TELEVISION

NEW DIMENSION IN TV SOUND

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SEMICONDUCTOR EDITOR

Stereo television may not be here, but this stereo synthesis system from RCA is the next best thing.

STEREO TELEVISION SOUND IS NOT HERE yet. Several systems have been suggested by various organizations, but apparently there has not been sufficient public demand to justify the development and implementation of a practical system. Perhaps we are waiting for improvements in monaural sound quality before the possibility of stereo seems real. RCA's CTC 101 color television chassis introduces a stereo-synthesis system called dual-dimension sound. Although it is not true stereo, the system produces a "stereo" effect from the transmitted monaural sound signal.

One of the most basic stereo-synthesis systems that has been proposed in the past is the simple division of the audio spectrum into two distinct bands—high and low frequency bands defined by high and low pass filters respectively. That method is far from satisfactory because of the obvious demarcation between the two sound channels. The new RCA system disguises the synthesis by intermixing the spectral sound ranges so they are not nearly as distinguishable.

Figure 1 shows the system schematic. The key components are a dual-section filter and two operational amplifiers, plus the two speakers necessary to produce the effect. A double-pole double-throw switch is used to disable the system when desired.

The audio signal from the sound detector feeds a 12-element R-C filter. The configuration may seem familiar to you—in fact it's dual-section twin-T filter. The two cascaded filter sections have been designed to have nulled responses at two different frequencies, one around 160 Hz and the other about 5000 Hz. The filter output is coupled to the noninverting input of amplifier IC1-a. Negative feedback between the amplifier's output terminal and the inverting input establishes the closed-loop gain. The output of amplifier IC1-a is coupled through a 100-μF capacitor and the system mode selection switch to the channel A speaker.

Now that channel A has been set up with three peaked bands of energy due to the filter's two nulls, a complementary response must be generated for the second channel. The second operational amplifier section produces the required double-peaked response without using any additional filter components. Op-amp IC1-b is configured as a difference amplifier; the original unfiltered input signal is routed to its inverting input, while the output of the twin-T filter is fed to the amplifier's noninverting input. Subtracting the flat input spectrum from the triple-peaked filtered signal forms the double-peaked channel signal response shown in Fig. 2. A stereo balance control varies the level into the inverting input of the second amplifier and is used to optimize the stereo-synthesis effect.

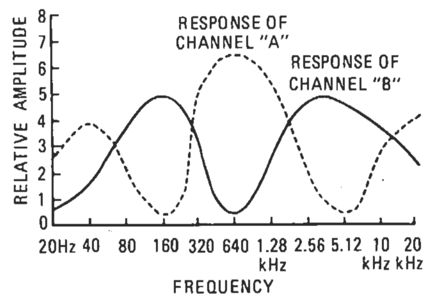


FIG. 2—FREQUENCY RESPONSE of synthesis sound circuit shows how the audio spectrum is split up to create a stereo effect.

The output of the second amplifier section drives the channel-B speaker. The spectrum in both channels is now distributed throughout the frequency range to give a pseudo-balanced effect. When the stereo balance control is properly adjusted, the total energy distribution in the two channels is the same as the energy content in the original input signal.

Negative feedback is also used around the second amplifier so that its gain matches the channel-A gain. Since the same filtered signal is applied to the inputs of both amplifiers and since the feedback is similar for the two channels, the output-signal levels are the same for both channels. The stereo balance potentiometer makes up for any tolerance imbalances that could disturb the important relationship between the A and B channel energy distributions.

When the mode switch is in the MONO position the system is bypassed, feeding the audio-input signal directly to the two speakers. **R-E**

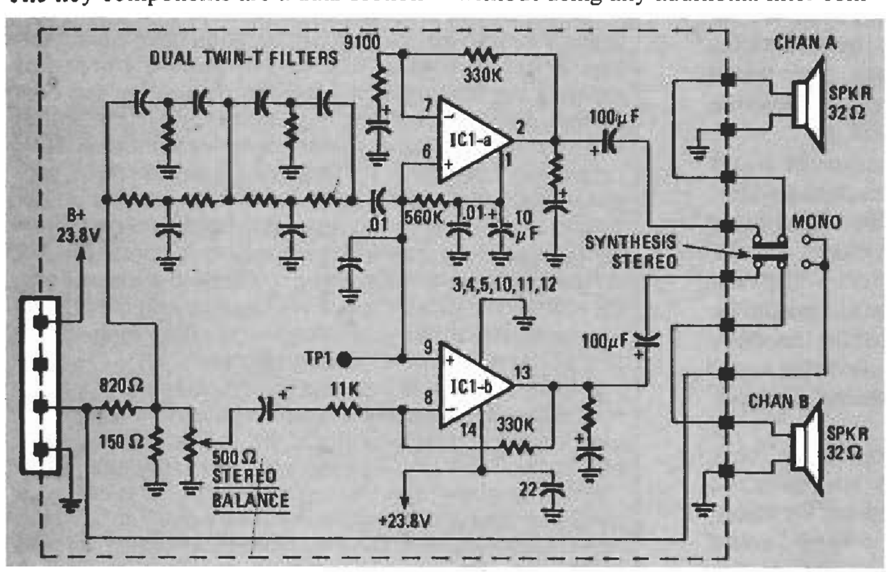


FIG. 1—SYNTHESIS SOUND CIRCUIT found in RCA's CTC-101 TV chassis produces a simulated stereo effect.

