POLYPHONY

$1.50
ISSN: 0163-4534
May - June
1979

ELECTRONIC MUSIC & HOME RECORDING

sample/hold and noise

voice processing

linear dac

timing signals

vcf circuits

john cage

pet music software

SWTPc ANALOG DELAY

INTRODUCING... TWO-CHANNEL ANALOG DELAY UNIT FOR AMBIENCE SYNTHESIS AND DELAY EFFECTS

FEATURES

* TWO INDEPENDENT CHANNELS
* 3072 STAGES OF DELAY PER CHANNEL
* ADJUSTABLE INPUT AND OUTPUT LEVELS WITH INPUT OVERLOAD INDICATION
* INTERNAL OR EXTERNAL VOLTAGE CONTROLLED DELAY TIME
* COMPANDOR IN EACH CHANNEL
* 3 MODES/CHANNEL WITH ADJUSTABLE MIX
* CONVENTIONAL REVERB OUTPUT FOR MUSIC EFFECTS

If you haven't heard what analog delay can do for home music reproduction, you're missing something. Let's face it, stereo in your living room is flat and 2 dimensional. Without the mixture of direct and delayed sounds that a large hall provides, almost all music reproduced in the home is lifeless. Quadraphonics has not proved to be the solution to this problem. The recent developement of bucket-brigade semiconductor technology has made it possible to offer a reasonably priced delay unit that can transform your listening room into a concert hall. Using your present stereo system, the 2AS-A, and whatever you have in the way of 2 additional speakers and 2 channels of power amplification— you have all the parts to put together an ambience system that is capable of creating the kind of 'space' you enjoy music in. You don't need state-of-the-art components to enjoy an ambience system. The secondary power amplifiers and speakers can be of very modest performance and yet still serve to create strikingly realistic spaciousness in your listening room. If you don't have 2 extra power amp channels on hand, we offer several low cost, low power amps in kit form that would be ideal for this purpose.

Although the 2AS-A has been designed for use in music reproduction systems as an ambience synthesizer, its voltage controlled clock and mixing capabilities allow it to be configured in a number of ways for delay effects such as phasing, flaging, chorous, and vibrato. External voltage control for special effects must be user supplied.

The 2AS-A is sold in kit form only and includes the circuit boards, components, chassis (11½" x 10" x 4"), cover 120VAC power supply, assembly instructions and application notes.

2AS-A Analog Delay Unit
$250.00 ppd. Cont. U.S.

SWTPc Southwest Technical Products Corp.
219 W. Rhapsody, San Antonio, Texas 78216

London: Southwest Technical Products Co., Ltd.
Tokyo: Southwest Technical Products Corp./Japan

MAIL THIS COUPON TODAY

☐ Enclosed is $_____ or BAC #__________

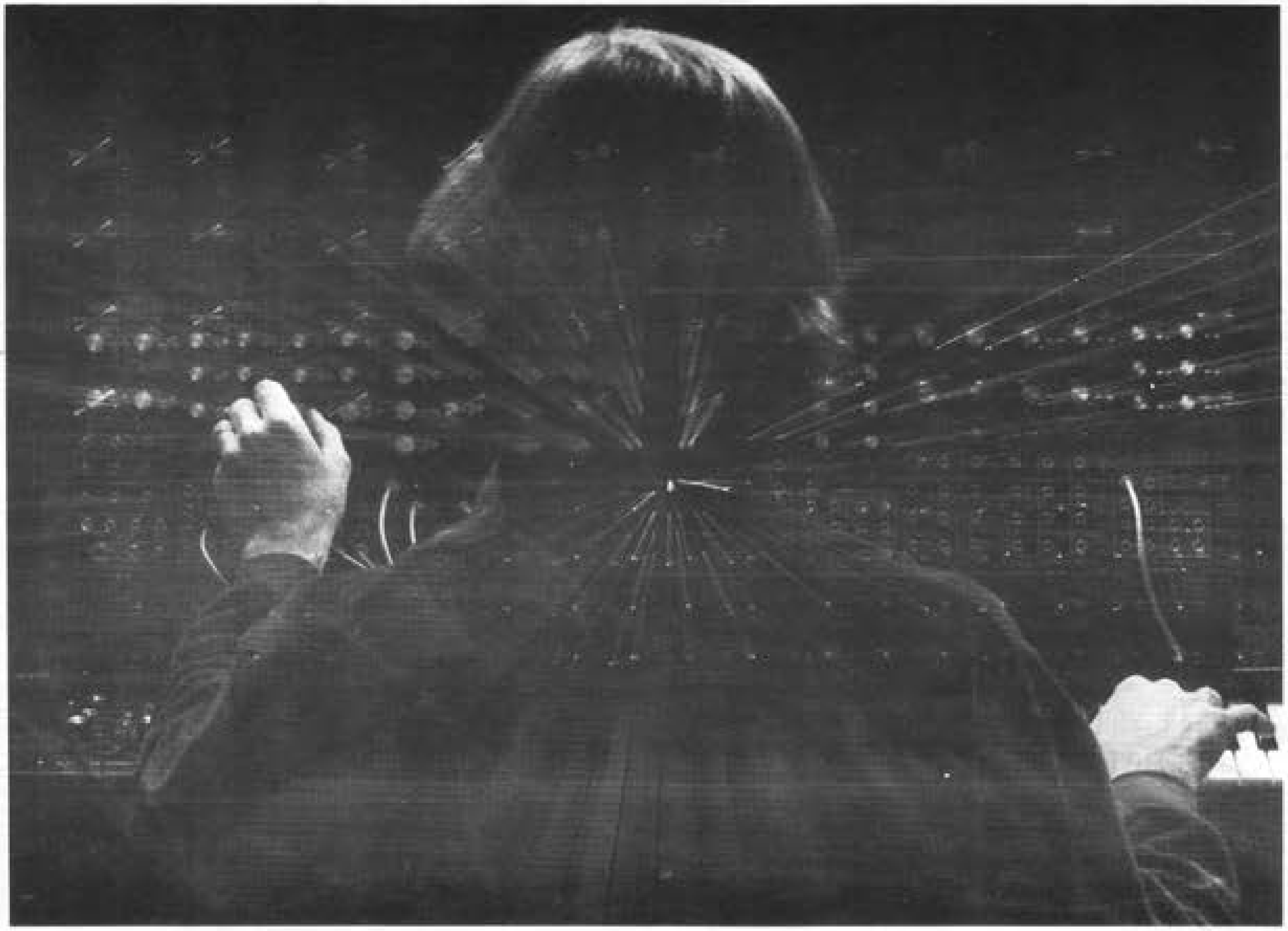
☐ or Master Charge #__________ Bank #_____ Expire Date_____

NAME____________________

ADDRESS____________________

CITY____________ STATE_____ ZIP_____

SOUTHWEST TECHNICAL PRODUCTS CORPORATION
Box 32040, San Antonio, Texas 78284



Voltage-controlled ADSR.

The many unique features of Aries Music systems make them the choice of the professional synthesist.

Voltage control of each of the four parameters of the ADSR envelope generator over an extremely wide range (2 ms. to 200 sec.) gives you more control over the single most important control signal in your synthesizer. By controlling each time parameter with the ADSR's output, envelopes of any shape can be created: , , , and so on.

You can easily make higher-pitched sounds attack and decay faster than low-pitched sounds, mimicking natural sound sources. And by storing control voltages in a sequencer, you can change envelope shapes at the push of a button: harpsichord, *switch*, tuba, *switch*, violin, *switch*, wood block.

Aries Music makes a complete line of modular synthesizers of unparalleled quality and performance at a price you can afford. Our music systems are available factory-wired or in easy-to-build kit form.

In future ads we will discuss other features of Aries Music systems. Watch for them, or send now for our 1979 catalog for descriptions, specs and prices.

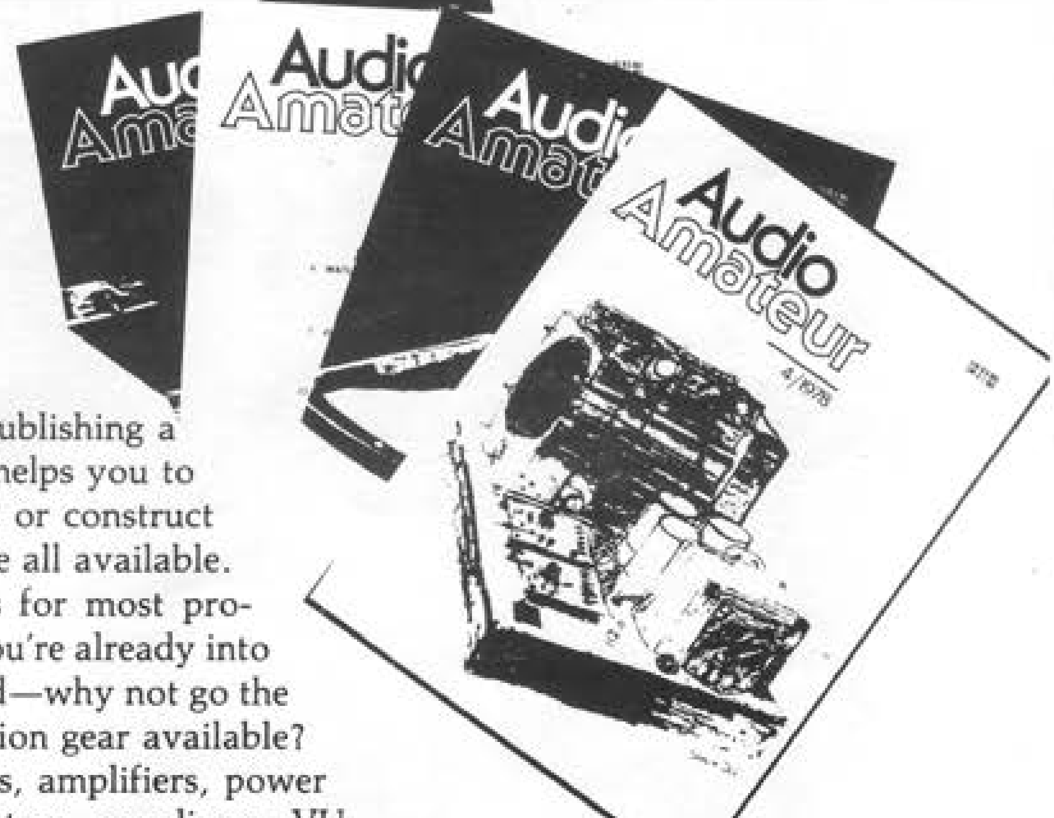
Other sophisticated Aries Music features include:

- □ phase-modulated sync
- □ dual-phase pulse waveforms
- □ voltage-controlled 10-stage phaser
- □ VCA sensitivity pan control
- □ hard and soft sync
- □ simultaneous linear and exponential fm
- □ 3 types of electronic switching

ARIES MUSIC

Shetland Industrial Park
35 Congress Street
Salem, Massachusetts 01970
617-744-2400

AUDIO
for
PLEASURE



Audio Amateur, now in its tenth year, is publishing a wide range of articles, all fully tested, that helps you to build your own sound system from scratch or construct the parts of it you need. Our back issues are all available. And *TAA's* subsidiaries offer kits of parts for most projects—or can tell you where to find them. You're already into the exciting world of direct creation of sound—why not go the rest of the way into the very best reproduction gear available? We offer full construction articles for mixers, amplifiers, power supplies, preamps, meters, overload indicators, equalizers, VU meters, ambient sound adaptors, tonearms and all sorts of speakers from small extensions to large electrostatics to 24" sub-woofers.

Great articles out of our past

1970 "Price, Time and Value" surveys nine years of the fortunes of used equipment. An all silicon, complementary output, 20W per channel amplifier, fail-safe overload protected by Reg. Williamson. A high efficiency bookshelf speaker by Peter J. Baxandall. How do update and improve your Dynaco PAT-4 preamp. A visit to the Heath Co.

1971 A superb, simple, high quality preamplifier by Reg Williamson; A 4+4 microphone mixer, using four ICs in a compact chassis, with eight inputs and two-channel output. A four channel decoder for adding a new dimension to listening: cost to build: $12.50. Two four-channel encoders, one with microphone preamps, to put four signals on two tape tracks. Three voltage/current regulated power supplies for better power amp performance.

1972 A nine octave graphic equalizer with slide pots by Reg Williamson. A 10½" reel tape transport, a full-range electrostatic loudspeaker and a 900 watt tube amplifier for driving the electrostatic panels directly. A high quality op amp preamp, Heath AR15/AR1500 modifications. A new type A+B, low cost 35W power amp, electronic crossovers for bi- and tri-amplifier operation. All about microphones, and tuning bass speakers for lowest distortion.

1973 Construction: Five transmission line speakers: 8" to 24" drivers, peak reading level meter, dynamic hiss filter, tone arm, disc washer, electrostatic amplifier II, and customized Dyna Mark II and Advent 101 Dolby. How to photograph sound, power doubling, microphones, Jung on IC op amps, Williamson on matching and phono equalization, and much more.

1974 A perfectionist's modification of the Dynaco PAS tube preamp, a mid/high range horn speaker, a wall-mounted speaker system, an IC preamp/console mixer by Dick Kunc, a family of regulated current limited power supplies, a switch & jack panel for home audio, grounding fundamentals, low-level phono/tape preamp with adjustable response, an IC checker, a lab type ±15V regulated supply. A series on op amps by Walt Jung and kit reports on an electret microphone and a Class A headphone amplifier.

1975 The superb Webb transmission line speaker construction article, how to test loudspeakers, a test bench set of filters, a variable frequency equalizer, building and testing Ampzilla, a power amp clipping indicator, a compact tower omni speaker, controls for two systems in three rooms. A visit to Audio Research Corp., an ultra low distortion oscillator, all about filters by Walt Jung, a universal filter for either audio garbage or crossover applications. An electrostatic speaker and complete schematics for Audio Research Corp.'s SP-3A-1 preamp, Heath's XO-1 and the Marantz electronic crossovers.

1976 Three mixers by Ed Gately, a vacuum system for cleaning discs, a 60W per channel amp for electrostatic speakers, a silent phono base, a perfectionist's tonearm, re-mods for Dyna's PAS preamp, Jung on active filters, a white noise generator/pink filter, A-Z tape recorder set-up procedures by Craig Stark, modifying the Rabco SL-8E, a high efficiency speaker system for Altec's 604-8G, uses for the Signetics Compandor IC, modifying Heath's IM (tube) analyzer, simple mods for Dyna's Stereo 70 amp, a tall mike stand. Kit reports: the Ace preamp, Heath's 200W per channel amp, Aries synthesizer, Heath's IO-4550 oscilloscope.

1977 Walt Jung's landmark series on slewing induced distortion, a wood/paper/epoxy horn, Reg Williamson's Super Quadpod, experiments with passive radiator speakers, a high efficiency electrostatic speaker with matching low-power direct-drive amplifier, modifying the AR turntable for other arms, do-it-yourself Heil air motion loudspeakers, a $10 Yagi FM antenna, Ed Gately's 16-in/two out micromixer, the speaker saver; complete stereo system protection. Audio Research modifies the Dyna Stereo 70; the super output buffer, a 101dB precision attenuator.

1978 Modular equipment packaging, A PAT-5 preamp modification, a radio system for Hospitals, supply regulation for Dyna's Mark III amp, B.J. Webb on phono interfacing and record cleaning, a 24" common bass woofer, a TV sound extractor, modifying the Formula 4 tonearm, a phono disc storage cabinet, Jung on IC audio performance and noise control, a visit to Peter Walker's Quad factory, a small horn enclosure, an audio activated power switch, the Nelson Pass 40W class A amplifier, a thermal primer, a capacitor tester, recording with crossed cardioids. Kit reports: Heath IC 1272 audio generator, Heath's IM5258 harmonic distortion analyzer, Hafler preamp, Dynaco's octave equalizer, West Side Electronics pink noise generator.

Subscription order form:
The Audio Amateur, P.O. Box 176, Peterborough NH 03458, USA
Enter my subscription order for *The Audio Amateur* as follows:

- ☐ The eight issues of **1979-80 @ $23.00**
- ☐ The four issues of **1979 @ $12.00**
- ☐ The four **1978** issues @ **$12**
- ☐ The four **1977** issues @ **$10**
- ☐ The four **1976** issues @ **$10**
- ☐ The four **1975** issues @ **$8**
- ☐ The four **1974** issues @ **$8**
- ☐ The four **1973** issues @ **$8**
- ☐ The four **1972** issues @ **$8**
- ☐ The four **1971** issues @ **$6**
- ☐ The four **1970** issues @ **$6**
- ☐ Send back issues UPS @ $1 extra

☐☐☐■☐☐

Name ____________________

Street & No. ____________________

City __________ State ______ ZIP ______

NOTE: I understand that the unexpired portion of my subscription will be refunded after my first issue if the magazine is not satisfactory for any reason. Please make checks and money orders payable to *The Audio Amateur*. Rates above are for USA ONLY. Central America & Caribbean add 50¢ for each year; Canada and all others add $2. per year. Non U.S. checks MUST be drawn in U.S. currency ONLY.

REMITTANCE DETAILS

I enclose $________

☐ Check ☐ Money Order

☐ Charge Card authorization

Card No.

☐☐☐☐ ☐☐☐ ☐☐☐☐☐☐☐ ☐

☐MasterCharge ☐Visa/BAC

Bank No.______ Expires___/___

Minimum Charge Order $10.

Signature ____________________

Telephone: (603) 924-6526

Please allow up to six weeks for delivery.

STAFF

EDITOR
Marvin Jones

EDITORIAL ASSISTANT
Jarice Kirkendoll

CONTRIBUTING EDITORS
Craig Anderton
Gary Bannister
David Ernst
John S. Simonton, Jr.

GRAPHIC CONSULTANT
Linda Brumfield

PRINTING
"Dinky" Cooper

POLYPHONY is published bimonthly at 1020 W. Wilshire Blvd., Oklahoma City, OK 73116, by Polyphony Publishing Co. Application to mail at second class rates is pending at Oklahoma City, OK.

POLYPHONY
PO Box 20305
Oklahoma City, OK 73156
Phone (405) 842-5480
ADDRESS CORRECTIONS REQUESTED

CONTENTS

ISSN: 0163-4534

Volume 5, Issue 1
April-June 1979

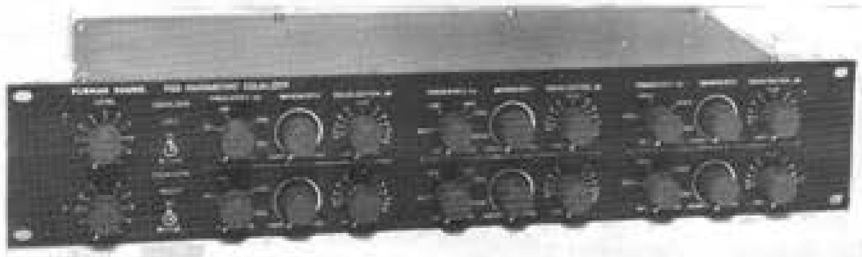
FEATURES

CONSTRUCTION & MODIFICATION

COLUMNS & REGULARS

Tell them you saw it in POLYPHONY

INDUSTRY NOTES:



Parametric EQ

The Furman Sound model PQ-6 is simply the equivalent of two model PQ-3 parametric equalizers in one package, with a newly redesigned black anodized front panel. Among its uses are: a stereo musical instrument preamp for use with an electric instrument or pickup plus a stereo power amp or conventional musical instrument amp; as a general purpose patchable equalization for studio or broadcast applications; in a PA system for feedback suppression; or for room equalization in studio control rooms or the home.

The Furman PQ-6 features three continuously variable and broadly overlapping frequency controls per channel so you can zero in EQ exactly where you want it. Each band can be boosted by up to 20 dB, or cut to cancellation, with infinitely deep notching capability. Equalization curves are the non-reciprocal kind, offering the user a range of bandwidth adjustments of maximum usefulness in musical work. For boosting, the bandwidth varies from 1/3 to over 4 octaves, yet offers a much narrower range for cutting of 1/10 to 1 octave, making the PQ-6 suitable for use as a notch filter if desired. Each channel has a seperate BYPASS switch, and an LED indicating EQ IN. Other features are high and low level inputs and outputs, 1/4" phone jacks, and a detailed instruction manual. The PQ-6 has calibration adjustments for absolute accuracy in the equalization controls, and all ICs are socketed for easy serviceability. The unit mounts in 19" racks, and is available in 115V and 230V versions.

For more information, contact: Furman Sound, Inc., 616 Canal St., San Rafael, CA 94901, Ph. 415-456-6766.

Music Workshop

The Schools Office of Chautauqua Institution (Box 28, Chautauqua, NY 14722) announces an Electronic Music Workshop to be held from July 30 - August 10 under the direction of Reynold Weidenaar. Classes will meet for one to three hours of lecture and demonstration per day, followed by five to six hours of semi-private "hands-on" sessions with the synthesis and recording equipment. No musical or technical background is required; there is a minimum enrollment of 22 required by July 30. Two hours credit will be given by SUNY at Fredonia. The fee is $110.

Following the two week workshop, the studio equipment will remain at the university for an additional week to allow individual use by a limited number of the students. Maximum enrollment is 10 students. Fee is $65.

Reynold Weidenaar received his BM degree from Cleveland Institute of Music, and has studied with Donald Erb, Robert Moog, and Vladimar Maleckar. He was editor of Electronic Music Review, and is director of the Electronic Music Studio of the Cleveland Institute of Music.

Mixer Update

TEAC Corporation has announced a new version of their highly successful Model 2 mixer, called the Model 2A. The new unit features more reliable modular construction and the addition of treble and bass controls. The suggested retail of $400 has remained the same, as well as compatibility with the MB-20 meter bridge, as shown. The MB-20 provides a 4 X 2 monitor mix plus four large averaging VU meters, independent buss/tape monitor switches for each channel, and an internal headphone amp.

The 2A features straight line faders, push button channel assignment, and mic and line inputs. Mic connections are via quarter inch jacks, while line connections are through phono jacks. Output connections are provided for buss and aux outputs, as well as cue outputs and accessory send/receive patch points. Stereo panning is automatically enabled when more than one output channel is selected for any input channel.

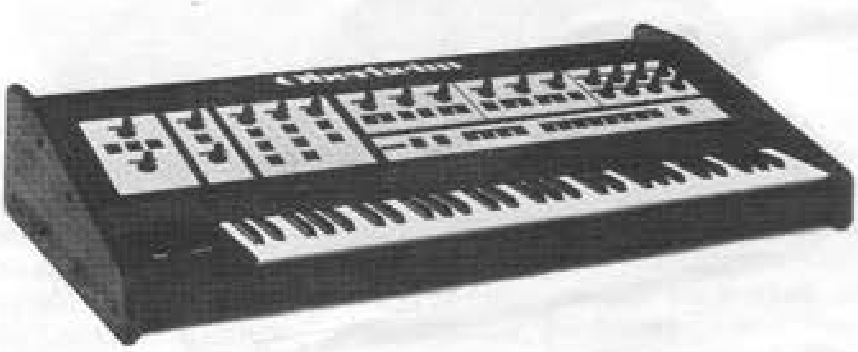
For more information, contact: TEAC Corporation of America, 7733 Telegraph Road, Montebello, CA 90640, Ph. 213-726-0303.

Two From Oberheim

The Dual Manual Eight Voice Polyphonic Synthesizer with Programmer consists of eight Synthesizer Expander Modules under the control of dual manuals, keyboard controllers, and an eight voice polyphonic programmer. The top keyboard has a four voice capability while the lower keyboard can use all eight voices. The lower keyboard acts as a master for pitch bend and filter control. In mode four, four voices will be controlled by each keyboard; in mode six, two voices appear on the upper keyboard and six on the lower; in mode eight, all voices are controlled by the lower manual. Also included are CV IN and CV OUT for all eight modules. Suggested list price is $12,500.

The new OB-X is a fully programmable four, six, or eight voice polyphonic synthesizer with 32 program memory storage plus direct to tape cassette storage for building a library of patches. The OB-X is microprocessor controlled to provide the ultimate in versatility while maintaining a portable package. Features include a five octave keyboard, auto tune, edit mode, polyphonic portamento, polyphonic sample and hold, noise generator, dual modulation levers, ADSR envelope generators, and two oscillators per voice. Foot pedal control inputs are provided for vibrato, volume, and filter; plus sustain and program advance foot switches. The factory offers a "roadie kit" to facilitate quick and easy field repairs. The construction techniques employed ensure high reliability and serviceability. Retail prices are $4295 for the four voice, $4995 for the six voice, and $5495 for the eight voice.

For information on any of the above units, write directly to Oberheim Electronics, 1455 19th St., Santa Monica, CA 90404, or call 213-829-6831.



