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Dec. 1984

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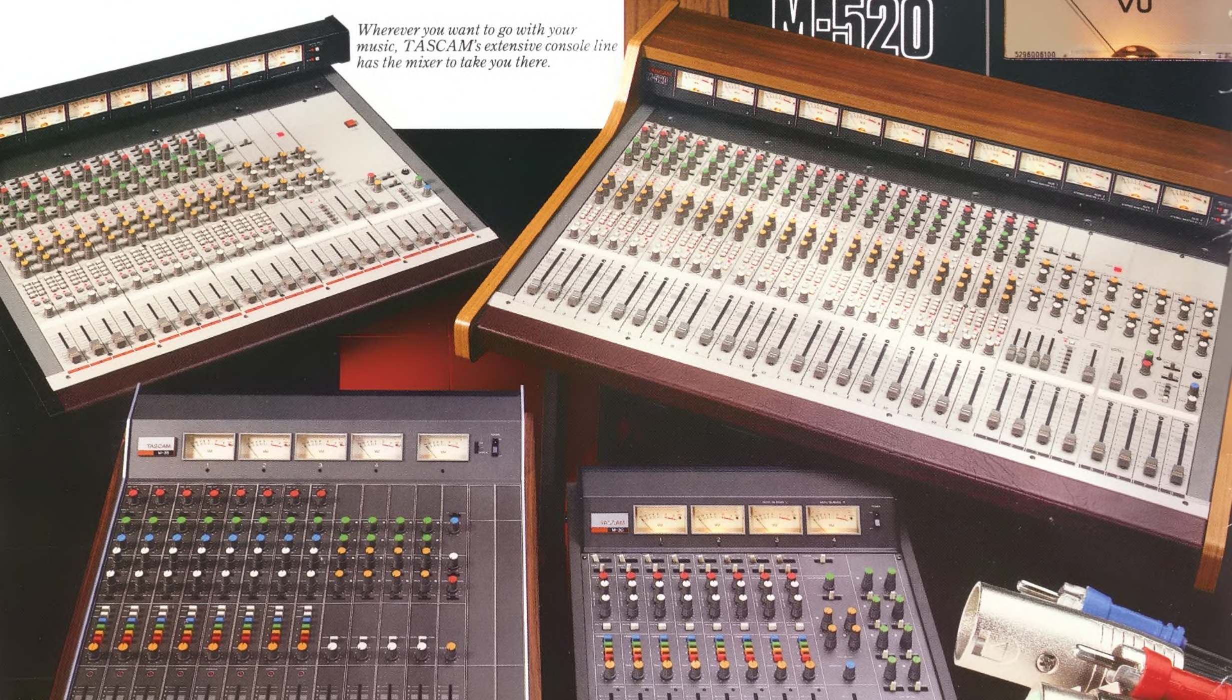

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Letters

INTERFACE BLUES — THE SEQUEL

In regard to the letter from C. L. Jaszberenyi in the August 1984 issue of Polyphony ("Interface Blues, Part 2"), please note that a series of articles detailing interfacing procedures for the Timex/Sinclair 1000 appeared in the July, August, and September 1984 issues of Radio-Electronics magazine. I have built an output-port decoding circuit based upon these articles which I use to drive a D/A converter for control voltages, and to provide trigger pulses for an envelope generator. There is probably some fairly simple way to adapt the circuit to the devices he wishes to control. Additional detailed hardware and programming ideas are also contained in the book Control Things with Your Timex Sinclair by Robert L. Swarts (Dilithium Press, 1984).

David Snow
Gaithersburg, MD

GR/XPANDER INTERFACE?

What are the possibilities of an interface project between the Roland GR series guitar synthesizers and something like the Oberheim Xpander? The Xpander, with its six voice configuration, seems ideal for this use. Is Oberheim planning something like this? (I wrote them to ask, mentioning that I saw their ads in Polyphony, but I thought you might have the inside scoop.) Thanks for helping, I think you've got a great magazine.

Dave Ogden
Newton, MA

Dave -- If you have a GR-700 synthesizer, of course, you can drive the Xpander directly via the MIDI port. As a matter of fact, the people at Oberheim are well aware that the Xpander is an excellent guitar synth add-on; however, I know of no plans by the company to come up with a MIDI guitar controller at present. One other controller, the SynthAxe (distributed in the USA by Fairlight) also produces a MIDI output and has been demoed with the Xpander. Unfortunately, these are pricey solutions. However, trying to make a custom interface to derive a control voltage from a GR-series guitar, then turning that CV into MIDI data, might end up costing you more in time and aggravation than just going for something like the GR-700 synthesizer.

For more information on MIDI and guitarists, see my column in the January and February issues of Guitar Player magazine. Also, one of our authors is working on a "hands-on" oriented GR-700 review, so stay tuned for further developments.

ECHOTRON SYNC

Is it possible to trigger the DeltaLab "Echotron" from something like a DMX, LinnDrum, keyboard trigger, etc.? Any help would be appreciated. Also, are there any other comparable units that would do this besides the E/H Instant Replay?

Nicky Moroch Jr.
Yonkers, NY

Nicky -- I've often wanted to make the same modification to my Echotron, but have not been able to pursue this due to lack of schematic and time. Perhaps a reader can help. Regarding comparable units, the Decillionix DX-1 (also see Current Events for a related story) uses an Apple II for sampling, and the recently introduced CompuEffectron not only allows for triggered sampling, but also lets you edit the sample -- something that greatly enhances the unit's musical usefulness.

Robert Carlberg's re-view

Wendy Carlos **Digital Moonscapes** (CBS 39340). Contemporary classical music which just happens to be realized on a GDS digital synthesizer. Carlos has always said she's a composer first, although her output of original works has been lamentably scarce. Here at last she gives us two long suites of fully-orchestrated music which, in style, fall somewhere between Sibelius and Lopatnikoff; i.e. contemporary but with strong Romantic leanings. There are echoes here and there of "Timesteps," her only previous large scale work, but neither of these suites develops the momentum of that piece. Still, that may be considered an advantage if you wish, as the ebb and flow here causes the listener to follow the music as it unfolds rather than rushing ahead to an anticipated conclusion.

The tone colors Carlos chooses for her "synthetic orchestra" sound on the whole more synthetic than orchestral, but remarkably full and satisfying. They fall into three categories: a very imitative string section, some decidedly non-organic bonging, and many tones reminiscent of a small pipe organ, particularly the reedy "Vox Humana" stop. Bell sounds (at which digital synthesizers seem particularly adept) also pop up -- although one brief section featuring marimba-like tones points up the generally non-percussive nature of these works. In "Timesteps" the percussion definitely powered the aggressive single-mindedness of the piece; here Carlos composes in a more all-encompassing, feminine (if you will) mode. Whether this is a reflection of her increasing piece of mind one can only speculate.

This could be considered the first large-scale "serious" work for synthesizer. You will find yourself evaluating this disc on the basis of the music in it rather than the synthesizer "patches" on it, which might be some sort of first and is obviously an enormous step forward.

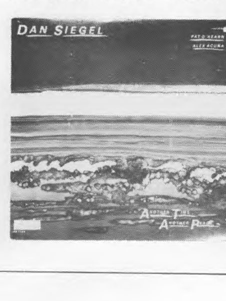
Erik Von Heiden **PKS2000-330** (cassette). Slow chords on string synth. One can envision a whole band mixed in with these tracks because, by themselves, they don't quite hold center stage. $7.50 from Scientific Creative Intelligence, 5351 S. Rosemead, Pico Rivera CA 90660.

Brainfood **Great Society** (cassette). On the other coast is Brainfood, an 8-track homebrew that sounds like the latest top-40 album. Professional production, tasteful synth, Van Halen-inspired guitar and vocals reminiscent of Andy Latimer (Camel) make this trio one of the most polished independents I've ever heard. Peter Lovi, P.O. Box 6564, Ithaca NY 14850.

Harold Budd **Abandoned Cities** (Cantil 384). Two long drones, moody and dark as you please, done on synthesizers with occasional interjections by harp and guitar. Very funereal.

Harold Budd/Brian Eno **The Pearl** (Editions EG 37). Drifting acoustic pianos, awash in echo, aided by subtle treatments. Ten years ago it was a great concept.

Dan Siegel **Another Time, Another Place** (PaUSA 7164). There's a small movement, populated by Mark Isham, Group 87, Steps ahead and now Siegel, toward instrumental rock/jazz featuring digital synthesizers. It's probably the offspring of the trailblazing Weather Report (whose former percussionist Alex Acuna appears here), and let's not quibble, it's great.



Rickie Lee Jones **The Magazine** (Warner Bros. 25117-1). An oblique and personal statement that gives off sparks but is too modest (or perhaps too guarded) to come out and yell at you. As always, Ms. Jones makes effective use of electronic keyboards.

Eberhard Schoener **Sky Music/Mountain Music** (Kuckuck 071). Fairlight, Oberheim and Roland synthesizers on a pair of 20+ minute mood doodles. Another "Celestial Harmonies" meditation background.

Anthony Davis **Middle Passage** (Gramavision 8401). Three solo pieces of experimental piano music and one with Serge accompaniment. All are frenetic and desultory.

Dave Stewart & Barbara Gaskin **I'm In A Different World** (Broken Records 7) (EP). Ex-art rocker Stewart turns out another (see also Oct. '82) fairly hard-rocking set in which he squeezes surprising sounds from the usually-docile Prophet. A whole album would be nice, Dave -- quit teasing us.

DEVO **Shout** (Warner Bros. 25097-1). DEVO's masterpieces sound just like the rest of their songs. They've MIDIfied and Fairlighted their set-up, resulting in some new digitones, but the main innovations here are homages to Hendrix and The Beatles.

(continued on page 29)

USING THE MODEL 64 MIDI SEQUENCER

by: Christopher Simmons

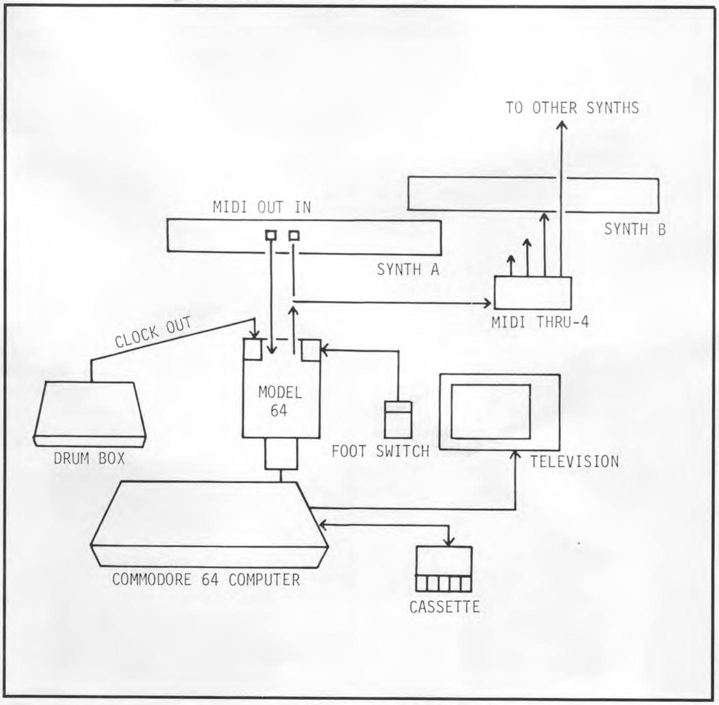
After using the Sequential Circuits Model 64 Sequencer for nearly six weeks, I am both elated and disappointed by its capabilities.

Basics. The Model 64 device is a compact (3-1/2" x 7-1/8" x 2" thick) interface that plugs into one of two "expansion ports" on the rear of a Commodore 64 home computer. It's like a video game cartridge; all of the software is "built-in", so it's ready to go on power-up. You don't **need** to load software to use the Model 64, but there is a provision to load expansion software. Therefore, it can be used for many specialized purposes that a dedicated sequencer box could not. Conceivably, with the proper information from the manufacturer, you could even write your own software for the device.

Because of its built-in LED indicators, a television/monitor is optional for basic sequence recording, playback, and overdubbing but is a necessity for song construction, editing functions, and song loading from disk. In the studio or on stage, a small black-and-white mini-television with a 3- or 5-inch screen would be ideal (even the newer Sony Watchmans with a video-in jack would be a possibility). However, the Model 64, like many plug-in boards for the Commodore, does seem to broadcast its share of RF interference which may or may not impair what you see on the TV screen. Wrapping aluminum foil around the plastic case seems to help. The Model 64 has one MIDI "in" and one MIDI "out" jack, a start/stop footswitch input (this controls the sequencer in exactly the same way as the SCI Prophet 600's sequencer), and one Clock In jack for use with virtually any drum machine. Note that the footswitch is not included, but is required. If you have an SCI Prophet 600 (as I do) or a Prophet T-8, you could use the footswitch that comes with your synthesizer.

Creating a sequence. The initial memory capacity is about 4870 notes. To record a sequence, you need only press one of the four function keys at the right side of the Commodore computer, then start up your drum machine (if using one), then play your sequence. The Model 64 holds only one song in memory at a time; this can include up to eight sequences, with up to six tracks per sequence.

Song Mode lets you create a song by stringing together sequences created in Sequence Mode. For a simple example, you might record a simple bass-line as Sequence One, then switch to Song Mode. Here you would build your song by "linking" together, say, eight blocks of Sequence One. While in Song Mode, you can also go into a sub-mode to edit; one use would be to transpose Sequence One (which, incidentally, affects all tracks recorded in Sequence One) within the song. For instance, you might choose to shift the third, sixth, and eighth blocks that you've linked together up a major fifth.



To make things more interesting, you could go back to Sequence Mode and "overdub" another track on Sequence One. Here you would select Sequence One, start your drum machine, and play an accompanying sequence (perhaps in the midrange of the keyboard) that would complement the previously recorded bass-line. Note that this overdub has not used up the second of your eight sequences; it has simply added the second of up to six tracks in Sequence One, right beside Track One on which you recorded the bass part. Each "track" can be edited or erased separately, or even muted during playback (Ed. note: selective muting lets you do such tricks as recording multiple solos, then selecting a different solo each night in order to prevent "computer boredom" from setting in). Now when you re-enter Song Mode and play back the song you've already constructed, you have two separate sequences playing and both will transpose as specified earlier, regardless of what key they were in originally. Some very pleasing counterpoints can be woven this way.

So far, we've haven't done anything too astonishing -- most sequencers can provide these kinds of functions. But since we've entered the MIDI age, the Model 64 can do quite a bit more than you might expect.

MIDI tricks. We now have two tracks, One and Two, both recorded in Sequence One. If you have two synthesizers, or one of those slick new keyboardless "expander" modules coming out from just about every manufacturer, we can "assign" each of these two voices to the different synths. Since I do have two keyboards -- and fortunately, each keyboard can be set to receive on one specific channel -- Track One could go to the Korg Poly 800 and Track Two to the Prophet 600. (Incidentally, older Prophet 600s can be updated by an authorized service center for MIDI channel selection capabilities.) Because updated software is just being made available for the Model 64 as this is being written, I won't go into specifics since whatever I say would be subject to change. However, here is one possibility: Alternating channel specification from block to block within a song would allow for some pretty complex orchestration, as Track One could play on one synth, Track Two on another, while a third track could appear on both; then Track Three could appear on one synth with the other two tracks on the other synth, and so on.

To use more than one keyboard with the Model 64, you need a "MIDI thru" box (Roland makes a "thru-4" box that serves this purpose quite well) to provide multiple outputs, since the Model 64 has only one MIDI out. (Ed. note: If you have an SCI Six-Trak or other multi-timbral MIDI synth, it is possible to get several different voices from one keyboard. Also, if your keyboard has a "MIDI thru" jack, this should be able to drive the "MIDI in" jack of a second synthesizer.) A thru box basically turns one MIDI output into several MIDI outputs; each keyboard then "listens" to its particular output for information coming in over its assigned MIDI channel.

The Model 64 can control certain patch selection functions, although only with SCI products (Prophet V, 600, T-8) unless the synth has a built-in parameter for "outside channel selection" where it can send the present patch setting to the Model 64 to read. Again, the software initially released with the Model 64 is undergoing revision and updating is expected to provide some additional functions.

Conclusions. Generally, I am pleased with the Model 64. Since I have a a fairly low-cost set-up it provided a reasonable way to find out whether or not a sequencer could change my life. I've always done some of my most creative work with the built-in real time sequencers on the Prophet 600, and always wished I could do more with them. Now I can. However, I am very disappointed in the user's manual that comes with the Model 64. It is presented in a simple, dare I say it, "user-friendly" manner that misleads you at first glance into thinking that someone is going to tell you how to use the thing. The problem is that is is hard to follow, at least beyond the extremely simple basic instructions required to turn it on, record a sequence, and so forth. The two, brief, widely-spaced pages on MIDI tell you little. There is mention of certain functions, without telling you specifically how to use them. It is a far cry from the manuals I have for many other Sequential Circuits products.

I was very happy to find that old sequences saved to tape from my Prophet 600 could be sent to the Model 64, where they could be looped, transposed, and synched to a drum machine. A large number of people who've never had a sequencer before will be looking at the Model 64, and a whole lot of people getting into synthesizers for the first time will be looking at it as well. Because it can only hold one song in memory at one time, a faster-than-cassette disk drive would be a certain requirement for performers wanting to use the Model 64. It takes between 40 and 90 seconds to load in a song from cassette, not counting the time to specify to the computer you want to load something and starting/stopping the cassette. Incidentally, one of the Model 64's best features is that after you have fully constructed your song, you still have an additional track for laying a melody over all your accompanying sequences, bass lines, or whatever.

Overall, the Model 64 is an excellent sequencer with which to start; but I would not recommend it for a professional who would probably need much more power than this little device can currently provide.

FAST FOURIER SYNTHESIS IN ASSEMBLY LANUAGE

BY: Arnold Driessen & Hans de Bont

Nowadays, digital sound synthesis has become an important tool for the simulation of natural sounding instruments and the creation of sounds which cannot be synthesized by analog means. Prices of digital synthesizers that use these digital synthesis techniques are very high and not likely to be affordable for the home-synthesist. However, prices of personal computers decrease from day to day and although the quality of digitally synthesized sounds using this equipment cannot compete with those obtained with the Synclavier II or Fairlight CMI, good results are possible.

The best results can be obtained by means of a software driven DAC. With carefully written software, sounds which are dynamically altered can be produced. The computer switches at a certain rate through several waveform tables, which are filled with an approximation of the dynamic alteration of the sound in time. Data from these wavetables are sent to a D/A converter which produces a corresponding varying voltage at its output. The software used for this technique has mainly been developed by Hal Chamberlin, Frank Covitz and Cliff Ashcraft of Micro Technology Unlimited. Using this technique with an Apple computer, you are capable to produce four voices with a sample rate of 6 kHz. Faster computers of course can produce more than four voices with a broader frequency bandwidth. The low cost of this method makes it very suitable for the home synthesist; you need add only an inexpensive D/A converter if you already own a small computer. You can use the sounds produced with this technique in conjunction with your analog gear to create a score.

This article does not intend to describe Hal Chamberlin's wavetable scanning technique; for an excellent description of this technique, and the software involved, see his article in the April 1980 issue of Byte, in the Byte Book of Computer Music, or in his book titled "Musical Application of Microprocessors" (available through Polymart). Instead, we will describe a "very" fast Fourier synthesis program written in 6502 assembly language. You need a 6502-based or compatible microcomputer, such as the VIC-20, Commodore 64, Atari or Apple II, and at least 32K bytes of memory. The programs described in this article run directly on an Apple II computer and can be used to calculate the waveforms needed for the generation of dynamically altering sounds, at very high speed, with Hal Chamberlin's program.

Fast Fourier Synthesis in BASIC. As you probably know, any periodic waveform can be approximated by a harmonic series of sine waves having certain amplitudes. Thus the waveform can be written as a sum of sine waves of harmonically increasing frequency (1,2,3 etc. times the key frequency) multiplied by a factor belonging to each of these components. A

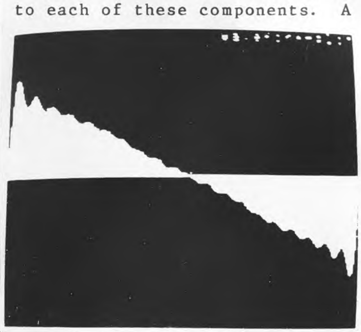


Fig. 1a The waveform which results from the harmonic specifications in Fig. 1c; it's a good approximation of a sawtooth wave.

sine wave does not contain any higher harmonics because the first component is a sine wave already. A waveform frequently used with analog synthesizer is the sawtooth; the sawtooth shown in **Fig. 1a** can be created if one adds sine waves whose amplitudes are in accordance with the following equation:

Amplitude Nth harmonic = 1/N

(Eq. 1)

The square wave is another common waveform. This waveform consists only of odd harmonics; equation 1 allows you to determine the amplitudes. **Fig. 1b** shows this waveform, while **Fig. 1c** and **Fig. 1d** plot the harmonic partials required to create an approximate sawtooth and square wave. The addition process is called Fourier synthesis or additive synthesis. Just as Fourier **analysis** breaks down a waveform into its harmonic components, Fourier **synthesis** creates a waveform from a set of harmonic amplitudes.

