied by the DEPTH control R4, to create either a light or a deep vibrato effect. The vibrato rate (frequency) is adjusted with RATE control R11, or may be turned off with switch S1 for normal sound.

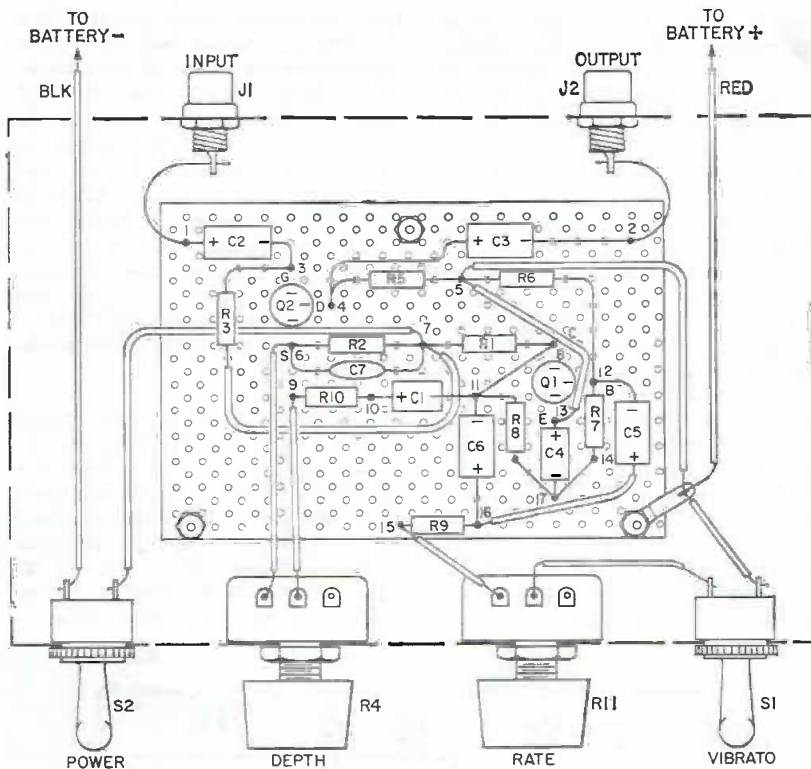
Operating the vibrato is easy. Connect any high-impedance micro-

phone or pickup device to input jack J1. Connect output jack J2 to the high-impedance input of any amplifier system. Place the VIBRATO switch and the POWER switch in the ON position. Adjust potentiometer R4 and R11 to approximately midrange. Then play single notes or chords on the musical instrument being used and adjust the RATE and DEPTH controls to obtain the desired effect. When operated this way, Q2's gain is effectively used, and you'll find it handy if you are using a low-power amplifier.

Complete construction details can be found in the diagrams, although parts layout is not critical. Cut the perforated board to size and carefully drill the three mounting holes. Next insert the push-in terminals where indicated. Then wire in all parts. Watch out for battery and capacitor polarity and you're home.



*A simple, yet effective vibrato circuit. Vibrato effect is produced by a 6-Hz twin-T oscillator that modulates the amplitude of the signal in the source circuit of the input transistor. Input and output impedances are high.*



*Physical layout of parts in the vibrato. Observe polarity of the electrolytics. The numbered circles represent the push-in terminals used as tie-points. Thin insulated hook-up wire is used throughout. The perforated-board chassis is suspended from the top of the case on three screws. These are visible in the head photo. Controls and switches are mounted on the front of the case; jacks are on the rear.*

#### PARTS LIST

B1, B2, B3—9-V mercury battery (Eveready E146X or equiv.)

Capacitors—25 volts or more

C1—5  $\mu$ F, electrolytic

C2, C3, C5, C6—1  $\mu$ F, electrolytic

C4—2  $\mu$ F, electrolytic

C7—0.1  $\mu$ F

J1, J2—phono jacks

Q1—HEP251 pnp transistor

Q2—HEP801 n-channel FET

Resistors— $\frac{1}{2}$ -watt 10% unless noted

R1—4,700 ohms

R2—5,600 ohms

R3—1 megohm

R4—50,000-ohm potentiometer (Mallory U35 or equiv.)

R5, R7, R8—47,000 ohms

R6—2,200 ohms

R9—1,000 ohms

R10—15,000 ohms

R11—3,000 ohm potentiometer

(Mallory U8 or equiv.)

S1, S2—spst toggle switch

#### Miscellaneous Parts

Chassis, 5" x 7" x 2"

Chassis cover, 5" x 7"

Perforated terminal board 2½" x 3¾" (Key-

stone 1753 or equiv.)

Push in terminals (18) (Keystone 1499 FT or equiv.)

Battery clips (3) (Keystone 72 or equiv.)

Battery holders (3) (Keystone 79 or equiv.)

Transistor sockets (2)

Knobs (2)

Terminal plug

Machine screws, 6-32 x ¾" (3) (for terminal board mounting)

Machine screws, 4-40 x 3/16" (6) (for battery holder mounting)

Nuts, 6-32 (9)

Nuts, 4-40 (6)

Rubber feet (4)

## RHYTHM LIGHTS

By R. T. MONTAN'E

TODAY'S POP MUSIC IS WILD AND EXCITING. Tie into it with a light-and-music show! Put Rhythm Lights near live musical instruments—or radio or hi-fi—and flood lamps will fill the room with color in rhythm with the music. This sound-to-light converter is all solid-state for trouble-free operation. No special wiring is necessary—just place the unit near the sound and the lights flash dynamically filling the room with a brilliant color array.

Assembly and wiring of Rhythm Lights is easy. All the components are laid out on a printed-circuit board which is enclosed in a simple, one-piece chassis. All components used are available at most electronic distributors.

### Circuit operation

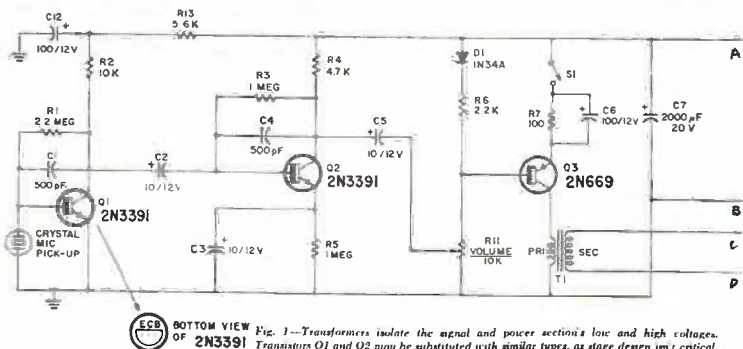
A crystal microphone picks up sound from radio, hi-fi or musical instruments. The high-impedance microphone is directly coupled to transis-

tor Q1 (Fig. 1); this transistor is a grounded-emitter stage with feedback from collector to base for stability. A 100-pF capacitor (C12) decouples the power supply from this low-level first stage. The signal next goes to transistor Q2. Resistor R3 established Q2's operating point and capacitor C4 reduces high-frequency signals. Final amplification takes place at transistor Q3. Volume is controlled by 10,000-ohm pot R11, which acts as part of the bias network for Q3. Diode D1 compensates for temperature effects on Q3.

Transformer T1 isolates the low-voltage amplifier stages from the high-voltage, lamp-driving stage. The signal is rectified by diode D7. Capacitors C8 and C9 limit the frequency response to about 500 Hz. Trigger diode D13, and R12—BRIGHTNESS—set the level at which the SCR (Q4) conducts. The SCR drives the lamps to various brightness levels and is powered through bridge rectifier circuit D9-D12.

Neon lamp PL1 indicates power is on. Transformer T2 supplies 12 volts for the amplifier.

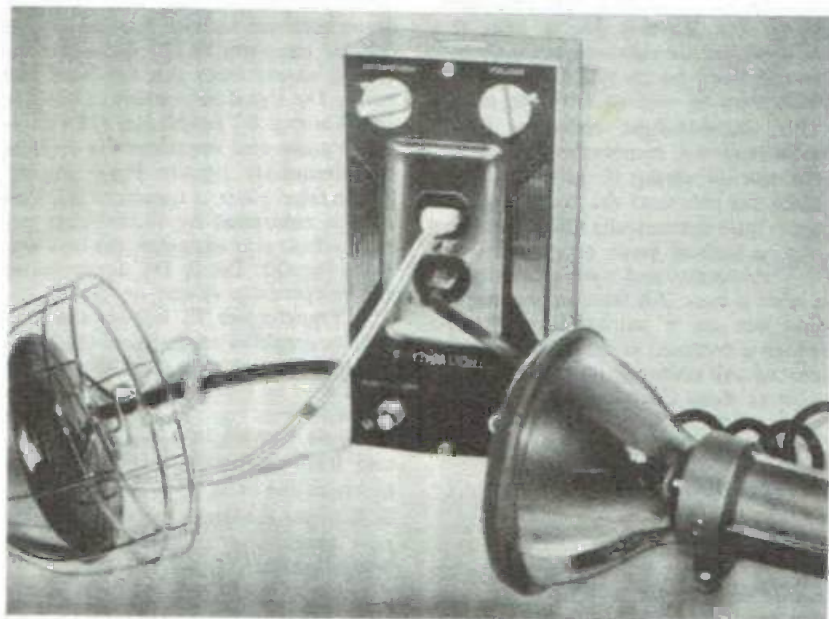




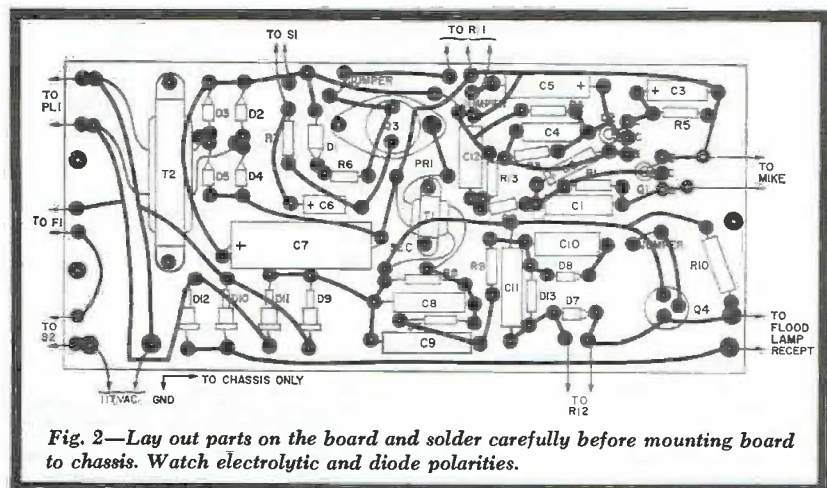
# PARTS LIST

C1, C4—500-pF, 150-volt ceramic capacitor  
 C2, C3, C5—10- $\mu$ F, 12-volt, electrolytic capacitor  
 C6, C12—100- $\mu$ F, 12-volt, electrolytic capacitor  
 C7—2000- $\mu$ F, 20-volt, electrolytic capacitor  
 C8, C11—0.1- $\mu$ F, 150-volt, paper capacitor  
 C9—0.047- $\mu$ F, 150-volt, paper capacitor  
 C10—0.008- $\mu$ F, 150-volt, paper capacitor  
 D1—1N34-A diode

D2, D3, D4, D5—Diode, at least 500 mA, 100 piv (Lafayette 19H5001 or similar)  
 D6, D8—1N4001 diode  
 D7—1N4003 diode  
 D9, D10, D11, D12—Diode, at least 1.5 amps, 200 piv (RCA 40267 or similar)  
 D13—Pnpn trigger diode, T1-42 or similar  
 F1—Fuse, 5 amps, and holder  
 Q1, Q2—2N3391 transistor  
 Q3—2N669 transistor  
 Q4—2N3528 silicon controlled rectifier





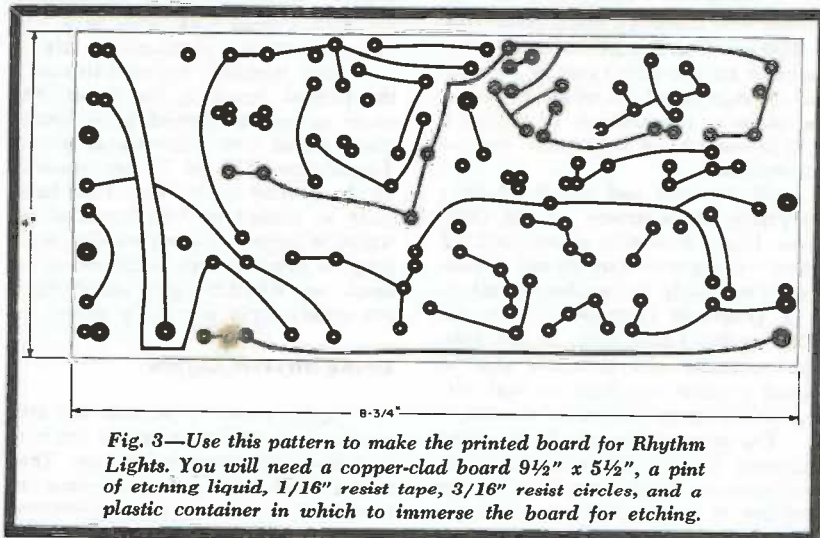


**Fig. 2—**Lay out parts on the board and solder carefully before mounting board to chassis. Watch electrolytic and diode polarities.

Furnish music at a normal level and turn up VOLUME until the music turns on the light. For best results, two 150-watt flood lamps of different colors should be used in a semi-dark room. The results are dramatic.

Switch S1 is a remote on-off device. You can use lamp cord of almost any length, since only 12 volts dc is

present in the circuit. This switch opens the emitter circuit of the final amplifier stage (Q3), disconnecting the input signal. Lamp intensity can still be controlled, however, with the BRIGHTNESS control. Another effect you might want to try: Hook up a string of Christmas-tree lights to the unit. Use your imagination; you'll find many interesting applications for Rhythm Lights.



**Fig. 3—**Use this pattern to make the printed board for Rhythm Lights. You will need a copper-clad board 9 1/2" x 5 1/2", a pint of etching liquid, 1/16" resist tape, 3/16" resist circles, and a plastic container in which to immerse the board for etching.

# ELECTRONIC TREMOLO

By R. H. KEENAN

AN EFFECT COMMON TO MOST KINDS of classical and popular music is *tremolo*: periodic, fairly rapid variation in loudness. It is particularly common in wind instruments, even including the pipe organ. What it amounts to is amplitude modulation of the musical note by a low-frequency, subaudible signal (usually around 5 to 8 cycles per second). In conventional musical instruments it can be produced by varying the wind pressure applied to them.

Tremolo is *not* the same thing as *vibrato*, which is slow frequency modulation (FM) and sounds quite different. Pipe organs never have vi-

brato. The two words are often confused.

Tremolo (AM) is easy to add to an existing music source. The simple device described here can be used with electronic guitars, organs or other instruments, or with recorded music or noise to produce special effects for electronic music or sound-and-light shows. The circuit is simple and can be added quickly to any amplifier or tape recorder.

## How it works

The subaudible tremolo signal is produced by Q1, a unijunction transi-

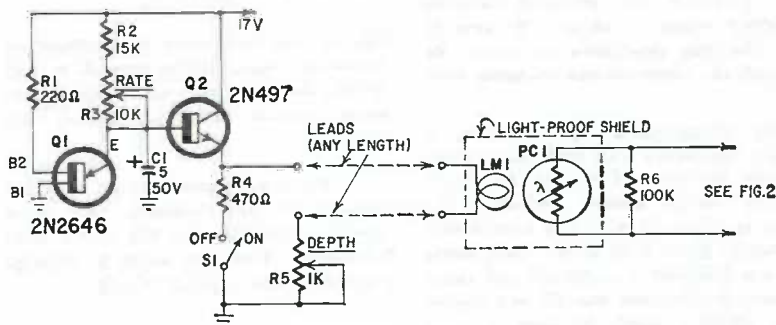


Fig. 1—Two 9-volt transistor batteries in series will power the tremolo nicely. You can install the lamp and photocell any convenient distance from controls.

## PARTS LIST

C1—5-μF, 50-volt electrolytic capacitor  
R1—220-ohm resistor  
R2—15,000-ohm resistor  
R3—10,000-ohm potentiometer (RATE)  
R4—470-ohm resistor  
R5—1000-ohm potentiometer (DEPTH)  
R6—100,000-ohm resistor for shunting PC1 (you may need to select best value from 47,000 to 470,000 ohms)

PC1—Photocell (Clairex CL607)  
LM1—Incandescent lamp, 28 volts, 40 mA (G-E 327 or similar)  
Q1—Unijunction transistor (2N2646)  
Q2—Large-signal npn transistor (2N497 or similar)  
S1—S.p.s.t. switch of any convenient kind. Case and perforated board or other means for mounting Q1-Q2 circuitry.

tor, used in a simple relaxation oscillator (Fig. 1). Q2 amplifies the signal and drives lamp LM1, which flickers at the rate set by the oscillator. The varying light falls on cadmium-sulfide photocell PC1, causing its resistance to vary accordingly. The photocell is part of the series arm of a voltage divider (attenuator), so the audio level varies periodically as the oscillator swings. R3 is the tremolo rate control; R5 controls the depth of modulation.

The photocell and lamp (see parts list) are placed end to end and rolled together into a strip of black

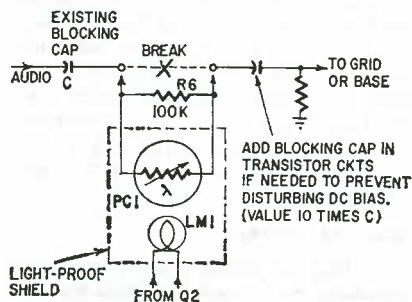


Fig. 2—Insert the photocell between amplifier stages as shown. Be sure to use blocking capacitors to avoid dc through the photocell and changing bias.

plastic electrician's tape, making a single, lightproof unit with leads coming out the ends. Total current drain for the circuit shown, including the lamp, is about 20 mA. The circuit will probably work with other transistors; Q1 is a 2N2646, a common and inexpensive unijunction, and Q2 is a 4-watt npn 2N497, used because it was handy. The circuit could be rearranged for use with a pnp transistor as Q2. This transistor should have fairly high beta ( $h_{FE}$ ) and be capable of dissipating at least 200 mW.

### Circuit connection

Connect the tremolo device between amplifier stages, either tube or

transistor, as in Fig. 2. You may want to trim the 100,000-ohm resistor, depending on the values of the resistors in the circuit. Keep the leads on the photocell short, else they may pick up hum. The leads to the lamp may be any length; this means that the unit containing Q1 and Q2 can be put in any convenient place.

When the tremolo is used with an electronic organ, it is usually desirable to keep tremolo off the pedal tones (tremolo in low bass notes just doesn't sound right). Thus you may want to insert the photocell in the lead that carries signals from one or more manuals only, before the pedal tones are mixed in (Fig. 3). For more specific information, consult the diagram of your organ.

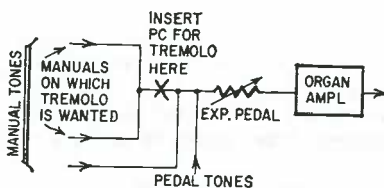


Fig. 3—You may want to split-feed an electronic organ, adding tremolo to only certain manuals, and not to others. See organ manual before attempting this.

Pipe organs generally do not have tremolo on all manuals. The *great* almost never has it, the *swell* does frequently. You may want to arrange your electronic organ similarly.

### Controls

Potentiometer R3 is for rate, R5 for depth. Although electrically the two controls are quite independent, psychologically there seems to be some interaction between rate and depth adjustments. You may want to rig up a switch with two or three values of fixed resistance so that you can select



quickly, while playing, different rates or depths of tremolo.

To check out the circuit, connect an ohmmeter across the leads of PC1. If the circuit is oscillating and the lamp is glowing, the needle should fluctuate.

The 2N2646 is fairly easily dam-

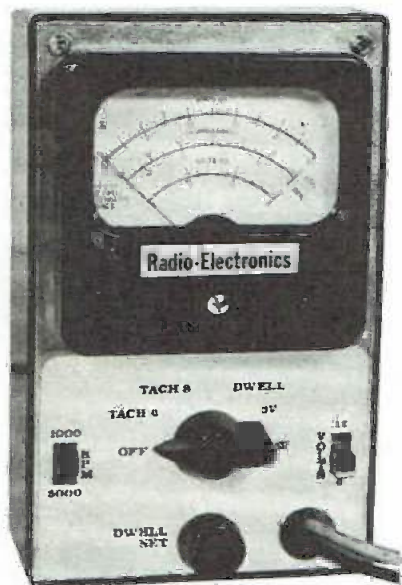
aged by heat, so use heat-sink clips when soldering, and solder quickly. Its case is connected internally to base 2, so be sure the case doesn't come into contact with any alien wires. A one-turn wrap of tape or a tight-fitting plastic tubing will prevent mishaps.

## PART 3

# Projects for the Auto

## TACH-DWELL VOLTMETER

By J. COLT and L. M. BOGGS



THE IDEAL AUTOMOTIVE TEST INSTRUMENT should be able to measure everything from headlamp candlepower to the water content of exhaust gases. It should be self-powered and thin enough to carry in your wallet. Cost should be less than 50 cents.

You may have to wait a few years for that instrument. For the time being, you might like an instrument that measures rpm and dwell (4, 6 or 8 cylinders) and includes a 3-volt and a 15-volt range for individual-cell and

system voltage measurements. It costs approximately \$10.

The TDVM's (Tach-Dwell-Voltmeter) low cost and professional appearance come from using low-cost components and a predrawn meter face and from combining circuit functions (Fig. 1). Many of the electronic parts come from discount-house "2-for" or "5-for" buys—for example, the switches, Zener diodes and the 0.1- $\mu$ F capacitors three of which are paralleled to make C1. Even if you decide to build the TDVM with name-brand components, the price of the completed unit should be well under what

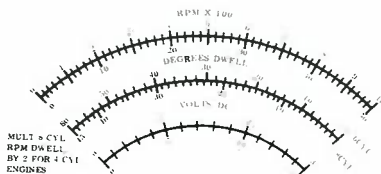


Fig. 1-Meter scale for the TDVM is reproduced here actual size. Drawing can be traced or photostated.

you'd expect to pay for a comparable commercial unit.

Try to use the meter specified. If you have a different 1-mA movement you want to use instead, you may have to make a new meter face and choose correct multiplying resistors for the voltmeter ranges. Substitutions for transistors Q1 and Q2 are not recommended unless you're *sure* your substitutes have an  $h_{FE}$  of at least 100.

For clarity, the description will be divided into three parts, each section describing one of the three function-

ally separate subcircuits (see Fig. 2).

The tachometer consists of a monostable (one-shot) multivibrator whose output is a constant-width pulse. Since the output is of constant width, the dc component of the output pulses varies linearly with the repetition rate of the pulses fed into the one-shot multivibrator. These pulses are taken from the engine breaker points.

Speed range is increased by switching the timing resistors; R4 is for low speed, and R5-R6 for high speeds. This changes the discharge time of C1.

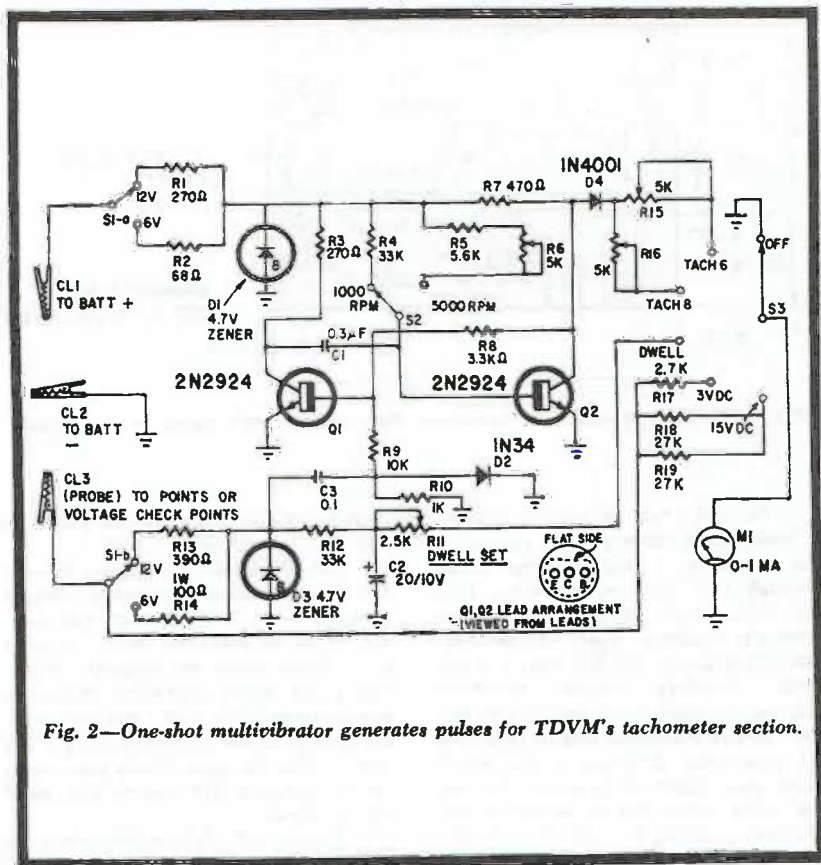


Fig. 2—One-shot multivibrator generates pulses for TDVM's tachometer section.

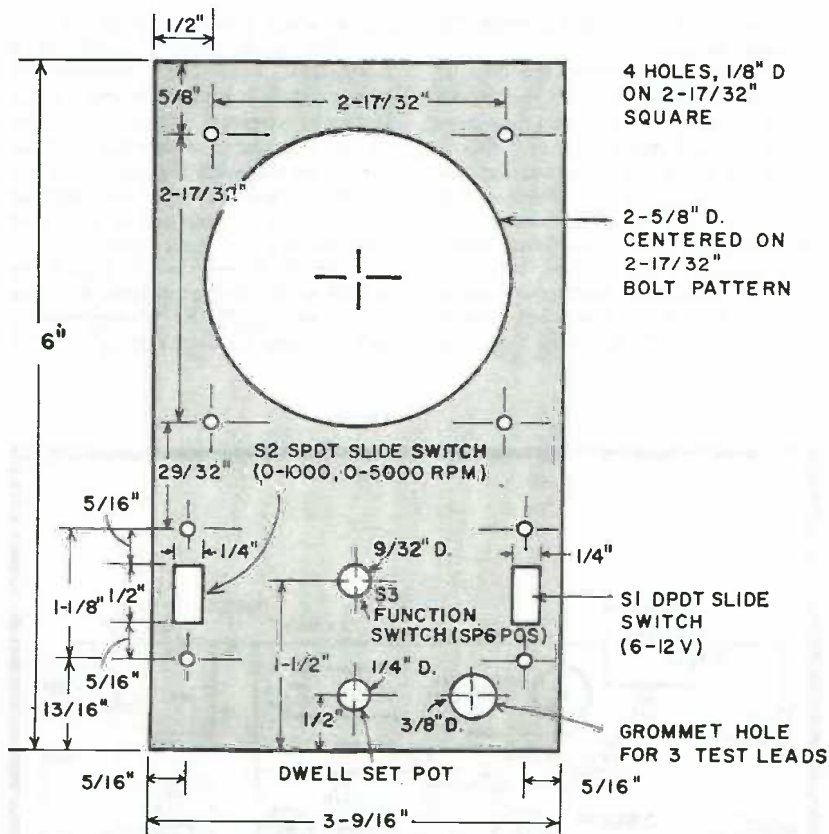


Fig. 3—Author's front-panel dimensions. Parts and controls layout is not critical.

To make both 6- and 8- (or 4-) cylinder measurements, the output of the one-shot is fed to the meter through two different resistors, R15 and R16, which are adjusted to give accurate readings. Both are current-limiting resistors. At 600 rpm, a 6-cylinder, 4-stroke engine generates (across the points) 30 pulses per second, and an 8-cylinder engine generates 40 pps—two different rates which must give identical currents through the meter. Since the dc output of the one-shot is going to vary, the obvious way to obtain identical readings is to

provide two different current paths for the two conditions.

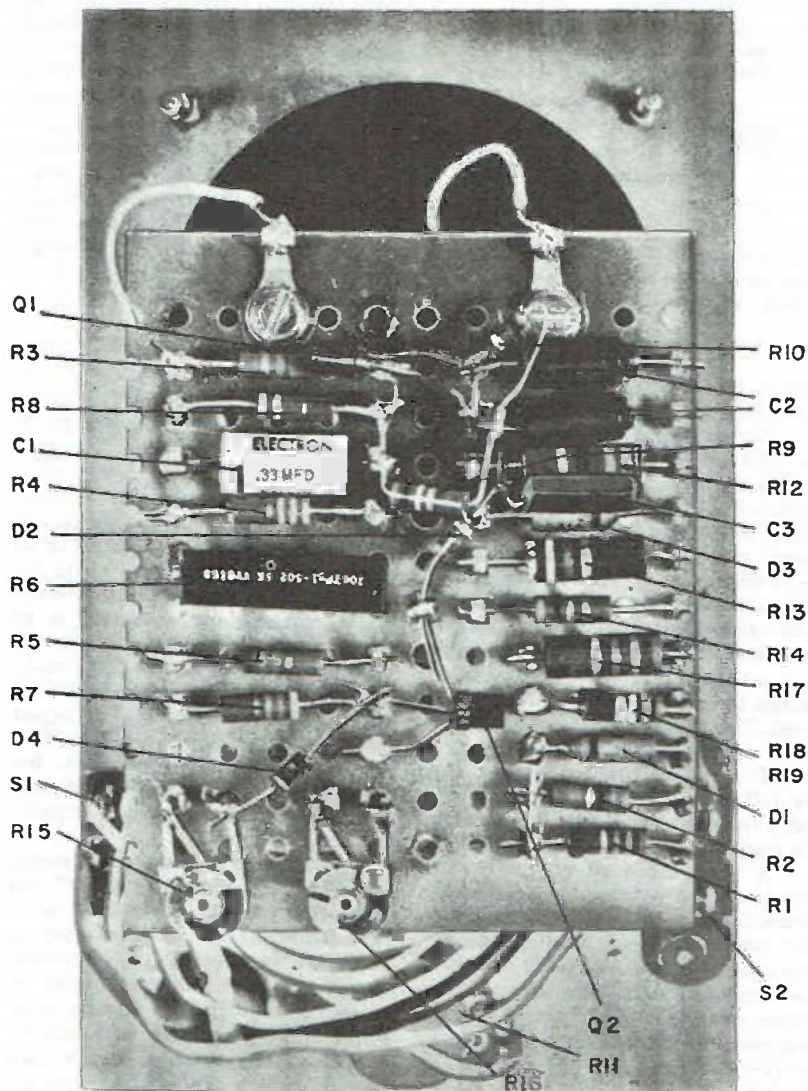
D4 is included because transistor Q2 has a small saturation voltage across it when it is fully on. This voltage tends to bias the meter in error everywhere except an original calibration point. Q2's saturation voltage is approximately 0.2 volt, and the turn-on voltage of D4 is approximately 0.6 volt, so that the meter reads zero when Q2 is saturated. D4 can be any good silicon diode.

Power for the multivibrator is supplied from the car battery via the

shunt regulator combination R1 or R2 and D1. R1 and R2, as well as R13 and R14, are switched to provide essentially constant voltage across the

Zener diodes used in the tachometer and dwell-meter sections.

Note that the clip lead marked (+) need be connected to the car



*Construction is not critical. Components are mounted on push-through pins. The perforated board is held in place by the two terminals on the meter. Observe polarity of the diodes. Avoid excessive heating of the leads from the diodes and the transistors.*



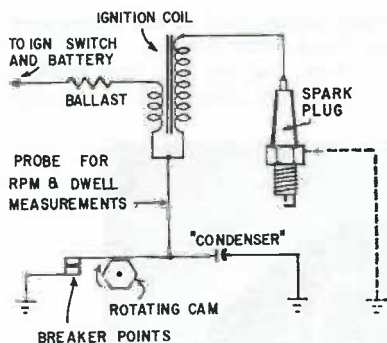


Fig. 4—Connect TDVM probe to conventional ignition system as shown, with ground clip to a convenient point.

battery only when using the TDVM tachometer function.

The dwell-meter section is made up of R13, R14, D3, R12, C2 and R11. It operates on the principle that, for a particular dwell angle, the duty cycle (ratio of "on" to "off") of the pulses across the car's points is constant. Since constant amplitude is assured by D3, we have a train of constant-amplitude, constant-duty-cycle pulses which has a certain dc component. The only way to change this dc component is to change the duty cycle, i.e., change the dwell. The dc component is read at the meter as dwell angle.

To calibrate the dwell meter apply full battery voltage (engine running) to PROBE and adjust R11 so that the pointer reads full scale (0° dwell).

A short comment on dwell settings is in order. Most auto manufacturers design their ignition systems for 40° (or thereabouts) dwell for 6-cylinder cars, 30° for 8-cylinder and 60° for 4-cylinder cars. This amounts to points open one-third, points closed two-thirds of the time or a duty cycle of 33⅓%. This is also the reason that the 40- and 30-degree marks coincide on the 6- and 8-cylinder dwell scales. If you're ever caught without specs for a particular auto, you will almost al-

ways be within a few degrees of correctness by setting dwell to the 40°-30° mark.

The dwell-meter section is common to the tachometer portion in that input pulses are provided to the one-shot through a high-pass filter (differentiator), R10-C3. D2 eliminates negative spikes from the one-shot. R9 is an isolating resistor. The differentiator is required because at low engine speed—or 4-cylinder operation—input pulses from across the auto's points are longer than the output of the one-shot. Without the differentiator, this would lead to inaccurate readings, since the input would tend to hold the multivibrator on for the duration of the input pulse.

The voltmeter is a basic series-resistance ammeter configuration. In the interests of economy, the voltmeter input is provided from the probe common to all other tests. The only drawback of this scheme is that when measurements are made of voltages higher than the Zener voltage of D3, D3 conducts. Dissipation in D3 is no problem for input voltages up to 15. The main disadvantage is that when  $V_z$  is reached, the voltmeter is no longer a basic 1000-ohms/volt device, because the voltage source being measured must supply current to D3 as well as to the meter. If you expect this to be a problem, a separate voltage-test lead could be provided. But the circuit as it stands has proved more than adequate for everyday automobile system checks.

When used with the meter movement specified, the values of R17 and R18-R19 provide acceptable accuracy. These are not precision resistors, just good name-brand 10% resistors.

Disassemble the meter and remove the original face. Be very careful. The movement can be ruined by excessive force on the pointer. A good rule of thumb is that *any* external force is too much. It might be a good idea to engage the help of a calm, nearsighted friend for this step. Also,

turn off fans and air conditioners, avoid drafts and try not to breathe into the works.

Copy the meter face shown in Fig. 1. Using the white dots as guides for the meter-face screw holes, rubbercement the new scale to the back of the old one. (You might want to use the 1-mA scale again some day.) To prevent wrinkles, press the new scale until it is dry, then trim the new scale to the size of the meter face. A good idea is to make a couple of copies or photostats of Fig. 1 in case the first attempt flops, or in case you want to build another unit later.

### Construction

The dimensions given in Fig. 3 should make layout easy. Of course, you may wish to "human engineer" to suit your particular needs or desires.

Circuit wiring was done on a 3" x 3 3/4" piece of perforated phenolic board with push-through terminals. Drill two holes in one end of the board, and mount it cantilever style with the meter posts as support. This arrangement has proved more than adequate; if you decide that you need added rigidity, the board can be supported at other points.

As part of the usual precautions, observe correct meter, diode and electrolytic-capacitor polarities. Pay particular attention to the basing diagram of transistors Q1 and Q2, shown in Fig. 2 along with the schematic.

### Calibration

Calibration of the dwell meter has been described; it should be re-emphasized that the meter should be set to 0° dwell *with the engine running*. System voltages are higher with the engine running, and this is, of course, when the dwell meter will be used.

Tach calibration should need be done only once, barring unforeseen circumstances such as the calibration resistors being jarred out of position.

If you have a friend with an accurate tachometer, you've got it made, as far as tach calibration goes. If you don't, another method will be described shortly, but we'll assume for now that you do have access to an tachometer and an 8-cylinder auto.

Start by setting R6, R15 and R16 to mid-range. Set S1 to either 6 or 12 volts, depending on the car; S3 to TACH 8 and S2 to 1000 RPM. Attach CL1 and CL2 to battery positive and ground, respectively, and CL3 to the low-voltage wire connecting the distributor and coil. (See Fig. 4 for a typical auto ignition schematic.) Adjust the engine idle to 600 rpm (as measured by the reference tach), and set R16 so that the meter reads 600. Now switch S3 to TACH 6 and adjust R15 so that the meter reads 800. (The reason for this is that, for a given rpm, a 6-cylinder engine generates 3/4 as many pulses across the points as an 8-cylinder engine; conversely, for a given point rep rate, the 6-cylinder engine is turning at 4/3 the speed of an 8-cylinder engine.) Switch S2 to 5000 rpm and set the idle screw so that the engine is running at say 2000 rpm. Adjust R6 so that the meter reads 2000. The tachometer is calibrated.

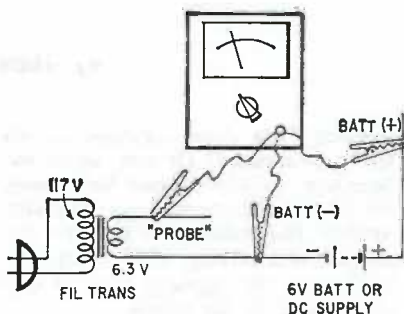


Fig. 5—Bench calibration requires only a dc supply and 6.3-volt filament transformer for 60-pulse/sec-signal source.

If you have a 6-cylinder car, calibration for the low end can be carried out as above, but with S3 set to TACH 6. Then switch to TACH 8 and adjust R16 so that the meter reading is  $\frac{3}{4}$  of the TACH 6 reading.

You can calibrate your instrument on the bench by using a 6.3-volt filament transformer and a 6-volt lantern battery, or other 6-volt dc supply, as shown in Fig. 5. For this calibration scheme, short out C3. Set S1 to 6 volts, and then plug in the transformer and turn the tach on, having preset R6, R15 and R16 to mid-range. Set S2 to 1000 RPM, S3 to TACH 8 and adjust R16 for a meter reading of 900 rpm. Then switch S2 to 5000 RPM, and adjust R6 so that the meter reads 900 rpm in the high scale. Now switch S3 to TACH 6 and adjust R15 so that the meter reads 1200 rpm. The tach is now calibrated, having never left the bench.

If you're concerned about the accuracy of the voltmeter with the 10% components specified, you can substitute appropriate adjustable resistors of the type used for R6, R15 and R16, and calibrate the voltmeter to greater

accuracy, using an appropriate reference scheme. The cost of two more adjustable resistors of this type will not increase total cost of the instrument by more than about 60 cents.

### Use

The instrument is designed for use on negative-ground systems, since almost every auto made today is wired this way. The dwell meter, however, can be used on positive-ground systems by simply reversing the roles of PROBE and NEG (ground) leads.

If you own an auto with positive ground and want to build this instrument to help in tuning it up, all is not lost. Simply reverse all diodes, the meter and the electrolytic capacitor. Substitute pnp transistors for Q1 and Q2, but be sure they have an  $h_{FE}$  of 100 or more. Be sure you remember that, in this configuration, PROBE is connected to negative when making voltage measurements.

In use, the PROBE and GROUND leads are used for all tests, and the (+) lead is used only for tachometer measurements.