Check Your Phase

Sounder Electronics Inc. (21 Madrona St., Mill Valley, CA 94941, Ph. 415-383-5811) has released the Phase Checker to ease a number of tasks in studio and PA applications. The device generates high and low level signals, an audible tone, and power signals to drive external speakers; inputs are accepted from low and high level sources, or acoustically by an internal mic. The test is performed entirely within the hand held unit which uses two 9 volt batteries; LEDs indicate NORMAL or REVERSE phase for the device under test. The signal generator and speaker can acoustically drive microphones as well as provide electrical signals for component tests. Similarly, the built-in microphone can test for proper phasing of the acoustic signals output by speakers. Cables, connectors, and snakes can be checked with the Phase Checker as well as electronic or acoustic components.

Phase coherency throughout an audio system is important for proper bass frequency response. With few standards in the speaker wiring or reconing industry, and since it is easy to make a mistake when rushing through the repair of a speaker or mic cable before a big job, easy methods of checking phase polarity became a necessity for the quality sound man. The Phase Checker will finally check it all.

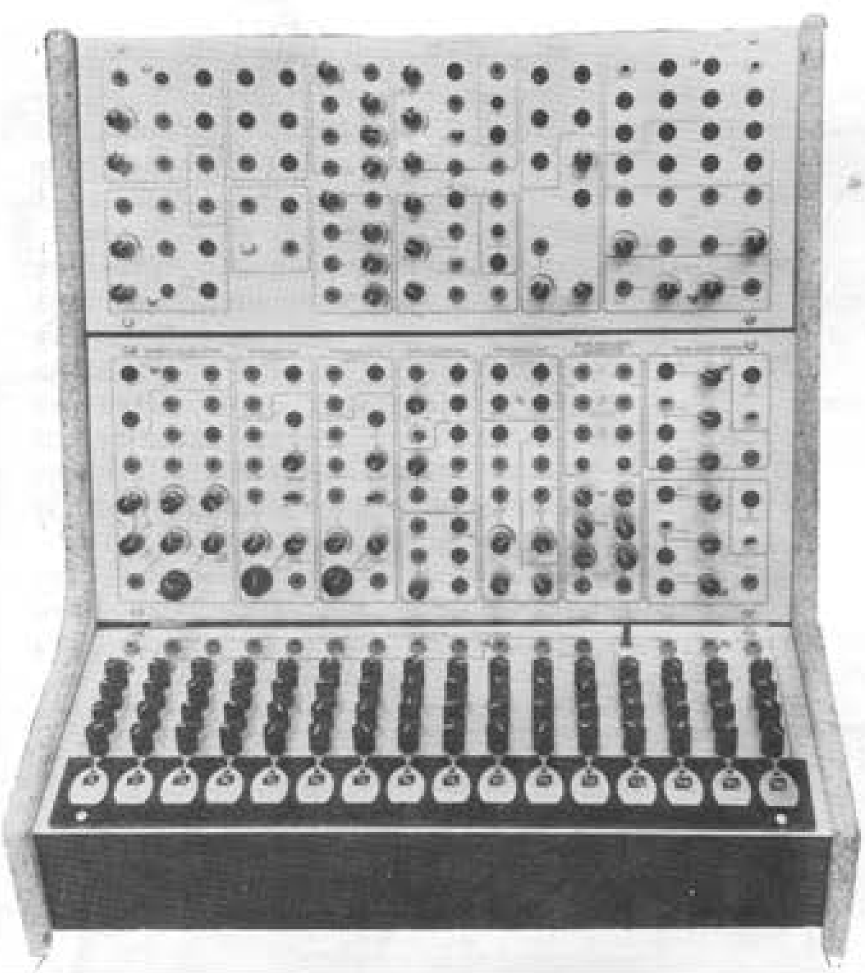
The price of the Model 500 Phase Checker Set is $495 plus shipping and tax. For more information, contact Sounder Electronics at the above address.

New System

Serge Modular Music Systems announces the 'Series 79' system as a small state-of-the-art synthesis system. The 'Series 79' is a versatile self-contained synthesizer, comparable in performance to the highest quality studio systems. Featuring patch programmable functions, the 'Series 79' is able to be patched to obtain 9 VCOs, 5 VCAs, low/high/band/notch filter, phase shifter, a variety of mixers, LFOs, and unique control voltage features.

In addition, the Serge 'Touch Activated Keyboard Sequencer' is included to provide sequential voltages (4 rows of 16 stages, 16 columns of 4 stages, or one 64 stage sequence); manual touch keys can produce control voltages in combination with the sequencer or independently. A finger pressure output is included to provide additional control over sophisticated synthesis.

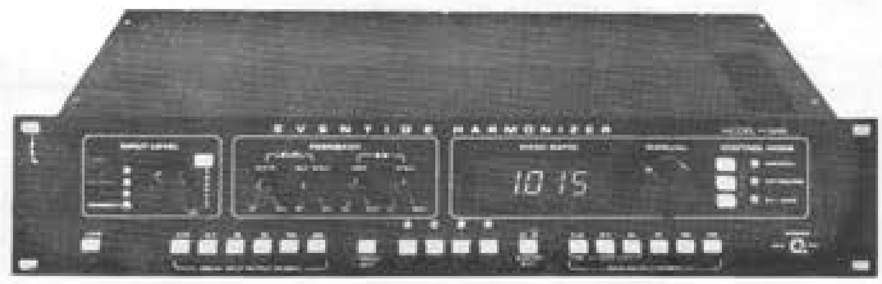
The unit is currently available for $2850, F.O.B. Los Angeles, from: Serge Modular Music Systems, 1107 1/2 N. Western Ave., Los Angeles, CA 90029, Ph. 213-461-7987. Write for a descriptive flyer on the 'System 79'.



Fun & Games

The new H949 Harmonizer from Eventide Clockworks can shift pitch from one octave up to two octaves down, has two outputs (each with 400 ms. of delay), a response of 15KHz, and S/N ratio of 96 dB. The H949 can produce flanging, repeat, random delay (for automatic double voicing), and a new effect- REVERSE. The micro pitch change function allows extremely precise settings, ideal for tuning in a late addition to a mix. There is also high and low EQ of feedback.

The H949 has two algorithms to handle the pitch change 'glitches', so the user can select whichever is optimum for the program material. Algorithm #2 provides, with the possible disadvantage of some 'coloration' of the sound, a glitch-free method of changing pitch by small ratios (roughly between .9 and 1.1). Included are a four digit pitch ratio readout, red/green LEDs for function indication, delay select buttons, and pitch control by panel control or optional musical keyboard. Delivery starts September 1, 1979; cost is $2400.



Eventide has also produced a real time audio spectrum analyzer which fits in a Commodore PET computer, and costs about 1/6 the price of currently available units. The analyzer divides the audio spectrum into 31 third-octave bands, and displays the bands and their amplitudes on the PET screen. The system can equalize hi-fis and PAs, check response of audio components, and do speech and sound pattern recognition (useful for voice control systems). With the capabilities of the PET, great flexibility in the manipulation of analyzed data is permitted. The PET can store and recall spectral data, and compare them with past, future, or other channel data. The PEAK HOLD feature determines if any preset levels have been exceeded. The three programs provided with the unit (Interactive Operation, Self Test, and Minimal Operation) are written in BASIC. The analyzer consists of one board which installs in the PET in about five minutes, and draws it's power from the PET transformer. Dealer inquiries invited; the cost is $595.

Industry Notes...

To introduce the analyzer, Eventide is running a CONTEST - write a program to recognize disco music. Prizes include Eventide T-shirts for any entries with merit, and either a real time analyzer or memory add-on for the PET. Programs must be submitted on a PET cassette, run in PET BASIC, and run on an 8K PET. All entries will be tried on 10 minutes of music and speech, and will be rated as follows: 50% for accuracy of program's decision, 30% for time taken to decide, and 20% for elegance, originality, documentation and other factors. All entries become the property of Eventide.

For more information on the products, or for a contest entry form and rule sheet, contact: Eventide Clockworks, Inc., 265 W. 54th St., New York, NY 10019, Ph. 212-581-9290.

PET's BUILT-IN SYNTHESIZER

By: Russell A. Grokett Jr.,

Interfacing your external electronic music equipment with the Commodore PET (Commodore Business Machines, 901 California Ave., Palo Alto, CA 94304) was the subject of my last article for Polyphony (September/ October 1978, p. 40). After more familiarization with the PET, I have found a more convenient method for generation of music. Those of you who haven't invested in a lot of signal generation equipment yet will be happy to know that the computer itself can do most of the work for you. PET has its own built-in synthesizer! (Well, part of one, at least.) Inside PET is a square-wave Digitally Controlled Oscillator (DCO), a Digitally Controlled Filter (DCF), and a controller for both.

INNARDS

The DCO and DCF are contained inside a 6522 VIA (Versatile Interface Adapter), an IC produced by MOS Technology, Inc. (950 Rittenhouse Rd., Norristown, PA 19401) This IC is used to control input and output operations for PET, and is accessable to PET owners through the parallel user port on PET.

Besides having an 8 bit parallel I/O port, PET has a pin designated CB2. It is CB2 that is accessed by the VIA and which will be our "synthesizer" output.

BREAKER-BREAKER GOOD-BUDDY

Contrary to popular belief, CB2 has nothing to do with the similarly named radio-telephone hobby, but is the essential ingredient in our PET music synthesizer. Connecting a line from CB2 to the input of an audio amplifier, you can give PET the ability to squeak. Here's how to do it:

1) Buy an edge connector for the PET user port (available from A B Computers, Box 104, Perkasie, PA 18944).

2) Select a length of shielded audio cable long enough to reach your synthesizer equipment or audio amp.

3) Connect a suitable plug to one end of the cable, and connect it to your amplifier or synthesizer.

4) Notice that the edge connector pins are labeled 1-12 on the top row, and A-F, H, and J-N across the bottom. Connect the ground (shield) on the computer end of the cable to pin N of the connector.

5) Connect the center conductor to pin M (CB2).

6) Plug the connector into the PET being sure that the two wires you just soldered go to the correct pins on the bottom row of the connector.

SOFTWARE TIME!

Before PET will "sound off", you will have to tell it HOW to speak!

The first thing we have to do is set the 6522 VIA so the shift register is in the "free-running mode" by:

```
POKE 59467,16
```

When you RUN this statement, you set the "music mode" by turning off the cassette recording system. You cannot record (SAVE) or LOAD a program now. To use your cassette again, you must "POKE 59467,0".

After putting the 6522 in the "free-running mode", set the DCF (Digitally Controlled Filter) in PET to a value between 1 and 255. Such as:

```
POKE 59466,15
```

This controls the tone of the DCO (Digitally Controlled Oscillator) inside PET.

Now, time for the big event. So far, you have heard nothing from PET, but that is about to change. Use caution, however- the square-wave output from PET is quite high (about 5v p-p) so it will be very loud (and perhaps damaging) if your amplifier's volume level is high! After levels have been checked and you are ready to synthesize, instruct PET to:

```
POKE 59464,115
```

The value 115 given in the example can actually be any number between 1 and 255, with 255 being the lowest note or pitch and 1 being the highest (you cannot hear it). Be sure to experiment with different DCF (tone) values to find which are the most pleasing. (10, 14, 15, 51, and 85 are interesting.)

One last (but important) point. Always turn the music mode off after the sound is finished. Otherwise the cassette recorder will not record properly!

```
POKE 59467,0
POKE 59466,0
POKE 59464,0
```

Now try this short little program:

```
10 POKE 59467,16
```

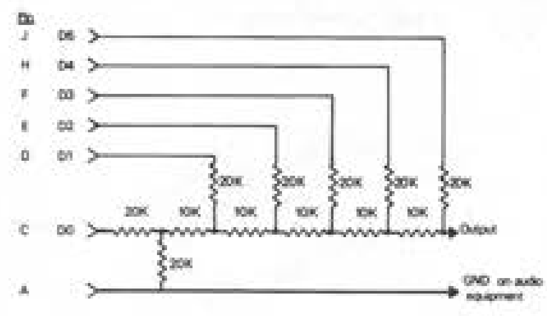
1. Values for Musical Note Equivalents

B0 = 251	(B below first C)	B = 124	(C above first C)
C = 237	(first C)	C1 = 117	
C# = 224		C1# = 111	
D = 211		D1 = 104	
D# = 199		D1# = 99	
E = 188		E1 = 93	
F = 177		F1 = 88	
F# = 167		F1# = 83	
G = 157		G1 = 78	
G# = 149		G1# = 73	
A = 140		A1 = 69	
A# = 132			

2. Program Listing for Music Control

```
   2  REM     PET MUSE 3.5 (C)
   3  REM     BY R. GROKETT, JR. 2/79
   4  REM
  10  POKE 59467,16: REM SET MUSIC MODE
  20  INPUT "TONE (15,51,81...)";TO
  25  INPUT "SPEED (10=FASTEST 75=AVG.)";SP
  30  PRINT "NOTE    .TIME"
  40  POKE 59464,0: POKE 59466,TO
  50  READ N$,T
  60  PRINT N$,T
  70  IF N$="END" GOTO 500
 100  REM     INTERPRETER SECTION
 110  IF N$="30"  THEN N=251
 120  IF N$="C"   THEN N=237
 130  IF N$="C#"  THEN N=224
 140  IF N$="D"   THEN N=211
 150  IF N$="D#"  THEN N=199
 160  IF N$="E"   THEN N=188
 170  IF N$="F"   THEN N=177
 180  IF N$="F#"  THEN N=167
 190  IF N$="G"   THEN N=157
 200  IF N$="G#"  THEN N=149
 210  IF N$="A"   THEN N=140
 220  IF N$="A#"  THEN N=132
 230  IF N$="3"   THEN N=124
 240  IF N$="C1"  THEN N=117
 250  IF N$="C1#" THEN N=111
 260  IF N$="D1"  THEN N=104
 270  IF N$="D1#" THEN N=99
 280  IF N$="E1"  THEN N=93
 290  IF N$="F1"  THEN N=88
 300  IF N$="F1#" THEN N=83
 310  IF N$="G1"  THEN N=78
 320  IF N$="G1#" THEN N=73
 330  IF N$="A1"  THEN N=69
 340  IF N$="R"   THEN N=0
 350  POKE 59464,N
 360  IF T (less than or equal to) 0 THEN 50
 370  FOR I=1 TO T*SP: NEXT I
 380  GOTO 50
 500  REM  END
 510  POKE 59467,0:POKE 59466,0:POKE 59464,0
 520  END
1000  REM   MUSIC NOTES AND TIME
1005  REM - TAPS - SPEED:100 TONE:51
1010  DATA C,2,R,0,C,1,R,0,F,4,R,1,C,2,
1020  DATA R,0,F,1,R,0,A,4,R,1,C,2,R,0
1030  DATA F,1,R,0,A,3,R,.4,C,2,R,.2
1040  DATA F,1,R,.2,A,3,R,.4,C,2,R,.2
1050  DATA F,1,R,.2,A,5,R,1,F,2,R,.2
1060  DATA A,1,R,.2,C1,5,R,.2,A,4,R,.2
1070  DATA F,3,R,.2,C,5,R,.2,C,4,R,.2
1080  DATA C,2,R,.4,F,5
5999  DATA END,0
```

3. PET User Port



4. PET Music Summary

POKE 59467,16	: Sets the 6522 chip for sound mode.
POKE 59466,T	: Sets tone of sound output. T=1 to 255
POKE 59464,N	: Sets pitch (note). N=1 to 255. Set N to zero (0) for Rest (silent).
POKE 59467,0 POKE 59466,0 POKE 59464,0	: Clear sound mode.

PET-MUSE software is available from:
Pet Library
401 Monument Rd.#177
Jacksonville, FLA 32211
SASE for list of available programs.

```
10 POKE 59467,16
20 POKE 59466,10
30 FOR P=1 TO 255
40 POKE 59464,P
50 NEXT P
60 POKE 59467,0:
   POKE 59466,0:
   POKE 59464,0
70 END
```

By adding and modifying the following statements in the above program, PET can generate (random) "Computer Music".

```
25 FOR I=1 TO 50
30 P=INT(RND(1)*255)
45 FOR J=1 TO 100 : NEXT J
50 NEXT I
```

Figure 1 is a list of values for musical note equivalents for PET. Combining these equivalents with a note-length value, you can have PET play sheet music.

Figure 2 is the listing of a program in which songs can be saved as DATA, and played on command. You can select a piece of sheet music, and write the DATA as note pairs, i.e.:

```
C,2    -a quarter note
D#,3   -a dotted quarter
R,0    -short rest
R,8    -long rest
```

WHAT NEXT?

Since CB2 music does not use any of PET's 8 bit port lines, let's add a Digital Voltage Controller (DVC) to PET. To do this, we will have to add a simple Digital to Analog converter to PET. If you have already built a DAC, then don't worry about the following - you already have a DVC! Using this DVC and your additional synthesizer modules (VCA's, VCF's etc.), you can further process your computer music.

First, we need to modify our CB2 line by unsoldering the wire from pin M, and soldering a .1uf capacitor between pin M and the previously unsoldered wire. This will avoid having the high DC content of the CB2 line applied to any of your audio processing modules which may have direct coupled inputs. Figure 3 shows a simple D/A converter for PET. Notice that only 6 bits of the 8 bit port are used. The reason for this primarilly hinges on the fact that the more bits used, the more resistors needed and the lower the accuracy of the D/A conversion due to resistor tolerances. 5% types will work for most purposes; if you can get them, use 1 or 2% precision resistors. Remember: the resistor D/A is NOT equally-tempered for controlling linear VCOs accurately, however it will control most processing modules and even some exponential VCOs without any trouble.

After you connect the D/A converter to the PET user port connector, run the following program and measure the output with a meter. If all is well, the output should rise slowly from 0 to 4 volts (approx.).

```
10 POKE 59459,255
   REM * SET PORT FOR OUTPUT
20 FOR I=0 TO 63
30 POKE 59471,I
40 FOR J=1 TO 100: NEXT J
50 NEXT I
```

Make sure you have connected pin A (gnd) to your audio equipment! You now have a DVC (Digital Voltage Controller).

Now lets use our DVC to add digital volume control to our CB2 music program. Connect your DVC output to a VCA (Voltage Controlled Amplifier) control input, and your CB2 line to one of the audio inputs. Next, add the following lines to your CB2 music program:

```
15 POKE 59459,255: V1=6
345 GET V: IF V=0 THEN V=V1
355 V0=V*7: POKE 59471,V0
365 V1=V: IF N=0 THEN 50
```

When you run the program, pressing the number keys 0-9 causes the volume to change.

THE FUTURE

With suitable programming, you can have PET control the pitch, tone, and now the volume of each note played. That certainly doesn't exhaust all the possibilities for this basic music control system. For example, there are still two bit lines remaining in the output port which can be used to trigger external equipment. Your experimentation is the best path to full realization of any number of peripherals and applications: Digital Filter Controller, Digital ADSR Generator, Digital VCO Controller, Digital Mixer, Digital....

COMPOSER PROFILES

JOHN CAGE

BY: DAVID ERNST

John Cage (b.1912) is among the most influential composers of the twentieth century. Born in California, his principal composition teachers were Henry Cowell and Arnold Schoenberg. In 1943 Cage moved to New York, and nine years later he organized the "Project for Music for Magnetic Tape," the first American center for the composition of electronic music. He is presently director and composer for the Merce Cunningham Dance Company.

It is impossible to examine the music of John Cage without discussing first his compositional aesthetic. His continual search for "new" sonorities is manifest in the early works "Imaginary Landscape No.1" (1939; for two RCA Victor test records, variable-speed phonograph, Chinese cymbal, piano interior), "First Construction In Metal" (1939; for brake drums, thunder sheets, water gong, piano interior etc.), "Imaginary Landscape No.2" (1942; for percussion, amplified coil of wire),"Imaginary Landscape No.3" (1942; percussion, amplified wire coil, tin cans, buzzer, oscillators etc.), and "Amores" (1943; for prepared piano, percussion). These compositions stem from the music of Henry Cowell, Edgar Varese, Harry Partch, et al, yet they also display Cage's fascination for all types of sound--especially "non-musical" or "noise", e.g. tin cans, brake drums etc.

Similar sonorities were introduced by the Futurists, Dadaists, and Bauhaus artists beginning in 1911. Among the most important are the "Intonarumori" (noise instruments) of Luigi Russolo; use of variable-speed phonographs by Darius Milhaud, Paul Hindemith, and Ernst Toch; the use of diverse noises (sewing machine, clock, water etc.) by Kurt Schwitters, Oskar Schlimmer, Moholy-Nagy, and Filippo Marinetti; and the reflected-light compositions of Ludwig Hirschfeld-Mack. The Dadaists were particularly important for their treatment of language, exemplified by the 'chance' and 'simultaneous' poems of Tristan Tzara. In these poems literal meaning is replaced by the sonorous characteristics of the words, often involving more than one language. In addition, the element of chance or indeterminancy is introduced, and this is prevalent in most of Cage's music.

D.T. Suzuki, the renowned Zen master, has also influenced John Cage. The philosophy of Zen is a perfect rationalization both for the admittance of all sounds in a "musical" composition and for the use of indeterminate compositional processes, for Zen is sensitive to the laws of Nature. When applied to the music of Cage this results in complete awareness of all sounds and their freedom to interact without limitations imposed by the composer such as predetermined formal schemes.

COMPOSITIONS WITH TAPE

"Williams Mix" (1952) is a solo tape collage of diverse sounds subdivided into six categories of city, country, wind produced, electronic, "small" (requiring amplification) sounds, and preexisting musical compositions. These materials are then fragmented via splicing and juxtaposed to form constantly changing textural masses. Although related to the subtractive methods of musique concrete composers, Cage extends this process by placing the individual fragments in such close proximity to one another that the listener is bombarded with rapid successions of haphazard events. "Williams Mix" is therefore an excellent illustration of the aforementioned principles of Dadaism and Zen, for Cage did not predetermine the succession of sounds; he employed the I-Ching Book of Changes for this task. By removing himself from the compositional process as much as possible, Cage was able to realize a sound-environment independent of predefined, i.e. personal, goals. For instance, "Williams Mix" cannot be defined generally as 'happy', 'sad' etc.; this is not to say that extra-musical content is absent. Cage simply realizes the futility of imposing his intentions upon an audience. Instead, he presents a tape as innocently as possible so that each member of the audience is free to respond to the aural stimulation in a truly personal manner. Such non-goal oriented composition follows precisely Zen ideology.

Similar techniques are present in "Fontana Mix" (1958) and "Aria with Fontana Mix" (1958). As its title implies, the latter is the juxtaposition of two compositions ("Aria" is for solo voice) resulting in an increased level of indeterminancy. Furthermore, "Aria" incorporates language in much the same manner as the simultaneous poems of the Dadaists. Cage couples five languages (English, French, Italian, Armenian, and Russian) with a tape collage whereby all traces of semantics are lost. Again, Cage establishes a non-predictible sound-environment so that the ears of the listener may be able to wander freely, thereby removing all restrictions on aural-visual associations.

A more recent example of instrumental-tape composition is "HPSCHD" (1967-69), for one to seven live harpsichords and one to fiftyone tapes. Lejaren Hiller, the noted computer music composer, collaborated with John Cage in this work. Among the most noteworthy aspects of "HPSCHD" are the methods of permutation applied to the live harpsichords, the role of computers, and the unique role of the listener.

The live harpsichord material is derived from the keyboard music of Mozart, Beethoven, Chopin, Schumann, Gettschalk, Busoni, Schoenberg, Cage, and Hiller; the music of Mozart predominates. Each of the seven harpsichord parts is composed of unrelated passages from the aforementioned composers - sometimes played as written, and sometimes played by reversing the treble clef to bass clef and vice versa. While the juxtaposition of such diverse musical styles again reflects Dadaist thought, the musical permutation of reversing the treble and bass clef is reminiscent of Marinetti's suggestions (in 1913) to play a Beethoven symphony backwards, and to reduce Wagner's "Parsifal" to forty minutes!

Cage and Hiller used the computer in two manners: to generate sound via software and D/A conversion, and to generate a numeric print-out to enable the listener to alter the overall performance of "HPSCHD." Each of the fiftyone computer generated tapes is based upon a different tuning system---from five pitches per octave (macrotonal) to fiftysix pitches per octave (microtonal). When combined with the standard tuning of the harpsichords the tonal textures become quite complex so that discrete pitch recognition (i.e. melody) is difficult, and at times impossible. Further textural complications arise due to the envelopes of the computer generated sounds, for they are based upon harpsichord envelopes.