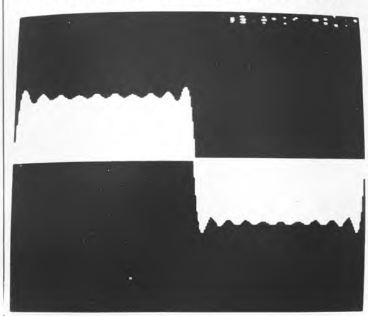


Fig. 1b The waveform resulting from the harmonic specifications in Fig. 1d; it approximates a square wave.

Listing 1 shows a simple BASIC program, similar to the program published by Hal Chamberlin in the April 1980 issue of

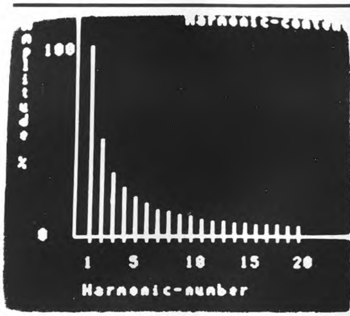


Fig. 1c A plot of the harmonic amplitude required to create an approximate sawtooth wave. For each odd and even harmonic specified the amplitude equals 100/N, where N is the harmonic.

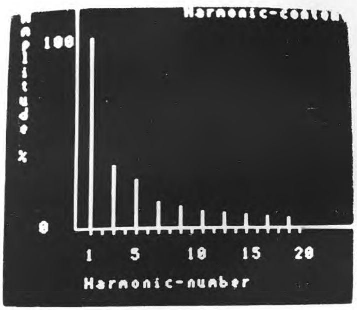


Fig. 1d A plot of the harmonic amplitudes required to create an approximate square wave. For each odd harmonic specified the amplitude is equal to 100/N, where N is the harmonic number.

Byte, which calculates a waveform from a user-defined set of harmonic amplitudes. The values of the calculated waveform will be printed on the screen. Although this program works correctly, it has the major drawback of slow speed. If you want to calculate more than thirty waveforms with up to twenty-four harmonics, you should have a lot of patience because the whole operation easily takes more than an hour! However, more than thirty waveforms are necessary to create a realistic-sounding instrument with minimal "buzz", due to switching between the wavetables. The 6502 assembly language program described later

Listing 1
A BASIC program that calculates and prints the values of a waveform. The user should specify the number of harmonics desired and moreover, the amplitudes of these harmonics should be defined. This results in a printed output of the waveform. One can fill a waveform sample table by POKEing the values into memory locations reserved for waveform tables. This program is a simplification of the program published by Hal Chamberlin (Byte April 1980).

Listing 1

```
10        REM FOURIER SERIES WAVEFORM TABLE FILLER
20        REM N IS THE HIGHEST HARMONIC
30        REM A IS THE AMPLITUDE ARRAY
40        HOME
50        INPUT "ENTER THE NUMBER OF THE HIGHEST HARMONIC:";N
60        DIM A(N)
70        PRINT:PRINT "ENTER AMPLITUDE'S HARMONICS (0-1)"
80        PRINT:PRINT "HARMONIC AMPLITUDE"
90        FOR I=1 TO N
100       HTAB 5:PRINT I;:HTAB 19:INPUT " ";A(I)
110       NEXT I
120       REM  CALCULATE AND PRINT WAVEFORM TABLE
130       HOME
140       PRINT " WAVETABLE LIST"
150       PRINT " LOCATION       VALUE"
160       FOR I=0 to 255
170       W=0
180       FOR J=1 TO N
190       W=W+A(J)*COS(0.02454369*I*J)
200       NEXT J
210       HTAB 5: PRINT I;: HTAB 20: PRINT INT (127*W)
220       NEXT I
230       END
```

Listing 2
A simple BASIC routine for setting up an array of amplitudes of the harmonics of a waveform table. Amplitudes should be defined as an integer between 0 and 255. The number of harmonics is saved in memory location #8300 (Decimal 33536) to serve as a counter by the assembler part of the program. A maximum of 24 harmonics may be defined and information will be stored from memory location #8301 (Decimal 33537). This program only works when you have saved the sine wave table and the fast Fourier assembler routine, both listed in listing 3.

Listing 2

```
10        REM SIMPLE ROUTINE FOR SETTING UP THE HARMONIC CONTENT
          OF A SOUND
20        REM N IS THE NUMBER OF HARMONICS
30        REM G IS THE NUMBER OF WAVEFORM TABLES
40        REM A IS THE AMPLITUDE OF AN HARMONIC
50        PRINT CHR$(4);"BLOAD FFS.OBJ0":REM LOAD ASSEMBLER PART
          FFS
60        PRINT CHR$(4);"BLOAD SINE":REM LOAD SINE TABLE
70        HOME : INPUT "ENTER NUMBER OF WAVETABLES (1-32): ";G
80        REM K IS THE WAVETABLE COUNTER
90        REM I IS THE HARMONIC NUMBER COUNTER
100       INPUT "ENTER HIGHEST DESIRED HARMONIC (1-24): ";N
110       FOR K=1 TO G
120       HOME : PRINT "WAVETABLE NUMBER";K
130       PRINT : PRINT "HARMONIC NUMBER:     AMPLITUDE (0-
          255):":PRINT
140       FOR I=1 TO N
150       HTAB 5: PRINT I;: HTAB 23: INPUT" ";A
160       H=33512+K*24+I:REM CALCULATE ADDRESS TO SAVE THE
```

takes only about 1 to 2 minutes to do the same operation.

Fast Fourier Synthesis in Assembly Language. First we will describe the program's principle of operation with the help of a flow chart, as well as some routines needed for the definition of the waveform amplitudes. These programs all run directly on an Apple II computer. One of the first things we have to do is to define the amplitudes of all harmonics; this can be accomplished with the simple BASIC program shown in listing 2. This program uses two additional machine language files, a sine wave table and a fast Fourier synthesis calculation program that we are going to discuss later on. The contents of the sine wave table is shown at the end of listing 3 (see pg. 14). You can enter this data via the monitor in the memory locations starting from #8700. When you have entered this data correctly you can save it with the following command: BSAVE SINE,A$8700,L$100. The assembler part of the program can also be entered via the monitor, however, it is easier to use something like the ASSEMBLER/EDITOR from the Dos 3.3 TOOLKIT for the Apple II. You should save the assembler part of the program under the name FFS.OBJ0. After using the monitor to enter the program, enter the following command: BSAVE FFS.OBJ0,A$8900,L$165. Remember listing 2 only works when you saved both the sine wave table and the fast Fourier assembler program.

Since we want to calculate a whole set of waveforms, we need to specify the amplitudes of the harmonics for each waveform. Then, we can vary the timbre of a sound by changing the relative amount of each harmonic. The amplitudes of twenty-four harmonics of each wavetable are loaded in a specific memory so that the assembler part of the program can use this data. Every amplitude should be defined as an integer number between 0 and 255. Great care should be taken to avoid overflow of the wavetable, which can happen when too many harmonics have a high amplitude. A general rule to avoid this distortion is to make sure that the sum of all harmonics is less than 255. The assembler program calculates with 16 bit values so that the amplitudes can best be defined in a range of 0 to 100 (hex). Later on

```
         AMPLITUDES (#8300 - #8700)
170      POKE H,A: REM SAVE AMPLITUDE IN TABLE
180      NEXT I
190      NEXT K
200      POKE 33536,N: REM SAVE NUMBER OF HARMONICS FOR
         ASSEMBLER PROGRAM
210      POKE 34559,G: REM SAVE NUMBER OF WAVETABLES FOR
         ASSEMBLER PROGRAM
220      HOME
230      CALL 35411
240      PRINT : PRINT "DONE": PRINT
250      PRINT "WAVETABLES ARE STORED FROM MEMORY"
260      PRINT "LOCATION #6300 TO #8300."
270      VTAB 22
280      INPUT "ENTER WAVEFORM NUMBER YOU WANT TO SEE    (0 tc
         END): ";O
290      IF O=0 THEN TEXT : HOME: END
300      HGR : HCOLOR= 7: REM SET HI-RESOLUTION GRAPHICS
310      H=25088 + (O*256):REM START ADDRESS OF WAVEFORM
320      FOR I=0 TO 255
330      A = INT (H + I): REM ADDRESS OF WAVEFORM
340      REM GET WAVEFORM AND CENTER FOR CRT SCREEN
350      F = INT (148 - ((PEEK(A))/2))
360      A = I + 20
370      HPLOT A,84 TO A,F
380      NEXT I
390      HOME
400      GOTO 270
```

Listing 3
The fast Fourier synthesis program. The program consists of a FOR-NEXT loop which performs the repeated calculation of more than one waveform table. Memory locations starting from #8301 (Decimal 33537) should contain the amplitude of the harmonics (see listing 2). At the end of this listing a HEXDUMP of the sine table is printed. Both the sine table, and the assembler, are automatically loaded from disk by the program in listing 2.

```
SOURCE FILE: FFS
0000:             *
0000:             *
0000:             *     ;********************
0000:             *     ;*                  *
0000:             *     ;*   FAST FOURIER   *
0000:             *     ;*     SYNTHESIS    *
0000:             *     ;*                  *
0000:             *     ;********************
0000:             *     ;
0000:             *     ;
0000:             *     ;      *** MULTI ***
0000:             *     ;MULTIPLICATION SUBROUTINE
0000:             *     ;
----- NEXT OBJECT FILE NAME IS FFS.OBJO
8900:                        ORG  $8900       ;LOAD ADRESS PROGRAM
8900:             *     ;
8900:             *     ;MULTI PERFORMS A MULTIPLICATION
8900:             *     ;IN 6502 ASSEMBLY LANGUAGE. TWO
8900:             *     ;8 BIT NUMBERS ARE MULTIPLIED
8900:             *     ;RESULTING IN A 16 BIT PRODUCT
8900:             *     ;
86F0:             MUPL   EQU  $86F0       ;MULTIPLICANT
86F1:             MUPR   EQU  $86F1       ;MULTIPLIER
86F2:             PLSB   EQU  $86F2       ;PRODUKT LSB
86F3:             PMSB   EQU  $86F3       ;PRODUKT MSB
86FB:             TSAV   EQU  $86FB       ;TEMP X
8900:             *     ;
8900:             *     ;
8900:A9 00        MULTI  LDA  #$00        ;LSB PRODUKT ZERO
8902:8D F3 86            STA  PMSB        ;ALSO MSB PRODUKT
8905:8E FB 86            STX  TSAV        ;SAVE X
8908:A2 08               LDX  #$08        ;8 BITS MULTIPLIER
890A:0A           SHIFT  ASL  A           ;SHIFT PRODUCT LEFT ONE BIT
890B:2E F3 86            ROL  PMSB        ;
```

when all the calculation has been done, a simple program can check the highest value in the wavetables. With this number a constant can be constructed so that multiplication of the defined amplitudes of the harmonics with this constant, and subsequent recalculation of the waveform, yields the highest possible dynamic range.

The program in listing 2 does not contain a graphical presentation of the entered amplitudes. However, it does contain a simple routine that enables you to plot the calculated waveforms on the HIRES screen. **Fig. 2** shows an example of the definition of the harmonic content of a sound. This plot is an approximation of a trumpet note. Amplitudes of the harmonics are graphically displayed on the VDU screen so that a nice representation of the harmonic content of a sound is available. With a little practice you can easily construct such a plot by yourself. Joystick or HIRES light pen owners can enter the amplitudes of the harmonics more quickly. (The Fairlight CMI uses a light pen for this same purpose.)

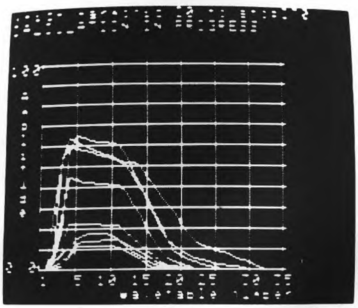


Fig. 2 The harmonic content of a trumpet note. Amplitudes of harmonics are immediately plotted on the VDU screen, in order to obtain a view of the harmonic content of a sound. The lines represent the envelope of a harmonic in time (wavetables).

When all the amplitude information has been loaded, the assembler part of the program performs the calculations and table fill-

```
890E:0E F1 86          ASL   MUPR      ;SHIFT MULTIPLIER LEFT
8911:90 09             BCC   CHCNT     ;NO ADDITION IF NEXT BIT IS ZERO
8913:18                CLC              ;ADD MULTIPLICANT TO PRODUCT
8914:6D F0 86          ADC   MUPL      ;
8917:90 03             BCC   CHCNT     ;
8919:EE F3 86          INC   PMSB      ;WITH CARRY IF NECESSARY
891C:CA        CHCNT   DEX              ;NEXT BIT
891D:D0 EB             BNE   SHIFT     ;
891F:8D F2 86          STA   PLSB      ;STORE LSB OF PRODUCT
8922:AE FB 86          LDX   TSAV      ;RESTORE X REG.
8925:60                RTS              ;RETURN
8926:          *     ;
8926:          *     ;
8926:          *     ;     *** ADD ***
8926:          *     ;ADDITION SUBROUTINE
8926:          *     ;
8926:          *     ;ADD PERFORMS A DOUBLE PRECISION
8926:          *     ;ADDITION
8926:          *     ;
86F0:          LSB1    EQU   $86F0     ;LSB NUMBER ONE
86F1:          MSB1    EQU   $86F1     ;MSB NUMBER ONE
86F2:          LSB2    EQU   $86F2     ;LSB NUMBER TWO
86F3:          MSB2    EQU   $86F3     ;MSB NUMBER TWO
00F9:          LSBP    EQU   $F9       ;LSB PRODUCT
00FA:          MSBP    EQU   $FA       ;MSB PRODUCT
8926:          *     ;
8926:          *     ;
8926:18        ADD     CLC              ;CLEAR CARRY TO START
8927:AD F0 86          LDA   LSB1      ;ADD LEAST SIGNIFICANT BITS
892A:6D F2 86          ADC   LSB2      ;
892D:85 F9             STA   LSBP      ;
892F:AD F1 86          LDA   MSB1      ;ADD MOST SIGNIFICANT BITS
8932:6D F3 86          ADC   MSB2      ;
8935:85 FA             STA   MSBP      ;
8937:60                RTS              ;RETURN
8938:          *     ;
8938:          *     ;
8938:          *     ;    *** INIT ***
8938:          *     ;INITIALISATION ROUTINE
8938:          *     ;
8938:          *     ;
8938:          *     ;INIT CLEARS THE CONTENT OF THE
8938:          *     ;WORKTABLE WHICH IS USED FOR
8938:          *     ;SAVING THE TEMPORARY RESULTS OF
8938:          *     ;THE ADDITIONS. IT ALSO
8938:          *     ;INITIALISES THE HARMONIC NUMBER
8938:          *     ;COUNTER
8938:          *     ;
8938:          *     ;
8800:          WRTB    EQU   $8800     ;WORKTABLE 8800-88FF
8300:          HANR    EQU   $8300     ;HARMONIC NUMBER
86F8:          HACNT   EQU   $86F8     ;HARMONIC NUMBER COUNTER
8938:          *     ;
8938:A0 00     INIT    LDY   #$00      ;CLEAR REGISTERS
893A:A9 00             LDA   #$00      ;
893C:A2 00             LDX   #$00      ;
893E:9D 00 88  CLRO    STA   WRTB,X    ;CLEAR WORKTABLE
8941:CA                DEX              ;NEXT LOCATION
8942:D0 FA             BNE   CLRO      ;IF NOT DONE NEXT
8944:A2 00             LDX   #$00      ;CLEAR X REG. AGAIN
8946:AD 00 83          LDA   HANR      ;GET HARMONIC NUMBER
8949:8D F8 86          STA   HACNT     ;SAVE AS COUNTER
894C:          *     ;
894C:          *     ;
894C:          *     ;     *** AMPL ***
894C:          *     ;AMPLITUDE LOOP-UP ROUTINE
894C:          *     ;
894C:          *     ;
894C:          *     ;AMPL SEARCHES THE AMPLITUDE OF
894C:          *     ;A HARMONIC IN THE ARRAY SET UP
894C:          *     ;WITH LISTING 2. IT TESTS IF IT
894C:          *     ;IS ZERO AND IF NOT SAVES IT FOR
894C:          *     ;LATER CALCULATIONS. THE
894C:          *     ;AMPLITUDE LOOKUP TABLE STARTS
894C:          *     ;AT #8301
894C:          *     ;
894C:          *     ;
86F9:          WTNR    EQU   $86F9     ;WAVETABLE NUMMER
86FA:          AMP     EQU   $86FA     ;AMPLITUDE
894C:          *     ;
894C:          *     ;
894C:AD F9 86  AMPL    LDA   WTNR      ;GET WAVETABLE NUMBER
894F:8D F0 86          STA   MUPL      ;PREPARE FOR MULTIPLICATION
8952:A9 18             LDA   #$18      ;MAXIMUM 24 HARMONICS
8954:8D F1 86          STA   MUPR      ;
8957:20 00 89          JSR   MULTI     ;DO MULTIPLICATION
895A:18                CLC              ;CLEAR CARRY BEFORE ADDITION
895B:AD F2 86          LDA   PLSB      ;USE PRODUCT AS OFFSET
895E:6D F8 86          ADC   HACNT     ;OFFSET WITH HARMONIC NUMBER
8961:8D F2 86          STA   LSB2      ;SAVE LEAST SIGNIFICANT BYTE
8964:AD F3 86          LDA   PMSB      ;CALCULATE MOST SIGNIFICANT BYTE
8967:69 00             ADC   #$00      ;ADD OVERFLOW
8969:8D F3 86          STA   MSB2      ;SAVE MOST SIGNIFICANT BYTE
896C:A9 E8             LDA   #$E8      ;GET LSB OF START ADRESS TABLE-1
```

ing. **Fig. 3** shows the flow chart for this part of the program. The program begins by reading the wavetable number and the number of harmonics of the waveform. It

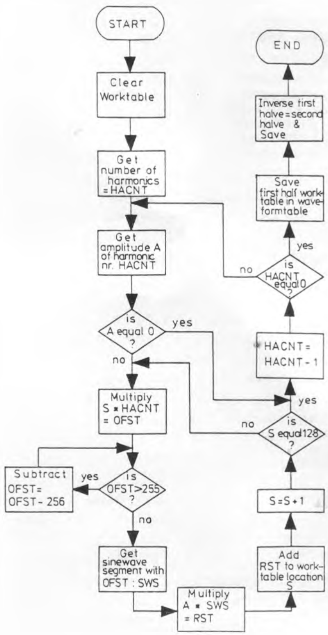


Fig. 3 Flow chart of the assember part of the fast Fourier synthesis program. The principle is explained in the text. Definitions of the abbreviations are:

HACNT : harmonic number counter

A : amplitude of an harmonic

S : sine wave table step pointer

OFST : offset for sine wave table scan purpose

SWS : sine wave sample, the result of the sine wave table look-up

RST : result of the multiplication of the amplitude of the harmonic and the sine wave sample

```
896E:8D F0 86             STA   LSB1        ;SAVE FOR ADDITION
8971:A9 82                LDA   #$82        ;MSB OF START ADRESS TABLE-1
8973:8D F1 86             STA   MSB1        ;
8976:20 26 89             JSR   ADD         ;AND DO ADDITION TO CALCULATE
8979:B1 F9                LDA   (LSBP),Y    ;AMPL. ADRESS AND LOAD AMPLITUDE
897B:F0 59                BEQ   NEXT        ;IF ZERO NEXT
897D:A2 00                LDX   #$00        ;OR PREPARE X REG. AS OFFSET
897F:8D FA 86             STA   AMP         ;SAVE AMPLITUDE
8982:            *
8982:            *      ;
8982:            *      ;     *** SLR ***
8982:            *      ;SINEWAVE TABLE LOOK-UP ROUTINE
8982:            *      ;
8982:            *      ;
8982:            *      ;SLR SEARCHES A VALUE FROM THE
8982:            *      ;SINE WAVETABLE ACCORDING THE
8982:            *      ;HARMONIC NUMBER AND ITS LAST
8982:            *      ;LOOKUP LOCATION. THIS VALUE IS
8982:            *      ;MULTIPLIED WITH THE AMPLITUDE
8982:            *      ;OF THE HARMONIC AND THE 16 BIT
8982:            *      ;RESULT IS SAVED IN THE WORK-
8982:            *      ;TABLE. THIS LOOP IS PERFORMED
8982:            *      ;128 TIMES, REPRESENTING HALF
8982:            *      ;OF THE WAVEFORM.
8982:            *      ;
8982:            *      ;
8982:8E F0 86    SLR      STX   MUPL        ;PREPARE FOR MULTIPLICATION
8985:AD F8 86             LDA   HACNT       ;THE STEP- AND HARMONIC NUMBER
8988:8D F1 86             STA   MUPR        ;SAVE AS MULTIPLIER
898B:20 00 89             JSR   MULTI       ;DO MULTIPLICATION TO DETERMINE
898E:A9 00                LDA   #$00        ;THE OFFSET AND ADD TO SINEWAVE
8990:8D F0 86             STA   LSB1        ;TABLE ADRESS
8993:8D F3 86             STA   MSB2        ;
8996:A9 87                LDA   #$87        ;PAGE NUMBER SINEWAVE TABLE
8998:8D F1 86             STA   MSB1        ;PREPARE FOR ADDITION
899B:20 26 89             JSR   ADD         ;DO ADDITION
899E:B1 F9                LDA   (LSBP),Y    ;TAKE A SAMPLE
89A0:8D F0 86             STA   MUPL        ;PREPARE FOR MULTIPLICATION
89A3:AD FA 86             LDA   AMP         ;WITH AMPLITUDE OF HARMONIC
89A6:8D F1 86             STA   MUPR        ;
89A9:20 00 89             JSR   MULTI       ;DO MULTIPLICATION
89AC:AD F2 86             LDA   PLSB        ;DO ADDITION TO THE WORKTABLE
89AF:8D F0 86             STA   LSB1        ;
89B2:AD F3 86             LDA   PMSB        ;
89B5:8D F1 86             STA   MSB1        ;
89B8:BD 00 88             LDA   WRTB,X      ;GET LSB WORKTABLE
89BB:8D F2 86             STA   LSB2        ;BECOMES LSB LATEST ADDITION
89BE:BD 80 88             LDA   WRTB+$80,X  ;GET MSB WORKTABLE
89C1:8D F3 86             STA   MSB2        ;BECOMES MSB LATEST ADDITION
89C4:20 26 89             JSR   ADD         ;DO ADDITION
89C7:A5 F9                LDA   LSBP        ;RENEW WORKTABLE
89C9:9D 00 88             STA   WRTB,X      ;AS LSB
89CC:A5 FA                LDA   MSBP        ;SAME FOR MSB
89CE:9D 80 88             STA   WRTB+$80,X  ;
89D1:E8                   INX               ;NEXT SAMPLE WAVEFORM
89D2:E0 80                CPX   #$80        ;ALL 128 ENTRIES DONE?
89D4:D0 AC                BNE   SLR         ;NO THEN NEXT SAMPLE
89D6:CE F8 86    NEXT     DEC   HACNT       ;YES THEN NEXT HARMONIC
89D9:F0 03                BEQ   PERM        ;ALL HARMONICS DONE? YES, CONTINUE
89DB:4C 4C 89             JMP   AMPL        ;NO THEN NEXT
89DE:            *
89DE:            *      ;
89DE:            *      :     *** PERM ***
89DE:            *      ;PERMUTATION ROUTINE
89DE:            *      ;
89DE:            *      ;
89DE:            *      ;THIS ROUTINE DOES THE
89DE:            *      ;PERMUTATIONS NEEDED TO REPOSITION
89DE:            *      ;THE 16 BIT CALCULATED WAVEFORM
89DE:            *      ;TO THE 8 BIT FORMAT, SO THAT
89DE:            *      ;IT OSCILLATES AROUND #80. THE
89DE:            *      ;SECOND HALF OF THE WAVEFORM
89DE:            *      ;IS CALCULATED BY THE CREATION
89DE:            *      ;OF A IMAGE OF THE FIRST HALF
89DE:            *      ;TO PI. THE COMPLETE WAVEFORM IS
89DE:            *      ;SAVED IN THE APPROPRIATE WAVEFORM
89DE:            *      ;TABLE
89DE:            *      ;
89DE:            *      ;
86FC:            MID      EQU   $86FC       ;MIDDLE POSITION WAVEFORM
00FB:            YLSB     EQU   $FB         ;ADRESS INDIREKT Y-INDEX ADRESSING
00FC:            YMSB     EQU   $FC         ;ADRESS INDIREKT Y-INDEX ADRESSING
89DE:            *      ;
89DE:            *      ;
89DE:AD 80 88    PERM     LDA   WRTB+$80    ;GET MIDDLE VALUE
89E1:8D FC 86             STA   MID         ;SAVE
89E4:AD F9 86             LDA   WTNR        ;GET WAVETABLE NUMBER
89E7:8D F0 86             STA   MUPL        ;PREPARE FOR MULTIPLICATION
89EA:A9 FF                LDA   #$FF        ;MULTIPLY BY 256
89EC:8D F1 86             STA   MUPR        ;
89EF:20 00 89             JSR   MULTI       ;DO MULTIPLICATION
89F2:AD F9 86             LDA   WTNR        ;PREPARE TO DETERMINE WAVETABLE
89F5:8D F0 86             STA   LSB1        ;POSITION
89F8:A9 62                LDA   #$62        ;FIRST TABLE AT PAGE #62+1
89FA:8D F1 86             STA   MSB1        ;
```

starts its calculation with the highest harmonic and the harmonic number is decremented until all harmonics have been done. The amplitude of the harmonic is loaded and a test is carried out to determine whether the amplitude equals zero or not. If the former is the case the program continues with the next harmonic. Otherwise, the amplitude will be multiplied with the contents of a sine wave table saved as a pre-computed version in memory. Samples of the sine wave table are taken according to the number of the harmonic (e.g. steps of two are used for the second harmonic, steps of three for the third harmonic, and so on). When this step-pointer reaches the end of the 255 bytes long sine wavetable, it starts all over again. This routine is repeated until 128 samples of the waveform have been calculated. Since the second half of a periodic waveform is symmetric and the inverse of the first half, only the first half has to be calculated. This trick almost doubles the calculation speed. Samples of the calculated waveform are added to the contents of a "worktable", which contains the temporary results of the additions. The arithmetic routines are done with double precision (16 bits). When all harmonics have been done, the most significant byte (MSB) of the worktable is saved in the appropriate waveform table. To calculate the second half of the waveform, a routine creates a mirror image of the first half around the center (Pi) of the wavetable. This results in a waveform which oscillates around 128. In order to calculate more than one waveform table, the assembler part of the program (see listing 3) incorporates a FOR-NEXT loop.

The assembler part of the program starts at memory location #8A53 (decimal 35411) and uses two arithmetic subroutines: MULTI (multiplication) and ADD (addition). These are necessary because the 6502 microprocessor has no hardware equivalent. As with addition, multiplication is done with double precision. **Fig. 4** shows a list of the most important memory locations used by the assembler part of the program. The first half of page #88 is used by the least significant byte (LSB) of the worktable, while the second half of this page is filled with the MSB (most significant byte).

```
89FD:20 26 89              JSR   ADD          ;DO ADDITION
8A00:A5 F9                 LDA   LSBP         ;STORE FOR INDIRECT Y INDEXED
8A02:85 FB                 STA   YLSB         ;ADDRESSING
8A04:A5 FA                 LDA   MSBP         ;ALSO FOR MSB
8A06:85 FC                 STA   YMSB         ;
8A08:A2 00                 LDX   #$00         ;CLEAR ALL REGISTERS
8A0A:8A           SAROUT   TXA
8A0B:A8                    TAY                 ;
8A0C:38                    SEC                 ;SET CARRY FOR SUBSTRACTION
8A0D:BD 80 88              LDA   WRTB+$80,X   ;GET MSB OF WORKTABLE
8A10:ED FC 86              SBC   MID          ;AND SUBSTRACT MIDDLE POSITION
8A13:18                    CLC                 ;
8A14:69 80                 ADC   #$80         ;PLACE RESULT AROUND MIDDLE
8A16:91 FB                 STA   (YLSB),Y     ;POSITION AND SAVE IT
8A18:E8                    INX                 ;NEXT SAMPLE
8A19:E0 80                 CPX   #$80         ;HALF WAVEFORM DONE?
8A1B:D0 ED                 BNE   SAROUT       ;NO THEN NEXT SAMPLE
8A1D:A0 00                 LDY   #$00         ;CLEAR Y REGISTER
8A1F:38           SAROUT2  SEC                 ;NEXT
8A20:8E FB 86              STX   TSAV         ;SAVE LAST SAMPLE
8A23:A9 FF                 LDA   #$FF         ;PREPARE TO CREATE IMAGE
8A25:ED FB 86              SBC   TSAV         ;INVERSE LAST SAMPLE
8A28:8D F0 86              STA   LSB1         ;AND REPOSITION THE SYMMETRY
8A2B:98                    TYA                 ;
8A2C:8D F1 86              STA   MSB1         ;
8A2F:A5 FB                 LDA   YLSB         ;
8A31:8D F2 86              STA   LSB2         ;
8A34:A5 FC                 LDA   YMSB         ;
8A36:8D F3 86              STA   MSB2         ;
8A39:20 26 89              JSR   ADD          ;DO ADDTION
8A3C:B1 F9                 LDA   (LSBP),Y     ;AND RENEW ENTRY COUNTER
8A3E:8D FB 86              STA   TSAV         ;
8A41:8A                    TXA                 ;
8A42:A8                    TAY                 ;
8A43:38                    SEC                 ;
8A44:A9 00                 LDA   #$00         ;
8A46:ED FB 86              SBC   TSAV         ;INVERSE AND SAVE IT IN THE
8A49:91 FB                 STA   (YLSB),Y     ;WAVEFORM TABLE
8A4B:A0 00                 LDY   #$00         ;
8A4D:E8                    INX                 ;NEXT ENTRY
8A4E:E0 00                 CPX   #$00         ;ALL DONE?
8A50:D0 CD                 BNE   SAROUT2      ;NO THEN DO NEXT SAMPLE
8A52:60                    RTS                 ;NO THEN RETURN
8A53:                 *    ;
8A53:                 *    ;
8A53:                 *    ;     *** START ***
8A53:                 *    ;START IS THE MAIN LOOP
8A53:                 *    ;
8A53:                 *    ;
8A53:                 *    ;START PERFORMS A LOOP FOR ALL
8A53:                 *    ;WAVEFORMS.
8A53:                 *    ;
8A53:                 *    ;
86FF:             WAVENR   EQU   $86FF        ;LOOP COUNTER
8A53:A9 01                 LDA   #$01         ;PREPARE
8A55:8D F9 86              STA   WTNR         ;
8A58:20 38 89     LOOP     JSR   INIT         ;CALCULATE WAVE
8A5B:EE F9 86              INC   WTNR         ;NEXT WAVE TABLE
8A5E:CE FF 86              DEC   WAVENR       ;
8A61:D0 F5                 BNE   LOOP         ;ALL DONE?
8A63:60                    RTS                 ;OTHERWISE RETURN TO BASIC

*** SUCCESSFUL ASSEMBLY: NO ERRORS
```

```
8926 ADD          86FA AMP          894C AMPL         891C CHCNT
893E CLR0         86F8 HACNT        8300 HANR         8938 INIT
8A58 LOOP         86F0 LSB1         86F2 LSB2           F9 LSBP
86FC MID          86F1 MSB1         86F3 MSB2           FA MSBP
8900 MULTI        86F0 MUPL         86F1 MUPR         89D6 NEXT
89DE PERM         86F2 PLSB         86F3 PMSB         8A0A SAROUT
8A1F SAROUT2      890A SHIFT        8982 SLR          86FB TSAV
86FF WAVENR       8800 WRTB         86F9 WTNR           FB YLSB
  FC YMSB
```

This table will be used as a temporary storage of the results of the additions. Page #87 contains the sine wave table, with memory location #8300 reserved for the number of harmonics defined. The remaining location on pages #83 to #87 are filled with the amplitudes of the harmonics for each waveform. A number of page #00 locations are used by the program, as shown in **Fig. 4**.

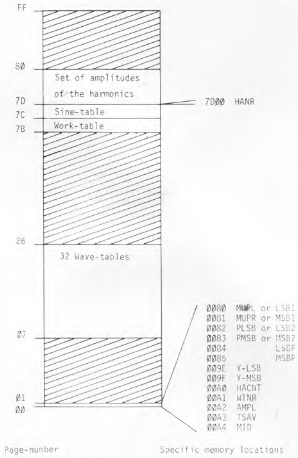


Fig. 4 Memory map of the locations used by the fast Fourier synthesis program shown in listing 3. The abbreviations are explained in listing 3.

```
  F9 LSBP          FA MSBP          FB YLSB          FC YMSB
8300 HANR        86F0 LSB1        86F0 [illegible]PL  86F1 MUPR
86F1 MSB1        86F2 PLSB        86F2 L[illegible]B2 86F3 PMSB
86F3 MSB2        86F8 HACNT       86F9 WTNR        86FA AMP
86FB TSAV        86FC MID         86FF WAVENR      8800 WRTB
8900 MULTI       890A SHIFT       891C CHCNT       8926 ADD
8938 INIT        893E CLRO        894C AMPL        8982 SLR
89D6 NEXT        89DE PERM        8A0A SAROUT      8A1F SAROUT2
8A58 LOOP
```

(contents of Sine wave table):

```
HEX DUMP     uBJECT: SINE

       0  1  2  3  4  5  6  7  8  9  A  B  C  D  E  F

8700: 80 83 86 89 8C 8F 92 95 98 9C 9F A2 A5 A8 AB AE
8710: B0 B3 B6 B9 BC BF C1 C4 C7 C9 CC CE D1 D3 D5 D8
8720: DA DC DE E0 E2 E4 E6 E8 EA EC ED EF F0 F2 F3 F4
8730: F6 F7 F8 F9 FA FB FC FD FE FE FF FF FF FF FF FF
8740: FF FF FF FF FF FF FE FE FD FC FC FB FA F9 F8 F7
8750: F6 F5 F3 F2 F0 EF ED EC EA E8 E6 E4 E3 E1 DE DC
8760: DA D8 D6 D3 D1 CE CC C9 C7 C4 C1 BF BC B9 B6 B4
8770: B1 AE AB A8 A5 A2 9F 9C 99 96 92 8F 8C 89 86 83
8780: 80 7E 7A 77 74 71 6E 6B 68 65 62 5F 5C 59 56 53
8790: 50 4D 4A 47 44 42 3F 3C 3A 37 34 32 2F 2D 2B 28
87A0: 26 24 22 20 1E 1C 1A 18 16 15 13 11 10 0E 0D 0C
87B0: 0A 09 08 07 06 05 04 04 03 02 02 01 01 01 01 01
87C0: 01 01 01 01 01 01 02 02 03 04 04 05 06 07 08 09
87D0: 0A 0B 0D 0E 0F 11 13 14 16 18 19 1B 1D 1F 21 24
87E0: 26 28 2A 2D 2F 31 34 36 39 3C 3E 41 44 47 49 4C
```

Of course, you can change all memory locations according to the free space of your computer. Labels should be changed in listing 3 according to your implementation of the actual program listing. Comments are placed behind the opcodes to make the program easier to understand.

Application. As already mentioned, this program can be used to calculate the wavetables needed for digital sound reproduction with Hal Chamberlin's wavetable scanning technique. We use his technique with an Apple II computer. 8K bytes are used for the waveform tables for one voice consisting of thirty-two wavetables. This is enough to obtain a clear sound and a reasonable approximation of a natural sounding instrument. Two voices are used with a 6 kHz frequency range and both can have their own wavetables. In order to obtain a fuller sound, the voice frequencies may be offset. **Figs. 5a,b, and c** show a 3-dimensional image of the 32 waveform tables of the trumpet sound, along with two wavetables of this trumpet sound.

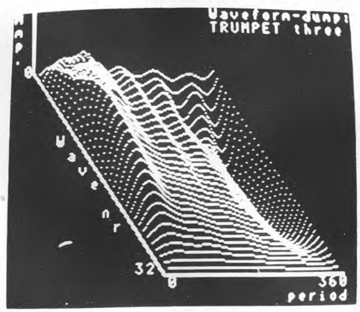


Fig. 5a Three-dimensional image of the wavetable contents of a trumpet note and two waveform plots.

While the resulting sound does not have the quality of a Synclavier II or Fairlight CMI, it sounds quite realistic. In fact, the Fairlight uses similar techniques, although wavetables are read with higher speed when a note becomes higher. With the table scanning technique, samples are taken from the wavetables with larger inter-

(continued on page 16)

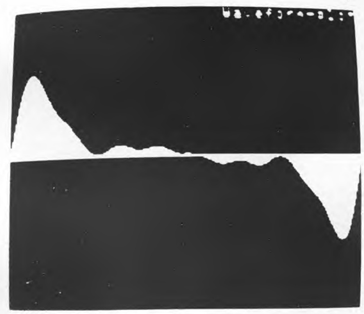


Fig. 5b Left waveform harmonic amplitudes are 59(1), 62(2), 65(3), 44(4), 23(5), 15(6), 13(7), and 12(8).

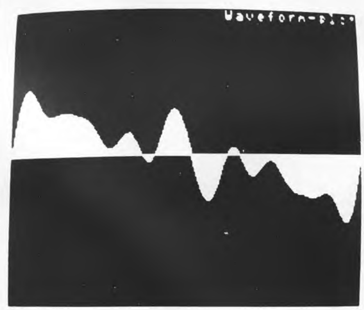


Fig. 5c Right waveform harmonic amplitudes are 80(1), 45(2), 52(3), 0(4), 37(5), 0(6), 42(7), 0(8), 6(9), and 13(10). The harmonic number is enclosed in brackets.

Second Annual Polyphony Awards Banquet

by: Robert Carlberg

The year of Big Brother, women in space, and Doonesbury's return has seen Electronic Music mature tremendously. Not only are synthesizers being used frequently -- and well -- in popular music, but "Electronic Music" itself has grown more professional and accessible, blurring the distinction. No longer a novelty as it is absorbed into the mainstream of music, E.M. has gained respect and passes on its vitality to a new generation.

Album of The Year: Group 87 - **A Career in Dada Processing** (reviewed Aug). A prime example of the new maturity in electronic music, brought on by the influx of extablished musicians and sophisticated hardware. Honorable mentions: Laurie Anderson - **Mister Heartbreak** (Aug) and Thomas Dolby - **The Flat Earth** (Apr): Adroit use of electronics in more conventional frameworks.

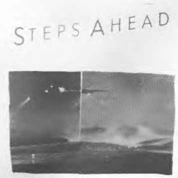
Song of the Year: "Mr. Moto's Penguin (Who'd Be An Eskimo's Wife?)" from Mark Isham's **Vapor Drawings** (Feb). One of the principals of Group 87, Isham proves his success was no fluke.

Tape of the Year: John Wiggins - **Anagenic** (Apr). An example of the type of music that best fits the cassette format, being perhaps too complicated and heroic for the informal LP.

Best Album Title: **The Flat Earth** by Thomas Dolby (Apr). Acknowledges and gently spoofs the inexplicable resurgence in reactionary thought these days.

Best Song Title: "Ghosts Before Breakfast" from Michelle Musser's **A Cast of Shadows** cassette (Oct). No reason -- just like the sound of it.

Best Popular Synthesizer Album: Take your pick -- there were a million of them on the radio. A couple hundred thousand anyway.

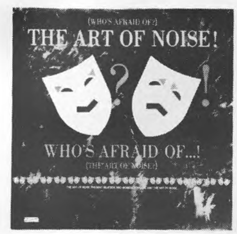


Best Jazz Synthesizer Album: **Modern Times** by Steps Ahead (Aug). Shows how to be "modern" while still being timeless.

Best Avant-Garde Electronic Album: **Computer Music** by Michael McNabb (Aug). Comes with the kind of lengthy 'explanations' that usually spell trouble, but this one's a gem of shifting tonalities and surprising transformations.

Most Improved: Laurie Anderson for **Mister Heartbreak** (Aug). After a couple of albums which were not substantially advanced over her debut, Laurie came out with a winner.

Most Likely to Succeed: The Art of Noise -- **Who's Afraid of?** (Oct). If not them, then someone like them. It's too dumb to fail.



Best Group Name: Bluetoy -- **Reinventing the Wheel Without a Third Eye** (cassette, Oct). When asked, Chris admitted it's his license plate.

Best Mixing: Iverson and Walters **First Collection** (Oct) for mixing bluegrass and EM. Tri Atma **Yearning & Harmony** (Oct) for mixing jazz guitar, Indian percussion and EM.

Irony of the Year No. 1: Spinal Tap, the parody of what's worst in Heavy Metal, played concerts and

(continued on page 16)

FAST FOURIER...

vals at higher frequencies. This results in loss of resolution when playing high notes; therefore, best results are obtained in the lower octaves. It is possible to get a good simulation of plucked string instruments like cello or banjo. Of course, steady-state waveforms can also be produced.

Beside the already mentioned poor high frequency response of this technique, a slight background hiss due to interpolation noise resulting from playback rounding errors also occurs. This effect can be minimized by using triangle waves instead of sine waves during the fast Fourier synthesis. (This probably occurs due to the more linear nature of a triangle wave.) Noise can also be reduced by coupling an expander to the D/A converter.

We hope many of you will use this program with Hal Chamberlin's program, and we are most interested to hear about your applications and/or extensions. Write: **Hans de Bont, Bontekoestraat 44, 6826 SX Arnhem, The Netherlands.**

References

Chamberlin, Hal. "Advanced Real Time Music Synthesis Techniques." Byte (April, 1980), page 70.

_______________. "A Sampling of Techniques for Computer Performance of Music." Byte (September, 1977), page 62. (Also published in the Byte book of computer music Ed. Christopher P. Morgan).

Moorer, J. and J. Grey. "Lexicon of Analyzed Tones: The Trumpet." Computer Music Journal Vol. II, No. 2 (1978), page 23.

Leventhal, L. A. 6502 Assembly Language Programming. Osborne, 1979.

For Listings:

Location	Label or function
#8900 - #8A80	Assembler program FFS.OBJ0
#8800 - #88FF	Worktable
#8700 - #87FF	Sinetable SINE
#86FF	WAVENR
#86FC	MID
#86FB	TSAV
#86FA	AMP
#86F9	WTNR
#86F8	HACNT
#86F3	MSB2 or PMSB
#86F2	LSB2 or PLSB
#86F1	MSB1 or MUPR
#86F0	LSB1 or MUPL
#8301 - #8601	Harmonic table
#8300	HANR
#6300- #82FF	Thirty-two wave-tables
#00FC	YMSB
#00FB	YLSB
#00FA	MSBP
#00F9	LSBP

AWARDS...cont.

put out an album just like a real band. The bandmembers were flabergasted to be taken seriously.