## SOLID-STATE GAUGES

by JACK SADDLER

HAS YOUR CAR EVER OVERHEATED AND left you stranded? Or how about the time you ran out of gas? Both times the dash indicators read o.k. Probably neither the sending units *nor* the gauges were wrong. Most likely, the culprit was the regulator used to supply voltage to the gauges.

When something is wrong with the gauging system, usually both gas and temperature gauges will be influenced. The instrument voltage regulator is a small device mounted in

a plug-in socket behind the instrument panel. This regulator is an electromechanical unit and can be inaccurate.

### How most gauges work

A typical setup for indicating temperature and gas tank contents in a modern automobile is shown in Fig. 1. The sending unit in the gas tank is a potentiometer with its wiper arm attached to a float. Usually an empty tank will provide a 60–65-ohm re-

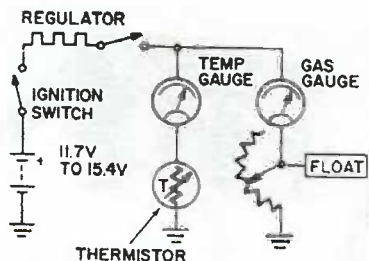


Fig. 1 Arrangement for gas and temperature gauges in most cars today.

sistance reading, while the full reading is 10–11 ohms.

The temperature sensor in Fig. 1 is a thermistor (temperature-dependent resistor) mounted in a brass cylinder. It is located in the water jacket of the engine close to the thermostat. A cold resistance reading would be 70–75 ohms, and the resistance just before the water boils would be about 9 ohms.

The gauges are hot-wire milliammeters. This meter type consists of a bimetallic strip connected to the pointer. The bimetallic element is wrapped with insulated heater wire. Voltage applied to the heater causes the bimetallic strip to bend; the amount of bending is proportional to the current.

The regulator contains a bimetallic strip heated by the passage of current through a separate heater wire wrapped around it. As the bimetallic strip heats, it bends because the two metals have different expansion rates. When a predetermined amount of current has passed through the winding, the contact is broken. When the bimetal strip cools, it makes contact again and the cycle is repeated.

The ratio of the time *on* to the time *off* determines the effective value of the resulting voltage. It is impossible with any easily available test instrument to determine the output

voltage of this regulator. Auto manufacturers' service manuals usually recommend checking with known-good gauges and sending units.

Input voltages to the instrument regulator vary widely. With the engine turning over at 2000 rpm on a cold morning the output from the alternator may be as high as 15.4 volts. The voltage may drop to 11.7 on a hot summer day.

Each panel gauge has a resistance of about 11.5 ohms. The highest load on the regulator occurs when the engine is hot and the tank is full. The current drains from the fuel-gauging and temperature-sensing system total about 500 mA.

When engine water temperature is low and the gas tank almost empty, the regulator consumes the most power internally but has to put out the least. Total drain for both systems in this case is 127 mA.

### Zener-resistor

The simplest solid-state replacement for the electromechanical regulator is shown in Fig. 2. It consists of a shunt regulator made from a resistor and a Zener diode.

The resistor-Zener combination is correct for most U.S.-made autos except those made by General Motors. It was specifically designed for American Motors cars made from 1961 to 1967. Formulas for calculating values for other voltages and current requirements are given later.

The first step is to open the old regulator. The metal case is crimped around a phenolic board (Fig. 3). The best tool to use for uncrimping

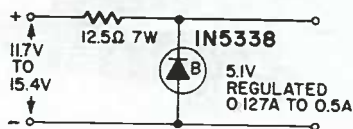


Fig. 2 Simple shunt regulator replacement for electromechanical devices.

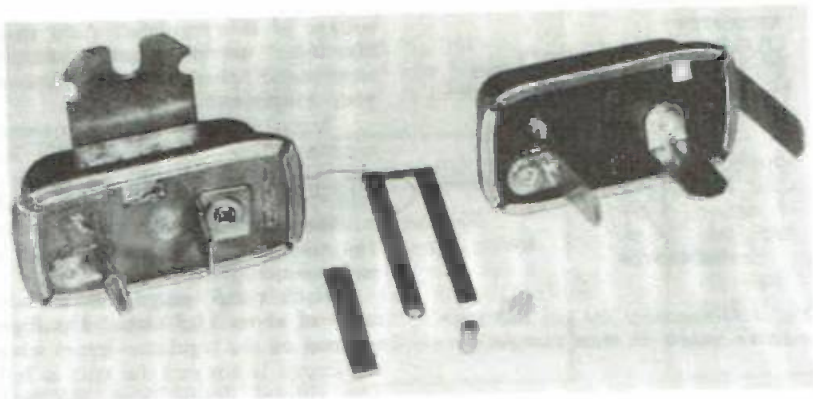


Fig. 3 Typical cases for bimetallic-strip regulators with components.

is a pair of diagonal cutters. The metal lip is loosened all around. After you open the case, bend the bimetallic element up and clip it off even with the terminal. Remove the silver-tipped brass contact blade similarly. Then back out the adjusting screw. All three of these parts can be discarded.

Now solder a 12.5-ohm, 5-7-watt resistor between the two terminals (Fig. 4). The input from the ignition switch is marked IGN. The output terminal is unmarked.

Drill a small hole in the metal cover near the spot-welded ground



Fig. 4 Photo shows positioning of resistor-Zener regulator combination.

strap. The negative end of the Zener diode is threaded through this hole. Check to see that neither of the resistor leads nor the positive Zener lead touches the case.

Complete the assembly by re-crimping the can with pliers. Bend the Zener diode lead parallel to the can and solder it. The original can is tin-plated and easy to solder. For the final test, connect the positive terminal of a 12-volt battery to the IGN terminal and the negative lead to the case. A voltmeter connected between the unmarked terminal and case should read 5.1 V.

After assembly and test, merely plug the solid-state regulator in place of the electromechanical unit. No alterations are required in the wiring.

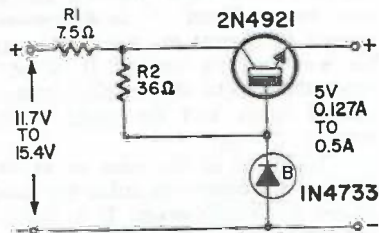


Fig. 5 Transistor-Zener circuit.





Fig. 6 (photo)—Layout for transistor regulator in tape-insulated mounting case.

### Transistor-Zener regulator

The simple resistor-Zener unit has a great advantage over electro-mechanical unit: with no moving parts, it is quite reliable. Its main disadvantage is that it consumes more power than the bimetallic unit. And since it must regulate for all possible

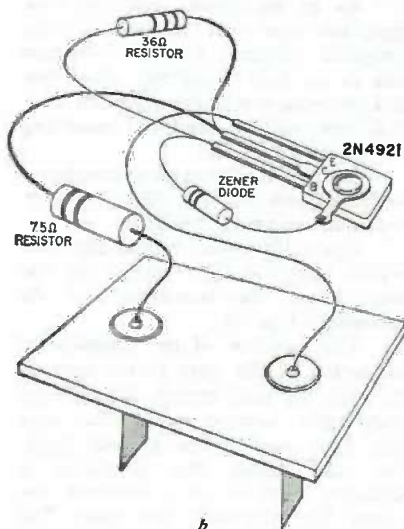
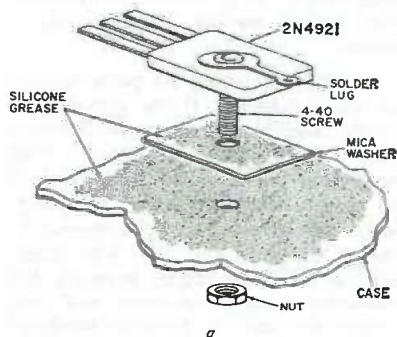
engine-temperature and fuel-tank conditions, it is designed to deliver the regulated voltage under all conditions.

This means some power will be wasted under most conditions. The Zener unit consumes about 8 watts in the worst case. When things are going well with the car there's no problem. But there are times when even the power consumption of the transistor radio is bad. For this reason the second circuit in Fig. 5 is shown.

While this system is slightly more complex, it is a more efficient regulator. It consumes only about 2 watts when the input voltage is low and output current demands are at a minimum.

How does it work? The transistor is in series with the positive supply line. It acts as a variable resistor. The Zener diode and R2 together act like the sliding contact on a potentiometer. As the supply voltage rises, more current flows through R2. The Zener diode and the transistor base

Figs. 7 a,b—Drawing shows transistor isolation and connections for the components.



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$$\text{Min. output current} = \frac{\text{regulator voltage}}{(\text{empty tank resistance} + \text{gauge resistance})} + \frac{\text{regulator voltage}}{(\text{cold thermistor resistance} + \text{gauge resistance})}$$

$$\text{Max. output current} = \frac{\text{regulator voltage}}{(\text{full-tank resistance} + \text{gauge resistance})} + \frac{\text{regulator voltage}}{(\text{hot-thermistor resistance} + \text{gauge resistance})}$$

$$\text{Series resistance} = \frac{\text{min. battery voltage} - \text{regulator design voltage}}{1.1 (\text{max. output voltage})}$$

$$\text{Resistor pwr. rating} = (15.4 - \text{regulator voltage}) (\text{max. output current} + 0.1)$$

(The power rating will be conservative.)

$$\text{Max. Zener current} = 1.1 (\text{max. output current} - \text{min. output current})$$

$$\text{Zener power rating} = \text{regulator voltage} (1.1 \text{ max. current} - \text{min. current})$$

$$\text{Zener voltage} = \text{regulator design voltage}$$


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split this current to maintain the output voltage at 5 volts. The ability of this circuit to maintain a constant output voltage is even better than the Zener alone. The control effect of the Zener is transferred to the transistor so that output voltage is rock-solid regardless of changes in input voltage or output current.

As in the Zener regulator, first open the case and remove the old bimetallic element. Drill a 7/64-inch hole in the case along the center line at a convenient distance from the end. This hole will be used for mounting the transistor.

Before mounting the transistor, its leads must be bent away from its heat sink at about a 30° angle.

Cover the inside of the can with plastic electrical tape except for the area where the transistor will be mounted (Fig. 6).

The collector of the transistor is connected to the bare metal spot on its case. In this circuit, it must be electrically isolated from the heat sink (the case of the plug-in unit). For this reason the transistor is mounted with a mica insulator between the transistor and case. The

insulator is supplied by the manufacturer. Use silicone grease on both sides of the insulator to provide a good heat path to the case.

The mounting is completed with a solder lug atop the transistor as shown in Fig. 7-a. The connections are shown in Fig. 7-b. When soldering the leads and wire to the transistor, make sure the leads are freshly tinned. Clamp the transistor lead with diagonal pliers or an aluminum clip-on heat sink. The silicon transistor is quite rugged, but it's best not to take a chance of damaging it by overheating. The 7.5-ohm resistor is an IRC type BWH. It is 1-watt composition size but is capable of dissipating 2 watts.

After soldering all parts together, close the case. If the unit is correctly assembled, an ohmmeter should read infinity with its negative lead connected to IGN and positive lead to case. Reversing the ohmmeter leads will give a low reading—something between 100 and 350 ohms. Similar readings taken between the unmarked output terminal and case should be identical to input readings.

### Zener regulator with other cars

The resistor-Zener combination shown in Fig. 2 is designed specifically for 1961 through 1967 Rambler cars. It is also useful with Fords and Chryslers. Some cars measure oil pressure electrically and the regulator must supply the current. It is best to review wiring diagrams to determine what circuits are connected to the regulator. If your service manual shows different resistance values for the sending units or the meters, jacket and gas tank, here are the measurements needed:

1. Normal voltage output from instrument regulator.
2. Empty-gas-tank sending-unit resistance.
3. Full-gas-tank sending-unit resistance.
4. Cold-engine sending-unit resistance.
5. Hot-engine sending-unit resistance.
6. Gauge resistance.

And here are the calculations:

(The 1.1 is an empirical safety factor used to assure correct performance of the Zener diode when it draws minimum current.)

Generally the Zener diode can be mounted by its leads. In my system the diode is capable of dissipating 5 watts at all temperatures to 167° F

with the case  $\frac{3}{8}$  inch from the solder joint. The 1N5333 series is available with voltages from 3.3 to 200 volts. Should more dissipation be required from the Zener, the 1N3993R series can be used. In this series, the 1N3996R is the 5.1-volt Zener. These are stud-mounted units and can be fitted inside the case. It's snug-fit with the resistor, which should be as small as possible.

### Transistor-Zener calculations

The 2N4921 transistor is quite adequate for any system requiring 8 volts or less. All calculations below are based on the use of this transistor. Measurements in the car or from a service manual are the same as for the Zener unit. Minimum and maximum currents required by the sending units are also the same. To make calculations easy some elements of the equations are combined.

$$\text{Zener voltage} = \text{required output voltage} + 0.6 \text{ V}$$

The 0.6 volt is the maximum voltage drop from collector to emitter for the transistor used.

$$RI = \frac{E_{\text{min}} - E_z - 2.6}{I_{\text{maz}}}$$

Table I—Zeners for Output Voltages

Desired output voltage	Zener voltage	Type	Max. current amps	Resistance ohms	Factor*
3.0	3.6	1N4729	0.252	10.0	1.38
3.3	3.9	1N4730	0.234	9.0	1.28
3.7	4.3	1N4731	0.217	9.0	1.16
4.1	4.7	1N4732	0.193	8.0	1.06
4.5	5.1	1N4733	0.178	7.0	0.98
5.0	5.6	1N4734	0.162	5.0	0.90
5.6	6.2	1N4735	0.146	2.0	0.82
6.2	6.8	1N4736	0.133	3.5	0.74
6.9	7.5	1N4737	0.121	4.0	0.68
7.6	8.2	1N4738	0.110	4.5	0.62
8.5	9.5	1N4739	0.100	5.0	0.56

\*This factor combines several design factors and is useful only with the 2N4921 transistor. The complete design calculations are in the Motorola Zener Diode Handbook.

The 2.6 volts is composed of 0.6 volt for the collector-emitter voltage drop, and the 2.0 volts is a collector-base voltage that will keep the transistor out of saturation.

$$R2 = \frac{52}{I_{max} + \text{factor from chart}}$$

$$I_z = \frac{E_{max} - E_z}{R2 + R_z}$$

where  $E_{min}$  is minimum input voltage

$E_{max}$  is maximum input voltage

$E_z$  is Zener voltage

$I_{max}$  is maximum output current

$I_{z\ max}$  is maximum Zener current

$R_z$  is Zener resistance (see Table 1)

The maximum current through the Zener diode must be calculated

to insure its power rating is not exceeded. The maximum currents shown in Table I limit dissipation of the Zener to about 1 watt. The 1N4728 series is capable of handling twice the current shown if the leads are less than  $\frac{3}{8}$  inch long and the heat sink is adequate.

The only remaining calculations necessary are the power ratings of the two resistors:

Power rating,  $R1 = (I_{max} + 0.1 I_{z\ max})^2 R1$

$R1$

Power rating,  $R2 = (I_{z\ max})^2 R2$

Either of these regulators in your automobile will give you more confidence in the gauge readings. You might save a long walk to the gas station or a costly repair bill.

## ELECTRONIC IGNITION

by BENNETT C. GOLDBERG & RAYMOND G. WILKINS

DURING RECENT YEARS A NUMBER of transistorized automobile ignition systems have appeared. The majority of these systems (manufactured units and construction projects) had malfunctions after a few thousand miles of engine use due to exceeding germanium power transistor capabilities and poorly designed circuits.

The detrimental effects on germanium transistors occur with continuous operation at junction temperatures of 100°C. Temperatures in the engine compartment can range from +70° to +110°C or higher on hot summer days.

### Silicon transistors

To beat the temperature problem, two transistor ignition systems were developed: a negative-ground system for American automobiles, and a positive-ground system for some foreign automobiles. A reliable high-voltage silicon power-switching transistor, having excellent beta with large collector currents at cold and hot temperature extremes, was selected. Silicon transistors have fast switching times and a junction temperature range in excess of +125°C with the proper heat sink.

These units were "worst case" designed for high reliability and lab-

oratory-tested at ambient temperatures  $-55^{\circ}$  to  $+125^{\circ}\text{C}$ .

The negative-ground system has been in operation in a 1963 6-cylinder 85-h.p. Ford Falcon and a 1965 8-cylinder 325-h.p. Chevrolet. The systems have exceeded 10,000 miles without malfunction.

### Circuit operation

On the negative-ground system (Fig. 1), current flows through R3 to the base of Q1 when the ignition breaker points are closed. This saturates the transistor and current flows through R4 to the base of Q2. Transistor Q2 then saturates and its collector current flows from the +12-volt battery terminal through ballast resistor R1 and transformer T1. Energy equal to  $\frac{1}{2} LI^2$  (where L is the inductance of the transformer primary and I is the current flowing in the transformer primary) is stored in the T1's primary.

When the points open, Q1 no longer receives base drive, turning it off, and Q2 is cut off when it loses base drive from R4. Energy is stored

in T1's primary until it reaches the breakdown voltage of Zener diode D2. This flux change in the transformer builds up a voltage in the secondary. The secondary voltage is the ignition pulse which jumps the spark plug. Ballast resistor R1 sets the primary current to the required amount. Zener diode D2 limits the voltage across transistor Q2 to a safe 100 volts.

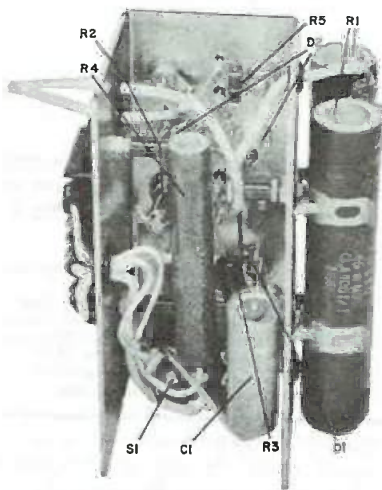
During normal running, the battery is fully charged to 14-15 volts. Under these conditions, the ballast resistor is set to give a peak coil current of approximately 12 amps. When the car is starting, the battery voltage drops to 8-10 volts because of the starter load. This reduced battery voltage could mean less coil current and therefore a weak spark.

Therefore diode D1 is added to bypass the ballast resistor during engine cranking. This maintains the desired peak coil current and a full spark.

On the positive-ground system (Fig. 2), when the points are closed current flows through R3 to the base of Q1. This saturates the transistor, and current flows through R1, Q1, R4, R5, R6 at the primary of the transformer T1. When the points open the transistor no longer receives base drive. Current flow through the transistor is interrupted, and the operation is the same as the negative-ground system, including R1, D1 and D2.

### Construction

The unit is built in a  $2\frac{1}{2} \times 2\frac{1}{2} \times 5$ -inch aluminum box. The heat sink for D1, D2, Q1 and Q2 is the outside top of the box (see photo), D1, D2, Q1 and Q2 must be electrically insulated from the heat sink and ground. This is done by mounting the components with their insulating hardware and applying silicone grease on the metal surface. The grease aids in conducting the heat away. Arrangement of the components inside the box is



*Inside the ignition system there's plenty of room to fit all parts without cramming.*



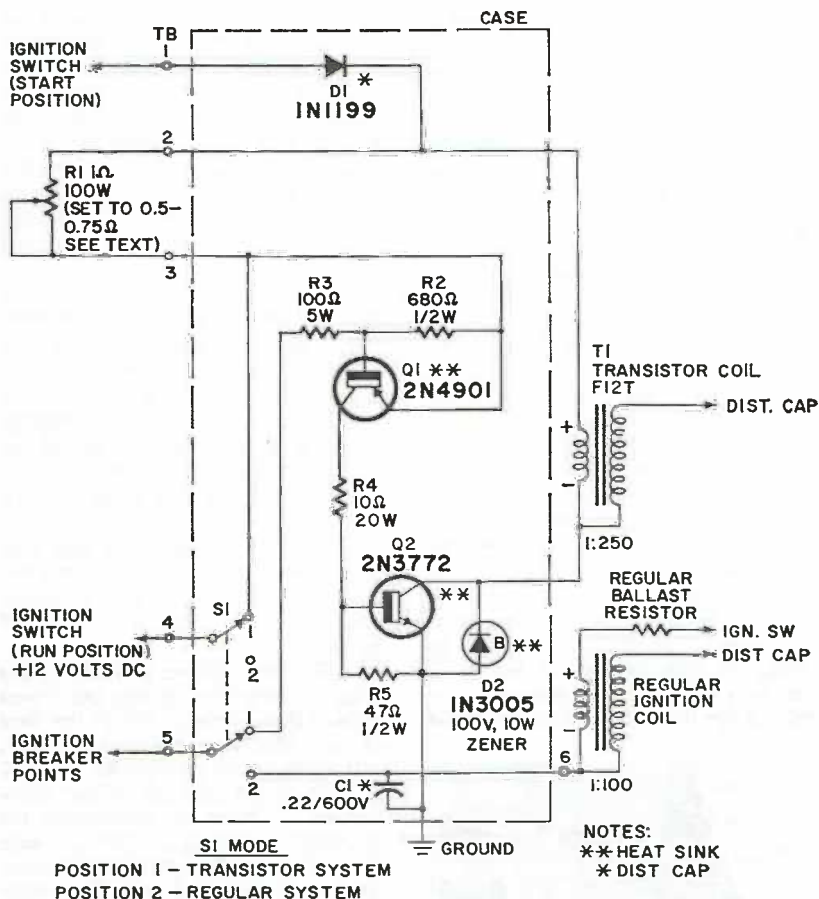


Fig. 1—This circuit is the one to use if you are building an ignition system to use with a negative-ground car. This includes almost all American made automobiles made in recent years. If you have a car with a positive-ground system try the circuit in Fig. 2.

#### PARTS LIST (Negative Ground)

- R1—1-ohm 100-watt adjustable resistor Clarostat VK100NA or equal
- R2—680-ohm 1/2-watt  $\pm 10\%$
- R3—100-ohm 5-watt  $\pm 5\%$  Clarostat VPR5F or equal
- R4—10-ohm 20-watt  $\pm 5\%$  Clarostat VPR-20H or equal
- R5—47-ohm 1/2-watt  $\pm 10\%$

- C1—22  $\mu$ F 600-volt ignition-point capacitor (see text)
- CR1—1N1199 50-volt, 12-amp silicon diode
- CR2—1N3005 100-volt, 10-watt Zener diode
- Q1—2N4901 pnp silicon transistor (Motorola)
- Q2—2N3772 npn silicon transistor (RCA)
- T1, S1, MISC—See positive ground TO-3 Power Transistors, 2 req.

not critical except D2 should be mounted near Q2.

Capacitor C1 (negative-ground system) should be the same value as the capacitor in the distributor next to the

ignition breaker points. It may be placed inside the box or at terminal TB-6. The original breaker-point capacitor may be used later after the transistor ballast-resistor R1 is ad-

justed. This adjustment will be discussed in more detail later.

Do not remove the capacitor in the positive-ground system. The time constant of R3 and the capacitor is

small and has little effect on the input trigger pulse.

Ignition coil T1 is mounted on the engine firewall as close to the distributor as possible. Number 14 wire

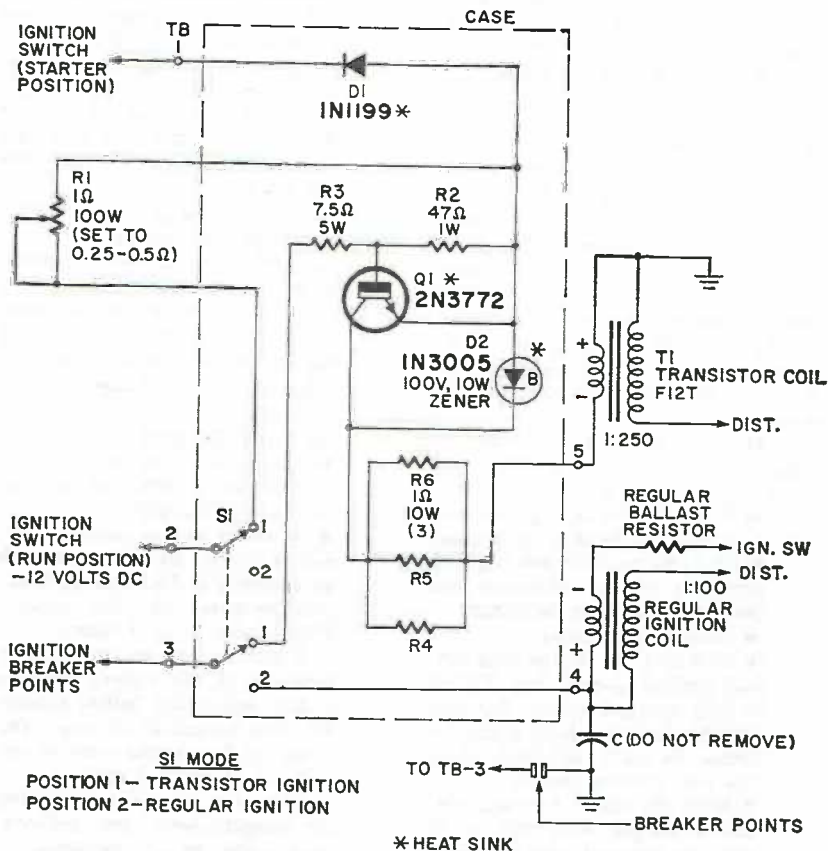


Fig. 2—Here's a circuit for an ignition system for a positive-ground car. This covers many European makes and some older U.S. models. Note that in this version, circuit ground floats. If your car has a positive ground, the positive battery terminal will be connected to the frame of the car. A glance under the hood is all you need.

#### PARTS LIST (Positive Ground)

- R1—1-ohm 100-watt adjustable Clarostat VK100NA or equal
- R2—47-ohm 1-watt  $\pm 10\%$
- R3—7.5-ohm 5-watt  $\pm 5\%$  Clarostat VPR5F or equal
- R4, R5, R6—1-ohm 10-watt  $\pm 5\%$  Clarostat VPR10F or equal
- DRI—IN1199, 50-volt, 12 amp silicon diode

- CR2—IN3005 100-volt, 10-watt Zener diode
- Q1—2N3772 npn silicon transistor (RCA)
- T1—Transistor ignition coil, 1:250T, Mallory F12T
- SI—dpdt 15-amp 125-volt toggle switch, Arrow-Hart 82609
- MISC—Barrier terminal block, Cinch Jones 6-142. Mini-Box  $2\frac{1}{2} \times 2\frac{1}{2} \times 5"$ , Bud CU-3004A. transistor mounting kit, Motorola MK-15. (For TO-3 power transistor, 1 req.)

was used in all external wiring to the coil, ignition switch and starter. It is important that good grounds be used and all connections are secure.

The ignition-system package was mounted on the engine firewall. It could be mounted in another convenient location, preferably away from the exhaust manifolds. Tests had indicated it was not necessary to mount the package near the radiator or fan, thus simplifying installation.

The change-over arrangement from a conventional system to the transistor ignition is simple. To operate in the conventional mode, S1 is placed in position 2 and the center high-voltage distributor lead is plugged into the conventional coil. To operate in the transistor mode, place switch S1 to position 1 and plug the center high-voltage distributor lead into the transistor coil.

Here is a seven-step sequence for adjusting your negative ground transistor ignition system:

- With the transistor ignition system bolted to the firewall, connect wires to the positive and negative terminals of the transistor coil (**no wires to other terminals**).
- Place S1 in position 1.
- Start and run engine with normal ignition system until battery is fully charged. (Note: No connections have yet been made between the car's electrical system and the terminal strip.)
- With the engine running, connect a jumper wire with a 15-amp dc ammeter and ballast-resistor R1 (set to 1 ohm) in series between TB-4 and the positive terminal of the battery.
- Connect TB-5 to ground and adjust ballast-resistor R1 for a current reading of 12 amps. This

will set R1 to the correct value, which should be 0.5–0.75 ohm.

● Stop the engine, remove jumpers, etc., and connect wires to all terminals; remove the distributor capacitor (negative system only), and place it in the transistor box or at TB-6.

● Place the center high-voltage lead from the distributor to the transistor coil. Make sure switch S1 is in position 1. Your car will now start and run in the transistor mode.

This is the adjustment procedure for positive-ground systems:

- Bolt the transistor ignition system to the firewall. Connect a wire from TB5 to the negative terminal of transistor coil T1 and the positive terminal to frame (no other wires to terminals).
- Place S1 in position 1 (transistor ignition mode).
- Start and run engine with normal ignition system until the battery is fully charged.
- With the engine running, connect a jumper wire in series with an ammeter to TB2 and the negative terminal of the battery. (Set resistor R1 to 1 ohm.)
- Connect TB3 to the positive terminal of the battery or auto frame, and adjust ballast resistor R1 for a current of 12 amps. The value of R1 should now be between 0.25 and 0.5 ohm.
- Stop the engine and remove the jumpers, meter, etc., and connect wires to all terminals as shown in Fig. 2. (Do not remove the distributor capacitor.)
- Place the center high-voltage lead from the distributor to the center hole of the transistor coil. Switch S1 to position 1.

# ROAD ICING ALARM

By JAMES E. PUGH, JR.

Relative humidity near 100% combined with a temperature near freezing or below can make driving or flying very dangerous, since these are the conditions that can cause unexpected formation of ice. Anyone can see ice forming on his windshield when conditions are *beyond* the critical range, but when both humidity and temperature are *near* the critical point, one may not realize that ice is forming just a short distance beyond.

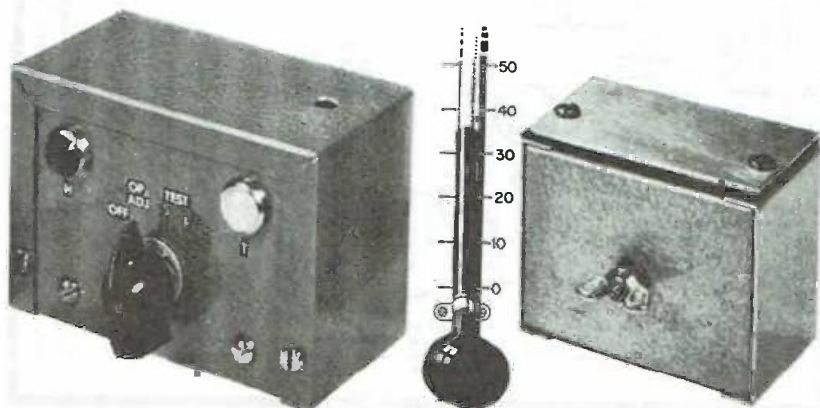
The icing-condition indicator (ICI) described here is intended to keep the driver alerted to critical moisture and temperature conditions, individually and in combination. Construction is simple and easy, the cost is reasonable, and once installed and adjusted, it is stable over a wide temperature range and should require no further attention.

## How it works

Humidity sensing element R3 (see schematic) is a sensitive resistive element (humistor) that increases in resistance as the ambient moisture increases. It controls the collector current in common-emitter dc amplifier Q1 by controlling the base-to-emitter resistance. As the humidity increases, R3's resistance increases, Q1 base-to-emitter voltage increases, Q1 collector current increases, and indicator lamp LM1 lights.

Potentiometer R2 (LEVEL H) sets the lamp brightness to a suitable level during critical humidity conditions, and diode D1 maintains a constant Q1 amplification over a wide ambient temperature range.

Inverse feedback is provided by connecting the positive end of the base bias network to the collector of Q1.







## PARTS LIST

C1, C3—10 $\mu$ F, 6-volt electrolytic capacitor  
 C2, C4—50 $\mu$ F, 6-volt electrolytic capacitor  
 D1, D2—5A4 silicon diode, International Rectifier Corp  
 D3—1N1518 Zener diode, 3.9-volt, 1-watt, IRC  
 LM1, LM2—Type 49 lamp  
 Q1, Q2—2N3242-A silicon transistor, NPN  
 R1, R7—1.5k ohm, 1/2-watt resistor  
 R2, R8—3.0k ohm potentiometer, linear taper  
 R3—Humistor, DeVry Hygropak HA-26  
 R4, R10—680 ohm, 1/2-watt resistor  
 R5, R11—7.5 ohm, wire wound resistor, IRC type BWH  
 R6—1.0k ohm potentiometer, linear taper  
 R9—Thermistor, 200 ohm @ 25° C, Fenwal KB 22J1

R12—470 ohm, 1/2-watt resistor  
 R13—15 ohm, 2-watt wire wound resistor (6-volt operation)  
 50 ohm, 5-watt wire wound resistor (12-volt operation)  
 S1—3-pole, 6-position rotary switch, shorting, Mallory 3136J  
 2—miniature lampholders, Leecraft 7-20  
 2—lens, 11/32" diameter, Dialco 514  
 1—knob, 1/4" shaft  
 1—4 x 2 x 2 3/4 aluminum box, PMC 1015  
 1—2 3/4 x 2 1/8 x 1 1/8 aluminum box, PMC 1000  
 1—4-wire cable, 22 AWG stranded, 7 x 30, Belden 8444  
 2—6-terminal miniature tie points  
 wire, grommets, solder, plastic strip, tube socket, decals, etc.

This improves the overall circuit stability and, combined with a silicon transistor, decreases the transistor warmup time to a negligible interval. Resistor R4 is used for testing the humidity portion of the indicator.

The temperature sensing portion of the circuit is the same as the humidity section, except that thermistor R9 senses temperature. Since the thermistor resistance increases as the temperature drops, LM2 lights when the temperature drops to the critical range. Thus, LM1 lights steadily when the humidity nears 100%, and LM2 lights steadily when the temperature nears freezing.

In addition, to provide a distinctive indication when *both* conditions are nearing the critical point, capacitors C2 and C4 were added to form an astable multivibrator circuit. The GAIN control, R6, sets the multivibrator operating point to cause alternate flashings of LM1 and LM2 when the humidity reaches approximately 95% any time that the temperature is about 35° or below. Resistor R12 is made approximately equal to R6 at the starting point of the multivibrator. This gives approximately equal lamp intensity when they are flashing.

A Zener diode, D3, is used to maintain the operating potential approximately constant at 3.9 volts with normal variations in battery voltage. The input voltage can be any dc supply

desired, provided that a suitable dropping resistor (R13) is used. The 15 ohm value of R13 given is for a 6-volt supply voltage. If a 12-volt battery is used, increase R13 to 50 ohms (5 watts) and mount it outside the case.

As shown in the photos, bolt the humistor inside the top section of the box with an intervening block of foam rubber. Make the top clamp for a section of thin plastic, in which a 5/8" x 3/4" window is cut to permit unrestricted air flow to the humistor. Fasten clips removed from a 7- or 9-pin miniature tube socket to the plastic to connect to the humistor pins. Solder 2-inch flexible leads to each clip before mounting to prevent melting the plastic clamp. Use a 4-terminal tie point to mount the thermistor and to make cable connections to the control unit. Solder the thermistor to the tie point last to minimize the chance of heat damage.

Drill about 20 small holes (about 1/16" diameter) in both sides of the bottom section of the box to permit a free flow of air to the sensing elements. If the box is to be mounted where rain will strike it, make a small metal plate to mount above the vents on the top side. This will allow air to flow in the bottom vent and out the rear slot formed by the box top and the plate.

A machine screw can be mounted in the back of the box, or in any other suitable location, for mounting the

sensing unit to the vehicle. After construction is completed, cover all openings with plastic tape. A rubber gasket can be used between the top plate and the outer edge of the box. Mount the unit where air can flow over it freely, but preferably where water cannot strike it directly. Also, if dusty roads are encountered frequently, mount the unit in a sheltered place to minimize contamination of the humistor.

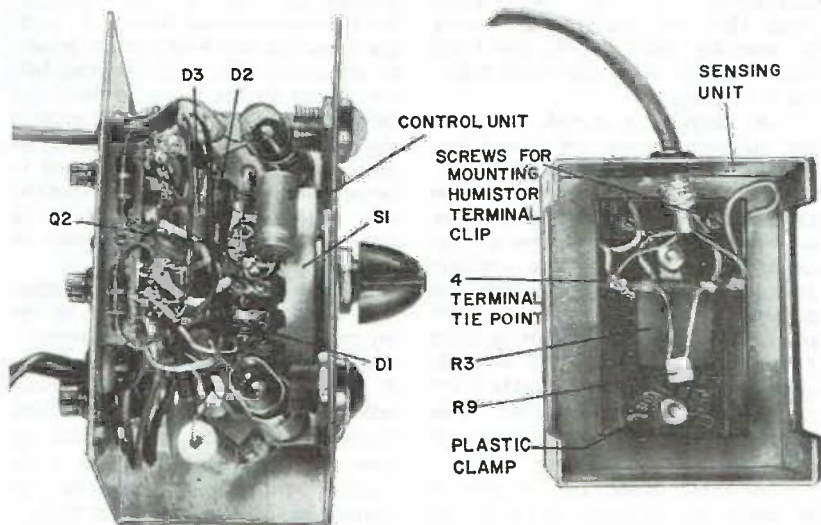
Make the 4-wire cable connecting the sensing unit to the control unit long enough to reach the control unit mounting position easily. Allow plenty of extra length for routing around obstacles in the engine compartment. Make sure that the cable is located where it won't interfere with any maintenance or repair work on the engine. Wrap the cable with thin cord just inside the sensing unit box to prevent strain on the tie point.

All parts except the sensing elements are mounted in the control box which will normally be located inside the vehicle. The metal box listed is

suitable where mounting space is limited, but if you have plenty of space, a larger box can be used to make wiring easier. After drilling all holes and mounting the switch, lamp holders, and potentiometers, make two subassemblies on 6-terminal tie points, using the parts shown. Q1, Q2, C4, R5, and R11 can be soldered to unused switch contacts. Point-to-point wire all other parts, and install the subassemblies. Be sure to insulate all bare leads.

After mounting the control unit in the vehicle, connect the 4-wire cable from the sensing unit to the terminals provided on the top tie point. Then connect the positive and negative leads to the vehicle battery via the chassis and the ignition switch. Make sure that the "hot" lead from the control unit is connected to the coil side of the ignition switch as shown.

An "OFF" position is provided on the control unit switch, but it does not disconnect the supply voltage. It simply provides an idle position by grounding the bases of both transistors



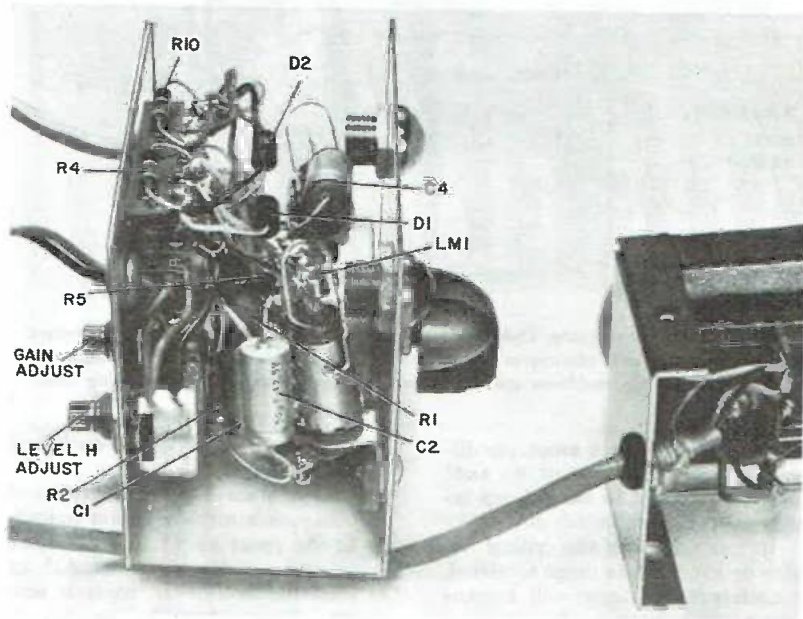
*Take a close look at the inside of the sensing and control units. Match this parts placement as closely as possible and you will have little trouble building the alarm.*

to reduce the collector current to the minimum. Wire potentiometers (R2, R6, and R8), so resistance decrease when the shaft of these controls is rotated clockwise. This causes lamp brightness to increase and flashing to start with clockwise rotation, making adjustments easier.

