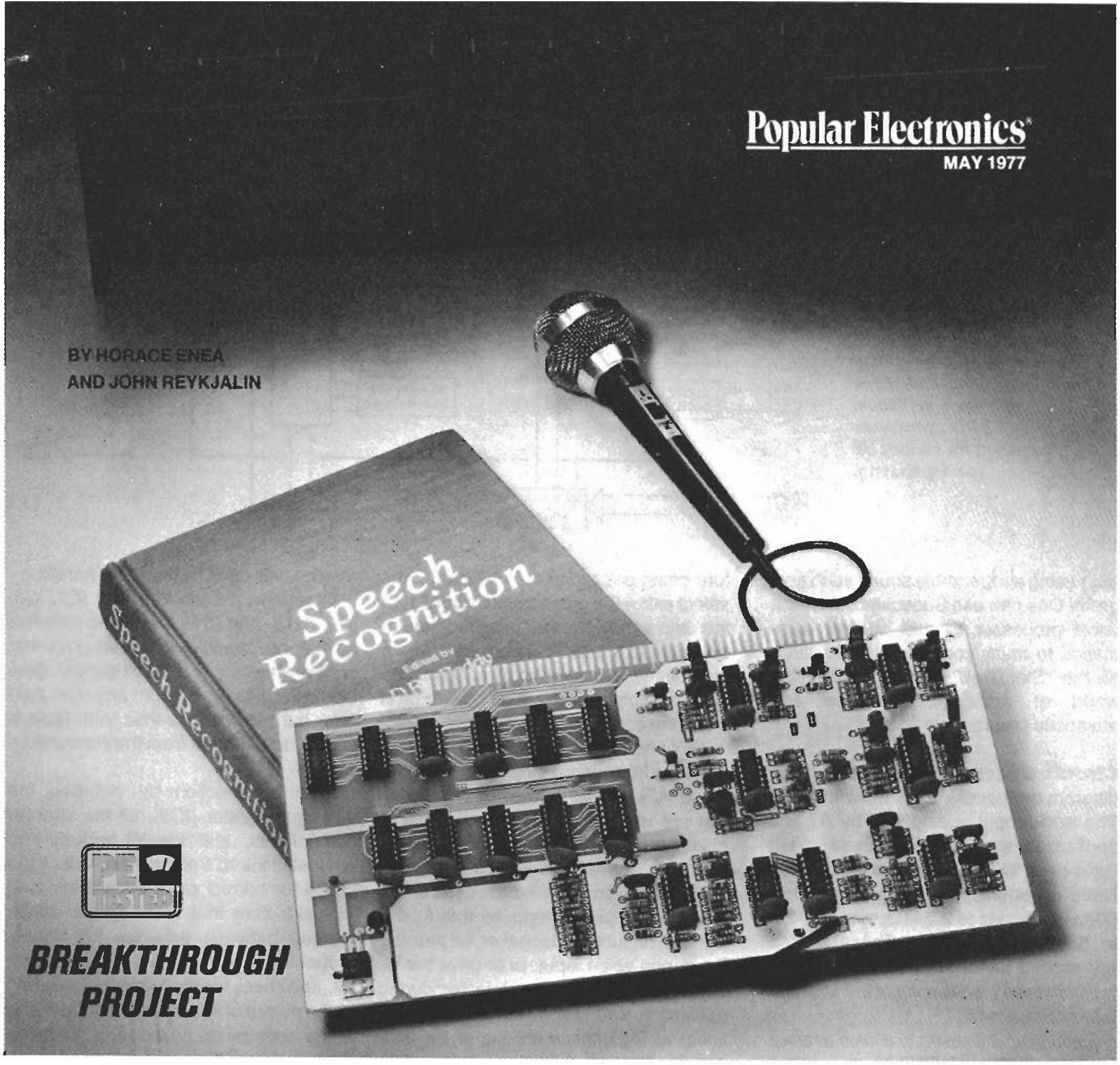


INTRODUCING SPEECHLAB

THE FIRST HOBBYIST VOCAL INTERFACE FOR A COMPUTER!

Now your computer can respond to vocal commands by the simple addition of a $250 single-board unit.

IMAGINE being able to *talk* to your computer and have it respond by way of a hard-copy device or by activating some external appliance! Computer hobbyists can now enjoy this facility by building "Speechlab," a new, low-cost (under $250) computer peripheral. To use it, all one does is plug the single Speechlab pc board into an Altair-bus connector (used by many microcomputer manufacturers), enter a special program, and the computer does the rest. It's a state-of-the-art approach at a moderate cost.

One section of the program allows the user to "train" the computer to accept a vocal input (via a microphone), analyze the spoken word, and create a digitized version that is stored in memory. The second part of the program allows the user to speak to the Speechlab and have the computer generate the output selected for that particular sound.

The vocabulary size of Speechlab is a function of the speech recognition algorithm used and the amount of memory available. For the program used in this article, it is 64 bytes per spoken word.

The unique characteristics of Speechlab open many formerly closed doors. Since Speechlab will operate with any audio input (not necessarily a recognized language), a person who's vocally handicapped can operate almost any number of appliances (TV receiver, stereo system, solenoid-operated door,

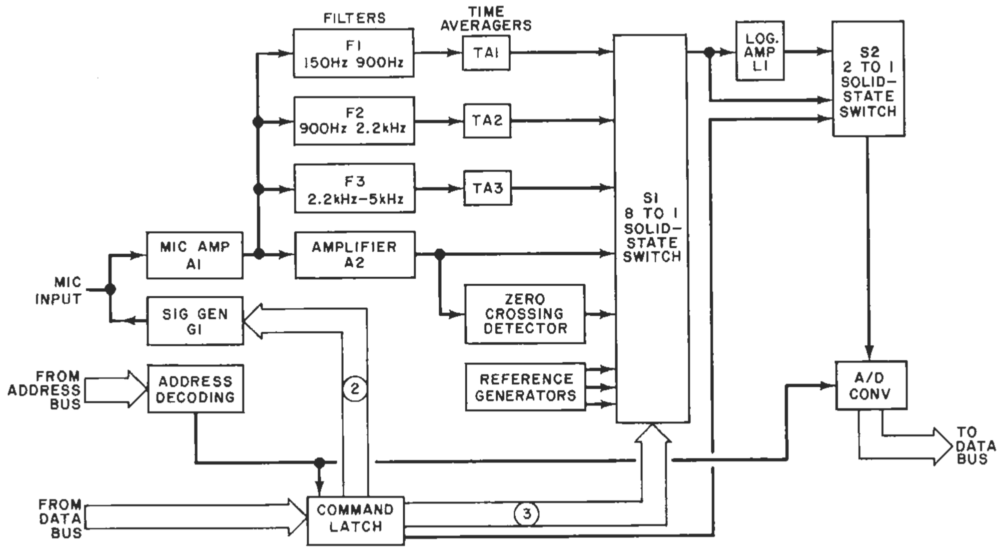


Fig. 1. The mic input is amplified, filtered and applied to S1 along with raw audio, zero-crossing detection, and three reference voltages. Output of S1 is computer selected by switch S2 for digitizing.

etc.) using a repeatable sound such as a grunt. One can use Speechlab, too, as a vocal processor to add spoken commands to many computer games (such as the "Star Trek" game), or enter the world of artificial intelligence and advanced programming.