Irony of the Year No. 2: "New Age Music," the current euphemism for minimalism, began as electronic "Cosmic Music" in the early 70s. It has been adopted by the California-based healing movement, made up of aging ex-hippies who shun technology. Well, not all technology, just the "bad" technology.

Irony of the Year No. 3: Noiseless, wearless digital Compact discs are flooding the music scene, but in their rush for "product" most of the labels are releasing some really crummy old analog recordings. These could cause the new CD owners to toss their players down the outdoor plumbing.

This truly is a mad, mad, mad, mad, mad world. And welcome to it. Grab the bull by the tail and face the situation. See you next year.

SHOCK HAZARD GROUNDING

by: Jack Sondermeyer

(Editor's note: The following is reprinted with permission from **The Monitor**, a publication from Peavey Electronics Corporation. We feel that the information in this article should be disseminated as widely as possible.)

The power supply cord used on most modern electronic equipment has a three pin plug. This article will explain why shock hazards will result if the ground system is defeated.

The power distribution system used in the United States is 120 volts "alternating current" at 60 cycles. For many years this standard AC (mains) system was a two prong plug and socket combination wired with one side "hot" (usually the black wire at 120 volts) and the other side neutral (usually the white wire at 0 volts or ground). However, in the early days there were no true standards. Often there was no color code and sometimes no hot or neutral wires (in this case both wires were hot). Unfortunately, this two wire system is still found in many older buildings throughout the country.

There has always been a certain amount of "respect" for the chance of getting shocked by the power system but most people are not sure how or why. Let's review the system. The "hot" wire is at 120 volts and the other wire is neutral or ground. If a person were to touch the neutral wire only, no shock would result simply because there is no voltage on it. If he were to touch the hot wire only, again nothing would happen to him unless some other part of his body were to become grounded. A person is considered to be grounded if he comes in contact with a water pipe, metal conduit, the neutral or ground wire, or stands barefoot on a concrete floor. A person is usually insulated from electrical ground by rubber or leather shoes. In other words, neither wire is a shock hazard unless a person is grounded, and then only the hot is a potential shock hazard. Of course if a person were to touch both wires at the same time, he would get shocked simply because his body is completing connection between "hot" and "ground" wires.

As the use of electricity became universal, it became apparent that the existing system had some serious problems, and various "standards" agencies began to review power distribution products and practices. Out of these efforts came Underwriters Laboratories (UL) and many others which were created to help design a system which would reduce the risk of shock hazard. Ultimately, dramatic changes were instituted which brought about the three prong plug, among other improvements in our system. However, these changes have caused many problems to manufacturers of equipment which plugs into this power system. These problems are due primarily to the fact that many older buildings still have the old two wire systems and two prong wall sockets. In order to use modern equipment in these older buildings, some people simply "break off" the ground prong from the plug. Although "ground adaptors" are readily available, a few users simply "can't be bothered." Also, newer three prong sockets are often wired (or miswired) into older two wire systems with horrible results including shock hazards (which can cause destruction of electronic equipment) or, in the worst case, personal injury or a fire. Unfortunately the integrity of such installations is rarely questioned and the user often doesn't discover problems until it's too late. Back in the early days, equipment and appliances fitted with the two wire power plug were readily accepted to be safe from shock hazard because the metal housing was not connected to either wire of the line cord. In other words, the metal case is said to be "floating." Appliances such as toasters and irons are still supplied this way. Why not connect the neutral wire to the metal frame of such an appliance? This might seem like a good idea until you consider that the two-pronged plug or the receptacle might not be polarized. Now, you have a 50/50 chance of plugging the appliance in backwards, putting 120 volts on the metal frame of the appliance. A polarized plug has a larger prong on the neutral side so that it can only be plugged in one way. Many of today's appliances do have polarized plugs which will be discussed later in this article.

Whenever audio equipment is operated without a ground (floating chassis), strange things can happen. Under certain conditions the amplifier will be more susceptible to radio frequency interference (picking up radio stations or C.B. radio). Also, without a suitable ground, amplifiers sometimes "hum" more when the musician picks up his instrument and provides a "pseudo" ground through himself.

Both of these problems are, of course, very annoying. The only solution is to find a ground point to connect to the chassis, such as a water pipe. Sometimes this may just cause more problems, when what appears to be ground turns out not to be!

One of the problems with appliances and equipment which have a "floating metal case" is that a shock hazard exists if the case comes into contact with the hot wire. This so-called "fault condition" may happen in many ways with some of the more common causes being a "pinched" line cord, failure of installation systems, or movement of components due to shock or vibration which will cause the "hot wire" terminal to touch the case. Naturally, if for any reason the case does become "live," then a person touching it may be shocked if he is grounded. If this "hot chassis" is connected to another chassis or instrument by a typical shielded cord, then that chassis or instrument will become hot also. The entire purpose of the present three wire system is to provide a separate ground path which will effectively eliminate any possibility of shock.

Today's modern (U.S.) mains cable consists of three separate wires: **black**, **white**, and **green**. The green wire is always connected to the large ground pin on the plug, and the other (green) end connected to the chassis of the equipment.* The black wire is always considered to be the "hot wire," and as such, is always the leg which is connected to the switch and fuse. The white wire is always the neutral or common wire. The neutral wire is sometimes also wired to the power switch assembly but...it is rarely fused...

***Vital Safety Note!!! This applies only to U.S.A. products. Other nations have different color coding.**

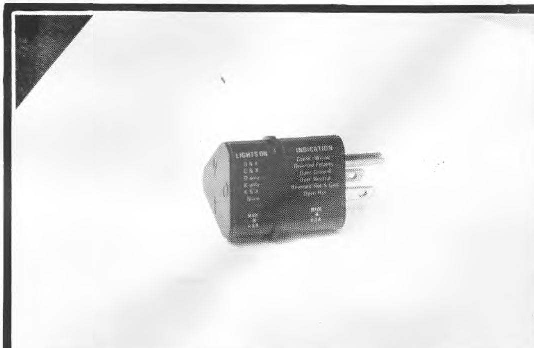
The integrity of the separate ground path is directly related to the quality of the chassis/green wire/ground pin combination. When the ground pin is removed, the separate ground path is destroyed and then fault conditions may cause shock hazards. Any modification of the 3 wire mains system completely eliminates the protection given by the three wire configuration. The integrity of the separate ground path is also directly related to the quality of the receptacle and the wiring system in the building itself. Today's three wire 120 Volt receptacle has some very important features which should be understood. First, the ground pin socket hole always has a green-colored screw for the ground wire attachment, and nothing but the "ground wire" should be attached to this. A correctly installed receptacle should always be vertical with the ground pin beneath the two parallel blade slots. In this position the **right** slot has a brass or **copper** screw for "hot wire" attachment, and the left slot has a **silver** or **chrome** screw for "neutral" wire attachment. The left slot is also larger than the right slot. This is the polarizing feature previously mentioned. Many television and stereo components as well as appliances are being fitted with polarized plugs which have a wider spade for the neutral line cord wire. Since the receptacle has a larger neutral slot, the manufacturer can connect the neutral side of the line cord to the chassis and be assured that the 50/50 chance of plugging the line cord in backwards is eliminated...that is, **if the receptacle is wired correctly!**

The standard wiring for the power receptacle is as follows: The black (hot) wire goes to the brass or copper screw which is connected to the right (smaller) slot. The white (neutral) wire goes to the silver or chrome screw which is connected to the left (larger) slot. The bare wire (ground) goes to the green screw for the separate ground path. The wiring system for the receptacle must be three separate wires (black, white, and bare or green, usually found in 2 wire + ground Romex and B-X) and must be connected correctly to the respective "hot," "neutral," and "ground" connections at the power source. **These connections should always be made be a qualified electrician!**

When the three wire line cord is plugged into a correctly wired receptacle, the wires in the line cord are: **black-hot**, **white-neutral**, and **green-ground**. The key word here is "correctly" wired. The whole system is no good if the receptacle is not wired according to accepted wiring standards of codes. The other key point is the use of a **third**, separate wire for ground. Often, older two-wire systems are "updated" to use the three wire receptacles. Since there is no third bare ground wire in the system and might be very costly to provide, many "electricians" simply connect the white wire to the neutral and ground connections at the receptacle. This is marginally better than no ground at all, but again defeats the purpose of the three wire system, which is to have a **separate** wire to ground for shock hazard protection. Also, at times the green "ground" screw is left open (unwired), which is the same as breaking off the ground pin on the line cord.

Even in most older buildings, one of the wires in a two wire system is "cold," but since the receptacles are not polarized the plug can be inserted either way. If you have a piece of equipment with one side of the line cord connected to the chassis, you have a shock hazard if the plug is inserted backwards. The usual "solution" to this dilemma is the "bypass capacitor" which is usually wired between the **white** (neutral) wire and the equipment chassis ground. A capacitor is a device which offers a relatively low impedance to high frequencies (such as generated by radio stations and C.B. radios) thereby providing them with a "short path" to ground in order to eliminate this type of interference. On the other hand, a capacitor offers a high impedance (AC Resistance) to low frequencies. Since, in the U.S., the frequency of the 120 volt supply is 60 Hertz (a relatively low value) the capacitor will not offer much of a signal path for this frequency. Thus, the bypass capacitor will minimize the shock hazard problem. To understand this, let's assume we are using equipment with a three prong plug and a three wire line cord on an older two wire system with a suitable ground adaptor which is not grounded. Plugging into this two wire system, the 50/50 chance could result in the line cord white wire being "neutral." In this case the bypass capacitor, which is connected to that white wire inside the equipment, will help minimize outside interference due to the grounding action of the neutral wire. Now let's take the opposite (and worst) case...the line cord white wire is "hot." In this case the bypass capacitor will be connected to 120 VAC...and the chassis will be "hot"...but...because of the capacitor characteristic, the current flow will be very small. There will be enough current to be felt, but not enough to cause bodily injury. If the user determines that he has a hot chassis with a system connected like this, the solution is to simply reverse the plug (if he can). A more modern solution to this problem is the use of an on-off switch like that found on many Peavey amplifiers. This switch flips the bypass capacitor from one side of the line to the other, so the user can keep the bypass capacitor on the neutral leg. A newer version of this is the three position ground reversal/lift switch found on many of the newer Peavey amplifiers. This switch not only reverses the polarity of the bypass capacitor, but the center position removes the capacitor from the circuit completely for use in most situations where the three wire power system in intact...In this case its presence may actually cause more hum and noise than if it were not there. This 3 way switch is a convenient way to remove it.

In order to assure that the bypass capacitor is the correct value and will protect the user from a serious shock hazard, a few audio manufacturers (including Peavey) perform what is referred to as a "high-potential" (hi-pot) test. During this individual product test, the integrity of the third wire ground is tested and then a very high voltage (1500 Volts) is applied between both line cord wires and the third wire ground. This test is **mandatory** to pass the requirements of CSA and other testing and approval agencies mentioned earlier and must be performed on each and every Peavey unit before it is shipped. This test also checks for faulty insulation and pinched wires inside the amplifier. If the equipment has passed this test, the user can usually be confident that the manufacturer has done all that can be done to assure a safe and shock-free product. It is then up to the user to operate the equipment safely and rely on the integrity of the power (mains) system being used. Fortunately, there are new devices available which can effectively check the integrity of the power plug. Peavey offers such a device called a "ground monitor" in our Accessory Program. We highly recommend its use each time a musician plugs into a strange power plug.



The following is a list of problem areas which should be avoided and suggestions to prevent a serious shock hazard:

1. Never use two wire "extension" cords.

2. Never use extension cords with non-polarized plugs or ones with broken-off ground pins.

3. Never break off the ground pin on electric equipment.

4. If necessary, always use a suitable ground adaptor...and if possible, ground that extra wire on the ground adaptor.

5. If no ground exists...find one...but make sure it is ground.

6. Always check the integrity of a "strange" power plug with a ground monitor device...if it checks "bad" don't plug into it!!!

7. Always use a qualified electrician to do all your "wiring."

NEXT ISSUE:

ON LOCATION: Musicom 84, Holland

OPTICAL DISK DRUM MACHINE

Video Focus: Getting Started in Video

Applied Synthesis

Practical Circuitry

Re-View ...and more!

Practical Circuitry

Build An Electric Drum Pad

by: Thomas Henry

Computer drums are great, but there may come a time when you'll want to arrange for a "hand-played" percussive part. For example, suppose you've written a song which requires a special bridge or intro in a strange meter. Getting the computer to play something like this might be a drag, so it might be easier to switch over to a hand-played passage. And if you do stage work with a band, the value -- both musical and visual -- of a real-time drum unit will be appreciated. Well, here's some good news! Adding an electric drum pad to your setup is not only easy but inexpensive as well, due to the miracle of "conductive foam." As a matter of fact, in this installment of **Practical Circuitry** we'll see exactly how to build just such a unit, with quality rivalling that of commercial drum pads.

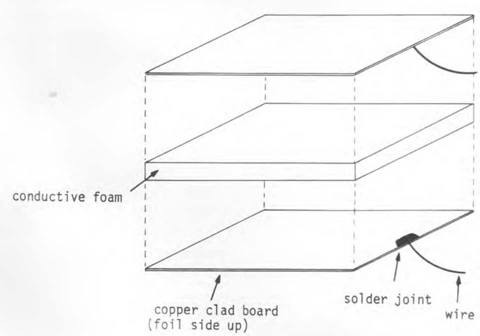
An Inexpensive Pressure Transucer. The electric drum pad consists of two main components: a pressure sensor and the supporting circuitry. Let's handle the mechanical aspects of this project first by examining how to construct a pressure sensor, then we'll cover the supporting circuitry.

Fig. 1 gives the basic idea. A piece of conductive foam is sandwiched between two copper clad circuit boards, each of which has a wire attached to establish the electrical connection. This creates a transducer whose resistance changes with pressure. Compress the pad and the resistance of the unit drops to around several hundred Ohms; release the pad and the resistance shoots back up to fifty kilohms or more. As we'll see in just a bit, the exact resistance values aren't important since the supporting circuitry responds to resistance **changes**, not absolute values.

work.

A pressure transducer, like that detailed in **Fig.** 1, can generate a variety of signals. For example, see my short design idea in **Electronics**, "Conductive Foam Forms Reliable Pressure Sensor", May 19, 1982, pp. 161, 163, which shows how to use this pad to generate simultaneous control voltage, gate and trigger signals, suitable for use with a synthesizer. Since we don't need all of these features for an electric drum pad, we'll concentrate more on getting a suitable trigger from the unit whenever it is struck by a drum stick. If this sounds like an exciting project to you, then let's get cracking and build the pressure sensor, since it forms the heart of the electric drum

fig.1

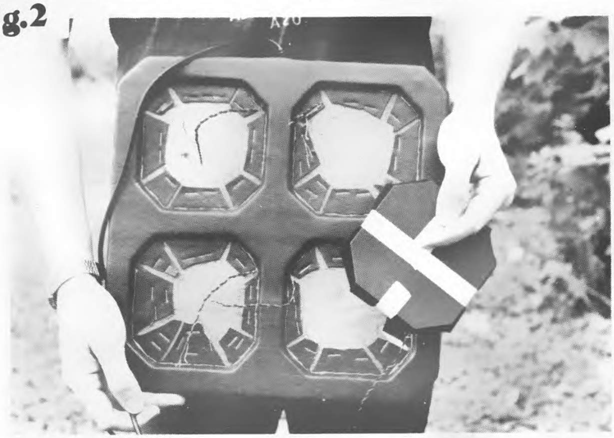


INEXPENSIVE PRESSURE TRANSDUCER MADE FROM COPPER CLAD BOARDS AND CONDUCTIVE FOAM: EXPLODED DIAGRAM

You may be wondering where you can find conductive foam which has this magical property of changing resistance under pressure. Surprise -- you see it every day! The foam employed by this project is the sort commonly used to pack CMOS integrated circuits because of its conductive properties. You know the type; it's a rather coarse, jet black substance. By keeping all of the IC pins at roughly the same potential, static electricity (which is the natural enemy of CMOS) doesn't have a chance to do its dirty pad.

Making an Electric Drum Pad. Any musical instrument should be attractive and rugged, so think carefully about how you will fabricate the sensor. A number of implementations are possible; here's how I did it. I decided to mount all four sensors in a single unit to make the instrument tight, sturdy, and easy to carry in case I decided to use it on stage. Another "must" was that it should be easy to connect to the supporting circuitry, thus avoiding myriad plugs and wires hanging all over the place. As it turns out, these features were simple to achieve.

fig.2



Refer to **Fig. 2**, which shows the construction of the quad sensor package in progress. Start by cutting out two pieces of plywood sufficient in size to house four pressure sensors in a two-by-two array. One piece of the plywood should be left intact, but cut four holes in the other to accommodate the four pressure sensors. You will note from **Fig. 2** that I used a hexagon shape both for the plywood pieces and for the individual pads. There's no particular reason for this other than it adds a bit to the aesthetics of the project.

After fabricating the two pieces of plywood route out four channels, each about 1/4" wide, in the wood. These channels allow the connecting wires of the pressure sensors to be brought out conveniently to a single termination. The wires run through the wood in much the same way that electric guitar pickup wires run to their associated volume and tone controls. After creating channels for the wires, string four twisted pairs through their respective openings, and terminate them at the top of the unit. Examine **Fig. 2** carefully; note that a single cable emanates from the top of the drum pad. We'll have more to say about this a little later.

When all of the wires are in place, apply a liberal dose of carpenter's glue to the two pieces of plywood and secure them together with small brads or tacks. At this point, the drum pad package will be a solid single piece, with the wires already running through the wood to the appropriate openings.

The unit can now be painted or covered in a variety or ways, but covering the package with vinyl fabric gives very attractive results. This sort of material is available at any fabric or department store. To apply vinyl, spread liberal amounts of carpenter's glue to the back surface of the fabric, then stretch it over the wood. Flip the drum pad onto its face and then secure the fabric to the back side with a staple gun. (This back surface will be covered in a later step, so don't worry about the ugly appearance of the staples just now.) You will have to cut and tuck the corners at various times, but this will be obvious to you when you start to tackle the project. After stapling the back flaps in place, turn the drum pad over so that you're looking at the front again, then cut, staple and tuck the access holes as required, making sure to avoid the wires you installed earlier. Again, these staples and flaps will be covered later on, so don't be alarmed at the seedy appearance.

At this point, we've caught up with the picture in **Fig. 2**. Careful examination of this photograph reveals the vinyl covering, various flaps and staples, and the wires emanating from the access holes in anticipation of loading the four individual pressure sensors.

On the left hand side of **Fig. 2** you will see one of the pressure sensors in a partially assembled state. Essentially, the foam has been sandwiched between the two pieces of copper clad board and the connecting wires soldered on to their respective plates. To keep the unit from slipping and sliding around, strap two pieces of masking tape around the affair. When strapping the sensor together, make sure that the plates are snug against the foam without compressing it too much.

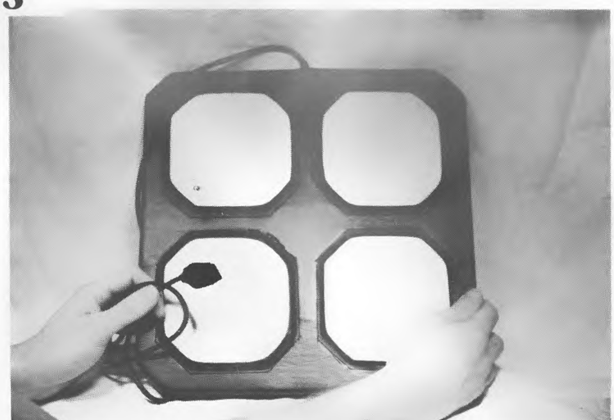
By the way, Radio Shack is a good source for conductive foam. They stock a 5" by 5" piece (#276-2400) which is the perfect size for a single pressure sensor. The price is under a buck, so you can tell that this is a very economical project!

To add to the reliability of the electric drum pad, scrub the copper clad boards thoroughly with some 000 steel wool before assembly. You might even want to consider tinning the copper surfaces as an extra precaution against corrosion. I was able to get our local university to do this for me, but you'll be glad to know that a number of mail order houses now stock a chemical which will perform this step at home -- <u>just be sure to follow the directions and obey all safety warnings.</u>

Okay, we have the main holder board done and the individual pressure sensors partially assembled. Let's try to tie up the loose ends now. The sensors need to be protected somehow and at the same time given a surface conducive to being struck by a drum stick. I cut out pieces of vinyl, with appropriate flaps, and actually wrapped each sensor in its entirely. If you go this route, use industrial strength rubber cement (like Weldwood) to secure the vinyl to the pad; this type of glue adheres to just about any surface. Again, each drum pad is covered in vinyl, with all of the various flaps wrapped under the sensor as required. Don't worry about the ugly appearance of the side with the flaps; we'll be sure to glue it into its receptacle with this side down.

After covering the pressure sensor, solder the two leads to the pair of wires emanating from one of the receptacles. Cover any exposed surfaces of the wires with electrician's tape. Now, coat the inside of the receptacle with a thick layer of silicone bathtub caulking, and press the pressure sensor down into it, dressing the wires underneath. Fill any cracks between the sensor and the board with additional caulking. Repeat this for the other three pads. **Fig. 3** shows the final result.

fig.3



After giving the silicone sufficient time to dry (several day, since the lack of air underneath increases the drying time somewhat), outline each pressure sensor with a strip of rubber tape. Rubber tape is available at most department stores. Sometimes the glue on this is rather inferior, so if you need to, spread a little more of that industrial strength rubber cement on the tape strips. Use a razor knife to trim up any unsightly seams or edges.