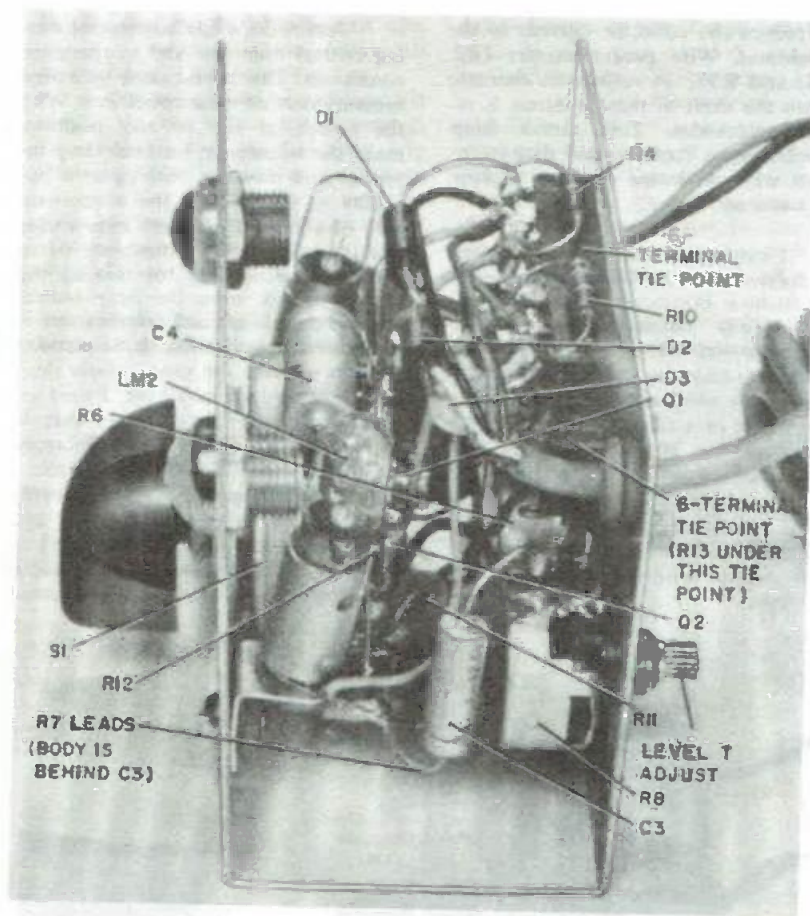
First make a coarse adjustment as follows: Set the three potentiometers to their maximum resistance position (fully counterclockwise). Set control switch S1 to TEST S (steady) and then rotate the LEVEL H and LEVEL T controls clockwise until the lamps are visible in a slightly darkened area. (A vtvm or a 20,000 ohms/volt vom connected across LM1 or LM2 should indicate about 0.85 volt at a suitable lamp intensity.) Now set the switch to TEST F (flash) and carefully rotate the GAIN control until the lamps just start to flash.

Next make a fine adjustment during critical humidity and temperature conditions. This is best done by actual measurement of these conditions. With the switch at the OP/ADJ position, make the adjustment for even lamp intensity and then for flashing at a humidity of about 95% and a temperature of about 35° F. Any hygrometer and thermometer of moderate accuracy will be suitable for making the measurements, since the lamp indication of humidity and temperature is not a precise indication. It is intended only to indicate the *range* where dangerous conditions may exist. I used an inexpensive slide hygrometer that is available at Lafayette Radio Electronics (Stock No. 99C9006) for \$2.69.

After making the fine adjustment, check that the lamp intensity is approximately the same at the TEST F and OP/ADJ positions during critical temperature and humidity conditions.



*Another view of the control unit. This time we're looking at it from the humidity circuit end. Obviously, if you fit a larger case in your car, it will make the construction job considerably easier as it will leave more room for parts and avoid squeezing.*



*Here's that control unit again. This time from the opposite end where the temperature circuit is located. Parts placement is not critical, but if you follow the arrangement here, your assembly problems will be eased and you can reduce assembly time.*

If there is more than a moderate difference, change the value of R4 and/or R10 as required to make lamp intensity equal at both switch positions.

In use, as either the critical humidity or temperature range is neared, the corresponding lamp will become brighter. When both conditions reach the critical point simultaneously, the lamps will start flashing at a greater intensity and will continue to do so until

either or both conditions drop below the critical range.

Note: Since it is not always possible to simultaneously obtain humidity in the range of 95 to 100% and temperature in the range of 32° to 35°, set the two LEVEL controls separately. Then when the weather is suitable, the GAIN control can be set for flashing. If a suitable hygrometer is not available, a satisfactory adjust-



ment of the humidity indicator can be made when driving through a light fog. Do not try this during rain or snow,

however, because the humidity is often considerably less than 100% at such times.

## AUTOMATIC WINDSHIELD WIPER PAUSE CONTROLLER

By S. B. GRYNKEWICZ

LIVES THERE A DRIVER WHO HASN'T encountered this: A light mist hangs in the air. Traffic is just heavy enough so that those @#!90-drivers-in-a-hurry periodically whizzing by you leave a thin coating of dirt on your windshield.

If you're like me, you hate to see those wiper blades flicking back and forth, wearing away, and giving a grit-polish to an expensive windshield. Turning the wipers on and off and reaching for the knob every few minutes didn't appeal to me, so I decided to let electronics do the job.

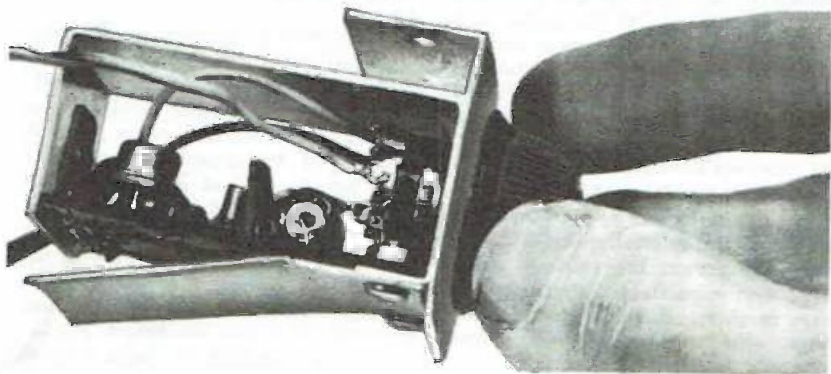
With this Automatic Pause Controller (APC), you can operate the windshield wiper at its normal speeds in a normal manner (no pause between sweeps) or set the control knob to obtain a desired pause between sweeps . . . 1 second, 2 seconds . . .

15 seconds . . . It's the pause that makes the difference. However, each sweep occurs at normal speeds.

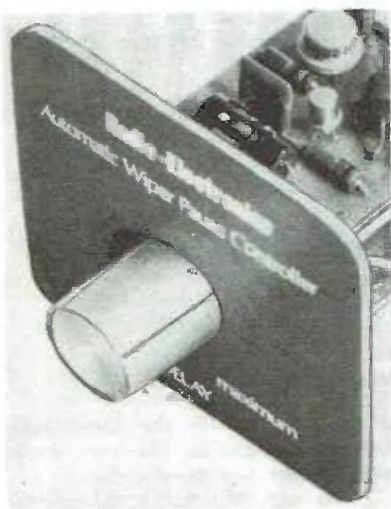
The unit will not operate with vacuum-operated wipers. Electric wipers have been in vogue for the past 15 years, and unless you have an older car the chances are you can use APC. Federal regulations require that all cars built after 1967 have two-speed wiper systems.

All windshield-wiper motors have a built-in cam-operated "park" switch (S3 in Fig. 1) to keep the wiper motor running until the wiper blades return to their park position; even after the wiper switch on the dashboard is turned off. It's this cam-activated switch that makes the APC possible.

Two silicon semiconductors, a unijunction transistor (Q1) and a sili-







*Mechanical layout of the automatic wiper-pause controller is not critical. See variation above. Build it to fit your car.*

con-controlled rectifier (SCR1) are used in the APC. The SCR, which switches the wiper motor on, is triggered by preset pulses from relaxation oscillator Q1. Once the SCR conducts, it stays on until its anode voltage is zero with respect to its cathode.

Operation of the relaxation oscillator is simple. When power is applied to the APC through S1 (S2 off), C1 begins to charge to the supply voltage through R1 and R2. The charge across C1 also appears across the emitter and base 1 of Q1. When the charge reaches a critical level, it drains off through Q1's emitter, base 1 and R3 until the emitter drops enough to stop conducting. (The critical level depends upon the transistor's characteristics.) Capacitor C1 recharges and the cycle is repeated. Value of R1 and C1 and the setting of R2 determine the charging time. The greater the value of resistance and capacitance, the longer it takes to charge the capacitor and the longer the pause between wiper sweeps.

When Q1's emitter voltage goes positive enough, current flows through R3, Q1's base 1 and base 2, and R4. The greater the current flow, the greater the voltage drop across R4. This voltage is also across the SCR's gate and cathode. When the voltage drop across R4 reaches a sufficient level it causes the SCR to conduct and turn on the windshield wiper motor.

Once the motor starts to operate, the cam-activated switch closes and keeps the motor running even though it shorts out the SCR and halts its conduction. The motor will continue to operate until the cam opens S3. When S3 opens, the motor will remain at rest until the SCR is fired once again by another pulse.

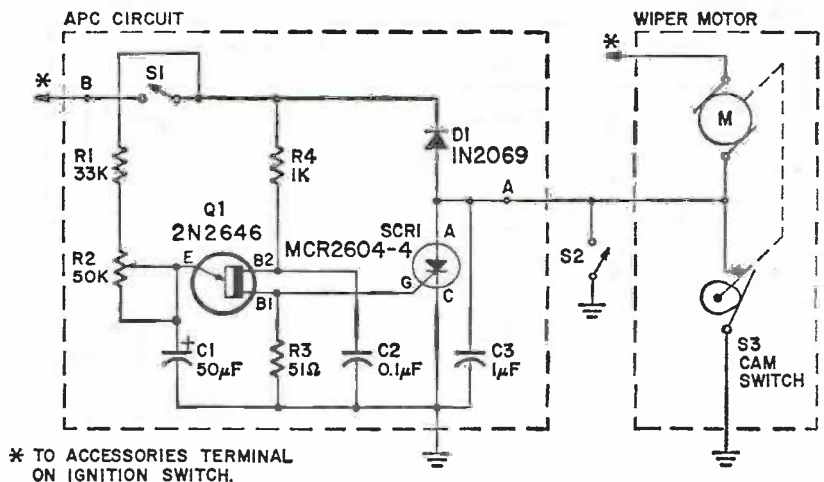
To operate the wipers in a normal manner simply close S2. When S2 is closed it overtakes the APC regardless of the setting of S1 and R2. However, both S1 and S2 must be off for the wipers to remain at rest. Note: On many cars S2 is actually a 3-position switch (off, low, high). If you are fortunate enough to own a Cadillac you have still another position (medium) at your disposal.

Capacitors C2 and C3 and diode D1 prevent the SCR from "false" firing due to inductive kickback from the motor when it is shut off, and prevent other undesirable effects brought on by switching transients.

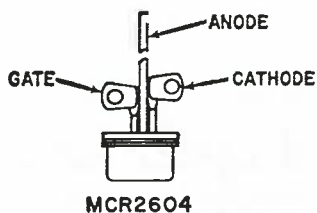
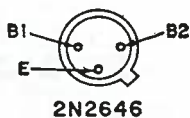
## Construction

A 1½" x 2" printed circuit board was used, but a plain perforated board and push-in terminals can be substituted, wiring and parts layout is not critical. However, you must observe polarity of the capacitors and must hook up D1, Q1 and SCR1 properly.

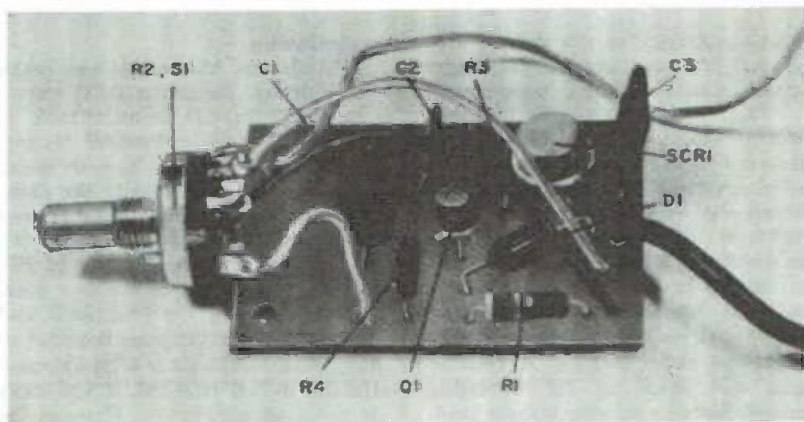
A minor modification of the SCR is necessary to allow insertion into the board. Carefully cut off the solder lugs brazed to the end of the gate and cathode terminals. The SCR is not mounted flush with the board, but inserted only enough to make a good



FOR 6-VOLT SYSTEM, REPLACE R4 WITH JUMPER WIRE, REDUCE R1 TO 15,000 OHMS



*Fig. 1—Uniunijunction transistor controls the wiper by firing the SCR at regular intervals.*



*The APC can be built on a PC board or a perforated board for point-to-point wiring.*

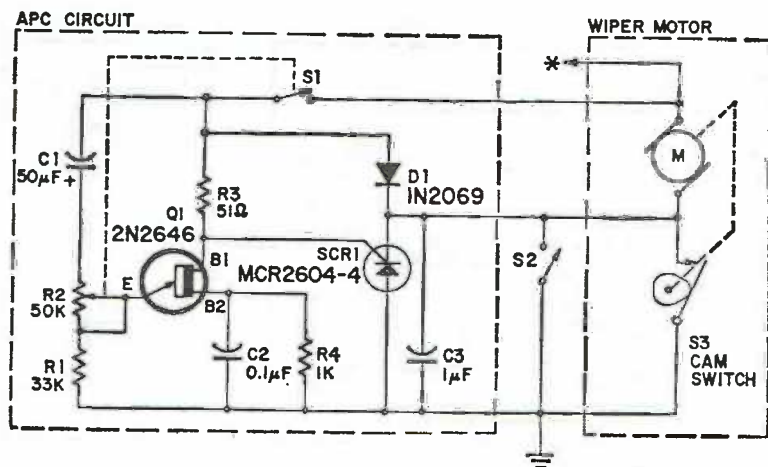


Fig. 2—For cars with 12-volt positive-ground system. See text for 6-volt operation.

#### PARTS LIST

- C1—50- $\mu$ F, 15-volt electrolytic capacitor
- C2—0.1- $\mu$ F, 25-volt disc capacitor
- C3—1- $\mu$ F, 25-volt disc capacitor
- R1—33,000 ohms (15,000 ohms for 6-volt systems)
- R2—Linear potentiometer, 50,000 ohms with switch S1
- R3—51 ohms, 5%
- R4—1000 ohms (replace with jumper for 6-volt systems)

All resistors  $\frac{1}{2}$  watt

- Q1—Unijunction transistor (2N2646, GE, Motorola or similar)
- SCR1—Silicon controlled rectifier (MCR2604-4, Motorola or similar)
- D1—Silicon rectifier, 200-volt or higher PIV (1N2069 or similar)
- S1—spst. (on R2)
- S2—Standard wiper switch (see text)
- MISC—small piece of perforated board, aluminum sheet, knob, wire, solder.

solder connection to the shorter gate terminal. You can shorten the longer terminal after soldering it to the board.

Mounting brackets can be made from  $\frac{1}{32}$ " aluminum sheet, cut and bent as shown. The mounting bracket can be custom fitted for installation on the bottom edge of your dashboard if you don't want to drill any new holes on the dashboard's face. The sides of the unit should be left open to help dissipate heat. Insulation paper or plastic tape should be used where necessary to prevent short circuits. A coat of paint will dress up the mounting bracket. Install a suitable pointer knob to match the car's decor to complete the construction.

#### Installation

Mount the APC within easy reach of the driver. Since the APC circuit is grounded through its metal case, as are most automobile electrical devices, it is necessary for the case to make good electrical contact with the dashboard. If for any reason a good ground connection cannot be made in this manner, run a wire (about No. 20) from the APC's ground bus to a good ground on the car. Connect the B lead from S1 to the accessories terminal of the ignition switch, or to a lead already attached to this terminal, if you can't get to the ignition switch. Connect the A lead from the anode of the SCR to the windshield wiper motor switch S2.

Use No. 16 stranded wire for the A lead. A smaller gauge wire (about No. 20) can be used for the B lead.

If the wiper-motor switch is not accessible in your car or if you prefer not to work under the dash, connect the number 16 wire directly to the wiper motor wire going to the dash-

board switch.

A schematic for constructing an APC for a 12-volt positive-ground system is shown in Fig. 2. For 6-volt operation, reduce R1 to about 15,000 ohms and replace R4 with a jumper wire.

## PART 4

# Projects for the Home

## AUTOMATIC ENLARGER TIMER

by HERBERT ELKIN

AN AUTOMATIC PHOTOGRAPHIC ENLARGER timer is an invaluable darkroom accessory. The timer leaves the photographer's hands and mind free to concentrate entirely on the photographic creation.

The schematic in Fig. 1 is a solid-state, automatically resetting timer. The advantages of a solid-state device over a mechanical timer are the elimination of noise, vibration, manual resetting, and increased reliability. The device is capable of timing intervals from 1 second to 69 seconds in 1-second intervals.

The time interval is set by two rotary switches. Switch S3 sets intervals of 10 seconds and switch S2 sets unit intervals additively. The timer is activated by momentarily pressing START switch S4. When the timing cycle terminates, the device is automatically reset and is ready to be activated again by pressing the momentary START switch. STOP switch S5 provides an abort capability before the end of the timing period if a timing interval has been erroneously set. The unit's recovery time is in the millisecond range, which is advantageous when making test strips.

Timing accuracy is better than 2% up to 30 seconds, after which accuracy falls off slightly due to the leakage current of tantalum capacitor C1. The unit provides exact time interval repeatability, which is important for consistent print results. Time interval

resetability is of more importance than the slight loss of accuracy (2-5%) at intervals above 30 seconds. Most enlarging situations call for less than 30 second exposures.

### How it works

Relay RY1 is energized during timing and applies 117 Vac to the enlarger via the ac receptacle. The power supply is a half-wave type consisting of R21, D1, C2, R22, D3 and R24, and supplies an unregulated dc voltage to operate relay RY1 and a +27 volts dc, regulated by Zener diode D3. The regulated voltage provides an accurate reference voltage to the timing section of the device.

The heart of the timer is unijunction transistor Q1, used as a single-cycle relaxation oscillator. The timing interval is varied by switching resistance values (R1-R16 additively) in the unijunction emitter circuit to vary the RC time constant.

At the end of the timing cycle (when C1 charges to the proper stand-off voltage at the unijunction emitter), a pulse is generated at Base 1 of the unijunction. The pulse is applied to the gate of silicon controlled rectifier (SCR1), which then conducts current around RY1 causing the relay to release and end the timing cycle.

Here is an outline of the timing cycle:

1. Momentarily depressing START switch S4 applies the unregulated





C1—47-uF, 20V, 10% electrolytic tantalum capacitor (Mallory type CSR13-E47K6L or equal; see note below under relay RY1)  
C2—100-uF, 150V electrolytic capacitor  
C3—0.005-uF, 600V ceramic capacitor  
All resistors 1/2W, 10% or better unless noted  
R1—2000 ohms, 5%  
R2—20,000 ohms, 5%  
R3-R10—22,000 ohms  
R11-R16—220,000 ohms  
R17—270 ohms  
R18—47 ohms

115

#### Other parts

RY1—4pdt, 125-mW, 2500-ohms, 8.4-mA dc relay (Allied Control TS154-4C. Available from Newark Electronics Corp., 500 N. Pulaski Rd., Chicago, Ill. 60624 for \$3.96 plus 2 oz. postage ppd. Stock No. 59F512, 1969 catalog). Capacitor C1 is \$2.82, Stock No. 17F1542.

S1—slide switch

S2, S3—1-pole, 12-position rotary switch (Mallory 32112J or equal)

S4—N.O., momentary pushbutton switch

S5—N.C., momentary pushbutton switch

S6—rocker switch

MISC—ac socket, 6 $\frac{3}{8}$  x 3  $\frac{3}{16}$  x 1 $\frac{7}{8}$ -inch bakelite case with aluminum panel (Lafayette 99T 6272), knobs, perf board, transistor socket

voltage through R23 to RY1, which puts the four relay contacts in the activated position.

2. When the START switch is released, the relay remains activated as the holding current is supplied through the normally closed STOP switch S5 and through the closed relay contacts 6 and 7. If it's desired to terminate a timing cycle before the interval set, the STOP switch can be momentarily pressed, interrupting the relay holding current and releasing the relay.
3. With RY1 activated, the enlarger lamp receives 117 volts ac through contacts 15 and 16, +27 volts is supplied to the unijunction circuit through contacts 12 and 13, and R20 is removed (contacts 8 and 9 are open) from across C1.
4. When Q1 fires at the end of the timing interval, the positive pulse generated at Base 1 triggers the SCR, which then bypasses current around the relay coil, causing the relay to release.
5. Since the SCR anode voltage is supplied through relay contacts 6 and 7, the SCR anode voltage is removed when the relay releases. Therefore, the SCR is restored to a nonconducting state and the timer is automatically ready for a new timing operation initiated by the START (S4) button.
6. When the relay releases, contacts 15 and 16 open, the enlarger lamp is turned off, the regulated +27 volts is removed from the unijunction circuit, and R20 is placed across C1. R20 completely dis-

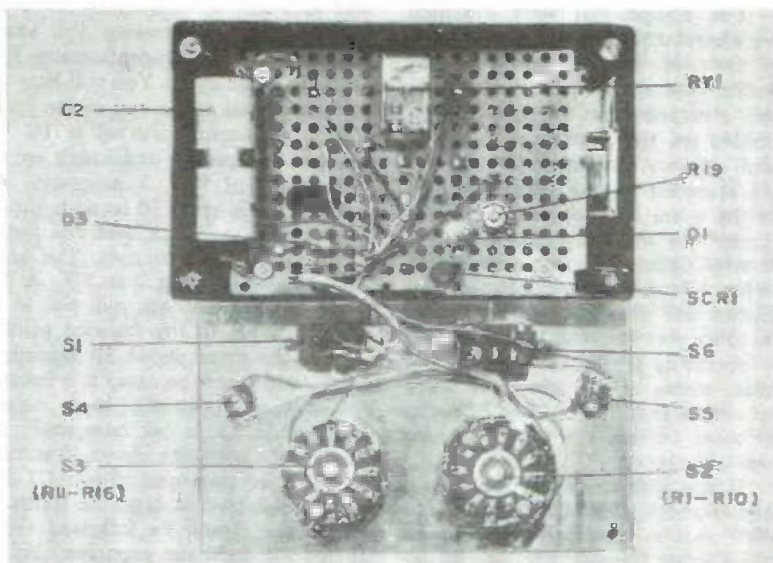
charges capacitor C1 within milliseconds to assure that when a new timing cycle is started the initial voltage on C1 will always be zero. This insures an accurate timing interval.

Capacitor C3 is across relay contacts 15 and 16 for arc suppression. Diode D2 is across the coil of RY1 to eliminate transient spikes during relay turnoff, which could damage the SCR and/or the relay coil. The 1-second timing resistance is divided into two resistors, R1 and R2. R1 is always in the timing circuit and provides protection for Q1 if the timer is started with switches S2 and S3 accidentally left in the 0 second position. R1 contributes only a small error (less than 1%) when S2 is set to 0 and the S3 (x10) resistances are in use.

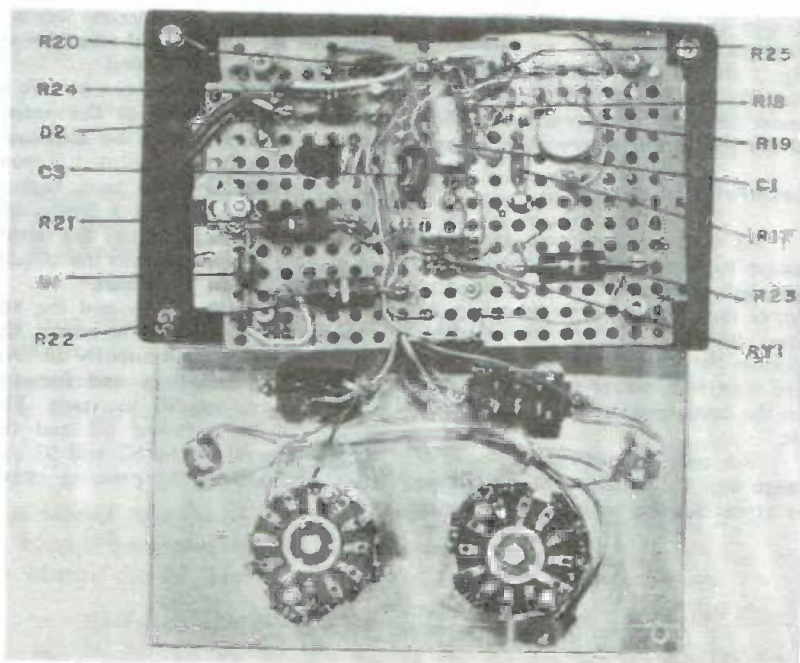
Provision is made for accurate calibration using potentiometer R19. By means of R19, the interbase voltage can be varied, thus controlling the firing point of the unijunction transistor (the firing point voltage at Q1's emitter is equal to the stand-off ratio ( $\eta$ ) x the voltage across Q1, Base 1 to Base 2). The  $\pm 10\%$  tolerance of C1 and the variation in the stand-off ratio of Q1 (0.68–0.82) are balanced out using R19. Although the timing resistors are  $\pm 5\%$  tolerance, it can be shown that the timing accuracy is better than 2% (if the leakage current in C1 is considered negligible).

#### Construction

The cabinet is a 6 $\frac{3}{8}$  x 3 $\frac{3}{8}$  x 1 $\frac{7}{8}$ -inch bakelite utility case. A 117-volt isolation transformer was not used.



**Top view of mounting perf board. Make sure S2 and S3 have adequate clearance.**



**Bottom view of component board. An external relay extends wattage capability.**

The components were mounted on a phenolic perforated board which was fastened to the bottom of the bakelite case using spacers (legs) to allow clearance for the components mounted on the bottom side of the board. Rotary switches S2 and S3 were mounted on the front panel so that the space left between them was sufficient for the relay clearance. FOCUS-TIME switch S6 is a "see-saw" (or "rocker") type which adds to the convenience and simplicity of operation. The relay was selected because of its low operating power (175 mw). The contact rating of the relay will allow it to switch an enlarger lamp of up to 100 watts (within the range of most home enlargers).

If you wish to use the timer with a higher-wattage lamp, use the unit to switch another external high-contact-current relay. Capacitor C1 is a tantalum capacitor and no other type should be used since a low-current-leakage capacitor is required. The ac common points on the schematic in Fig. 1 should be joined together and connected to the indicated side of the power line; do *not* ground this side of the power line to the front panel.

### Calibration and use

As with all electronic gear, the power supply should be checked out first before connecting the circuitry. Leave the connection to relay pin 13 open until the voltage is checked. Double check the power supply wiring and then turn power on. The voltage on the open lead should be +27 volts dc.

Set potentiometer R19 to mid-range and the FOCUS-TIME switch S6 to TIME. Set the timer for 5 seconds

(S2 to 5 and S3 to 0). Start the timer by momentarily pressing the START switch S4. Adjust potentiometer R19 for a 5-second cycle. You will hear the relay drop out at the end of the cycle. Check the timing interval at 10, 15, 20, and 30 seconds and touch up the adjustment of R19 if necessary. All time intervals up to 30 seconds should be accurate to 2%. Check the timing at 40, 50, and 60 seconds. The accuracy should be from 2 to 5%. A 5% error at these settings will not at all be noticeable in the finished print.

If the error above 30 seconds is large (which will be rare) it is because C1 has exceptionally high current leakage. If C1 is not completely defective, the accuracy above 30 seconds can still be improved. Since C1 is a relatively high-cost item, the following procedure should be followed rather than to discard the capacitor. Set S3 to the shortest time at which the accuracy is to be improved (say 50 seconds) and S2 to 5 seconds. Replace R15 with a 500K pot and adjust for an accurate 55-second interval.

The potentiometer can be left in as part of the circuit or the potentiometer resistance can be measured and a fixed resistor substituted. Repeat the procedure for the next highest time setting of S3 (60 seconds and R16 in this example) after the selected resistor is soldered into the circuit. Leave S2 set to 5 seconds.

With switch S1 on and the FOCUS-TIME switch S6 set to FOCUS, the enlarger will be continuously on. After selecting negatives and focusing the enlarger, set S6 to TIME. The lamp will be switched off and the timing interval set by S2 and S3 can then be initiated by pressing START switch S4.

# HALF-HOUR INTERVAL TIMER

by FRANK H. TOOKER

RC ELECTRONIC INTERVAL TIMERS usually have a maximum time lapse of about 10 minutes. Beyond this delay, timers have been costly to build, have required relatively hard-to-get parts or have been inexpensive but inaccurate.

The timer described here is different. It's not cheap to construct, but it won't flatten your wallet either. All components are available from catalog sources. The unit has two timing intervals, the longer is  $\frac{1}{2}$  hr. Accuracy is  $\pm 5\%$  for the low range;  $\pm 10\%$  for the upper range.

You can use a timer such as this to get your breakfast eggs just right or for timing long-distance telephone calls. In the winter it will tell you when you've been under the sun lamp long enough; in the summer it tells you when to move the lawn sprinkler to another location. If you're a photography enthusiast, it will let you know—even in total darkness—when your negatives are developed, fixed and washed. You'll find a host of other uses once you have the instrument built.

## How it works

A schematic of the  $\frac{1}{2}$ -hour interval timer is in Fig. 1. Potentiometers R1, R2 and resistor R3, together with tantulum capacitor C1, determine the timing interval. Resistor R5 has much too small a value to affect the intervals significantly.

The voltage slowly developed across C1 is fed to the gate of field-effect transistor Q1 via R6. The FET is connected as a dc source follower.

Gate junction leakage in the FET is very low—perhaps 1 nA at room temperature. Thus, loading of the RC timing components is negligible. Leakage resistance of the timing capacitor is the only significant loss in this part of the circuit.

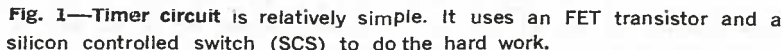
The dc potential at the source junction of Q1 rises as the voltage across C1 rises. This potential appears across voltage-divider resistors R7 and R8. The potential across R8 is fed to the cathode gate of the silicon controlled switch via R9. When this potential rises to the triggering level of the SCS (silicon controlled switch) (about 0.5 volt), the SCS turns on, applying voltage to the Sonalert unit.

The power supply uses a center-tapped transformer, a pair of rectifier diodes and a fairly large filter capacitor, C4. The dc output across C4 is fed through resistor R11 and regulated at 15 volts by Zener diode D2. On the timer PC board, this potential is regulated to 10 volts by R10 and Zener diode D1 before being applied to the sensitive RC timing circuit. Rigid control of the timing-potential supply is thereby maintained. The remainder of the timer is supplied at the regulated 15-volt level.

When the Sonalert sounds, and the instrument is switched off, section S1-b of the ON-OFF switch opens the transformer primary circuit, while section S1-a discharges timing capacitor C1 by short-circuiting it through R4 and R5. The negative pulse that appears across R5 at the instant of discharge is fed to the cathode gate of



R10—470 ohms  
R11—220 ohms, 1 watt  
D1—10V, 1-watt, Zener diode  
D2—15V, 1-watt, Zener diode  
D3, D4—rectifier diodes (1N3754 or similar)  
Q1—2N5457 field-effect transistor  
SCS1—3N84 silicon controlled switch  
S1—dpdt slide switch  
S2—spst slide switch  
T1—30V ct, 50-mA power transformer  
LM1—120V ac neon pilot lamp (Leecraft 36N2315 or similar)  
Mallory Sonalnet type SC628  
MISC.—5 x 4 x 3-inch aluminum cabinet, power cord & plug, pointer knob, rubber feet, 3-lug terminal strip, small L-brackets, rubber grommet, hardware

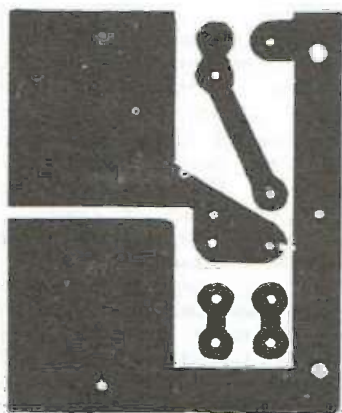


the SCS via C2, triggering the SCS off. Without this feature, the Sonalert would continue to sound, even though the timer is switched off, until filter capacitor C4 becomes nearly discharged.

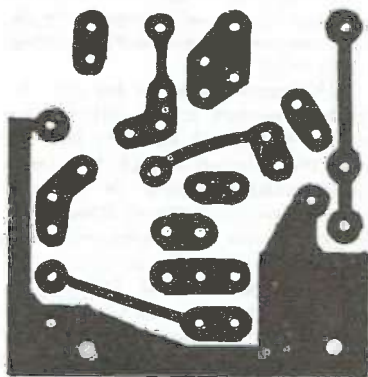
Capacitor C3 reduces the sensitivity of the SCS to line-voltage transients. The circuit, however, may at times be triggered by a strong transient pulse, when the potential at the cathode gate is raised near the firing level.

The instrument is equipped with a neon pilot lamp to indicate when the unit is on and timing. (It's discouraging to wait for a ½-hour interval and discover you've forgotten to plug the power cord into a wall receptacle!)

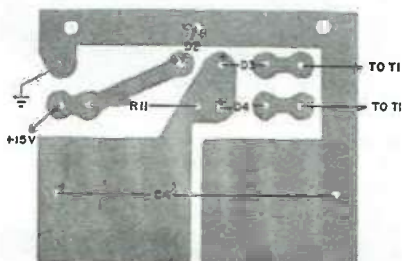
With S2 closed, any time interval between 2 minutes and 16 minutes may be obtained by setting potentiometer R2. When switch S2 is opened, a fixed 14 minutes is added to



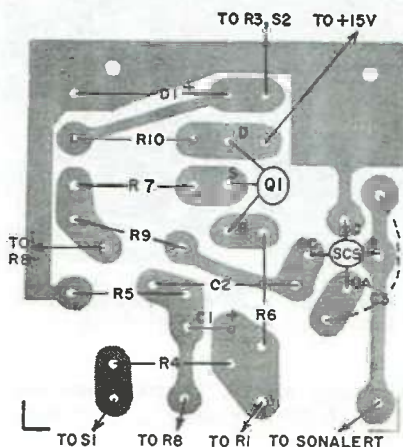
**Foil pattern** for the power supply circuit board. This is actual-size drawing.



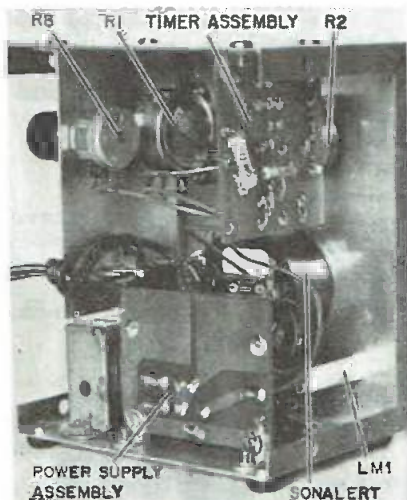
**Foil pattern** of the timer circuit board. Drawing is actual-size pattern.



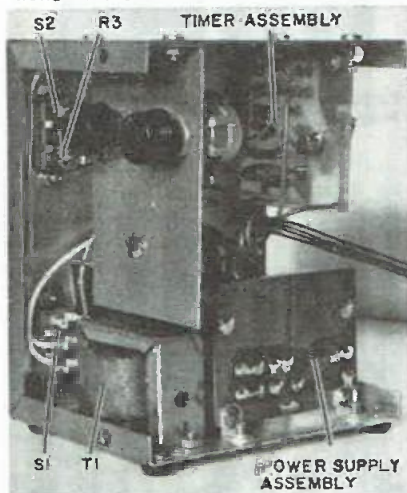
**Parts layout** on the power supply board. Phenolic board material is ok here.



**Parts layout** for the timer board. For accuracy use a glass-epoxy board.



Inside the timer case you get a close look at parts placement. Note the positions of the two circuit boards.



Another internal view shows parts locations from a different vantage point. Follow this arrangement closely.

the timed interval. The calibrated settings of R2 then cover a 16–30-minute range.

The setting of R8 determines the voltage level across capacitor C1 at

which SCS1 turns on. This adjustment is made to obtain the 16-minute interval, when switch S2 is closed and potentiometer R2 is set at *maximum* resistance in the circuit. Potentiometer R1 is adjusted to obtain the 2-minute interval, when switch S2 is closed and potentiometer R2 is set at *minimum* resistance in the circuit.

The setup exhibits some variation in the timed intervals as a result of variations in room temperature. Over the range of 65–90°F, these variations fell within the specified tolerances. A variation of 3 min in ½ hr represents a 10% tolerance.

### Building a timer

The timer shown in the photos is constructed in a 5x 4x 3-inch utility cabinet. Two PC boards are employed: one for timing circuit components, the other for power supply components. The timer-circuit PC board is preferably of glass-epoxy base material. The power supply board may be phenolic.

*Be particularly careful not to overheat timer components when soldering this board.* Even slightly excessive heat applied to C1 or the FET can materially affect circuit performance. If the timer cannot be calibrated as described later, or if the calibrated points appear to shift without reason or are otherwise erratic, and all solder joints are clean and properly made, then component overheating should be suspected.

The parts list calls for two 10-megohm resistors in parallel for R3. Using two parallel resistors like this allows the value of one of them to be adjusted slightly higher or lower to bring the longer range of timed intervals within the 10% tolerance. This adjustment is desirable because, as the timing extends beyond 16 minutes, the leakage resistance of C1 begins to have a slight but increasing influence on the timed intervals. As a result of this influence, R3 will appear to have a somewhat larger value when R2 is

set at the 16-minute calibration point than when it is set at the 2-minute point. Thus, without compensation, all this effect could appear at the upper end of the range of the longer timed intervals. By carefully adjusting the value of R3 the effect can be divided between the high and low settings of R2, providing a more reasonable tolerance over the entire range. The setting of potentiometer R8 determines the duration of the *longer* timed intervals, while the setting of potentiometer R1 determines the duration of the *shorter* timed intervals. Before turning the timer on initially, set R1 to put about two-thirds of its resistance in the circuit, and set R8 to just a little below midposition.

