

# BUILD THIS



## Mini Music Synthesizer

Turn your voice into a versatile musical "instrument" with this fun-to-build mini music synthesizer.

THIS MINI-MUSIC SYNTHESIZER IS NOT TOO complex, but it is a load of fun to build and use. The ingenious little unit samples and holds the frequency of a signal from a microphone or other sound source and outputs that signal as a single note. The input "signal" can be a whistle, hum, or some other sound. The duration and amplitude of the note can be controlled. The frequency of the note can be halved or doubled with a flick of a switch. A vibrato effect can also be applied via a variable tremolo control. To add to all of that, two channels are available to give chorus or stereo effects. Each of those channels has a separate volume control.

The signal source is derived from a low-to medium-impedance microphone. That microphone can be almost any dynamic or electret type. The types used with low-cost cassette recorders will work well.

The synthesizer's output is connected to an external amplifier of some type. A home-stereo system would be suitable; feed the signal to the auxiliary or tuner input. A PA amplifier could also be used.

### How it works

The schematic of the synthesizer is shown in Fig. 1. As you can see there, its input section is basically a microphone amplifier and signal shaper with three stages. Twin Voltage Controlled Oscillators (VCO's) each with a Voltage

Controlled Amplifier (VCA) output stage make up the main section. The remaining circuitry is a simple, adjustable low-frequency oscillator that's used as a tremolo-effects generator.

The first stage of the input section is made up of IC1-a and its associated circuitry. That stage has a gain of about 100 at 1 kHz. The op-amp circuit used in that stage is very basic. Note, however the presence of the capacitor in the feedback loop. That capacitor causes a roll-off in gain as the input frequency increases. The output of the amplifier is AC coupled to the next stage by C7.

The second stage consists of IC1-b and its associated circuitry. That stage has a gain of about 20. Like the previous stage, the high frequency response is limited by a capacitor in the feedback loop. That response tailoring has been done to reduce the normally rich harmonic content of the human voice; the only signal we want to process is the fundamental note. The harmonics are low in amplitude and can be partially rejected by the simple approach used in that circuit.

The last stage of the input section is the most significant and needs some explanation. The op-amp there, IC1-c, is configured as a Schmitt trigger with offset. The hysteresis components are R22 and R21. Normally a Schmitt trigger is bistable, and in the quiescent state the output would be either positive or negative depending on the last signal transition. The difference in this case is that the offset volt-

age at the inverting input is greater than the hysteresis voltage at the noninverting input; resistors R20 and R39 establish that offset. The purpose of the offset is to assure that the output at pin 8 of IC1 is always low (negative) when no signal is present. The output will only switch when the input exceeds the sum of the hysteresis and offset voltages. The circuit acts as a gating system to the following phase-locked loop circuits.

The Schmitt trigger is the second stage of the processing that rejects the harmonics and noise in the input signal from the microphone. The input level, set by R41, is adjusted by turning that control until reliable response is achieved from a normal-level input (such as a singing voice). It is not adjusted farther than that minimum amount. In that way, the Schmitt trigger will tend to switch only on the peaks of the fundamental. The levels of the harmonics and noise content of the signal are below the threshold switching points and will be rejected. The purer the voice or note, the more reliable the circuit action. A "gravelly" or "rough" voice thus will tend to prove unreliable as a signal source. The best results will be from a whistle, because that produces a note that is relatively free from harmonics of any amplitude.

From the output of IC1-c, the signal is split and fed to both the right and left channels. Since the circuitry in the two channels is identical, we will look at only one of those: channel 1.

Adapted from a project that originally appeared in Dick Smith's *Funways into Electronics*, volume three.