Circuit Operation. The basic block diagram of Speechlab is shown in Fig. 1. The audio input is amplified by *A1* and applied to three 80-db/decade rolloff band-pass filters *F1, F2*, and *F3*. These filters encompass the ranges of 150 to 900 Hz, 900 Hz to 2.2 kHz, and 2.2 kHz to 5 kHz, respectively. These ranges correspond to the frequency ranges of the first three resonances of the average human vocal tract.

Each filter is passed to a time averager (*TA1, TA2,* and *TA3*) to generate a voltage proportional to the level of the speech waveform within each band.

The amplified audio signal from *A1* is further amplified by *A2* to generate an unfiltered waveform that can swing ±2 volts about a rest level of 2 volts. This signal is also applied to a zero-crossing detector that generates a voltage proportional to the number of times the speech waveform crosses the 2-volt rest level in a given period of time, thus generating a measure of the dominant frequency in the speech signal.

These five voltages—*TA1, TA2, TA3, A2*, and *ZCD*—are fed to solid-state switch *S1* along with three reference voltages used for calibration and self test. A computer output command selects one of these five voltages to be passed through *S1*.

The selected output from *S1* is passed to a second solid-state switch (*S2*), and to a logarithmic amplifier (*L1*) that emphasizes the low-level signal before being passed to *S2*. Switch *S2* can select either the direct output from *S1*, or the output from *L1*, and pass this selected signal to a 6-bit A/D converter where the voltage is converted to a digital value. The output of the A/D converter is fed to the computer data bus.

All operations of the Speechlab are controlled through a single I/O port (address AF_{hex}). As shown in Fig. 2, six bits are used: bit-5 disables the 8-to-1 multiplexer (*S1*), and is used when switching between bands; bit-4 controls signal generator *G1* which is used either to drive the microphone so that it acts like a miniature loudspeaker for prompting during voice input, or to drive the filters and zero-crossing detector during calibration and test; bit-3 selects either linear or logarithmic scaling of the voltage applied to the A/D converter; while bit-2, bit-1, and bit-0 select one of the eight signals from *S1* for A/D conversion.

The input data word contains the 6-bit A/D output in bits 0 through 5, bit-6 is unused and is always 0, while bit-7 is the A/D converter status with a 1 corresponding to busy, and 0 corresponding to finished.

Speechlab is physically configured to occupy one slot in the Altair bus, and the complete schematic is shown in Fig. 3 through Fig. 7.

Construction. The two foil patterns (Speechlab uses one double-sided pc board) are shown half-size in Fig. 8. (Blow up to full size on *film* only.) Component layout is shown in Fig. 9.

All the components are mounted on one side of the board, with all the soldering done on the noncomponent side. Sockets are recommended for all IC's since most of them are MOS-types that may be damaged by improper handling.

Integrated circuits *IC1, IC4, IC7, IC8, IC9, IC15,* and *IC16* should be selected so they are capable of delivering a 4-volt output when using a 5-volt supply. Dual flip-flop *IC14* can be from any manufacturer but Fairchild, as their truth table is somewhat different from the conventional table.

Start construction by installing the voltage regulator (*IC6*), all the discrete components, and the IC sockets—do not install the IC's at this time. Check the board for correct parts installation, and to make sure that there are no solder bridges between adjacent foil traces. Mount the board in an Altair bus connector, and check for the presence of 5 volts at the output of the voltage regulator and at the appropriate socket pins. Remove the board from the computer.

Install *IC2* through *IC5, IC10* through *IC14*, and *IC17* through *IC22*. Install the board back in the Altair bus connector, and turn on the computer. Load the test

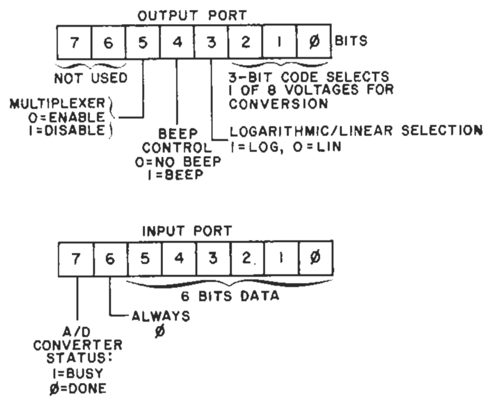


Fig. 2. Input and output port bit configuration.

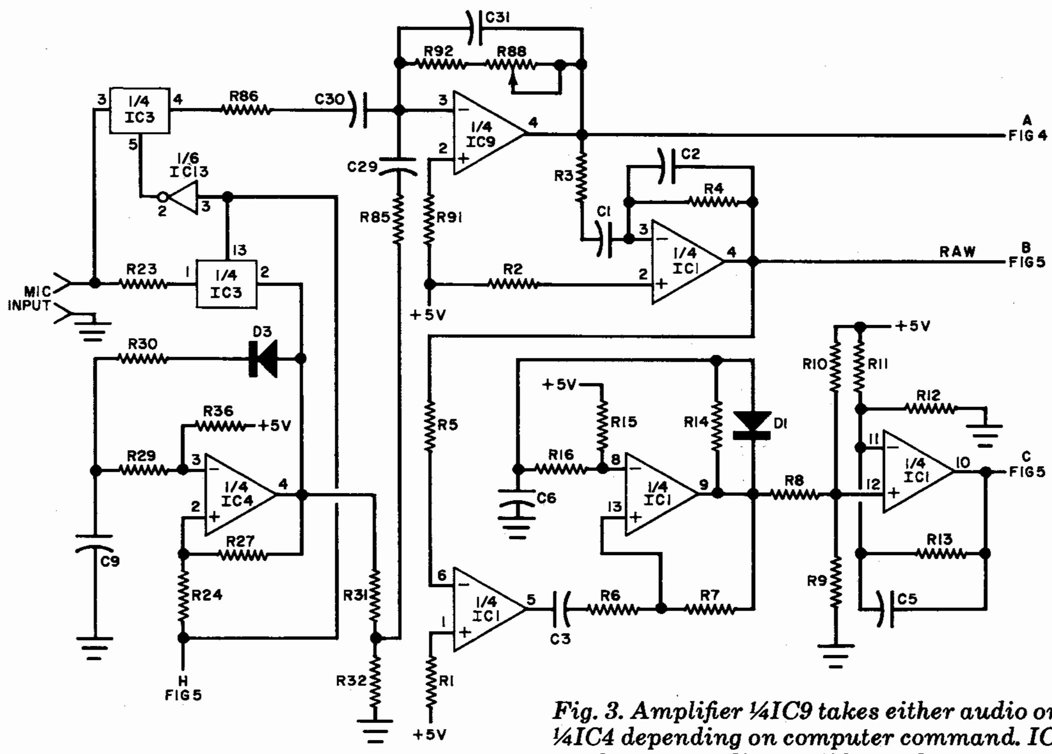


Fig. 3. Amplifier ¼IC9 takes either audio or tone from ¼IC4 depending on computer command. IC1 circuits are used as raw audio amplifier and zero-crossing detector.