This may sound like a lot of work, but if you've successfully completed these steps, then you will have a unit which is not only strong but also attractive. You will notice that all of these steps have insured that each pressure sensor is securely fastened in its receptacle, in such a way that the pad actually seems to be part of the surface. Again, refer to **Fig. 3** and examine it carefully, noting that the final product is very homogeneous in nature. By the way, I used blue vinyl for the main surface and yellow for the pressure sensors. The result is a decent looking instrument!

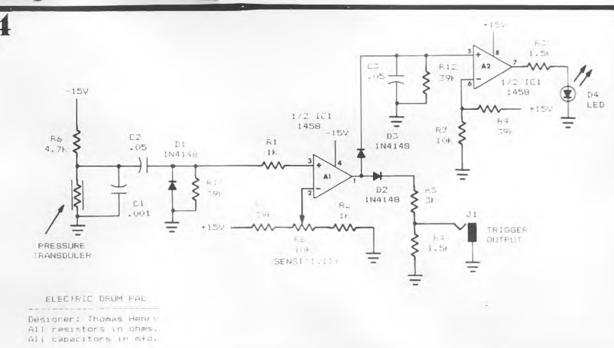
One final problem needs to be rectified. Recall that the back side of the drum pad is a mass of flaps and staples. We can't let that go unattended! Here's how to fix it up. Check the automotive section of your local department store and look for one of those rubber mats intended to sit on the floor of a car. Most stores carry several kinds, and they range from under a dollar to more that twenty dollars in price. Get the cheap one! (The fancy ones have all sorts of exotic carpet materials on them; we don't need that. All we want is a plain piece of rubber with a slightly pebbled appearance.) Now cut this piece of rubber matting to size and affix it firmly to the back surface of the drum unit, again using some industrial strength rubber cement. Besides covering up the unsightly staple and flap mess, the rubber provides a nice non-slip surface for the musical instrument.

Okay, we're done with the hard part. If you have followed all of these steps, then you are now the proud possessor of a complete, four-in-one drum pad. At this point we need to switch gears and come up with some sort of interface circuitry, but as you'll see, this is a snap.

How the Support Circuitry Works. Fig. 4 shows the schematic for the electric drum pad support circuitry. If you are building a quad drum pad, then you will need to repeat this circuit four times. However, all parts are inexpensive and this shouldn't impose too great a financial burden.

Refer to **Fig. 4** now. The pressure transducer is hooked up in series with R6 and the entire voltage divider is strung between the -15V supply and ground. At rest, the pressure transducer will have a resistance many times greater than R6, so the voltage at the tie point will be very close to -15V. When the transducer is depressed, however, this resistance drops considerably and forces the tie point towards ground. Thus, striking the pad with a drum stick creates a positive going trigger. Notice that C1 is wired in parallel with the transducer; this dumps any scratchiness and noise to ground, thus improving the reliability of the unit considerably.

fig. 4



The positive pulse generated by the transducer is AC coupled to the rest of the circuitry by C2. This serves to eliminate the negative bias, among other things. Diode D1 eliminates negative excursions of the pulse, while R11 acts as a load resistor for the voltage passed by C2.

Comparator A1 senses the pulse generated by the transducer and swings positive. Since the quiescent output of A1 is normally -15V, diodes D2 and D3 restrict the travel so that the output actually swings from 0V to +15 V. Notice that the threshold of A1 is set by potentiometer R8 (along with R10 and R2). This allows you to set the sensitivity of the electric drum pad to accommodate a variety of playing styles. The values of R8, R10 and R2 were selected under the assumption that the drum pad would be struck with a drum stick; change these if you have something else in mind. Although the circuit is quite tolerant of different types of foam and playing styles, you can increase the circuit sensitivity by lowering R2's value (or lower the circuit sensitivity by increasing R2's value). My prototype develops a pulse voltage of about +2V as measured at the junction of C2 and R11.

A1's output splits off in two directions. D2 sends the output to voltage divider R5 and R4. The output of this divider has an amplitude of +5V and an impedance of 1K. These are our old standards again. In addition, the pulse width will be somewhere in the neighborhood of 1 millisecond, although this may vary slightly depending on the setting of R8 and the force with which the pad is struck.

D3 couples the output to the peak detector composed of C3 and R12. In this situation, the peak detector serves as a pulse stretcher so that LED D4 turns on for a reasonably long period of time, thus giving positive indication that the pad has been struck.

Finishing Up the Project. And that's all there is to it! As mentioned, if you are building a quad drum pad (as illustrated in **Figs. 2 and 3**), then you will want to build four interface circuits. I used a D-9 plug to patch the pads to the support circuitry; this is the same type of plug used with joysticks on many personal computers. In fact, when my old VIC-20 joystick bit the dust, I clipped the plug and accompanying cable off and saved it just for this purpose. You can see this cable if you examine **Fig. 3** closely.

The D-9 plug can accept up to nine wires (hence the name), but we need only five here: four hot wires and one ground. When I wired up my unit I employed the same pinout as used by Commodore for their VIC-20 and C-64 computer joysticks. This gives the bonus of being able to plug a joystick into the circuit as well as the drum pad, thus adding the capability of joystick generated triggers to the synthesizer! If you want to go this route, check the owner's manual for your own computer and follow the pinout detailed there.

You will need a female D-9 connector to mount on the front panel. Fortunately, these are very easy to find (Radio Shack part number #276-1538).

After building the support circuitry, mount the entire unit behind a front panel. I found that a standard 1-3/4" by 19" rack panel was sufficient to comfortably house the D-9 connector, the four jacks, four pots and four LEDs, while giving a very professional appearance. Mount the circuit board behind the panel with some small angles and #4 hardware, and then complete the final wiring. After hooking up the power supply, plug in the drum pad and confirm that it works. Strike each of the pads and rotate the pots while noting the effect on the sensitivity. You will probably find, as I did, that each pad dictates an optimal setting for its associated pot. In general, you will simply set the sensitivity and leave it for the duration of a session.

As mentioned above, although various types of foam will have different resistances, and a hundred other gremlins might try to invade the project, the electric drum pad gives reliable operation since the support circuitry simply doesn't care about these parameters. It responds to <u>changes</u> in resistance and nothing more. Thus, in spite of the "quick and dirty" nature of the pressure transducer, the unit has proved extremely reliable and very easy for creating music. Best of all, it's a do-it-yourself project costing under $30...so there <u>is</u> an alternative to all those expensive drum pads!

CURRENT EVENTS

'Tell Them You Saw It In Polyphony'

All prices quoted are suggested retail prices, as supplied by the manufacturers.

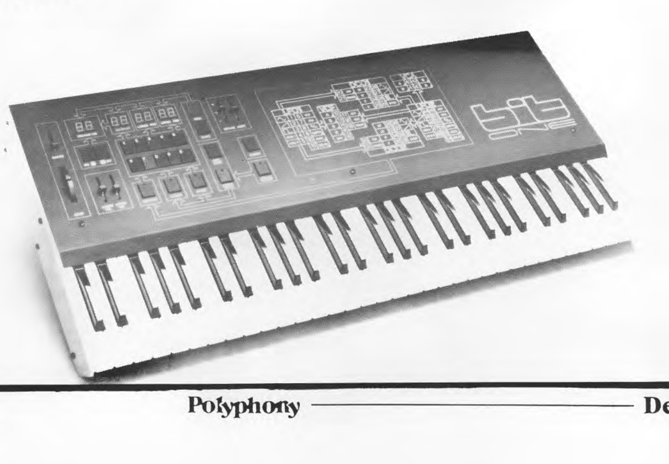
Interchangeable pickup guitar. The Player MDS-1 uses pickups that are factory-installed in plastic modules. These load into the guitar from the back, thus allowing pickup changes in seconds. Available in 1, 2, or 3-pickup versions; 22 fret neck; Kahler tremolo; full line of DiMarzio pickups available. Player Instrument Corp., PO Box 1398, Scarsdale, NY 10583-9398. Tel. 914/472-7805.

Keyboards. The six-voice Bit One ($1395) offers keyboard touch control over the VCFs, VCAs, LFOs, pulse width, and all 12 DCOs. Includes double/layering mode, stereo output, keyboard split, and MIDI. Digital Keyboards, 105 Fifth Ave., Garden City Park, NY 11040.

The Chroma Polaris ($1995) is a six-channel, analog-sound-with-digital-control synthesizer. Features include 132 programs, velocity sensitive keyboard, keyboard split, MIDI and Chroma interfaces, and sophisticated sequencer. Includes cassette and Apple II interface. Fender, 1300 E. Valencia Dr., Fullerton, CA 92631. Tel. 714/879-8080.

The Emulator II ($8645 with two disk drives) is a re-design of the original Emulator. Features include 17 seconds of high-fidelity sampling time; 1 Megabyte of disk storage; multiple-parameter, velocity-sensing keyboard; analog signal processing (VCF, VCA, LFO, etc.); digital signal processing (splice, mix, etc.); 8 channel sequencer with overdubbing and autocorrect; MIDI, SMPTE, and RS-232 ports; and updating via diskette. E-mu Systems, 2815 Chanticleer, Santa Cruz, CA 95065-1891. Tel. 408/476-4424.

New software. "Effects II" ($349) is an updated version of the Decillionix DX-1 sound sampling system for the Apple II. Additional features include sample playback over a five octave range in forward or reverse, external triggering, sequencing menu, and choice of Syntauri or Passport keyboard interface. Decillionix, PO Box 70985, Sunnyvale, CA 94086. Tel. 408/732-7758.



A second Sound Kit ($249) chip set is now available for the Oberheim DX. Sounds include three congas, three timbales, cowbells, tambourine/rimshot, ride cymbal, and "fat snare". Oberheim Electronics, 2250 S. Barrington Ave., Los Angeles, CA 90064. Tel. 213/473-6574.

More drums. Dynacord's new drum set consists of seven 13" pads ($150 each), one 20" pad ($230), and the Percuter 8-channel digital drum machine ($895) with interchangeable EPROM cartridges ($60 and $90). The Percuter may be driven from the pads or trigger microphones, sequencers, etc. Europa Technology, 1638 W. Washington Blvd., Venice, CA 90291. Tel. 213/392-4985.

Help is on the way! The Mini-Midi series includes 7 compact interface boxes for MIDI systems including a Channelizer (stamps channel info onto MIDI signals), Channel Filter (feed in 16 channels, select the one you want as output), CV Out (provides CV and gate in response to MIDI input), CV In (provides MIDI output for a mono CV and gate input), Drumu-Driver (slaves Drumulator to MIDI output), Sync 1 (generates MIDI based clock from 24, 48, and/or 96 pulse systems), and

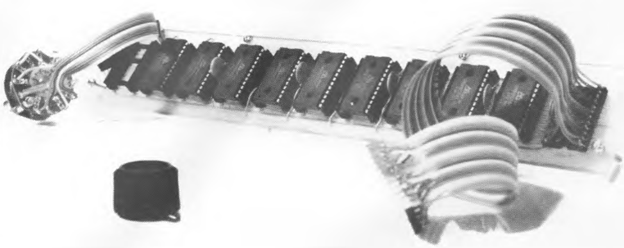
(continued on page 35)

The ColorTone two-octave membrane keyboard ($79.95) lets beginners play music in correct keys and tempos, choose different instrument sounds, and play along with pre-programmed melodies. Includes a defeatable "no-fault" mode that insures non-dissonant playing. Can run independently with a Commodore-64, or with Musi-Calc 1. Waveform, 1912 Bonita Way, Berkeley, CA 94704. Tel. 415/841-9866.

The MidiMate interface ($495) works with Atari 800 computers and provides up to 16 channels of MIDI sequencing. The package includes an interface box, cables, and software. Hybrid Arts, PO Box 480845, Los Angeles, CA 90048. Tel. 818/508-7443.

Veteran session player Denny Seiwell (Wings, Billy Joel) has released two rhythm program tapes for the LinnDrum ($35 each). One contains pop rhythms, the other reggae patterns. MusicData, 8444 Wilshire Blvd., Beverly Hills, CA 90211. Tel. 213/655-6356.

Expand Drumulator memory. The X5 ($275) is a factory-approved retrofit that increases the Drumulator's memory (song and segment) by 500%. The mod consists of a pre-assembled memory board that installs inside the Drumulator (and also leaves enough room for the J L Cooper three-kit mod), and a five-position bank select switch. SD Simpson, 11907 Brookmont, Maryland Heights, MO 63043. Tel. 314/739-6082. (Editor's note: I purchased and installed an X5 in my Drumulator. It was easy to install, works great, and now I can use the Drumulator live without feeling like I have to get a **disk** drive.)



Four Book Reviews by: David Doty

1

ACOUSTIC TECHNIQUES FOR HOME AND STUDIO, 2nd Edition, by F. Alton Everest (TAB Books, Blue Ridge Summit, PA)

According to its cover, Acoustic Techniques for Home and Studio is "an essential sourcebook for every dedicated audio enthusiast." While "audio enthusiast" is a label that might appropriately be applied to most Polyphony readers, I have serious doubts as to whether most of us will be able to make much use of the information presented here. If one has the time and money to build a small studio from scratch, or, at least, to make serious structural modifications to an existing room, then this book may prove to be a valuable resource. For the majority of us, who would like merely to improve our understanding of how room acoustics affect sound recording and reproduction, this book is at once too narrow in its focus and too vague in its overall organization to be of much use.

After the usual introductory chapters on basic musical acoustics, the book goes on to deal with standing waves in small rooms, the diffusion of sound, the control of interfering noise, and the decay of sound in small rooms. In each chapter, statistics, experimental data, anecdotes, and personal observations are intermingled in a way that makes it difficult to distinguish fundamental principles from irrelevant details. Charts and tables abound, but they are frequently separated by several pages from the text which they are intended to clarify. This results in the incessant flipping of pages in search of the correct chart. Editors and designers of such books would do a service to their readers by placing text and related graphics on the same or facing pages wherever possible. Throughout the book, the emphasis is on the use of proper building materials and construction techniques as a means of preventing or correcting acoustical problems. Plans for various types of walls, doors, windows, diffusers, absorbers, etc. are found scattered among the various chapters. It might have been better, both for the would-be builder and the general reader if the emphasis had been placed in a special section.

The final chapters include a case history of the design and construction of a Japanese radio studio, in which Mr. Everest participated, involving such exotic techniques as the pouring of floating floors, and including detailed accounts of the effects of the Tokyo fire code on materials selection. Probably the most useful chapter for the typical Polyphony reader is one which deals with the use of analyzers, equalizers, and test records and tapes. This technology, at least, is within economic reach of many readers. The book concludes with a photographic tour of studios around the world. Most of these, curiously, are Christian radio studios in Third World countries.

In conclusion, if you are looking for plans for a control room window or a sound-proof door, or need to know the absorption characteristics of various types of insulation, then you will probably find this book to be a useful tool. If you wish to gain a good general understanding of the behavior of sound in small rooms, you would be better advised to look elsewhere.

2

MUSIC SYNTHESIZERS, A MANUAL OF DESIGN AND CONSTRUCTION, by Delton Horn (TAB Books, Blue Ridge Summit, PA)

Longtime Polyphony readers and electronic music hobbyists in general are probably already familiar with the work of Delton Horn. In addition to his articles in Polyphony, he is the author of two other TAB books on electronic music, Electronic Music Synthesizers (TAB 1167) and the Beginner's Book of Electronic Music (TAB 1438). The purpose of the current work, according to the introduction, is to provide a collection of circuits from which a reader with a basic knowledge of electronics can construct a customized synthesizer to meet his or her specific needs. While this book does indeed contain numerous circuits representing just about every known synthesizer module, plus several that were previously unknown, I have some reservations as to whether one could create a professional-quality system using only the circuitry presented here. While I have not yet attempted to build any of the devices in this book, it appears to me that simplicity and low cost rather than precision and stability were the primary design criteria for many of these circuits. As an extreme example, consider a VCO consisting of two transistors, five resistors, and one capacitor. Nowhere in the text is anything said about the frequency response or stability of such a device. In general, circuits based on discrete components or general purpose ICs predominate, with custom electronic music ICs, such as the CES or SSM lines, appearing only occasionally.

Scattered through the book are a number of apparent typos or minor errors of fact that more diligent editing and proof reading might have prevented. There are also some curious ideas about musical technique, such as the following statement: "Filter banks generally aren't suitable for use on melodic lines because of the fluctuating tonal quality the fixed filters will produce." (Has Mr. Horn not heard of formants?) In addition, in discussing impedance matching between modules, Mr. Horn advocates a one-to-one match for maximum power transfer, rather than the generally accepted practice of feeding high impedance inputs from low impedance outputs in order to achieve maximum voltage transfer and fan-out capability.

On the positive side, the book includes a number of interesting circuit designs which I have not previously seen in print. Among these are several unusual oscillators, including staircase wave and programmable waveform generators, which may prove very useful as control voltage sources. There are also a number of dedicated sound effects circuits, such as gong, triangle, and crash cymbal generators which, with some modification, might be valuable additions to your analog percussion section. Also included are four analog sequencers, including one thirty-two step device which alternates sixteen forward and sixteen backward steps.

On a whole then, while this book seems to be good source for inexpensive circuits for the experimenter, or for unusual devices to add to an existing electronic music systems, it cannot provide the necessary constituents of a reliable, professional-quality synthesizer. Readers with such a goal in mind would be better advised to consult Barry Klein's Electronic Music Circuits (available through Polymart, see p. 41), Electronotes' Preferred Circuits collection (see Databank, p. 39, for Electronotes' address), or Thomas Henry's "Practical Circuitry" columns in Polyphony.

3

THE MUSICIAN AND THE MICRO, by Ray Hammond (Blanford Press, Dorset, U.K.; distributed in the U.S. by Sterling Publishing, 2 Park Ave., New York, NY 10016)

Hooray for the future! Cheap microprocessors will free us from all manner of musical drudgery and allow our musical genius to flow unimpeded. Programmable drum machines and digital sampling instruments are the most important musical developments in the last twenty years. These are the dominant idea expressed in The Musician and the Micro. Published in 1983, this book is primarily an inventory of the microprocessor controlled or assisted instruments commercially available at the time of its writing. If is serves no other purpose, this book at least illustrates how rapidly the field is developing, for despite its recent publication date, it contains no mention of such now-pervasive developments as MIDI or the Yamaha DX series.

The book is organized into ten chapters. Chapter One is devoted to a brief but not particularly concise discussion of what a microprocessor is and how it is changing the recording and performance of music. Chapters Two through Nine each cover one loosely defined type of instrument. The topics covered include drum machines, devices which "add-on" to personal computers, sequencers, live-performance keyboards and automated mixing consoles. Instruments singled out for chapters of their own include the Emulator, which Mr. Hammond identifies as a "hybrid" (of micro and instrument, rather than in the more usual digital and analog sense) and the Fairlight, which he presents as the crowning achievement of contemporary technology. Much to my surprise and delight, the chapter on keyboards makes mention of the Prophet Five's Just Intonation capability, and suggests that, as computer controlled instruments become more common, more musicians are likely to discover the superiority of Just Intonation.

All of this is presented in a very non-technical manner. Indeed, Mr. Hammond is at his best in describing the manner in which the user interacts with the instruments, rather than how the hardware and software go about doing their work. Believing wholeheartedly in the potential of computer power to advance musical evolution, he praises the manufacturers who put this power entirely at the user's disposal through fully programmable instruments, while condemning the more conservative companies who lock the power away behind unalterable presets. Although he advocates maximum user control over the power of microcomputers, Mr. Hammond seems in fact to be rather in awe of those who write computer programs; an attitude which might lead the beginner to believe that such accomplishments are the exclusive realm of those with advanced degrees in computer science. The fact is that he does not appear to have a really clear understanding, at the nuts and bolts level, of the way computers operate or how sound can be represented electronically. When he uses technical terms, it is often in such a vague way as to cast doubt on their precise meaning. At one point he suggests that all music composition software ought to be menu-driven, and that software that is not menu-driven is likely to be useful only to experienced programmers. Most experienced computer users (whether programmers or not) have discovered that while menu-driven programs are easy to learn, they can, once learned, be rather tedious to use. Consider the time consumed wading through a series of nested menus to reach a desired function, as opposed to simply pressing an appropriate control key.

Following the chapters dealing with specific instruments are short interviews with four musicians: Warren Cann, John Lewis, Hans Zimmer, and Peter Gabriel, each of whom discusses, in remarkably similar terms, how the Fairlight (or Linn Drum or Microcomposer) changed his life. Only Gabriel lets on that there may be a cloud behind this silver lining, touching on the musicians unions vs. Linn Drum controversy and the whole home-taping/software piracy can of worms.

The book concludes with a "Glossary of Jargon" which, unfortunately, is likely to do more harm than good. Mr. Hammond is on shaky ground when he attempts to give definitions of technical terms. His definitions are often sufficiently muddled as to confuse rather than enlighten the novice. For example: "Wave- As in sound wave. The shape of the graph (wave-like) that represents the frequency of a sound" (Polyphony readers know, of course, that a graph of a sound's waveform represents the sound's timbre, rather than its frequency). Perhaps the

(continued on page 38)

E-Drum Review

By: Craig O'Donnell

It's tough to review something too good, but really -- no one paid me off for this one! With the E-Drum, E-mu has capitalized on a head start in digital sampling/playback and designed a useful, idiotproof budget-priced digital Simmons rival that is also useful as a beatbox extender.