Compared to "Williams Mix" and "Fontana Mix," the sound-environment of "HPSCHD" is much more complex. Perhaps this is one of the reasons that prompted Cage and Hiller to develope the computer program KNOBS, a unique method of admitting yet another level of indeterminancy to the recorded performance of "HPSCHD." Each record album contains a computer print-out that includes seven columns: time (beginning at 0.00 and preceeding in 5-second increments to the end of the piece at 21 minutes); volume (channels 1 and 2); treble (channels 1 and 2); and bass (channels 1 and 2). Volume, treble, and bass control settings are defined within the range 0-4 (0=minimum, 4=maximum). As the record plays the KNOBS print-out should be followed by changing the appropriate volume, treble, and bass levels at the prescribed 5-second intervals. It should also be mentioned that each KNOBS listing is unique, so that each recorded performance will be different if the KNOBS print-out is followed.


photo by: Lori Seid

If any readers are skeptical concerning the various results of KNOBS, or if some feel that it is not necessary even to change any of the control settings, then I encourage you to do two things. First, listen to "HPSCHD" without changing any controls, and notice the complex textures; try to follow some melodic lines. Second, follow the control changes proposed by KNOBS, and notice how the textural complexity diminishes. It is easier, for instance, to trace individual melodic lines, but they quickly disappear and reappear---very much like a game of hide-and-seek. It is this highly indeterminate game-like situation that draws the listener into the continually changing sound-environment of "HPSCHD", thereby helping the listener to appreciate not only the aesthetic of Cage and Hiller, but also that of their forerunners, e.g. Futurists, Dadaists, Bauhaus artists, etc.

LIVE ELECTRONIC COMPOSITIONS

The live electronic works of Cage also rely heavily upon aleatoric methods, while standard notation systems are rarely employed. Cage does not use conventional voltage-controlled synthesizers in these pieces since his sound sources are always 'natural', and subsequent transformations are generally derived from amplification-related techniques, e.g. feedback. Perhaps the two most significant aspects of Cage's live electronic works are their affinity with musique concrete sources and their almost exclusive use of amplification as means of sound modification.

"Cartridge Music" (1960) is a fine example of live musique concrete, for Cage deals exclusively with natural sounds. Since many of these sounds are practically inaudible (Cage refers to them as "small" sounds) they are picked up via phonograph cartridges and contact microphones, and then amplified. Three stages of timbral transformation result from application of various levels of amplification: moderate gain simply makes the "small" sounds audible; excessive gain produces distortion, and extreme amplification results in feedback. Movement within this timbral continuum is indeterminate, thereby yielding unique performances of "Cartridge Music."

Phonograph cartridges and contact microphones are also used in "Variations II" (1961). The former are placed directly on the strings of a piano, while the latter are placed on resonant surfaces such as the sounding board. Like "Cartridge Music," "Variations II" reveals principles of musique concrete within an indeterminate performance. Similarly, a timbral continuum based upon degrees of amplification is

(David Ernst with John Cage)

photo by: Lori Seid

established. In this instance the pianist (David Tuder) refrains from playing the instrument in a traditional manner; all sonorities arise from manual excitation of the strings, sounding board, metal cross-bars, etc.

Cage's interest in natural and "small" sounds, evident both in his tape and live electronic works, tends to emphasize our physical environment. This is the case with "Variations IV" (1964). Cage and David Tuder placed microphones in the street among pedestrians and traffic to capture the sounds of city life. More recently, Cage has been commissioned to do a children's piece for Italy, which will be the amplification of a city park; and in the Fall of 1978 Cage prepared a tape for the Merce Cunningham Dance Company that included sounds of a tea pot and water passing through conch shells.

Percussionist-composer Max Neuhaus employed contact microphones, amplifiers, loudspeakers, and various percussion instruments to produce a complex network of feedback loops in "Fontana Mix---Feed" (1964-65). The percussion instruments, with microphones attached, were placed in close proximity to the loudspeakers so that the microphones functioned as the focal point of the feedback network, i.e. the sound from the loudspeakers caused the microphones to vibrate, and these vibrations initiated a chain reaction with the percussion instruments. Pitch, rhythmic, and timbral changes therefore result from purely physical considerations: distance between instruments and loudspeakers, amplification levels, room acoustics, microphone placement, and the choice of instruments.

Cage's search for "small" sounds is not restricted to inanimate objects, and in "Solos for Voice 2" (1966) he places contact microphones on the lips and throat of each member of a mixed chorus. As in the "Aria," the vocal part is indeterminate; phonemes, syllables, and vocal effects constitute the entire 'text.' The voices are then modified by filtering and ring modulation. The resultant sounds contain obvious vocal textures along with a wide variety of less voice - oriented sonorities, especially when throat and lip noises are modulated. At times this work gives the impression of speech transmissions from astronauts, obtained by filtering (high-pass) a ring modulated voice.

Without a doubt, John Cage is among the most influential artists of the twentieth century - not only because of his tremendous sensitivity to all sounds, but also because he is aware of past artistic movements. Cage is not simply a composer, he is a sound-artist, philosopher, teacher... The philosophy and aesthetic contained in his writings are expressed in his compositions and, along with his compositional techniques, Cage has helped musicians face the reality of the twentieth century.

DISCOGRAPHY

"Aria with Fontana Mix." Mainstream 5005.
"Cartridge Music." Mainstream 5015.
"Fontana Mix." Turnabout TV-34046 S.
"Fontana Mix---Feed." Columbia MS-7139.
"HPSCHD." Nonesuch 71224.
"Solos for Voice 2." Odyssey 32160156.
"Variations II." Columbia MS-7051.
"Variations IV." Everest 3230; Everest 3132 (excerpts).
"Williams Mix." Avakian JC-1.

SELECTED BIBLIOGRAPHY

Bunger, Richard. The Well-Prepared Piano. Colorado Springs:The Colorado Music Press, 1973. continued on page 29....

Prerecorded Timing Signals: Techniques and Applications

By: John Duesenberry

Did you ever encounter a problem like this?:

You've just laid down, on your multitrack deck, a basic percussion track, the timing of which was under control of a sequencer. (Digital or analog is irrelevant at this point). Now you want to record a sequencer bass line on another track. So you set up the bass patch, and "program" the sequencer to play the line. You're ready to record; all you have to do is monitor the previous track, and cue in the sequencer by pressing its "start" button on the beat. Right?

WRONG.

For one thing, with human reaction time being on the order of 0.1 second, you can be assured that the sequencer will never be cued in exactly on the beat, as long as a human is cueing it in. Unless you are capable of anticipating the beat by .01 second (good luck!) the bass line will start a bit late and will be rhythmically out of sync with the previous track, although possibly by an imperceptible amount.

(You might wonder, if this business about human reaction time is true, why it's not impossible for a performer to stay in sync with a prerecorded track while playing in real time, etc. Well, psychoacousticians have made some pretty startling measurements about the accuracy of rhythm and tempo with which humans can play; startling if you're used to electronic rhythmic "perfection" that is. However, there is a self-correcting feedback loop between performers and other performers or prerecorded tracks that enables them to stay in sync; there is obviously no such feedback between a sequencer and anything. But we digress.)

Even if you did succeed in cueing the sequencer in with absolute precision, there would be no guarantee that the taped track and the sequencer would stay in sync, unfortunately. Slight instabilities in the tape speed and sequencer clock frequency would eventually cause rhythmic errors to accumulate to the point where the two lines would sound grossly out of phase. It's essentially a problem of phase drift that is equivalent to the problem of trying to keep two tape decks synced: you usually can't, at least not for long periods of time.

Some composers, of course, have exploited this rhythmic drift problem extensively; cf. Steve Reich's Come Out, for example. The whole rationale of the piece is that the same tape loop is played back on two decks, which start togather and then get out of phase; the results are very interesting. But if your objective is to avoid this process, you'll need to use some form of prerecorded timing signal on your tape. Such a signal is often called a clock track, click track, or a sync track.

Let's look at our problem from another standpoint. Suppose you had two sequencers available, one for the percussion and one for the bass. Would you have run them at the same time with separate clocks? We hope not. Most likely you would have ganged them together on the same clock. Both voices could then be recorded in real time, possibly saving a track. The clock-track technique is just an extension of the same idea; it enables you to do the same thing but with only one sequencer, and not in real time. The technique is essentially a 4-step process:

(1) Set up the patch you want to record (say, the percussion sound in our example.) Adjust the rate of the clock which is timing your sequencer, sample/hold, envelope generators, or whatever, to the desired tempo.

(2) Record the output of the clock (as opposed to the output of the patch) on one track of the tape. There are various methods of doing this, discussed below in more detail.

(3) That track now becomes your master timing track for subsequent recording passes. Patch the output of the timing track to the timing-signal inputs of all the devices which were formerly controlled by the original clock signal. Then play back the recorded clock track (in sel-sync mode) while recording, on another channel, the musical output of the patch which it controls.

(4) Repatch the synthesizer for whatever sound or sequence you want to record next; record on another track as in (3). And so on until all your "automated" music tracks are done. The track on which the clock signal is recorded may then be erased and used, along with whatever free tracks remain, for overdubbing of keyboard solos or other types of manually-timed material.

Regardless of the type of signal you're using as a clock track, or the applications to which you put it, the process is basically always the same as that outlined above. Let's consider some details.

Signals of several types may function as clock tracks:

A. Subaudio signals, directly recorded. Basically, this category is limited to subaudio waveforms (such as sawtooth, square, or pulse waves) in which a sharp voltage spike is present. What gets recorded on audio tape is simply a series of clicks, which function as triggering signals to the appropriate synthesizer modules. To use the clicks to step an analog sequencer, for example, is simply a matter of amplifying the clicks and patching the signal to the sequencer's "step" or "external clock" inputs. As shown in figure 1, the most convenient way to amplify is to use a mic input/ external signal preamp such as is commonly found on most synthesizers these days:

The reason for amplification is that most devices that run off trigger signals need to see a much higher-level signal at their timing inputs than the line-level signals from tape outputs.

B. Subaudio signals, recorded indirectly: If a subaudio pulse or square wave, probably the most common type of clock waveform, is recorded directly onto audio tape, what is preserved on tape is actually only its leading and trailing edges. For example, if you taped the "original signal" of Fig. 2 and played it back, you'd get something like the "recorded signal" of Fig. 2.

This happens because an audio tape deck cannot record any steady DC level. Since the amount of time a pulse remains "on" (i.e. its pulse width) is often an important parameter of a timing signal --for example, in varying the articulation characteristics of an envelope generator being clocked by that pulse-- it is sometimes necessary to use indirect methods for the storage of timing information on tape. A simple example of AM-demodulation storage will illustrate this.

Suppose that we want to use the "original signal" from Fig.2 as our recorded clock signal. What we record is not the pulse wave itself, but rather an audio-range signal being amplitude-modulated by our desired timing signal. The audio signal involved is known as a carrier signal, because it literally "carries information about" the subaudio timing signal. In other words, we record the output of the patch shown in figure 3. This creates a series of tone bursts on the tape similar to those shown in figure 4.

If this signal is now played back through a typical preamp/ envelope follower setup featured in figure 5, what the envelope follower will output will be close to the original subaudio pulse or square wave, and the envelope follower output is what is used as a timing signal.

However, you may encounter a problem when trying this. Due to the slow response time of some envelope followers, the rise-time of the "pulse" the follower puts out may not be fast enough for triggering purposes. The best way to solve this is to use a comparator in the patch, between the follower and whatever devices are to be triggered. By adjusting the comparator's threshold level, we can derive a pulse with a very sharp leading edge from the follower output; the comparator output is then used as the timing signal. (Some envelope-follower modules, such as the Aries AR-331, Paia 2720-11, and the Moog 912, feature a comparator or Schmitt trigger as part of the module for precisely this sort of application.)

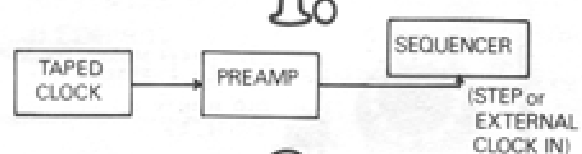
Another method for indirect storage of timing signals is via FM-demodulation. The carrier signal is frequency-modulated on tape and the tape is played back through a frequency follower or better yet an FM-demodulation device similar to those found on radio receivers. Since frequency followers are much more expensive than envelope followers, and other devices for demodulation are custom-built, this method seems to be less common among synthesists. For this particular application, there doesn't seem to be much advantage over AM-demodulation anyway.

C. Using a music track as a timing track is an extension of the AM demodulation technique; it works especially well if you have a comparator because the envelope follower/ comparator combination is capable of deriving a clean trigger off virtually any signal with a clearly-defined amplitude envelope. Thus, you might put your first percussion track down on tape and use it as the "clock" for the remaining automated tracks. One advantage is that you free a track since you don't have to save one especially for the timing signal. But a disadvantage is that it may result in all subsequent tracks being too closely tied rhythmically to the first track.

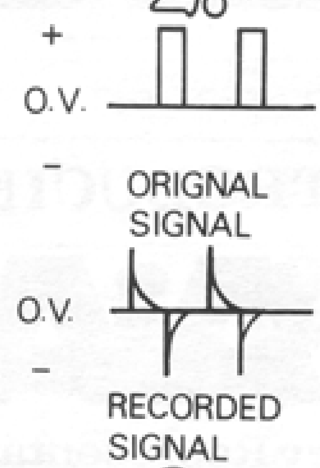
D. Audio-range timing signals. To use a signal of more than 20 hz. as a rhythmic control signal may seem senseless at first. Occasionally it's necessary to use one, however; particulary with digital sequencers. Here's a situation that could easily arise in the course of composing with a Synthi 256 digital sequencer, for example (or a Sequential Circuits 800, for that matter, if it's being "programmed" in certain way.):

Suppose you want to realize a complex rhythm sequence like the one scored in figure 6, at an indicated tempo.

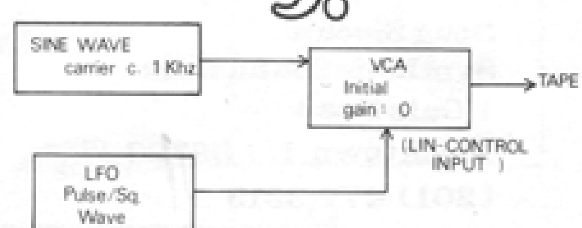
1.



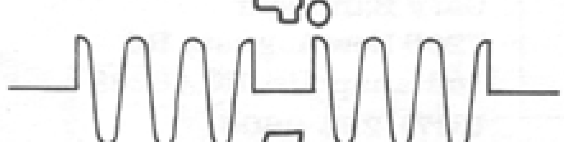
2.



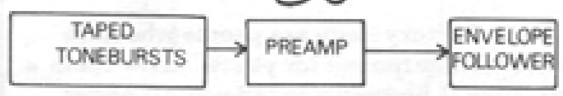
3.



4.



5.



6.

It must be understood that durations are stored in the Synthi's memory in the form of a number equivalent to the number of system clock pulses (whether from an internal oscillator or not) which elapse between two key-depressions on the input keyboard. Since the Synthi has a readout which displays the count of elapsing pulses, we can program a sequence of durations by (a) determining how many clock pulses long each note-value should be and then (b) using the input keyboard while monitoring the counter. This is not a real-time process. On input, a slow clock is used so that we can watch individual pulses elapsing; on playback, any clock speed may be use.

Now in order to realize the above sequence of note-values accurately, the quarter-note would have to be 84 clock pulses long; the triplet eighth 28 pulses long; the sixteenth 21 pulses long, and the septuplet sixteenth 12 pulses long. To play back the sequence at the indicated tempo, the clock frequency must be exactly 84 hz! This is true whether we're talking about a recorded clock track or not. Assuming we're using a taped clock, we now have several alternatives:

(1)directly record an 84 hz pulse or square wave as the clock signal.

(2)directly record an 84 hz sine or triangle; derive timing pulses from it via a caomparator.

(3)indirectly record clock signals by the AM or FM method as described above. This may or may not work well, depending on how well your demodulating device can follow rapid changes in the incoming signal. Generally, method (1) is adequate, provided that the recorded signal is clean.

In closing, here are a few cautionary notes based on personal experience with recorded timing signals.

(1) Record and play back the signal with just as much care about avoiding distortion and optimizing the S/N ratio as you would take in recording music signals. A noisy clock signal, one that has been over-recorded, or one with dropouts can ruin your composition. Noise can cause random spurious triggering; dropouts can cause subsequent tracks to "drop a beat." Keep in mind that tape noise is amplified and added to by the preamp through which you play back the signal.

(2) Make sure your clock signal is clean before you start laying down any music, by setting up a test patch and playing the tape back for its full length; listen for dropouts and unwanted triggers. If these occur you may be able to compensate by adjusting the signal level. If you can't, maybe you need to clean or demagnetize the heads, buy a better - quality tape, or re-recorded the signal more carefully.

(3) If you're going to use the AM-demodulation method, use a relatively high-frequency (c. 1Khz) carrier signal. If the carrier frequency is too low, an envelope follower may "ripple" in response to it. (i.e., may follow individual cycles of the signal instead of detecting its overall amplitude contour.

(4) Record a length of clock tape that exceeds the anticipated length of your composition. It's better to waste tape than to run out of clock signal.

(5) Once you've recorded a clock signal, you're stuck with an inflexible tempo. Make sure you set the clock frequency you want before recording.

(6) It's possible to record the clock signal and the output of a patch it's controlling at the same time on different tracks. This saves you one pass in the overdubbing process. However, if you do this, don't record the two signals on adjacent tracks. If there is any crosstalk into the clock channel during the recording it could ruin the signal for use in later passes.

If you're careful about these considerations, the use of recorded timing signals can considerably expand the musical capabilities of your synthesis system.

LETTERS...

MORE TAPE EXCHANGE

Now that I've had some time to read the recent Polyphony, and have had some time to absorb some of the discussion that has taken place, I would like to say a few things.

First of all, concerning Craig Anderton's letter- recent news coming out of Synapse indicates that there is already in existence a company that will distribute and promote (on a limited scale) music that an artist has produced (that is, recorded and pressed out of his own pocket). I have no further information, as I have not yet received a reply to my inquiry. This should be investigated further.

Craig's letter was well thought out. But, there is one flaw that exists. An important flaw. He talks about a group of insiders who make decisions as to whom else will be included in future samplers, etc. If homebrew music is to be encouraged, there can be no such cliques formed. Otherwise, what is the difference between this operation and the big record companies, other than size?

That is why I suggest a system similar to that which Trouser Press employs- the performing artist pays all recording and pressing costs. He then purchases advertising space in the magazine (at a reasonable rate; some kind of special rate for such artists). The recording is then reviewed on a take it or leave it basis, just like the records the biggies send in. As an alternative, the magazine could distribute a limited quantity of the recordings.

It's just that I percieve a danger when a publisher gets involved financially with such a venture. If you were to start laying out cash to support people while they do their work, then you can only act in a manner that will get you your money back. But, if you were to stay aloof from the production and recording end of the business, then you are in a position to do more good.

Another danger of the system wherein you lay out the cash is that you end up listening to a lot of crap. Apple Records had this problem. The Apple Corporation, as you recall, was the Beatles' attempt to give an avenue of distribution to any and all. As a result they were inundated by a lot of crap, and there was less for deserving artists. And eventually, bankruptcy.

But, if the artist is required to put up his own cash, then he will think twice about what he is doing. He will be in a position where he will really believe in his art. He can get feedback from a number of sources (my synthesizer exchange is one such source- through tape exchange).

The most important aspect of such an enterprise is the reviewing of material. After all, it is the review that will eventually persuade people to lay out a few bucks. And to depend on the hit or miss reviewing of all publications is to miss out on good music, or waste money on bad. <u>ALL</u> such music that is submitted must be reviewed. That is why I suggest a paid ad by the artist- to help cover the costs of the pages needed to accomodate the reviews. This way you are still making money, and so still able to continue doing this.

There are a number of other benefits related to such a system, including a possible increase in circulation due to people who want to know what is going on in homebrew music. The field has come to where it is a dominant force in electronic and synthesizer music, perhaps in the whole field of pop music. If not now, very soon.

Returning to Will Nordby's library system and the magazine devoted to audio visual art, no reason why Polyphony cannot begin to accomodate. Slowly, of course. Here again, as with records, there can be room for videotape reviews also. Todd Rundgren has performed on video tape. There have been one or two others, but I don't have the information available.

There are two approaches to the videotape thing- recordings of bands in concert (very unimaginative, and probably pretty dull after a couple of showings), or there is the more artistic approach. For example, I am currently working on putting together a video show of some astronomical phenomena which I will then do a soundtrack for. Suppose I thought this was good enough to sell- I could operate through the homebrew music system; submit a copy for review, buy a paid ad, and so on. I'm sure that there are other even more imaginative things going on with video.

So again, you are in a position to be a leader in this field. And without losing your commitment to synthesizers. Yours could be a highly technical magazine, and at the same time devote considerable space to the promotion and consideration of home produced art, because in many instances it is the artist working in his living room who is the leader in the avant field.

Just a little imagination, and you could go a long ways beyond what other magazines are doing for the artists. As I said earlier, I feel the crux of this is the artist footing his own

continued on page 38....

POLYPHONY 'Bookpage'

BOOKPAGE is provided to help Polyphony readers find the special publications their projects require. The items we currently stock are a must for every active music experimenter's lab or studio bookshelf. We are looking for additional items which may interest you. Let us know of books (and publisher, if you know it), records, tapes, or anything else which would be useful to you or others. Circuit boards for Polyphony projects? Patch chart pads?

TO ORDER: Use the convenient fold-up in the center of this issue or, if someone beat you to it, the order form at the bottom of the next page. We cannot invoice; payment must be included with any orders. As a bonus, we pay postage on shipments within the U.S.. Foreign customers must enclose $1.00 per item to help defray additional costs. ALL foreign orders MUST be paid in U.S. FUNDS, preferably drawn on a U.S. bank. Better yet, send certified check or money order.

Craig has prepared two fine books to serve as continuous reference manuals in building, and using musical electronics. Electronic Projects for Musicians is a perfect introductory manual for the musician with no previous experience in electronics. Home Recording for Musicians outlines the selection and operation of recording equipment for the musician with BIG ideas and small budgets. The Craig Anderton Music Tape is a collection of original compositions recorded by Craig while he was writing HRFM.

#EPFM	$7.95
#HRFM	$9.95
#CAMT	$5.95

Order CAMT with either book and deduct $1.00!

Howard Sams' cook books are an excellent way to stock your library with materials that are not only heavy on theory, definitions, and educational material, but chock full of practical applications as well! These books can easily replace stacks of manufacturers data sheets and applications notes all in an easy to use reference. "Audio IC Op Amp Applications" is a smaller version of the Op Amp Cookbook. This edition contains only the sections which are pertinent to audio circuitry.

#OACB Op-Amp Cookbook	$12.95
#AFCB Active Filter Cookbook	$14.95
#CMCB CMOS Cookbook	$10.50
#AUOA Audio Op-Amp	$ 7.95

stamp

YPHONY

OX 20305

naCity, OK 73156

a
er
on
ns
hes

SUBSCRIBE to POLYPHONY

T.M.

CONVENIENT ORDER FORM

FOR POLYPHONY BOOKPAGE

QUANTITY	ITEM	PRICE
______	EPFM	$ 7.95
______	HRFM	$ 9.95
______	CAMT	$ 5.95
______	OACB	$12.95
______	AFCB	$14.95
______	CMCB	$10.50
______	AUOA	$ 7.95
______	CYCLO	$39.95
______	EGAH	$10.50
______	Source	$ 4.00
______	Bind-S	$ 2.50
______	Bind-L	$ 4.95
______	TOTAL	______

Foreign Postage Add $1.00 per item. TOTAL ______

SUBSC

to PO

The magazine for
you're a beginner, adva
material, Polyphony ha
includes original circui
equipment design for a
theory, recording techn
digital synthesis equipn
and record with electro

Join the growing l
the latest developments
person's guide to new ic

SUBSCRIPTION R

Mailed in the USA
Mailed to Canada c
International Subsc

Don't miss a

ENTER MY ONE YEAR

NAME: ______

ADDRESS: ______

CITY: ______

CRIBE

LYPHONY

T.M.

he electronic music enthusiast and experimenter. Whether
ced user, experimenter or into home recording of your own
something for you. Now published bi-monthly, each issue
designs, modifications, expanded applications and custom
ariety of electronic music equipment; electronic and music
ques, patch charts for synthesizer and programming for
ent. All written by those who design, experiment, perform
ic music equipment.
st of subscribers who find that Polyphony is the source of
n electronic music. Subscribe to Polyphony, the creative
eas in electronic music.

TES:

....	$8. 00/year (6 issues)
Mexico	$9. 00/year (6 issues)
iption	$10. 00/year (6 issues)

informative issue. Subscribe Today!