PARTS LIST

Unless otherwise noted, the following capacitors are 10% Mylar types, and all picofarad sizes are CM05 types.
C1, C16, C21, C43, C47, C49, C52, C57—0.0047 µF
C2, C31—100 pF
C3, C17, C20—270 pF
C4, C7, C8, C10, C12, C19, C27, C32, C33, C34, C35, C36, C37, C44, C55, C61, C62—0.1 µF, 25-V disc
C5, C14, C18, C24, C54, C60—0.01 µF
C6, C42, C45, C53, C56—240 pF
C9, C40, C48—0.022 µF
C11, C29—47 pF
C13—15 µF, 25 V tantalum
C15, C22, C51, C59—0.0015 µF
C23—0.0022 µF
C25, C26, C28, C38—1 µF
C30, C39, C46—0.047 µF
C41—0.1 µF
C50, C58—0.001 µF
D1, D3 through D6—1N4148 or 1N914 diode
D2—1N746 diode
IC1, IC4, IC7, IC8, IC9, IC15, IC16—LM3900 quad amp
IC2—4051 8-to-1 analog multiplex
IC3—4016 quad analog switch
IC5—LM311 comparator
IC6—78M05 5-volt regulator
IC10—4024 7-stage binary counter
IC11, IC18—74C174 D flip-flop
IC12—4050 hex buffer
IC13, IC22—4049 hex buffer inverter
IC14—4013 (see text) dual-D flip-flop
IC17—74LS30 8-input NAND gate
IC19—8097 three-state hex buffer
IC20—8093 three-state quad buffer
IC21—4001 NOR gate
MIC—Mura DX-121 dynamic microphone (part of stereo set Mura DX-242)
L1—22-µH choke
Unless otherwise noted, the following resistors are ¼-W, 5%
R1—619,000 ohms, 1%
R2—1 megohm, 1%
R3—6810 ohms, 1%
R4—332,000 ohms, 1%
R5—200,000 ohms, 1%
R6,R20,R21—30,000 ohms
R7, R100—3 megohms
R8, R9, R10, R12, R14, R16, R104—1 megohm
R11—910,000 ohms
R13—2.7 megohms
R15, R48—10 megohms
R17,R18—20,000 ohms
R19, R22, R106—10,000 ohms
R23—1000 ohms
R24, R27—1.2 megohms
R25, R34, R39—470,000 ohms
R26, R38—750,000 ohms
R28, R31—100,000 ohms
R29—110,000 ohms
R30—39,000 ohms
R32—47,000 ohms
R33, R41—68,100 ohms, 1%
R35, R96,R102—75,000 ohms
R36—3.9 megohms
R37, R46—357,000 ohms, 1%
R40, R50, R52, R54, R56, R58, R60 R61—10,000 ohms, 1%
R42—12,100 ohms, 1%
R43, R49—4750 ohms, 1%
R44—4320 ohms, 1%
R45, R47—681,000 ohms, 1%
R51, R53, R55, R57, R59—4990 ohms, 1%
R62—274,000 ohms, 1%
R63—7500 ohms
R64, R66, R72, R75—160,000 ohms
R65, R71—12,000 ohms
R67, R70—300,000 ohms
R68—931,000 ohms, 1%
R69—2 megohms
R73—620,000 ohms
R74, R76, R90, R92—62,000 ohms
R77—15,000 ohms
R78, R83, R84—147,000 ohms, 1%
R79, R80, R87—51,100 ohms, 1%
R81,R82,R89—174,000 ohms, 1%
R85—330,000 ohms
R86—680 ohms
R88—100,000-ohm pc trimmer potentiometer
R91—270,000 ohms
R93—249,000 ohms, 1%
R94—4300 ohms
R95, R97, R103, R105—360,000 ohms
R98, R101—820,000 ohms
R99—845,000 ohms, 1%
R107—158,000 ohms, 1%
R108—4700 ohms
R109, R111, R117, R119—82,000 ohms
R110, R116—5100 ohms
R112, R115—180,000 ohms
R113—549,000 ohms, 1%
R114—1.6 megohms
R118—510,000 ohms
R120—6800 ohms
R121—2000 ohms
Misc.—Sockets (one 8-pin, thirteen 14-pin, seven 16-pin), regulator mounting hardware, tie-wrap etc.
Note 1: The following is available from Heuristics Inc., 900 N. San Antonio Rd. (Suite C-1), Los Altos CA 94022 (Tele: 415-948-2542): complete kit of all parts including pc board, sockets, microphone, hardware manual, and 200-page lab manual, SpeechBasic, and assembly language programs at $249. (California residents please add 6½% sales tax.)