ENHANCE TV SOUND WITH STEREO

Synthesizer produces impressive stereo sound from the mono output of any TV receiver or video cassette recorder

BY JOEL M. COHEN

STEREO sound for TV broadcasts has not yet been established in the U.S. Anyone who has listened to it in Japan, however, quickly becomes aware that it provides a quantum jump to a higher level of listening pleasure, even with inexpensive speakers. Now you, too, can enjoy the benefits of stereo TV sound through use of the device presented here.

The project to be described is a stereophonic sound synthesizer. It generates synthetic stereo from a monophonic source such as produced by TV sets, video cassette recorders, and most video disc machines when interconnected with any stereo system.

Synthetic Stereo Theory. The perceived advantage of stereo over mono sound reproduction is a clear sense of more spaciousness and, thus, more realistic audio. Theoretically, accurate stereo reproduction relies on precise time and phase relationships between the channels and requires a centrally located listening position. This is because a person senses the location of a real sound source by the difference in arrival times of sound at the ears. Much stereo material, however, uses differences in amplitude between the two channels as a less critical way of indicating the relative positions of particular sounds. This amplitude-based stereo does not give as fine a focused sonic image, but it is more tolerant of listening position. For example, a sound which appears only in the left speaker cannot sound as if it is coming from the right side of the room.

On the same basis, if a mono signal could somehow be broken up so that some of it were put in the left channel and some in the right channel, it would begin to sound very much like most stereo material. Thus, the basic technique for generating synthetic stereo is to divide the mono signal and distribute it between the two channels. The simplest example of such a device is a crossover network, where the bass is fed to one channel and the treble to the other. As you can imagine, there's a substantial difference between that and true stereo.

The professional technique for generating synthetic stereo involves the addition and subtraction of mono signal components. The mono signal is presented to both the left and right channels. It is also passed through one

or more bandpass filters. The output of these filters is then added to one channel and subtracted from the other. The frequencies passed by the bandpass filters then increase in volume in the channel to which they are added and decrease in volume in the channel from which they're subtracted. This technique has been used for many years to process old monophonic recordings into simulated or "rechanneled" stereo records.

How the VSP-1 Creates Stereo. The Stereo Synthesizer uses a simpler, but much more effective "add and subtract" system to create a very believable stereo illusion. It is based on the comb filter effect.

If a signal is delayed and then mixed back with itself, the frequency response of the system ends up with a series of peaks and dips. When there are many of these, the amplitude vs frequency plot looks like the teeth on a comb.

In this synthesizer, a bucket-brigade device (BBD) is used to generate the delay for the comb filter. With a BBD delay line, you can break the audio spectrum up into as many pieces as you wish with much less complexity than using many individual bandpass filters. But, if the spectrum is too finely divided, the difference between channels will disappear. Each individual sound source will seem to spread across the whole width of the space between the speakers because every slight shift in pitch will move the sound from one channel to the other.

Extensive listening tests have shown that the most stable and convincing stereo simulation is obtained by dividing the spectrum evenly into approximately 1-kHz pieces. The delay line in the device is therefore set for approximately 0.5 ms.

Circuit Operation. Figure 1 is a block diagram of the synthesizer, and Fig. 2 is a plot of the frequency response for the two output channels. The signal fed into the delay line is derived directly from the input. Since the delayed output feeds a phase splitter, the original signal (delayed by 0.5 ms) is added to the right channel and subtracted from the left.

At very low frequencies, the delay is much less than the wavelength of the signal. Thus, the delayed signal cancels the original signal in the left channel and adds to the level in the right channel. Low bass, then, is approximately 6 dB louder in the right channel and almost totally missing from the left channel. At 1 kHz, however, the delay of one-half millisecond equals one-half the wavelength of the signal. Therefore, the two phases are effectively reversed and the level drops in the right channel and doubles in the left channel.

For the purpose of the preceding description, the delayed signal was considered to be exactly equal in amplitude to the main signal. Therefore, the maximum amplitude was increased by 6 dB, and the minimum amplitude was a complete null. (If the delayed signal were slightly lower in amplitude than the main signal, the peak would be less than double in one channel and above zero in the other. This would result in *less* of a sense of stereo spread.)

In the synthesizer, however, the delayed signal is actually set at a slightly higher level than the main signal. The result is that, while one channel is at maximum amplitude, the opposite channel has a small out-of-phase output. This effectively widens the image slightly beyond the boundary of the loudspeakers and, at the same time, adds presence, moving the apparent sound source forward into the room.

Figure 3 is a detailed schematic of the circuit. The main power supply is a conventional, regulated ±15 volts. The additional +14-V source is a special low-noise positive voltage for the BBD and phase splitter stages.