With R1 and R8 set as described, set potentiometer R2 in its maximum-clockwise position (maximum resistance in the circuit). Make sure switch S2 is closed, shorting R3. Then move S1 to On. Using an accurate electric clock with a sweep second hand, check the duration of the timed interval. It should be 16 minutes. If it is less, switch the timer off and adjust R8 to put a little less resistance in the circuit. Then check the timed interval again. Try to set R8 so that the interval is accurate to within  $\pm 10$  sec. Check the interval several times, to

make sure of the best setting of R8, and to determine the repeat accuracy.

With R8 properly adjusted, turn potentiometer R2 to its maximum counterclockwise position (minimum resistance in the circuit), and check the timed interval. It should be 2 min. If it is less than this, adjust potentiometer R1 to put a little more resistance in the circuit, and check the interval again. With R1 properly adjusted, go back and recheck the 16-minute interval. Readjust R8 slightly, if necessary. Then recheck the 2-minute interval, and readjust R1 slightly, if necessary. Go over these checks and settings several times.

Calibrating the settings of R2 consists of locating the pointer knob positions for the 4-, 6-, 8-, 10-, 12- and 14-minute intervals, and of making a tiny mark on the cabinet front at each position, using a sharp scriber. This can take time, especially when the exact settings for the longer intervals are involved. Having a good book to read during the interim each time will make the task seem less tedious. Be sure to make a mark at the extreme settings of R2 to locate the 2- and 16-minute intervals.

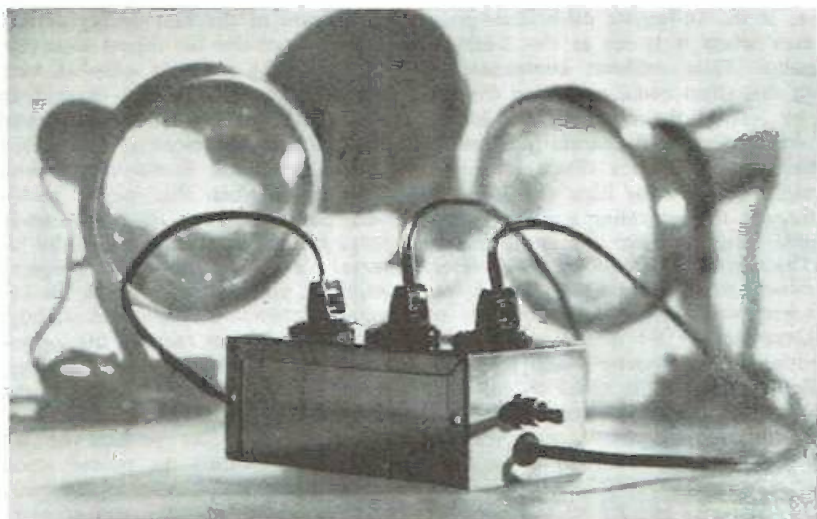
The optimum value for R3 may now be determined, as described previously. When this has been done, the timer is ready for use.

## DANCING CHRISTMAS LIGHTS

By R. W. FOX\*

REMEMBER THE COLOR WHEEL YOU bought last year, and the blinking lights from the year before? They're obsolete. Here's the color blender, the latest in special lighting for this Christmas season. No more sudden flashes; no more monotonous repetition

of red, yellow, green, for hours. This year you can get every color of the rainbow with a cycle time of—well, it still hasn't been measured. All of this in a tiny box, waiting to please both family and friends.



The Color Blender consists of three identical circuits, one for blue, one for red, and one for green. Combinations of these colors will produce every color in the rainbow. Figure 1 shows graphically what is happening and, by the way, the first time you turn

it on could very well be a happening! Due to the variation in component tolerances, these circuits will not be identical in performance. One may come on faster than the rest; one may stay off longer. This will give a true randomness to the colors, which is not

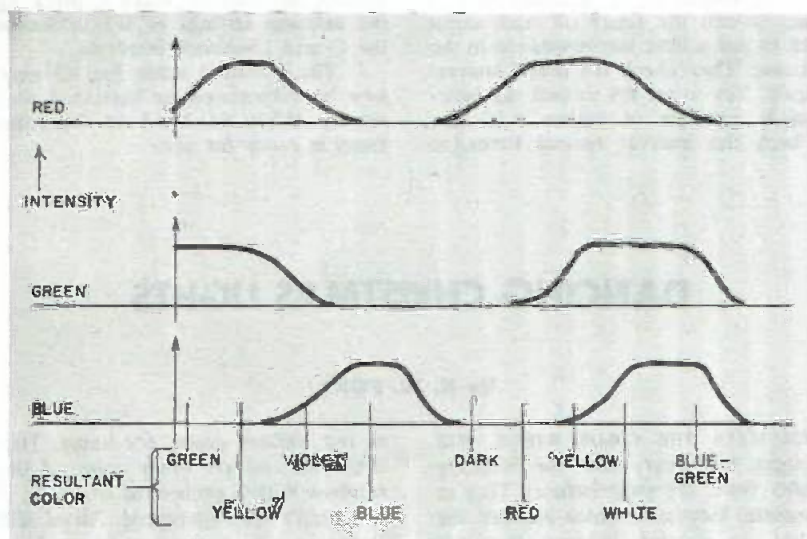


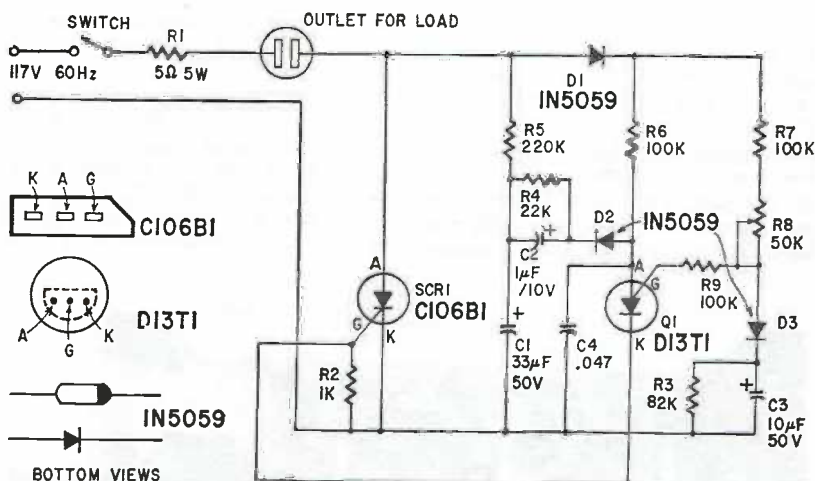
Fig. 1—Output amplitude of three light channels as a function of time. Variation in component values cause each channel to randomly vary lamp pattern and intensity.



end the red lamps go out to leave a blue-green scene.

Inside, the Color Blender will handle up to 25 miniature lights (6 watt) per outlet on your Christmas tree. If you intermingle three strings, one each of red, blue and green, with balls and tinsel to reflect the light, your tree will seem to come to life. Or, as another way, you could use up to 150 watts per outlet of flood lamps with color filters shining on a metallic tree.

Outside, you could use strings of lights on a tree (be careful not to exceed the load limit on your circuit), or flood lights on the house or on a



**Fig. 2—Unijunction transistor Q1 is used in conjunction with the D13T1 SCR to turn the lamps on and off. The charging relationship between C1, C3 and C4 set the pattern.**

## PARTS LIST

R4—22,000 ohms  
R5—220,000 ohms  
R6, R7, R9—100,000 ohms  
R8—50,000 ohms, potentiometer  
Q1—D13T1  
SCR1—C106B1

Miscellaneous hardware—box, line cord, outlets (Amphenol 61-F1 or equiv) circuit board, switch (Except for the box and line cord, three sets of parts will be needed to build a three-lamp controller)

R1—5 ohms, 5 watts  
R2—1000 ohms  
R3—82,000 ohms

door covered with wrinkled foil. The wrinkling keeps the colors separated but allows them to blend in some areas to give a multiple color effect.

After Christmas, there is no need to put the Color Blender away. Just build a foil reflector with a large picture frame and shine floodlights on it for a great addition to your game room.

### How it works

The G-E type D13T1 Programmable Unijunction Transistor, allows us to build this unique circuit. The D13T1 can be thought of as a complementary SCR. When the gate voltage drops below the anode voltage, current flows from anode to cathode. This circuit uses this feature to phase-fire the C106B1 SCR (see Fig. 2). When the Color Blender is initially turned on, both C1 and C3 have no charge. Capacitor C4 quickly charges to a voltage greater than the D13T1 gate voltage and triggers the D13T1, which in turn triggers the SCR, causing the light to come on brightly. On each succeeding cycle of operation capacitors C1 and C3 have a higher

initial charge so that C4 cannot charge to a voltage which would trigger the D13T1 until much later in the cycle. Since C3 charges at a faster rate through R7-R8 than C1 and through R5, R6, R4, the lamp dims slowly. When the lamp goes dark, C1 discharges faster than C3 and the triggering angle of Q1 is advanced and the light brightens again.

The purpose of R1 in the circuit is solely for the protection of the C106B1 SCR. This resistor keeps the peak current through the SCR within its ratings. Although it wastes a considerable amount of power, about 4 watts at full load, it is better than replacing the SCR whenever a lamp burns out.

In addition to adding to the beauty of your Christmas display, the Color Blender also increases lamp life at least 25 times. This increase in life is due to operating the lamp on half wave. An added consequence of half wave operation is lower bulb temperature, decreasing the chance of fire, hence giving greater safety.

With the board mounting, all three circuits can be built at the same

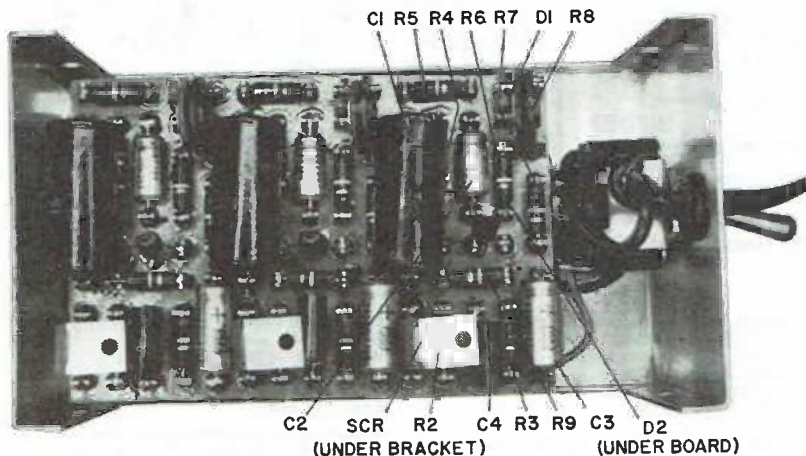


Fig. 3—Component layout for all three identical control channels. Parts are identified on one channel only. Resistor R8 is adjusted to establish proper "on" sequence.

time, reducing the chance for mistakes. Since one side of the line is common to many components, a single bus running the entire length of the board may well be a good way to start. To make the circuit compact, R1 should be left off the board and wired directly to the outlet. This also keeps R1 far enough away from the SCR. Watch the connections of the SCR and Programmable Unijunction Transistor. Reversing the lead order could very well destroy these units. The polarities should also be observed on the electrolytic capacitors and the three diodes.

*For safety's sake do not ground the box to the line, for a reversal of the plug could be dangerous.* If you have a three wire system, though, by all means use that third wire as a ground for the case.

Figure 3 shows the board as it was constructed with all three circuits so that it would fit in a 5 x 2 x 3 inch box (Bud-CU-2106-A or equal). You will note in Fig. 3 that the leads and

tab of the C106B1 are bent. **CAUTION:** The tab should be held by long-nosed pliers between the body and the bend while you are forming the device.

To prepare your box, punch three 1¼ inch diameter holes in the top, one small hole in one end for the line cord, one for the switch and the box is ready. Insert into the box the three outlets and fasten them in place with spring clips. The R1 resistors can be attached on one terminal and their free ends connected to the switch. Use heavy bus wire from the other outlet terminals to the board, and no other supports should be needed to hold the board in place.

Once the circuit board is mounted in the box and the final connections are made, but before you put the Color Blender to use, you may have to adjust the R8 resistors. If an associated light comes on but damps out to a level between full on and off, R8 should be decreased. If the lamp snaps on, R8 should be increased.

## PHOTOTACHOMETER

By BRICE WARD

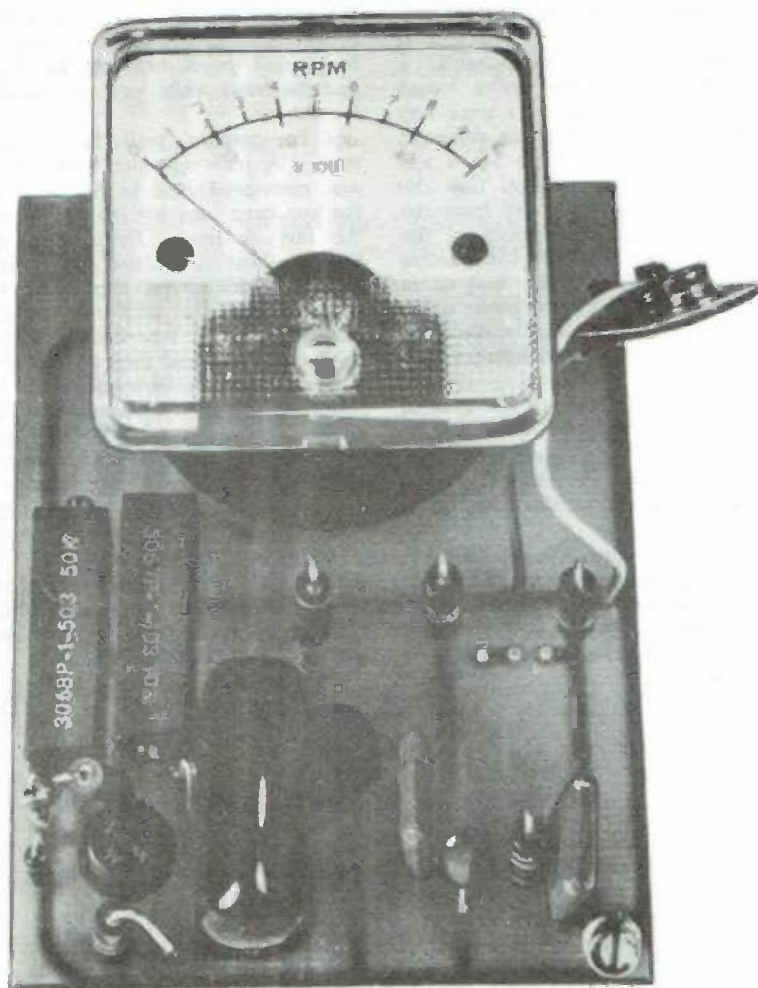
HOW WOULD YOU LIKE TO READ THE speed in rpm of any rotating shaft without making physical contact with it? Drill motors, lathes, drill presses, slot-car wheels, capstan drives, turntables and dc tape recorder drives are just a few possibilities. Best of all, how would you like to do this without fiddling with dials or knobs?

This little hand-held tachometer measures rpm this way: Place a contrasting mark on the shaft, pulley,

capstan or what have you, and make sure a steady illumination falls on it (from an incandescent lamp or flashlight). Aim the probe, press a push-button switch and read the rpm directly from the calibrated meter.

### How it works

The sensing circuit uses a cadmium selenide cell designed to respond quickly to changes in light level. Other cells could probably be used, but the



*Meter for the phototachometer is mounted directly on board.*

Clairex unit (see parts list) will provide excellent response out to the limit of the rpm range and probably beyond this range.

Output from the photocell is coupled through capacitor C1 to Q1, used as an amplifier with a voltage gain of about 50 (Fig. 1). The amplified voltage pulse is coupled through C2 to the multivibrator (Q2-Q3).

Transistor Q1 and the photocell, then, form the sensing element circuit. Any device that can deliver a 1-volt or better, positive-going pulse to the multivibrator could be used to drive the meter circuit.

A pulse standardizing circuit utilizes a monostable multivibrator followed by a Zener diode. The monostable multivibrator, variously called a





one-shot or univibrator, has only one stable state, determined by the feedback arrangement used. Here, the pulse width is determined by the variable time constant of C3-R7, and the pulse height is clamped to a 5.2-volt level by Zener diode D3.

Pulse counting is achieved with a simple half-wave rectifier. It applies the average power of the pulses to the meter, which smooths the pulses by acting as an integrator. A series of waveforms illustrate what occurs in the circuit. All waveforms are positive-going above and negative below the applied zero reference line.

The voltage at point A in Fig. 1 is negative-going with an amplitude of about 0.8 volt starting from a 4.2-volt level. This 4.2-volt level will change with the light level on the photocell or the voltage applied to the circuit. (An 18-volt supply should give a 1.6-volt negative-going signal, permitting the probe to be moved back from the pulsing light source.)

Since a fluorescent lamp temporarily extinguishes on each half-cycle of 60-Hz line voltage, pulse frequency is 120 pulses per second, or (multiplying by 60) 7200 rpm. This gives a good source for examining circuit operation, and was used to calibrate the phototachometer.

This voltage pulse output of the cell is coupled by C1 to the base of Q1 (B), overcoming the positive bias on Q1 and cutting it off. The result is decreased current flow through R4 and a consequent reduction in the voltage drop across R4. Since R4 and Q1's collector-emitter resistance represent a voltage divider to ground (from the 9-volt source), if the voltage across R4 decreases, the voltage across Q1's collector-emitter increases and the voltage on the collector of Q1 (C) increases with respect to ground.

## Multivibrator operation

In the multivibrator, Q2 is normally cut off and Q3 is conducting. The collector-emitter current of Q3 is then determined by the resistance of R8. With a low Q3 collector voltage, less voltage is returned to the base of Q2 (D) than is required to cause forward base-emitter junction bias.

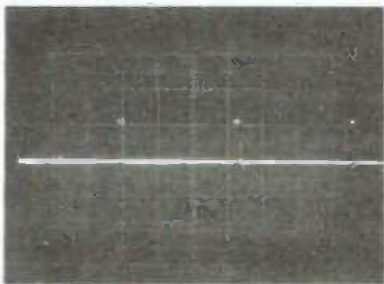
When the signal on the collector of Q1 swings positive, the voltage rises on the coupling capacitor C2. In trying to charge to this rising voltage, C2 draws current through Q2's base-emitter junction, causing Q2 to conduct and its collector voltage to drop (E).

Capacitor C3, which was charged to about 7.6 volts, begins to discharge through R7, and drives the anode of D1 (F) sharply negative, cutting the diode off. Since Q3's base-emitter current path (G) is through D1, Q3 is cut off and the voltage at point H rises to a level controlled by D3. The 5.2-volt pulse developed serves to drive the metering circuit, and supplies regenerative bias to Q2.

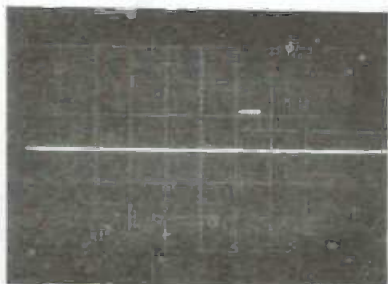
As long as D1's anode remains negative with respect to its cathode, Q3 will remain cut off, and the length of time that the anode remains negative is determined by the time constant of C3-R7.

In short, the multivibrator has developed a pulse of a width determined by the C3-R7 time constant and a height determined by the Zener diode at its output. On completion of the C3 discharge, the voltage on the anode of D1 goes positive with respect to its cathode and begins to draw current through Q3's base-emitter junction. Next, the voltage drops on Q3's collector feed a negative-going voltage back to the base of Q2, causing Q2 to begin cutting off. The positive-going voltage developed on Q2's collector starts to charge C3, reinforcing the switching cycle by positive feedback.

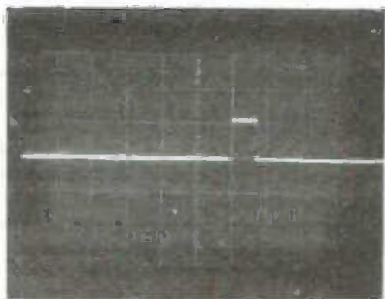
Switching times for Q3 are extremely fast (probably in the nanosec-



*Fig. 2—Scope photo shows pulses at point H with a 3600 rpm (60 Hz) input instead of the 7200 rpm calibrating input (120 Hz) shown in Fig. 1. Pulses appear every 3 cm instead of 1.5 cm, halving the current to the meter.*



*Photo obtained at point H indicates a pulse width of 0.56 msec.*



*Adjusting R7 increases the pulse width to 0.7 msec.*

ond range), and the entire standard pulse is generated in about 0.56 msec.

Figure 1-h shows pulses generated at 7200 rpm, while Fig. 2 shows them generated at 3600 rpm. The 7200-rpm pulses appear 8.5 milliseconds apart. This means that roughly 14 more pulses could be generated between existing pulses, or a pulse rate equivalent of 100,000 rpm.

At this rate, due to the extra energy generated, the meter range would have to be increased to 500  $\mu$ A to read the total current.

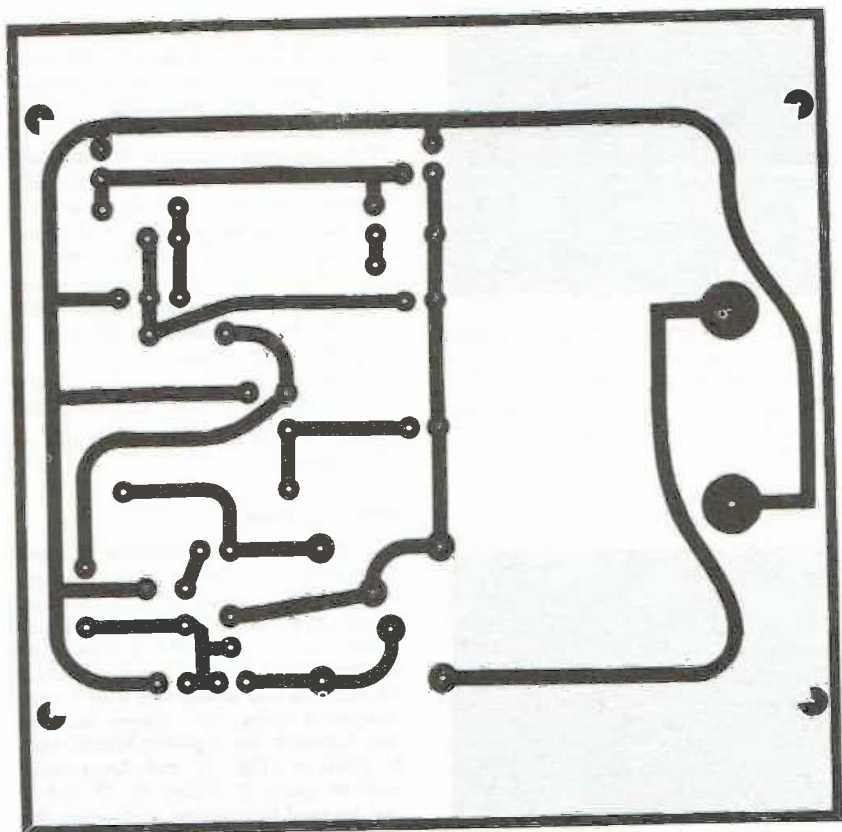
As the system is set up, 10,000 rpm is the upper limit, and R9 is adjusted to give a full-scale reading on the meter.

## Construction

For ease of construction, the unit was built on a circuit board. Before installing R7, adjust it with an ohmmeter to about 3500 ohms or, if you prefer, use a 3600-ohm resistor. The photocell (R1) can either be installed on the board or connected with a two-conductor cable for remote location use. Connect the negative battery lead to point A (Fig. 3) and the positive lead to point B. Points C, D and E can be used to raise the voltage on the photocell.

To do this, remove R2 from its present location and install it from point C to point D. Install a second battery with the negative lead going to the old R2 connection (extension from point B) and the positive lead to point E. With this arrangement a pair of 9-volt batteries will apply 18 volts to the photoresistor (increasing its sensitivity), but the circuit still receives only 9 volts.

With 18 volts on it, the probe can be moved farther from the device being measured. Depending on the amount of light striking the rotating shaft, surrounding ambient light and other circumstances, a 9-volt probe will function about 6" from the shaft. With the extra battery, this distance



could probably be increased to 12".

The meter is installed with the *plus* (+) terminal to point G and the *minus* (-) or unmarked terminal to point H.

If you're a "scratch" builder or experimenter, the transistors and other circuit components are not critical. For example, any reasonably good npn silicon transistors should work in the circuit. Reversal of all polarity-sensitive elements and the battery should allow operation with pnp types of transistors.

The Zener diode is not critical either. If it's rated at less than the voltage on the collector of Q3, and clamps the pulse amplitude, this can be corrected by adjustment of R9.

The metering diode should be germanium, but this diode and D1 would probably work with silicon devices.

The phototachometer can be used without a case, but since the meter movement supports the circuit board it could be mounted in a case. Make a cutout for the meter and allow the meter to support the circuit board inside the case.

A larger meter movement could be used by drilling holes spaced a little farther apart on the printed circuit board and again hanging the circuit board on the meter.

### Adjusting your tach

If you use a graduated 50- $\mu$ A meter, every reading will have to be

multiplied by 200. (This is done quickly by adding two zero's and doubling the result.) As mentioned earlier, a fluorescent lamp will provide a 7200-rpm pulse source. Since we want to determine what the meter should read for 7200 rpm, we reverse the procedure: divide 7200 by 2 (3600) and remove two zero's (36) to obtain a setting point for our calibration.

Hold the light cell about 12 inches from the fluorescent light and move it in. The meter should jump to some

intermediate value, and then double this value as the probe is moved a little closer. It appears to be characteristic of fluorescent lamps that the light pulse output is higher on one cycle than the other. Therefore, the first meter jump represents a reading of *every other pulse* from the lamp or 60 cycles per second (3600 pulses per minute).

At the point where the meter value doubles, adjust R9 for a reading of 36 on the meter, and the tachometer is calibrated. By proper adjustment

#### PARTS LIST

- C1, C2—0.1  $\mu$ F, 75-V Mylar capacitor
- C3—0.22  $\mu$ F, 100-V EIMenco 1DP-3-224 or equal
- R1—Clairex CL903C fast-response photocell
- R2—1-megohm,  $\frac{1}{2}$ -watt resistor
- R3—3.3-megohm,  $\frac{1}{2}$ -watt resistor
- R4, R5—22,000-ohm,  $\frac{1}{2}$ -watt resistor
- R6—47,000-ohm,  $\frac{1}{2}$ -watt resistor

- R7—10,000-ohm, Bourns E-Z Trim No. 3067-1-103 potentiometer
- R8—1500-ohm,  $\frac{1}{2}$ -watt resistor
- Q1—2N2712 transistor (G-E)
- Q2, Q3—2N697 transistor
- CR1, CR2—1N456 germanium diode
- CR3—1N709 silicon zener diode
- M1—50  $\mu$ A, 1 9/16" square dc meter (Lafayette Radio 99 H 5049 or equiv)
- MISC—Circuit board, battery clip, 9-V battery

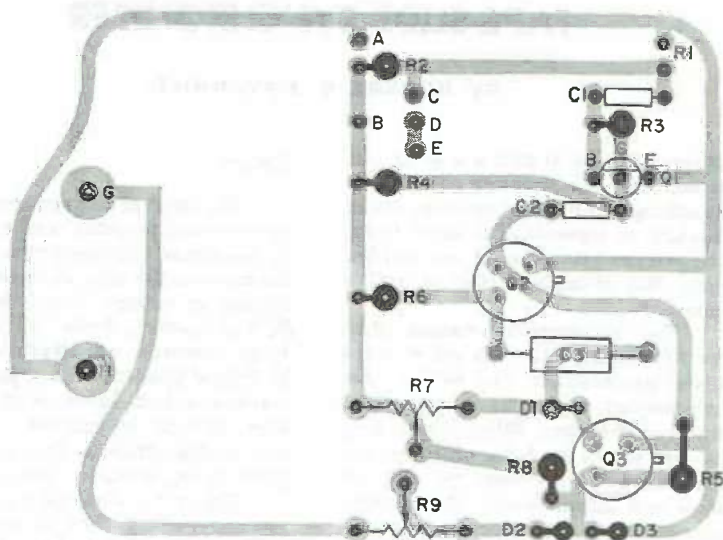


Fig. 3—Component placement on the PC board. Text explains letters A-H.

of R9, the phototachometer should be capable of reading pulses out to the equivalent of 100,000 rpm. The photocell response time may become a limiting factor at or before this rate is reached.

### Trying it out

First, in reading something like the propeller on a model airplane engine, remember that if the prop has two blades the indicated rpm will be twice the true rpm since the light is being chopped at twice the actual rpm. And if a disc has 30 segments the indicated rpm will be 30 times the actual.

This applies for any rotating device. Determine the number of full cycles for each revolution and divide the rpm reading by this figure to obtain true shaft rpm.

If you measure very high-speed devices, such as a model airplane engine, it might be necessary to reduce the overall pulse width by reducing either C3 or R7, and to reset R9 to indicate, say, 50,000 rpm full scale.

Other possibilities? Connect the photocell to the circuit board using a long cable. The very small probe can then be mounted at a remote location to monitor some shaft rpm not otherwise easily read.

The unit could be used as a fail-safe. The univibrator pulses could be applied to a rectifying system used to energize a relay. If monitoring shaft rotation, the relay would de-energize and sound an alarm when rotation stopped.

In short, the potential uses for the circuitry are restricted only by your imagination.

## TAPE-SLIDE SYNCHRONIZER

By ROBERT S. HAVENHILL

SHOWING COLOR SLIDES is a good way to recreate your vacation for friends. By adding commentary on tape, you're relieved of repeating the same thing, and you can't forget what you said last time. But changing slides is still a nuisance.

The synchronizer changes slides for you automatically. All you need is a stereo tape recorder. You put the voice commentary on one track, and a sync signal on the other. When friends drop in, you simply load the projector and the recorder, turn them on, and the show runs automatically. The recorded sync pulses advance slides as your commentary moves ahead. And there are no relays to chatter or stick.

### Theory

The heart of this synchronizer is a bidirectional thyristor, called a Triac. As you know, a silicon controlled rectifier can conduct only on one-half of an applied ac voltage. Why? Because an SCR is simply a diode with a gate. A Triac consists of essentially *two* diodes in reverse parallel, with a gate. It can conduct on both halves of an ac waveform and can be used as a full-wave control and switch. A Triac can also be gated by ac, which is quite useful.

Figure 1 shows the symbol for a Triac and one method of triggering it. When S1 is closed the gate is connected to anode 2 through current-lim-



iting resistor R2. The Triac then conducts and switches on the ac motor. (Some slide changers are operated by a solenoid, but the principle is the same.)

Figure 2 illustrates a second method of triggering: An ac signal is applied to the gate-anode 1 circuit of the Triac through isolation transformer T1. This second method is used in the tape/slide synchronizer on playback. The recorded sync signal on the tape is fed to the primary of T1. The signal then triggers the Triac, which conducts and activates the motor (or solenoid) in the projector. The slide advances.

In the record mode, it's necessary to simultaneously trigger the Triac (to

advance the slide) and produce an ac signal (to put sync on the tape). The circuit is shown in Fig. 3. Closing S1 triggers the Triac and advances the slide. At the same time the 60-Hz ac to the motor passes through R1, causing a 1-volt drop. This 1-volt signal is passed through T1 to the second tape channel, while voice commentary is going on the first channel. Fig. 4 shows outline and terminal connections of the Triac.

#### Operation—record

The synchronizer (Fig. 5) is connected to one recording input of the tape recorder through J1, and, to the slide changer through P1. S1 is set to the record function. The tape recorder

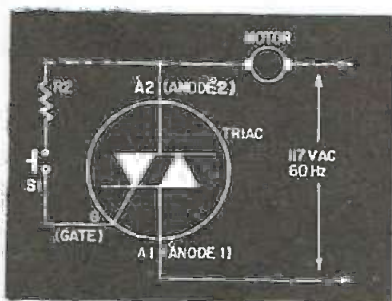


Fig. 1—Triac is turned on with voltage between gate and anode 2. It can conduct on both halves of an applied voltage.

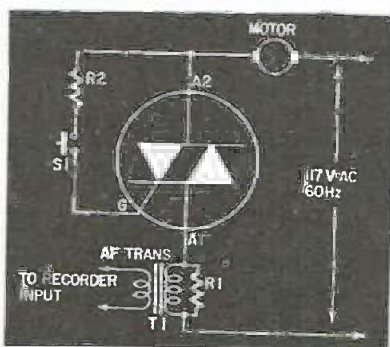


Fig. 3—Drop across R1 is sync signal.

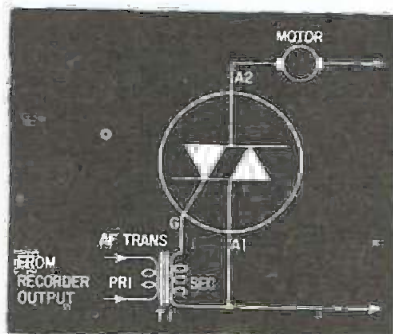


Fig. 2—Either ac or dc will trigger the Triac. Signal can also be applied between gate and anode 1 to turn the device on.

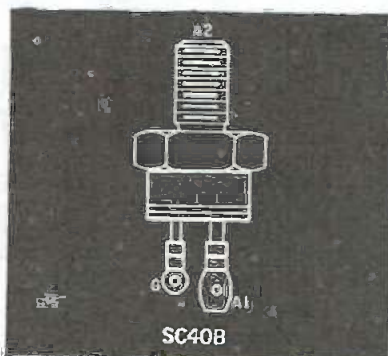
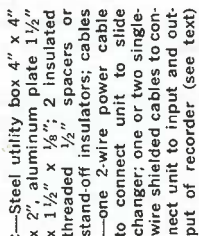


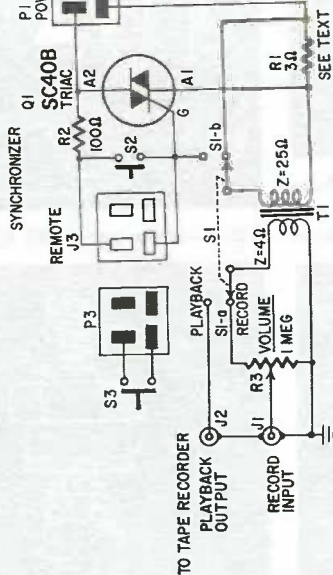
Fig. 4—Typical Triac outline, terminal connections, with mounting stud at top.



**Fig. 5—You need build only the synchronizer (left) but your slide changer must have a remote change facility (right).**

- R3—1-megohm, audio-taper potentiometer  
 S1—D.p.t. slide switch  
 S2—S.p.s.t. pushbutton switch, normally open contacts  
 S3—S.p.s.t. cable mount pushbutton switch (Switchcraft 913 or similar)  
 T1—Output transformer, 25 ohms to 4 ohms (Stancor TA-62 or similar)

**MISC**—Steel utility box 4" x 4" x 2", aluminum plate 1½" x 1½" x ½"; 2 insulated threaded ½" spacers or stand-off insulators; cables—one 2-wire power cable to connect unit to slide changer; one or two single-wire shielded cables to connect unit to input and output of recorder (see text)



## Parts List

- |  |  |  |  |   |
|--|--|--|--|---|
| J1, J2—Phono jacks   |  |  |  | mote control socket on projector                            |
| J3—4-prong chassis (Cinch-Jones S-304-AB or similar)       |  |  |  | P3—4-prong cable plug (Cinch-Jones P-304-CCT or similar)    |
| J4—4-prong cable socket (Cinch-Jones S-304-CCT or similar) |  |  |  | Q1—Triac G-E type SC40B; Texas Instruments T1C25 or similar |
| P1—4-prong chassis (Cinch-Jones P-304-AB or similar)       |  |  |  | R1—3-ohm, 5-watt resistor (see text)                        |
|  |  |  |  | R2—100-ohm, 5-watt resistor                                 |

- R1—3-ohm, 5-watt resistor (see text)  
R2—100-ohm, 5-watt resistor

is then adjusted to record voice from a microphone on the other input channel. The tape is started and when you want to change slides, S2 (or remote switch S3) is pressed. This triggers the Triac into conduction and starts up the slide-changer motor (or solenoid). In a motor-operated changer, when S2 is released the cam switch in the slide changer continues to feed ac to the motor or solenoid until the cycle is complete.

Meanwhile, as S2 is closed, the sync signal is passed through T1, volume control R3 and J1 to the input of the recorder. The level of the sync signal can be set with R3; thus the volume control of the recorder does not need to be changed in going from record to playback.

S3 is an optional remote-control switch. It lets you put sync on tape without having to be near the unit.

### Operation—playback

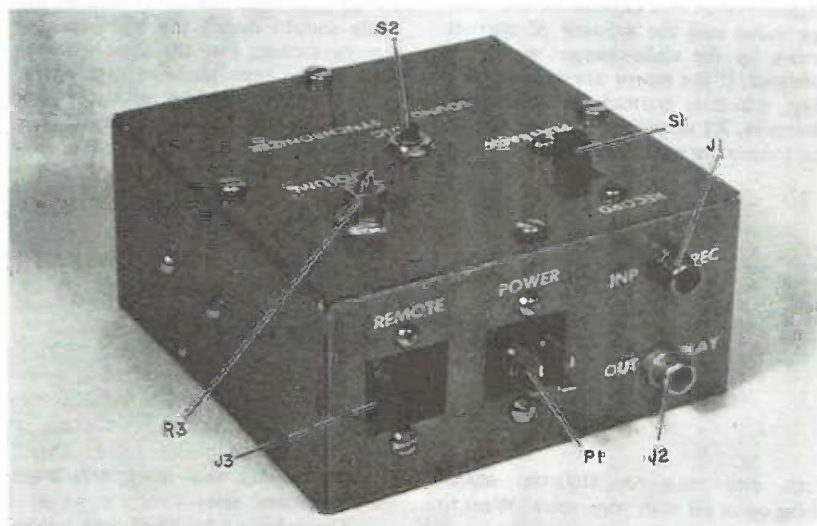
The cable connecting J1 to the recorder microphone input is now used

to connect J2 to the external speaker jack of the recorder. (This jack must be on the same channel the sync signals were recorded on.) S1 on the tape/slide synchronizer is set to the playback function, and the recorder controls are set for stereo (or split-channel) playback.

When you play back the tape, you'll hear commentary as usual through the tape speaker. But the sync-channel speaker should be muted, and the sync signal goes into J2 and the primary of T1. The signal is stepped up about 2.5 times by T1 and is then applied to the gate of the thyristor.

Note that closing either S2 or S3 will override all automatic controls and cause a slide change during either recording or playback.

This synchronizer can be used with nearly any slide changer with remote pushbutton operation, and with any stereo or split-channel tape recorder with power amplifiers. It cannot be used with a tape deck since preamp output isn't high enough to trigger the Triac. The SC40B requires 1.5 to 3



*Fig. 6—Remote jack J3 isn't required . . . it is useful for remote operation of S2. Locking control is used for R3 to avoid accidental misadjustment of sync level.*

volts and 25 to 100 mA for triggering, and will control up to a 6-amp load at 120 volts.

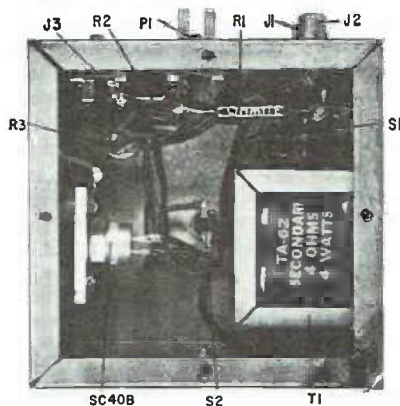
Transformer T1 isolates the power line from the recorder and gives a fair impedance match between recorder output and the Triac gate. In the record function the impedance match to the recorder input is not good, but this is not important as there is plenty of signal for recording.