A black foam pad covers most of the top striking surface of this hefty 7" X 7" X 2" package. On the front: PITCH clocks your memory chip faster or slower; two SOUND SELECT buttons access up to four sounds; PITCH SENSITIVITY controls dynamic pitch modulation and "pitch domain"; DECAY, BASS, and TREBLE are self-explanatory. On the rear: banana jacks for DC IN and OUT, BATTERY ON button, a large slot for the drum cartridge which edge-connects to the PC board, TRIGGER (PITCH MOD) IN and AUDIO OUT 1/4" jacks.

An optional E-Drum power supply daisychains through the bananas to a full rack of five E-Drums on a standard Rototom stand. One E-Drum munches two 9V nicads pretty effectively, and the power supply might be an economy for those who own more than a couple. On the bottom is a tongue/slot arrangement for the stand adapter.

"ED" lived in our Chicago studio for a few months without ill effects. Engineers and musicians were floored by its flexibility and great sound. It was accidentally dropped, used as an ashtray, and as a beverage stand. Sucker's tough. No one blew the trig in or broke the drum sound cartridges; I can appreciate that. In fact, the E-Drum survives the "beer test". Properly stated, the BT is scientifically derived from Murphy's Law: If something survives (1) a dose of beer from a topsyturvy can, or (2) a musician who's seen too much brew, then (3) you've got quality gear.

To use, insert a sound cartridge, push SELECT buttons if appropriate, and pound or trigger away. There's a power-on LED but it's recessed to avoid flying sticks. More than once I was at an angle where I didn't see it glowing in its little foxhole.

I got six test cartridges. Each one can hold up to four drums, but only the blank-label cart had more than one. "SNARE 1" was cardboardy: useless. "GRAND PIANO" got a gig as a percussive pedal tone where accidental multiple triggering from a tape track gave it an odd shimmer. The max E-Drum decay time is too short to make a convincing piano, especially at lower pitches. Pulse triggers produce a sound like a piano bonked with a sledge-hammer. The attack is too abrupt; I like it.

"ROTO 2" and "ELECTRIC TOM" are totally convincing. The anonymous SNARE is wonderful, with a pitch, a thwack, and a rattle reminiscent of old Motown. The second sound, a "Simmons" snare, is indistinguishable from Bill Bruford's very own. "TYMPANI" is quite real except for the somewhat attenuated envelope. E-mu says carts cost $59.95 and that the catalog to date includes the above plus Tom 1, Bass Drum, Gong, and Rock Ride. You choose one free cart when purchasing an E-Drum module ($299 list); additional cartridges can be purchased as needed.

Digitization isn't obvious, especially in a mix. Heard alone, a slight amount of hiss (quantizing noise?) grows as the drum sound decays. Low batteries plus heavy bass boost can aggravate a slight "thumping" at the end of the envelope/clock cycle. Sonically, attack transients are there and some peak limiting can actually help! The natural feel must be due to the dynamic volume/pitch. The response follows tape tracks nicely within a slightly limited dynamic "window".

You can stick ED, or bop it with your hands like a conga. Mics, preamped items, and line levels all triggered the drum. Output varies with trigger as befits a dynamically-controlled device, so it might drive a line input here and need a preamp there. Any positive-going waveform acts as a trigger: With pulse trigger, the dynamics are missing. Drumatix triggers go in just fine, but my Korg (negative triggers) needed an inverter. Let's congratulate Dave Rossum and E-mu for developing a simple, almost universal trigger input!

Any isolated single-drum track recorded on tape has no trouble generating a digital cousin; but for the acid test I tweaked the signal chain to extract a snare beat buried in a stereo drum-overhead mix. The tape fed a low pass filter (highs and mids rolled off), which fed a parametric EQ tuned to find and boost a snare harmonic around 650 Hz, and ended with a dbx 118 expander set 2:1. A trigger extractor would have been helpful! After a few hours juggling levels and EQ I got a clean track laid that paralleled the original snare, so IT CAN BE DONE.

PAIA's "The Drum" sensor puck works great. Dynamic, cheap condensor, and Radio Shack $39 PZM mics gave a discernible but muzzy trigger. I imagine a gated, close-miked drum will trigger cleanly; I tried (without real success) to trigger from handclaps and shouts on one tape.

About external pitch modulation: When pounding ED, higher Pitch Mod knob settings didn't exaggerate the pitch-bending too much. The harder you hit, the higher the initial pitch. Simple envelope stuff.

When the external modulation feed gets used, the sensitivity knob has a very different effect. Mod CVs must be ±5V, and the 0V CV setting will neutralize modulation at the pitch setting you twist to. Pitch sensitivity interacts a great deal; as you raise the Sensitivity the "pitch domain" tends to rise. Deep sounds like kick-drum settings occur with the knob fully OFF (CCW). Top CW tends towards pitch sameness. Middle settings vary in effect. This control bias is fine because lowered percussives sound natural, just "bigger". Fast clocked percussives sound phony and gimmicky. This must be a psychoacoustic experience and intuition working in sync: Nobody believes a 4-inch tympani! It's play-as-you-go, which I like since you discover nifty sounds just fiddling. That the controls become more interactive in the mode best suited to studio work is probably serendiptious, but someone at E-mu might be really clever.

The modulation will take sample-and-hold, envelope, and mod osc if you'd like to apply randomness or create special effects. You could sequence a melody. I had a cybernetic Buddy Rich playing an immense Roto set by triggering rapidly and using a 0.5 Hz falling sawtooth to modulate pitch. Sounded like George Jetson's futuristic drumpanel on the "Eep Opp" episode! Low-frequency sawtooth or triangle will make toms and snares sound DDLed. Tympani makes an alien spacecraft by using a falling sawtooth at about 5 sec/Hz on the trigger clock and pitch mod. Some dum-dum will program "Sketches of Spain" or "The Planets" in dogbarks when the Dog Woof cart comes out. It's that versatile.

The E-Drum is a great as-is drumkit item or synth/beatbox expander module. Every live or studio engineer needs one to clean up those problem mixes. I'd like to see four sounds per cart at the advertised price in two types: Selection (like snare/electrosnare/tom/electrotom) and Special (snare miked at different ambient distances, or via four different microphones). Third-party drum sound "software" will be the key. Have they learned from Apple?

The price is a little steep for me -- five modules plus power, rack, and Anvil cases must run nearly $2000. Still, you end up with something more flexible, economical, perhaps even more resaleable than other drum kits. I do have some user feedback: The two 9V batteries fit too snugly; a jeweler's Phillips screwdriver is de rigeur to remove the very tiny screw and pry out the batteries. Lighted Select buttons or LED indicators might be useful so musicians on a dark stage could see what sound is up without feeling about or pounding a few times to hear. A defeatable retrigger suppressor would be handy to keep the trigger disabled for a miniscule period of time (like 1/12 second) after an initial hit; this would help speed up E-Drum salvaging operations such as snare drum dredging.

In my opinion, "ED" is worth running out and buying. As an add-on to analog drum machines it cannot be surpassed. I want a whole ED family, with: More carts, a tiny beatbox-trigger-only version, a compact Synsonics-type four sound striker. My Drumatix, PAIA "The Drum" synth-kick, "Snare+" and E-Drum gave me so many sonic options cutting beat tracks I sat there muttering "Huh? What? Linn? Simmons? Drummer?"

re-view

(continued from page 5)

Port Said **Traveller's Companion** (cassette). Tara Cross & Stefan Tischler **Searchlight & Torch** (cassette). Keeler **Planet of Lovers** (cassette). New Yorkers Tischler and Keith "Keeler" Walsh prove there's still a lot of life left in lo-tech. Often we tend to think the latest megatoys somehow abet the creative process, but quite the opposite is possible. 132 West 24th Street, New York NY 10011.

Talking Heads **Stop Making Sense** (Sire 25121-1E). This is not a live album -- it's the soundtrack to a movie about a concert (they just had a live album, right?). An excuse for elaborate staging and digital recordings of old songs -- and they're not in a "dry spell."

Mike Gould **The Hy James Rap** (flexidisc). The Christmas Party Joke That Wouldn't Die -- a tongue cramping rap about the chores and foibles of the employees of this Michigan audio dealer. Now it's a novel promo gimmick for Hy James, 24166 Haggerty Road, Farmington Hills MI 48018.

Apollonia 6 **Apollonia 6** (Warner Bros. 25108-1). For my money Apollonia Kotero was the best thing about "Purple Rain" -- campy, cocky and electric, much like this album. With her are the other 2/3 of Vanity 6 and backing by Prince's band **The Revolution**. It is more than slightly cheeky, but the tongue's in there too.

Vanity **Wild Animal** (Motown 6102). Her first solo since being replaced by look-alike Apollonia. Bill Wolfer wrote the music and plays synthesizers and drum units while Vanity sings her own lyrics. She may be stretching herself a bit here and there, but I'd take this over Prince's extravagant soundtrack.

PLUCKED STRING OSCILLATOR USING THE 6502 MICROPROCESSOR

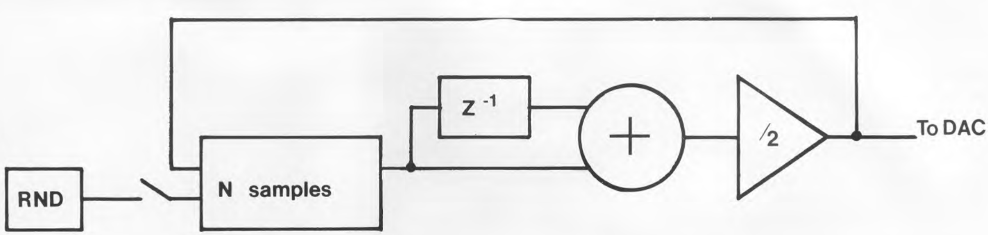
by: Jack Deckard

The summer 1983 issue of Computer Music Journal presented an article on the technique of simulating plucked string and drum sounds using inexpensive microprocessors. This basic technique is called the Karplus and Strong Plucked String Algorithm after its authors, Kevin Karplus and Alex Strong. This article presents the assembler code for implementing the basic algorithm on a 6502 microprocessor. Also covered will be some general techniques on controlling the Karplus Strong oscillator (hereafter referred to as the K. S. Oscillator), including a short program to implement the K. S. Oscillator on a PAiA 8700 computer.

Brief Review of the K.S. Algorithm. The algorithm is a modification of the circular wavetable scanning technique for producing sound. As each sample is read from the circular wavetable, it is placed in a one sample delay line. The current sample is averaged with the previous sample (the delayed sample) and is then sent to the DAC and written back into the wavetable. This produces a smooth decaying amplitude envelope, and a timbral envelope in which the higher harmonics fade away more quickly than the lower ones. To maximize the advantages of the timbral envelope characteristics, the wavetable is initially filled with random numbers. As the random numbers are being repeated, with a small evolving change brought about by the averaging, noise is not produced. **Fig. 1** shows a block diagram of the K.S. Oscillator.

The timbre of the sound is greatly influenced by what is initially placed in the circular wavetable. To keep successive notes played on the same key from sounding the same, two independent pointers are used to load the oscillator's wavetable. These pointers reset at different times, giving a different string of random numbers with each load. In addition the loading subroutine takes into account the size (pitch) of the table it is going to load, and chooses from one of four random tables to load. This means that you can change the basic timbre across the keyboard. As written, the highest notes (wavetable size of 32 to 255) will

FIG.1



load from the random table at \$B000 to \$BFFF. The next random number table to load from is at \$A000 to \$AFFF, and is used for oscillator wavetable sizes of 256 to 511. Two more tables at \$9000 to \$9FFF and \$8000 to \$8FFF are used for each increment of 256 oscillator wavetable samples. More will be said later about filling these random tables.

In my implementation of the K.S. Oscillator, an external controller supplies the pitch data. This is because in sequencer type implementation, note timing is difficult to control. If one uses a counter in software which is decremented every time a sample is sent to the DAC, the counter range must be over 16 bits to really be useful. A 20 bit counter would be about right for note timing but would significantly lower the sample rate of the K.S. Oscillator (in eight bit machines). One could decrement the counter every time the wavetable pointer is reset to the beginning, but this would require calculating different timing values for each different pitch value. For example, the number to count down for a quarter note of C in the fourth octave would not be the same as a quarter note of C in the third octave. A hardware counter for note timing could be built and would be a viable alternative if a suitable external controller is not available. I chose to use my PAiA 8700 computer, equipped with an EMC expander board, for the external controller. Note that a MIDI equipped keyboard could also be used for control. In this case the K. S. Oscillator would have to perform MIDI data to K.S. oscillator pitch data conversion.

The assembler code for the K. S. Oscillator is written for page zero and uses two tricks to make it run as fast as possible. First, it uses a trick called instruction self-modification. Normally when the 6502 processor uses an instruction which addresses page zero, there is a separate buffer in page zero set outside the program. I put this buffer (WAVPTR) inside the program to save a few clock cycles. The second trick is the use of an invalid instruction LDAX. There is no documentation from any of the 6502 manufacturers on this instruction, but it will execute properly. (Check the references at the end of this article for more information on some of these invalid instructions.) The assem-

```
 10                    ;HARDWARE LABELS
 20                    ;
 30 DF01=              DAC    = $DF01
 40 C70F=              CRB    = $C70F  6821 IC
 50 C70E=              PRB    = $C70E
 60 C70D=              CRA    = $C70D
 70 C70C=              PRA    = $C70C
 80                    ;
 90                    ;OSCILLATOR CODE FOR KARPLUS AND STRONG
100                    ; ALGORITHM FOR PLUCKED STRING TIMBRES
110                    ; VERSION 1.0  8700 KEYBOARD CONTROL
120                    ;
130 00BF=              LDAX   = $BF   LOAD A & X ABSOLUTE,Y ADDR MODE
140 8000=              RNDTBL = $8000 - $BFFF RND DATA TABLE
150 7C00=              WAVTBL = $7C00 - $7FFF CIRCULAR NOTE BUFFER
160                    ;
170                    ;ZERO PAGE BUFFERS
180                    ;
190 0060=              DELAY  = $0060 DELAY BUFFER VOICE 1
200 0097=              FRND   = $0097
210 009A=              SRND   = $009A
220 00A3=              RNDMAX = $00A3
230 00DD=              WL1    = $00DD
240 00DF=              WL2    = $00DF
250                    ;
260 0070               *=$0070
270                    ;
280                    ;NEXT NOTE - FETCH LOW & HIGH BYTES OF
290                    ; BEGINNING OF CIRCULAR NOTE BUFFER. USE HIGH
300                    ; BYTE TO CALCULATE WHICH 4K RANDOM TABLE TO
310                    ; LOAD CIRCULAR NOTE BUFFER FROM.
320                    ;
330 0070 ADOCC7 FILL   LDA PRA         GET PITCH BYTE LOW
340 0073 85DE          STA WL1+$1      SAVE IN INNER LOOP
350 0075 A8            TAY             & IN Y INDEX
360 0076 A599          LDA FRND+$2     PREPARE TO CALCULATE
370 0078 290F          AND #$0F        PROPER RNDTABLE TO
380 007A 8599          STA FRND+$2     LOAD FROM
390 007C EA     F1     NOP
400 007D AD0DC7        LDA CRA         PITCH HIGH BYTE READY?
410 0080 10FA          BPL F1          LOOP IF NOT READY
420 0082 AD0CC7        LDA PRA         GET PITCH BYTE HIGH
430 0085 85E0          STA WL2+$1      SAVE IN INNER LOOP &
440 0087 859C          STA SRND+$2     RANDOM DATA STORE LOOP
450 0089 0A            ASL A           CALCULATE RANDOM DATA
460 008A 0A            ASL A           TABLE TO LOAD INTO
470 008B 0A            ASL A           CIRCULAR NOTE BUFFER
480 008C 0A            ASL A
490 008D 0599          ORA FRND+$2
500 008F 29BF          AND #$BF        RANGE $80-$BF
510 0091 8599          STA FRND+$2     SAVE IN STORE LOOP
520 0093 090F          ORA #$0F        CALCULATE MAX HIGH
530 0095 85A4          STA RNDMAX+$1   BYTE OF RNDTABLE
540                    ;
550                    ;LOAD CIRCULAR NOTE BUFFER WITH RND NUMBERS
560                    ;
570 0097 AD0080 FRND   LDA RNDTBL      REFILL CIRCULAR
580 009A 99007C SRND   STA WAVTBL,Y    BUFFER WITH RND #
590 009D E698          INC FRND+$1     RND # POINTER + 1
600 009F D008          BNE S1          BRANCH IF NO OVERFLOW
610 00A1 E699          INC FRND+$2     INC RNDTBL HIGH BYTE
620 00A3 A98F   RNDMAX LDA #$8F
630 00A5 2599          AND FRND+$2
640 00A7 8599          STA FRND+$2
650 00A9 C8     S1     INY
660 00AA D0EB          BNE FRND
```

bler code should be compatible with any 6502 machine with memory up to $BFFF except for the loading of the data from an external controller. I load the data through a 6821 chip located at addresses $C70C to $C70F. If you have a 6821 or 6520 chip in your system and are going to use it to receive the control data, you only need to change the addresses relative to that chip. A 6522 chip could be used almost as easily. Note the 6821 chip has already been set up with port A serving as the input port in the pulse handshake mode outside of the assembler code presented. You will also have to change the address of the DAC to where it is in your system.

Normally the code for the K.S. Oscillator will first be loaded to some temporary location outside of page zero and then be relocated to page zero just before running it. A short assembler code program to do this might look like this:

```
       LDY #$60       ;set up index
                       pointer
STLOOP LDA TEMP,Y     ;load osc code
                       from tempo-
                       rary location
       STA $0000,Y    ;save in page
                       zero
       INY            ;increment
                       pointer
       BNE STLOOP     ;loop until
                       done
       JMP $00DD      ;jump to label
                       WL1 in osc
                       program
```

Note: the K.S. oscillator program is entered at address $00DD (label WL1) which insures that the wavetable pointers are set and playing a very low note, thus giving you audible feedback that the oscillator loop is functioning properly.

The math needed to calculate the size of the wavetable for a certain pitch is:

$$Ns = (Fs/P) - 0.5$$

where **Ns** is the number of samples in the wavetable, **Fs** is the sampling rate, and **P** is the desired pitch of the note in Hertz. The following finds the sampling rate:

$$Fs = 1/(Cs/M)$$

where **Fs** is the sample rate, **Cs** is the number of machine clock cycles per sample (50 in this program), and **M** is the clock speed of the microprocessor. You will need to

```
00AC E69C              INC SRND+$2      INC HIGH BYTE NOTE BUFFER
00AE A980              LDA #$80         COMPARE MAX HIGH BYTE
00B0 C59C              CMP SRND+$2      ADDR WITH CURRENT
00B2 D0E3              BNE FRND         LOOP IF CURRENT < MAX
00B4 4CDD00            JMP WL1
                ;
                ;PLAY WAVEFORM LOOP
                ; BALANCED, 50 CLOCK CYCLES
                ;
                ;                       # OF CLOCK CYCLES/INSTR
00B7 EA         VLOOP  NOP              2 BALANCE TIME OF INNER
00B8 EA                NOP              2 LOOP
00B9 EA                NOP              2
00BA EA                NOP              2
00BB EA                NOP              2
00BC EA                NOP              2
00BD AD0DC7     VLOOP1 LDA CRA          4 CHECK FOR NEW NOTE
00C0 30AE              BMI FILL         2/3 BRANCH IF NEW NOTE
00C2 A500              LDA $00          3
                ;
00C4 BF         VLOOP2 .BYTE LDAX       4 LOAD SAMPLE IN A & X
00C5 007C       WAVPTR .WORD WAVTBL       WAVTBL + Y INDEX
00C7 18                CLC              2 PREPARE TO ADD
00C8 6560              ADC DELAY        3 ADD CURRENT & DELAY BYTE
00CA 6A                ROR A            2 AVERAGE BYTES
00CB 8D01DF            STA DAC          4 UPDATE DAC
00CE 91C5              STA (WAVPTR),Y   6 SAVE IN WAVETABLE
00D0 8660              STX DELAY        3 NEW BYTE TO DELAY
00D2 C8         IW     INY              2 CIRCULAR POINTER + 1
00D3 D0E2              BNE VLOOP        2/3 BRANCH Y NO OVERFLOW
00D5 E6C6              INC WAVPTR+$1    5 HIGH BYTE POINTER + 1
00D7 A5C6              LDA WAVPTR+$1    3 CHECK HIGH BYTE POINTER
00D9 C980              CMP #$80         2 = TO MAXIMUM LENGTH?
00DB D0E0              BNE VLOOP1       2/3 NO, BRANCH
00DD A000       WL1    LDY #$00         2 RESET LOW BYTE TO START
00DF A97C       WL2    LDA #$7C         2 RESET HIGH BYTE TO START
00E1 85C6              STA WAVPTR+$1    3
00E3 4CC400            JMP VLOOP2       3 & KEEP LOOPING
                ;                       ---
                ;                       = 50 CLOCK CYCLES MAXIMUM
                ;                         TIME AROUND LOOP
```

```
                ;PROM FIRMWARE LABELS
                ;
0D4E=           LOOK   = $0D4E
0D21=           INIT   = $0D21
                ;
                ;HARDWARE ADDRESS LABELS
                ;
0A10=           ORB2   = $0A10  6522 IC
0A1C=           PCR2   = $0A1C
0A1D=           IFR2   = $0A1D
0820=           DSPY   = $0820  8700 DIGIT DISPLAY
                ;
                ;KARPLUS AND STRONG STRING ALGORITHM
                ; 8700+E CONTROL KEYBOARD PLAYER
                ;
                ;PITCH INFORMATION (TABLE LENGTH) IS SENT TO
                ; OSCILLATOR CPU VIA ORB2 PORT ON 8700
                ; EXPANDER BOARD. KEYBOARD IS SCANED VIA LOOK
                ; ROUTINE IN PROM AND CORRECT PITCH DATA TO
                ; BE SENT IS DERIVED VIA A TABLE LOOKUP
                ; SCHEME. ORB2 HAS ALREADY BEEN INITIALIZED
```

subtract the number of samples from the top of the wavetable buffer ($7FFF) to find the actual number with which to load the oscillator for the desired pitch. Thus, with a machine clock rate of 1MHz, the sample rate would be 20,000 samples per second and the number sent to the K. S. Oscillator for a desired pitch of 440 Hz would be $7FD2 (in hex). The low byte $D2 should be sent first, followed by the high byte $7F. In the 8700+E control program I have included, all pitch information was calculated for an oscillator machine clock rate of 1.0MHz.