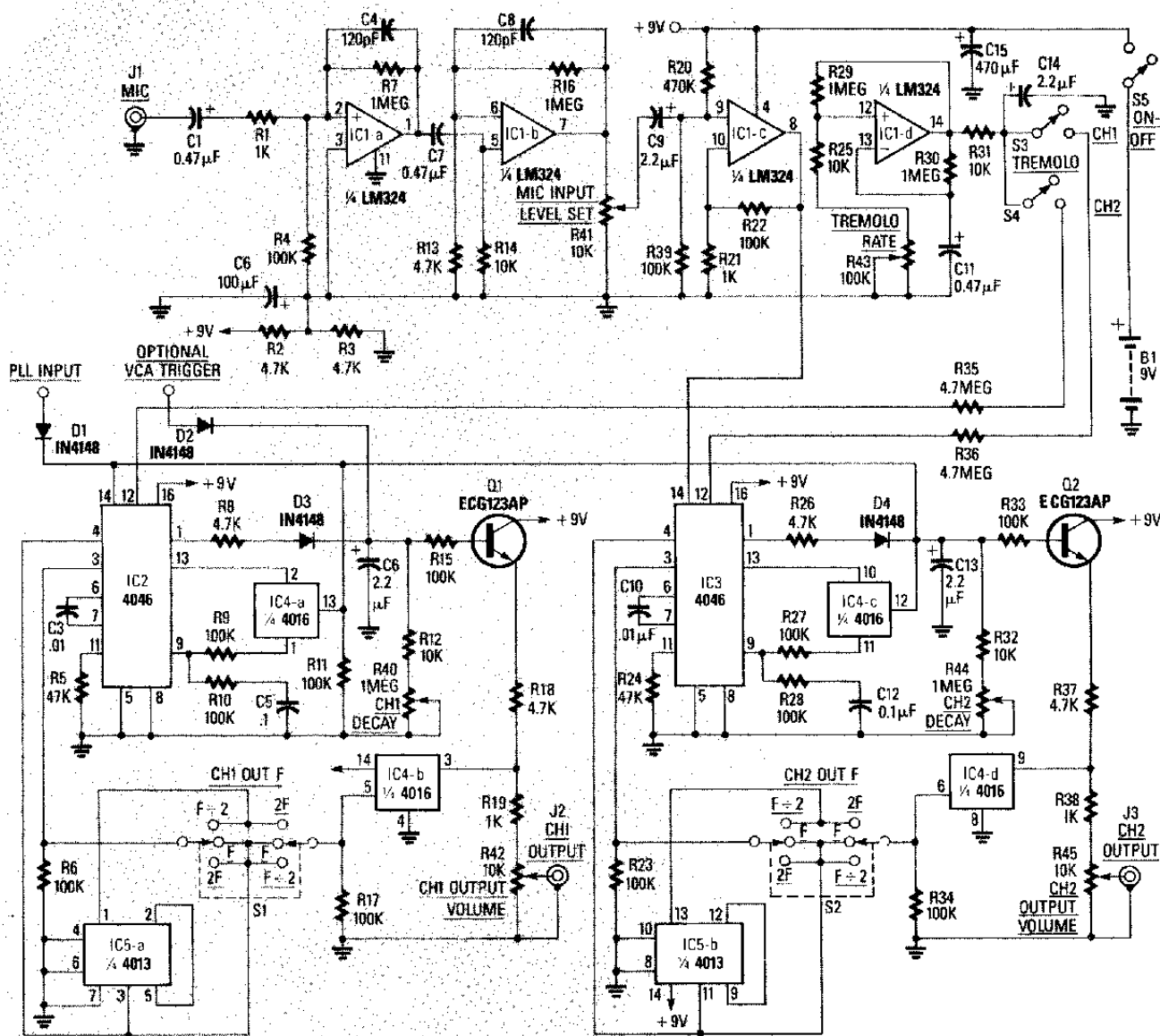
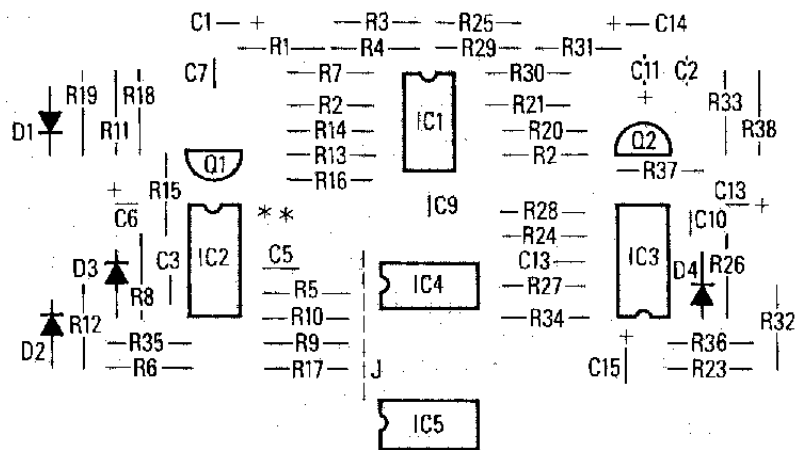