The left input signal is selected by *S1B* from one of two inputs. It passes through a 16-Hz, high-pass, subsonic filter formed by *C1* and *R1* plus *R2*, to the input of the mixer/buffer amplifier *IC1D*. Resistor *R27A* (one-half the volume control) sets the gain of that amplifier, which directly drives the output via a current-limiting resistor, *R24*.

Since the gain is a maximum of one, the volume control acts as an attenuator. This is necessary because television-derived audio signals can be much higher than conventional audio sources (the typical output of a tuner, for example, is 500 mV).

Switch *S2* selects either the stereo or mono mode. In the stereo source mode, the right input signal, which is selected by *S1A*, passes through a high-pass filter (formed by *C2* and *R21* plus *R22*) and into the mixer/buffer amplifier *IC1A*. In the mono source mode, the junction of *R21* and *R22* is grounded, thereby removing the right input signal from the right-channel amplifier. The input to the right-channel amplifier now comes via *R23* from the output of the delay-line driver stage *IC1C*.

The noninverting delay-line driver, *IC1C*, takes its input from one-half the left-channel input signal at the junction of *R1* and *R2*. Its output is also one-half the left input signal, which assures that the signal peaks in the delay line are well below maximum. Note that since *R23* is exactly one-half of *R21* plus *R22*, the output level of the right channel will be the same in both stereo and mono source modes. The output of the delay-line driver passes through a 15-kHz, low-pass, anti-aliasing filter formed by *R3* and *C4*. Capacitor *C3* blocks the BBD input bias voltage set by *R5* and filtered by *C5*. The stereo synthesizer function is bypassed by grounding the junction of *R3* and *C3* through *S3A*, thereby removing the signal from the delay-line input.

The BBD is driven by a two-phase clock formed with five of the six

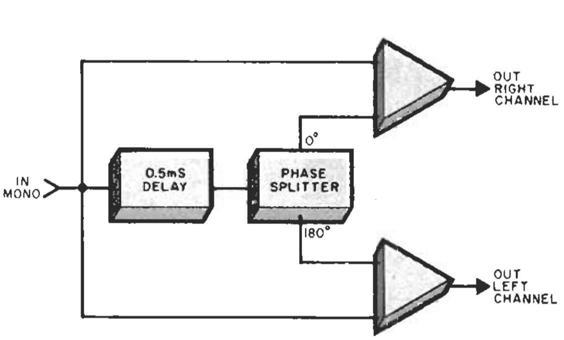


Fig. 1. The delayed signal is added to the right channel and subtracted from the left.

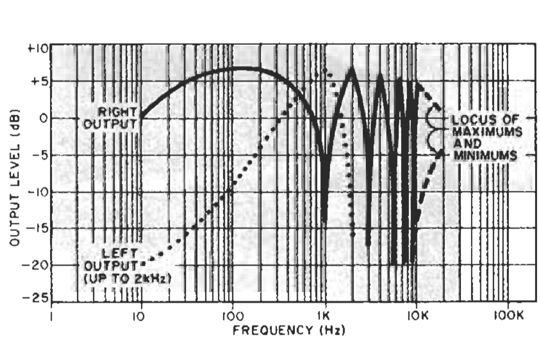


Fig. 2. At low frequencies, the delayed signal cancels the original in the left channel.