### Construction

The parts layout is not critical as all circuits are low impedance. Figs. 6 and 7 are top and underside views and show the location of the few parts used.

The Triac is conveniently mounted on the small aluminum plate which is insulated from and fastened to the case with  $\frac{1}{2}$ " insulated, threaded spacers. Transformer T1 is mounted on the opposite side of the cabinet (Fig. 7). P1, J1, J2 and J3 are mounted on the front. As Fig. 6 shows. R3, S1 and S2 are mounted on the top of the case.

Resistor R1 should be chosen to produce a 1-volt sync signal. The resistance value will depend on the voltage value and the amount of current drawn by the slide-changer motor or solenoid. If the motor draws about 0.3 amp, which is normal for home-type changers, R1 should be 3 ohms. Check the nameplate on your projector to find



*Fig. 7—Parts placement isn't critical within the box as the circuit is chiefly low impedance. Mount Triac carefully.*

out how much current it draws. This figure may include the projector lamp wattage, which you must subtract. Better still, use an ac voltmeter across R1 and try different values until you come up with a 1-volt signal.

You will need a two-wire power cable with a receptacle to mate with P1 on one end. The other end of the cable should match the slide-changer remote socket. For the sync connections to the tape recorder, use phone plug on one end, and whatever you need on the other to match the recorder jacks.

## IC SOUND-OPERATED RELAY

By LYMAN E. GREENLEE

LIKE THE IDEA OF TURNING SOMETHING on or off with your voice? Want to control a transmitter or recorder so it operates only when you speak? For a very small outlay of cash and time, you

can do it with this integrated-circuit sound-actuated relay.

The relay circuit itself uses a conventional transistor, but what provides the audio amplification before that point

is a tiny integrated circuit, a Westinghouse WC 183, encased in a flat plastic package that measures 0.14 by 0.25 by .055 inch. At 4.5 volts supply, it draws less than 4 mA and gives a gain of around 90 dB. The IC, a differential amplifier with four direct-coupled stages, comes in two kinds of packages—the “flat-pak” (WC 183G) and a 12-lead TO-5 metal transistor-type case (WC 183T). The “G” is the smaller, and that’s the one I used.

### How the circuit works

Sound is picked up by the microphone or small speaker and fed into terminals 3 and 9 of the WC 183G (Fig. 1). The audio voltage across the secondary of the output transformer is rectified by a diode and charges a large low-voltage capacitor in the base circuit of a pnp transistor. When the charge across this capacitor builds up to about 0.3 volt, the transistor conducts and trips the relay, which then remains closed until the voltage across the capacitor has dropped to about 0.2 volt.

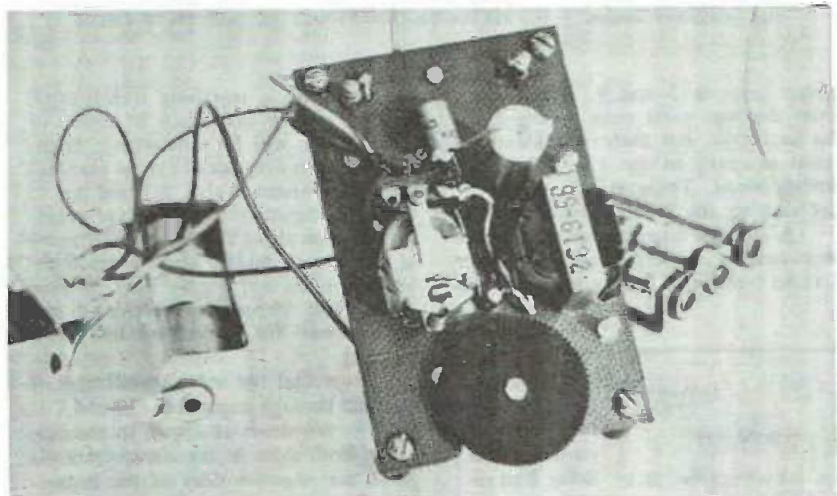
The delay must be long enough to allow for normal speech pauses. It can be adjusted by choosing the proper val-

ues for R1 and R2. R1 can be a fixed resistor for predetermined fixed delay, or the arrangement shown in Fig. 2 can be used for some control over the delay. If you use the transistor to drive the relay, capacitor C1 can be much smaller than if it were connected directly across the relay coil—500  $\mu$ F is about right. If C1 is too large, there will be too much delay in pull-in and the first syllable or so will be lost.

I used separate batteries for the amplifier and relay because of the tendency of the circuit to cycle itself on and off when the relay is operated from the amplifier batteries.

### Construction hints

The sound relay can be built inside most portable tape recorders. Components can be arranged to fit the available space. Layout is not critical, but input leads from the microphone (or speaker) used for sound pickup should be kept short to avoid feedback, hum and noise. If these leads are more than 2 or 3 inches long, use a two-conductor shielded mike cable. The shield goes to terminal 1 or ground and the two micro-



*This complete “chassis” is just a piece of hardboard measuring less than 2 x 3 inches.*





the device. If you check with your vtvm, you will find that the input terminals are slightly above ground. Westinghouse engineers suggest not using any external components that will alter this bias voltage.

C2 and C3 are 10- $\mu$ F decoupling capacitors. Solid tantalums rated at 20 volts were used in the prototype, but any small low-voltage electrolytics should work equally well. Just be sure they are good before you use them.

Gain is controlled by a miniature transistor-radio volume control (R3). It can be bolted to the board that holds the other components. It will require adjusting for maximum sensitivity in different locations because background noise level varies. It can be turned up as battery voltage drops. Maximum gain is with pins 4 and 8 shorted (minimum resistance).

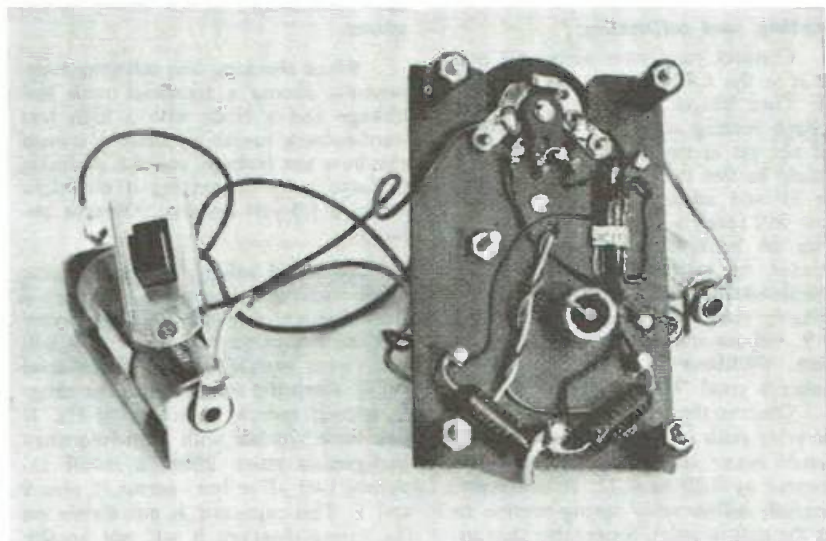
Output transformer T2 is a special modulation transformer for a 27-MHz CB transceiver (available from Lafayette Radio Electronics Corp., 111 Jeri-

cho Turnpike, Syosset, N. Y. 11791). The 3,000-ohm winding goes to the relay circuit, and the 8-ohm winding can feed a pair of low-impedance headphones for monitoring or testing.

The relay is made especially for radio control in model airplanes or boats and is available from any supplier of radio-control equipment. The contacts will carry 500 mA, which will do to start and stop a miniature tape recorder. If heavier currents are to be interrupted, or if higher voltages than 9 or 12 are to be used, an additional relay should be used in cascade to handle the extra load.

### Working with the WC 183G

Handle an IC with tweezers! The most convenient way to mount one is probably with a drop of cement. (Goodyear Pliobond cement is fine, and available at local Goodyear tire stores and many hardware stores.) Put one drop on the Bakelite board and another drop on the back of the WC 183G. Al-



*The other side of the board. Can you find the IC? It's the flat thing with the marking "WC183" barely visible—on the right side of the board just above center.*

low both to dry for a few minutes until the cement is tacky, then carefully and firmly press the IC against the board.

Carefully straighten and press each lead against the circuit board so that all are held down by the cement. Leads are identified by the black dot next to pin lead No. 1. Use phono pickup arm wire for connections. Use the smallest soldering pencil when soldering IC leads, get it *hot* and work fast.

Use nothing but the finest grade of rosin-core solder. First, tin all leads to be joined; then sweat them together with a quick dab of the hot iron. A clamp-on heat sink is handy for holding wires in place while soldering. Practice on some scrap pieces of wire until you can make a good clean sweated joint fast. Just a quick touch of a hot iron is all you need. After all the connections are soldered to the WC 183G, coat it with more cement to anchor everything firmly.

Connect the battery with the right polarity! IC's cost money. If you goof, you pay! Check all connections *before* you install the batteries.

### Testing and adjusting

Connect your vtvm across C1 and set it to the 1.5-volt dc range. Turn S1 off. Turn S2 on and take a no-signal voltage reading across C1 with R1 and R2 not yet in the circuit. This reading should be less than 0.1 volt. Now turn on S1 and advance the gain control. You will reach a point where the relay trips on background noise. With the speaker or microphone disconnected, you should be able to trip the relay by bringing your hand close to terminal 3 or 9, without actually touching either of them. (Without a mike, this device makes a good "touch" relay.)

Observe the voltage levels at which the relay pulls in and drops out. Pull-in should occur at 0.28 or 0.3 volt, and dropout at 0.20 to 0.22. If necessary, carefully adjust relay spring tension to set the pull-in point. Remember that increasing the spring tension will also shorten the time cycle. You can omit R2

if the background-signal voltage across C1 falls below the relay dropout point without it. This will depend on several factors and can be determined only by experiment.

If the background-noise level is high enough to trip the relay with R2 out of the circuit, connect a variable resistor across C1 and reduce the resistance gradually while observing the result with the vtvm. Choose a fixed resistor closest to the resistance that will hold the voltage across C1 to below the relay dropout point of about 0.2 volt. R2 should be from 10 to 50K ohms. If it is less than 10K, try a different transistor and use a rolloff capacitor across pins 5 and 7 of the IC.

Any resistance across C1 acts as a bleeder and reduces the sensitivity somewhat. Adjust R1 for the required holding period. If you need an adjustable time delay, use a 50K pot and series limiting resistor for R1 (Fig. 2). If you do not need the variable delay, make a temporary hookup like Fig. 2 to determine the value you want and then substitute the closest value in a fixed resistor. R1 is not shown in the photos.

When checking and selecting components, choose a transistor with low leakage and a diode with a high forward-to-back resistance ratio. C1 must also have low leakage, and it is desirable to form it by connecting it overnight across a 1.5-volt battery. (Observe polarity!)

To avoid talking yourself hoarse while making adjustments, connect a small speaker to your audio generator and use it as a sound source. This will give you a variable source of low-level audio. Keep the frequency in the range of normal speech, under 3,000 Hz. If you have trouble with high-frequency background noise, install a rolloff capacitor (.05  $\mu$ F or less) across IC pins 5 and 7. This capacitor is not shown on the wiring diagram; it will not usually be required, and it reduces sensitivity somewhat.

By connecting a pair of headphones across the 8-ohm secondary of T, you can monitor the signal and check for noise, motorboating, hiss, hum and lack of response from the amplifier. The headphone load will affect relay operation, so disconnect the phones while setting up the timing and using the sound relay. You can use the 8-ohm output to drive a miniature speaker also.

To make a recording with the sound relay, set the tape recorder on RECORD and adjust its volume correctly. Everything must be set ready to go. You will probably have to cut one of the battery leads to your recorder and run a pair of wires to the sound-powered relay contacts. The regular switch in the recorder will then be left on all the time. Add a shorting switch across the sound-

relay contacts to bypass them when the sound relay is not in use.

You may also want to install a switch to stop the recorder when the tape is all used up. On recorders in which the on-off switch is not a part of the record-playback transfer, all you will need to do is to hook the sound-relay contacts across the power switch. If you do that, include another switch in series to disable the sound relay in case the points should stick. This will also enable you to stop the recorder without disconnecting the sound relay.

All recorders take a moment to get up to full speed. Experience will tell you how to work with yours. You may need to say "uh" into the mike while the recorder comes up to speed.

## PART 5

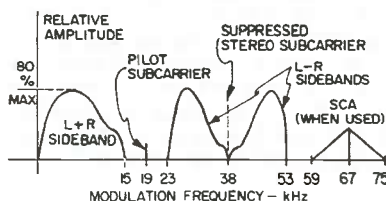
# Test & Measuring Equipment

### FM STEREO MULTIPLEX GENERATOR

by AL FRANSON

The stereo generator described here is designed for low cost and simplicity. It compares favorably to commercial units in stereo performance, but does not have self-contained audio signals or i.f. test signals. Only a 1-kHz signal and a general-purpose scope are required for aligning this generator. An FM receiver should also be available. The generator provides over 30 dB stereo separation between 300 and 20,000 Hz. Total parts cost is around \$50, but many parts are common and can be found in your spare parts box.

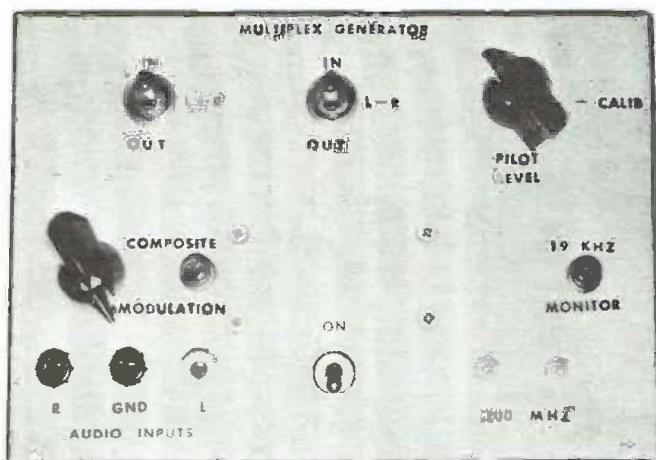
This unit does have an rf output to check an entire receiver's performance without an external rf generator. The high-frequency oscillator works at  $106 \pm 2$  MHz and is frequency modulated by a variable-capacitance diode. A regulated power supply is incorporated and gives exceptional rf stability.



**Fig. 1—FREQUENCY SPECTRUM in a frequency modulated stereo broadcast.**

#### Theory of operation

A basic understanding of FM stereo transmission will aid the user in understanding the generator operation and design theory. The FCC controls the method of stereo transmission. The base-band spectrum is shown in Fig. 1. The monaural information or average left plus right (L+R) signal is contained in the 0 to 15 kHz useful audio range. The





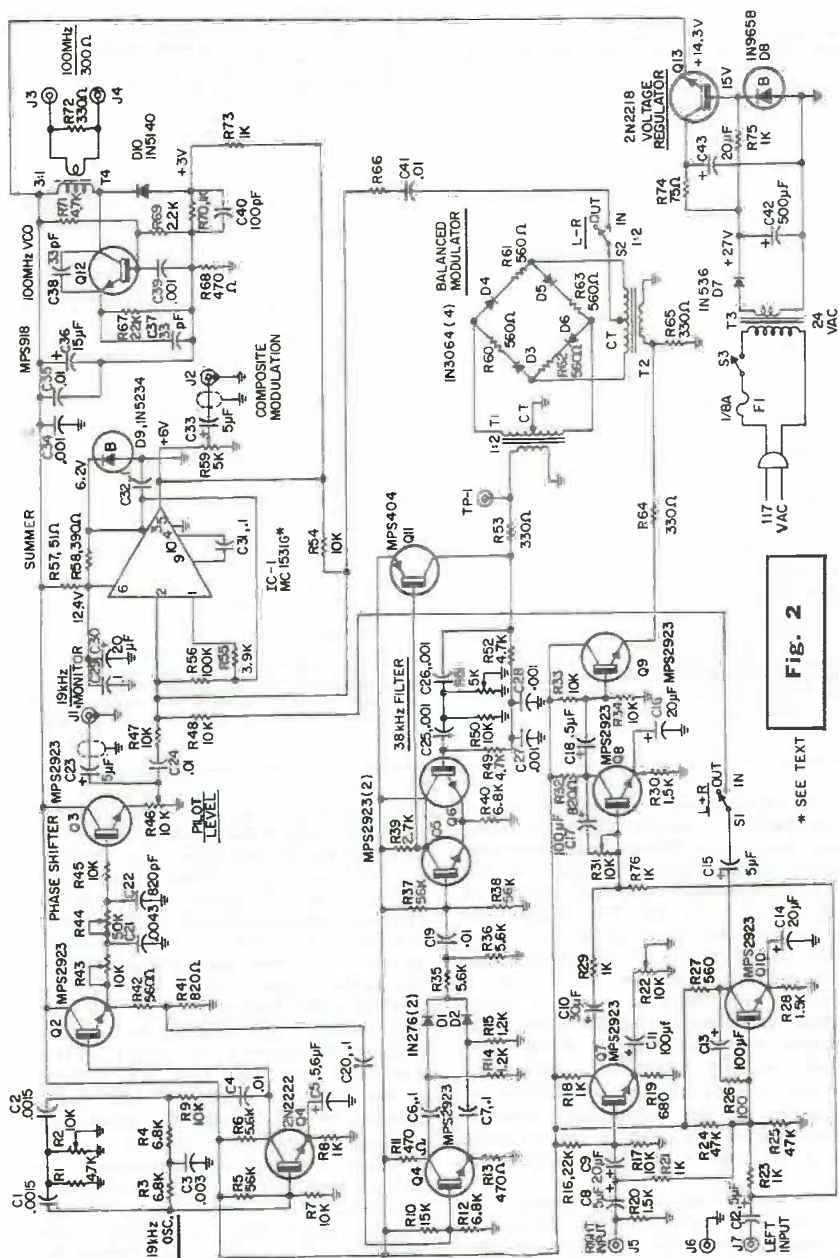


Fig. 2

\* SEE TEXT

**All resistors 1/4 watt, 5% unless noted**

R1, R49, R52, R71—4700 ohms  
 R2, R22, R31, R43, R46—potentiometer, 10,000 ohms, linear taper  
 R3, R4, R12, R40—6800 ohms  
 R5, R37, R38—56,000 ohms  
 R6, R35, R36—5600 ohms  
 R7, R9, R17, R33, R34, R45, R47, R48, R50, R54—10,000 ohms  
 R8, R18, R21, R23, R29, R70, R73, R75, R76—1000 ohms  
 R10—15,000 ohms  
 R11, R13, R68—470 ohms  
 R14, R15—1200 ohms  
 R16—22,000 ohms  
 R19—680 ohms  
 R20, R28, R30—1500 ohms  
 R24, R25—47,000 ohms  
 R26—100 ohms  
 R27, R42, R60, R61, R62, R63—560 ohms  
 R32, R41—820 ohms  
 R39—2700 ohms  
 R44—potentiometer, 50,000 ohms, linear taper  
 R51, R59—potentiometer, 5000 ohms, linear taper  
 R53, R64, R65, R72—330 ohms  
 R55—3900 ohms  
 R56—100,000 ohms  
 R57—51 ohms  
 R58—390 ohms  
 R66—27,000 ohms  
 R67, R69—2200 ohms  
 R74—75 ohms, 5 watts  
 C1, C2—1500 pF, 2%, mica  
 C3—3000 pF, 2%, mica  
 C4, C19, C24, C35, C41—.01  $\mu$ F, ceramic  
 C5—56  $\mu$ F, 15 volts, tantalum  
 C6, C7, C20, C29, C31, C32—.01  $\mu$ F, ceramic  
 C8, C12, C15, C18, C23, C33—5  $\mu$ F, 6 volts, tantalum

C9, C14, C16, C30, C43—20 F, 50 volts, tantalum  
 C10—30  $\mu$ F, 15 volts, tantalum  
 C11, C13, C17—100  $\mu$ F, 6 volts, tantalum  
 C21—4300 pF, 5% mica  
 C25, C26, C27, C28—1000 pF, 2% mica  
 C34, C39—1000 pF, ceramic  
 C36—15  $\mu$ F, 20 volts, tantalum  
 C37, C38—33 pF, 5% mica  
 C40—100 pF, 5% mica  
 C42—500  $\mu$ F, 30 volts  
 C22—820 pF, 5% mica  
 T1, T2—Three 60-turn windings on CF111-06 core (see text), Newark Electronics Stock No. 59F1510. \$2.05  
 T3—24 volts, 0.3 amp transformer (B.A. #13A903)  
 T4—3-turn primary, 1 turn secondary, #26 enameled wire, 3/16" dia form, brass slug  
 S1, S2, S3—spst toggle switch  
 D1, D2—1N276 diode, Hughes  
 D3, D4, D5, D6—1N3064 diode, TI  
 D7—1N536 rectifier, G.E.  
 D8—1N965B, 15-volt Zener, I.R.  
 D9—1N5234, 6.2-volt Zener, Mot.  
 D10—1N5140, Mot. or MV1624 (see text)  
 Q1—2N2222, Mot.  
 Q2, Q3, Q4, Q5, Q6, Q7, Q8, Q9, Q10—MPS2923, Mot.  
 Q11—MPS404, Mot.  
 Q12—MPS918, Mot.  
 Q13—2N2218, T.I.  
 IC-1—MC1531G op amp, Mot. or MC1709C (see text)  
 J1, J2—phone jacks  
 J3, J4—teflon feedthru  
 J5, J6, J7—bannana Jacks  
 F1—1/4 amp fuse  
 Case—3" x 5" x 7" aluminum minibox (Bud CU3008A)  
 Miscellaneous: 12 pin printed circuit connector, 300 ohm twinlead, fuse holder, 1/16" single sided p.c. board—4 1/2 x 6 inches.

L-R information required for stereo demodulation is transmitted as an amplitude modulated 38-kHz suppressed-subcarrier signal. This will result in frequency components  $\pm 15$  kHz around 38 kHz, or 23 to 53 kHz. The 38-kHz subcarrier must be reinserted in the correct phase at the receiver to obtain the complete stereo information. To do this, a low-level 19-kHz pilot signal is transmitted which has a definite phase relationship with the original 38-kHz subcarrier at the transmitter. This pilot phasing is one of the most important properties of the transmitted composite signal and is discussed in greater detail in the section on alignment procedures.

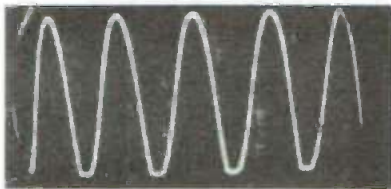
A few FM stations also transmit a 67-kHz storecast subcarrier. This signal is a frequency modulated subcarrier occupying the spectrum from 59 to 75 kHz.

This subcarrier was not included in this generator since it has nothing to do with aligning stereo separation of a receiver.

The composite baseband signal shown in Fig. 1 now frequency modulates the rf carrier between 88 and 108 MHz at a maximum deviation of  $\pm 75$  kHz peak. Broadcast stations use pre-emphasis of the audio inputs which increases the level of audio frequencies above 2 kHz. This makes a de-emphasis necessary in stereo FM receivers to restore correct frequency response. This pre-emphasis at the transmitter occurs before multiplexing. In other words, the subcarriers previously mentioned are not pre-emphasized. This makes it unnecessary to include pre-emphasis in a stereo generator which is used for alignment purposes. If pre-emphasis is desired, it can be applied to the audio input signals before they are fed to this generator.

### Circuit description

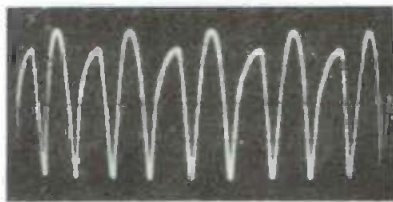
Transistor Q1 in Fig. 2 operates as a 19-kHz oscillator with the exact fre-



**Fig. 3—PILOT SUBCARRIER** signal generated in 19-kHz Twin-T oscillator Q1.

quency of oscillation determined by the twin-T feedback network. The twin-T network makes a very stable oscillator once R2 is adjusted for 19 kHz. The 11-volt peak-to-peak output of the oscillator (Fig. 3) is fed to a phase-shift network through buffer amplifier Q2. The phase-shifter is also isolated from the summer through buffer amplifier Q3. The phase-shifter provides a means of correcting phase relationship between the 19-kHz pilot and the 38-kHz subcarrier. The range of phase shift available is about 120°. A maximum phase shift variation of 90° is necessary.

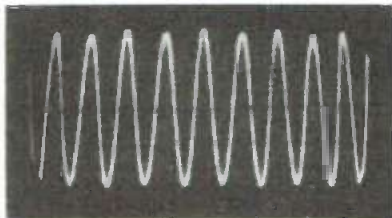
Transistor Q4 is the doubler amplifier which delivers two equal but opposite polarity signals to diodes D1 and D2 which full-wave rectify the 19-kHz signal. Germanium diodes are used for highest rectification efficiency. This distorted waveform (shown in Fig. 4) is



**Fig. 4—DISTORTED SIGNAL** results when 19-kHz subcarrier is rectified.

passed through an active bandpass filter consisting of Q5, Q6, and Q11. This is a twin-T filter which has excellent selectivity. The filter output is shown in Fig. 5.

The two audio inputs representing left and right channels must be conditioned to give L+R and L-R signals for proper stereo transmission. The left and right inputs are fed to the summer, Q10,



**Fig. 5—THE STEREO SUBCARRIER** is cleaned up by sharp 38-kHz Twin-T filter.

which is a feedback amplifier that provides greater than 20 dB isolation between L and R inputs. The actual isolation is partially determined by the generator internal impedances used to feed the L and R signal inputs. Source impedances below 100 ohms should be used to give greater than 40 dB isolation. Transistor Q7 is operated as an inverter to give -R at its output. This is summed with L by Q8 to give R-L at its output. R31 and R22 must be adjusted to give proper stereo separation.

The R-L signal is used to modulate the 38-kHz subcarrier delivered by Q11. This is done with a ring modulator with diodes D3-D6. This type modulator is doubly balanced with two center-tapped transformers so the 38-kHz component is canceled at the output as desired. The output of the modulator consists of the amplitude modulation components of the 38-kHz suppressed subcarrier. Resistors R60-R63 are used to improve balance or carrier suppression. The carrier is more than 40 dB below the input, which is adequate.

Transformer T1 must respond past 50 kHz without any resonances. *Most commercial audio interstage transformers do not meet this requirement.* Therefore, a special toroid transformer was designed. The construction details for this transformer are in Fig. 6.

Transformer T2 can be identical to T1. I used a commercial 2:1 center tapped transformer instead. At audio frequencies below 300 Hz the reactance of T2's primary introduces an undesired phase shift which results in decreased stereo separation. This can be improved only at the expense of a larger transformer winding or by reducing R65 and R64 which requires more transistor current. A higher current transistor could be used for Q9.

The total composite modulation consists of L+R audio, L-R information on the 38-kHz suppressed subcarrier, and the 19-kHz pilot signal. These are summed together by operational amplifier IC1. An op amp is used to give greater than 60 dB isolation between the three inputs. Summing resistors R47, R48, and R66 can be varied to adjust the individual gains of each input. The individual gain is equal to R54 divided by the series resistor. I used an MC1531G which is rather expensive. Newer and cheaper op amps are available, such as the MC1709C, and can be used instead.

The summer output modulates variable-capacitance diode D10, which is part of the resonant tank circuit of the 100-MHz oscillator. The diode, a Motorola 1N5190 Epicap, makes this a voltage-controlled oscillator which results in an FM signal output. The center frequency of the oscillator can be varied about  $\pm 2$  MHz by adjusting the slug in T4. My unit oscillated between 102 and 106 MHz. The exact frequency will be affected by stray wiring capacitance and layout. Be sure to prevent ground currents from modulating the oscillator. This is the reason for the decoupling consisting of R68, C35, and C36. The oscillator is especially susceptible to 19-kHz ground currents. Too much leakage will cause the pilot to be transmitted even though R46 is set at minimum. The oscillator output can be suitably loaded with a 300-ohm twin lead. Diode D10 can be any of a number of 10-pF units now available. One low cost unit is the MV1624 (\$1.42). The difference in operation between it and the 1N5140 (\$5.85) will be a slight difference in modulation sensitivity but should be negligible.

A regulated dc power supply and an ac rectifier circuit is used. The Zener diode reference provides a low output impedance. This is necessary to keep both oscillators at their correct frequencies and prevent power line modulation of oscillator Q12.

### Construction and alignment

The circuit layout is not critical except for the 100-MHz oscillator, for which component leads should be kept

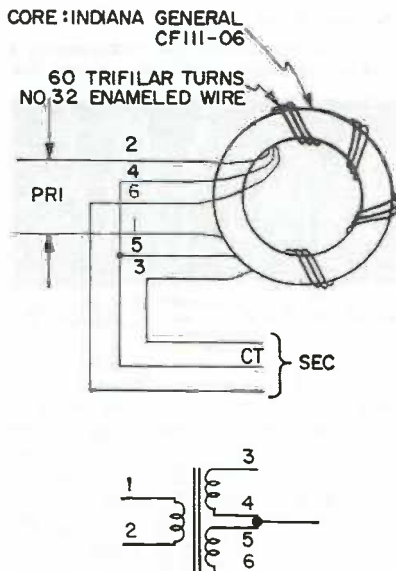
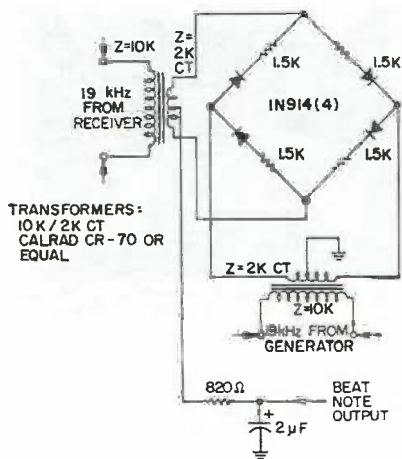


Fig. 6—TOROIDAL CONSTRUCTION increases bandwidth without resonances.

to an absolute minimum. A piece of 300-ohm twin lead connects the oscillator coil to the feedthru terminals in the chassis. The circuit board is laid out so it can be unplugged from the chassis. It is held in place by one screw and a threaded standoff to the chassis.

The first step in aligning this generator is to set the frequency of the 19-kHz oscillator. The easiest way to do this is to monitor the signal at J1 with an electronic counter while adjusting R2. Another way was devised which doesn't require a counter. A 19-kHz signal is available in all multiplex demodulators when the FM radio is tuned to a station broadcasting stereo. This signal can be used to compare with the generator pilot frequency using the simple phase detector shown in Fig. 7. R2 is adjusted until the phase-detector output contains a low-frequency beat note. This oscillator can be adjusted for a beat note of around 2 Hz which is plenty accurate—meaning the two frequencies are within 2 Hz of each other. One of the phase detector inputs should be greater than 4 volts p-p to turn the diodes on. The





**Fig. 7—SIMPLE PHASE DETECTOR eases adjustment of the 19-kHz oscillator.**

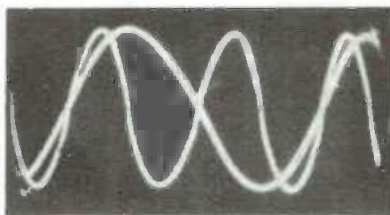
chassis cover should be in place for this test in case stray capacitance changes the oscillator frequency. This requires an access hole in the side or back of the unit for adjusting R2.

Next, the 38-kHz filter is aligned by adjusting R51 for maximum output at Q11's collector. Oscillation in this type of active filter is possible. Therefore, the 19-kHz oscillator should be disabled by shorting R7 and checking to see that no signal appears at Q11. The ac voltage at Q11's collector should be near 10 volts p-p and can be adjusted by selecting the value of R41.

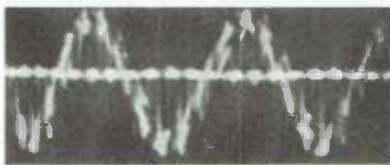
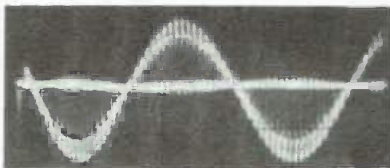
The next step is to adjust the phase relationship between the pilot signal and 38-kHz subcarrier. To do this, you must either sync the scope on one of the signals or use a chopper input on the scope if available. The chopped input method allows direct viewing of both waveforms simultaneously. I used another method. The scope is externally synchronized by the 38-kHz signal at test point TP-1. Then the 38-kHz waveform here is viewed on the scope and the scope settings adjusted so the sine-wave zero crossing occurs at the center of the grid. Next the scope input is placed at J1 and R3, R4 are adjusted for a zero crossing of 19-kHz at the same point as the 38-kHz waveform. This is shown by the

double exposure photo in Fig. 8. Now the pilot signal transmitted is in the proper phase relationship for stereo demodulation.

The audio mixing circuits must now be adjusted to give proper stereo separation. First apply a 1-kHz signal to the left channel. The pilot level is turned to minimum. The L+R and L-R switches must be closed. With the scope input at J2, adjust R31 until one of the 1-kHz envelopes is minimum in amplitude as shown in Fig. 9-a. Figures 9-b and 9-c

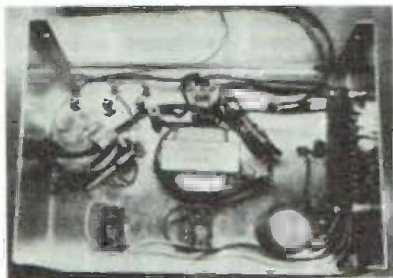


**Fig. 8—PILOT AND SUBCARRIER are in phase in this double-exposure photo.**

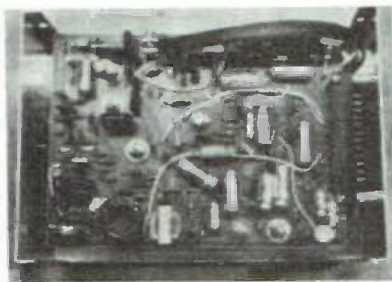


**Fig. 9—MODULATION WAVEFORMS. a—Left-only or right-only signal with 1-kHz input at zero pilot level. b— and c—Waveforms at J2 with right or left inputs at 500 and 5000 Hz, respectively.**





**POWER TRANSFORMER AND JACKS** are on front panel with the pots and switches.



**COMPONENTS ON PLUG-IN BOARD** are visible when the rear cover is removed.

show the same point for audio input frequencies of 500 Hz and 5 kHz respectively. These photos demonstrate a separation of about 40 dB. Next, remove the signal from L and apply to R only. A waveform similar to Fig. 9 will appear at J2 when R22 is adjusted for maximum separation.

The only calibration remaining is the 19-kHz pilot level required to give the proper oscillator deviation. I determined this with the help of an FM stereo receiver. For this measurement the L+R and L-R switches are in the OUT positions. Somewhere in the receiver multiplex you can check the 19-kHz level being received from the discriminator with a scope. Knowing this level, you can receive the signal from this multiplex generator and adjust the pilot level to equal that from a broadcast station. The exact level is not extremely important but some multiplex demodulators are sensitive to pilot level as separation is affected. I found that 0.2 volt p-p 19 kHz at J1 output is correct and should be accurate enough for alignment. At this point the knob on the PILOT LEVEL control was adjusted to read "CALIBRATE." I found it convenient to place an access hole in the chassis to permit external adjustment of T4's slug. This allows changing the oscillator frequency to a spot where no local station is operating.

### Using the generator

The composite modulation output is used mainly for applying the stereo signal to a multiplex demodulator for checking it separately. For this, the 19-kHz level is not calibrated and must be set to a predetermined level which the discriminator will supply from broadcast stations. This is also true for the remaining composite signal level. The audio input levels to the generator must be kept below about 1.5 Vrms or saturation occurs.

Receiver separation versus frequency can be checked between 300 Hz and 20 kHz with this generator. This also provides a convenient way to determine if the receiver de-emphasis circuit is correct. The easiest way to measure receiver separation is to apply an audio tone to only R or L of the generator and measure the ratio of outputs on L and R at the receiver.

The rf VCO in this unit is very convenient. It lets you check the effect of i.f. amplifier selectivity on overall separation. It also lets the technician adjust a multiplex unit incorporated in the radio without disconnecting it. A direct connection from the rf output is not required as the receiver will pick up the signal from a few feet away.

This stereo generator will result in a professional alignment job on any FM stereo receiver. Its relatively low cost and versatility make it a valuable piece of test equipment for any technician.

# FET DUAL-TRACE SCOPE SWITCH

by THOMAS B. MILLS

TO THE INDUSTRIAL ELECTRONICS maintenance man, the experimenter and the service technician the value of the oscilloscope is unquestioned. But its role in sniffing out faulty amplifiers, measuring phase shift and analyzing circuit operation can be greatly enhanced by adding a dual-trace switching unit.

Modern laboratory oscilloscopes, in the thousand dollar bracket, include dual vertical-input amplifiers with a choice of chopped-trace or alternate-trace sweeps, while some actually have dual-beam CRT's.

The cost of a laboratory version is not practical for most of us, but you can inexpensively add a chopped-trace capability to your present scope at the vertical input.

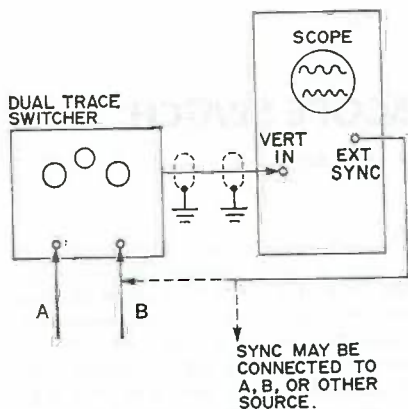
With the dual trace, or dual sweep, you will be able to look at two waveforms simultaneously. For instance, when testing amplifiers you

can observe the input and output together, instantly compare these signals and note if distortion or phase shift is present. For working with electronic counters and other digital circuits, a dual-trace presentation is almost indispensable to find out the relationship of one flip-flop output to another.

You might also use the dual trace for checking a modulation signal against a modulation envelope or for demodulation while testing a radio transmitter. Uses are limited only by the frequency response of the dual-trace switcher and oscilloscope. In some cases the frequency limitations may be circumvented by converting a high-frequency rf signal to a lower frequency within the bandpass of the amplifiers.

This switcher may be constructed in a small box external to your scope. It is connected to the scope as shown in Fig. 1. The two unknown signals are





**Fig. 1**—Inputs to switcher are combined and fed to scope vertical input. External sync is taken at input or source.

applied separately to the two inputs. The switcher output goes to the single vertical input of the scope, and a sync line from one of the inputs is taken to the scope's external-sync input.

Frequency response of the input amplifiers is 300 kHz over the range of the gain controls. With the gain controls wide open so no attenuation of the signal takes place, the frequency response is up to 1 MHz. The dual-trace switcher is easier to use if the scope has a triggered sweep to make syncing of the trace independent of the input frequency.