FIG. 1—SCHEMATIC DIAGRAM of the mini synthesizer. The potentiometers can be either panel-mounted or PC-mounted.



\* SEE TEXT

FIG. 2—THE LOCATIONS OF ALL PC-board mounted components are shown here. To disconnect the microphone input stage, cut the trace between the pads marked with an asterisk.

From the Schmitt trigger, the signal is passed to IC2, a 4046 CMOS Phase Locked Loop (PLL). That IC consists of two separate circuits. One is a VCO that runs from subaudio to over 1 MHz. The other is a dual-output phase comparator. By adding just a few external components, a complete PLL can be formed. (For more information on the 4046, see the manufacturer's data sheet.)

Without an input signal, the output of the Schmitt trigger will be low, and therefore pin 13 of IC4-a, a 4016 analog switch, will also be low. That means that the analog switch will be off, and pin 13 (PLL COMPARE-OUTPUT 2) of the 4046 will be disconnected from pin 9, the input to the VCO. The voltage present on the lowpass filter capacitor (C5), together with the RC timing components (C3 and R5), will determine the frequency of the

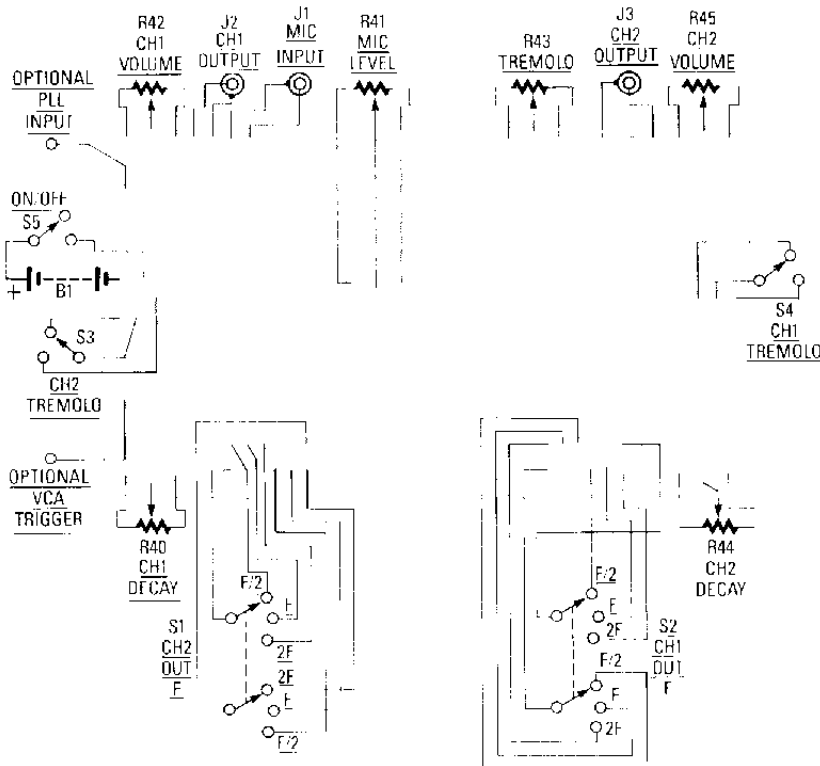


FIG. 3—THE OFF-BOARD COMPONENTS should be connected to the PC board via wires.