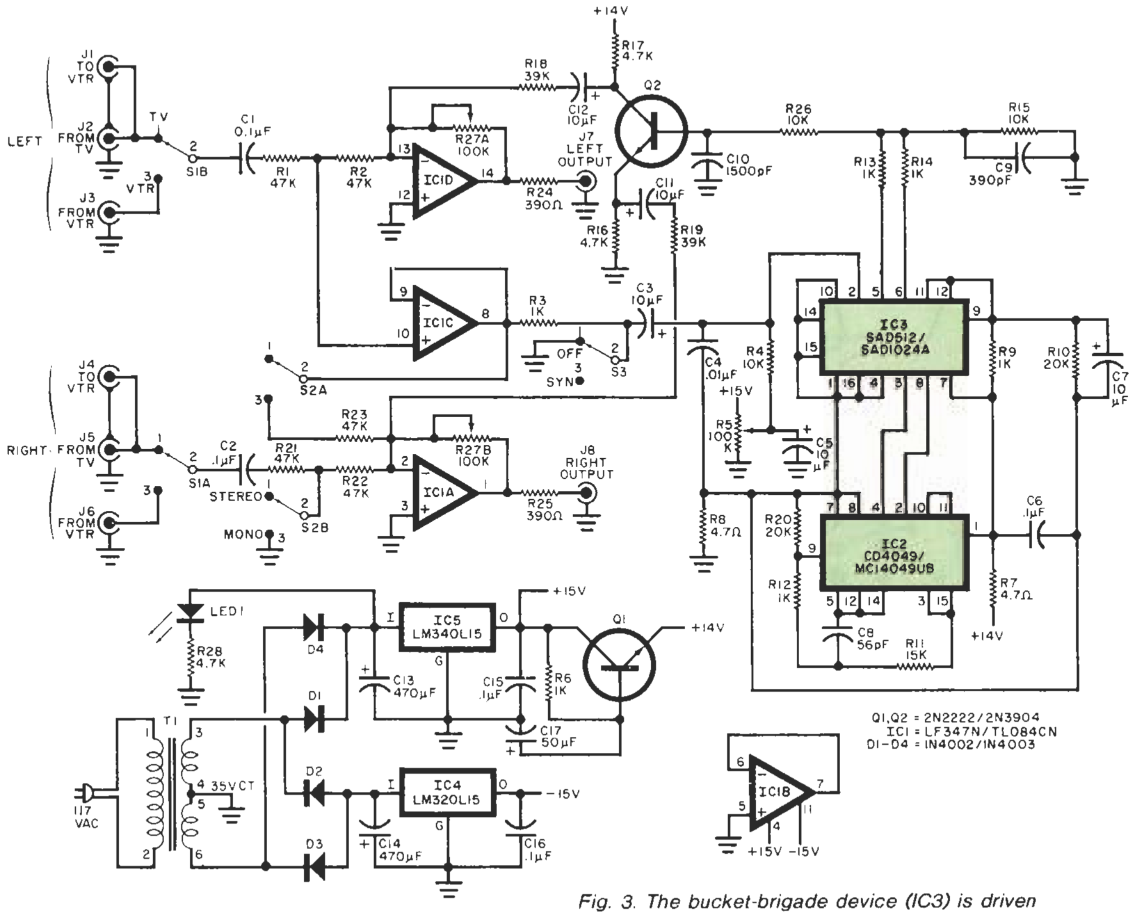


Fig. 3. The bucket-brigade device (IC3) is driven by a two-phase clock formed by inverters in IC2.

CMOS inverters in *IC2*. The nominal clock frequency is 550 kHz. The SAD512 (*IC3*) is a 256-stage, bi-phase, n-channel, bucket-brigade circuit that operates as an analog shift register. It is identical in function and pinout to one-half of the SAD1024A, which may be used in its place.

The delay time through *IC3* is 465 µs. (256 times the period of 550 kHz.) Its balanced output goes through *R13* and *R14* and is filtered by *C9* and *R15* and *R26* and *C10*. The signal is then applied to *Q2*, a buffer and phase splitter. The in-phase delayed signal at the emitter of *Q2* is ac coupled by *C11* and summed via *R19* into the right-channel mixer amplifier. The out-of-phase delayed signal at the collector of *Q2* is ac coupled by *C12* and summed via *R18* into the left-channel mixer.

When the delayed signal is exactly in phase with the original, the output from the right channel increases to about 6.5 dB and the output from the left channel goes out-of-phase at −20 dB. In Fig. 2, the actual amplitude of these peaks and nulls can be seen to vary across the spectrum as a result of the selected filter corner points, so that the out-of-phase component for

PARTS LIST

C1,C2,C6,C15,C16—0.1-µF disc ceramic capacitor
C3,C5,C7,C11,C12—10-µF, 35-V aluminum electrolytic
C4—0.01-µF axial ceramic capacitor
C8—56-pf axial ceramic capacitor
C9—390-pf axial ceramic capacitor
C10—1500-pf axial ceramic capacitor
C13,C14—470- or 1000-µF, 35-V aluminum electrolytic
C17—50-µF, 35-V aluminum electrolytic
D1 through D4—IN4002 or IN4003 rectifier
IC1—LF347N or TL084CN quad BiFET amplifier
IC2—CD4049 or MC14049UB CMOS hex inverter
IC3—SAD512 (single) or SAD1025 (dual) n-channel BBD
IC4—LM320L15 −15-V regulator, 100 mA
IC5—LM340L15 +15-V regulator, 100 mA
J1 through J8—dual RCA phono jack, right-angle pc mount
LED1—Fairchild MV5053 or equivalent
Q1,Q2—2N2222 or 2N3904 NPN transistor
The following are ¼-W, 5%-tolerance resistors unless otherwise noted:
R1,R2,R21,R22,R23—47-kΩ
R3,R6,R9,R12,R13,R14—1-kΩ
R4,R15,R26—10-kΩ
R5—100-kΩ trimmer potentiometer
R7,R8—4.7-Ω
R10,R20—20-kΩ
R11—15-kΩ
R16,R17,R28—4.7-kΩ
R18,R19—39-kΩ
R24,R25—390-kΩ
R27—dual 100-kΩ potentiometer
S1 through S3—3-switch assembly (3 dpdt or 1 dpdt and 2 4 pdt).
T1—35-VCT, pc-mount, Dale PL-12-09 or equivalent
Misc.—Printed-circuit board, chassis, cover, line cord, strain relief, LED clip, hardware, and knobs.