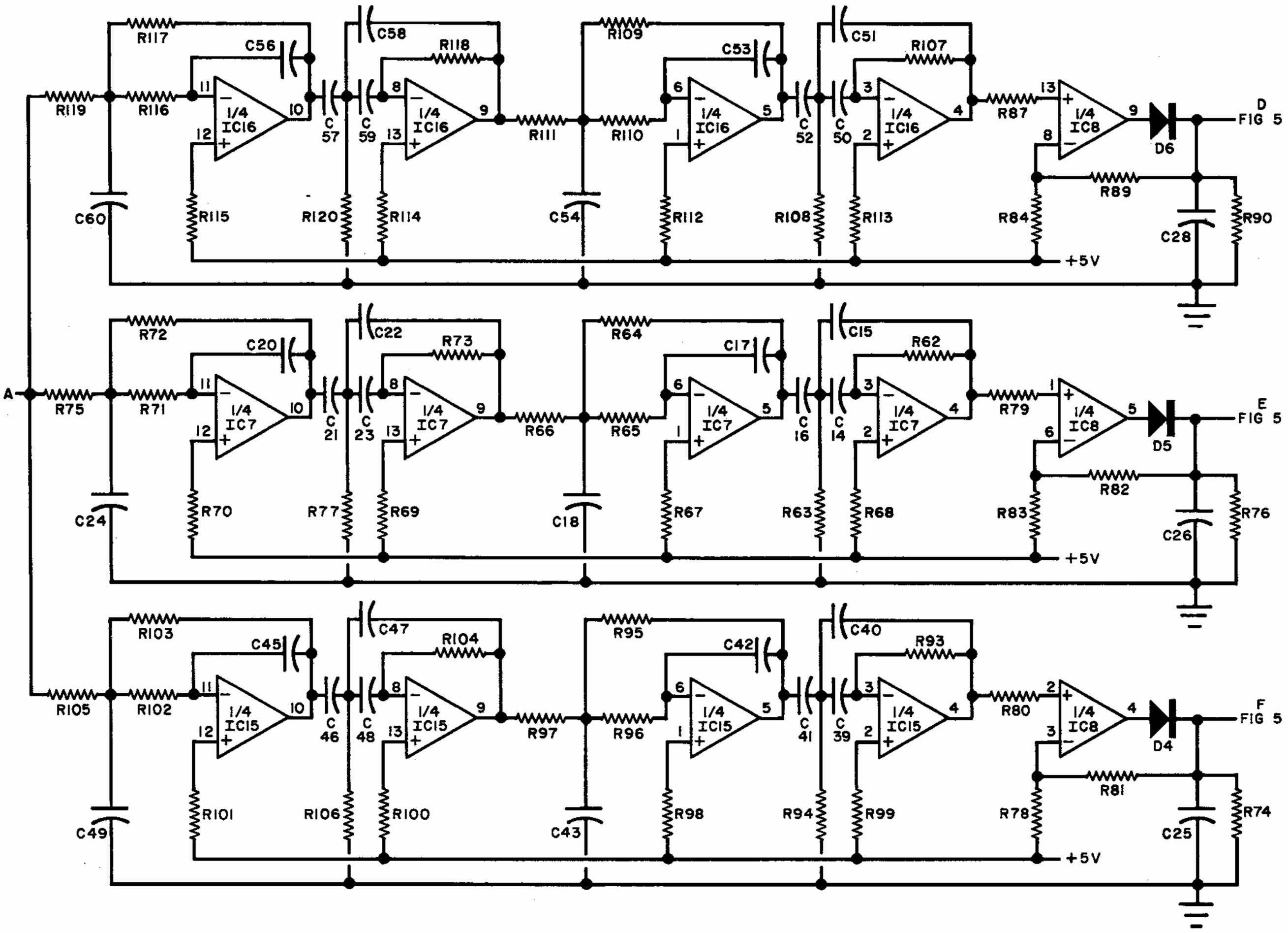


Fig. 4. Three bandpass filters and their associated time averagers. They encompass three ranges corresponding to frequency ranges of the first three resonances of an average human vocal tract.

program of Table I at 100 (hex). *NOTE: all program data in this article is in hex.*

You must jump to your monitor routine at address 0164-0165. Load address 195 with 05 and run the program. This will input the fixed reference voltage levels to the A/D converter and check the signal paths from switch *S1* to the computer data bus.

After running this program, examine locations 200 through 20F, 300 through 30F, and 400 through 40F. Location 200 through 20F should contain 12 ±4, 300 through 30F should contain 24 ±4, and 400 through 40F should contain 36 ±4.

Insert the remaining IC's in their sockets, load location 195 with 10, and run the test program (Table I). This test uses the signal generator (*G1*) to create an input for the filters, amplifiers, and zero-crossing detector, and thereby checks the remaining signal paths on the board and calibrates the microphone preamplifier. After running the program, examine locations 200 to 20F to see if it contains 16 to 18. If not, adjust potentiometer *R88* and rerun the program until these outputs occur.

Calibration and Test Program. The test program (Table I) is a general-purpose calibration, test, and diagnostic program for the Speechlab. It loads at location 100 and requires memory from 100 to 600 for program and data areas. Locations 163-165 should be loaded with a jump to your monitor address so that the program will return control to your monitor after execution. If you do not have a monitor, place a halt at this location.

The program collects four 256-byte buffers of data from four of the eight pos-

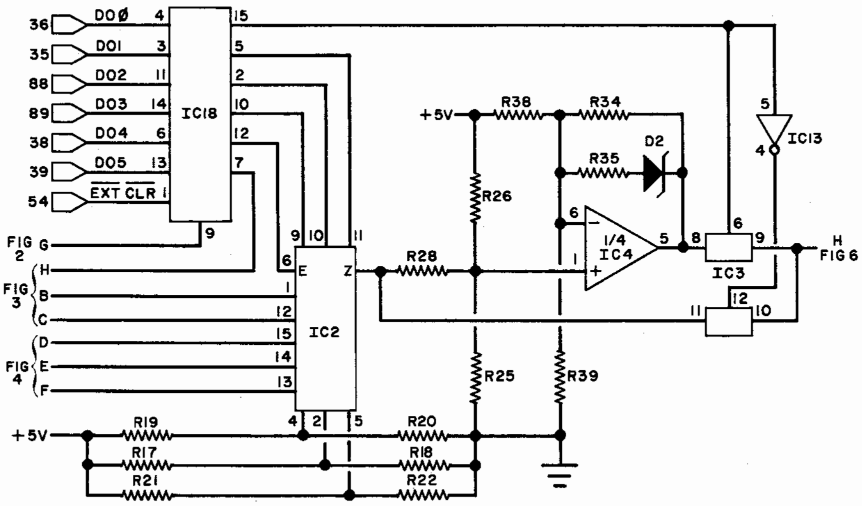


Fig. 5. Command latch (IC18) can activate tone generator and switch S1 (IC2). Op amp (¼IC4) is logarithmic amplifier.