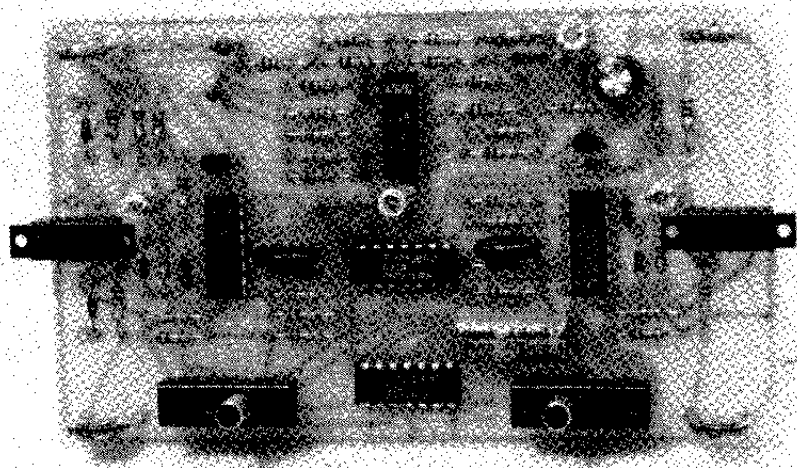


FIG. 4—THE FINISHED PC-BOARD is shown here. Note the use of PC-mounted trimmers in this version.

VCO. The voltage across C5 will not dissipate for a long period of time, because of the very high input impedance at pin 9 of the 4046, and the low-leakage characteristic of that 0.1- $\mu$ F polyester capacitor. That, in turn, means that the output frequency of the VCO at pin 4 will remain stable for a long period of time.

Now let us look at what happens when an input signal is present. Pins 13 of the analog switch and 12 of the PLL will go high. The analog switch is then on, and the phase-comparator output of the PLL (IC2, pin 13) will be connected to the VCO (IC2, pin 9) via the lowpass filter (R9, R10, and C5). Provided that the input signal continues at a fixed frequency for a

short period of time, the PLL will lock in and follow any frequency variations. For the most part, in that locked state, the PHASE PULSE output (pin 1) will be high (there will, however, be some narrow negative-going pulses).

With pin 1 of the 4046 high, D3 will be forward-biased via R8, so C6 will charge. The voltage across the capacitor is applied to the base of Q1, which acts as a voltage follower/buffer. That transistor is half of the VCA output stage, the other section being another 4016 analog switch (IC4-b). The control gate (pin 5) of IC4-b is connected directly to the output of the VCO (pin 4 of IC2), or indirectly via the 4013 (a dual D flip-flop), depending on the

## PARTS LIST

### Resistors

All resistors are 1/4-watt, 5%, unless otherwise noted.

R1, R19, R21, R36—1000 ohms  
 R2, R3, R8, R18, R26, R37—4700 ohms  
 R4, R6, R9–R11, R15, R17, R22, R23, R27, R28, R33, R34, R39—100,000 ohms  
 R5, R13, R24—47,000 ohms  
 R7, R16, R29, R30—1 megohm  
 R12, R14, R25, R31, R32—10,000 ohms  
 R35, R36—4.7 megohms  
 R40, R44—1 megohm, potentiometer, linear taper  
 R41, R42, R45—10,000 ohms, potentiometer, audio taper  
 R43—100,000 ohms, potentiometer, linear taper

### Capacitors

C1, C7, C11—0.47  $\mu$ F, 10 volts tantalum  
 C2—100  $\mu$ F, 16 volts, electrolytic  
 C3, C10—0.01  $\mu$ F, ceramic disc  
 C4, C8—120 pF, ceramic disc  
 C5, C12—0.1  $\mu$ F, polyester  
 C6, C9, C13, C14—2.2  $\mu$ F, 16 volts, electrolytic  
 C15—470  $\mu$ F, 16 volts, electrolytic

### Semiconductors

IC1—LM324 quad op-amp  
 IC2, IC3—4046 CMOS PLL  
 IC4—4016 quad analog switch  
 IC5—4013 dual D flip-flop  
 Q1, Q2—ECG123AP NPN transistor  
 D1–D4—1N4148 silicon diodes

### Other Components

S1, S2—DP3T miniature slide switch  
 S3–S5—DPDT miniature slide switch  
 J1—miniature phone jack  
 J2, J3—phono jack  
 B1—9-volt battery