Note: The following is available from Sound Concepts, Inc. P.O. Box 135, Brookline, MA 02146: complete kit with all electrical and mechanical parts (KVSP-1) at $90.00. Also available separately: pc board (KVSP-2) at $16.00; transformer (KVSP-3) at $7.50; phonojacks, switches, pots (R27,R28) and knobs (KVSP-4) at $12.50; semiconductors D1-D4, LED1, Q1 and Q2, IC1-IC5, and sockets for IC1, IC2, and IC3 (KVSP-5) at $18.00. Add $2.00 for shipping and handling. Massachusetts residents, add 5% sales tax. If possible, give a street address for UPS delivery. Outside the continental U.S., add 10% or $5.00 minimum for parcel post.

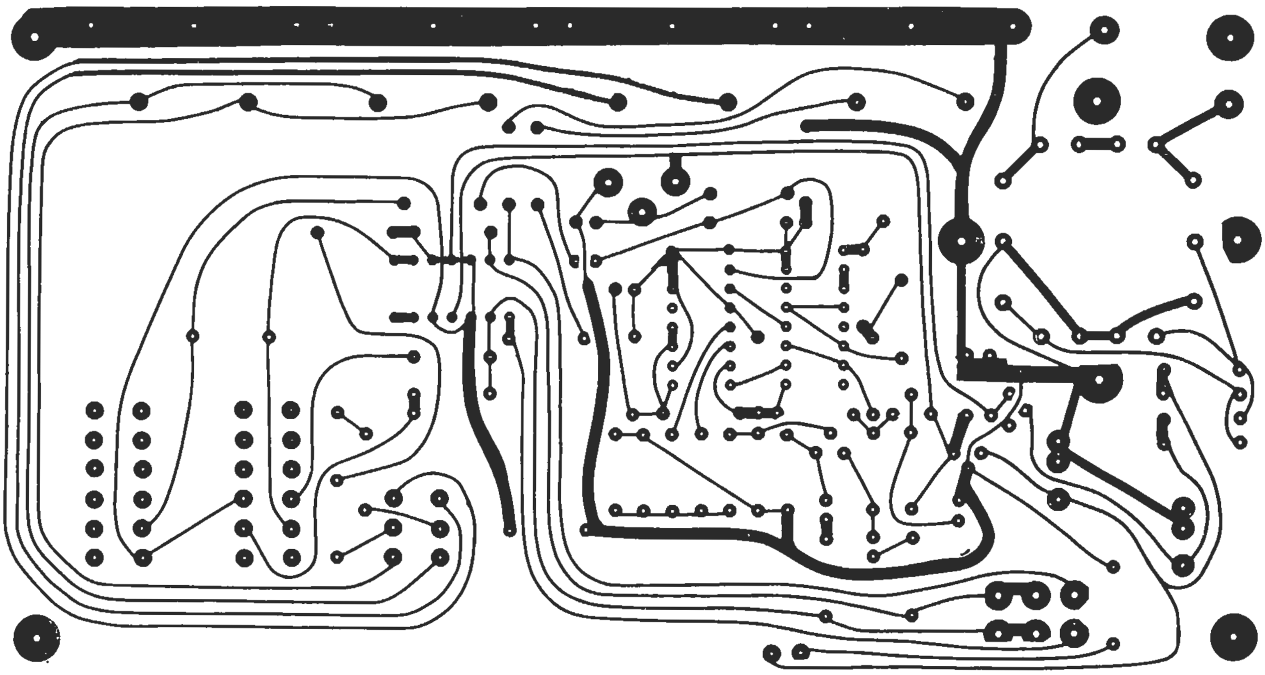


Fig. 4. Actual-size foil pattern for printed-circuit board.

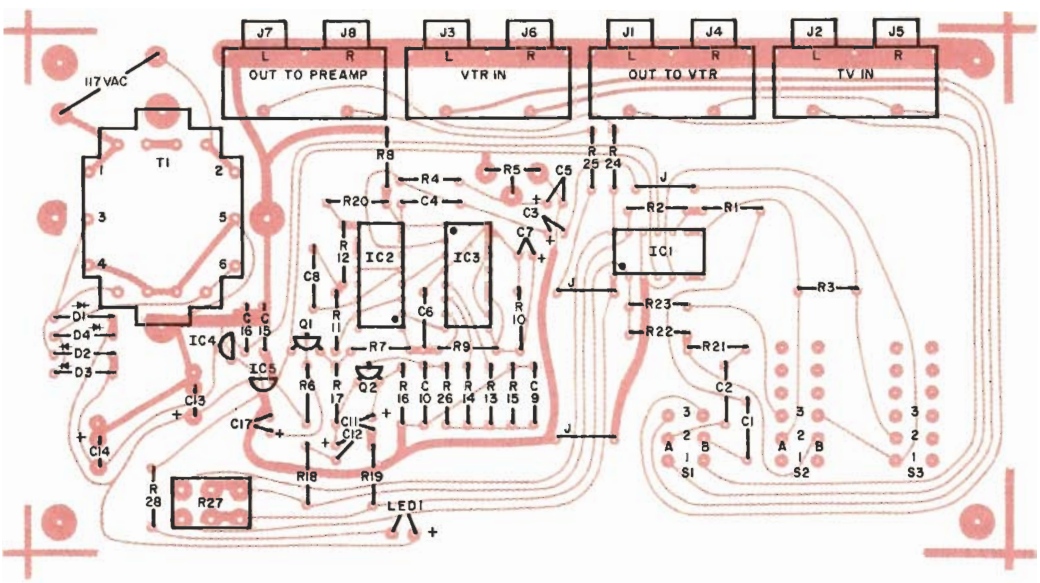


Fig. 5. Follow this component layout scheme for the pc board.