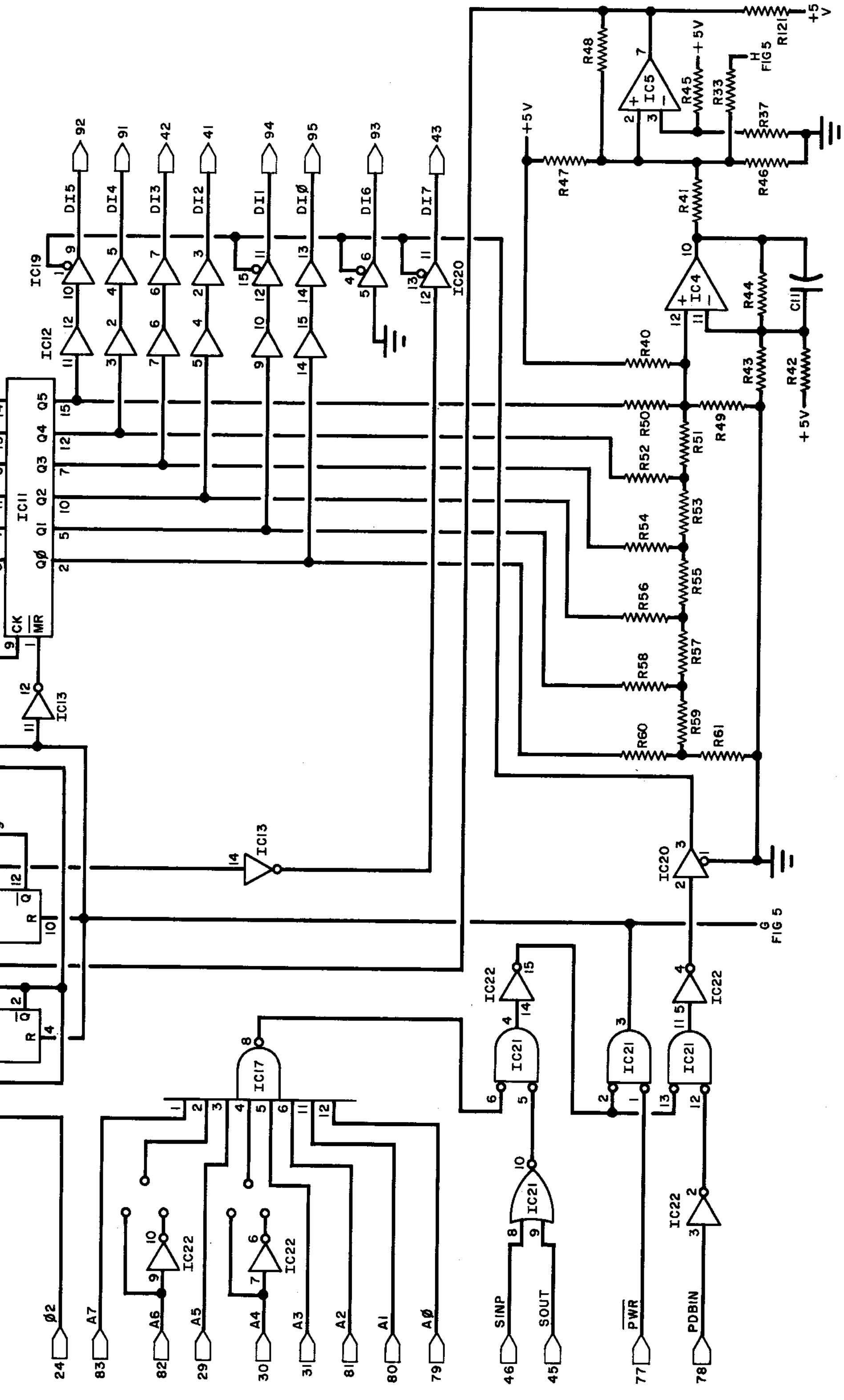


Fig. 6. IC17 circuit selects board address and IC14 forms S2. IC10 and IC11 form 6-bit A/D converter. Digitized data is then passed to computer.

sible inputs to the A/D converter. The first of the four bands is specified by the Test Command word, which also specifies beeper on/off and linear or logarithmic scaling. The next three bands are 1, 2, and 3 greater than specified by the Test Command word. Each band is sampled every five milliseconds until 256 samples have been collected from each of the four bands. Data from the first band is stored in 200 to 2FF, the second band from 300 to 3FF, the third from 400 to 4FF, and the fourth from 500 to 5FF.

For example, if the Test Command word is set to 00, after the test program is run, the four data areas will contain numbers representing the outputs of band-0 (low frequency), band-1 (mid frequency), band-2 (high frequency), and band-3 (zero-crossing detector). Anything that was spoken into the microphone while the program was running, is filtered, converted into a binary number, and stored in the data areas.

If the Test Command word is set to 05, the first three data areas will contain

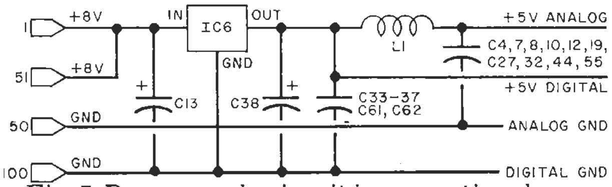


Fig. 7. Power supply circuit is conventional. Note bypass capacitors that are actually mounted between IC power supply pins and ground.

constant numbers corresponding to the three reference voltage levels to the A/D converter on band 5, 6, and 7. This is useful for checking the A/D converter operation and isolating problem areas to one side or the other of the 8-to-1 analog switch *S1*.

If the Test Command word is set to 10, signal generator *G1* is enabled which begins to "beep" the microphone and connects the signal-generator output into the microphone preamplifier *A1*. The four data areas contain data from bands 0, 1, 2, and 3 as when the Test Command word was 00, but the input signal comes from the signal generator rather than from the microphone. This allows calibration of the microphone preamplifier and isolates problems in one of the filter-averager chains.

Adding bit-3 to the command word will cause logarithmic rather than linear data scaling and will isolate problems to the log amplifier or either of the two analog switches comprising *S2*, the 2-to-1 analog switch.

Various combinations of bits in the Test Command word will allow quick calibration and fault isolation, and also provide a quick way to look at raw data from any input through the microphone.

Software. A simple technique for speech recognition of the digits zero through nine with a recognition rate of 90% or better, is shown in the flowchart of Fig. 10. An 8080 program for this algorithm is shown in Table II. The program starts at memory location 0100 and requires less than 4K bytes of storage including table space.

There are two modes of operation—training and performance. During training, speech examples of the digits are read into the microphone and the parameters of the speech input are extracted and placed in the tables. In the performance mode, an unknown utterance is presented and recognized.

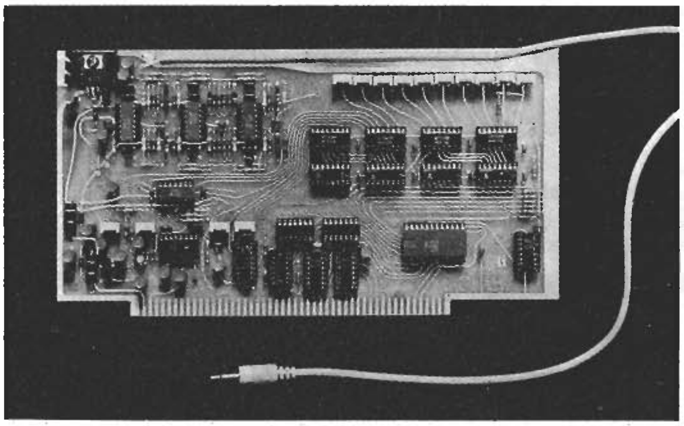
To use the program, enter it into the computer starting at location 0100, and then run the program. The Teletype will respond with "T" (train) or "P" (perform). Type a "T" and the Teletype will respond with "NUMBER?" which can be between 0 and F. Type the digit you desire, and the microphone will emit a "beep" indicating that the speech window is open. When this beep occurs, vocalize the same digit you just typed in. The microphone will beep again to indicate that the speech window is now closed. The machine will then type T or