**Miscellaneous:** PC board, case, knobs, battery snap, wire, etc.

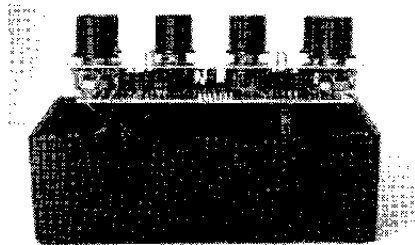
**A kit of parts (Catalog Number K-2669) is available from Dick Smith Electronics, PO Box 8021, Redwood City, CA 94063. The kit includes the PC board, but not the case, the battery or the jacks. The price is \$19.95.**

setting of S1. The squarewave output from the VCO opens and closes the analog switch. It can be seen that that action directly gates the voltage available to the output terminal via the two current limiting resistors and the potentiometer.

When the input signal disappears, pin 1 of the 4046 returns low. As described previously, the VCO remains running at a frequency determined by the voltage on capacitor C5. The voltage across C6 then begins to discharge via the transistor follower and the two resistors, R12 and R40. Those two resistors control the "decay" rate. If R40 is set to its maximum value (1 megohm) the discharge time will be long (decay will occur slowly). As the discharge is taking place, the continuous output from the VCO switches the analog gate IC1-b on and off to "sink" the decaying voltage on the emitter of Q1 via the load resistor (R18) at the VCO rate.

The frequency of the note reaching the output stage can be changed from that of the original. By using a flip-flop as a divide-by-two element, the output can be halved or doubled depending on where it is coupled into the circuit. With the octave-select switch (S1) in the center position (F), the output frequency will be identical to input. When the 1/2 position is selected, the 4013 (configured as a clocked flip-flop) divides the output from the VCO (IC2, pin 4) by two (lower octave). In the 2x position, the flip-flop is connected between the output of the VCO and the comparator of the PLL. That results in a frequency that is twice that of the input (upper octave) at pin 4 of the VCO.

So far we've only used three sections of the LM324 quad op-amp. The fourth section (IC1-d) is used as a tremolo generator. The op-amp is configured as an astable multivibrator to give a squarewave output. Potentiometer R43 is included to adjust the frequency (tremolo rate). The squarewave is then smoothed somewhat to give a more natural tremolo effect to the output note. That waveform is then applied to the VCO (at pin 12, IC2) via a selector switch and a 4.7 megohm resistor.



ONCE COMPLETED, the circuit should be installed in a case.

### Optional inputs

The VCA of either channel can be triggered from an external source. A positive pulse input via D2 to the VCA will charge C6. If that input is held high, the output from the VCA will also remain high. If the input is a pulse from a sequencer or even a simple switch, the output from the VCA can be triggered without the need for an audio signal at the microphone input. That could be used with rhythm generators, etc., to create different effects. The frequency of the VCO can still be changed by using the microphone input. If you require that the input signal from the microphone-amplifier stage not trigger the VCA, create an open circuit by removing D3 from the board. (The foil pattern for the project is provided in our "PC Service" department; the parts-placement diagrams are shown in Figs. 2 and 3.)

By disconnecting the input stage from the VCO, the PLL can be used separately by providing an external input signal. Isolation is performed by cutting the PC

board trace between the pads marked with an asterisk in Fig. 2. The input is applied via D1. That input should be a square-wave. The peak-to-peak voltage of the input should not be greater than the circuit's supply voltage, and should not be less than 75% of the supply voltage. If the input stage is disconnected from the PLL in the manner described, the circuit could be used to modify the output of an electronically amplified instrument or a sound generator (provided that the input signal meets the requirements of the 4046). For example, the output from a simple monophonic organ can be used to create more interesting tones and sound effects. The decay control could be used to vary the note shape, and the octave switch can change the note frequency.

To reconnect the PLL to the microphone input stage, the link destroyed when the PC trace was cut must be restored. That is done by installing a jumper between the pads marked with an asterisk in Fig. 2; for convenience, that jumper could be replaced by a switch.