additional stereo-spread effect occurs only at the low and middle frequencies. In between the maximum and minimum extremes, the relative amplitudes in the two channels vary with frequency to create a full pseudo-stereo spread.

Construction. There are no panel-mounted parts in the project. All electrical-mechanical components are on the circuit board (Fig. 4). Almost everything is obvious from the parts-layout diagram (Fig. 5), but you should be careful about a few things.

Be sure that the transformer is mounted with pins 1 and 2 facing the back of the board. Once the transformer has been firmly pushed down and seated into the board, the two tabs should be bent under the board for mechanical strength and soldered to the pads on the foil side, grounding the transformer's frame. The extra pads around the transformer are for an alternate dual-primary substitute.

Cut off the plastic tabs at the rear of the dual phono jack assemblies. Snap the front ground tabs into the slots in the ground bus at the back of the board and solder them in place. Also solder the center terminals of the jacks, and be sure that the jacks themselves are lined up parallel to the board's surface. The volume control should be firmly inserted so that its shaft is parallel with the top surface of the board before you solder it. The switches either have shoulders on the pins or three plastic spacers that set them 3/32 inch above the board's surface. It is important that all these components be firmly seated and parallel with the board if they are to pass properly through the holes in the kit chassis.

The negative terminal is marked on the parts layout for each of the electrolytic capacitors. Those supplied in the kit have the negative terminal marked on the body of the capacitor. If you purchase your own, check to see whether the negative or positive terminal is identified.

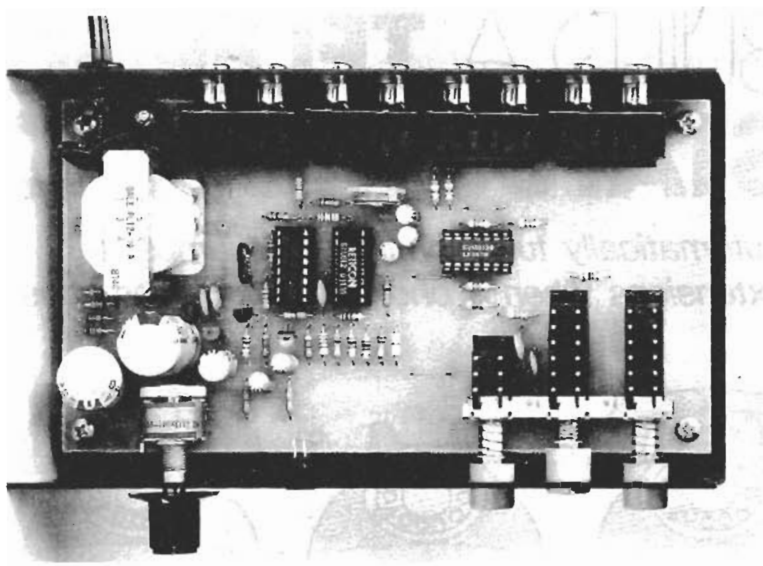
Integrated circuits *IC4* and *IC5* are voltage regulators that look like transistors. They mount with the center lead bent back toward the curved side of the body. The two npn transistors mount differently, with the center lead bent forward toward the flat side of the body before insertion. The dual-in-line ICs have a small indented dot adjacent to pin 1. Be sure to use that for location, and do not pay attention to the way the brand is oriented. In addition to the normal components, three wire jumpers are required as shown on the parts layout.

The leads on the LED should be preformed, so that it aligns with the window in the kit chassis. Hold the LED so that it faces you with the positive lead on your right. Then make a 90-degree bend downward 3/16 inch behind the back of the lens body. Insert the leads into the circuit board so that there is 1/2 inch from the bend down to the top surface of the board (LED facing forward). Then solder the leads.

To insert the board into the chassis, push *S1*, *S2*, and *S3* to the "in" position and tilt the front edge of the board down into the chassis. Put the volume control shaft through its hole and push the front edge of the board against the inside front panel.

At this point, the back edge of the board can drop down, and the board can move back so that the phono jacks extend through the rear panel. The four pc-board mounting holes should line up with the standoffs in the bottom of the chassis. Four No. 6, 1/4-inch sheet-metal screws are used to mount the board onto the standoffs.