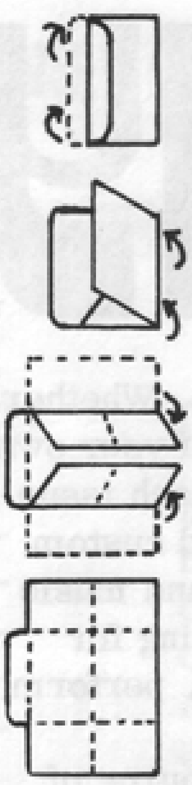
UBSCRIPTION TO "POLYPHONY", ($________ enclosed)

____

____

__________ STATE: ______________ ZIP: __________

USE THIS HANDY SUBSCRIPTION ENVELOPE TODAY!



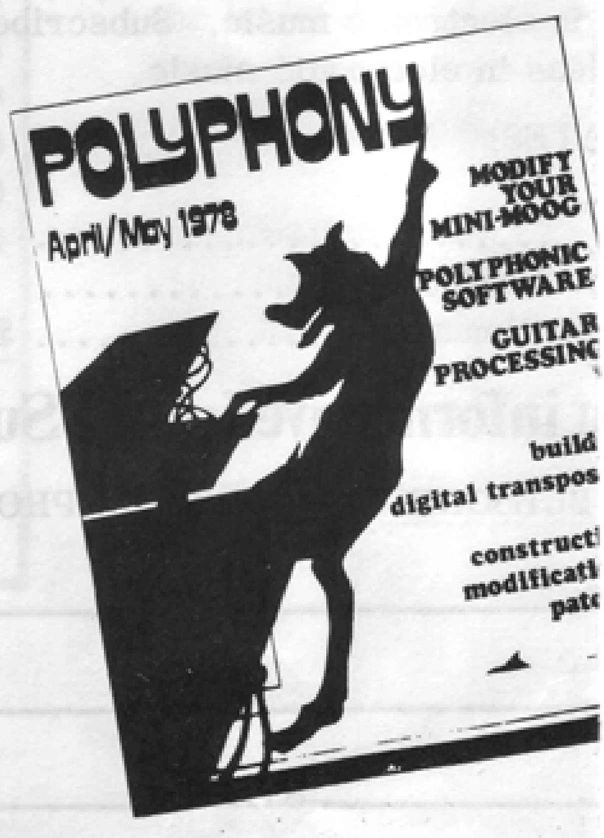
from:

POL

P. O. BO

Oklahon

Circulation Dept.



Jack Darr's Electric Guitar Amplifier Handbook is stuffed with schematics for popular amplifiers and effects units from major manufacturers. This book is an aid to service shops as well as the advanced experimenter. Contains thorough discussions of typical amplifier problems, locating them, and repairing them.

#EGAH $10.50

Audio Cyclopedia is subtitled "the most comprehensive and authoritative reference volume on audio ever published". With 1760 pages, 3650 entries, and hundreds of illustrations and schematics, that description can't be too far off.

#CYCLO Audio Cyclopedia $39.95

The Source is finally here! Over 125 pages of patches, synthesizer notation, and computer software for music applications. Regardless of the synthesizer you own, these patches serve as an intense study of the capabilities of synthesizers. As an inspirational guide or departure point for deriving your own patches, The Source can't be beat! Convenient thumb index leads you to the patch you're after! Tonal patches, sound effects, patch techniques -- it's all here!

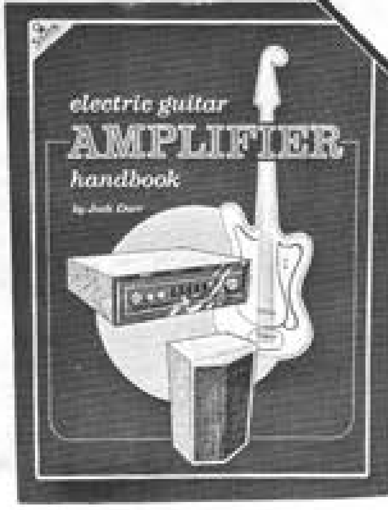
#Source $4.00

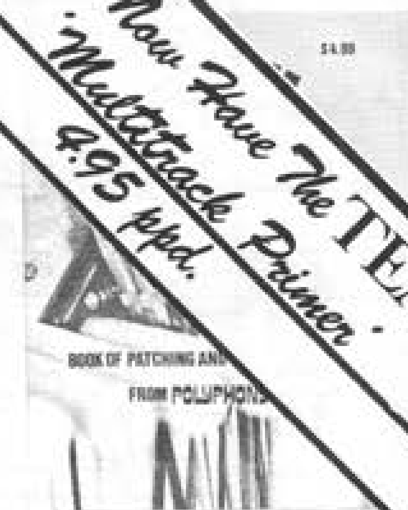
We have new binders to hold the large format Polyphony -- or any other 8.5" X 11" publication. Black vinyl over hardboard is silkscreened with the Polyphony logo and a spot to index the contents. Wire-type magazine holder eliminates punching your magazines for ring binders, and keeps issues like new for reference use.

#Bind-L $4.95

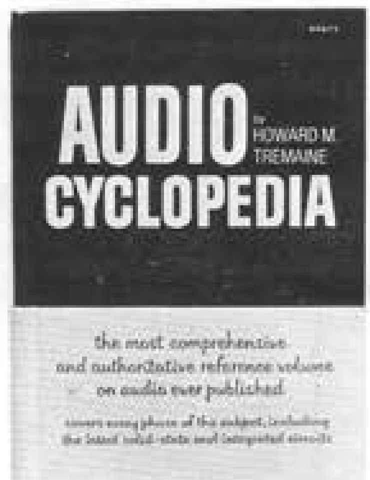
ALSO: We have a few of our old style magazine binders left. These were designed for our 5.5" X 8.5" issues, but will serve well to hold other small publications, such as manuals from Paia, Strider, or old copies of Musicians Guide, etc. 1/2 price while they last!

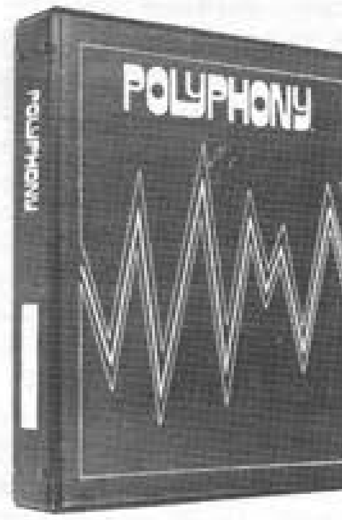
#Bind-S $2.50





We Now Have The TEAC 'Multitrack Primer' 4.95 ppd.





The wide variety of practical applications and construction projects in past issues make a binder full of Polyphonys a frequently used reference to keep near your synthesizer or workbench. All Polyphony back issues are still available; there was one issue in '75, four in '76, two in '77, and five in '78. For a more complete rundown on the features in each issue, send us a SASE and request our "back issue list".

Order by issue cover date $2.00 each

ORDER BLANK

NAME: ____________________

ADDRESS: ____________________

CITY: __________ STATE: __________ ZIP: ____

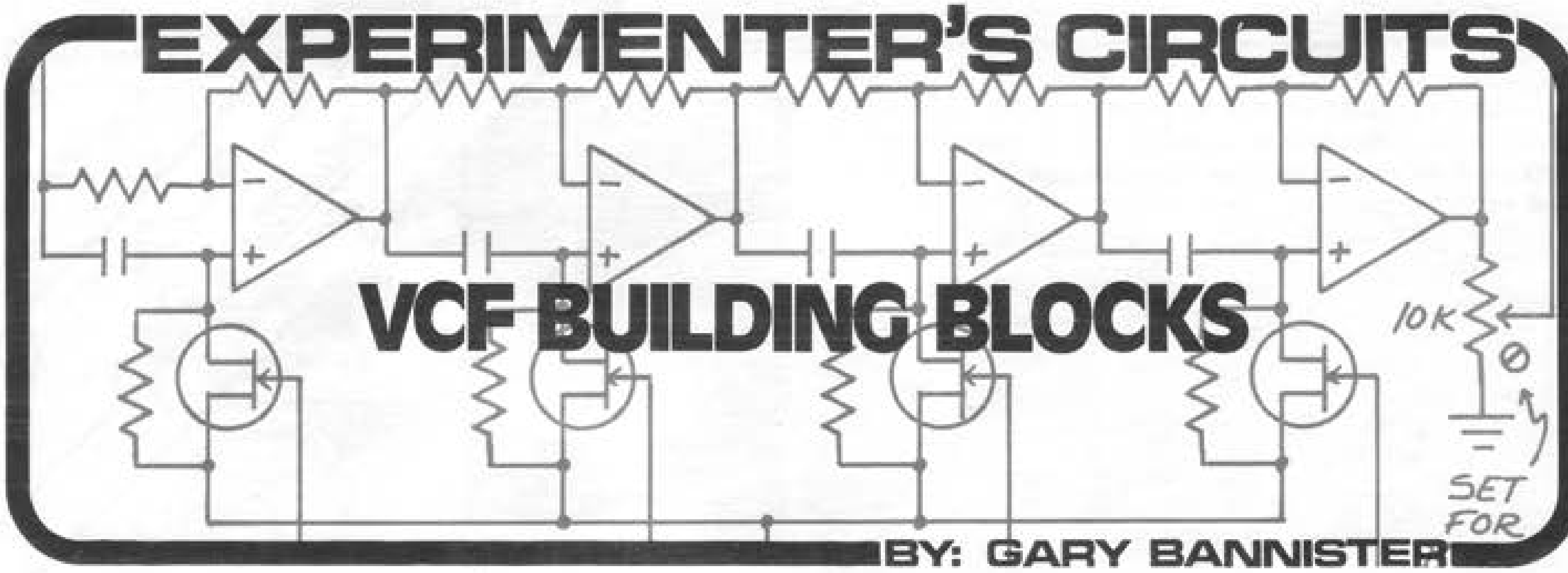
SEND TO: POLYPHONY Bookpage, P.O. BOX 20305, Oklahoma City, OK 73156

QUANTITY	ITEM	PRICE
		$
		$
		$
		$
		$

Foreign Postage ($1.00 per item) $ ______

TOTAL $ ______

FOREIGN ORDERS BY CERTIFIED CHECK IN U.S. FUNDS



EXPERIMENTER'S CIRCUITS

VCF BUILDING BLOCKS

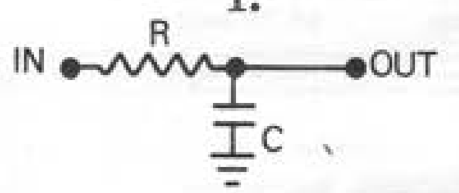
BY: GARY BANNISTER

The VCA's presented in the last two columns should have whetted your appetite for voltage controlled devices, and you should have some experience with the CA3080 by now. I'm sure you're aware that amplitude control may be one of the least critical parts of electronic music. Maybe even more important than pitch is harmonic content, so this time we'll try a voltage controlled filter.

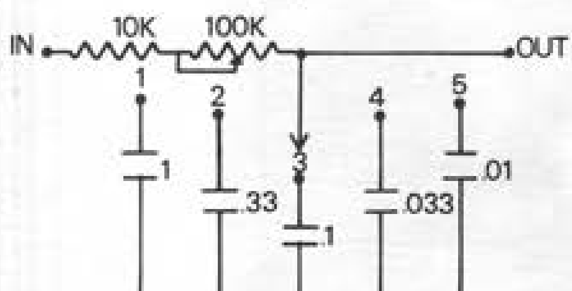
Obviously, we're going to use the 3080 as the basis for this module. However, it's not a good idea to jump into anything without looking first, so we'll look at some other filter designs first.

The most common type of filter is the simple RC network displayed in figure 1.

1.



2.



You've all seen this somewhere. One very common use is power supply decoupling. If you think of DC as absolutely Zero Hz, then this becomes a filter that allows only DC to pass. Anything higher in frequency is attenuated. This, then, is a simple low pass filter.

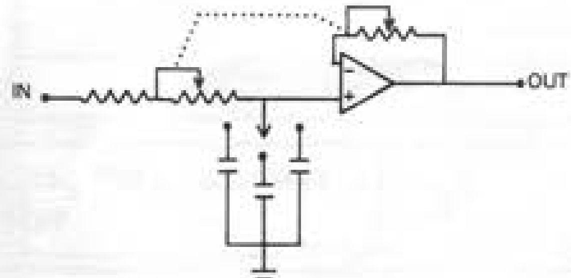
Low pass filters can be 'tuned' two ways: by changing the capacitor (C) or the resistor (R). As shown in figure 2, you can put several capacitors on a rotary switch to change ranges in steps, and add a potentiometer to smoothly vary the frequency.

Not that this is extremely important, but a similiar unit is found on some of the Moog Modular Accessory Panels. It's effective for its simplicity.

If you build this little system, you will immediately find its drawback: With that resistance in series with the signal path, the amplitude of the output signal drops considerably. Fortunately, modern science has a cure!! We should install an op-amp as in figure 3.

This has some complications, too. The dotted line means that both pots are turned by the same knob. Not difficult, but bulky.

3.



Besides, this filter is only 6db per octave. We could sure use 12db; even 24db if we can get it.

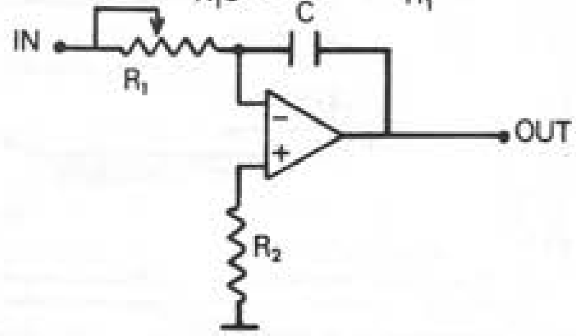
AND WE CAN. Let us look at another type of filter in figure 4, the integrator.

While rather difficult to explain, it's easy to use. Most multimode filters are made of these integrators, in fact. ARP, Aries, Paia, and Oberheim all use variations. Moog gets a multimode response by cross coupling a separate high pass and low pass filter. However, we're not looking for a multimode filter. What we would ultimately like to end up with is a voltage controlled 24db per octave low-pass filter. Actually, by using a group of integrator 'blocks' we can produce just about any filter we need. Figure 5 shows just such a block.

There are two things to notice about this circuit. First is the FET (MPF 102). This is hooked up as a source follower (note that the resistor goes to -V, not ground). This gives us a synthetic FET opamp. Note, too, that the integration capacitor (470pf) goes to the gate of the FET, not to the input of the opamp. This serves a very

4.

$$OUT = -\frac{1}{R_1C}\int IN\,dt - \frac{R_2}{R_1} IN$$



important purpose. Unless specially selected, opamps such as the 748 or 301 have a certain amount of INPUT OFFSET. Simply put, the opamp requires a certain small signal just to operate. Without the FET, this offset must be supplied by the 3080, robbing us of the extreme low end range the 3080 is capable of giving us. As well, the high impedance of the FET lets the 3080 work practically effortlessly. Adding the FET allows the 3080 to work with high precision over a very wide range. In fact, this circuit, when properly assembled, can sweep FROM 4 HZ TO GREATER THAN 20 KHZ IN ONE RANGE!!

"Properly assembled." The great grabber. A certain amount of care <u>must</u> be taken with this circuit. With the high impedance of the FET this circuit may tend to oscillate. To prevent this, the FET, 470pf capacitor, and the 748 opamp should be as close to each other as possible. Make your wiring as short as possible. Do not use great globs of solder. As well, I found a few opamps that would not work in this circuit. They worked fine every where else, just not here. It would be a good idea to socket the FET and opamp so that exchange is easy.

Now the easy part. This is one 'block' for a filter. It has a 6db per octave response. If you want 12db, put two of these in series. 18db gets three and 24db gets four. It really is about as simple as that. A block diagram for a 24 db per octave filter appears in figure 6.

Note the connections to point A on stage 1 and 4. Let's go into some detail. The connection to the first stage is to stabilize things. The unlabeled resistor should be approximately the same size as the resistor 'R' in FIG. 5. It may be juggled somewhat to produce unity gain through out the system. The connection to the fourth stage is the RESONANCE ('Q') control. Figure 7 outlines the details for the output stage. The 1 meg resistor is the front panel RESONANCE control (I don't like the term 'Q'. It's meaningless for lowpass filters.) The 50K trim pot should be adjusted so that the RESONANCE control gives the greatest effect without oscillating. In fact, it may be possible to get this filter to oscillate and produce a good sine wave, so adjust the trim pot so that maximum RESONANCE produce a good clean sine wave over the full control voltage swing.

You may find some loss of gain with this filter. You may not get out the same amount that you put in. This can be adjusted best by changing resistor 'R' in stages B, C, and D. The circuit, as it stands, was designed to work with PAIA equipment. If you have ARP, Moog, or something else, you will have to do some slight adjusting. This amounts to changing the value of 'R' so that the 3080 does not distort (remember? about 100mv input?). Below is a table of suggested values for 'R' for different input voltages.

.5 volt	5k
2.5 volt	25k
5.0 volt	50k
10.0 volt	100k

When using other equipment, start with ALL 'R's the same, and then adjust for unity gain as already discussed.

There are some nifty things that can be done with this circuit. High pass filters can be derived as shown in figure 8.

Note the addition of the extra FET on the + input of the opamp.

The hipass and lowpass can actually be combined into one circuit using figure 9 as a guide.

Notice how the SPDT center off switch is connected. With the switch in the center off position, the filter acts as a notch filter or phase shifter.

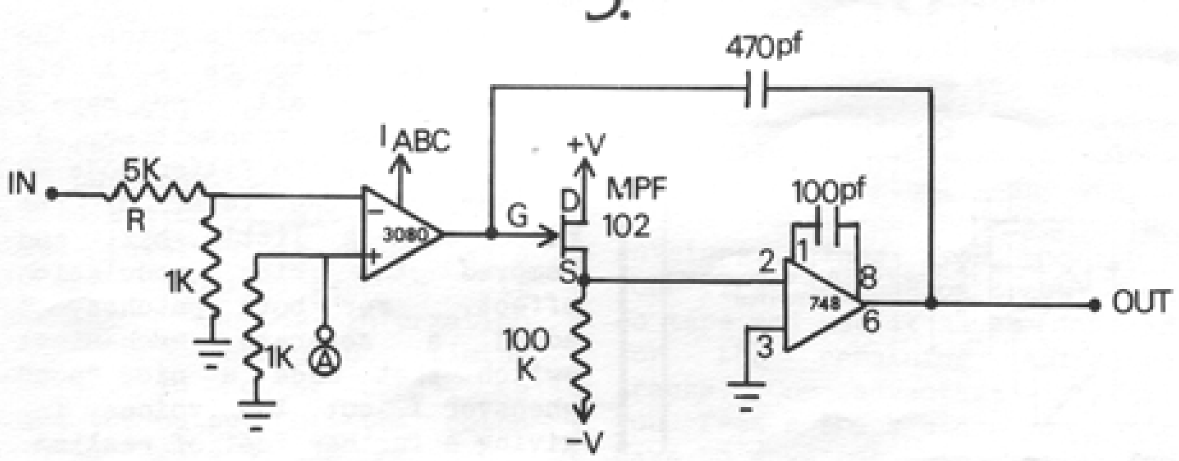
As a matter of fact, I built two of these filters in a 12db/octave configuration (by leaving out stages B and C), and by cross coupling them in the manner depicted in figure 10 you can get a very 'wide' band pass filter that adds a certain haunting straining to horn-type sounds.

The resistor reduces the amount of ADSR fed to the hipass so that it does not sweep as high as the lowpass. The low pass is responsible for the 'blat' and the high pass adds a certain 'thinness' that imparts a strained sound to the patch.

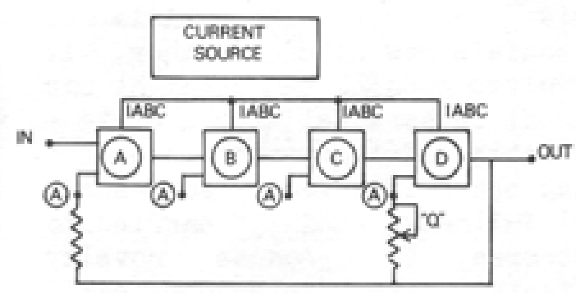
I purposely have left the I_{ABC} sources until last . Not that they're unimportant, but they will most likely be slightly different for almost every brand

continued on 35....

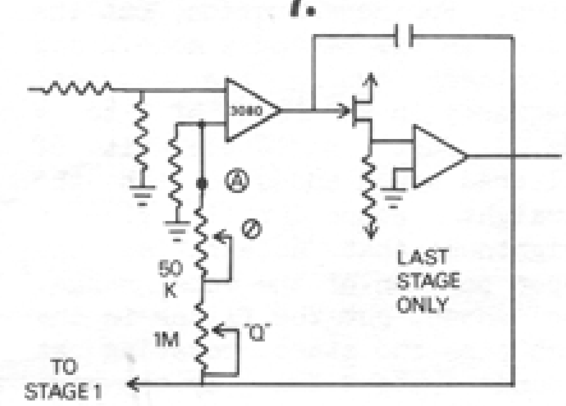
5.



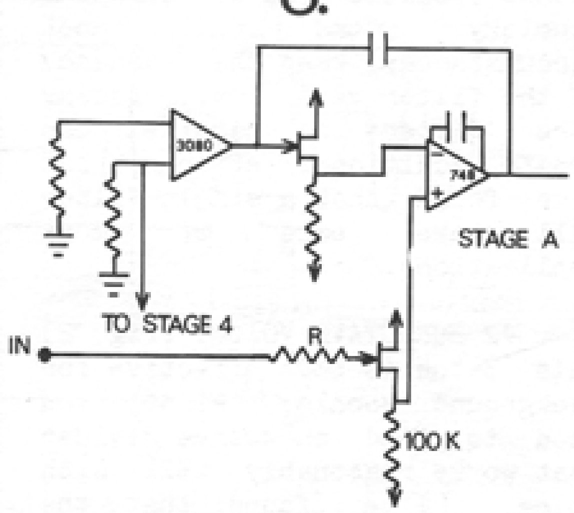
6.

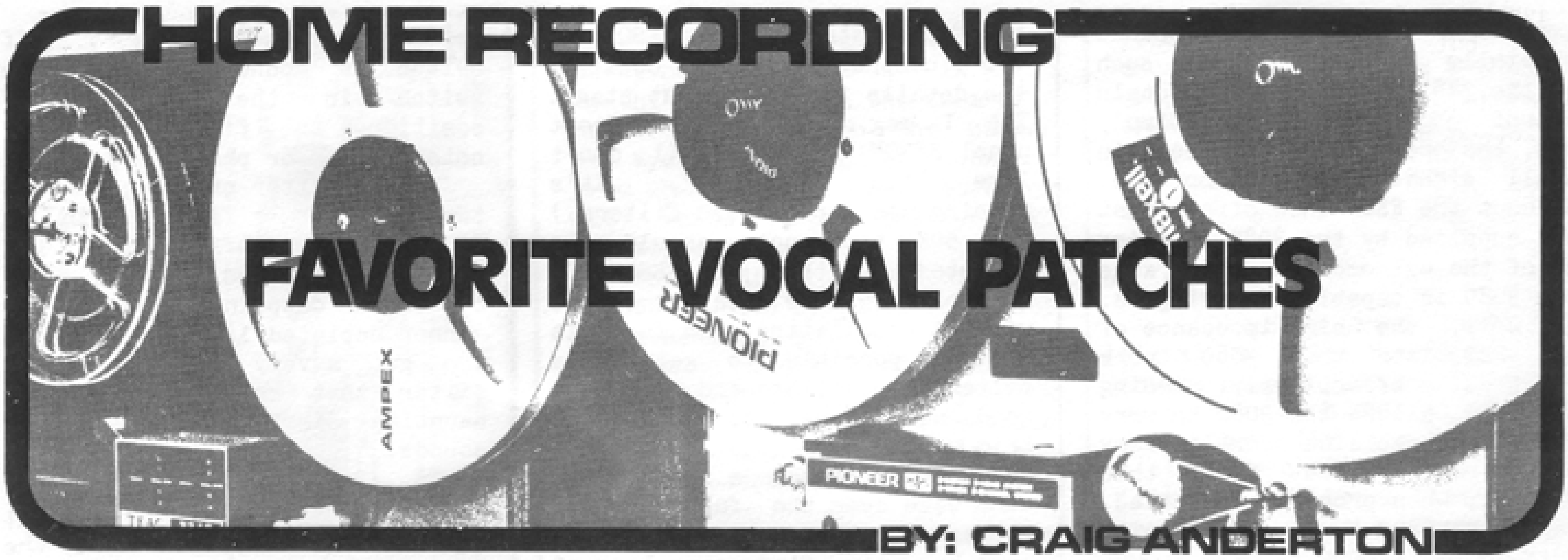


7.