TABLE I

```
0100                    ORG 100H
0100 =                  START EQU 100H
0100 210002             LXI H,START+100H
0103 228D01             SHLD TEMP1
0106 210003             LXI H,START+200H
0109 228F01             SHLD TEMP2
010C 210004             LXI H,START+300H
010F 229101             SHLD TEMP3
0112 210005             LXI H,START+400H
0115 229301             SHLD TEMP4
0118 3A9501     GO      LDA COMAND
011B CD6601             CALL INPUT
011E 2A8D01             LHLD TEMP1
0121 77                 MOV M,A
0122 2C                 INR L
0123 228D01             SHLD TEMP1
0126 3A9501             LDA COMAND
0129 C601               ADI 1
012B CD6601             CALL INPUT
012E 2A8F01             LHLD TEMP2
0131 77                 MOV M,A
0132 2C                 INR L
0133 228F01             SHLD TEMP2
0136 3A9501             LDA COMAND
0139 C602               ADI 2
013B CD6601             CALL INPUT
013E 2A9101             LHLD TEMP3
0141 77                 MOV M,A
0142 2C                 INR L
0143 229101             SHLD TEMP3
0146 3A9501             LDA COMAND
0149 C603               ADI 3
014B CD6601             CALL INPUT
014E 2A9301             LHLD TEMP4
0151 77                 MOV M,A
0152 2C                 INR L
0153 229301             SHLD TEMP4
0156 CA5F01             JZ STOP
0159 CD7701             CALL DELAY
015C C31801             JMP GO
015F 3E00       STOP    MVI A,0
0161 D3AF               OUT 0AFH
0163 C3XXXX             JMP SYSTEM
0166 F620       INPUT   ORI 20H
0168 D3AF               OUT 0AFH
016A E6DF               ANI 0DFH
016C D3AF               OUT 0AFH
016E DBAF       LOOP    IN 0AFH
0170 17                 RAL
0171 DA6E01             JC LOOP
0174 DBAF               IN 0AFH
0176 C9                 RET
0177 C5         DELAY   PUSH B
0178 3E05               MVI A,5
017A FE00       DEL0    CPI 0
017C CA8B01             JZ RETDEL
017F 0669               MVI B,69H
0181 00         DEL1    NOP
0182 00                 NOP
0183 05                 DCR B
0184 C28101             JNZ DEL1
0187 3D                 DCR A
0188 C37A01             JMP DEL0
018B C1         RETDEL  POP B
018C C9                 RET
C000 =          SYSTEM EQU 0C000H
018D            TEMP1 DS 2
018F            TEMP2 DS 2
0191            TEMP3 DS 2
0193            TEMP4 DS 2
0195 YY         COMAND EQU $
0100 21 00 02 22 8D 01 21 00 03 22 8F 01 21 00 04 22
0110 91 01 21 00 05 22 93 01 3A 95 01 CD 66 01 2A 8D
0120 01 77 2C 22 8D 01 3A 95 01 C6 01 CD 66 01 2A 8F
0130 01 77 2C 22 8F 01 3A 95 01 C6 02 CD 66 01 2A 91
0140 01 77 2C 22 91 01 3A 95 01 C6 03 CD 66 01 2A 93
0150 01 77 2C 22 93 01 CA 5F 01 CD 77 01 C3 18 01 3E
0160 00 D3 AF C3 53 00 F6 20 D3 AF E6 DF D3 AF DB AF
0170 17 DA 6E 01 DB AF C9 C5 3E 05 FE 00 CA 8B 01 06
0180 69 00 00 05 C2 81 01 3D C3 7A 01 C1 C9 9B F7 F0
0190 FE 57 DC 49 E2 A6
?
```

"HANDS ON" EXPERIENCE WITH A *TALKING* COMPUTER

BY LESLIE SOLOMON, Technical Editor



While testing the Speechlab, we borrowed an Al Cybernetic Systems (Box 4691, University Park, NM 88003) Model-1000 Speech Synthesizer ($325, assembled) to see if our microcomputer could "talk" as well as "hear." The Model 1000 is designed to fit into one slot of an Altair bus and delivers its output via an audio cable that can be plugged into any audio amplifier system. The output level is 0.6 volt p-p; impedance is 1000 ohms; and frequency range is 150 to 4500 Hz.

This synthesizer is phoneme-oriented. Accordingly, you can program it to say anything, as opposed to speech synthesizers that have only several words fixed in ROM. Essentially, the Model 1000 is a hardwired analog of the human vocal tract and various portions of the circuit emulate the vocal cords, the lungs, and the variable-frequency resonant acoustic cavity of the mouth, tongue, lips and teeth.