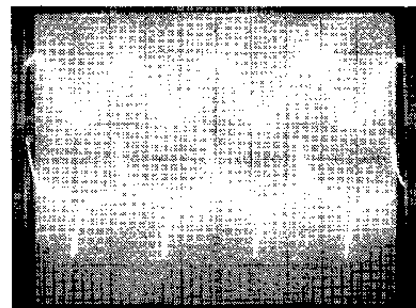
### Assembly

Assembly is very straightforward if you follow Figs. 2 and 3. Start by mounting all of the low-profile components on the board. Those include the resistors, diodes, and the jumper. Next, install the capacitors. Be sure to observe the polarity of the electrolytics. After that, install the IC's and transistors, taking care to observe the proper orientation. Finally hook up the switches, potentiometers, and other off-board components. Note that the board has been designed to accept PC-mount potentiometers, but you can use panel-mount potentiometers and connect them to the board with wire.

To test the system you will need to use an audio amplifier of some type. Feed the output of the synthesizer to the amp, plug in the microphone and battery, and turn the unit on. Whistle a few times close to, but not directly into the microphone. Turn up the MICROPHONE INPUT LEVEL SET control until some response is heard from the outputs. Set the VOLUME controls of both channels to an appropriate level. Vary the pitch (frequency) of your whistle; the output should vary in kind.

Next, try varying the DECAY controls. With the controls set to maximum, you will hear the outputs change in frequency as the note of your whistle changes. Notes that are wide apart in frequency will take a little longer to lock. Now try changing the setting of the OCTAVE SELECT switches. Also try out the TREMOLO RATE control; be sure that the tremolo section is switched into the circuit when you do that.

If all is working, the completed board (see Fig. 4) can be mounted in a case to complete assembly. If you detect any problems, go over your work carefully to find the cause of the problem.



THE UPPER TRACE is a 3-kHz input signal from a signal generator. The bottom one is the output that results.

(Photo courtesy of Tektronix)

### Operation

Connect the microphone and switch the unit on. Now slowly whistle a tune a short distance from the mouthpiece. Do not blow directly into the microphone. Best results are obtained with the microphone at the side of your mouth so that any air currents do not hit it directly.

Turn the input-level control up until you get a reliable response from the unit every time a note is whistled at the same volume. Turn the output-volume controls to a suitable level to avoid feedback. The whistled note should be a short, clean burst. Try changing the DECAY controls so that the generated sound varies from a short staccato to a long, slowly-decreasing tone.

Note that the synthesizer is sensitive to all sound. Thus, when the microphone picks up the output sound of the amplifier, feedback occurs and the system locks up in an uncontrolled state. To avoid that, you must keep the microphone away from the speaker system.

When you find a suitable input level, try the other functions. By changing the OCTAVE SELECT switches, the output frequency can be set to be the same as the input, or one half of the input, or twice the input. By adding tremolo to one or both channels, interesting tonal effects can be achieved.

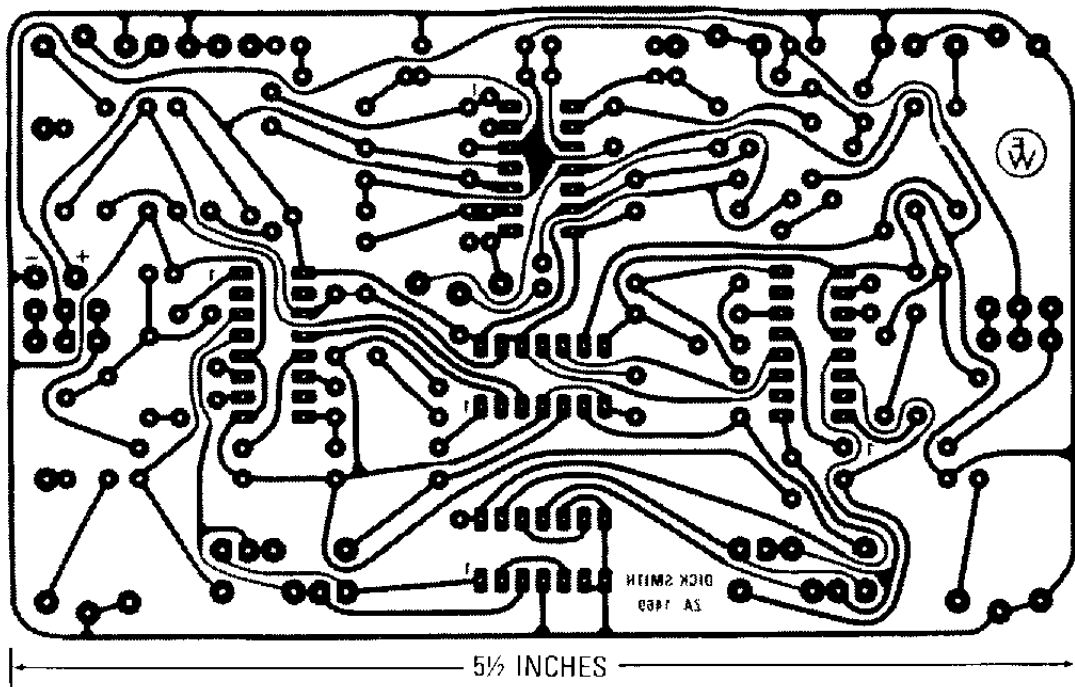
Try singing or humming into the microphone as the sound source. The system will respond best to pure, clean notes. Rough voices will not be reliable.