8.





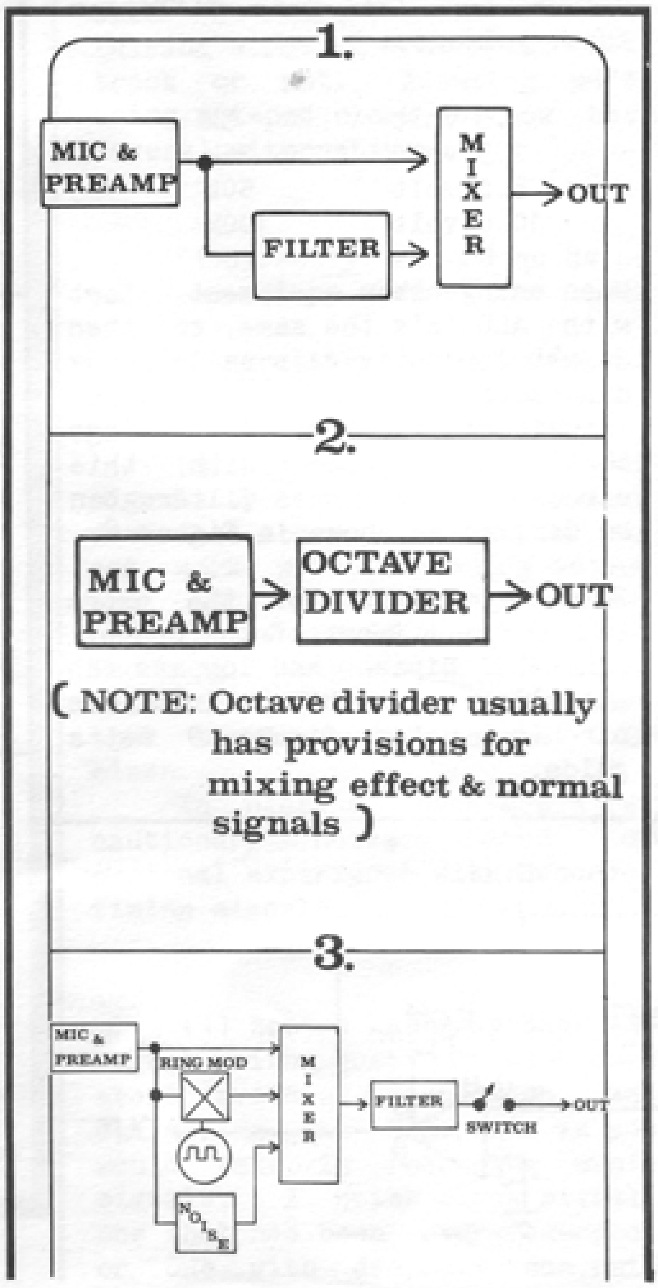
FAVORITE VOCAL PATCHES

BY: CRAIG ANDERTON

During the years I've been working with getting a good vocal sound, I've come up with some favorite patches. Most of them require synthesizer-type modules, so I figured this article would be of great interest to POLYPHONY readers who are into home recording. I hope you enjoy using these patches - they'll really perk up your vocal sound.

#1 PARALLEL FILTERS. No, this is not the title of Blondie's new album; rather, it involves running the straight mic signal in parallel with a state - variable filter to accentuate weak spots in a vocal, add sheen or fullness, and if carried to extremes, to produce novelty effects like "telephone call" voices. Figure 1 shows the basic patch. For accentuation, put the filter in the bandpass mode, and experiment tuning the resonant frequency in the range of 1 to 4 KHz. Just a little bit of filtered sound added in with the straight sound will give a brightness that accentuates the upper portion of the vocal range. For sheen, put the filter in the high pass and start boosting at around 1 to 3 KHz. By the way, for achieving the most subtle sounds possible (you don't want a gimmicky sound under most circumstances) keep the resonance of the filter very low. Adding more filters in parallel can create additional effects, but I've found that a single filter will take care of most applications.

#2 SUBOCTAVE VOICE. (fig. 2) This patch is most effective for background vocals. First, you need to find an octave divider that works reasonably well with voice. I've found that the Mutron Octave Divider, with the stabilize switch off, does a pretty good job; nonetheless, be prepared for the fact that no octave divider is perfect, and you'll have to pay some attention to giving it the type of input signal it wants. Second, you have to adjust the blend between the suboctave and normal voice sound VERY carefully. I generally use this effect with "ooooh" and "ahhhh" type of background parts; I mix in just enough octave so that if it's taken out, you notice something missing. Avoid making it prominent enough so that you can listen to the vocal and say, "hey - listen to that suboctave effect". When used sparingly, the result of adding the suboctave is a very thick type of vocal sound that works amazingly well with chordal parts.



#3 "COMMUNICATIONS" VOICE. I was recently working on a cut called "Airplanes" that required a 'pilot to tower' conversation to appear at certain points in the piece. The process of simulating it was pretty interesting.

Figure 3 shows the basic patch. For the pilot's voice, the filter was set to bandpass mode in the lower midrange region, with a moderate Q (adjust for whatever sounds most like a communications mic to you). While this does an OK type of pilot voice, more drastic measures are required for authenticity. The ring modulator, if mixed in as a background effect and driven with a square wave carrier, gives some of those weird sideband effects you encounter in radio communication. The noise source adds a lot to the authenticity, too.

For the tower's voice, the fidelity needed to be a little better (after all, they have a ground based transmitter...). So, I changed the filter mode to lowpass, cut the level of the noise down a little bit, and removed the ring modulation effect. For both patches, I added a separate mechanical switch that made a nice "pop" whenever I cut the voices in, giving a further feel of realism.

Maybe that's getting carried away, but I've fooled a lot of people into thinking I had recorded an actual tower to pilot communication instead of recording it in the studio!

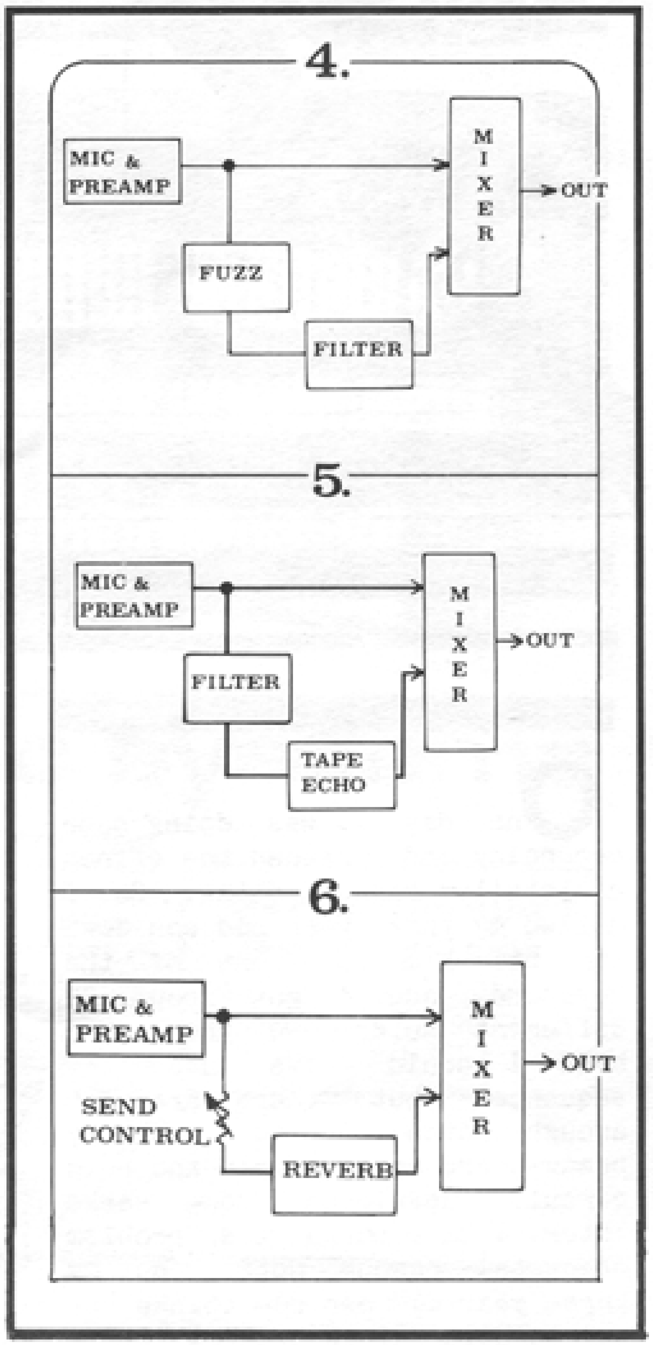
#4 "LIVE" VOCAL EFFECTS. Here's another oddball requirement that elicited a somewhat unconventional approach: I wanted to get a vocal sound that gave the effect of being sung through a PA at a club. One thing I've noticed about cheap PAs is that certain resonant peaks in the system will distort before anything else does. To create a synthetic distorted peak, I used a fuzz/filter combination as shown in figure 4. You'd think the filter before the fuzz would be better, but for some reason the set up shown gave best results. Again, this has to be mixed in subtly unless you're really going for a maxed out, punk rock vocal effect (and it does come in handy for that).

#5 SELECTIVE ECHO. Here's a good one for pop music people; the patch works best with reasonably clean vocals in a ballad-like situation (but don't let that discourage you from trying it in other contexts.) Referring to fig. 5, put the filter in the bandpass mode, right where the sibilant range occurs in the voice (try around 4 to 5 KHz for starters). Then feed this into an echo unit. Tape echo is best for this, since analog delay lines typically don't have sufficient high frequency response.

Normally, the vocal does not produce enough energy in the upper end of the audio range to make a noticeable echo. But when you hit an "S", the filter lets it through and gives an echoing "S" sound that is really nice to feed in with the original vocal. It's even more effective if this echo is panned off to one side (if the vocal is in the middle), with conventional reverb being panned off to the other side.

#6 VARIABLE SPEED VOCAL DOUBLING. I'm certainly glad manufacturers are starting to include variable speed capabilities in home recorders; I've always felt variable speed was right up there with head cleaner in terms of usefulness. Anyway, everybody knows the old trick about recording a vocal, then going back and re-recording another, identical vocal on a separate track and combining the two togather for a "thicker" sound. If you have speed control, though, try recording the second vocal a little faster or a little slower than the original...it makes just enough of a difference to add some extra richness to the sound.

#7 THE EXPLODING REVERB VOICE. Figure 6 shows a fun patch, but one that is more applicable to mixdown rather than recording - unless you can sing and fool around with the controls at the same time. Here we have a reverb system in parallel with the mic, and a reverb send control. Big deal, you say. Well, here's how to make it a big deal: turn up the reverb send very briefly whenever your voice produces a plosive sound (b, p, etc.). This will shoot a jolt of reverb down the reverb unit and into the output, especially if you really crank the reverb up at the appropriate time. Leave the reverb send control down at all other times, except during those peaks you want to accentuate. This effect softens the plosive, and yet, accents it at the same time. It's hard to explain the right way to get this effect on paper, so I'd suggest experimenting with it to discover what works for you. If you have any favorites send 'em to Polyphony's patch section, I'm sure other readers would be interested.



CAGE

.... continued from page 15

Cage, John. A Year From Monday. Middletown, Conn.: Wesleyan University Press, 1961.

_____. "The Future of Music." Numus West, 5:6.

_____. Notations. New York: Something Else Press, 1969.

_____. Silence. Middletown, Conn.: Wesleyan University Press, 1961.

_____ Writings '67 - '72. Middletown, Conn.: Wesleyan University Press, 1973.

Cage, John, and Hiller, Lejaren. "HPSCHD." Source 4(1968): 10-19.

Coutts-Smith, Kenneth. Dada. New York: E.P. Dutton, 1970.

Cunningham, Merce. Changes: Notes on Choreography. Ed. Frances Starr. New York: Something Else Press, 1969.

Fuller, Buckminster, Operating Manual for Spaceship Earth. Carbondale: Southern Illinois University Press, 1969.

Kostelanetz, Richard. John Cage. New York: Praeger, 1970.

_____. Master Minds. New York: Macmillan, 1969.

The John Cage Catalog. New York: C.F. Peters, 1962.

Tomkins, Calvin. The Bride and the Bachelors. New York: Viking Press, 1965.

SCORES

"Cartridge Music." Peters P6703.

"Fontana Mix." Peters P6712.

"HPSCHD." Peters P6804.

"Solos for Voice 2." Peters P6768.

"Variations II." Peters P6768.

"Variations IV." Peters P6798.

"Williams Mix." Peters P6774.

BUILD IT!

SAMPLE/HOLD AND NOISE SOURCE

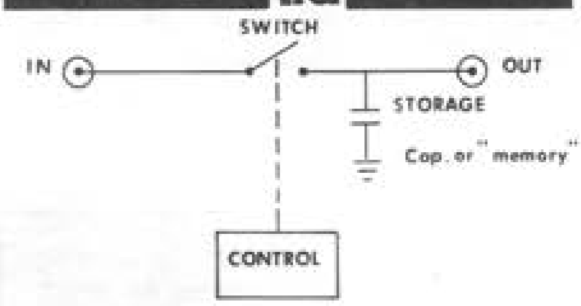
By: Larry Pryor

One day I was doing some recording and I needed the effect of totally random notes. So I called my three year old son down and put him in front of the keyboard, and I got about 37 different notes. On the other hand I could have used my sequencer but there are not enough notes. So out came my bread-board and a sample and hold circuit developed some weeks later. I have cured the problem of total random notes, but my three year old son now thinks he has to help dad record!

Figure 1a shows the basic sample and hold circuit. It requires a switch and a capacitor. The switch can be a relay or a toggle switch, and the capacitor should be a quality low leakage type. For a source voltage, let's use a sawtooth. See Figure 1b. By closing and opening the switch, a voltage is stored on the capacitor; if there are no leakage paths, this voltage will stay there until it has been changed by closing and opening the switch again. The resultant voltage that appears on the output is a stair step type waveform. Most analog synthesizer keyboards take advantage of this principle. The switch is a step pulse from the keyboard and the voltage input comes from the resistor divider string. Closing a key samples the voltage and holds it until the next sample or key depression.

From this simple beginning, several improvements can be made to make this sample/hold more versatile and stable. Additional circuitry is required to allow standard interface to additional synthesizer circuitry. It's pretty hard to find an "ideal" capacitor for the storage of the sample voltage, and input

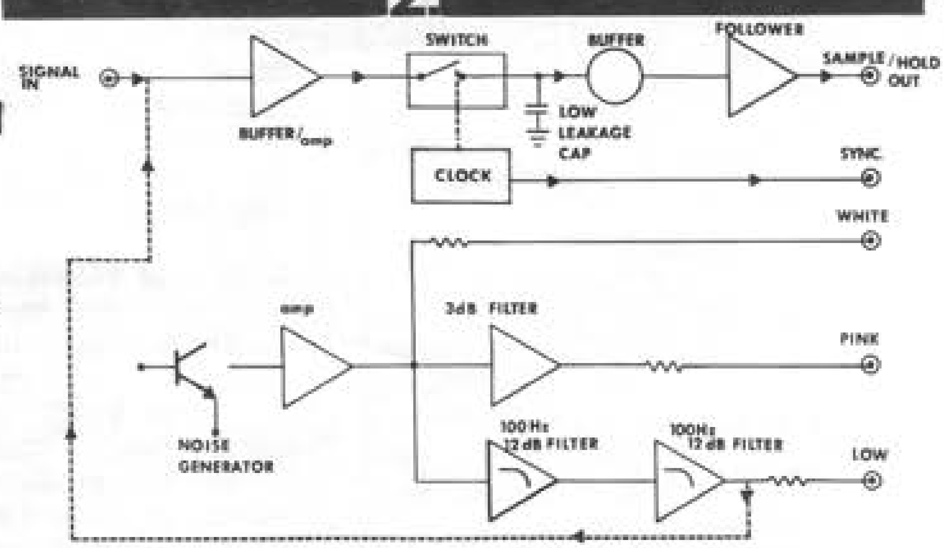
1.a



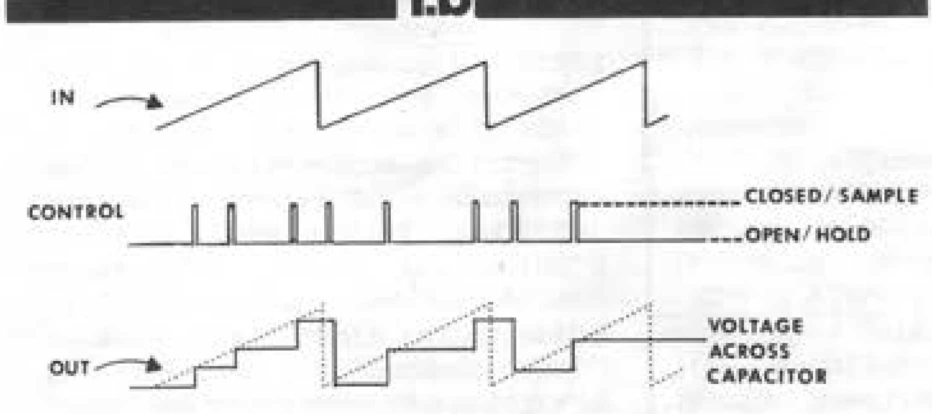
buffering is always nice to add. Another nice thing to have is an automatic switch. To make the output truly random, the voltage input should be of a random nature. I have chosen in my unit to use a low random noise (or "red noise") signal for this. Figure 4 shows what a low random signal looks like. Figure 2 shows an overall block diagram of the sample/hold and noise source module.

The sample and hold circuit is pretty much a straight forward design. The low random noise source goes back a few steps and is derived from a white noise source. Since I started with white noise, I took advantage of this and brought it out to a front panel jack. While you are doing that, you might as well add a few more components to make a pink noise filter. Then we get down to the low random noise which uses a 24 dB filter tuned to about 100 Hz. Now we have a multi-purpose module that includes sample and hold, white noise, pink noise and low random. See Figure 3 for the complete schematic.

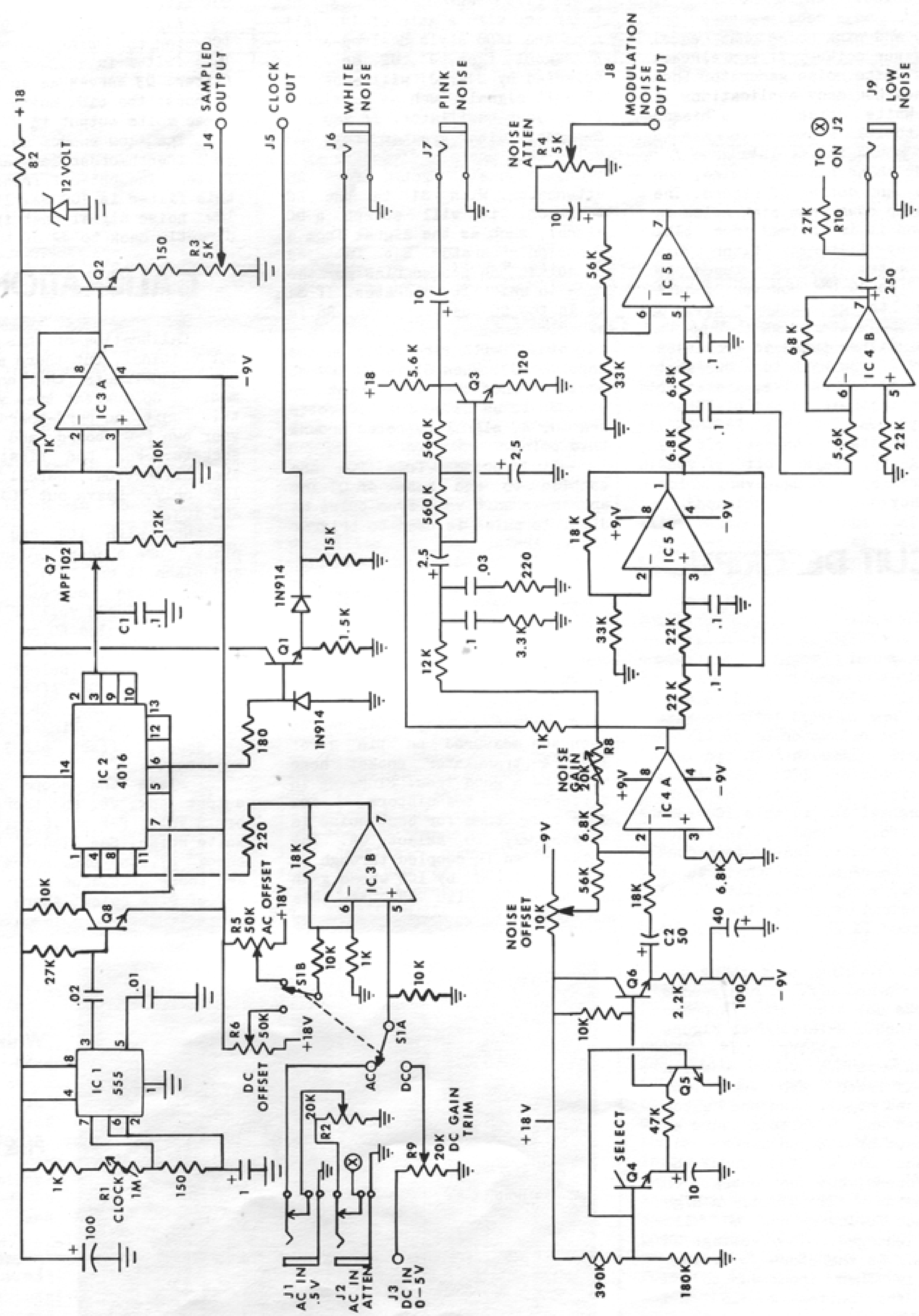
2.



1.b



3.


IC 1 555
IC 2 4016
IC 3 A
IC 3 B
IC 4 A
IC 4 B
IC 5 A
IC 5 B
Q7 MPF102
12 VOLT
J1 AC IN .5 V
J2 AC IN ATTEN
J3 DC IN 0–5V
J4 SAMPLED OUTPUT
J5 CLOCK OUT
J6 WHITE NOISE
J7 PINK NOISE
J8 MODULATION NOISE OUTPUT
J9 LOW NOISE
TO ⊗ ON J2
NOISE ATTEN R4 5K
NOISE OFFSET 10K
NOISE GAIN 20K R8
R5 50K AC OFFSET
R6 50K DC OFFSET
R9 20K DC GAIN TRIM
R1 CLOCK 1M
R3 5K
SELECT

Now a word about white and pink noise. White noise is defined as equal energy per cycle, and pink noise has equal energy per octave. If you already have a white noise generator then you know how many applications it has. White noise has a hissss sound to it. Pink noise has a deeper sound to it. It's good for rolling thunder sounds, wind, gun sounds and dozens of others. The next step down from pink noise is red, and it is derived from pink noise by filtering it through a -3 dB filter that is tuned to about 7 to 10 Hz. However, I have chosen to go another way. I filter the white noise through a 24 dB filter tuned to about 100 Hz. Now if you want to know what this noise sounds like, listen to the first minute or two of "Space Fantasy" on Tomita's "KOSMOS". There he starts things off by peeling the paper off of your speakers by using very low frequency noise.

CIRCUIT DESCRIPTION

The clock is a free running clock; an NE555 is used. R1 is a front panel mount pot and controls the sample time or speed of the sample and hold clock. Pulses are coupled into the base of Q8. The collector of Q8 is a bilateral pulse that is fed into the controls of IC2. IC2 is a 4016 linear analog switch. There are four switchs in this IC, four inputs, four outputs, and four control inputs. The typical "on" resistance of one switch is about 300 ohms, so by paralleling all inputs and outputs the total resistance is around 75 ohms or so.

The clocking, or turning the switch on and off, either passes or blocks any signal that appears on the input. Referring to Figure 1b, if the voltage of the sawtooth is .5 volts when a sample is taken (when the clock line goes high), then the switch is closed and .5 volts appears at the output of IC2 and across C1. When the clock line drops the switch opens, but the charge on C1 remains. FET Q7 is a high impedance follower and C1 will remain charged. The voltage on the drain is therefore .5 volts. This is then inputted to an op-amp for buffering to the outside world.