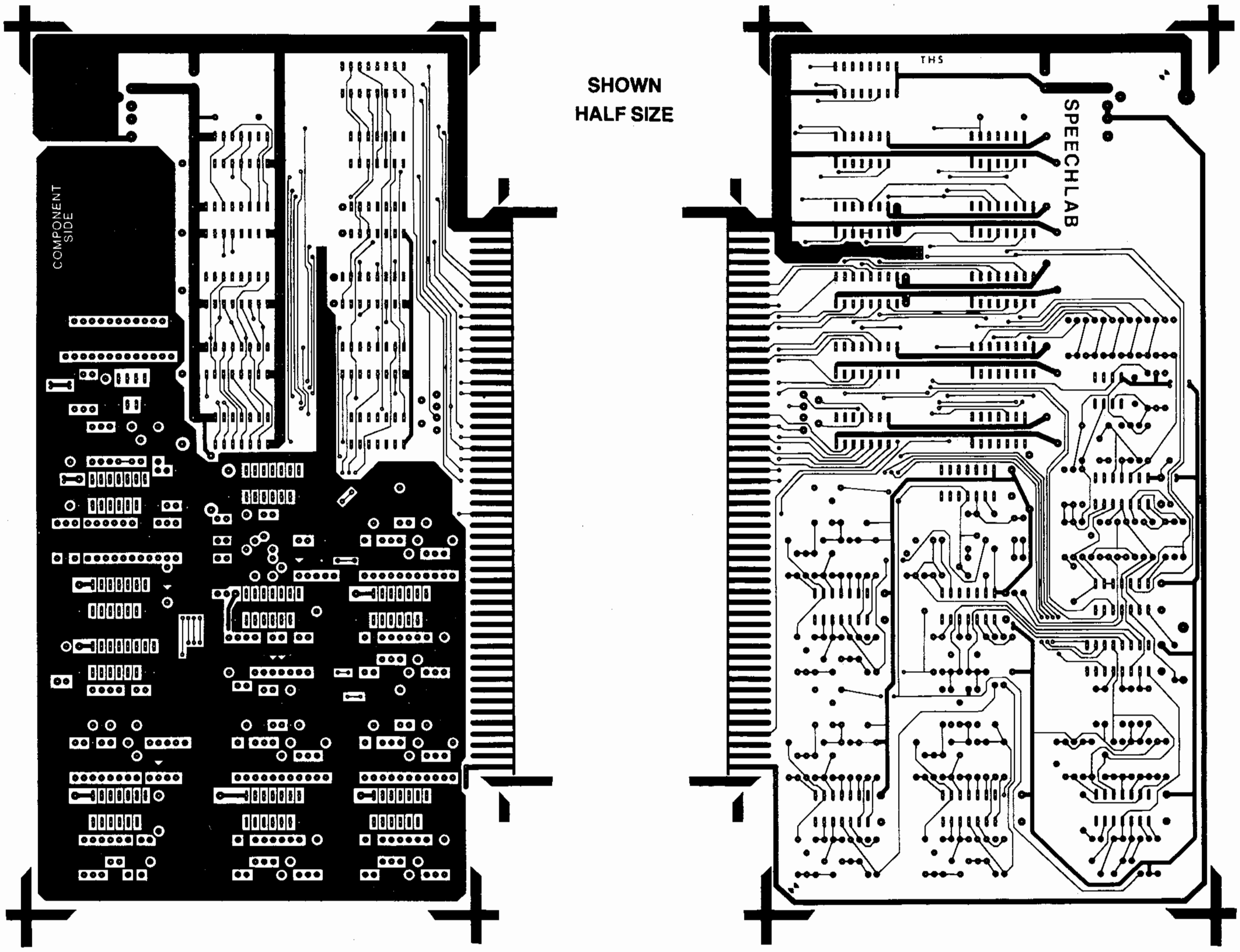


Fig. 8. Etching and drilling guides for pc board are shown half size. Guide at left is the component side. Component layout is in Fig. 9.

P again. You answer with a T, and the process is continued as long as you want. Do not exceed 16 entries with this sample program.

Once you have some vocalized digits in memory, run the program again. This time, when the Teletype asks T or P, answer with a P (for perform). Now, as you *speak* the digits into the microphone, the Teletype will respond by typing that digit. When used in a quiet room, with the same vocalization, this algorithm can be

All the information necessary to perform the synthesis functions are located within a ROM that is accessed by the program. Words and sentences are formed by supplying a string of ASCII characters as would be done when outputting to any port, except that these strings also use some non-alphanumeric characters (i.e., the "+" is used to form "th" as in "thaw" or "earth"). Each ASCII character represents a particular phonetic sound or phoneme. If desired, you can create a program that produces simultaneous printout and "voiceout" of the same string.

The device requires very little software to implement: less than 50 bytes of assembly language or a handful of BASIC statements. The manual accompanying the synthesizer covers speech generation in detail, how it is created, and what is involved. It also illustrates how to "mechanize" speech, with several examples shown.

After working with the synthesizer for a couple of weeks, we found that we have a lot to learn about how humans create speech. After many hours of studying, experimenting, and redoing programs, we made the Model-1000 utter some recognizable sentences. It is not easy, our experience showed, even when one uses the wealth of instructions provided.

Working with a phoneme-oriented speech synthesizer is a little like learning to use a microprocessor. All the logic is there, but programming it properly is another story. Like working with a processor for the first time, one must crawl frustratingly before walking. Slowly, however, the ideas start to percolate. Our computer still talks with a rather heavy "robotic" accent, but we have hopes that someday it will "humanize."

To paraphrase Sam Johnson: "Sir, a computer talking is like a dog walking on its hind legs. It is not done well; but you are surprised to find it done at all." We have a long road ahead to the "HAL-9000," but the first step has been taken.

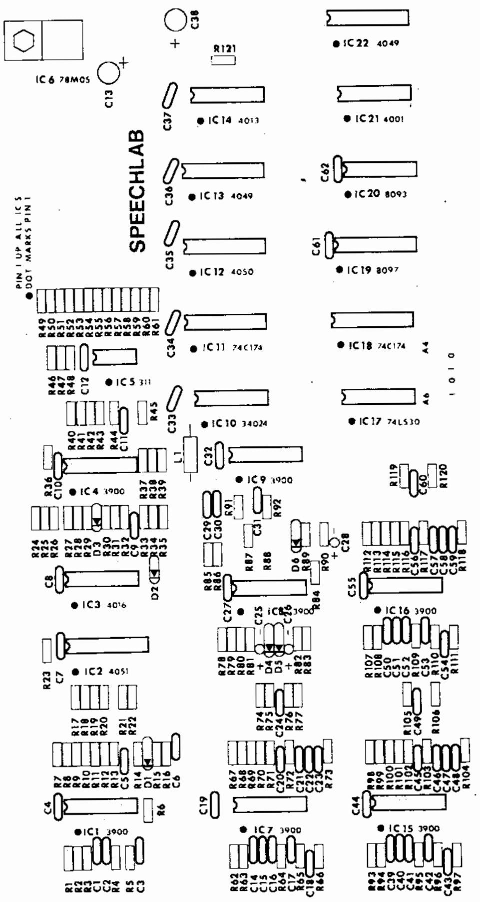


Fig. 9. Component layout for the Speechlab. See etching and drilling guide on previous page.

expected to have a recognition rate greater than 90%.

The program works as follows: the sampling subroutine is entered to obtain a sample of the amplitude every 10 milliseconds in each of the three frequency bands and to estimate the number of zero crossings during each time period. One hundred and fifty samples are collected, allowing up to 1.5 seconds of speech (between microphone "beeps"). A preset threshold is used to find the beginning and end of the word. The duration of the word can now be computed by a simple subtraction. Typically, this duration will be about 400-milliseconds for the digits. The duration time is divided by 16 to select 16 evenly spaced parameters from the three bands and zero-crossing information.

The 64 bytes obtained (16 parameters from each of the four bands) are compared with similar parameters which were collected during the training mode. A summation (running total) of the difference between the 64 parameters of the sample and the parameters of the training "templates" is computed. The totals represent a measure of the difference between the sample and each of the previously stored templates. The template with the smallest difference from the sample is then selected as the answer (output).