The volume control and switch knobs can now be attached and the line cord on its strain relief brought through its hole on the back panel. Be careful of stray strands from the power cord hitting the grounded mounting screw in the left rear corner when you solder the line cord to the terminals on the pc board. The cover slips over the chassis and is held in place with four No. 440 Phillips-head screws. Four self-adhesive, black-rubber, furniture-protecting bumpers may be affixed to the bottom of the chassis or the cover to complete the assembly.



Internal view of the author's prototype shows convenient arrangement of controls on front panel and jacks on rear.

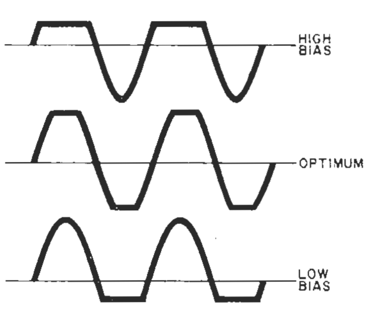


Fig. 6. Optimum bias adjustment causes minor clipping top and bottom.

Test and Alignment. There is only one adjustment necessary. Start by setting the bias on pin 2 of the BBD, (*IC3*) to approximately 4.5 V. To optimize the adjustment, put a high-impedance dc voltmeter probe on the arm of *R5* to ground. Place an oscilloscope probe at the junction of *C11* and *R19* to observe the filtered output of the BBD. Put a signal generator into the left TV-input jack. The generator should be at 1000 Hz and 1.5 V rms. Switch *S1* should be out, and *S2* and *S3* in for this test.

If the bias is set too low, the bottom of the sine wave will be clipped; if it is too high, the top of the sine wave will be clipped. The optimum adjustment is in the center, where there is either no clipping or just a slight symmetrical clipping of the top and bottom of the waveform (Fig. 6).

Connection and Use. Although labeled TV and VTR, the two inputs may come from any mono or stereo source in the signal range of 100 mV to 1 V. You could, for example, put the synthesizer in the tape loop of your preamp or receiver and use it with mono records and AM or mono FM broadcasts. Whatever your source, if it only has a single mono output line, it should be connected into the left input terminal on the unit. When listening to it, *S2* should be in the mono position.

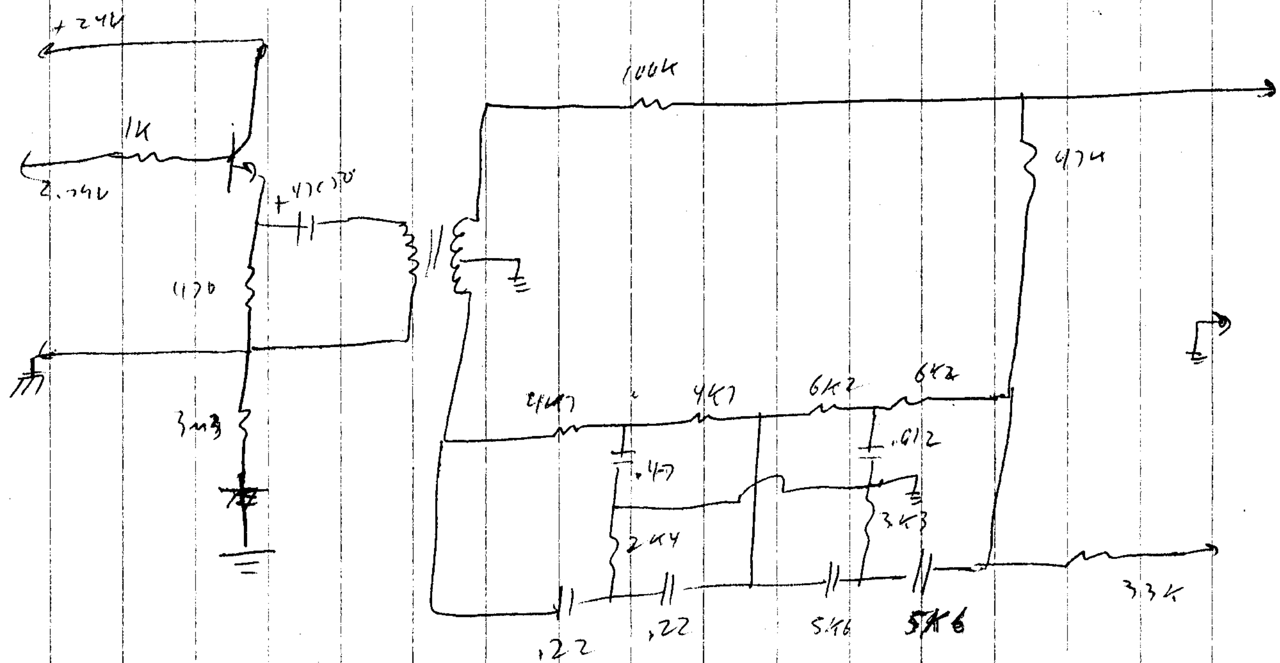
With the volume control turned all the way up, the synthesizer has a gain of one. Its primary function is to equalize the volume of signals coming out of the unit to the others in your stereo system. As mentioned previously, this is done because video sound voltages are typically 10 to 20 dB higher in level than FM tuners or phono preamps.