The input signal to the 4016 comes from one half of IC3, and is an amp with a gain of 19. All amps are 1458 style dual op-amps. A signal from J1, J2, or J3 is selected by S1a. J1 will accept a .5 volt signal, such as a signal from your oscillator. If you use a signal with greater than .5 volts then use J2 and use R2 as a front panel pot for an attenuator. With S1 in the DC position, it will accept a DC signal, such as the signal from a function generator. S1b is used to select the proper bias for the mode in which IC3 operates. If S1 is in the DC position then, R6 is adjusted until zero volts appears on pin 7 with zero volts on the input of J3. When S1 is in the AC position then the bias point on pin 7 is adjusted for 4.0 volts and the AC signal operates around this point.

Clock pulses from IC1 are coupled to the base of Q1 and appear as an 8 volt sync pulse on J5. This pulse is used to trigger ARs or ADSRs.

NOISE SOURCE

Q4, Q5, and Q6 make up the white noise source. Q4 is a 2N2712 or 2N3392 selected for good noise quality. This can be seen and measured on pin 1 of IC4. A transistor socket here would be a good idea. Plugging in different transistors and observing them for best noise is an easy way to select Q4. The white noise is coupled through C2 and amplified by IC4 whose gain can be varied with R8. At this point the white noise is brought out to J6.

Also coupled from pin 1 of IC4 is the pink noise filter. This filter is a -3 dB per-octave filter. Q3 serves as an amplifier to boost the pink noise to about three volts output to J7.

The low random noise filter is a fourth-order low-pass -24 dB filter. The cut-off frequency of this filter is 100 Hz. It is this low noise signal that is coupled directly back to J2 of the input.

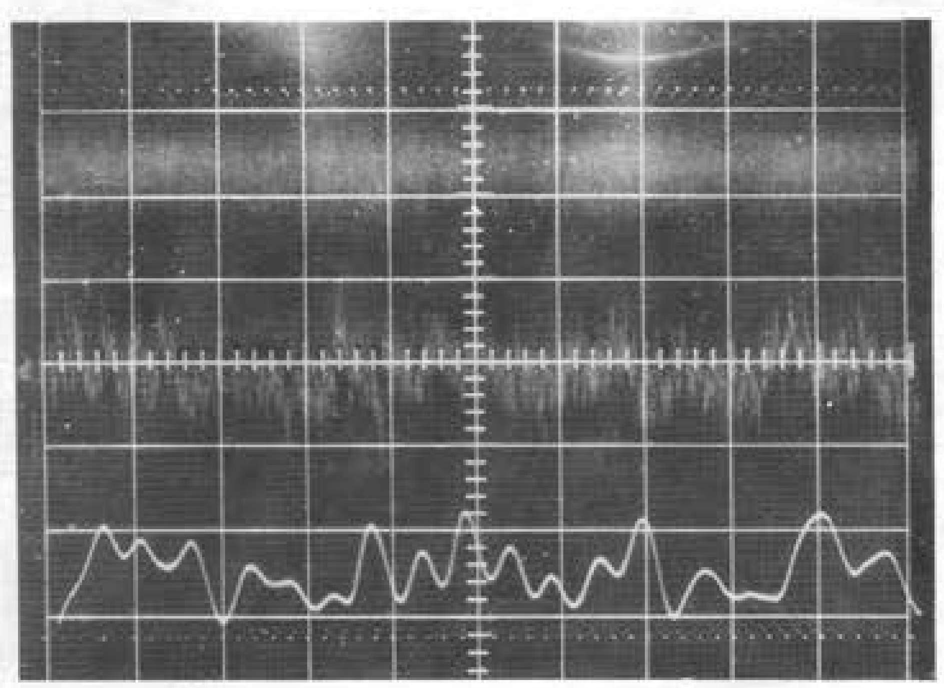
CALIBRATION

Calibration of this unit is not critical, but there are a few key adjustments that have to be made. I do suggest that you make this unit on perf-board or make your own P-C board and use IC sockets. For the first part of the calibration, install IC4 and IC5 only. Leave out IC1 and IC2 and IC3.

Now with IC4 and IC5 in place, use a scope or an AC VTVM and place it on pin 1 of IC4, the white noise output. Adjust trimmer R8 for 3 volts P-P on the scope or .6 volts AC on the VTVM. If the gain cannot be adjusted for 3 volts then select another transistor for Q4. After this has been done, remove Q4 and adjust R7 for 0 volts, using a DC VTVM or scope, offset pin 7 of IC4. Replace Q4.

Place the scope on the output jack, J6, and there should be 3 volts P-P or .6 volts AC of white noise. See Figure 4. Next check J7, the pink noise output, and there should be about 3 volts P-P of pink noise there, or .6

4.



volts AC on the VTVM. See Figure 4. By the way, if your VTVM has a dB scale on it then these readings should read about -2 dB. Place the scope on J9 and 3 volts of low random noise should be seen, or .6 volts AC on the VTVM. See Figure 4.

After this is all up and running, IC1, IC2, and IC3 may be installed. If you have your scope handy, check the collector of Q8 for a clock impulse as shown in Figure 5a. (Advance the clock pot fully clockwise.) Now check J5 for about 8 volt clock pulses and vary the speed of the clock and check its operation. See Figure 5b. Advance the clock fully again.

Next advance R3, the output attenuator pot, fully clockwise. Now there are three more adjustments, R5, R6, and R9. Leave the clock pot full clockwise for the rest of the calibration.

Place S1 in the DC position. Place a DC VTVM or DC coupled scope on J4, the output. Now input a DC bias source into J3 and turn down the bias to zero volts. Adjust R6 for zero volts out on J4. Input 5 volts to J3 and adjust R9 for 10 volts at J4's output. Repeat these two steps again.

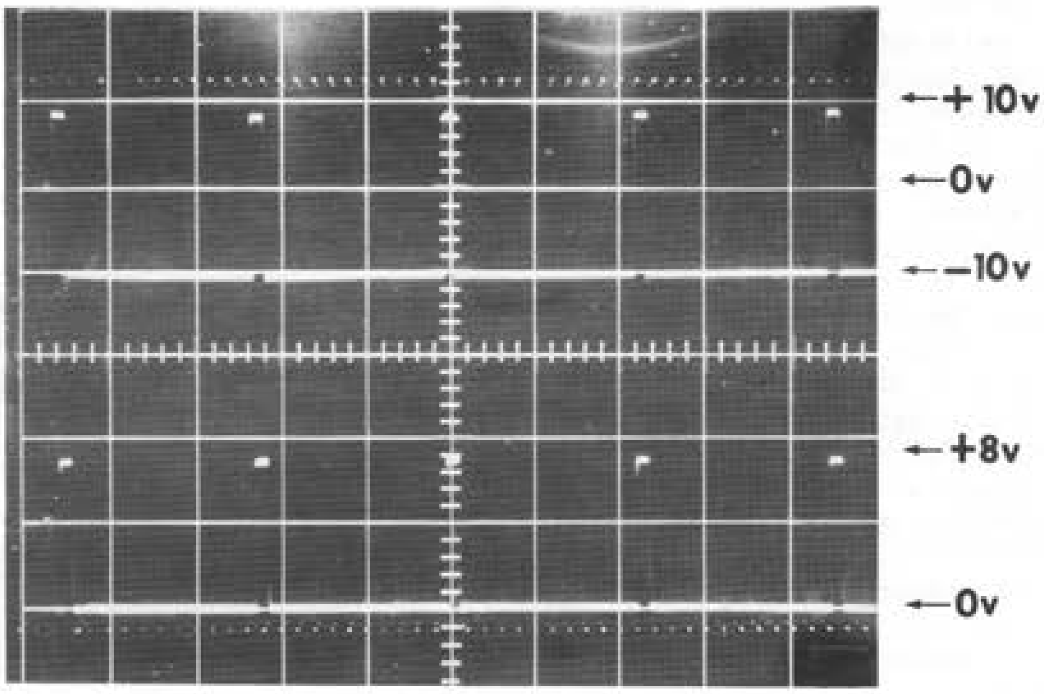
Place S1 in the AC mode. Insert one end of a patch cord into J2. Leave the other end free at this time. This will open the jack and prevent any low random noise from entering IC3. With no signal input, R5 may now be adjusted. With the scope or VTVM on J4 output, adjust R5 for 5.0 volts DC. Now place the other end of the patch cord into the triangle output of an oscillator. Adjust the oscillator for a frequency of about 16 Hz. With the scope still at J4, the output signal should look like Figure 6a. If R5 has not been adjusted correctly, then the signal will look like Figure 6b.

Turn the clock speed down and the sample triangle, as seen at J4, now takes on a different shape.

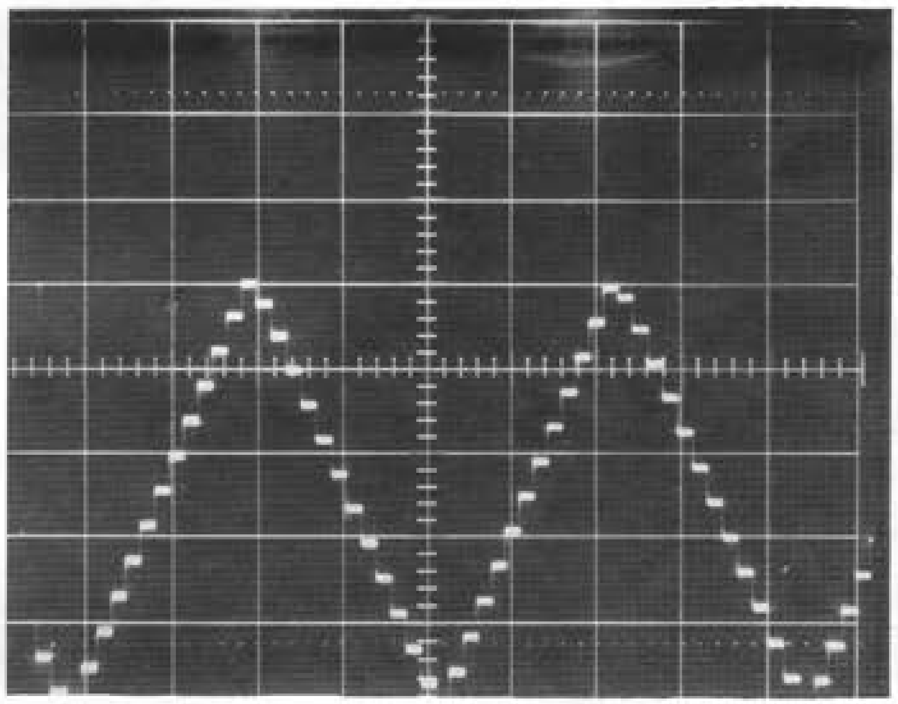
Now I know that you are just dying to try this out, so try the patch in Figure 7a. In place of the oscillators, try a filter as in Figure 7b.

Other possibilities include inputting a sawtooth from an oscillator and sampling that. By changing the clock speed, many nifty patterns can be produced.

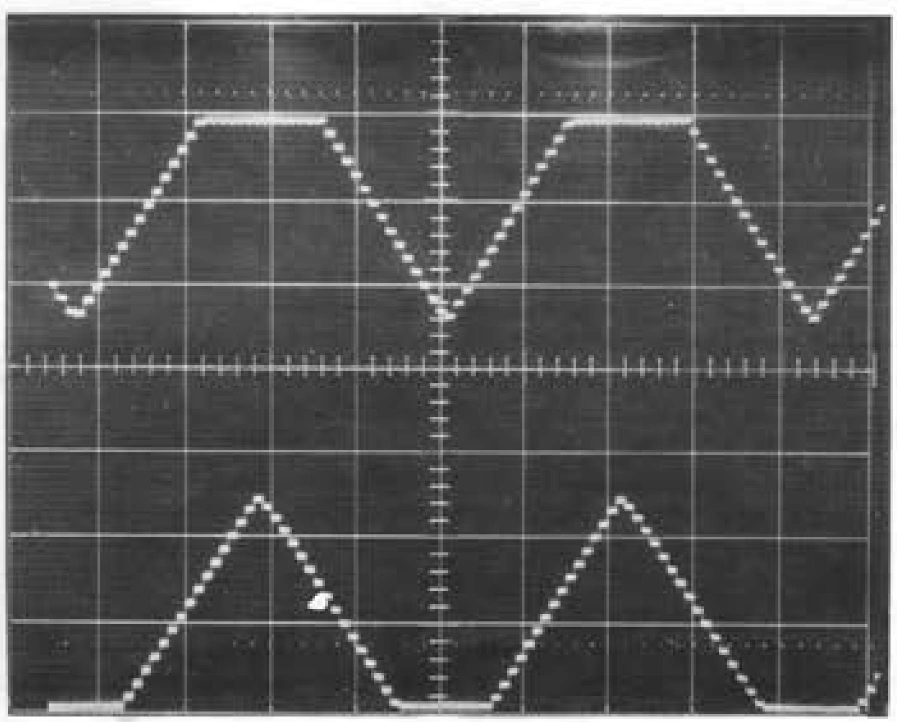
5.



6.a



6.b



FINAL NOTES

I would suggest you using a good mylar or polystyrene capacitor for C1. Also, as stated in the calibration, the white, pink and low noise outputs have been adjusted for a 3 volt P-P signal. If you think this is too much, then adjust R8 to your own specifications. However, when doing this, reduce R10, pin 7 of IC4. For it is this P-P voltage that lets the output swing its full 10 volts. Adjust R10 so there is a .5 volt signal of low random noise at the input of J2.

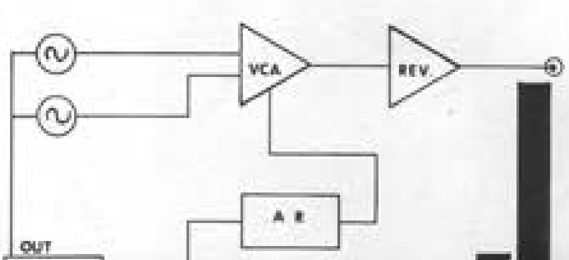
All of the op-amps are type 5558, dual op-amps. The NPN transistors are 2N5129 or similar. The suggested power supply is also shown in Figure 8.

There is one more thing I forgot to mention -- J8 and R4, the noise modulation output. The low random noise is outputted to a pin-jack, J8, and attenuation pot R4. This low random noise modulation can be used on VCOs, VCAs, filters, or many other things. The depth can be adjusted by the noise attenuation pot R4. This is a control voltage output level, not an audio signal like the other low noise output.

A perf-board or a "roll your own" P-C board may be used. All controls and jacks will fit a four by four inch panel. See Figures 9 and 10.

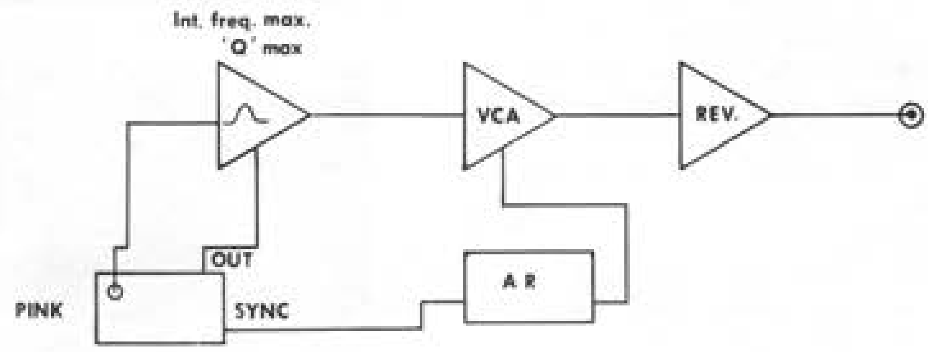
I could fill the rest of this page with many different patches that can be used with this unit, but it's more fun finding them out for yourself. Don't forget the use of the pink noise and low random. If you are using small shelf speakers for your system, don't expect to hear very much. But when you plug in the low random and you are using large speakers, be prepared to explain why the plaster is on the floor!

If you are cost conscious, the cost of this project is around twenty-five dollars including pots, knobs, sockets, etc. However, if your junk box is good, then the price should be much lower.

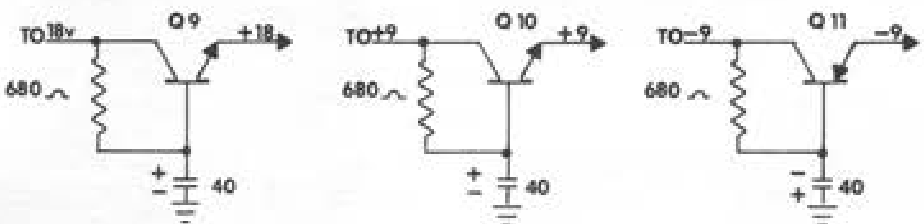


7.a

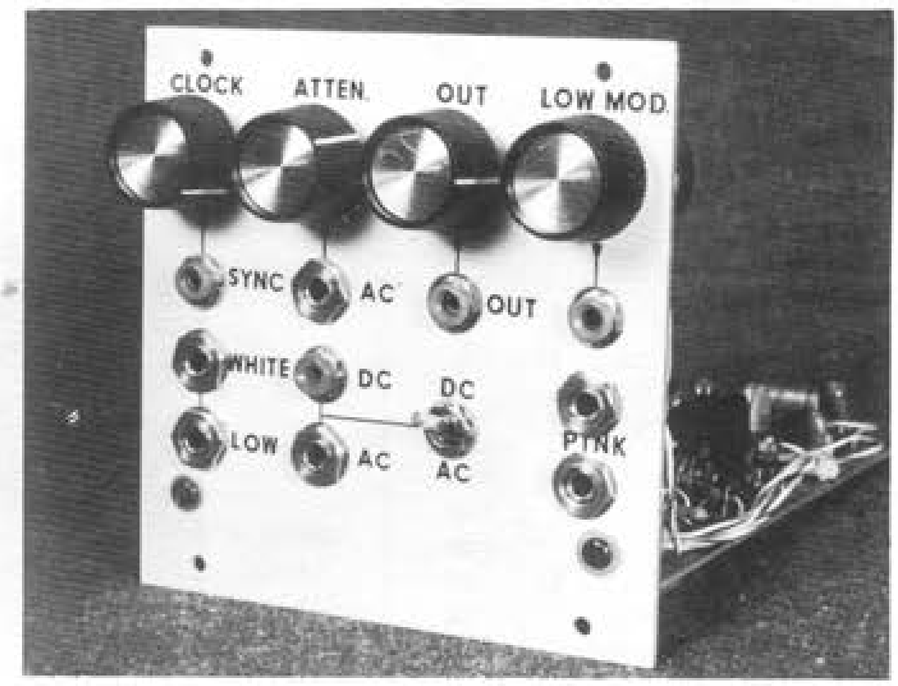
7.b



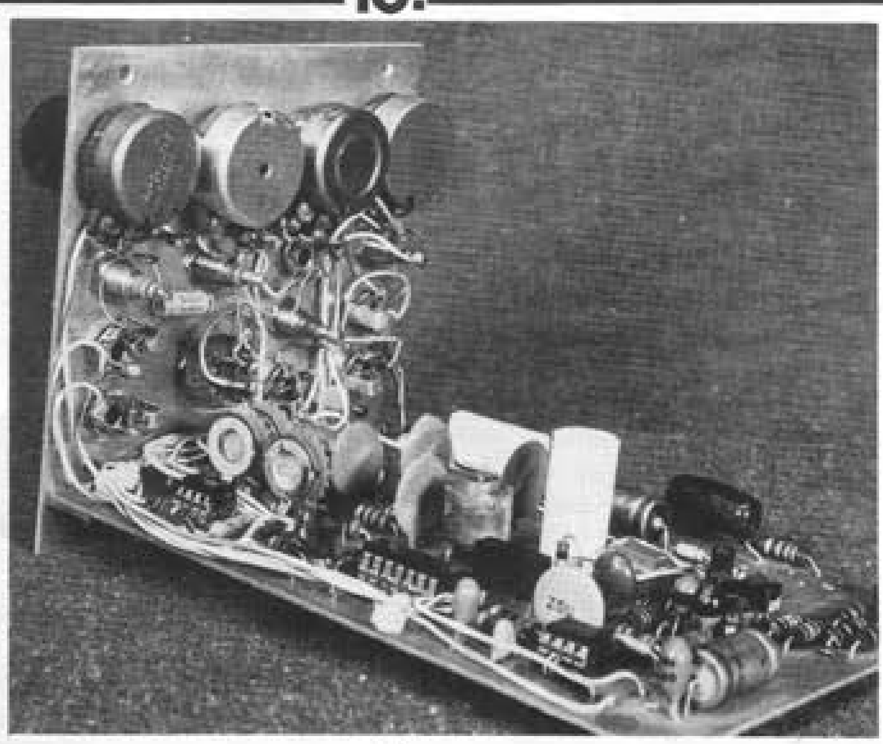
8.

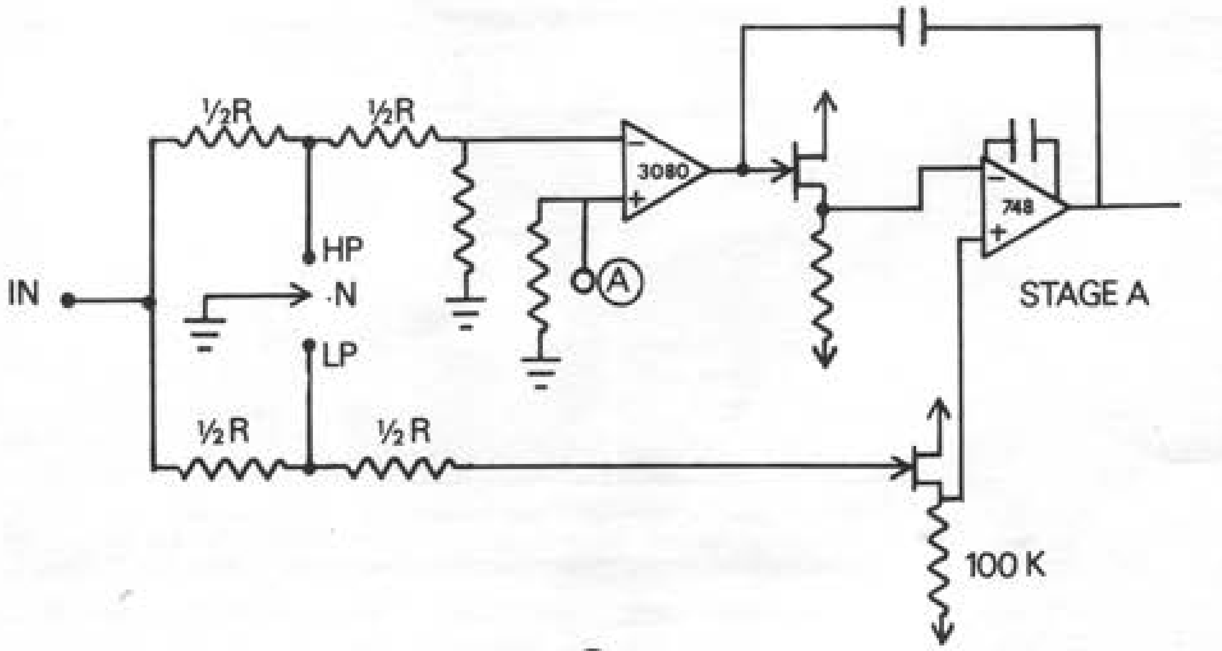


9.



10.



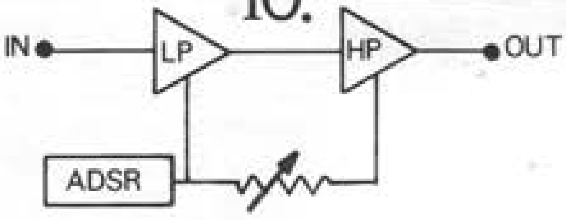


9.

.... continued from page 27

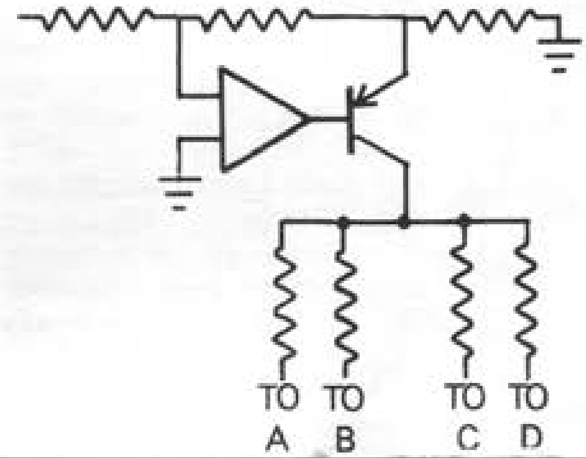
of equipment. If you own PAIA equipment, you could copy the linear converter from the 4730. Don't forget to add two extra resistors out of the current source as shown in figure 11 to allow for driving four of the 3080 filter stages.