The above algorithm, while relatively simple, illustrates many of the basic concepts of speech recognition. A manual supplied with the Speechlab kit contains descriptions of other approaches to speech recognition, along with sample programs to demonstrate the techniques of speech recognition. ◇

TABLE II

0100	31	00	10	CD	5B	03	21	35	0B	CD	3A	03	CD	8F	03	FE
0110	54	C2	1A	01	CD	33	01	C3	00	01	FE	50	C2	2B	01	CD
0120	5B	03	CD	79	01	CD	81	03	C3	00	01	3E	3F	CD	70	03
0130	C3	00	01	F5	C5	D5	E5	CD	5B	03	21	3D	0B	CD	3A	03
0140	CD	DA	03	32	6E	04	3C	47	3A	6D	04	B8	DA	2B	01	CD
0150	5B	03	CD	A3	02	CD	99	01	CD	07	04	3A	6E	04	4F	16
0160	40	CD	F7	02	11	13	07	CD	A3	03	60	69	0E	40	11	C8
0170	06	CD	8C	01	E1	D1	C1	F1	C9	C5	D5	E5	CD	A3	02	CD
0180	99	01	CD	07	04	CD	3D	02	E1	D1	C1	C9	F5	EB	7E	23
0190	EB	77	23	0D	C2	8D	01	F1	C9	0E	00	21	6F	04	CD	F9
01A0	03	FE	06	D2	B2	01	79	FE	96	C2	9E	01	21	44	0B	C3
01B0	9D	02	79	3D	32	08	07	47	CD	F9	03	FE	06	DA	CC	01
01C0	79	FE	96	C2	B8	01	21	4E	0B	C3	9D	02	79	3D	3D	32
01D0	09	07	90	3C	32	C7	06	FE	0A	DA	A6	01	0E	01	CD	F9
01E0	03	FE	06	D2	ED	01	79	FE	0A	C2	DE	01	C9	3A	C7	06
01F0	81	4F	3A	08	07	81	4F	C3	B8	01	21	FF	FF	22	13	0B
0200	3E	00	32	6E	04	3A	6D	04	3D	4F	06	00	C5	79	87	4F
0210	21	15	0B	09	46	23	4E	21	13	0B	56	23	5E	CD	B2	03
0220	DA	2B	02	C1	0D	FA	39	02	C3	0C	02	21	13	0B	70	23
0230	71	C1	79	32	6E	04	C3	24	02	3A	6E	04	C9	CD	64	02
0240	06	00	48	16	40	3E	00	5F	21	13	07	CD	D5	03	CD	75
0250	02	1C	7B	FE	3F	C2	48	02	04	3A	6D	04	B8	C2	42	02
0260	CD	FA	01	C9	3A	6D	04	87	4F	3E	00	21	15	0B	77	23
0270	0D	C2	6E	02	C9	E5	D5	C5	F5	21	C8	06	16	00	19	96
0280	F2	85	02	2F	3C	57	78	87	21	15	0B	4F	06	00	09	23
0290	7A	86	77	D2	98	02	2B	34	F1	C1	D1	E1	C9	CD	3A	03
02A0	C3	00	01	3E	FA	CD	E3	02	CD	4B	03	3E	64	CD	E3	02
02B0	21	6F	04	1E	96	16	00	7A	FE	04	CA	C6	02	CD	D3	02
02C0	77	23	14	C3	B7	02	3E	0A	CD	E3	02	1D	C2	B5	02	CD
02D0	4B	03	C9	F6	20	D3	AF	E6	DF	D3	AF	DB	AF	17	DA	DB
02E0	02	1F	C9	C5	FE	00	CA	F5	02	06	69	00	00	05	C2	EB
02F0	02	3D	C3	E4	02	C1	C9	F5	D5	06	00	1E	09	79	1F	4F
0300	1D	CA	0E	03	78	D2	09	03	82	1F	47	C3	FD	02	D1	F1
0310	C9	F5	D5	1E	09	78	47	79	17	4F	1D	CA	2F	03	78	17
0320	D2	27	03	92	C3	16	03	92	D2	16	03	82	C3	16	03	17
0330	5F	3E	FF	A9	4F	7B	1F	D1	F1	C9	7E	FE	00	CA	47	03
0340	CD	70	03	23	C3	3A	03	CD	5B	03	C9	F5	3E	10	D3	AF
0350	3E	64	CD	E3	02	3E	00	D3	AF	F1	C9	F5	3E	0D	CD	70
0360	03	3E	0A	CD	70	03	3E	00	CD	70	03	CD	70	03	F1	C9
0370	F5	DB	00	07	D2	71	03	F1	D3	01	C9	00	00	00	00	00
0380	00	E6	0F	C6	30	FE	3A	DA	70	03	C6	07	C3	70	03	DB
0390	00	E6	40	CA	8F	03	DB	01	E6	7F	C3	70	03	00	00	00
03A0	00	00	00	EB	09	EB	42	4B	C9	F5	79	93	4F	78	9A	47
03B0	F1	C9	32	0A	07	78	BA	CA	BE	03	3A	0A	07	C9	79	BB
03C0	3A	0A	07	C9	F5	C5	D5	CD	F7	02	09	85	6F	D2	D1	03
03D0	24	D1	C1	F1	C9	CD	C4	03	7E	C9	CD	8F	03	E6	7F	FE
03E0	30	DA	2B	01	FE	3A	DA	F6	03	FE	41	DA	2B	01	FE	47
03F0	3F	DA	2B	01	D6	07	D6	30	C9	AF	C5	06	03	86	23	05
0400	C2	FD	03	C1	23	0C	C9	21	C8	06	22	0B	07	3A	C7	06
0410	4F	16	0A	CD	F7	02	16	10	CD	11	03	26	00	69	22	11
0420	07	21	00	00	22	0F	07	1E	10	D5	2A	0F	07	44	4D	16
0430	0A	CD	11	03	06	00	3A	08	07	5F	16	00	CD	A3	03	16
0440	04	CD	F7	02	21	6F	04	09	54	5D	2A	0B	07	0E	04	CD
0450	8C	01	22	0B	07	2A	0F	07	44	4D	2A	11	07	54	5D	CD
0460	A3	03	60	69	22	0F	07	D1	1D	C2	29	04	C9	10	CD	EF
0B30	DA	0C	FE	02	C2	54	20	4F	52	20	50	3F	00	4E	55	4D
0B40	42	45	52	00	4E	4F	20	53	50	45	45	43	48	00	4F	55
0B50	54	20	4F	46	20	57	49	4E	44	4F	57	00	00	32	5C	00

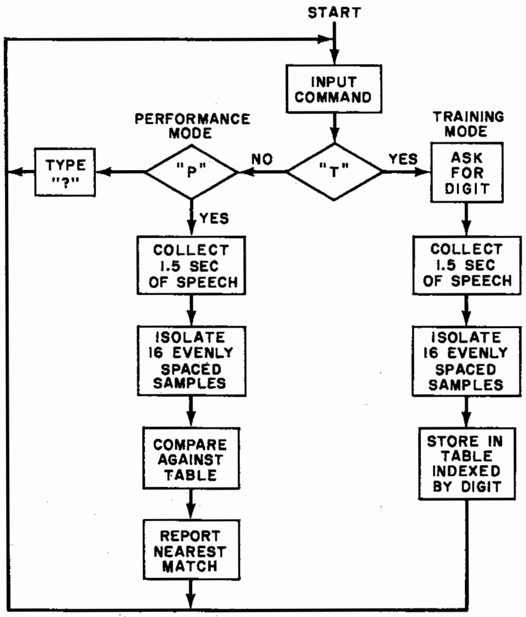


Fig. 10. Flow chart of a simple program that is used to "T" (train) and "P" (perform) a vocal operation. The program is shown in Table II.