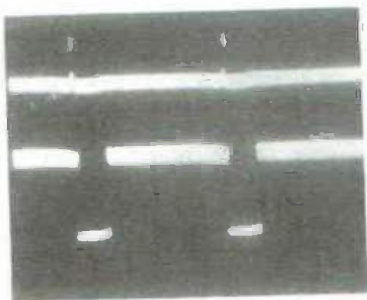
Two examples of dual-trace switcher use are in Fig. 2. Fig. 2-a is an oscillogram of a trigger pulse on the upper trace, which drives a one-shot multivibrator giving an output shown on the bottom trace. The sweep speed is such that the chopping transients appear as a broadened base.

A second example, Fig. 2-b, is an oscillogram of a sine-wave modulating signal on the top trace and a demodulated signal on the bottom trace of the original modulation taken from an rf transmitter.

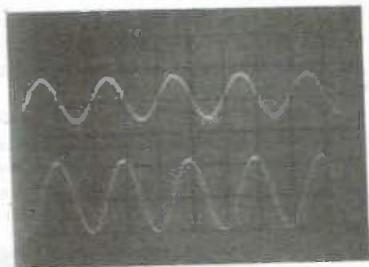
Fig. 3 shows a heterodyne zero beat in an FM signal that appears as a narrow band of horizontal lines run-

ning vertically from the negative sine-wave peak.

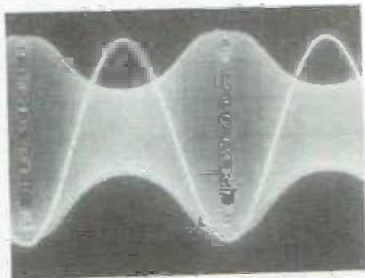
The superimposed sine-wave is the modulating signal that gives reference points for tuning to the maximum positive peak, maximum negative peak and carrier of the signal. In this way the peak-to-peak deviation can be measured with a heterodyne



**Fig. 2-a**—Trigger pulse in upper trace drives a multivibrator whose output (bottom) can be viewed simultaneously.



**Fig. 2-b**—Top trace is a sine-wave modulating signal, bottom trace is the demodulated signal from rf transmitter.



**Fig. 3**—Heterodyne zero beat in FM signal (vertical lines from negative peak).

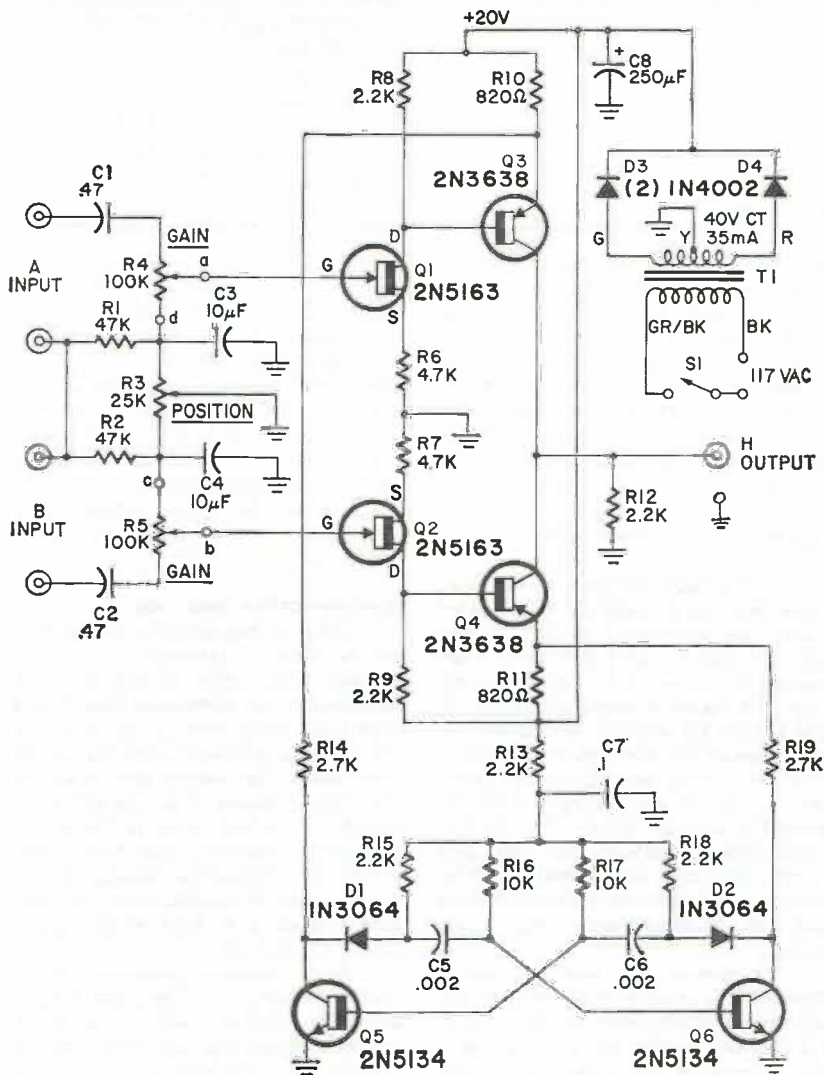


Fig. 4 (above)—FET's Q1 and Q2 boost inputs, which are chopped by Q3 and Q4 from Q5-Q6 pulses.

## PARTS LIST

### All resistors $\frac{1}{2}$ W, 10%

R1, R2—47,000 ohms  
R3—25,000-ohm linear potentiometer  
R4, R5—100,000-ohm linear potentiometer  
R6, R7—4700 ohms  
R8, R9, R12, R13, R15, R18—2200 ohms  
R10, R11—820 ohms  
R14, R19—2700 ohms  
R16, R17—10,000 ohms

### Capacitors

C1, C2—0.47  $\mu$ F, 600 volts  
C3, C4—10  $\mu$ F, 15 volts  
C5, C6—.002  $\mu$ F  
C7—.05  $\mu$ F  
C8—250  $\mu$ F, 30 volts

### Other parts

T1—Triad F-94X transformer or equiv.  
S1—spdt switch  
MISC— $3\frac{1}{2}$  x 7 x  $3\frac{1}{2}$ -inch cabinet (LMB W-2C),  
ac cord, knobs, terminal posts  
Kit with PC board and semiconductors only is  
available for \$10.00 from Solid State  
Services, 1720 Kimberly Dr., Sunnyvale,  
Calif. 94087

### Semiconductors

D1, D2—1N3064 diode or equiv.  
D3, D4—1N4002 diode or equiv.  
Q1, Q2—2N5163 transistor, Motorola or Fair-  
child (selected for  $I_{DSS}$  match)  
Q3, Q4—2N3638 transistor, Motorola or Fair-  
child  
Q5, Q6—2N5134 transistor, Motorola or Fair-  
child

converter, where the converter oscillator is on a low frequency that can be easily measured. The oscillator deviation times the harmonic number is the measured signal's deviation. The amplitude of the rf varies in the oscillogram, not because of modulation, but because of the bandpass of the low-frequency amplifier used after the converter.

## How the switcher works

A schematic diagram of the dual-trace switcher is shown in Fig. 4. Two inputs are attenuated by potentiometers R4 and R5 to a level within the range of the switcher's amplifier circuit. The signal is amplified by the input FET's, Q1 and Q2, and applied to the bases of Q3 and Q4, which act as switches. These are alternately gated on and off by the gating waveforms from Q5 and Q6. (Note that the diagram does not indicate the drain and source terminals on Q1 and Q2. This is because the devices are bidirectional and may be placed either way in the circuit.)

Transistors Q5 and Q6 are a free-running multivibrator whose frequency is determined by R16, R17, C5 and C6. Diodes D1 and D2 allow the voltage at the collectors of the multivibrator to rise very rapidly and thereby improve the switching speed.

As an example of how switching is accomplished, consider Q5 on; its collector is therefore at ground. The

emitter of Q3 is at a potential below its base and is therefore off. At the same time, Q6 is off and its collector is "floating" so that no current flows in load resistor R19. The emitter of Q4 is, therefore, 0.6 volt higher than the base and this transistor is operating in its normal mode. The output from either Q4 or Q3 is developed across R12, depending on which transistor is on. The output voltage across R12 is fed to the input of the oscilloscope in the setup.

## Construction and use

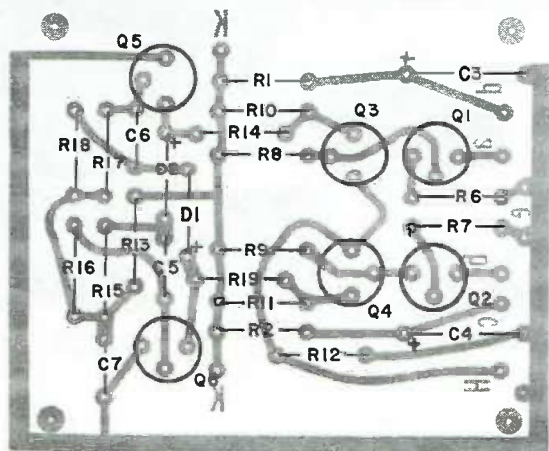
Most of the circuitry can be built on the same-size printed circuit board shown. The entire circuit may be mounted in an aluminum box. Fig. 5 shows an inside view of the box. The PC board is mounted vertically on the back panel. The output jack is on the rear panel, where it is convenient to attach the vertical input of the scope.

After assembly has been completed, an oscilloscope should show a square wave of approximately 20 volts peak-to-peak at 40 kHz on the collectors of Q5 and Q6.

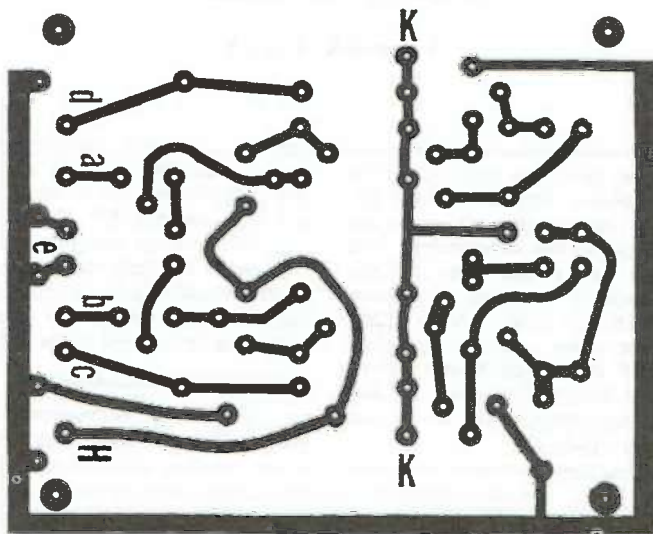
Next, connect a generator (either sine or square wave between 10 Hz and 100 kHz) to both inputs for a function check. The scope sync should always be obtained externally as shown in Fig. 1. A good sync source is at either input or the generator. The scope sensitivity should be set at or near maximum (0.5 volt p-p per inch or better).



Same size PC board and component placement are below.



Note that the line between the two inner-most input connectors must go to plus 20 volts as do points "K" on PC board. Points "e" on board are for ground connections.





**Fig. 5—**Chassis view with cover removed from the case shows PC board mounted on the back panel. You may want to mount the dual-trace parts inside your scope.

Operation of the position control should move the two traces on the screen to any desired position. Gain

controls R4 and R5 may be varied to produce a convenient display size on the scope.

## SCOPE CAMERA

by DALE E. COY

MOST EXPERIMENTERS HAVE AN OSCILLOSCOPE, yet nothing is quite so rare as the experimenter with an oscilloscope camera to keep a permanent record of the results of his experiments. One reason for this is the cost—a commercial camera designed for scope photography will make a \$400-\$1200 dent in the wallet, and can only take pictures of the scope screen. Much as we appreciate the value of this piece of equipment, few of us can justify the expense.

This section describes the construction of a simple, inexpensive scope camera which, like Superman, is often disguised as a mild-mannered camera for all the ordinary uses. You

can buy a Polaroid Swinger for about \$15. or the Big Swinger for around \$20. Conversion to a scope camera will cost another \$5.

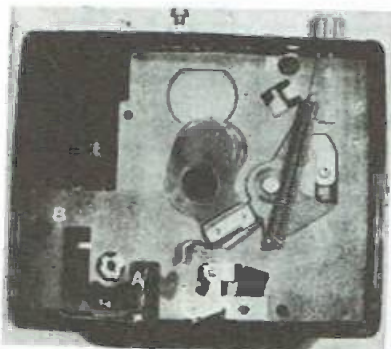
Three things are required to convert these cameras for scope photography; modifying the shutter, adding a supplementary lens; and a method for mounting the camera to the scope.

### Shutter modifications

The shutter mechanism for these cameras has one fixed speed of about 1/200 second, which is much too fast for scope photography. An added "bulb" position is needed. With this type of action, the shutter will open when the shutter button is pressed,

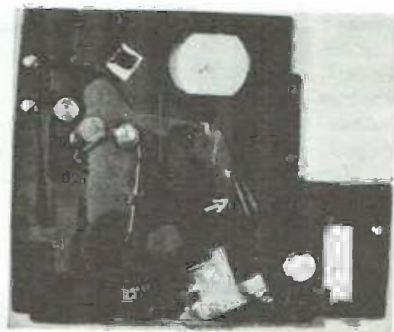
and will stay open as long as the button is held down. Since most scope photos require exposure times between 1 and 10 seconds, this is an easy shutter action to use. As an added bonus, the shutter mechanism does not have to be cocked between exposures, so multiple exposures may be made on each piece of film to place more than one trace on the picture.

To get to the shutter, first open the back of the camera and remove the battery compartment. Three small Phillips-head screws hold the lens-shutter housing to the rest of the camera. Two of them are chrome, and the other is black. Removing all of them lets the front of the camera drop off. Place the front housing on a clean cloth so the lens will not be scratched. Remove the plastic rear cover plate and the flashbulb ejector from the back of the shutter housing. You should now see something like Fig. 1.



*Fig. 1—Rear of the modified shutter. (A) is battery contact, (B) marks two screws holding shutter together. Lens opening is in the center of the plate.*

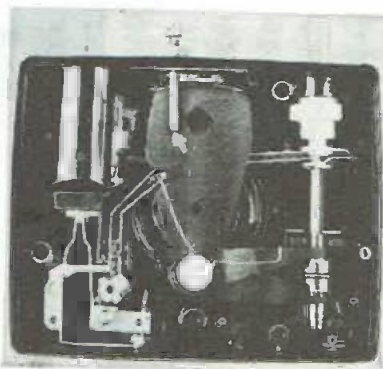
Although Fig. 1 shows the completely modified mechanism with the shutter open, it also illustrates the next operation. Remove the two screws at B. Grasp the tab at C with needle-nose pliers and pull gently upward until the plate snaps free. Then slide the plate out from under the battery contact at A, and turn it over.



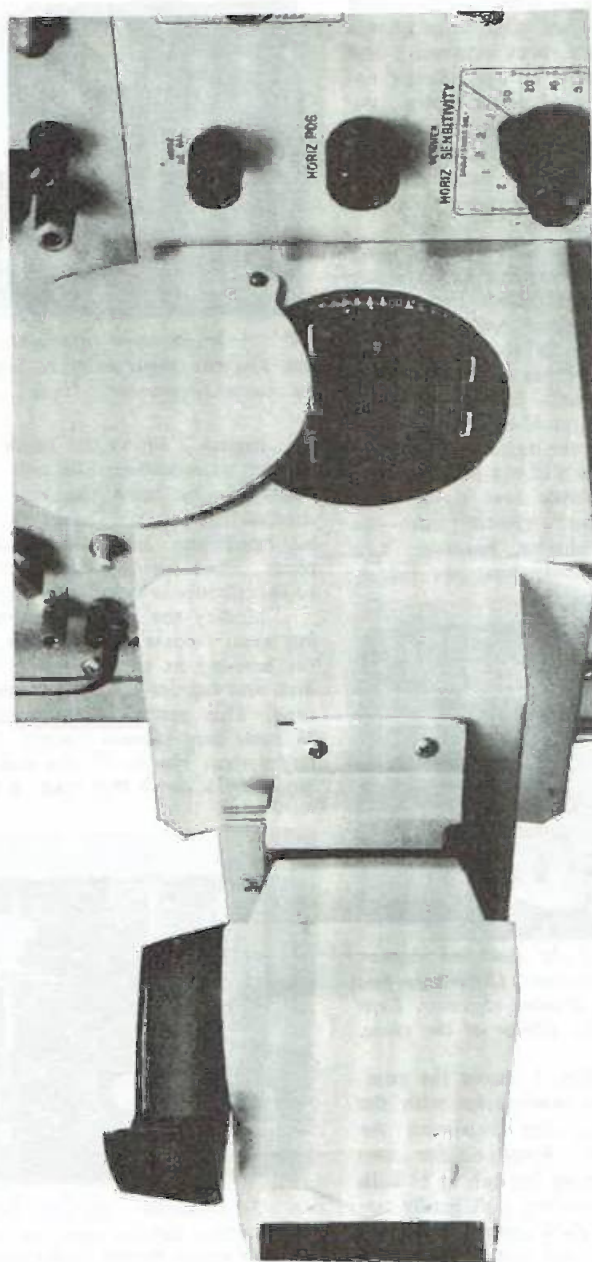
*Fig. 2—Inside view of modified shutter. The tiny arrow points to bend made in shutter to provide a "bulb" position.*

Figure 2 shows the inside of the shutter, as modified. The only change needed is to bend the end of the shutter leaf upward. Make the smallest bend you can with needle-nose pliers. Then file the bent part down so that about  $\frac{1}{16}$  inch extends upward.

Modify the plastic shutter housing next. Locate the spot on top of the housing as shown in Fig. 4, and drill and tap for the stop screw to be used. This screw must be at least  $\frac{3}{4}$  inch long (thread length), and no larger than No. 4. If you can find a No. 2 or 3 screw this long, it is even



*Fig. 3—Modified shutter housing. Added screw (arrow) stops the shutter leaf. The screw should lie flat and must be at least  $\frac{3}{4}$ " long (thread length).*



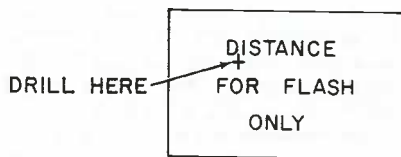


Fig. 4—Drill and tap for the stop screw between the D and I in DISTANCE.

better. Carefully drill and tap the hole so the screw will lie flat, as shown in Fig. 3. Since the top of the housing is angled, the hole will also have to be angled. Now insert the screw, and make sure it does not bind against the sliding blades as the red "brightness" knob is turned.

Put the pieces back together, pressing in at C (Fig. 1) to snap the plate back on. Install the screws. Press the shutter release a few times to check for binding inside. If there is some scraping or binding, take the

shutter apart again. Binding may be corrected by one of three methods: (1) filing the threads off the stop screw where it is inside the housing, (2) filing some more off the bent part of the shutter leaf, or (3) using thin shims or washers between the metal plate and plastic housing under the two screws at B in Fig. 1.

A few trial assemblies will correct any binding. When the shutter is assembled and working properly, hold the shutter release down and adjust the stop screw so that the shutter is fully open. For this operation, the red knob should be fully clockwise (away from the "darken" end).

This completes the hard part of the modification. You may want to mark the stop screw, or install a nut on the screw so that it can always be turned in to the same depth. By removing the screw and replacing it with a shorter screw (1/4 inch or less), the shutter is returned to its usual

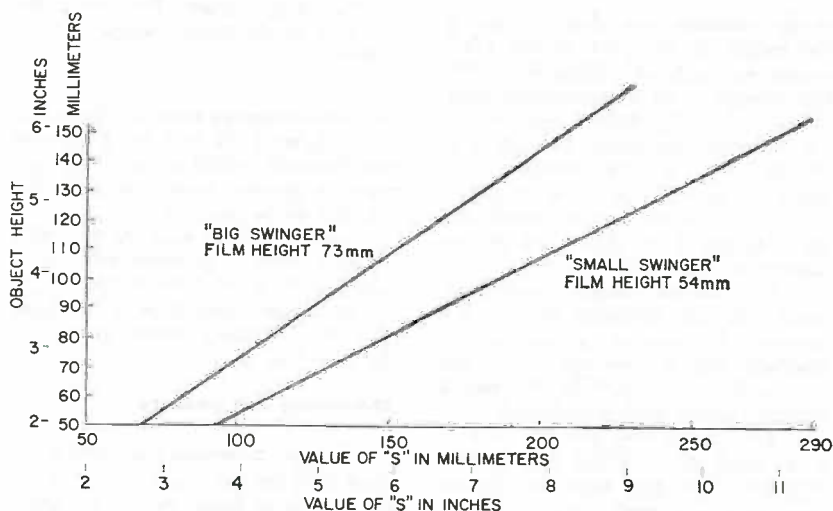


Fig. 5—Graph showing value of "S" (the focal length of the supplementary lens required and the focusing distance of the lens combination) for the 100-mm Swinger lenses. The actual focusing distance will usually be about 10% less than indicated on the graph.



operation and the camera will work normally.

Polaroid Corp. jealously guards the lens specifications for these cameras. A rough measurement shows, however, that the Swingers have f/16 lenses with a focal length of about 100 millimeters. Most lens measurements are traditionally made in millimeters (mm). There are about 25.4 mm to the inch, so the focal length of 100 mm means that the distance from lens to film is just shy of 4 inches. A supplementary lens is needed to let us focus the camera from a distance of something less than a foot, and to fill the picture with the image of the scope's CRT. Optics is a complicated subject, but since "rough" figures are all we need, we can use a "rough" formula:

$$S = \frac{(\text{Lens Focal Length}) \times (\text{Object Height})}{\text{Film Height}}$$

In this formula, the object height is the height of the part of the CRT screen we want to photograph. The film height is the top-to-bottom measurement of the film image (about 54 mm for the small Swinger and 73 mm for the Big Swinger). The value "S" is *both* the focal length of the supplementary lens we need, and the distance from the front of the camera to the CRT.

The object height chosen is usually not the diameter of the CRT because the pattern to be photographed usually does not go all the way to the top of the tube. Instead, a slightly smaller height is chosen.

To simplify the problem, Fig. 5 is a graph from which the correct supplementary lens may be chosen. The other necessary information is the diameter of lens needed. For the Swingers, the supplementary lens should be at least 20 mm in *diameter*, with the most convenient diameter for mounting about 28 to 30 mm. The best

place to obtain lenses is probably Edmund Scientific Co. You can obtain their free catalog by writing to 300 Edscorp Building, Barrington, N. J. 08007. The catalog lists a wide variety of planoconvex and positive meniscus lenses with prices around \$1, and a lot



*Fig. 6—Front of the modified camera. A plastic cable clamp holds supplementary lens in place. The added stop screw is on the top just to the right of center.*

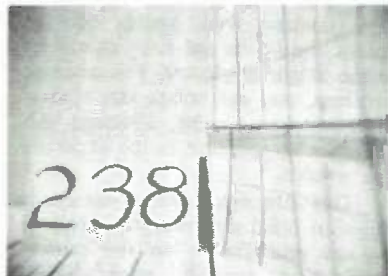
of other interesting gadgets. Their minimum order is \$2, so when you order your lens, also order a 2¼ x 3¼-inch piece of ground glass (No. 2143 for 50¢) for use in focusing.

Mounting the lens on the front of the camera is probably the easiest task of all. It may be simply taped to the camera lens tube or clamped on with a plastic cable clamp, as show in Fig. 6.

### Mounting the camera

Before mounting the camera, we must find the exact distance at which the lens is in focus. With the supplementary lens in place, install a piece of ground glass (or a piece of clear plastic sandpapered on one side) in the back of the camera, where the film is located. Open the shutter, and

move the camera until the image of some object (for instance, a ruler) is in focus on the ground glass. Then carefully measure the distance from the front of the camera to the object. This is the distance that the front of the camera must be mounted away from the oscilloscope. An alternate method of measurement is to take a picture of a test chart as shown in Fig. 7.



*Fig. 7—Test of the camera with a 238-mm supplementary lens. Camera focuses at slightly less than 9 inches and covers a field approximately 4 x 5 in.*

The actual distance from the front of the camera to the CRT should be slightly less than the distance measured, because the scope image is actually formed on the *inside* of the CRT. For most oscilloscopes, the image formed is actually about  $\frac{1}{4}$  inch *behind* the front panel.

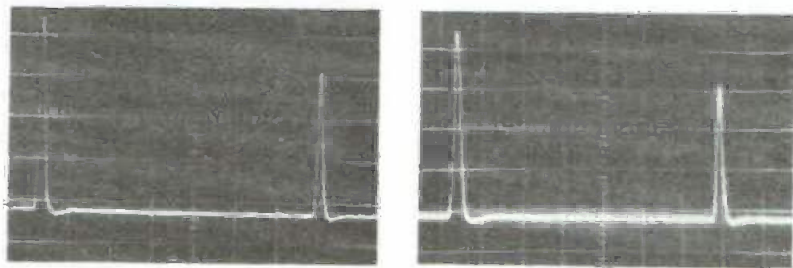
The scope mount is made from aluminum chassis and boxes, as shown

in the photo. Since scopes vary in size and style, the final design may not match the picture shown, which is for a 5-inch Hewlett-Packard scope. The mount is held in place by the oscilloscope bezel, located inside the box. The mounting screws pass through the bezel and through the mount, and are fastened to the front panel. The box is painted flat black inside to prevent reflections. Viewing holes with sliding covers are provided in each side, so that the screen may be seen. The covers are closed before the picture is taken. The piece of felt between the two boxes shown also covers a slot which may be used for viewing. The camera is fastened to the second box by two angle brackets bent from aluminum. The lens projects through a hole cut in the box.

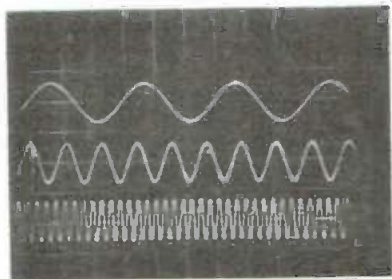
When the mounting is finished, check the focus by using a piece of ground glass inside the camera as outlined above. If the image of the scope trace is in focus, the camera is complete. Small adjustments in focus may be made by adding felt pads between the camera and the mounting, if the distance was made too short.

#### **Using the camera**

Using the oscilloscope camera is simple. Just set up the picture on the screen, close the viewing ports and hold the shutter open for a few seconds. Generally, exposure times will run between 1 and 10 seconds. Many



*Fig. 8—Comparison of waveform photos (left) taken with a \$500 professional oscilloscope camera and (right) \$20 Swinger modified for scope photography.*



*Fig. 9 Multiple exposure made by changing scope's input and trace position. This is an easy way to display waveforms when comparisons are required in the lab.*

things affect this exposure time, including the design of the oscilloscope, the frequency of the signal displayed and settings of the scope controls. The advantage of using Polaroid equipment is the ability to change exposure times and to see the results in just a few seconds. Because of the wide variation in setups, more specific exposure data cannot be given here. Shooting one roll of film with your equipment will make you an expert.

The scope camera may be used to make a permanent record of your experiments, to record transients and to photograph events which happen too fast or too slow for the eye to detect on the oscilloscope screen. Its versatility is limited only by your imagination. Samples of the work done by this camera are shown in Fig. 8 and 9. Quality compares favorably with equipment costing 20 to 30 times as much. For most pictures, the "darkness" control (red knob) on the camera should be set fully clockwise for the brightest picture. For extreme sharpness, you can rotate this control counterclockwise a bit, although this will cause an increase in the exposure time needed.

This inexpensive scope camera is simple to build and performs well. If you are reluctant to do the camera modification yourself, most camera repair shops will do it for a few dollars. By removing the supplementary lens and the stop screw, the camera is returned to service taking snapshots. This kind of bargain is hard to pass up.

## 3-WAY SCOPE CALIBRATOR

By JAMES ROBERT SQUIRES

ONE OF THE MOST VALUABLE COMMODITIES in a shop or laboratory is bench work space. I often find some little device that I want to build but never do because of the space it will rob from my workbench. For some time I've needed an oscilloscope calibrator like this one but I didn't decide to build it until I realized that the proper place for it was *inside* the scope.

Space requirements are small: A few square inches of front-panel space and less than 12 cubic inches of elec-

tronics for voltage and frequency calibration and a variable-delay trigger pulse with a full spectrum of scope uses.

The stick-anywhere calibrator system consists of three units. Fig. 1 shows the Box, the power supplies and the five panel controls. With the new miniature controls and switches, a minimum of scope front-panel space is needed. If space is not available on the scope's front panel a separate panel can be built.

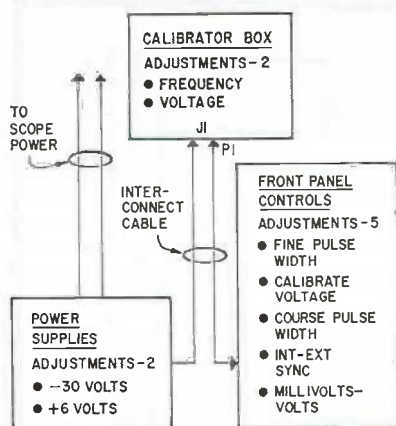
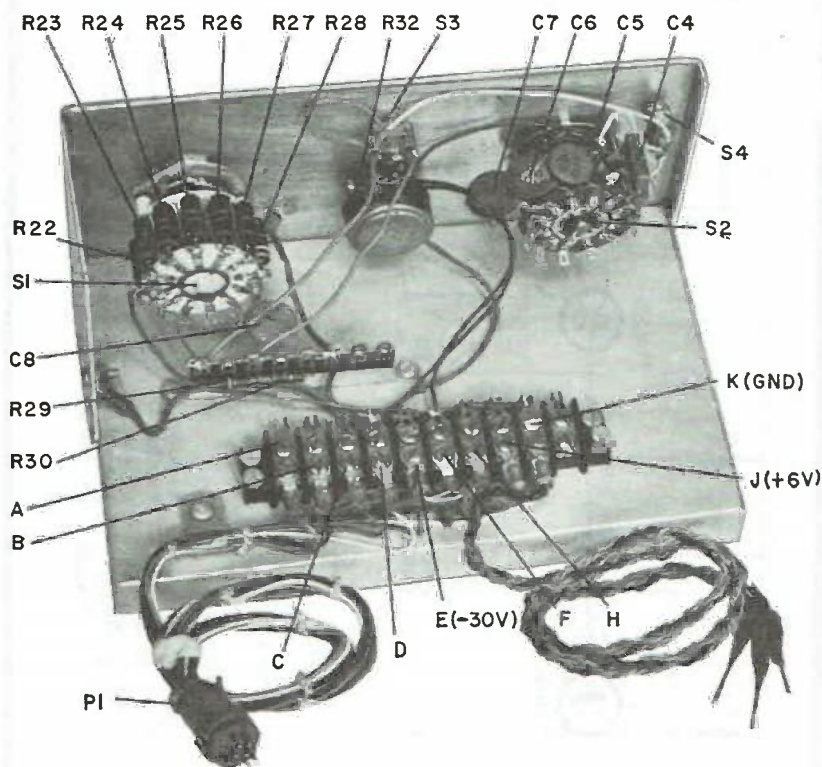


Fig. 1—Calibrator block diagram. Sections can be built into scope to save space.

Actually, the Box (Figs. 2 and 3) provides a calibration pulse variable in both frequency (300 to 3000 Hz) and voltage (16 Vdc to 25 Vdc), and a delayed trigger pulse. The delayed-trigger-pulse generator can be synced to either the internally generated calibrator pulse or to a negative-going external sync pulse of 30 volts or less. The external sync pulse may be a sine square or triangular wave. Fig. 4 gives pulse relationships.

The calibrator pulse is used to examine scope sweeps as a general check of frequency (horizontal time per division) and voltage sensitivity (vertical volts per division). This calibration pulse may also be passed through a positive diode clamp and applied to the Z-axis of the scope (the cathode of the



Controls may be mounted on panel scope or in a separate unit as shown here.

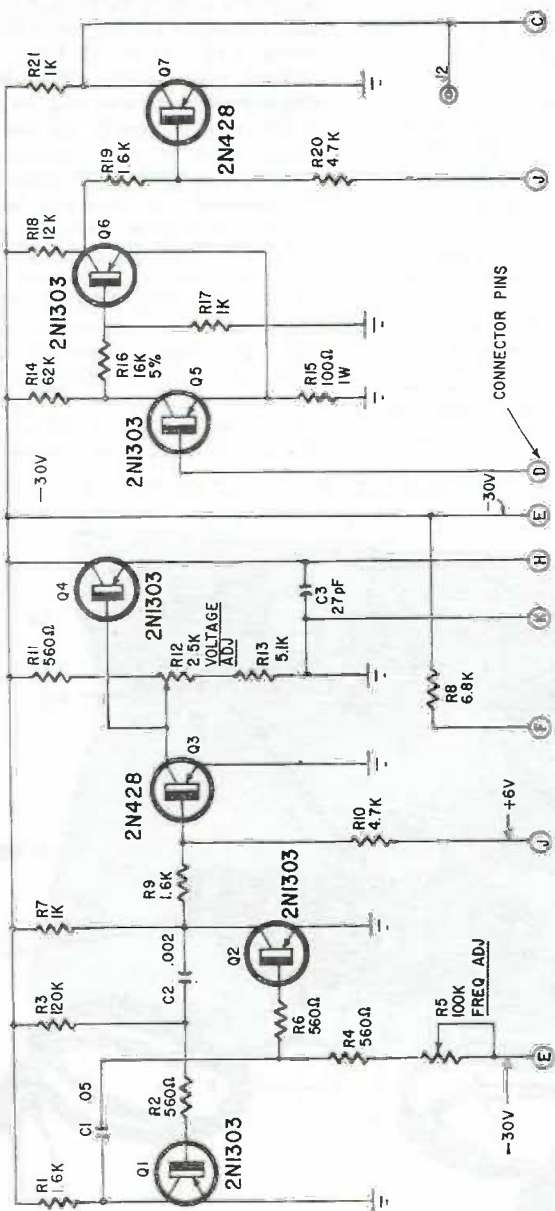
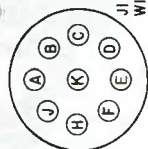


Fig. 2—Most of the components in the calibrator are in a small box and are connected to front panel controls through a cable.

F — OUT TO FINE WIDTH ADJ — GREEN  
H — CALIBRATOR VOLTS OUT — BROWN  
J — + 6 VOLTS IN — PURPLE  
K — GROUND — WHITE

A — BLANK — GRAY  
B — BLANK — BLACK  
C — DELAYED TRIGGER OUT — RED  
D — DELAY CIRCUIT IN — YELLOW  
E — -30V IN — BLUE





CRT) to provide variable blanking for timing and sweep marking purposes (see Fig. 5).

Either a simple Zener regulator or a more involved series and shunt regulator is the power supply. Fig. 6

gives data for using the Zener regulator at various voltages tapped from the scope supply. Assuming that you find a 60-volt tap in your scope power supply, you would use a 30-volt Zener in series with an 1800-ohm resistor and a

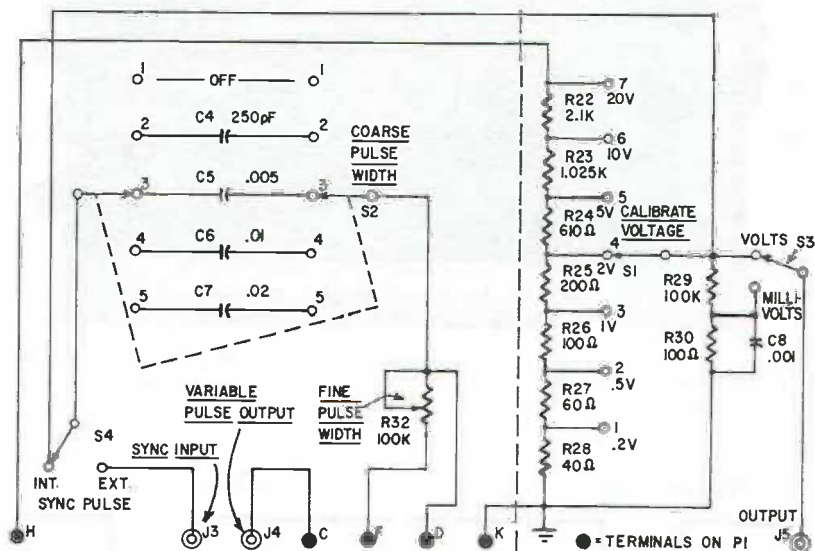
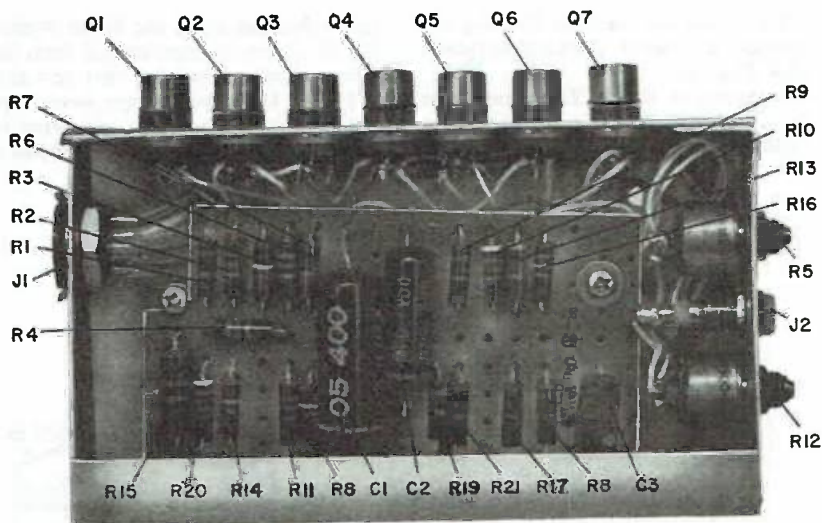


Fig. 3—Pulse-width and voltage-calibrator controls are mounted on front panel.

#### Parts List

C1—.05  $\mu$ F capacitor  
 C2—.002  $\mu$ F capacitor  
 C3—27 pF capacitor  
 C4—250 pF capacitor  
 C5—.005  $\mu$ F capacitor  
 C6—.01  $\mu$ F capacitor  
 C7—.02  $\mu$ F capacitor  
 C8—.001  $\mu$ F capacitor  
 R1, R9, R19—1600-ohm, 5%  $\frac{1}{2}$ -watt resistor  
 R2, R4, R6, R11—560-ohm,  $\frac{1}{2}$ -watt resistor  
 R3—120,000-ohm,  $\frac{1}{2}$ -watt resistor  
 R5—100,000-ohm miniature potentiometer  
 R7, R17, R21—1000-ohm,  $\frac{1}{2}$ -watt resistor  
 R8—6800-ohm,  $\frac{1}{2}$ -watt resistor  
 R10, R20—4700-ohm,  $\frac{1}{2}$ -watt resistor  
 R12—2500-ohm miniature potentiometer  
 R13—5100-ohm,  $\frac{1}{2}$ -watt resistor  
 R14—62,000-ohm,  $\frac{1}{2}$ -watt resistor  
 R15—100-ohm, 1-watt resistor  
 R16—16,000-ohm, 5%,  $\frac{1}{2}$ -watt resistor

R18—12,000-ohm,  $\frac{1}{2}$ -watt resistor  
 R22—2000-ohm, 5% 1-watt resistor, see text  
 R23—1000-ohm, 5%, 1-watt resistor, see text  
 R24—610-ohm, 5%, 1-watt resistor, see text  
 R25—200-ohm, 5%, 1-watt resistor, see text  
 R26—100-ohm, 5%, 1-watt resistor, see text  
 R27—56-ohm, 5%, 1-watt resistor, see text  
 R28—39-ohm, 5%, 1-watt resistor, see text  
 R29—100,000-ohm, 5%,  $\frac{1}{2}$ -watt resistor  
 R30—100-ohm, 5%,  $\frac{1}{2}$ -watt resistor  
 S1—2-pole, 7-position miniature non-shorting rotary switch (see lead photo)  
 S2—2-pole, 2-section, 5-position miniature non-shorting rotary switch  
 S3, S4—miniature spdt toggle switch  
 J1—miniature 9-contact socket connector  
 J2—insulated tip jack  
 J3, J4, J5—Miniature coax receptacles  
 Q1, Q2, Q4, Q5, Q6—2N1303 transistor  
 Q3, Q7—2N428 transistor  
 P1—Miniature 9-contact plug connector to match J1  
 MISC—Transistor sockets, terminal strips, hook-up wire, knobs



Interior of calibrator box. Transistors are in sockets mounted on top edge. Mounting board is  $1 \frac{3}{8} \times 3 \frac{5}{16}$ ", with  $\frac{1}{4} \times \frac{3}{4}$ " corner section removed.