With *S2* in STEREO and *S3* at OFF, only the volume-control function is active on the unit. You should note that, since there is only a single inverting amplifier between the input and output of the synthesizer, it inverts the phase of both channels by 180 degrees.

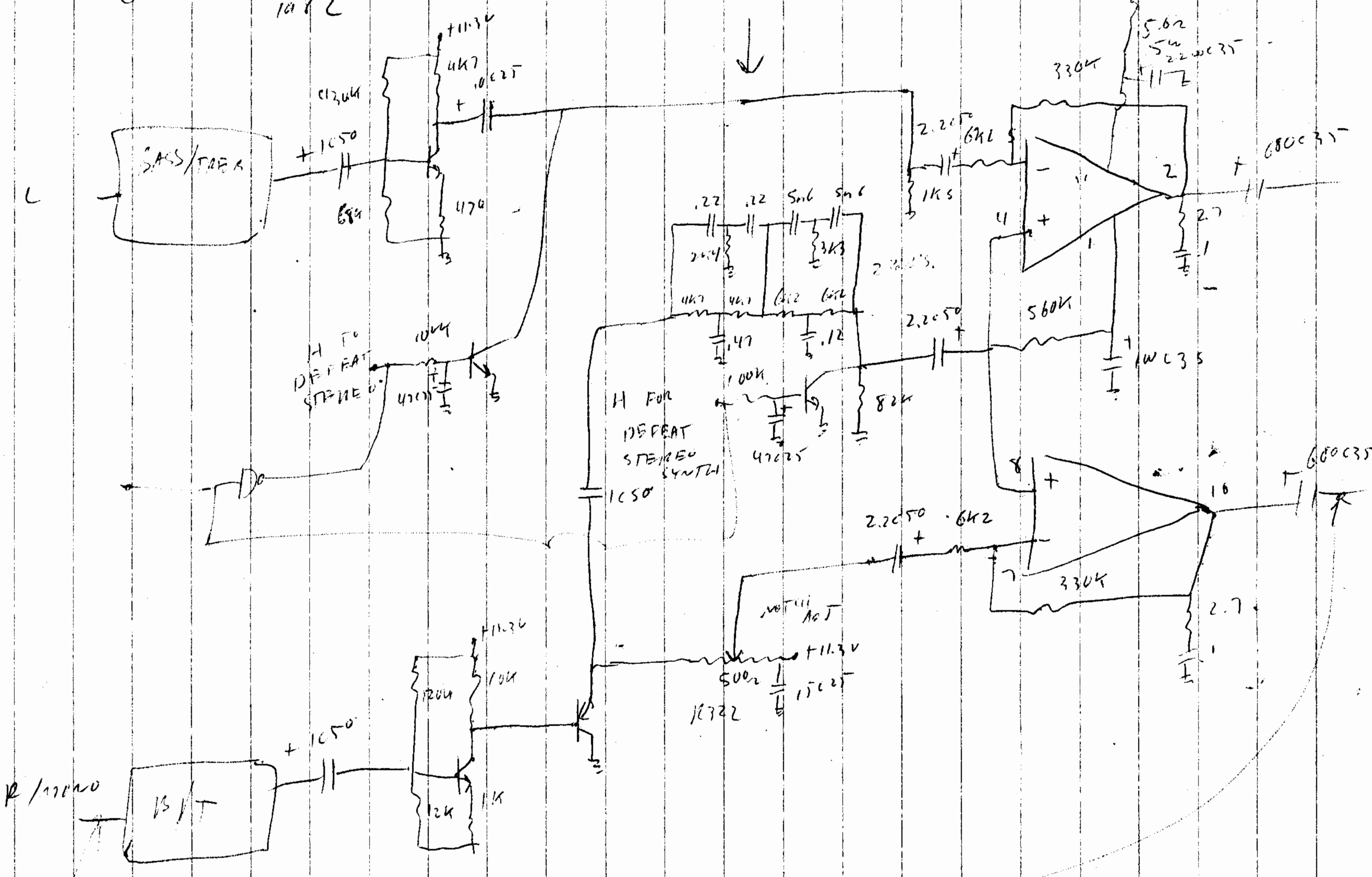
TV Connection. If your TV set has an audio output jack, it is a simple matter to connect it to the synthesizer. The next best alternative is an earphone jack, which also allows for easy connection. If you're audio-quality conscious, you have other options. The best would be to buy and use a separate TV sound tuner. This would enable you to take advantage of the hi-fi sound (50 to 15,000 Hz) broadcast by the networks. Should you wish to make a hardwire connection internally to, say, the sound detector output, you should check to be sure there will be no ac flow between the TV set and your stereo sytem. Another option is to connect to the loudspeaker in the television set, though this will diminish the sound quality. ◇

DW220 Dual Dimension Sound Output

1984-CTC121



CTC121 PW 44000 1072 — Blinco synth.



TO set R322
feed in 925 Hz ABT 30mV

adjust R322 for minimum output.