How good does the K.S. oscillator sound and how well does it mimic plucked strings? (All my comments are based on a sample rate of 33.3k samples per second except where noted.) The low to mid range (C2 to C4) sounds bright and buzzy. There is plenty of decay time in this range. In the upper octaves, the string simulation is much more realistic. The dynamic resolution of eight bits modulates the final fadeout in an unnatural but not uninteresting way. With a sampling rate of 61.7k the higher notes sound incredibly like a mandolin. I like to roll off all the frequencies beyond 10k for the best string simulation. Note that all the above comments relate to using Basic's RND function to fill the random number tables. I have also tried loading the random number tables with digitized waveforms. Using a segment of digitized speech, the oscillator takes on the sound of a filter with high resonance. The resonance frequency follows the pitch perfectly. I've also tried loading the random number tables with a digitized FMed oscillator (sine wave out) which resulted in very little timbral movement. It would appear that using different distributions of random numbers is the best thing with which to experiment for filling the random number tables. Karplus and Strong recommended using a two level random number table. To do this one would first fill the table with random numbers and then go back and change the sign bit (the most significant bit) so that approximately one half of the numbers are positive. I have not yet tried this but the result is supposed to be that the notes are about 5 dB louder.

```
                     ; (PORT B = OUTPUT, CB1 NEGATIVE ACTIVE EDGE,
                     ; CB2 OUTPUT ALWAYS HIGH) BY DOWNLOAD PROGAM.
                     ;
0200=                PTCHL  = $0200 PITCH LOOKUP TABLE LOW
0300=                PTCHH  = $0300 PITCH LOOKUP TABLE HIGH
00DF=                KTBL   = $00DF MUS1 FROM LABEL
00DF=                LAST   = $00DF LAST NOTE DOWN
                     ;
                     ;*=$0000
                     ;
0000 20210D START  JSR INIT   CLEAR KTBL, LAST
0003 A9AE          LDA #$AE   PULSE OUTPUT CB2, NEGATIVE
0005 8D1C0A        STA PCR2   ACTIVE EDGE CB1
0008 A9FF          LDA #$FF   MAKE SURE HANDSHAKE FLAG
000A 8D1D0A        STA IFR2   IS CLEARED
                     ;
000D 85DF   ZLOOP  STA LAST   CLEAR LAST IF NO KEY DOWN
000F 204E0D LOOP   JSR LOOK   SCAN 8782 KEYBOARD
0012 A5E7          LDA KTBL+$8 NOTE DOWN?
0014 F0F7          BEQ ZLOOP  NO, BRANCH IF ZERO
0016 C5DF          CMP LAST   SAME AS LAST KEY DOWN?
0018 F0F5          BEQ LOOP   YES, BRANCH
001A A8            TAY        USE FOR TABLE LOOKUP
001B B90002        LDA PTCHL,Y LOOKUP LOW BYTE
001E 8D100A        STA ORB2   SEND OVER INTERFACE
0021 EA     T1     NOP
0022 AD1D0A        LDA IFR2   CHECK FLAG FOR DATA TAKEN
0025 2910          AND #$10
0027 F0F8          BEQ T1     LOOP TILL TAKEN
0029 B90003        LDA PTCHH,Y LOOKUP HIGH BYTE
002C 8D100A        STA ORB2   SEND OVER INTERFACE
002F EA     T2     NOP
0030 AD1D0A        LDA IFR2   CHECK FLAG FOR DATA TAKEN
0033 2910          AND #$10
0035 F0F8          BEQ T2     LOOP TILL TAKEN
0037 8C2008        STY DSPY   SHOW KEY DOWN
003A 84DF          STY LAST   SAVE IN LAST
003C 4C0F00        JMP LOOP   & KEEP CHECKING KEYBOARD
                     ;
                     ;PITCH PARAMETER LOOKUP TABLES
                     ; C2 TO C5 TUNED TO 1.00 MHZ CLOCK 50 CLOCK
                     ; CYCLES PER SAMPLE
                     ;
0250                 *=$0250 ;TO $0274 LOW BYTE
                     ;
0250 CD              .BYTE $CD,$DE,$EF,$FE,$0C,$1A,$27,$33
0251 DE
0252 EF
0253 FE
0254 0C
0255 1A
0256 27
0257 33
0258 3E              .BYTE $3E,$49,$53,$5D,$66,$6F,$77,$7E
0259 49
025A 53
025B 5D
025C 66
025D 6F
025E 77
025F 7E
0260 86              .BYTE $86,$8C,$93,$99,$9F,$A4,$A9,$AE
0261 8C
0262 93
0263 99
0264 9F
0265 A4
```

→

PLUCKED STRING...

Now, if you don't have anything to use for external control, but would still like to hear the K.S. oscillator in action, here's what to do. Enter the oscillator assembler code into your 6502 assembler. Replace the DAC label address with the address relevant to your DAC. Change the address for label CRA to any non-zero page address, and store #$00 at that address when you run the oscillator program. Calculate the pitch you want and change the program so that at label WL1 you load the Y register with the low byte of the pitch and at WL2 you load the accumulator with the high byte of the pitch. Assemble the program somewhere in unused memory and save it. Fill addresses $8000 to $BFFF with random numbers. Reload the oscillator program, relocate it to page zero and execute it (jump to label WL1). You will only get one note, but at least you can hear the K. S. Oscillator sound and decide whether it merits getting an external controller.

I hope you enjoy this program and learn something from it. I am semi-seriously thinking of wire-wrapping up four dedicated K.S. oscillators. If I do, you can be sure I'll share the schematic with you in **Polyphony**.

References

Karplus, K., and A. Strong. 1983. "Digital Synthesis of Plucked-String and Drum Timbres." **Computer Music Journal** 7(2): 43-55.

Shepherd, J. 1983 "Extra Instructions." **Compute** 5(10): 261-264.


```
690 0266 A9
690 0267 AE
700 0268 B3        .BYTE $B3,$B7,$BB,$BF,$C2,$C6,$C9,$CC
700 0269 B7
700 026A BB
700 026B BF
700 026C C2
700 026D C6
700 026E C9
700 026F CC
710 0270 CF        .BYTE $CF,$D2,$D4,$D7,$D9
710 0271 D2
710 0272 D4
710 0273 D7
710 0274 D9
720                ;
730 0350           *=$0350 ;TO $0374 HIGH BYTE
740                ;
750 0350 7E        .BYTE $7E,$7E,$7E,$7E,$7F,$7F,$7F,$7F
750 0351 7E
750 0352 7E
750 0353 7E
750 0354 7F
750 0355 7F
750 0356 7F
750 0357 7F
760 0358 7F        .BYTE $7F,$7F,$7F,$7F,$7F,$7F,$7F,$7F
760 0359 7F
760 035A 7F
760 035B 7F
760 035C 7F
760 035D 7F
760 035E 7F
760 035F 7F
770 0360 7F        .BYTE $7F,$7F,$7F,$7F,$7F,$7F,$7F,$7F
770 0361 7F
770 0362 7F
770 0363 7F
770 0364 7F
770 0365 7F
770 0366 7F
770 0367 7F
780 0368 7F        .BYTE $7F,$7F,$7F,$7F,$7F,$7F,$7F,$7F
780 0369 7F
780 036A 7F
780 036B 7F
780 036C 7F
780 036D 7F
780 036E 7F
780 036F 7F
790 0370 7F        .BYTE $7F,$7F,$7F,$7F,$7F
790 0371 7F
790 0372 7F
790 0373 7F
790 0374 7F
```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CURRENT EVENTS

(continued from page 25)

Chromaface (MIDI to Chroma interface). J L Cooper Electronics, 2800 S. Washington Blvd., Marina Del Rey, CA 90921. Tel. 213/827-4884.

Quieters. The Dynafex DX-2 is a stereo, single-ended adaptive filter noise reduction system. It includes an exciter-like "brilliance" control to restore highs aurally perceived as lost during the noise reduction process. For highly critical applications, the DP-1 (single channel version) includes the "brilliance" control as well as several additional controls for optimizing the noise reduction effect. MICMIX, 2995 Ladybird Lane, Dallas, TX 75220. Tel. 214/352-3811.

New Components. The LM1875 power amp delivers 20W into 4 Ohms with ±22V supplies, or 30W into 8 Ohms with ±30V supplies. THD at 1 kHz is 0.015%. 5-lead TO-220 package; $4.60 in hundreds. The LM2879 stereo amp delivers 9W into 8 Ohms. 11-lead TO-220 package; $3.10 in hundreds. National Semiconductor, 2900 Semiconductor Dr., Santa Clara, CA 95051.

The MA-362-CP pro quality op amp ($13.99) features 0.0001% THD, 60 MHz Gain-Bandwidth Product, extremely low noise, ±24V supply for extra headroom, and 25 uV untrimmed offset. Analog Systems, PO Box 35879, Tucson, AZ 85740-5879. Tel. 602/293-4923.

Ministudio. The Porta One ($600) is a battery-operated 4 channel cassette mixer/recorder with switchable dbx noise reduction. Includes pitch control, 2-band EQ and panpot per channel, standard 1-7/8 speed. Optional AC adapter and flight case. TASCAM, 7733 Telegraph Rd., Montebello, CA 90640. Tel. 213/726-0303.

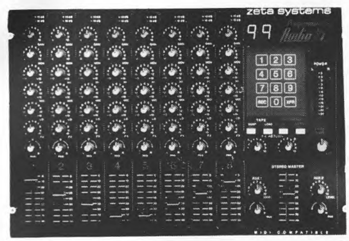
Blank cassettes. Acme Studios now sells high-tech style blank audio cassettes with see-through shells and premium quality tape. Chrome and normal bias tapes are available in 62 and 92 minute lengths. Standard cassettes are also available. Contact Les at Acme for more information; mention Polyphony, and you'll receive a free C-12 high-tech cassette sample. Acme Recording Studios, 3821 N. Southport, Chicago, IL 60613. Tel. 312/477-7333.

Au revoir, MXR. MXR has gone out of business. Repairs are now being handled by Rochester Audio Services, Inc., 982 Monroe Avenue, Rochester, NY 14620 (tel. 716/442-1070). Several of the people at MXR have gone on to form Applied Research and Technology, 215 Tremont St., Rochester, NY 14608 (tel. 716/436-2720). They will continue to market selected MXR products, with new product introductions slated for 1985.

Portable computer. The "Eve" ($1695) is an Apple and CP/M compatible portable computer with built-in dual disk drives, monitor, and detachable keyboard. The manufacturer notes one application is as a compact, all-in-one "road" computer for people using Apple II-based keyboards. Comp-U-Save, Turtle Creek, PO Box 1300, Skyland, NC 28776. Tel. 704/274-3003.

Digital tape standard. Expect digital tape players, the tape equivalent of the compact disc, to surface in late 1986. Standardization should be finalized by the middle of 1985; the two contending techniques are S-DAT (which uses stationary multi-track heads) and R-DAT (which records on a single track with a rotating head).

Pressure zone alternative. The SM91 ($300) low-profile mic is intended for unidirectional, surface-mounted applications and is claimed to offer less muddiness and reverberation than conventional, omni-directional surface mount mics. Shure, 222 Hartrey Ave., Evanston, IL 60204. Tel. 312/866-2534.



Programmable MIDI mixer. The Zeta Programmable Mixer ($2995) is an 8 in - 2 out board with programmable 3-band EQ, monitor send, effects send, pan, level, aux in, and effects return. Stores 99 front panel settings; cassette interface; tape sync for automated mixdown; MIDI compatibility insures communication with other programmable devices and keyboards. Zeta Systems, 1122 University Avenue, Berkeley, CA 94702. Tel. 415/848-7728.

Printed matter. OP magazine is no longer being published, but "Sound Choice" intends to carry on the services and traditions established by OP. $12 in USA for six issues (1 year). Sound Choice, PO Box 1251, Ojai, CA 93023.

"Goldmine" is a magazine for vintage record collectors; it contains lots of ads for used records and some interesting articles. A 1/2 year subscription (13 issues) is $22. For a free sample copy, write to Goldmine, Sample Copy Department, 700 E. State St., Iola, WI 54990. Tel. 715/445-2214.

DOD R-908 REVIEW

By: Jack Orman

The R-908 is a multi-function digital delay that offers up to 900 ms of delay time. There are four range switches, Flange (14 ms maximum), Chorus (56 ms), Double (225 ms), and Echo (900 ms). A control to the left these switches fine tunes the delay time from 0.125 to 1.0 times the maximum of each range. Flanging, chorus, and vibrato effects usually use delay time modulation and the R-908 has controls for speed (0.06 - 6.0 Hz) and sweep width. There are also front panel controls for input and output levels, output mix of direct and delayed signals, feedback, repeat hold, and effect bypass. The feedback control has a center detent for no feedback; turning the control to the left feeds back an inverted signal, while turning the control to the right feeds back a non-inverted signal. A power switch and a four stage LED readout (to aid in setting proper input levels) complete the front panel.

The rear panel has four audio jacks and three control jacks. The input impedance is about 470k Ohms and accepts instrument or line level signals. The Dry output parallels the input jack and obviously, is not delayed. The front panel mix control varies the proportion of straight and delayed signal available at the Mix and Phase output jacks. The difference between these outputs is that the Phase out delay signal is inverted. Sending the Mix out to one channel and Phase out to the opposite channel produces a pseudo-stereo effect. Output impedances are about 1k ohms.

Footswitches can be connected to control Repeat Hold and to turn the delayed signal on and off. Simple push on - push off switches that ground the center conductor (like the reverb footswitch that comes with many guitar amps) are used with these jacks. The remaining rear panel jack, "Control Voltage", allows for external control of the delay time. The operator's manual indicates that a variable resistance connected across this jack controls the internal VCO; this resistance is specified as 0-100 Ohms, although I think a more likely value is 0 - 100k ohms.

The DOD R-908 converts analog signals to 8 bit digital words and has 16K of memory (MK4116N, just like my TRS-80). When converting to digital form, the available dynamic range is about equal to six times the number of bits used in the ADC. For the R-908 this is about 48dB; however, companding (compression/expansion) is used to allow the R-908 to handle signals with larger dynamic ranges. The compansion, along with pre-emphasis/de-emphasis, also reduces the output noise level.

Opening up the delay reveals a well laid out and assembled unit. The analog and digital portions are on separate boards to prevent interaction, components are good quality (5% carbon film resistors, mylar caps, etc.), and the ICs are socketed. In operation the R-908 proved to be a very capable performer, and I was very pleased with its clean, tight sound. The only problem that I encountered involved patching the unit into the effects loop of my Tascam M-30 mixer. With the channel fader at its nominal value, the output of the delay was too low to be useable even with the DOD's output level at maximum. Only by running a very hot signal to the delay input, and setting the mixer channel fader higher than normal, could I get a strong enough signal. (Perhaps the gain of the output mixing amps of the R-908 could be increased to make the unit operate better with the Tascam mixer; removing the 47K resistors paralleling the output level pots would accomplish that.) The owner's manual indicated that the +3 dB LED should not light up during operation but I found that even if it lit on occasional peaks there was no noticeable distortion.

The operation manual is merely two pages stapled together and is too brief to be of much use to the purchaser. Also, the manual should include a copy of the sample patches printed on top of the unit (these can't be seen when the delay is rack mounted, and people in the public library look at you funny when you put a digital delay in the copy machine). I must compliment DOD for including schematic diagrams with their equipment; I learned a great deal about digital delays from tracing the diagram of the R-908, and I think other manufacturers should follow their example and include schematics with all electronic equipment.

From the circuit diagram one can see that the master clock frequency is 55 microseconds (about 18k Hz). Multiplying the sampling interval (55 microseconds) by the maximum number of memory locations (16, 384) gives the maximum delay time (901 ms). By allowing the address counter to cycle to fewer memory locations the delay time is correspondingly shorter. Since the clock frequency is only a little over 18 kHz, the audio bandwidth must be restricted to less than half this amount (8 kHz bandwidth) to prevent aliasing distortion.

I had been looking for a good delay unit to use in my system, and after using one on loan for a few days, I found it to be a fine quality delay and purchased three of them -- two for studio use and one that our band uses in live performance. At one point I considered replacing the RC4558 chips with some quieter models (NE5532, RC4559, TL072) but the sound is good enough without doing this (probably due to the restricted bandwidth). At $399.95 list, I consider this unit a "best buy" and a capable performer in the studio or on stage.

PARTS and KITS from: PGS

LINEARS

TL061	BiFet	.72
TL062	Dual BiFet	.99
TL064	Quad BiFet	1.95
TL071	BiFet	.65
TL072	Dual BiFet	1.15
TL074	Quad BiFet	1.95
NE555	Timer	.39
NE570	Compander	3.80
NE571	Compander	2.95
NE572	Compander	4.95
UA741	Comp. OpAmp	.29
MC1456	Low Noise OpAmp	.90
RC1556	Low Noise OpAmp	1.48
CA3080	OTA	.94
CA3280	Dual OTA	1.98
RC4136	Quad OpAmp	1.10
RC4739	Dual Low Noise	1.19
NE5532	Dual High Perf	3.70
NE5534	High Performance	2.65

SPECIAL PURPOSE

SAD-1024	Analog Delay	17.50
SAD-4096	Analog Delay	37.50
MK50240	Top Octave Div.	5.95
SN76477	Sound Generator	3.45

SANYO HYBRID POWER AMPS

STK050	50 Watt Power Amp	19.40
STK070	70 Watt Power Amp	24.20

SSM- SOLID STATE MICRO-TECHNOLOGY

SSM 2010	VCA	7.50
SSM 2011	PreAmp	5.75
SSM 2012	VCA	9.50
SSM 2020	VCA	7.50
SSM 2022	VCA	7.50
SSM 2030	VCO	7.50
SSM 2033	VCO	10.00
SSM 2040	VCF	7.50
SSM 2044	VCF	7.50
SSM 2050	VCTG	7.50
SSM 2056	VCTG	5.75

THERMISTER (Temp. Sensing Resistor)

TSR-Q81....Tel Labs Q81 1k $3.50

OPTO-ISOLATOR

CLM6000....Clairex CLM6000 .... $2.85

CAPACITORS (25 volt)

701-100	100 pf polystyrene	.25
701-180	180 pf polystyrene	.25
701-1000	1000 pf polystyrene	.25
701-2200	2200 pf polystyrene	.25
701-2200	3300 pf polystyrene	.25
701-3900	3900 pf polystyrene	.25
702-005	.005 uf mylar	.12
702-01	.01 uf mylar	.12
702-05	.05 uf mylar	.16
702-1	.1 uf mylar	.21
702-22	.22 uf mylar	.33
703-1.0	1.0 uf tantalum	.39
703-3.3	3.3 uf tantalum	.49
703-4.7	4.7 uf tantalum	.59
704-2.2	2.2 uf electrolytic	.21
704-4.7	4.7 uf electrolytic	.21
704-10	10 uf electrolytic	.21
704-100	100 uf electrolytic	.31
705-10	10 pf ceramic disk	.15
705-.01	.01 uf ceramic disk	.12
705-.1	.1 uf ceramic disk	.17

IC SOCKETS (soldertail)

IC-S-08	8 pin high quality socket	.27
IC-S-14	14 pin high quality socket	.30
IC-S-16	16 pin high quality socket	.34
IC-S-18	18 pin high quality socket	.40
IC-S-28	28 pin hgih quality socket	.60
IC-C-08	8 pin economy socket	.13
IC-C-14	14 pin economy socket	.15
IC-C-16	16 pin economy socket	.17
IC-C-18	18 pin economy socket	.20
IC-C-28	28 pin economy socket	.40

RESISTORS 5%, 1/4 watt

All EIA values available from 2.0 ohm to 5.1 Meg. Also availble is 10 Meg.

100 each of same value	$1.50
50 each of same value	.98
25 each of same value	.75
10 each of same value	.40
5 each of same value	.25

ASSORTMENTS

10 each of 10 values (100)	3.00
25 each of 10 values (250)	6.50
50 each of 20 values (1000)	16.00

CHORUS/DELAY KIT

This chorus/delay unit, designed by Craig Anderton and featured in Guitar Player magazine, provides flanging, slapback echo, and automatic double tracking effects. The delay range is from 2 ms to 80 ms. Due to the use of compression and expansion techniques, the unit has dead-quiet operation up to about 50 ms and only minimal noise out the full 80 ms. This project kit consists of all electronics, pots, jacks, etc. Also included are the two circuit boards (etched, drilled, and legended) needed for the project. Not included is wire, solder, case, knobs, etc. The Chorus/Delay unit also needs a well regulated bi-polar 15 volt power supply (not included). (A punched and legended rack mount panel will soon be available for this project.)