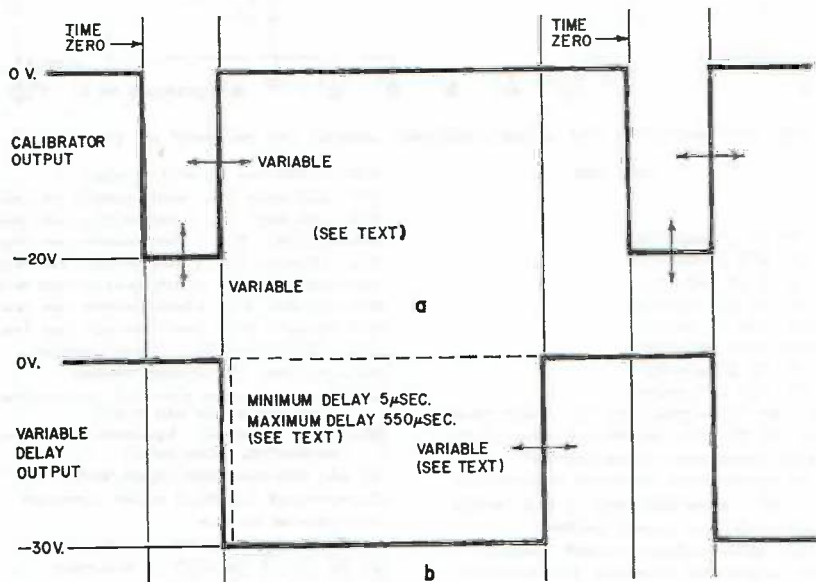


Fig. 4—Calibrator output waveforms. The one at a is a 1000-Hz calibration pulse;  $150 \mu\text{sec}$  ON and  $850 \mu\text{sec}$  OFF. Its amplitude and pulse duration are variable. At b the 30-volt trigger pulse may be delayed over an adjustable range of 5 to  $550 \mu\text{sec}$ .

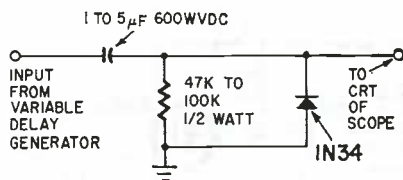


Fig. 5—The pulse output of the calibrator may be used for scope Z-axis blanking after being passed through a diode clamp.

6-volt Zener in series with a 3300-ohm resistor.

Figure 7 illustrates a complete -30-volt and +6-volt combined supply that could be installed within the scope. For pulse work this sort of supply gives much better high-frequency regulation and transient response than does a Zener regulator.

Panel controls are mounted wherever possible on the front panel of the scope. The photos show these controls in open breadboard construction. Follow this arrangement if you build the calibrator as a separate unit.

A 5" x 7½" piece of aluminum sheet is used for the chassis. You may want to look under the dust cover of your own scope to see just how much space is available to house the calibrator box. Location within the scope is not critical so long as the circuit card and other components are shielded by the metal chassis. Should you find that you have the room, you could purchase a larger chassis.

The circuit card is cut from Vector prepunched board. As the components are inserted into the board, their leads extend through the card to the other side and are used in wiring the circuits.

Transistor sockets are mounted on the Box chassis with mounting rings. Orient the sockets with the emitter pin nearest the bottom or open side. This facilitates testing when the unit is completed. Use No. 24 multi-strand wire to connect the transistor

sockets to the circuit card to minimize wire breakage during assembly.

### Calibrator box wiring

First connector J1 is wired, using nine 8" lengths of stranded wire color-coded as shown in Fig. 2. Solder 5" lengths of wire to the transistor sockets and fold out of the way for the present. Mount and wire the two potentiometers and test joint J2 as completely as possible.

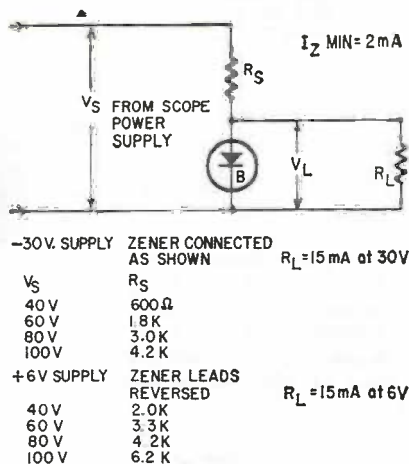
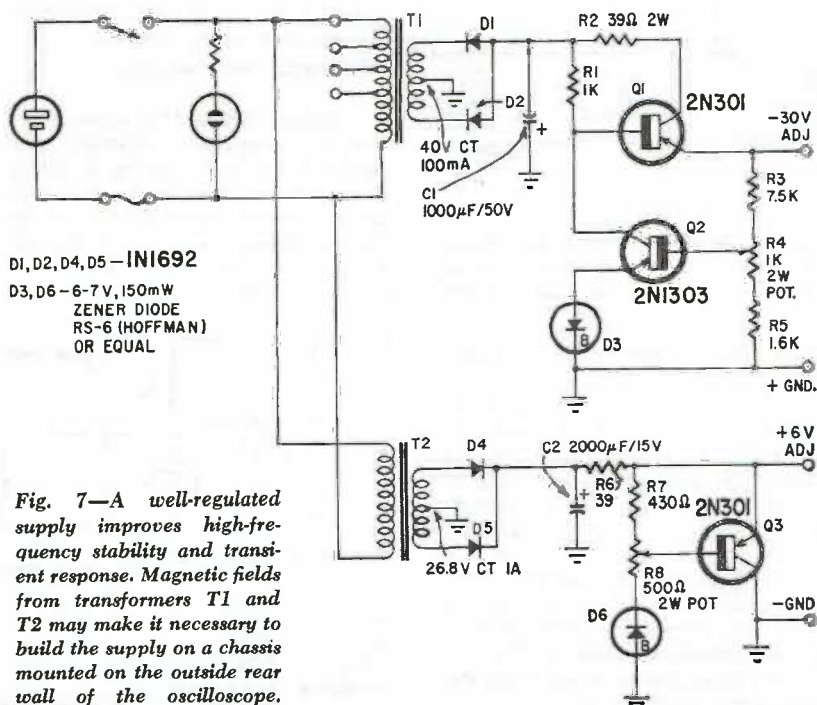


Fig. 6—How to obtain Zener-regulated -30 and +6 volts from scope's power supply. The Zener's cathode always connects to the positive side of the source.

Next the components are inserted into the circuit card and wired according to the photograph.

When the card has been wired and is ready to be mated to the Box, it should be placed component side down above the transistor sockets. All wiring between J1, the circuit card, the sockets and the potentiometers should be made from this position. Wiring the card in this manner permits the card to be swung out of the way for servicing. When all wiring is completed, tuck any loose wires under the card and assemble with two 6-32, ½" screws.



#### Parts List

- T1—Power transformer, 40-V ct, 100 mA (Triad F90-X or equiv)  
T2—Power transformer, 26.8 V ct, 1 amp (Triad F40-X or equiv)  
D1, D2, D4, D5—1N1692 diode  
D3, D6—6-7-V, 150-mW Zener diode (Hoffman RS-6 or equiv)  
C1—1000-µF, 50-V electrolytic capacitor  
C2—2000-µF, 15-V electrolytic capacitor  
R1—1000-ohm, ½-watt resistor  
R2, R6—39-ohm, 2-watt resistor  
R3—7500-ohm, ½-watt resistor  
R4—1000-ohm, 2-watt potentiometer  
R5—1600-ohm, ½-watt resistor  
R7—430-ohm, 5%, ½-watt resistor  
R8—500-ohm, 2-watt potentiometer  
Q1, Q3—2N301 transistor  
Q2—2N1303 transistor

When choosing a place to mount the calibrator, remember that the frequency and voltage controls must be easy to reach and adjust. The inexpensive transistors used in this unit must be kept away from heat. Heat arrives

by conduction, convection or radiation, so mount your transistors accordingly.

The emitter load of Q4 (Fig. 2) is made up of seven resistors (R22–R28, Fig. 3). The total value of these resistors is 4135 ohms. To produce the voltage steps of 20-, 10-, 5-, 2-, 1-, 0.5- and 0.2-volt, 1-watt carbon resistors are altered to give the exact resistances needed.

Using a three-cornered file, very carefully file through the phenolic outer insulating body of the resistor until the shaft of carbon is exposed. With a good ohmmeter, measure the resistor value in ohms. For example, let us use R24, a 610-ohm resistor. A 5% 610-ohm resistor was purchased; this means that the resistor might have any value between 580 and 640 ohms.

However, I have found that many 5% resistors are right on the money at 600 ohms. Let us assume the resistor you get is 600 ohms. You have measured its value on your ohmmeter and affirmed this. You have, using your three-cornered file, cut through the phenolic body of the 600-ohm resistor.

But R24 must be 610 ohms. File into the carbon of the resistance element a few strokes. Measure the resistance again. Notice that the resistor increases its value. Continue filing with the finesse of a brain surgeon until the higher value is obtained. Work slowly so as not to heat the carbon as you work on it. As you work, continuously check the zero of your ohmmeter. Alter all resistors requiring a resistance increase.

Careful work at this stage will result in a series of resistors providing exact voltage division ratios. As each resistor is filed to the needed value, apply a dab of Duco cement to the cut so that moisture will not cause future damage to the resistor.

It is true that the power rating of the resistors has been reduced, but that is why 1-watt units were used. Only  $\frac{1}{2}$ -watt units were needed, and the minute filing did not derate the resistors below that.

Use special care when installing these units as their body has been weakened. Once they have been assembled between decks of S1, you have a strong rigid assembly.

Remember that the filing technique only increases the resistor's value and do not forget to seal the cut! This may sound like a piece of work but conservatively speaking it can save more than \$5 for 1% precision resistors to gain the same ratios. If you have access to a precision or lab-type bridge, so much the better.

## Operation

The voltage calibrator consists of a variable-frequency multivibrator, Q1 and Q2, driving Q3, a variable-voltage

switch. The output of this switch is then fed to emitter follower Q4. Emitter resistance is ratio-divided to provide voltage ratios of 1,  $\frac{1}{2}$ ,  $\frac{1}{4}$ ,  $\frac{1}{10}$ ,  $\frac{1}{20}$ ,  $\frac{1}{40}$  and  $\frac{1}{200}$  to ground. When the voltage-control potentiometer (R12) is adjusted so that there are 20 volts at the emitter of Q4, the voltages are as noted on the schematic of Fig. 3. These dividing ratios apply for any voltage appearing at the emitter of Q4. S3 provides additional division either by 1 or 100. The ratios then become 1/100, 1/200, 1/400, 1/1000, 1/2000, 1/4000 and 1/20,000. In other words, the calibrate pulse output is in either volts or millivolts.

The variable-pulse generator is triggered either internally by the variable-frequency multivibrator or from an external negative-going square-wave or sine-wave trigger. Both course and fine control of the variable-pulse generator is available.

Pulse width can be varied from a few microseconds to 550  $\mu$ sec. Pulse widths for this Schmitt trigger depend to some extent on the input trigger waveform. Therefore no data are given for width values on switch S2. Output of the Schmitt trigger is buffered by Q7, a fast switch. Fig. 4 illustrates the time-pulse relationships available within the Box.

## Adjustments may be needed

Most scope calibrators operate at 1000 Hz. The variable-frequency multivibrator used in the Box covers from 400 to 4000 Hz. Frequency adjustment should be made against a known frequency standard such as a 1000-Hz crystal. Approach the setting from the high-frequency end of the potentiometer. R5, the frequency control, varies the symmetry of the multivibrator and can be set to shut off the circuit. When troubleshooting this circuit, first be certain that the potentiometer arm is near the high-frequency end of its travel.



When adjusting calibrator voltage, use at least a 20,000-ohms-per-volt meter to set the emitter of Q4 to 20 volts. You may want to cut holes in the dust cover of your scope to reach conveniently the two controls, R5 (frequency) and R12 (voltage), of the Box.

The variable-pulse generator has both coarse and fine pulse-width or delay controls. For example, with proper minimum settings it is possible to obtain a 5- $\mu$ sec pulse of 30 volts amplitude. As mentioned before, the range and extent of influence of these controls depend on the type of sync or trigger signal used.

Uses of the Calibrator Box are many; they include variable sweep delay, Z-axis blanking, amplifier calibration using voltage ratios, simple signal tracing in radios. In photographing scope traces, you will find that Z-axis blanking is a fine way to add another dimension of information to your sweep photographs.

Remember that neither the calibrator nor the variable-pulse generator is a power driver and should not be loaded excessively. As an extra bonus, the Box can often be tucked neatly away in the dark recesses of your scope.

## DOT-BAR GENERATOR

By BENNETT C. GOLDBERG

IF YOU'D LIKE AN INEXPENSIVE, functional and compact Dot-Bar generator to insure your color set always has "hi-fi" color, this project is for you.

Construction is simplified with the actual-size printed-circuit drawing shown. The unit (Fig. 1) operates on a 9-volt battery, fits in a small metal box, and puts out horizontal and vertical lines, dots and a cross-hatch pattern.

A second similar circuit (Fig. 2) is a Cross-Hatch generator that can be installed in any set with a service switch. It can be used when convergence circuits are adjusted.

### How does it work?

The function of the Dot-Bar Generator is shown in the block diagram in Fig. 3. The circuit is straightforward and explained as follows:

Transistors Q1 and Q2 form a free-running multivibrator which produces a square wave output. Because

of the variation in tolerances of the components, one transistor will conduct before the other.

An increase in collector current for transistor Q1 causes a decrease in its collector voltage and a corresponding reduction in regenerative feedback through capacitor C2 to the base of transistor Q2. As a result, the current through Q2 decreases steadily as the current through Q1 increases until Q2 is cut off. Capacitor C2 then discharges until forward bias is re-established across the base-emitter junction of Q2. Current through Q2 then decreases until Q1 is cut off. Capacitor C1 is coupled from the collector of Q2 to the base of Q1 so the cycle is repeated.

A positive input trigger pulse, 15,750-Hz, from the horizontal sweep circuits of the television receiver is used to sync the multivibrator. The frequency of the multivibrator is set for a multiple of 15,750 Hz by adjusting R22.

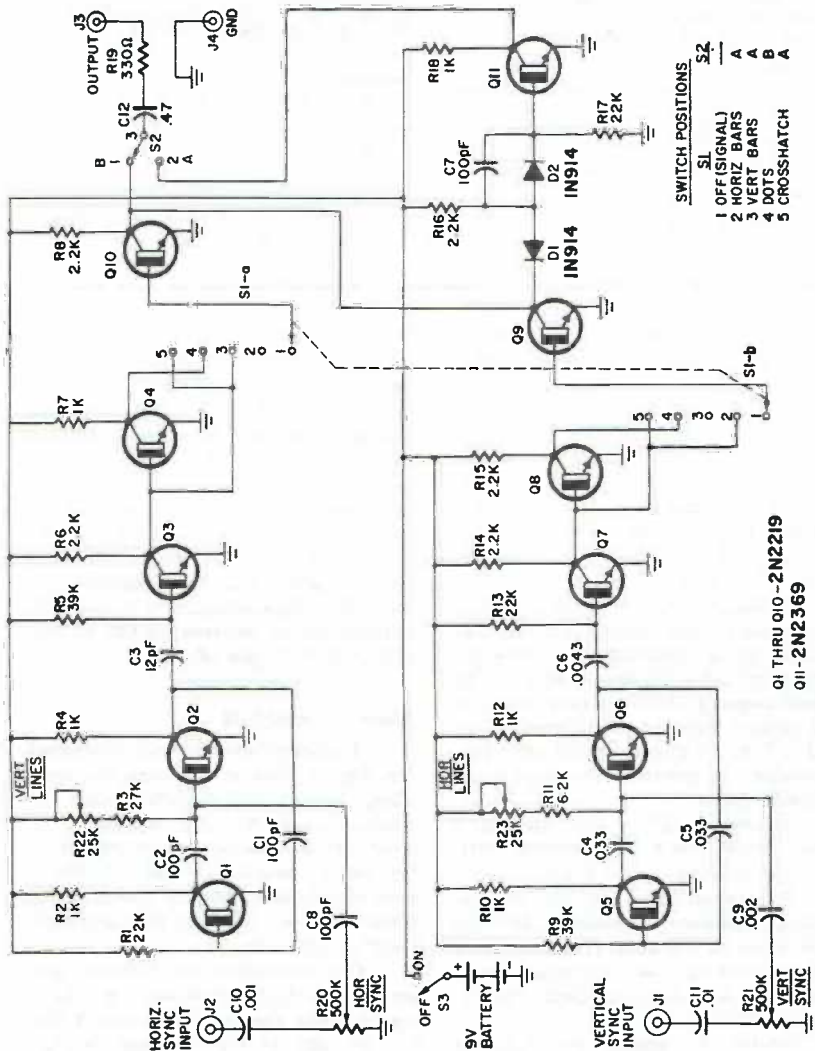


Fig. 1—Circuit for Dot-Bar generator. Transistors Q1–Q2 form a free-running multivibrator synchronized by 15,750-Hz horizontal sweep pulses, and Q5–Q6 operate as a 900-Hz multivibrator, synced by 60-Hz pulses from the vertical sweep circuit.

# PARTS LIST

## Resistors 1/4 watt or more, 10%

R1, R13, R17—22,000 ohms  
 R2, R4, R7, R10, R12, R18—1000 ohms  
 R3—27,000 ohms  
 R5, R9—39,000 ohms  
 R6, R8, R14, R15, R16—2200 ohms  
 R11—6200 ohms  
 R19—330 ohms  
 R20, R21—500,000-ohm 1/2-watt linear potentiometer, Mallory SU-50 or similar  
 R22, R23—25,000-ohm 1/2-watt linear potentiometer, Mallory SU-29 or similar

## Capacitors

C1, C2, C7, C8—100 pF, 100V, 10% mica  
 C3—12 pF, 100V, 5% mica  
 C4, C5—0.033  $\mu$ F, 50V, 5% disc ceramic  
 C6—0.0043  $\mu$ F, 100V, 5% mica  
 C9—0.002  $\mu$ F, 100V, 10% mica

C10—0.001  $\mu$ F, 500V, 20% mica  
 C11—0.01  $\mu$ F, 500V, 20% disc ceramic  
 C12—0.47  $\mu$ F, 50V disc ceramic

## Semiconductors

Q1—Q10 (Fig. 1), Q1—Q8 (Fig. 2)—2N2219  
 Q11 (Fig. 1), Q9 (Fig. 2)—2N2369

D1—D2—1N914 diodes

## Switches

S1—2-pole, 5-position, nonshorting rotary.  
 Centralab PA-1003 or equal

S2—Spdt

S3—Spst

MISC—11 transistor sockets, 9-V battery (NEDA No. 1600 or equivalent), tip jacks and plugs, circuit board, sloping-panel case at least 5 1/4" wide, 3" deep and 2 3/4" high, test leads and clips.

The multivibrator operates at 180 KHz, which produces approximately 12 lines. There is enough variation in the adjustment of R22 to provide more or fewer lines as desired.

Transistors Q5 and Q6 operate as a 900-Hz multivibrator. Its frequency is adjusted by R23. A 60-Hz positive trigger pulse from the vertical sweep circuits of the receiver is used to sync this multivibrator.

Transistor Q3 is a pulse shaper. Capacitor C3 and resistor R5 are connected as a differentiator network. Transistor Q3 is biased on by R5. The negative-going differentiated edge of the square wave of multivibrator Q1 and Q2 turns transistor Q3 off. This produces a positive, 0.1- $\mu$ sec-wide, 180-kHz pulse.

Transistor Q7 is also used as a pulse shaper, but a 50- $\mu$ sec-wide, 900-Hz rate positive pulse is obtained.

Transistors Q4 and Q8 are inverting amplifiers. Transistors Q9 and Q10 have a common-collector load and perform an "or" function. Transistor Q11 is a low-impedance inverting output driver.

Switch S1 selects the required mode of operation to form the desired output signal to the bases of transistors Q9 and Q10. Switch S2 selects the correct output phase to the receiver.

Circuit operation of the cross-hatch generator (Fig. 2) uses transis-

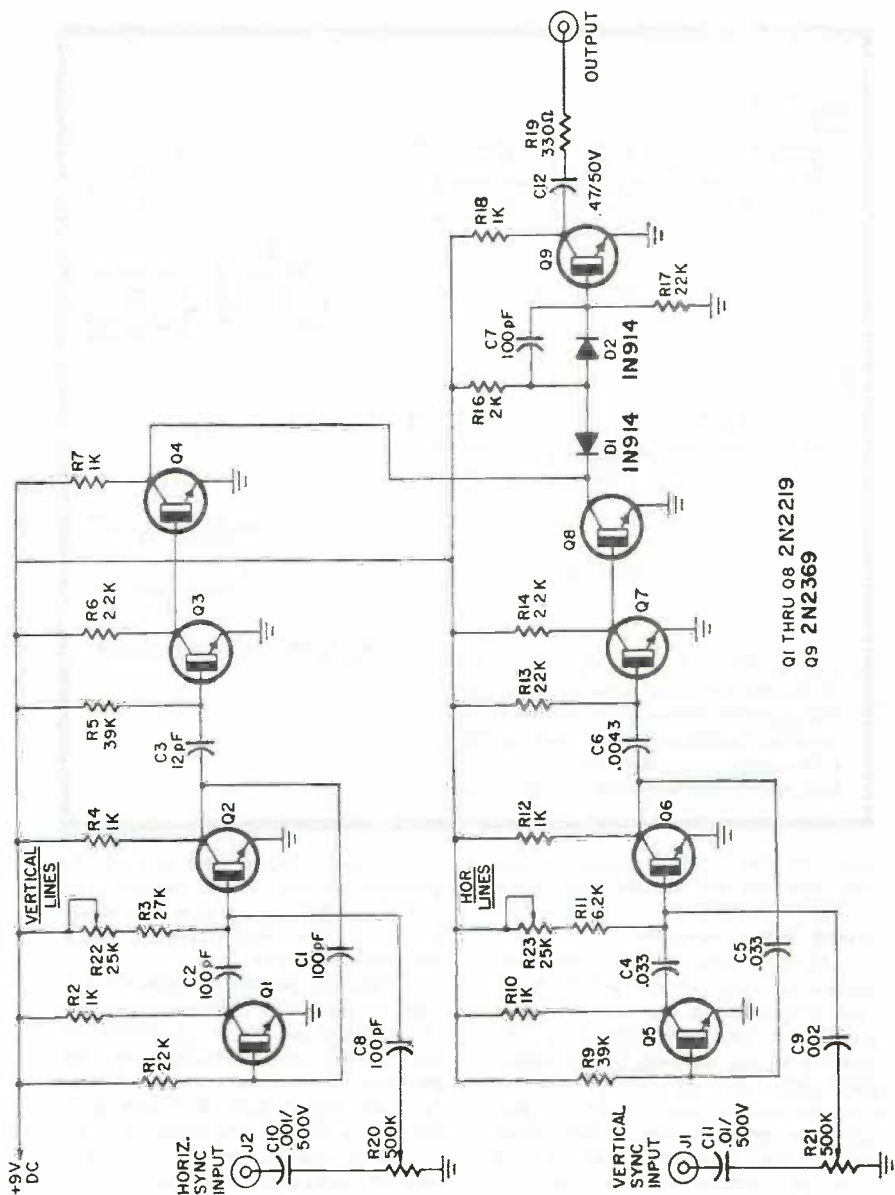
tors Q1 and Q2, Q5 and Q6 as multivibrators. Transistors Q3 and Q7 are pulse shapers, Q4 and Q8 is the OR amplifier and Q9 is the inverting and output amplifier. Since the switches are eliminated in this circuit, the outputs of Q3 (a 0.1- $\mu$ sec, 180-kHz positive pulse) and Q7 (a 50- $\mu$ sec, 900-Hz positive pulse) are applied to the bases of Q4 and Q8 combining these two frequencies at the common collector R7. This produces a cross-hatch pattern and is inverted by Q9 to give the proper output phase.

## How to build it

A printed circuit was developed for Fig. 1, but it can also be built using breadboard material such as Vector board. All parts should be of good quality, but nothing is critical to the circuit except C3 and C6. Since they determine the pulse widths, they should be as close to the specified value as possible.

The transistors are silicon npn small-signal high-frequency types having excellent rise and fall times. I did not try any of the economy planar passivated transistors such as the 2N3855 or the 2N3856, but do not see why they won't work in this circuit as they are very similar to the 2N2219 and the 2N2369 transistors.

After completion, apply power and check the current. It should be



Q1 THRU Q8 2N2219  
Q9 2N2369

Fig. 2—Cross-Hatch generator circuit that can be installed in sets with service switches. The 180-kHz output of Q3 and 900-Hz Q7 output are combined at R7. The signal is inverted and fed to the last video amplifier of the set through the service switch.

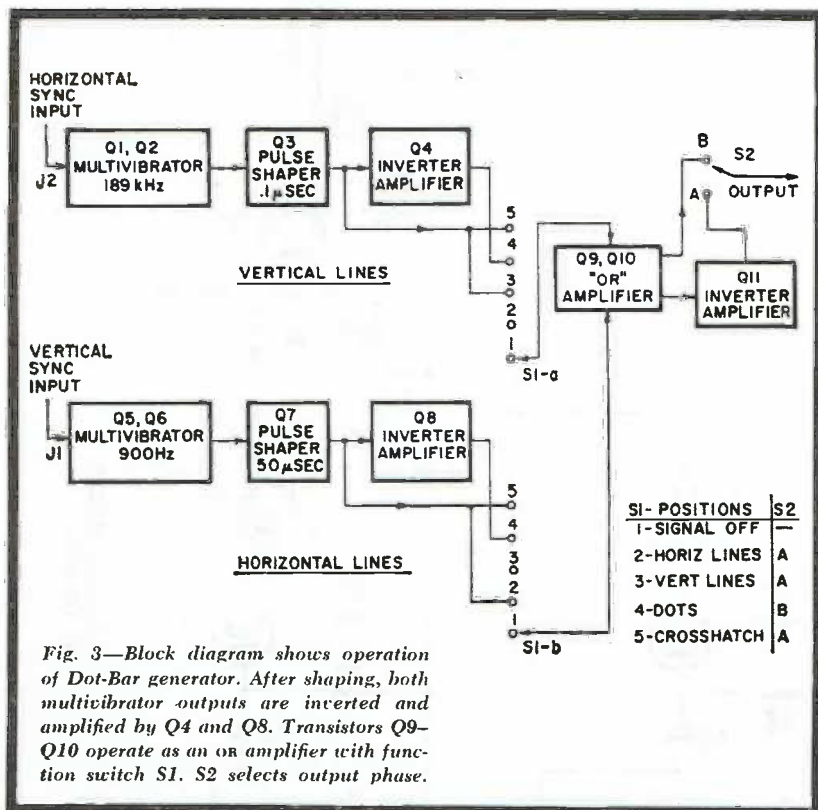


Fig. 3—Block diagram shows operation of Dot-Bar generator. After shaping, both multivibrator outputs are inverted and amplified by Q4 and Q8. Transistors Q9-Q10 operate as an OR amplifier with function switch S1. S2 selects output phase.

near 50 mA for the dot-bar circuit and about 40 mA for the cross hatch circuit.

### Using the generator

Both generators require external horizontal and vertical synchronization. To provide it, clip the horizontal lead, J2, of the generator to the insulation on the red lead of the deflection yoke (*clip only, do not solder*). Clip the vertical lead, J1, of the generator to the insulation on the yellow lead of the deflection yoke. If you have any problem with vertical sync, connect clip J1 to any of the leads of the vertical output transformers or terminal "C" on the convergence board of color receivers like the RCA CTC-15, 16 series.

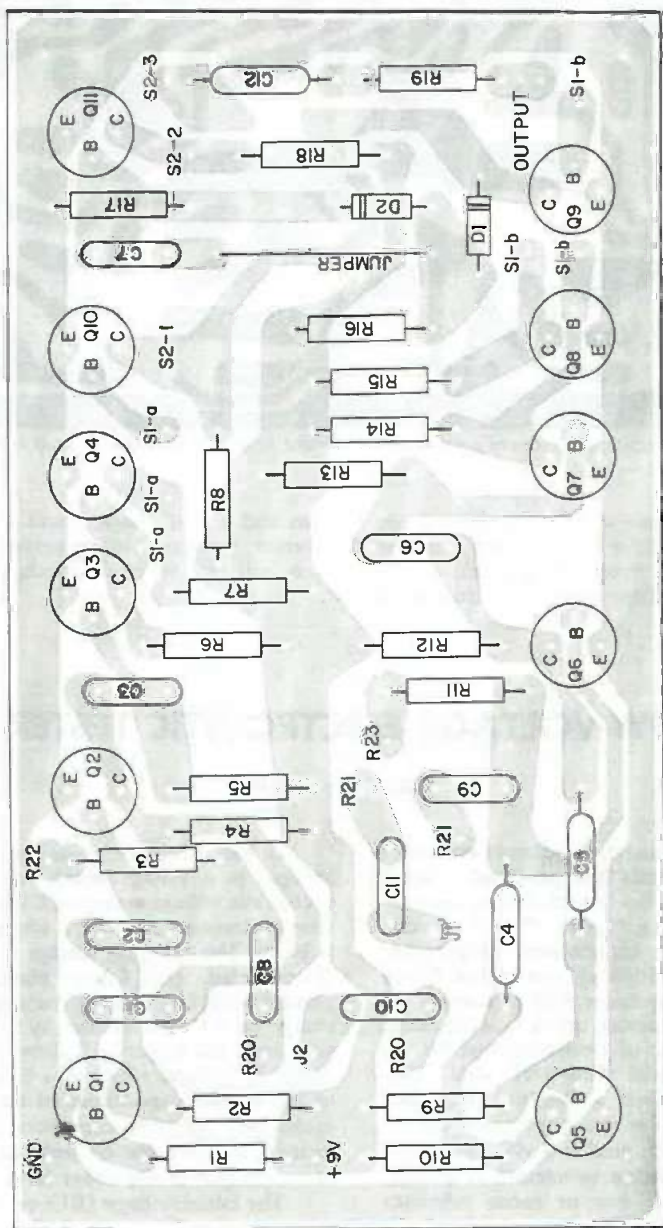
Connect the ground lead of the generator to the receiver chassis. Connect the output of the generator to the grid of the last video amplifier using a test signal adapter.

Now set the channel selector to a local station. This will provide synchronization for the sweep circuits of the receiver and generator. Next set generator circuit to cross-hatch pattern and adjust R20, R22 and R21, R23 until the pattern stays in sync.

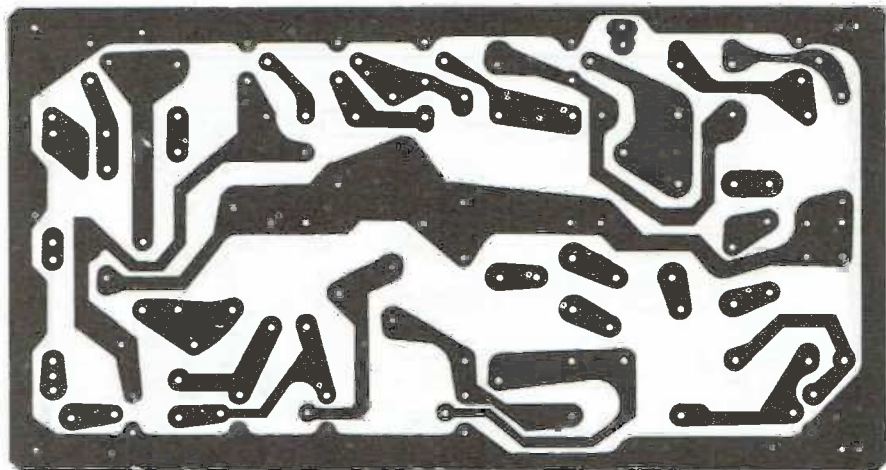
There should be 12 horizontal and 15 vertical lines when the generator is in sync. Some minor adjustment of the vertical hold on the color receiver may be required.

Adjust brightness and contrast controls for the sharpest patterns.





*Placement of components on PC board as seen from component side.*



*Reproduction of PC board used by author. Board is shown actual size as it is used.*

Now you can set S2 to position B for dots and A for other modes. This applies the correct phase signal to the video amplifier grid, producing white

lines and a black background on the receiver picture tube. Program pictures will also be in the background.

## LOW-VOLTAGE ELECTROLYTIC TESTER

By MELVIN CHAN

THE POPULARITY OF SOLID-STATE electronic circuits has produced a rapid growth in the use of low-voltage capacitors (e.g., 3, 10, 15 and 25 volts dc) whose capacitances range from 0.1  $\mu\text{F}$  to 5000  $\mu\text{F}$  and higher. There is a need to know with reasonable accuracy the actual, rather than the rated, capacitance of electrolytics which are manufactured with very wide tolerances. Here is a simple method that you can use to measure capacitance.

All you need is a vtvm (or other high-impedance voltmeter), a dc voltage source, one or more reference capacitors of known value and the simple circuit shown in Fig. 1.

In use, reference capacitor C1 is charged to a voltage which does not exceed the voltage rating of C1 or C2 (the unknown capacitor), whichever is lower. The charging voltage is then disconnected, and C2 is placed in parallel with C1, causing the original charge on C1 to be shared by C2. At this time, the meter will show a decrease in voltage, which is a function of the relative capacitance of the reference and unknown capacitors. (Obviously, the voltages on both capacitors are equal at this lower level.)

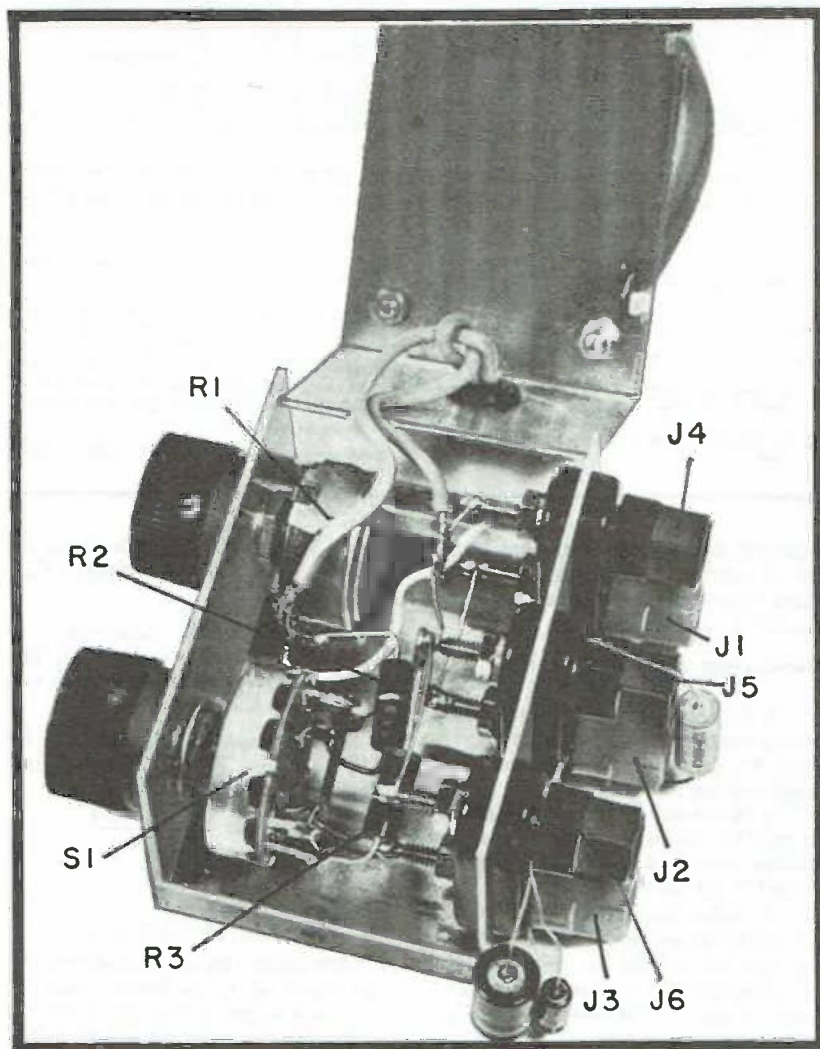
The initial voltage (E1) on reference capacitor C1, and the lower voltage (E2) resulting when C2 is in pa-

rallel with C1, are substituted in the formula

$$C2 = C1 \frac{(E1 - E2)}{E2}$$

Electrolytic capacitors which have been idle for a period of 2 weeks

or longer should be reformed before measuring or return to service. This process consists of applying (with correct polarity) the rated working voltage to the capacitor until a voltmeter across the capacitor stabilizes at that voltage. The time required for the voltage to stabilize is a function of capacitance, leakage current and the



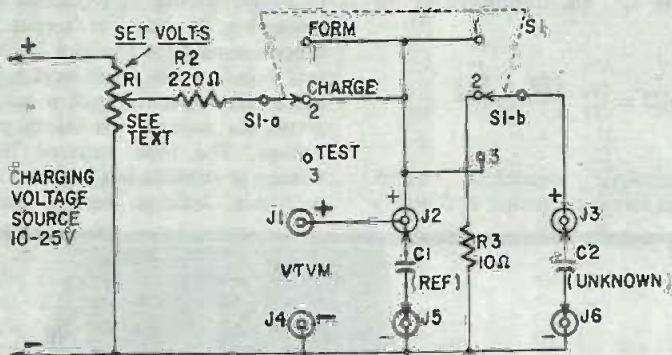


Fig. 1—Unknown electrolytic capacitors can be reformed, discharged and measured in a 1-2-3 procedure at the flip of S1. Voltage levels can be set by control R1.

#### Parts List

R1—1000-ohm potentiometer, linear taper (see text)

R2—220-ohm, ½-watt, 10% resistor

R3—10-ohm, ½-watt, 10% resistor

S1—2-pole, 3-position nonshorting rotary switch (Mallory 3323J or similar)

J1, J2, J3—Pin jacks or 5-way binding posts, red

J4, J5, J6—Pin jacks or 5-way binding posts, black.

C1—100-μF, 12-volt electrolytic capacitor (Sprague TE1119.3 or similar); **Optional:** 15-μF, 12-volt electrolytic capacitor (Sprague TE 1129 or similar) 450-μF, 3-volt electrolytic capacitor (Mallory TPG-4503T, Cornell-Dubilier type NLW Electro-mite or similar)

Chassis—aluminum box, 3¼" x 2½" x 1½"

length of time the capacitor has been out of service. Reforming time may range from only a few seconds to a minute or more.