By striking different shaped objects next to the microphone, the system will tend to pick up the fundamental resonance of the object and produce an equivalent note.

Musical instruments can also be used as the sound source. A simple recorder, for instance, can produce many and varied sounds. Other instruments, such as a guitar, can produce different notes and sounds. The combinations are endless.

To give greater versatility to the unit, channel one can be controlled from external sources as described above. With a little practice, this mini synthesizer can make you a "one-man band." **R-E**

# PC SERVICE



OUR MINI MUSIC SYNTHESIZER can turn anyone into a "one-man-band." The project is fun, and easy-to-build if you use the foil pattern shown here. The story begins on page 75.

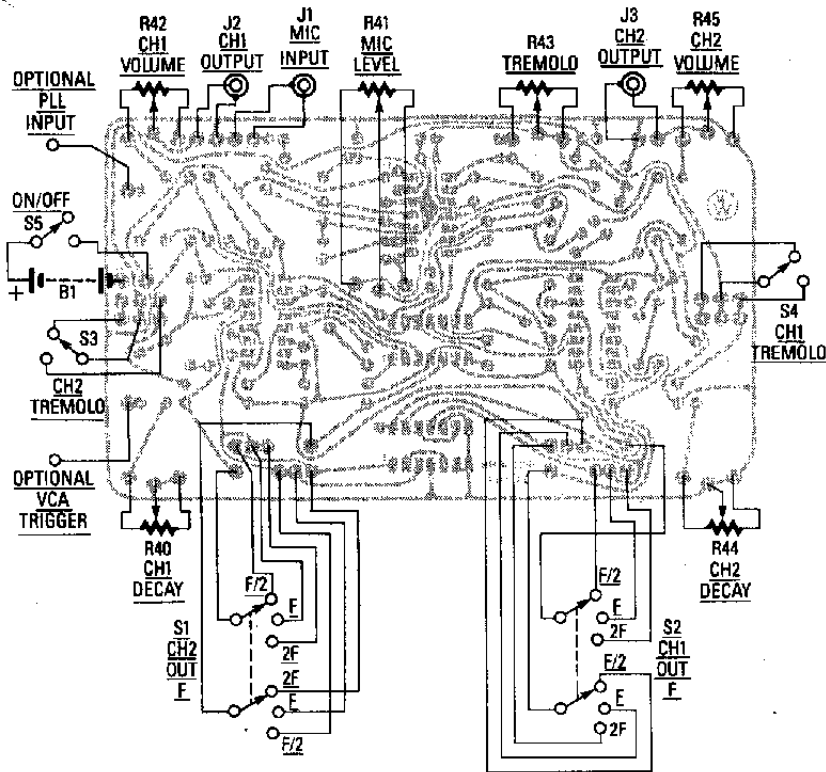


FIG. 3—THE OFF-BOARD COMPONENTS should be connected to the PC board via wires.