10.



If you own exponential gear, I hope to have some accurate exponential converters for you in a couple of months. For experimental use, try the converter given in the last VCA article. I don't think that it is of enough precision for VCOs or very tight filters, but it

11.



should get you off the ground. Besides, if you build a 12db version, they are not all that tight anyway and won't oscillate (for a sine source), so you should be able to get some nice brass sounds or add some color to your system.

One last note. This filter is a true resonant filter. With medium to high 'Q' settings the output can GREATLY exceed the input, possibly overloading any following VCAs, mixers, reverb, or other modules. BE CAREFULL! This type of distortion usually doesn't harm anything, it just sounds bad (at least when used to excess). You may wish to put an input attenuator pot on the panel so that the level may be adjusted to stay within system limits.

POLYPHONY REVIEWS;

PRINTED IN U.S.A.

<u>Air Pocket</u>
by Roger Powell

(To be released soon, label unknown at this time)

Roger Powell is one of the most versatile and accomplished electronic musicians to break into the popular market. His advanced and experimental techniques have been solid additions to Utopia and Bowie, and have been responsible for an increase in interest in good synthesis by a growing number of followers. Unfortunately, his short breaks with these bands coupled with widely spaced live performances have made us all wait anxiously for a new solo album so we could hear what he has been up to. Finally, it's on its way.

"Air Pocket" is a good re-entry into the solo album market for Roger. The material included covers a wide range of styles and techniques, and the time span covered is several years. So, this is, in effect, an anthology of Roger's activity since "Cosmic Furnace"; now let's just hope the next album comes sooner than this one did.

Side one starts with "Lunar Plexus", a powerful introduction with some good patching. The snare drum patch is very realistic, even down to that initial 'crack' at impact. A lot of sequencing takes place throughout the song to add background fill and special effects, and studio technique plays an important part in this realization with all the outboard processing and mixing going on. The title song, "Air Pocket" follows to give the listener a taste of some of Roger's more recent work. "Air Pocket" uses an Imsai computer to generate a Brown sequence and a number of permutations which are then used as the basis for melodic overdubs and rhythms. An RMI KC II is used to do some poly string- type fills and interludes. The solo work on this piece is typical Powell solo work, with well executed pitch bends and modulation control. "Landmark" is a fairly standard rock cut with vocals and rock instrumentation. This plus the following "Windows", which is a slightly slower tune with much richer vocal work, are much in the style of Utopia and sound as if they may have been originally written for inclusion in the Utopia repertoire. I have no liner notes for this material, but there may even be some performers on these cuts from Utopia; it sounds that much like them. Side one closes with "Sarls of Arakis" which starts with some unique synthesis of plucked string instruments in a very 'eastern' flavor. Tuned drums and atonal effects mixed into the back further the open 'eastern' tonality. This intro segues into a very spacey twisting piano and orchestra movement that goes through some intense stereo flanging before settling to a more free form flow with a lone melody and droning bass line.

"Morning Chorus" provides a mellow begining to an otherwise powerful side by fading into a synthesized meadow sunrise with numerous birds, pond life, and other animate sonic creations. Electric piano with 'pipe organ-size' swells and light sequenced effects randomly enter, but fit well with the random sonic background. This is more of an emotion or mood study which draws heavily on a variety of well done nature sounds. "March of the Dragonslayers" uses some interesting synthesized electric piano sounds to build a contrapuntal structure against an acoustic piano track. In the midst of this, a strange groaning voice patch (the dragon, I suppose) provides transition from one section to the next. Instantaneous addition or deletion of reverb and ambience to the mix is a well used effect that disorients the listener a number of times throughout the piece. The momentary accelerando near the end with all voices following the changes in rhythm is a tasteful use of sync techniques commonly overlooked by many musicians. "Dragons and Griffins", in two parts, is another rock piece, but with most voices synthesized. More good drum synthesis appears here, including a number of tightly tuned toms which sound as realistic as the snare drum which is also featured again. The second section is more jazzy in style and sounds much like work from the 'Cosmic Furnace' period. "Prophecy" continues in the rock flow which predominates this side, but throws in some interesting meter changes to keep the listener on his toes. "Emergency Splashdown" closes side two with a vocal studio cut that again has many Utopia influences.

Roger Powell has spent his life with synthesizers; he has learned most every instrument that's available, used them in both studio and stage applications, and even pushed ahead to design his own instruments (the Probe) and write programs for computer assisted composition and performance. His highly polished technique and intuitive wisdom with the artform shine through every part of this album; it's just been far too long in the works. I for one am anxious to hear more extended demonstration of his current endeavors in higher technology music. At least he's back with another album; I hope more will follow soon.

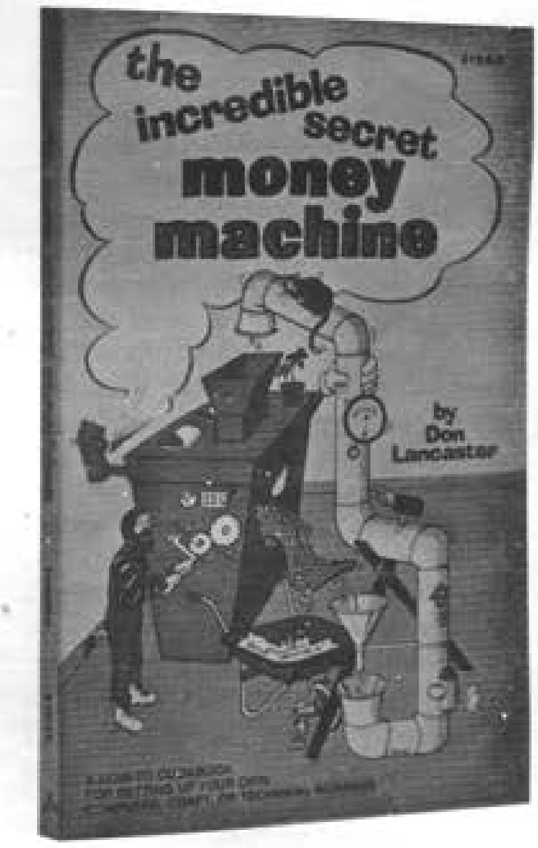
Robert Greenberg
by Robert Greenberg

Available only on cassette for $6.00 postpaid from: Future Now, 600 S. Muirfield Rd., Los Angeles, CA 90005.

Robert Greenberg is a newcomer to synthesis. From what I understand, this tape was recorded in the first six months of his experience with synthesis. Yet, much of his music shows a higher degree of electronic subtlety and sound structure than many of those who have been at it a while. The overall texture of the tape is very smooth. This is one of the mellowest collections I have heard in quite a while; it's very easy to totally relax while listening to Greenberg's work. Much of the composition seems to be derived from jazz background, as there is minimal rhythmic structure and a great deal of sonic overlap in the voices and lines presented. The only equipment used on the tape is an Oberheim 4 voice, an ARP Axxe, and a Wurlitzer electric piano. All tracks were recorded on a 3340S, and the final mix is in stereo.

Side one contains Oriental, Sahara Star, Saturn Sunrise, Ives, Hipstar, and Space Dance. Side two presents Last Year, New Beginning, Coming Home, Mercury Mountains, and Sunburst. In most cases, the Oberheim is used to provide a lush string or brass type background over which Axxe bass and melody lines are placed. "Sunburst" seems to show the strongest rhythmic structure, and is a bolero type of development. In "New Beginning", an up tempo break is taken which shows the strongest display of jazz improvisational skills. "Space Dance" is fun, with mechanical rhythm and high frequency content representative of what most people would expect from outer space.

Robert Greenberg's first tape shows great promise in the field of synthesis, and a solid background in music performance and composition. To produce such a smooth sounding release in the short amount of time he has worked with synthesis is an obvious asset, and is strong reason to watch for future releases.



The Incredible Secret
Money Machine
by Don Lancaster

Published by Howard Sams, available for $6.95 ppd from Synergetics, Box 1112, Parker, AZ 85344.

"Money Machine" is not like most of the books you have seen reviewed in Polyphony. But, in many ways, it is just as important and useful. Any type of new technology tends to be expensive to play with. So, to make the burden a bit lighter, it can be a wise move to turn your hobby equipment into a small business. Not only can money be made with the equipment, but most expenses associated with your interests can become tax deductable. Don Lancaster has made this philosophy work for him for years. Most of the electronic music industry started as someones part-time interest, and most of them have survived enough to make a living. In "Money Machine", Don gives a good overview of the types of things you need to be aware of to avoid hassles with the law, the taxman, and other types of authority. Yet, most sections provide enough detail that "Money Machine" can be used as a 'handbook' for running your business.

Philosophies involved in small businesses must obviously be quite different from corporate guidelines. The first three chapters of the book are dedicated to outlining those philosophies, tactics, and strategies which are unique to small business operation. The chapters on getting started cover special accounts for business use, building an image, nurturing the business, keeping informed so your business can be timely and useful, and how to start the cash flow rolling. Later chapters discuss the use of text and images as a primary source of income, and as a necessity for advertising and promotion. Final chapters cover legal hassles to avoid, income tax requirements and special benefits, and investments and tax dodges.

I'm sure there are many Polyphony readers who would like to make some extra money with their equipment and studios, not to mention avoiding paying as much for all that equipment. But I'll bet that most of you don't even consider the possibility because of the stories you've heard about the hassles of the business world. In "Money Machine" you have one of the best explanations (in everyday language) by one of the most experienced free-lance people I know. There's no excuse not to try starting your business now. And if you don't have a particular interest in starting a business, I still recommend this book as a valuable insight into how small businesses work (it will definitely change your buying habits), and as an opportunity to enjoy Don Lancaster's great sense of humor which never had an outlet in all his technical cookbooks.

LETTERS...

continued from page 19...

bill, and buying a paid ad in the magazine. I also hope that you will expand, if only temporarily, your letters column so that a lot of discussion can be handled. The ideas have to come out. And the people have to be put in contact with each other. Exciting times. I'm glad to see that you have taken the steps you have. And I hope that you will continue to do so.

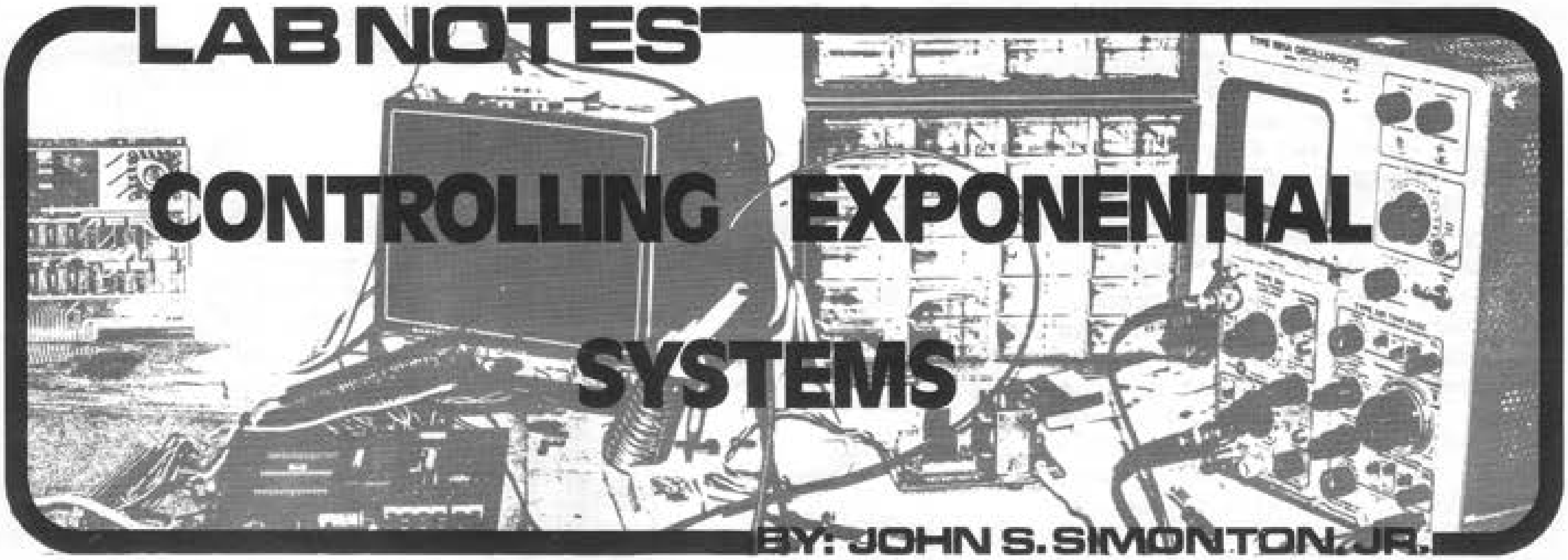
Chuck Larrieu
Corte Madera, CA

Chuck

You have a lot of good points. However, I feel that a record label operated by Polyphony would be beneficial to musicians who, after completing their tapes, are financially drained. I think that just completing a master is a good indication of an artists faith and dedication. Also, the pressing and distribution 'games' can be quite a hassle for an individual artist to handle. If every artist has to spend time completing every last detail of his album, fewer albums will be released. Somewhere along the line, the labor needs to be divided so certain people can concentrate on certain jobs. With a single organization handling these matters, better (and more consistent) pressings can be found, wider distribution obtained and, in general, more success for the individual artists as well as the label and concept. From it's inception, one of Polyphony's prime motives has been to increase communications so we can learn from the experiences of others. A record label seems to fit the nature and philosophies of our organization very well. I sincerely feel that most people would trust Polyphony to find quality material (please let me know if any of you feel otherwise!) without limiting the selection from an artistic standpoint. I DO feel that it would not be wise to release just anything, regardless of recording quality or musical technique employed. However, most of those submissions would be eliminated if the artist took the initiative to obtain the feedback you mentioned.

As always, we need to hear from as many of you as possible in order to devise a system that will really work. The "LETTERS" column is ALWAYS open to forum; let us hear from you.

ADVERTISER INDEX



LAB NOTES
CONTROLLING EXPONENTIAL SYSTEMS

BY: JOHN S. SIMONTON, JR.

The two most common questions I hear about the computer - based synthesizer systems we've been developing here are:

1) How do I use it with my exponential synthesis gear?

and

2) How do I use it with my Razmataz RMT-80 computer?

The answer to the second question is going to have to wait just a bit longer (though I expect to have a surprising answer soon).

The answer to the first question is what we're going to focus on this time by looking at a Digital to Analog converter that is designed to be compatible with almost every synthesizer in the world with the exception of the linear holdouts- Paia, Yamaha CS series, Unicord, some EML; you know who they are. For them, you use the stuff we've already covered.

By way of a very short review, the differences between D/As that are to be used with linear response elements and those that are to work with Moog, Arp, or any other exponential system are not great from a basic conceptual standpoint. A binary number is fed in one end, and a DC control voltage comes out the other. But, they do differ greatly in the character of the voltage that comes out.

For linear response equipment, the D/A must produce an output that has an exponential character- as the control voltage increases, the incremental change in voltage must also increase.

Since exponential response equipment has analog circuitry built into the front end of each control input which "bends" the linear control signal into an exponential curve, a D/A that is to be used with this equipment must produce a linear output voltage function. That is, the incremental change in output voltage must be constant. See figure 1.

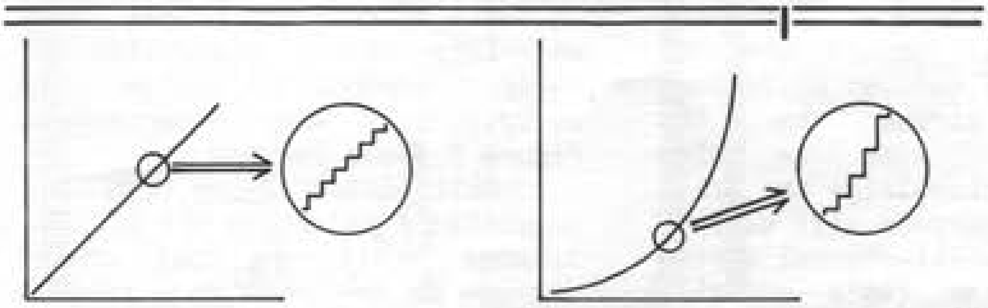
One of the nicer things about this linear D/A is that it's common, the kind that most applications require. Since it is common, we have a large number of parts to choose from. From that large number we've selected a "5008" type which is made by a number of manufacturers. When Signetics makes it and houses it in a 16 pin plastic package it becomes an NE5008N.

Inside, this chip is relatively simple. It looks like figure 2. The transistors shown are each a current source and the values of the resistors in the matrix that their emitters are tied to are such that if the source associated with D0 is pumping some current (i), the one that corresponds to D1 will pump twice that (2i). Similarly, the source that goes along with data bit D2 produces twice what the previous one did (4i), and so on.

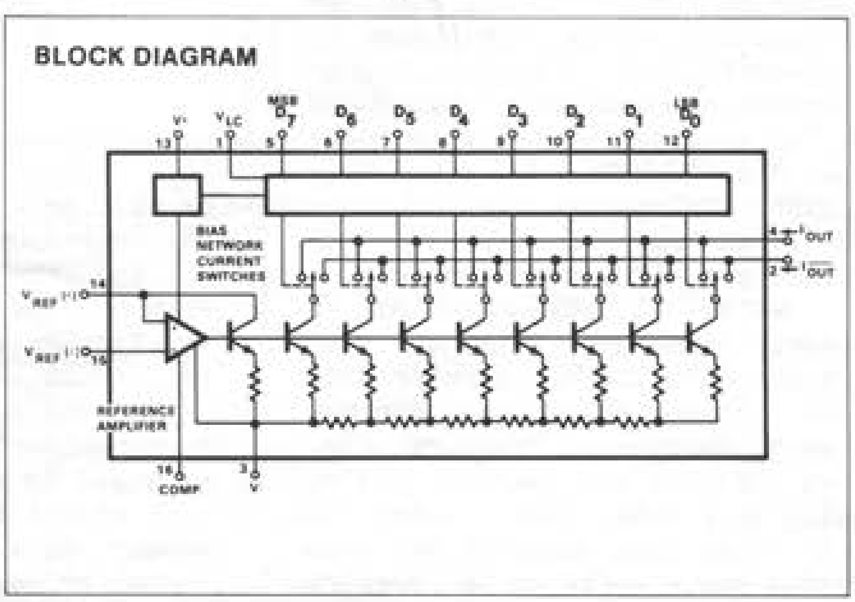
In response to a bit being set, the current produced by the source associated with that bit is switched so that instead of appearing at pin 2 of the IC it appears at pin 4 (Iout). At any given instant, this output current will be the sum of the currents corresponding to each input bit which has been set.

To turn this chip into a "system" that accepts data at the input and controls a synthesizer at the output, we need to add such niceties as latches to hold the data that the computer sent out, an I/V (current to voltage) converter to change the 5008's current output to a voltage that our synthesizers will like, and

1



2

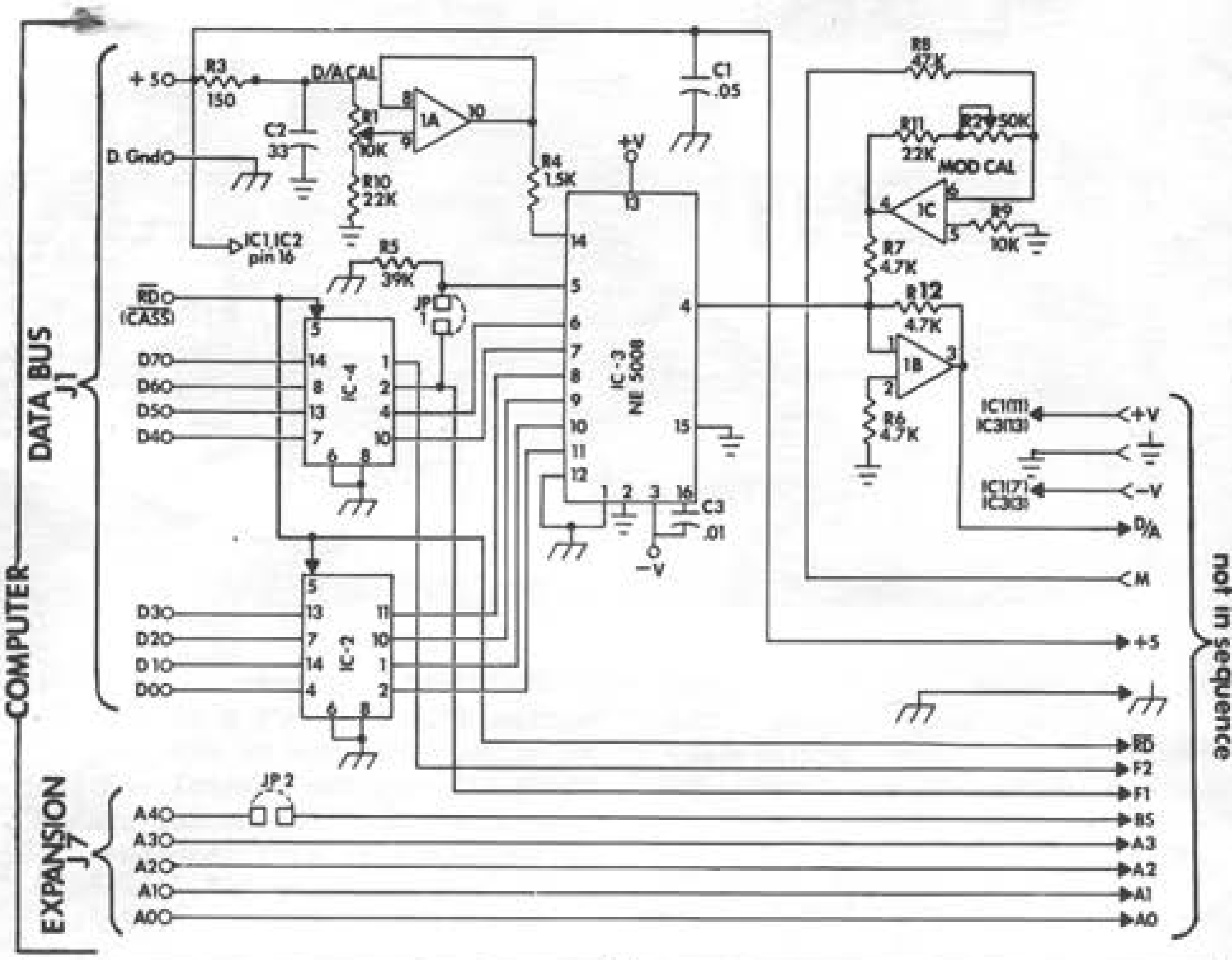


other bells and whistles as available.

When we do all this, the design looks like figure 3. It's pretty straight - forward. We've used 4042's to latch the data coming in and the $\overline{RD}$ line is the strobe on these latches which, when low, allows the data present at their inputs to appear at the outputs. When $\overline{RD}$ is high, whatever data was present at the latch inputs when the line went high will be held at the outputs. Notice that the two most significant data bits follow our previous protocols in that they come out simply as flags rather than being presented to the converter circuitry. But notice also the jumper JP1 which, as we'll see later, can be used to double the range of the D/A (although at what might be an unacceptably high cost).