#### Measuring capacitance

1. Connect the charging-voltage source and the vtvm as shown in Fig. 1. Turn R1 (SET VOLTS) to reduce the meter reading to zero.

2. Set switch S1 at FORM, connect C1 and C2. Adjust R1 until the meter reading reaches the voltage rating of C1 or C2, whichever is lower.

3. When the meter stabilizes at the rated voltage, throw S1 to CHARGE. Readjust R1 to hold the voltage (E1) on capacitor C1 at the level established in Step 2. (Meanwhile, C2 discharges through R3.)

4. After a few seconds throw S1 to TEST. Read the new voltage (E2) on the meter.

5. Substitute the capacitance of C1 and voltages E1 and E2 in the equation above and calculate the capacitance of C2.

For example, if C1 is 100 μF, E1 is 10 volts and E2 is 7.25 volts, the equation becomes:

$$C2 = 100 \frac{(10 - 7.25)}{7.25}$$

$$= 100 \frac{2.75}{7.25}$$

$$= 38 \mu F \text{ (approx.)}$$

Obviously, the more precisely the capacitance of C1 is known and the lower its leakage current, the more exact will be the value of C2 as calcu-

**Table I**  
**C1 multiplier (see text)**

$E_1 - E_2$	Charge Voltage	
	3 Volts	10 Volts
0.1	0.03	0.01
0.2	0.07	0.02
0.3	0.11	0.03
0.5	0.2	0.05
0.6	0.25	0.06
0.7	0.3	0.075
0.8	0.36	0.09
0.9	0.43	0.10
1.0	0.5	
1.25	0.7	
1.5	1.0	
2.0	2.0	0.20
2.25	3.0	0.20
2.4	4.0	0.31
2.5	5.0	
2.6	6.5	
2.75	11.0	0.38
3.0	O-C*	
3.5		0.54
4.0		0.67
5.0		1.0
6.0		1.5
7.0		2.33
7.5		3.0
8.0		4.0
8.5		5.5
8.75		7.0
9.0		9.0
9.25		12.3
10.0		O-C*

\*O-C = C2 open circuited

lated from the formula. If you avoid mathematics whenever possible, Table I is for you. It gives the multiplier of C1 capacitance for charge voltages of 3 and 10, at useful voltage-difference levels. Thus, if  $E_1$  is 3 volts and  $E_2$  is 2,  $E_1 - E_2 = 1$  volt. Find 1.0 in the  $E_1 - E_2$  column and read 0.5 on the same line in the 3-volts column. If C1 is 450  $\mu\text{F}$ , C2 is  $0.5 \times 450$  or 225  $\mu\text{F}$ .

The resistance of R1 should be selected to limit its bleeder current to about 15 mA. For example, with a 15-volt dc supply R1 should be 1000 ohms ( $15/0.015$ ). Resistor R2 limits the charging current; R3 fully discharges C2 after it has been reformed and before it is paralleled across charged capacitor C1.

#### Extending the range

Using a 100- $\mu\text{F}$ , 10-volt capacitor for C1 permits measuring capacitances ranging from 1  $\mu\text{F}$  to 1200  $\mu\text{F}$ . The use of 15- $\mu\text{F}$ , 12-volt and 450- $\mu\text{F}$ , 3-volt capacitors for C1 will extend the range from 0.15  $\mu\text{F}$  through 4950  $\mu\text{F}$ . If two or more standard capacitors are to be used for C1, they may be plugged into tip jacks (J2 and J5).

The charging voltage may be supplied by dry cells connected in series, or by any 10–25-volt dc supply. It is not necessary to charge C1 to more than 10 volts when measuring unknown capacitors.

The parts in the basic test unit cost between \$6 and \$10. Be sure to use the 10% tolerance tantalum capacitor as the reference unit for best accuracy.



# AUDIO TONE-BURST GENERATOR

By RICHARD J. DE SA, Ph.D.

HIGH ON THE LIST OF "HOT" ELECTRONIC items are those amazing little devices known as integrated circuits. Now that their prices have dropped so dramatically, it's quite simple to incorporate a couple of them in a multipurpose test instrument. Here's a practical and useful project for audio testing.

True to their acclaim, integrated circuits allow construction of the instrument with minimum fuss and few auxiliary components. At the same time, they provide versatility, small size and, most importantly, low cost.

When used alone, the instrument puts out a high-quality square wave (rise time under 450 nsec) at five switchable frequencies ranging from 20 Hz to 12 kHz. When the unit is used with an external sine-wave generator, variable-width pulses are produced. The square-wave and pulse outputs are both excellent signals for testing and troubleshooting audio components.

Most important, however, is the instrument's ability to gate or switch an external signal to produce tone bursts. The external signal to be gated can be any waveform—sine or square waves, white noise, even music or speech. Cost runs under \$20, and construction should take no more than two or three evenings.

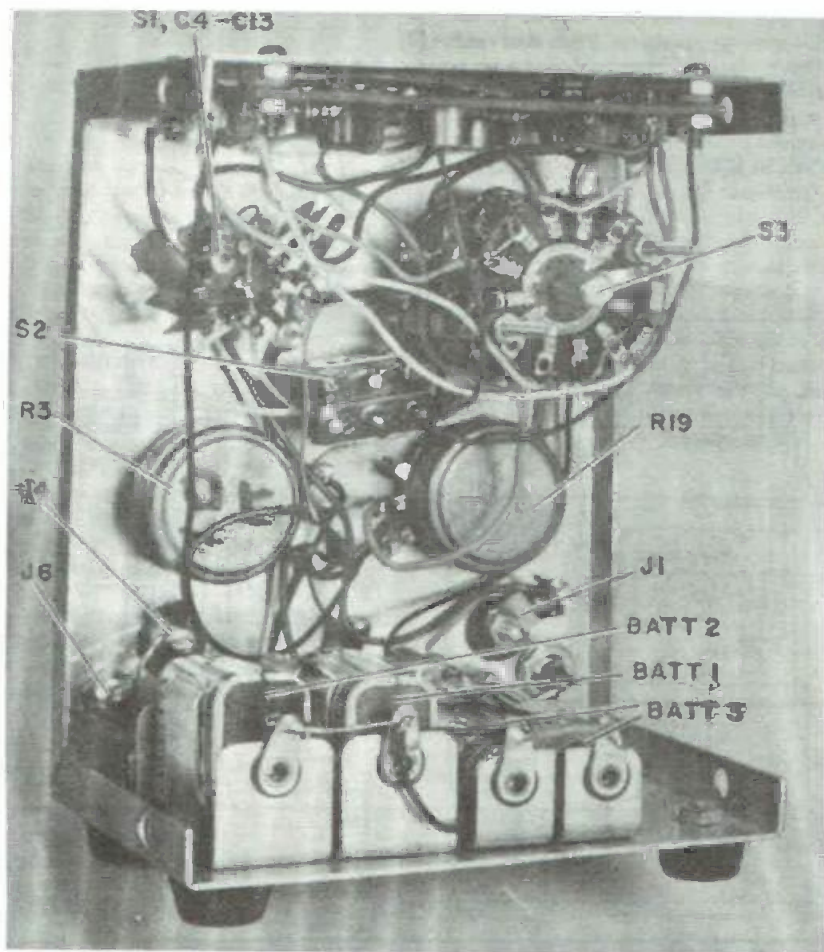
The tone burst is excellent for testing transient response in speakers, amplifiers, tape recorders, crossover networks and other audio components. The

gated or switched output provides a close simulation of the rapid attack-and-decay patterns of musical material. The tone-burst waveform is also useful in evaluating overload characteristics of power amplifiers.

Fig. 1 shows the schematic of the generator. Fig. 2 is a simplified block diagram showing its essential features. The unit consists of four subunits. IC1 is connected as a variable-width pulse generator (Schmitt trigger). IC2 is a free-running multivibrator. Q1 and Q2 are the gate driver—amplifier and the gate, respectively. Q3 and Q4 form a compound emitter follower to provide a low-impedance output. A direct-coupled npn–pnp pair is used here so the output is at ground potential.

Sine waves introduced at input jack J1 (Figs. 1 and 2) are converted to fast-rise pulses at the input frequency by IC1. Pulse width is controlled by R3. Pulses, applied via C2–R5 to IC2, serve to lock in the square wave produced by free-running multivibrator IC2 (the gating waveform) with the incoming sine wave (the gated waveform). The tone burst will start at the point at which the sine wave crosses the zero axis. The input sine wave also is routed to gate Q2 by R9–R15, to be gated by the output of IC2.

The multivibrator (IC2) produces a clean square wave without the rounded-cornered characteristic of the usual multivibrator output. In addition, a signal from IC2 is presented at J3 to sync the scope.



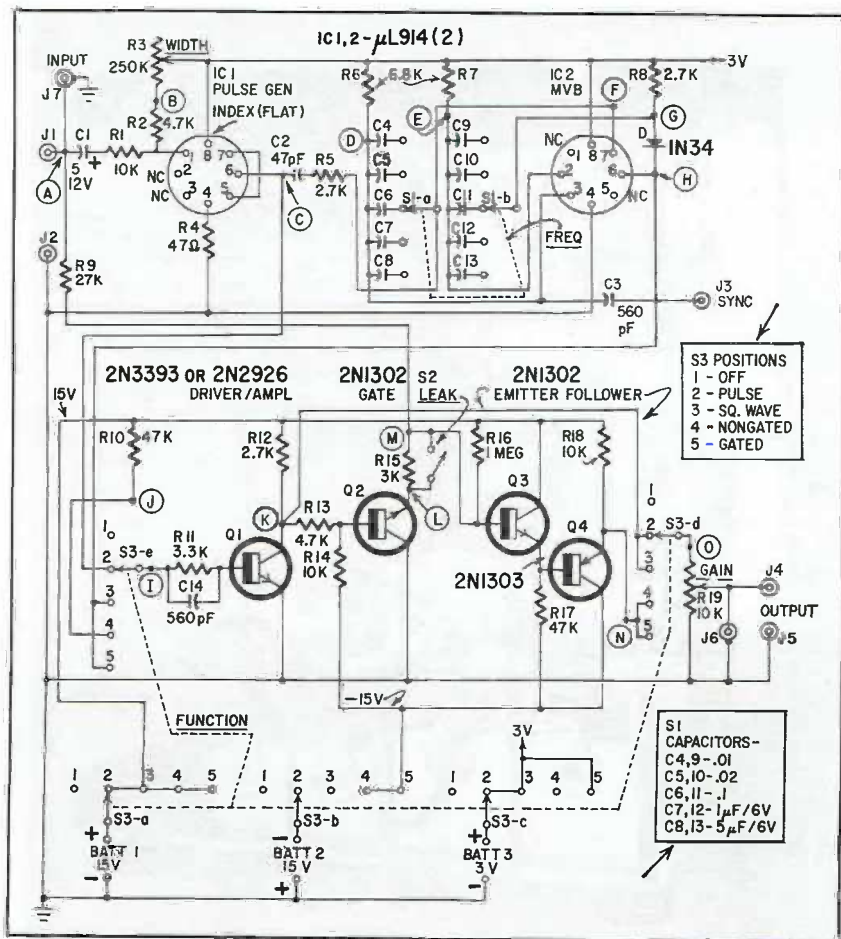
*In the box specified, there is just enough room for batteries to fit behind panel jacks.*

The gate (Q2) is a circuit frequently used in analog computer systems. Extremely simple, it is relatively free from switching transients. It is also highly effective. Signal through the closed gate is down more than 65 dB! Note, however, that, with R15 switched into the circuit by S2, "leak" through the gate is increased so that the signal is down only 20 dB from the main tone burst. This feature is useful when you are interested in noting the behavior of

the device under test after the main tone burst is switched off.

In addition to its function as gate driver, Q1 amplifies signals obtained from the integrated circuits. Waveforms up to 15 volts peak-to-peak are available at the output jack when the unit is operated as a square-wave or pulse generator.

Switch S3 selects the desired function (OFF, PULSE, SQUARE WAVE, NON-



GATED, GATED). Each battery is switched into operation only when it is necessary for the selected function. The signal applied to R19 also is switched to pass the desired output signal through the GAIN control. For the gate-open or nongated function, R10 is connected between 15 volts and the base of Q1. The driver is thus turned on, disabling the gate.

The entire unit is built into a 5 x 4 x 3-inch aluminum box. Most of the cir-

The circuit pattern can be readily copied using standard printed-circuit

etching techniques. Careful hand application of ordinary nail polish makes an ideal and inexpensive way to apply an etch-resistant coating to the copper-clad board. After etching in warm ferric-chloride solution, the nail polish is easily removed—with nailpolish remover, naturally!

The circuit board is fastened to the top of the box with 4-40 hardware and small spacers, and the batteries are installed in the bottom of the box. If you wire the lettered points on the board to the appropriate points, the unit can be wired quickly. Addition of rubber feet, knobs, etc., will then complete the assembly.

After the wiring is complete, install the batteries and connect output jacks J4 and J5 to the vertical input of

your scope. Connect sync jack J3 to the scope's external-sync input and turn on the instrument. With function switch S3 in position 3, you should get a clean square wave. S1 varies the frequency of the output. Now inject a 1-kHz 8-volt sine wave at J1 and J2. Place the function switch in position 4 (nongate). You should get an unaltered sine wave at J4. In position 5, the sine wave will be switched on and off at the frequency chosen by S1, producing tone bursts.

Adjustment of R3 will allow the tone burst to begin at the zero-axis crossing of the sine wave (this is not necessary, but it results in a more attractive display). Similarly, the gate can be made to close at the zero-axis crossing by trimming the input frequency slightly. Throw S2 and note that

#### SPECIFICATIONS

##### Input:

External signal generator—impedance approximately 7,000 ohms at 1 kHz. Requires 2 to 3 volts rms to drive pulse generator.

##### Output:

- Output impedance approx. 5,000 ohms (affected by position of level control). Maximum output 15 volts peak to peak (5.2 volts rms). Square-wave and pulse outputs positive-going from zero.
- External scope-sync output.

##### Performance:

- Gating and square-wave frequencies 20, 100, 250, 5,000, 12,000 Hz nominal.
- Pulse frequency—determined by external signal generator—usable to at least 500 kHz.
- Frequency response in gated and nongated modes—flat from dc to approx. 50 kHz.

#### PARTS LIST

R1, R14, R18—10K  
R2, R13—4,700 ohms  
R3—pot, 250K  
R4—47 ohms  
R5, R8, R12—2,700 ohms  
R6, R7—6,800 ohms  
R9—27K, 5%  
R10, R17—47K  
R11—3,300 ohms  
R15—3,000 ohms, 5%  
R16—1 meg (see text)  
R19—pot, 10K  
All resistors 1/4 watt 10%, except R3, R9, R15.  
C1—5  $\mu$ F, 15 volts, electrolytic  
C2—47 pF, disc  
C3, C14—560 pF disc  
C4, C9—.01  $\mu$ F, disc  
C5, C10—.02  $\mu$ F, disc  
C6, C11—.01  $\mu$ F, disc  
C7, C12—1  $\mu$ F, 6 volts, electrolytic  
C8, C13—5  $\mu$ F, 6 volts, electrolytic  
D—1N34  
IC1, IC2— $\mu$ L914 Fairchild integrated circuit Available through Fairchild distributors

Q1—2N3393, 2N2926 (GE)  
Q2, Q3—2N1302 (RCA, TI)  
Q4—2N1303 (RCA, TI)  
S1—2-pole 5-position rotary switch  
S2—spst slide or toggle switch  
S3—5-pole 5-position rotary switch  
BATT 1, 2—15-volt battery (Burgess U10 or equivalent)  
BATT 3—two 1.5V Z-batteries in series  
Box—5 x 4 x 3-inch Minibox or equivalent (Bud CU-2105A)  
J1 through J5—4-way binding posts  
J6, J7—Miniature phono or coax connectors  
Battery holders—2 for Burgess U10 (Keystone 166 or equivalent) 1 for 2 Z-batteries (Keystone 140 or equivalent)  
Perforated copper-clad board (Lafayette 19C3601 or equivalent)  
Screws, nuts, spacers, wire, solder, etc.

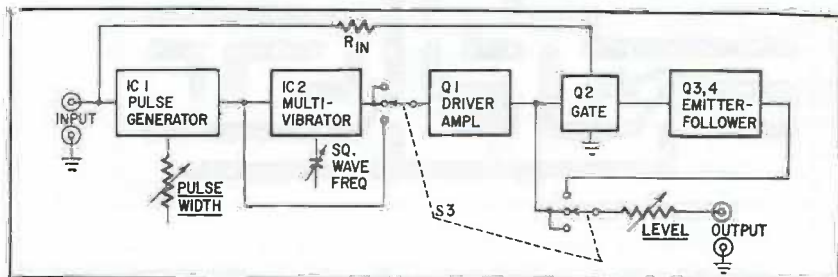
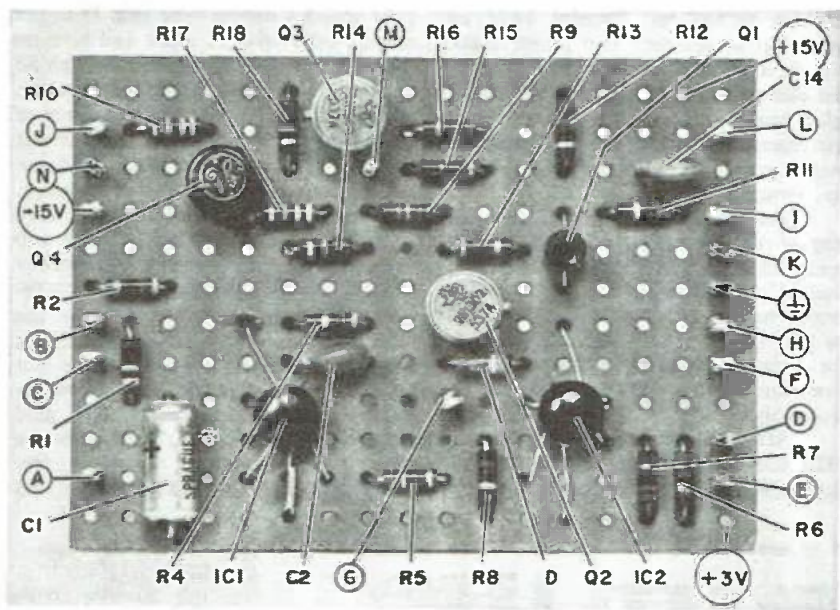
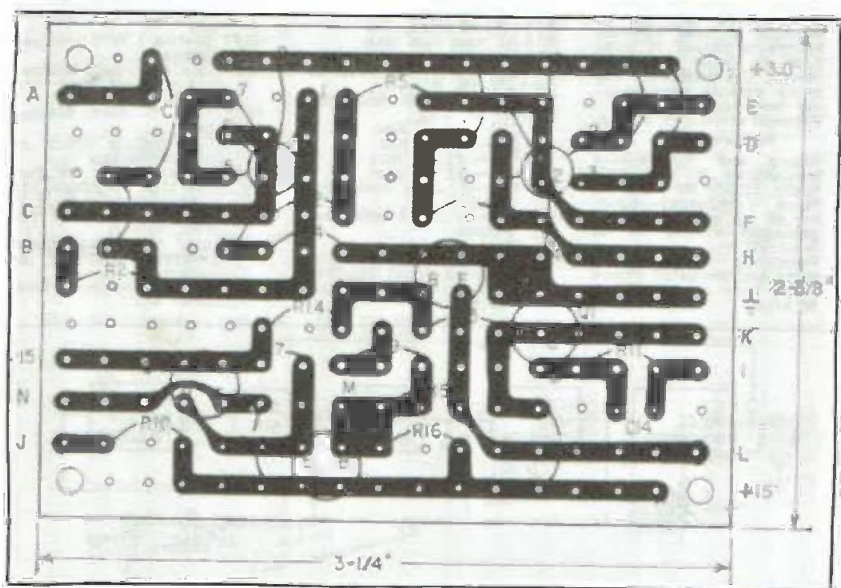


Fig. 2—Generator circuit operation depends on the position of function switch S3.





*Front of the perforated board contains most components with some room to spare.*



*Fig. 3—Back of the board above, showing copper connecting paths to all elements.*



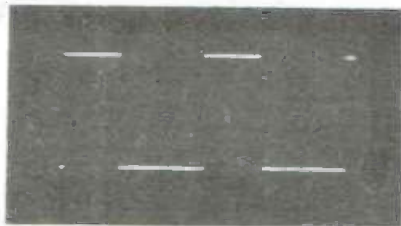
the "leak" during the gate-closed period is increased to -20dB.

Operation of the pulse generator can be checked by setting switch S3 to position 2 and varying R3. Note that R3 can be adjusted to give a symmetrical square wave; thus the unit will produce square waves at the frequency of

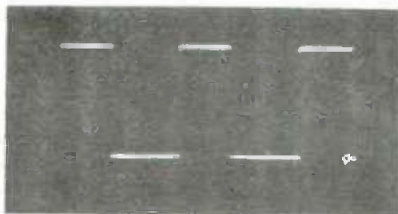
the external sine-wave generator in addition to the five internally produced frequencies.

With the unit in the "gated" mode, but with input jack J1 shorted to J2, check the gate offset. (This offset effectively places the tone burst on a slight pedestal.) It should be less than 0.1 volt.

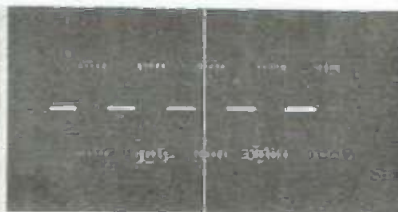
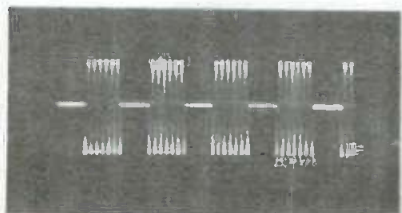
*Fig. 4—Samples of generator waveforms:*



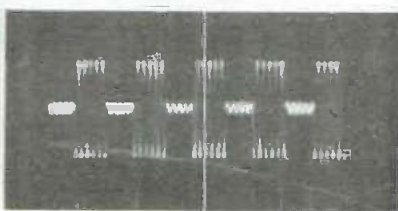
*Low-frequency square wave (about 20 Hz).*



*High-frequency square wave (12 kHz). Pulse output at approximately 10 kHz.*



*1,000-Hz sine wave, gated at 100 Hz. 50 kHz sine wave gated at about 5 kHz.*



*As above, but with LEAK increased. As above, but with LEAK increased.*

This offset is caused by two effects. The first and less significant one is the leakage current through Q2 when it is cut off. Second, the positive and negative 15-volt supplies are not necessarily equal in magnitude. In fact, as the batteries age, the offset will vary, since battery voltages do not change at the same rate. One could overcome this

problem by making R16 variable so the offset could be adjusted to zero occasionally as the batteries age. Such an adjustment was omitted, however, to reduce circuit complexity. In any event, the offset will not exceed about 0.1 to 0.2 volt through the useful life of the batteries.

## 3-WAY WAVEFORM GENERATOR

by NEIL HECKT

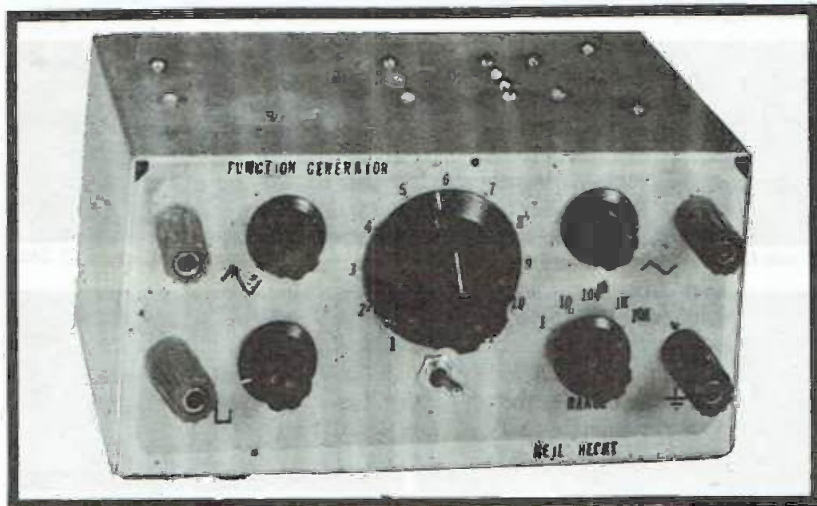
The heart of the generator is three Motorola MC1709CG integrated-circuit operational amplifiers. Each contains the equivalent of 13 transistors, 2 diodes, and 15 resistors in a standard TO-5 package.

### Theory of operation

The unit (Fig. 1) can be divided into three major parts. The oscillator sine-wave converter, and output amplifiers. The oscillator is a form of relaxation circuit called a hysteresis oscillator. It is composed of two

integrated-circuit operational amplifiers IC and IC2.

Integrated circuit IC2 is connected as a voltage comparator with hysteresis feedback. Because of the high gain of the amplifier, if the non-inverting (+) terminal is slightly more positive than the inverting (-) terminal, the output will be the positive saturation level of +14 volts. Conversely, if the non-inverting input is more negative, the output will be the negative saturation level, -14 volts.



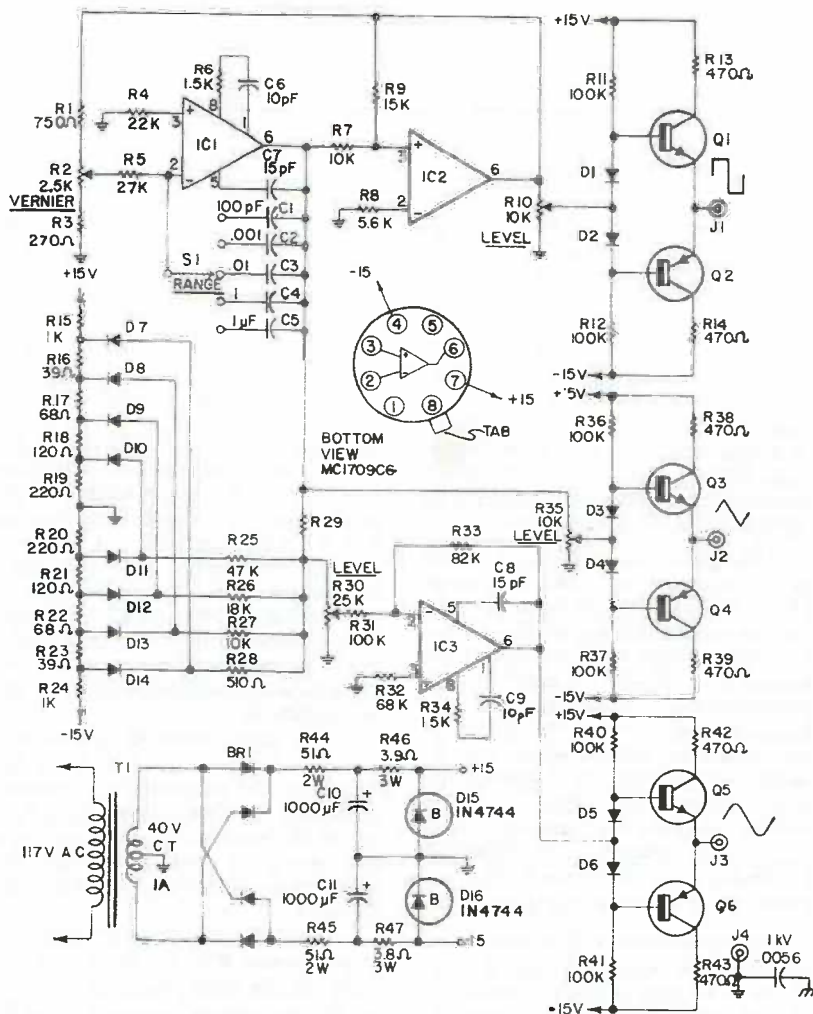
# PARTS LISTS

All resistors  $\frac{1}{2}$  watt, 10% unless noted

- R1—750 ohms
- R2—potentiometer, 2500 ohms, linear taper
- R3—270 ohms
- R4—22,000 ohms
- R5—27,000 ohms
- R6, R34—1500 ohms
- R7—10,000 ohms
- R8—5600 ohms
- R9—15,000 ohms

R10, R-35—potentiometer, 10,000 ohms, linear taper

- R11, R12, R31, R36, R37, R40, R41—100,000 ohms
- R13, R14, R38, R39, R42, R43—470 ohms
- R15, R24—1000 ohms
- R16, R23—39 ohms
- R17, R22—68 ohms
- R18, R21—120 ohms
- R19, R20—220 ohms
- R25—47,000 ohms
- R26—18,000 ohms



COMPLETE CIRCUIT OF WAVEFORM GENERATOR. Regulated power supply delivers +15 and -15 volts. Three IC amplifiers simplify construction drastically.

R27—10,000 ohms  
 R28—510 ohms  
 R29—27,000 ohms  
 R30—potentiometer, 25,000 ohms, linear tapes  
 R32—68,000 ohms  
 R33—82,000 ohms  
 R44, R45—51 ohms, 2 ohms, 5 %  
 R46, R47—3.9 ohms, 3 watts  
 C1—100 pF  
 C6, C9—10 pF  
 C2—.001 uF  
 C3—.01 uF  
 C4—.1 uF  
 C5—1.0 uF  
 C7, C8—15 pF  
 C10, C11—1000 uF, 30 volts, electrolytic  
 C12—.0056 uF

J1, J2, J3, J4—4-way jacks  
 T1—transformer, 118-volt primary, 40-volt center tapped secondary, 1 amp  
 BR1—full-wave bridge rectifier, 100 pV, 1A or four 1N4004 or similar diodes  
 D1 to D14—1N914  
 Q1, Q3, Q5—2N4123  
 Q2, Q4, Q6—2N4125  
 IC1, IC2, IC3—MC1709CG IC op amp  
 Miscellaneous; 6" x 4" x 3" utility box, knobs, stand-off terminals, hookup wire  
 D1 to D14—1N914  
 Q1, Q3, Q5—2N4123  
 Q2, Q4, Q6—2N4125  
 BR1—MDA 920A-3  
 IC1, IC2, IC3—MC1709CG

Assuming the output is +14 volts, positive feedback applied to the non-inverting input via R9, keeps the amplifier latched in the positive saturation state. An increasingly negative voltage is applied to the input resistor R7. The ratio of R7 and R9 is such, that when the input reaches -10 volts, it cancels the +14 volt feedback. As the input goes slightly more negative, the amplifier switches to the negative saturation level. It remains in that state until the input exceeds +10 volts when a similar, but opposite reaction occurs causing the amplifier to latch again at the positive level.

Integrated circuit IC1 is connected as an integrator. If a positive voltage ( $E_{in}$ ) is applied to the input resistor, R5, the output will try to swing negative. The voltage across the timing capacitor  $C_t$ , (C1-C5) cannot change instantaneously, however, and the negative output thru the capacitor cancels out the positive input in an attempt to maintain the inverting terminal at zero volts. The cancelling current charges the capacitor at a linear rate toward -14 volts at  $E_{in}/R5C_t$  volts per second. The opposite effect is true of negative inputs.

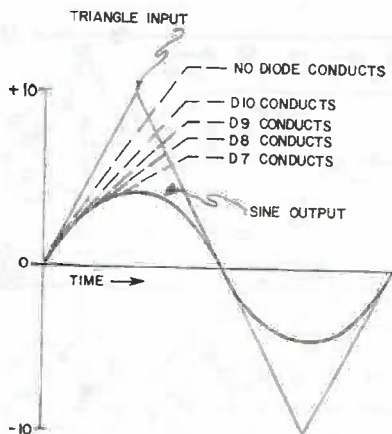
If comparator IC2 is latched in the +14-volt state, and some portion of the +14 volts is applied to the integrator input resistor, the integrator output will run down linearly toward -14 volts. Since the integrator output is connected to the comparator input,

the comparator will switch to -14 volts when the integrator output equals -10 volts. This applies the opposite polarity to the integrator, and its output runs up linearly toward +14 volts. At +10 volts the comparator switches back to +14 volts completing one cycle. The cycle then repeats and the circuit oscillates at a constant rate.

The frequency of oscillation is a function of the applied voltage to the integrator ( $E_{in}$ ), the integrator input resistor (R5), the feedback capacitor ( $C_t$ ), and the trigger levels of the comparator ( $E_o$  max) according to  $F = E_{in}/4 E_o \text{ max} \times R5 \times C_t$ .  $E_o$  max is set at 10 volts, R5 is 25,000 ohms, therefore  $F = E_{in}/C_t$  where  $C_t$  is in  $\mu\text{f}$ . Resistors R1, R2 and R3 are chosen to provide  $E_{in}$  from  $\pm 0$  to  $\pm 11$  volts, and  $C_t$  is switched from 1  $\mu\text{F}$  to 100 PF providing frequencies from 1 Hz to 110,000 Hz.

The output from the comparator is a 28-volt p-p square wave while the output of the integrator is a triangle of 20 volts p-p. The triangle wave output of the integrator is converted to a sine wave by the non-linear resistor/diode network R15-R30 and D7-D14.

Assuming the input to R29 and output across R30 are both zero initially, all the diodes are reverse biased by the bias voltage divider R15-R24. If the triangle wave is rising linearly toward +10 volts, its slope is reduced at the output by the voltage dividing



**FIG. 2—DIODES MAKE A SINE WAVE** out of a triangular wave.

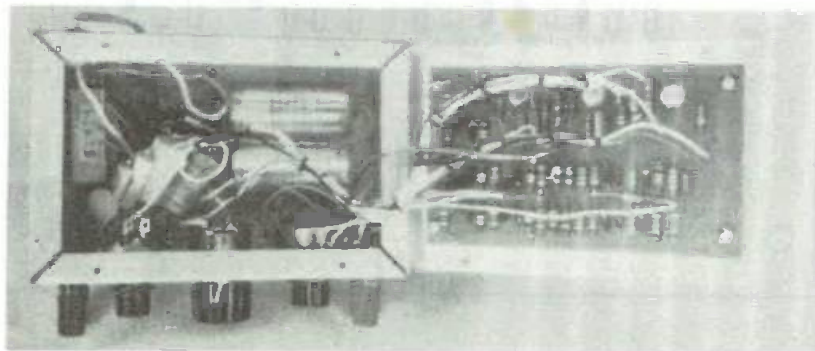
action of R29 and R30. When the output reaches the bias level of D10, R25 is switched in parallel with R30, further reducing the slope. As the output increases still more, diodes D9, D8 and D7 conduct in turn (Fig. 2), each reducing the output slope by switching in R26, R27 and R28. The input triangle is therefore "distorted" into a quarter sine wave. As the triangle voltage decreases back toward zero, each diode is cutoff in reverse order to produce the second quarter sine wave. For negative half-cycles,

There are three output drives, one for each waveform. They consist of complementary emitter followers. Diodes, such as D1 and D2, compensate for the base-to-emitter drop of the transistors. For positive inputs, the npn transistor supplies the current to the load, and for negative inputs the pnp supplies the current. The 470-ohm resistors limit the current to prevent damage to the transistors, as well as provide an output impedance of about 500 ohms.

### Construction techniques

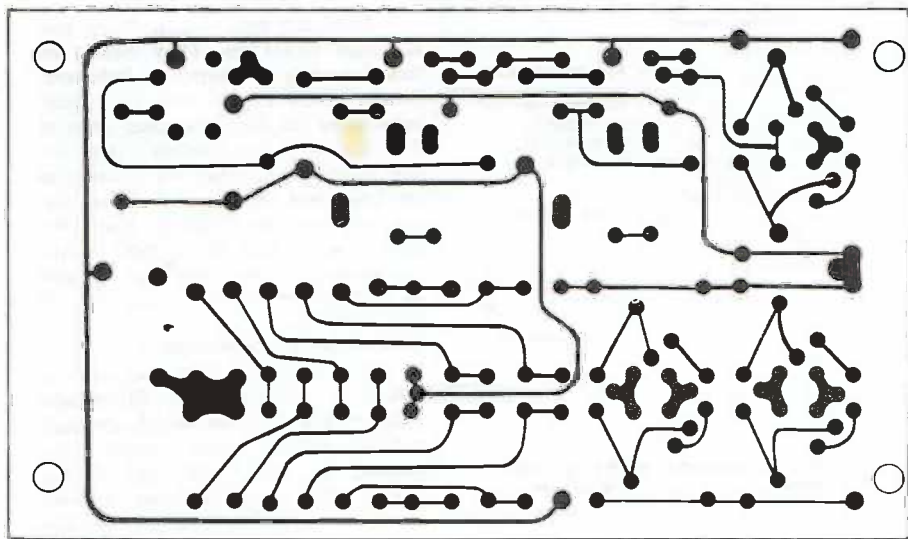
A 3 x 4 x 6-inch chassis, such as a Bud AC-430 with BPA-1505 bottom plate makes an economical, compact cabinet. Mount power supply components on the inside top of the chassis box using insulated standoff terminals (Grayhill type 18-1 or similar). Mount all controls and binding posts symmetrically on one 3 x 6-inch face. After all components have been mounted in the chassis and on the PC board, connect the two thru twisted diodes D11-D14 are cut in in a similar manner. The points at which the slope changes are not sharp, but are rounded by the non-linear forward resistance characteristics of the diodes as the current thru them increases.

The output of the converter is about 10 volts p-p. Operational amplifier IC3 is connected to give a gain of 2 so that 20 volts p-p are available.

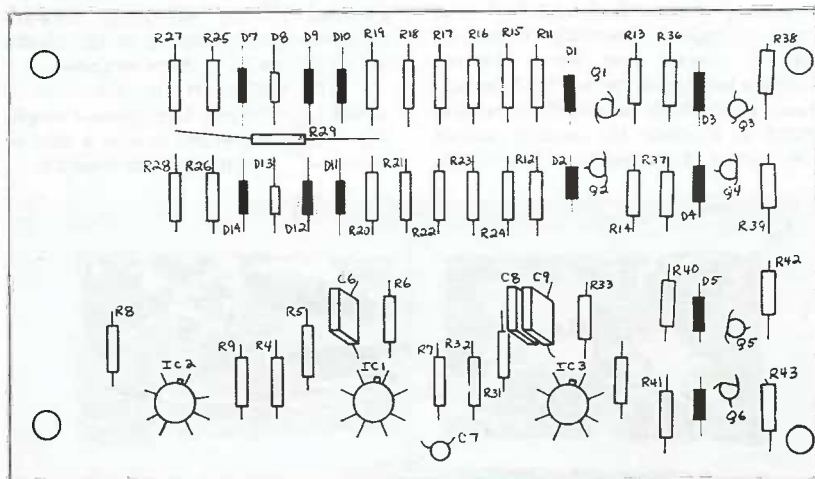


**INSIDE THE GENERATOR** you can see how parts are set up.





**PRINTED-CIRCUIT PATTERN**  
is half actual size.



**PARTS LAYOUT** shows component  
location on the circuit board.

pairs of wires as per the wiring list.

Resistor R1 is connected from the top of R10 to the top of R2, and resistor R3 is connected from the bottom of R2 to the ground post.

The completed PC board is attached to the center of the bottom plate with  $\frac{1}{4}$ -inch spacers.

The completed unit should be checked for proper operation. Notice

there are no adjustments to be made. If all components are good, and the wiring is correct, the unit will function immediately.

After checkout is completed, fold the PC board up into the chassis in a manner similar to closing a book. Attach the bottom plate with four  $\frac{1}{4}$ -inch No. 6 self-tapping screws.

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IND 30. M

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