Order KT-CD777................................ $78.00

"SNARE +" DRUM VOICE KIT

This percussion synthesizer was designed by Thomas Henry and appeared in POLYPHONY magazine. Here's what Craig Anderton had to say about the "SNARE +". "At last - an inexpensive drum voice that has a punchy, full sound. .....All in all, the Snare + delivers a lot of drum sounds, and I would unhesitatingly recommend it to anybody who's tired of the thin sound found in most electronic drum units."

We offer the kit with or without a panel. Kit 3770 contains all electronic parts, switches, jacks, pots, etc, as well well as etched, drilled, and legended circuit board. Kit 3772 includes all this plus a punched and legended rack mount panel (standard 1 3/4 by 19 inches) available in black or blue (both with white legends).

Not included with either kit is wire, solder, mounting hardware, etc. The SNARE + also needs a bi-polar 15 volt power supply (not supplied).

KIT 3770 Basic SNARE + kit............. $33.95

KIT 3772 SNARE + with rack panel... $44.94

THE "CLARIFIER" GUITAR EQ/PREAMP

The "CLARIFIER" is an onboard preamp/EQ module for guitar. This design, by Craig Anderton, was first seen in the pages of GUITAR PLAYER magazine. Here's what the CLARIFIER will do: Replace the guitar's standard passive tone control with a two control, active circuit which provides over 12 db of bass and treble boost and up to 6 db cut.... Buffer your pickups from external loading, giving additional output and improve high freq response.... Add a nominal 6 db of gain to give your signal a bit more punch, as well as improve the signal/noise ratio in multiple effects systems... make your guitar immune to the high freq loss caused by long cable runs.

The CLARIFIER kit is available in two options, both of which include a high quality drilled, legended, and masked circuit board, as well as complete step by step instructions. Kit 2450 contains everything needed for a complete unit.. Kit 2455 contains everything execpt the pots (for those who prefer a particluar brand of potentiometer). Batteries are not included with either kit.

KIT 2450....Complete CLARIFIER kit . $18.95

KIT 2455.....CLARIFIER less controls ..$14.95

TERMS: (Check, Money Order, Cashiers Check - Add .75 if under $10.00)— ($10.00 minimum on C.O.D. (UPS only) add $1.50)— (Mastercard and Visa: $10.00 minimum. You must supply exp. date.)— (Indiana residents add sales tax.)

SHIPPING AND HANDLING: $1.00 plus 5% of purchase. We will credit any amount over our standard rate.

SATISFACTION GUARANTEED!

SIGNAL DIODE

601-60...1N914 (1N4148) signal diode . 5/.35

TRANSISTORS

2N3904	2N3904 NPN Transistor	.25
2N3906	2N2906 PNP Transistor	.25

POTENTIOMETERS

(3/8 long shaft, 5/16 mounting hole)

854-401	10K Linear taper	1.09
854-501	100K Linear taper	1.09
854-505	500K Linear taper	1.09
855-401	10K Audio taper	1.09
855-501	100K Audio taper	1.09
855-505	500K Audio taper	1.09
856-401	10K Audio taper with on/off switch	1.25

TRIM POTS (vertical mount)

802-251	250 ohm trimmer	.40
802-103	10K trimmer	.40

MINI TOGGLE SWITCHES

403-20	SPDT (on/on) sub-mini (3A)	1.20
403-40	DPDT (on/on) sub-mini (3A)	1.50
405-10	SPST (on/off) bat handle (6A)	1.85

LED's

Please note that the typical DC forward current (I-fwd) of these LED's is less than those offered elsewhere making these LED's ideal for battery circuits or others where current consumption is a factor.

305-201	Red T-1¾ jumbo diffused (20 ma.)	.30
305-202	Green T-1¾ jumbo diffused (30 ma)	.40
305-203	Dual T-1¾ jumbo diffused (50 ma)	.90
305-204	Tri T-1¾ jumbo diffused (20 ma)	1.50

Note: 305-204 is a three lead, tri-color (green, red, yellow) device. It is essentially two separate LED's in one package. (The yellow is obtained by turning on both green and yellow.)

JACKS and PLUGS

1/4 In. PHONE JACKS

901-101	Mono standard phone jack	.45
901-103	Mono with n/closed contact	.52
901-105	Mono encl. jack (open back)	.55
902-211	Stereo standard phone jack	.70
902-213	Stereo encl. jack (open back)	.77

1/8 In. MINI JACKS

903-351	Mono with n/closed contact	.32
903-353	Mono encl. (open back)	.26
903-355	Mono enclosed with contact	.35

RCA JACKS

921-100	RCA jack, chassis mount	.34
921-200	RCA jack on phenolic mount	.25
921-300	Dual RCA on phenolic mount	.43

1/4 In. PHONE PLUGS

911-201	Mono, black phone plug	.48
911-203	Mono, red phone plug	.48
911-205	Mono, chrome (metal) plug	1.20
911-211	Stereo, black phone plug	.65

1/8 In. MINI PLUGS

913-251	Mono, black mini plug	.38
913-253	Mono, red mini plug	.38
913-255	Mono, chrome (metal) plug	.56

SWITCHING JACKS

These are stereo phone jacks that contain an independent switching sywtem that is controlled by the insertion of the plug. Jack 905-301 contains the equivalent of a DPST normally on switch. Jack 905-302 contains the equivent of a DPDT on/on switch making it ideal for switching bi-polar power supplies on and off in effects boxes, etc.

905-301	Stereo jack with SPST switch	.90
905-302	Stereo jack with DPDT sw.	1.00

PGS ELECTRONICS
Route 25 - Box 304
Terre Haute, IN 47802

(continued from page 27)
Book Reviews

problem is that Mr. Hammond regarded the words in question as "Jargon," which carries a connotation of double-talk or nonsense, rather than as precise technical terms, in need of equally precise definitions.

4

MUSIC AND SOUND FOR THE COMMODORE 64, by Bill L. Bhrendt (Prentice Hall, Inc., Englewood Cliffs, New Jersey)

This modest book attempts to do several things in its one-hundred plus pages (not counting program listings and appendices). It includes chapters on the fundamentals of microcomputers, the physics of sound, analog synthesis, digital synthesis, and musical notation, in addition to providing listings and explanations of the collection of music generating programs for the Commodore 64, which are the principal reason for the book's existence. Mr. Bhrendt's reason for taking this particular approach to his subject is that he wishes to serve the greatest variety of potential readers: computer users who are unfamiliar with music or synthesis, musicians who have never used a computer, and every other possible permutation of these elements that you might be able to imagine. The result of this approach is that, at least for Polyphony readers, less attention is paid to the techniques of synthesis peculiar to the Commodore 64 than one might desire. The fact is that there are already many books on the market on the fundamentals of microcomputers or basic music theory, but few on the musical use of computers. Given this circumstance, I would have preferred that this book gave more attention to programming techniques and the SID chip.

Mr. Bhrendt is, according to the cover blurb, a performing musician, in addition to being the author of several books on programming. We are not told what sort of music he practices, but judging from the chapter on analog synthesis, it probably doesn't involve synthesizers. This chapter contains its share of factual errors, and, altogether, reads as if was copied from other books without much understanding getting in the way. For instance, there are a couple of patch charts which seem to be thrown in mostly to impress or perplex the reader. In addition, Bhrendt confuses the rare track-and-hold, with the more popular sample-and-hold. Well, there are, altogether, twenty-eight Basic programs in this book. The majority of these are simple sound demonstrators of a dozen or two lines, not unlike those found in Commodore's Programmer's Reference guide. The more elaborate offerings include Soundep, a patch development program for the SID; a Sound Editor, similar in general form to those reviewed by Craig O'Donnell in Polyphony, Volume 9, Number 2; a "Simulated Sample and Hold"; and several short algorithmic music generators.

The idea of getting this much music software for the price of a paperback book was quite exciting, but unfortunately, the quality of most of the programs is not what a serious musician would desire. Soundep, the patch editor, uses an excessive number of screens; every time you change a parameter of your patch, you must go to the main menu and select your parameter, then go to the screen for that parameter before editing. Further, changes cannot be made while the patch is sounding. By contrast SIDMON, a program found on some Commodore demo disks, offers more control over musical parameters, allows changes on a patch while it is sounding, and uses only one screen. The Music Editor also possesses some severe limitations. Rhythmically, only whole, half, quarter, eighth, and sixteenth notes are permitted; dotted notes, triplets, and ties are not possible. Editing and printing selections created on the Music Editor presents other problems. Although compositions can easily be edited or printed out at the time of their creation, once a piece has been saved to tape or disk, it can neither be edited or printed. This effectively limits the work on a selection to what can be done in a single work session. Can you imagine what kind of acceptance a word processor with these limitations would receive? The algorithmic sound generators, such a Mugenchord and Mugenplay, produce totally uninteresting sounds. While a competent composer who is also a good programmer can generate a variety of unusual music using these techniques, the author only succeeds in generating the exercises of an unsuccessful student in a first year theory course.

The trouble with all of these programs is, I suspect, that the author created them for the purpose of putting them in this book rather than for actual use. If an active composer/programmer had set out to write programs with the necessary flexibility for day-to-day music making, the results would, I think, have been far different. Of course it is possible that if you are a fairly competent programmer, you may be able to alter the programs in this book so as to make them suit your musical needs. On the other hand, if you are a good enough programmer to do this, you probably can write useful music software without the aid of this book.

DATA BANK

ADC - Analog to Digital Converter.

CCW - Counter-Clock Wise.

Computer Music Journal - Information requests may be addressed to: The MIT Press, Journals Department, 28 Carleton Street, Cambridge, MA 02142. TEL: (617) 253 - 2889.

CV - Control Voltage.

CW - Clock Wise.

D/A - Digital to Analog Converter.

DDL - Digital Delay Line.

DCO - Digitally Controlled Oscillator.

Electronotes - Information requests may be addressed to: Electronotes, 1 Pheasant Lane, Ithaca, NY 14850.

Gain Bandwidth Product - In op amp specifications GBP indicates an op amp's ability to handle high frequencies with high gain. The higher the value the more accurate the op amp.

HIRES screen - High Resolution monitor screen.

LFO - Low Frequency Oscillator.

Nicads - refers to batteries of Nickle Cadmium composition.

RF - Radio Frequency. Radio Frequencies sometimes leak into or are generated by electronic equipment, thus causing unwanted interference.

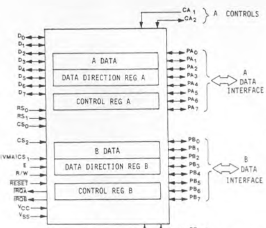
THD - Third Harmonic Distortion. In op amps this is a common type of distortion component. The lower the specified value the less prevalent is this distortion.

VCA - Voltage Controlled Amplifier.

VCO - Voltage Controlled Oscillator

VDU - Video Display Unit.

$BFFF - $ indicates numbers in hexadecimal, (base 16) where A=10, B=11, C=12, D=13, E=14 and F=15.



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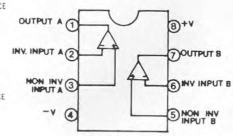


1458
DUAL OP-AMP

6502

PIN NAME	DESCRIPTION	TYPE
$R/\overline{W}$	Read/Write control	output
$\overline{IRQ}$	Interrupt request	input
$\overline{NMI}$	Non-maskable interupt	input
$\overline{RESET}$	Reset	input
Ø0	CPU clock	input
Ø1,Ø2	System clocks	output
DB0-DB7	Data Bus	Tristate, Bidirection
AB0-Ab5	Address Bus	output
RDY	Single Cycle control	input
SO	Set Overflow flag	input
SYNC	Identify op code fetch cycle	output
V_{CC}	Power	
V_{SS}	Ground	

EQUIPMENT EXCHANGE

PUT POLYPHONY TO WORK FOR YOU. List equipmment for sale or trade, job openings, positions wanted, etc. Equipment exchange classified rates for individuals offering goods or services for sale or trade: 25c per word, 20 word ($5.00) minimum charge; Commercial establishments: 50c per word. Prices, zip, phone numbers count as one word each. DISPLAY CLASSIFIED: $15 per column inch, one inch minimum, camera ready art to be supplied by advertiser. All classified advertising must be prepaid. Advertisers using a Post Office Box number for responses must furnish Polyphony Publishing Co. with a complete street address and phone number. Readers should respond directly to advertiser. Polyphony is not responsible for claims made in ads, or for the results of any transactions. Polyphony reserves the right to edit or refuse any ads submitted.

Music equipment

DON SLEPIAN Sells Out! Arp 2600 in good shape, body only $350. Korg PS-3100 synth, patchable, with 48 VCFs, 48 VCAs, 48 ADSRs etc. $750. CEM3660s, 10/$25, MN3005 or MN3011, $10 each, 4016 CMOS, 25/$5. Don Slepian, Box 836, Edison, NJ 08818 Call (201) 572-5088 around 11:00 PM EST.

CASIO 202 8-voice digital keyboard $375 includes insured UPS shipping. (916) 741-4008 or (206) 888-2161.

KORG MS-Q2 log/lin interface (connects PAIA and 1v./oct) $99; MS-Q3 pitch to voltage converter $75; Electroharmonix vocoder $250; Yamaha GE-5 stereo 10 band graphic with spatial expander $125. All minto except for MS-Q3). May trade for other synthesizer equipement. Chris Meyer, 999 West Hamilton #33, Campbell, 95008 . (408) 866-4361 evenings.

PAIA 4700/S Synthesizer system. Includes sequencer, 3 simultaneous filter outputs. Make reasonable offer. Call Maryann (201) 628-1224.

Parts

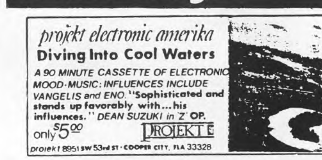
FOR SALE: SSM IC's and misc. from college project that never worked out. Stuff for sale as is, but is new & unused unless notes. 18-SSM 2020 VCA, 35-SSM2030 VCO, 9-SSM2040 VCF, 30 SSM 2050 ADSR ass $4.50 ea. CMOS 4016A 3/$1. 4069/74C04 use, tested 4/$1. 130 switchcraft Hi-D 1/4" mono phone jacks 3/$1. 75 Allen-Bradley 100K linear type J pots $1 ea. 159 red 1-1/4" dia. knobs with aluminum inset 3/$1. Al Duester, 19 Quissett, Woods Hole MA 02543.

LET ME BUILD YOUR PAIA KIT. I have extensive experience with them, nominal fees on most kits. Call (405) 751-5764 Evenings, CST.

Literature

INTERNATIONAL ELECTRONIC MUSIC ASSOCIATION. Join charter members Craig Anderton, Klaus Schulze, and Synergy in promoting the Art. Dues $12 annually include Syne Magazine. P. O. Box 176-P, Salamanaca, NY 14779.

Recordings

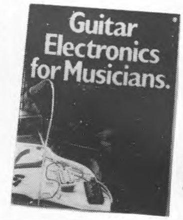


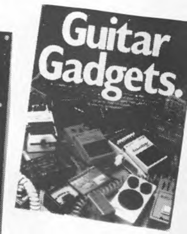
AD INDEX

POLYMART BOOKS

NEW BOOKS!

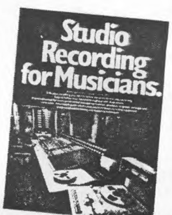
NEW

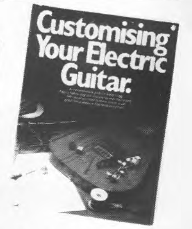




GUITAR ELECTRONICS FOR MUSICIANS by Donald Brosnac is a comprehensive guide for anyone interested in electric guitars. It clearly explains guitar electronics step-by-step with over 350 photos, drawings and schematics. Chapters include: types of pickups, design and function of hardware components, servicing electric guitar circuits, hot rodding electric guitars and more. Anyone who wants to increase his knowledge of guitar construction and function will benefit from reading this book. **# GEM..... Guitar Electronics for Musicians............$12.95**

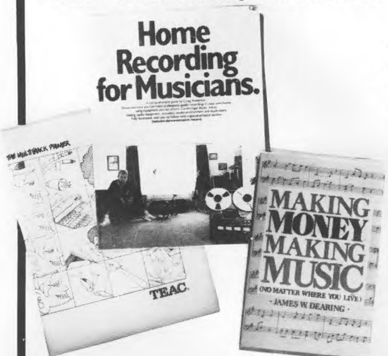
GUITAR GADGETS by Craig Anderton — A consumer's guide written by *the* expert on the subject. For the guitarist who wants to know all about electronic gadgets. How to buy them, fix them, and get the most out of them. Includes a demonstration record.
#GG Guitar Gadgets $14.95





CUSTOMIZING YOUR ELECTRIC GUITAR by Adrian Legg — An Easy to follow guide for customizing your guitar to turn it into a unique and personal instrument. Easy to follow diagrams and step-by-step instructions shows you how to get new and better sound from your guitar.
#CEG Customizing your Electric Guitar **$7.95**

STUDIO RECORDING FOR MUSICIANS by Fred Miller — Tells you everything you need to know about modern studio recording. Easy to follow text, backed throughout with illustrations. A must for professional and aspiring musicians — and for producers, engineers, arrangers and contractors.
#SRM Studio recording for Musicians $14.95

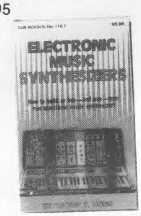


HOME RECORDING FOR MUSICIANS is Craig Anderton's original guide to outfitting and operating a budget studio for maximum results, includes mixer and other audio processing circuits and a sound sheet demo recording.
#HRFM HOME RECORDING FOR MUSICIANS $14.95

Synthesists must be well versed in a number of techniques and principles. "How To" and project oriented books are a great way to pick up these skills. **MULTITRACK PRIMER** by TEAC is a step-by-step guide to building, outfitting and operating your home studio.
#TEAC TEAC MULTITRACK PRIMER $4.95

MAKING MONEY MAKING MUSIC by James Dearing — Everyong dreams of being at the top, but there's an enormous amount of "middle money" out there for the taking. This is not a book about how to become a Millionaire Rock Star, but the strategies revealed will give you the knowledge you need to keep afloat if you decide to pursue a recording contract. A fresh and practical approach to staying alive in the music business. From the publishers of Writer's Digest.
#MMM MAKING MONEY MAKING MUSIC $12.95

REFERENCE



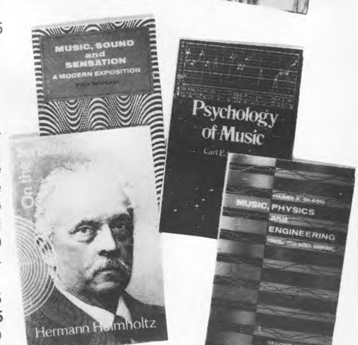
Often used reference materials to answer the many questions encountered in everyday synthesis. **THE SOURCE** Book of Patching and Programming from Polyphony has over 125 pages of patches in universal flow chart notation; the largest publication of its type.

ELECTRONIC MUSIC SYNTHESIZERS by Delton Horn devotes the first half to descriptions and functions of commercial electronic music synthesizers (Moog, Arp, PAIA, Oberheim, EML, and RMI); the second section provides schematics and projects for the experimenter.

#SOURCE THE SOURCE $4.00

#EMS ELECTRONIC MUSIC SYNTHESIZERS $6.95

SCIENCE OF SOUND



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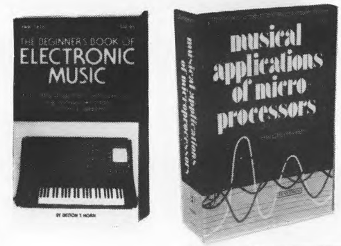

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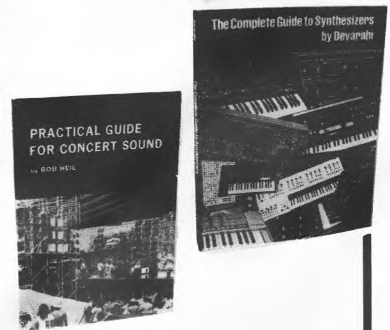

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- #0506: March/April 80: Computers in Music: real time audio processing hardware, Powell sequencer system, Max Mathews, advanced STG software, PortaStudio, phase modulation, Volume 5 index.
- #0601: May/June 80: Gary Numan, Microcomputers in Real Time Audio, Build a Digital Audio Delay Line, writing Documentation, Richard Hayman Composer/Performer Home Recording: Applying Harmonizing and Pitch Transposing Techniques by: Craig Anderton.
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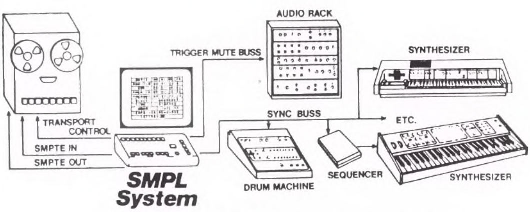

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The eight autolocator points let you get from section to section with a minimum of hassle and wasted time. And a separately programmable CUE point controls the recorder for a looping function at the end of rehearsals and takes. You concentrate on the art, the system attends to details.

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