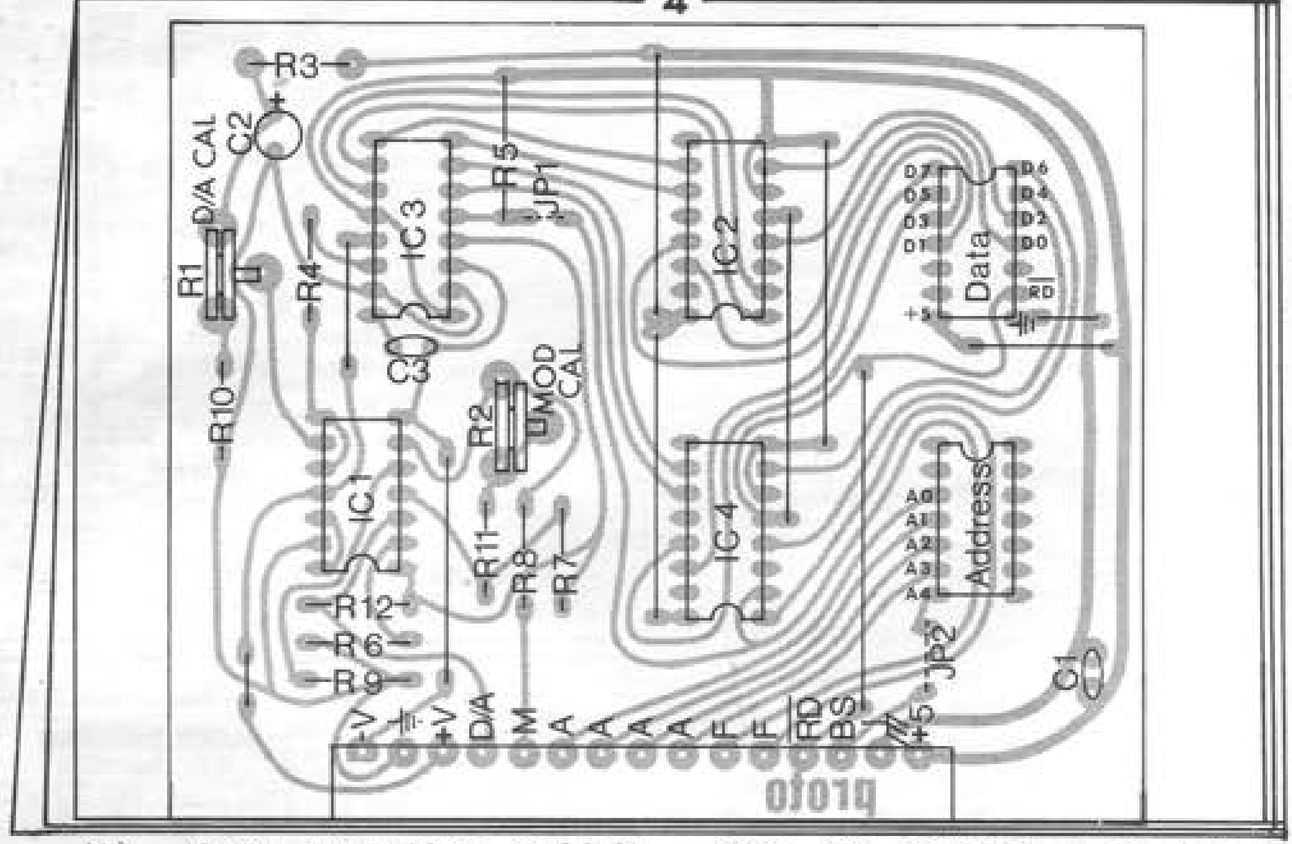
We've used a 4136 quad op-amp to provide all of the analog support that the 5008 needs; one stage serves as a buffer between the calibration trimmer and the 5008's Vref input (IC1a), another comprises a current to voltage converter (IC1b), and a third is an inverting summing amplifier that allows a modulation input (IC1c).

With the exception of the standard "be tidy" caveats, there's nothing very critical about this D/A system and you can build it using whatever construction techniques appeal to you, but the board which is available from Paia has enough interesting features that it's worth taking a special look at it. Check out figure 4.



3

4



I suppose the most interesting thing is the way the input, output, and control lines are configured. Notice that the connections to the computer all appear on two 14 pin dip outlines (J1 and J2), while connections to the synthesizer (including some computer address lines that QuASH in an expanded system will need; see "In Pursuit Of The Wild QuASH", Polyphony July '77, page 19) come out to the 15 pin Molex-type edge connector (J3).

We've already examined in general terms how this type of D/A connects at the computer side (see "The Polyphonic Synthesizer", Polyphony February '78, page 28). If the computer you're using is a Paia 8700 (which is not a bad idea since it has some useful music software to support it), these connections couldn't be simpler - there is a one to one correspondence between J1 and J2 and the connectors they mate with on the computer. Standard pre-terminated jumpers are used to connect the two. No soldering.

The wiring to the "synthesizer" side is also arranged to acknowledge the fact that almost everyone will want to expand to a multi-channel system sooner or later (it's actually what the computer stuff is best at!), so the Molex wiring is the same as that found on QuASH modules.

All of this means that from an inter-wiring standpoint, a fully expanded system is exceptionally easy to implement. Figure 5 shows you how.

Calibration of the 8785 D/A consists of adjusting the D/A CAL trimmer (R1) so that octave changes in the input data produce octave changes in the module being controlled; this can easily be done by ear. The MOD CAL trimmer (R2) should be set so that a one volt (or whatever represents one octave in your system) change at the modulation input produces a one octave change in the controlled element.

Before we wrap this column up, there are some little detail things that really need to be mentioned.

Going back to the schematic for a minute, observe that there are two "programming" jumpers (JP1 and JP2) indicated on the circuit board.

As we've mentioned again and again, the Paia protocols use the least significant 6 bits of an 8 bit word to specify an analog parameter while the two most significant bits are flags (D6 is used as a gate, and D7 is a general purpose control bit which QuASH recognize as a portamento control bit). Since the 5008 is an eight bit converter, obviously some bits will not be used. I decided to permanently not use the least significant bit (LSB) of the converter (pin 12) by grounding it. The only effect of this is to slide all the lines of the controller "up one" as far as the 5008 is concerned, and it has no electrical effect that we need to worry about.

The other unused 5008 bit is then it's MSB (most significant bit - D7, pin 5) and if the jumper JP1 is not in place, this bit is in fact not used. But, if you are one of those people for whom nothing is ever enough, you have the option of installing the jumper. This means that the MSB of the 5008 is tied to data bit D6, effectively doubling the range of the D/A from 64 notes (over 5 octaves) to 128 notes (almost 11 octaves).

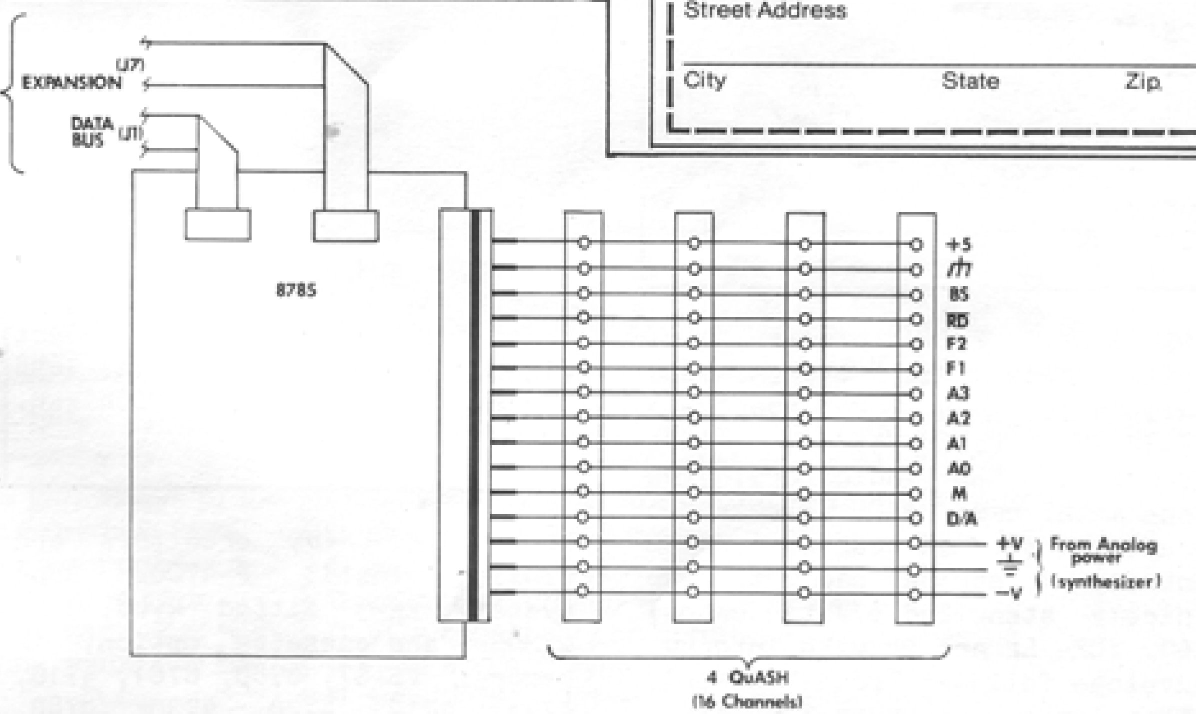
The cost of this "simple" modification is much greater than just a piece of wire, though, because if the option is selected the system is no longer compatible with our existing software (which might be just fine for your purposes). Maybe worse than that, it's no longer compatible with QuASHes either. But if you need it, it's there.

A second jumper (JP2) is meant to be used in systems with 4 or more QuASH and causes the fifth address bit from the computer to serve as the Bank Select (BS) line (see "In Pursuit Of The Wild QuASH" referenced earlier).

Something else to worry about is grounding. At some point in the system, digital power ground (recognized as a chassis ground symbol) and analog power ground (recognized as an earth ground symbol) must be tied together. However, they must have a common connection at only one point. Otherwise you run the risk of ground loop problems. I recommend that these two grounds be tied together at the Molex connector of the D/A, as shown in figure 5.

Finally, Moog "S" triggers must be pulled to ground rather than accepting the high logic level that our trigger outputs provide. The simple circuit in figure 6 takes care of this using almost any NPN transistor you happen to have laying around.

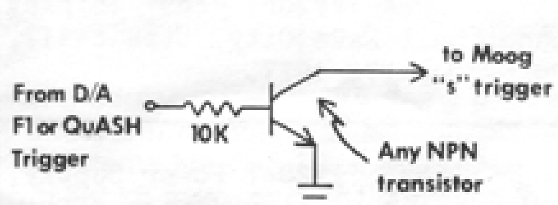
Synthesizers that have both "gate" and "trigger" inputs can use the scheme shown in figure 7 to derive both of these signals from the single gate that our D/A produces.



5

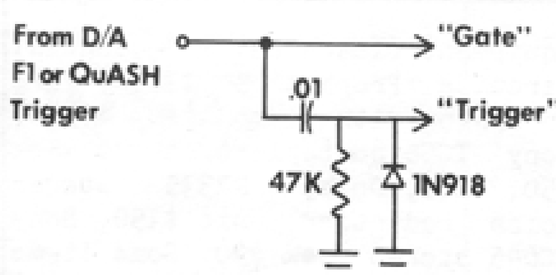
A complete kit for the Linear D/A including printed circuit board, sockets, headers and edge connector is available from Paia Electronics, Box 14359, Oklahoma City, OK 73113. Order #8785 Linear D/A. $22.95 ppd.

6



MOOG "S" TRIGGER ADAPTER.

7



"GATE" & "TRIGGER" ADAPTER.

EQUIPMENT EXCHANGE

A place for our readers to offer messages of equipment for sale, trade, or wanted, job openings or positions wanted, and so on. Keep listings as brief as possible, and enclose $1.00 for each listing. Persons responding to ads should write directly to the other party. DO NOT write to POLYPHONY. Polyphony is not responsible for any claims made in ads, or the results of any transactions. We reserve the right to edit or refuse any ads submitted.

FOR SALE: ARP OMNI II (new) $1800; ARP ODYSSEY $1150. Harry Poole, Charter Square 8, LaGrange, GA 30240.

MOOG Inventory Clearance: Polymoog with Polypedals $2900, MicroMoog $425, MiniMoog with travel case $900, Minitmoog $450, Graphic Equalizer $175, Ribbon Controller $150, Sample and Hold Controller $150, Glide / Decay foot switches $10 each, Pedal Controller $25. Most have warranty. Boston School of Electronic Music, 28 Highgate St., Allston, MA 02134, (617) 782-9100.

FOR SALE: Dynaco ST-120 stereo power amp, 60 WRMS/channel, TIP mod installed, all manuals, $100 ppd. Koss Pro-4AA phones with 20 ft. extension, $30 ppd. Marvin Jones, PO Box 20305, Oklahoma City, OK 73156.

FOR SALE: PAIA Phlanger, low noise mod, footswitch, scope calibrated, $100. Power supply- ± (5,12,15) volts (29v, lamp Xformer), LED indicators, two tone metal case $30. Electronic Drums- 1 pitched source, 1 noise source, 1 strike pad trigger (nicely stenciled plastic cases) $60. VCF- LP and BP with internal envelope follower (good add-on), cheap case, no knobs $25. Alpha Monitor- portable biofeedback $20. Ron Minemier, P.O. Box 153, Ripon, Wis. 54971, Ph. 414-748-2793.

Equipment Clearance: Sequential Circuits Prophet 5 $3750, Arp 2500 Wing with 7 modules $2000, Sony TC55 portable cassette deck $50, Sony/Dolby NR335 stereo noise reduction unit $150, Sony TC645 stereo deck $90. Some items still in warranty. BSEM, 28 Highgate St., Allston, MA 02134, (617) 782-9100.

FOR SALE: 4710 Balanced Modulator, $20. 2720-4 Function Generator, $20. 2720-9 Glide Retro-fit, $10. 2720-8 Sample/Hold Circuit, $10.00. All in excellent condition, postpaid. Tom Henry, 417 Shady Circle Drive, Rocky Mount, NC 27081

FOR SALE: Digital Group four board computer system with 10K ram, four PIOs, video and cassette interfaces, large motherboard, and much documentation. Includes supplies (+5v @ 12A, ±12v and -5v @ 1A ea.) and Radio Shack ASCII keyboard. $1150 list value, asking $750 or offers. Marvin Jones, PO Box 20305, Oklahoma City, OK 73156

FOR SALE: Brand new, never used P-4700/J computer synthesizer. All parts working. D/A and Quash need assembly finished. $750 firm. Also: 3 watt blocks, mixer, modulater and sequencer in road case. Some need work, $50. Keyboard and road case (not wired) $60. Frank Rogala, 4698 Barnum, Hastings, Michigan, 49058, Ph. 616-367-3951.

FOR SALE: (my creditors are closing fast!) P-4700J type system. EK-3 fitted kybd, 8700 w/MUS-1 and cassette option, 1K memory, PS-87, 8780, 8781, 4710, 4711, 4712, 4720, 4730, 4780, 2720-1, 2- -2A, -3L&3B, -4, -5, -7, -11, -12, -14, Phlanger, road cases, and 1 wing cabinet. 1 2720 needs work, 4730 filter uncalibrated. Asking $825, will deal. Greg Taylor, 4289 Shirley Lane, Salt Lake City, Utah 84117, Ph. (801) 272-1977.

FOR SALE: 2720-7 Power Supply, excellent condition, large filter caps for better stability. $18. 2720-2A VCO, works great except trouble keeping in tune. $10. Stereo Chord Egg, excellent condition. $15. Will sell seperatly or all for $35. Walt Simmons, 4616 Antelope Creek Rd., Lincoln, NE 68506, Ph. (402) 488-4285.

FOR SALE: 2720R with Glide circuitry, 4710, 4711, 4712, 4720, 4730, 4740, 3-4770, 4780, 2 wing cabinets. Works fine. $570 or will sell seperately. Martin Gmitter, 9300 Bundidge Rd., Richmond, VA 23235, Ph. 804-320-0193

FOR SALE: Monster Paia System- 7 VCOs, 5 VCFs, 6 EGs, 6 mixers, 5 VCAs, 2 bal mods, 2 seq, 3 LFOs, reverb, and more (but no microprocessor). Custom cabinets, regulated supplies, ribbon controller, road crates, matrix switching, patch cords, TWO keyboards, schematics, drawings, notes, pictures. Also smaller cabinet (certain modules unplug for a portable mini-synthesizer), built in ARIES S&H, VC clock, noises, LFO. Everything works great. $2000 plus shipping. John Deaton, 5959 Pinemont #156, Houston, TX 77092, Ph. 713-686-3389

FOR SALE: Paia Gnome/Oz combination, assembled. Works fine. Includes all manuals and a dust cover. $70 + postage. Thom Hogan, 719 Anna Lee Lane, Bloomington, IN 47401, Ph. 812-339-9013.

FOR SALE: 2720/R in excellent condition, with all cables and manuals, $225. Will send shipping COD. Steve Carlozzi, 40 Market St., Brockton, MA 02401.

FOR SALE: 1550 Stringz 'n' Thingz, superior condition, chorusing needs work, $250 or offer. Gnome, excellent condition, external trigger and signal mods., $30. Paul J. Drongowski, 644 South 8th East #12, Salt Lake City, UT 84102, Ph. 801-532-6694.

THE MS-20
Above and Beyond.

TWO VOLTAGE CONTROLLED FILTERS
Low-pass and high-pass for extended control of the entire frequency spectrum. Each filter features continuously variable resonance control: flat to self-oscillating.

ENVELOPE GENERATOR 1
Separate AR envelope generator with Variable Delay Control.

5-PART ENVELOPE GENERATOR
Features variable ADSR controls plus unique Variable Hold Control.

ADVANCED APPLICATION PATCH PANEL
The extensive complement of control voltage inputs/outputs expands your creative possibilities by creating new combinations of the MS-20 modules (such as interfacing with external synthesizers, sequencers, etc. separately or in conjunction with the MS-20's internal capabilities).

TWO VOLTAGE CONTROLLED OSCILLATORS
Each with 10-octave range. Selectable, switchable waveforms include: Triangle, Sawtooth, Variable Pulse Width (50%-0%), White Noise, and pre-patched Ring Modulator.

PROGRAMMABLE SECOND VCA
For controlling modulation depth, sample and hold, expression, etc.

PROGRAMMABLE SAMPLE AND HOLD
For "random" or "stepped" patterns.

EXTERNAL SIGNAL PROCESSOR MODULE
Contains advanced envelope and pitch follower, allowing any external instrument to actually "play" the MS-20. Module consists of pre-amp, variable bandpass filter, pitch-to-voltage converter, envelope-to-voltage converter, and variable threshold trigger detector. Outputs include amplified/filtered signals, pitch control voltage, volume control voltage and trigger output.

PROGRAMMABLE CONTROL WHEEL AND MOMENTARY SWITCH
Can be programmed to vary pitch, cut-off frequency, VCA gain, modulation, sample and hold, etc.

MODULATION GENERATOR
Features continuously variable waveforms: Triangle–Sawtooth; Rectangular–Pulse.

PINK AND WHITE NOISE GENERATOR OUTPUTS
For both audio and control voltage sources.

PROGRAMMABLE, MIXABLE FREQUENCY AND CUTOFF FREQUENCY MODULATION CONTROLS

When Korg set out to create the new MS-20 synthesizer, the idea was to build a professional, fully variable instrument for the serious synthesist that would be a cut above the rest.

Korg built in two VCO's, two VCF's, two VCA's, two EG's, an MG(LFO), a sample and hold circuit, extra noise generators and much more. (We're showing you the essential controls of the MS-20 so you can compare it, feature-for-feature, with anything on the market.)

If you're satisfied that the MS-20 compares favorably even to synthesizers costing far more (there's really nothing like the MS-20 in its price range), then consider what makes the MS-20 truly incomparable!

The MS-20 contains a built-in External Signal Processor module that lets any external instrument actually "play" the synthesizer! (To purchase this feature separately, you'd have to pay more than the entire purchase price of the MS-20.)

Considered solely as a keyboard synthesizer, the MS-20 stands above all the competition. But, with its ESP module, Korg has taken the MS-20 a quantum step beyond.

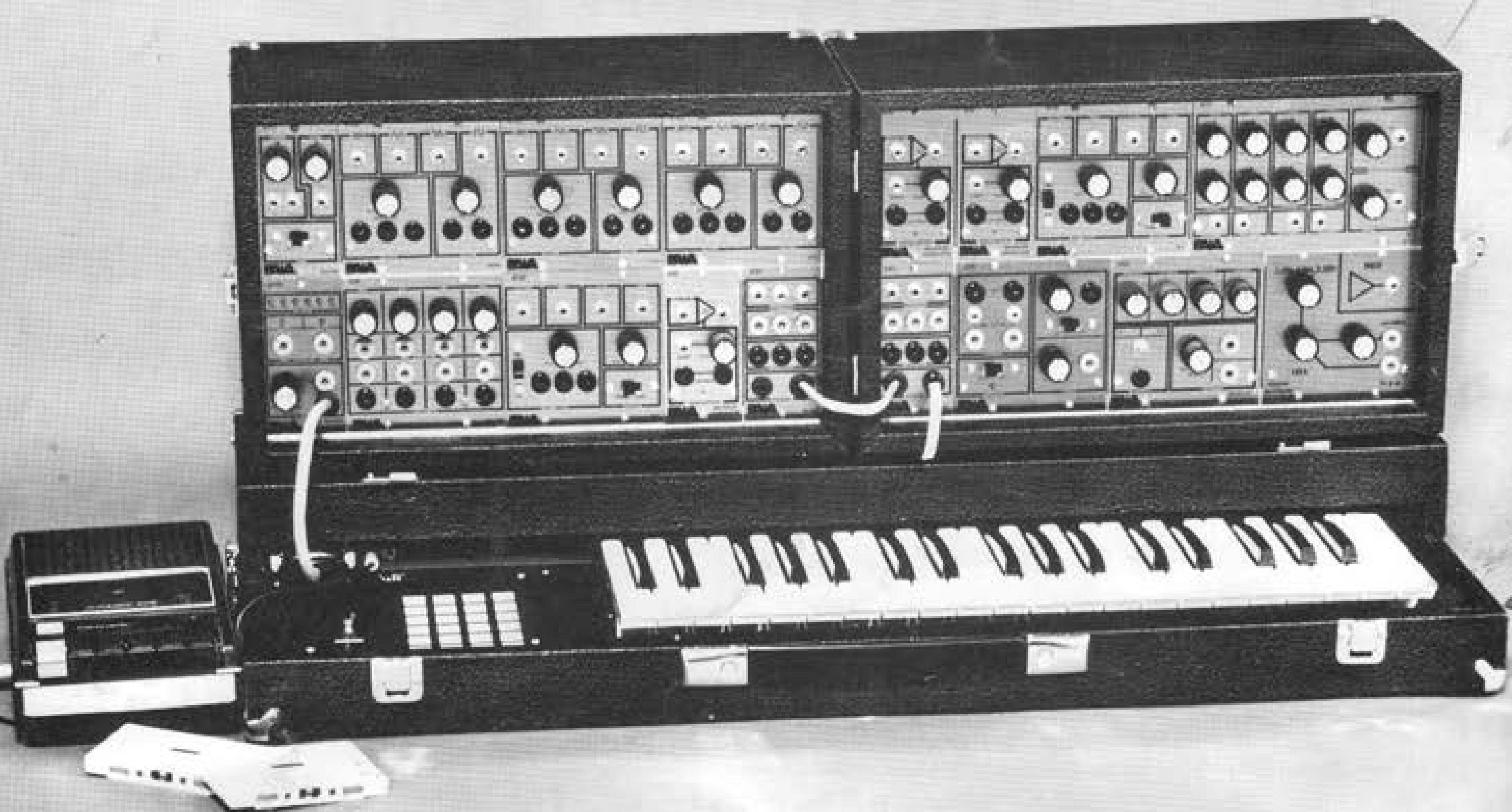
THE MS-SERIES

KORG Puts Synthesizers within reach.

Unicord Division of Gulf & Western Manufacturing Company, 75 Frost Street, Westbury, N.Y. 11590

PAIA P-4700/J POLYPHONIC...Of Course!

SYNTHESIZER/COMPUTER



But that's just the beginning, because only PAIA Synthesizer/Computers Allow you to use any of a growing number of personality programs. Including:

POLYPHONIC

MUS 1.0 - a 16 voice polyphonic synthesizer with software transient generators.

SEQUENCERS

SEQUE 1.0 - a general purpose monotonic sequencer.

POLY SEQUE - a 4 voice sequencer.

COMPOSERS

PINK TUNES- Composes 4 part harmonies

PINK FREUD- Composes 4 part canons.

SPECIAL EFFECTS

SHAZAM - Multiple keyboard split and chorusing.

AND MANY MORE COMING SOON!

The P-4700/J Synthesizer/Computer package includes the following module complement: two 4710 Balanced Modulator VCAs, 4711 Stereo Mixer, 4712 Reverb, three 4720 Wide Range VCOs, two 4730 Multi-Modal VCFs, 4740 ADSR Envelope Generator, 2720-5 Control Oscillator/Noise Source, 8780 Digital to Analog Converter, 8781 QuASH (Quad Addressable Sample & Hold), and the 8782 Intelligent Keyboard with 8700 Computer Controller housed in sturdy vinyl covered road cases.

System firmware includes: PIEBUG - system monitor; POT-SHOT - cassette interface and MUS 1.0 synthesizer operating system.

P-4700/J Synthesizer/Computer Kit $749.00
Includes all listed software (shipped freight collect)

( ) Sounds intriguing, but I need a lot more information. Please send the most recent edition of your "Friendly Stories About Computers/Synthesizers" ....$3.00 postpaid

( ) Please also send complete instruction manual set for the P-4700/J ....$10.00 (refundable upon purchase of P-4700/J Kit)

( ) I've been with you all along. Please send complete P-4700/J Synthesizer/Computer Kit. ....$749.00 ....shipped freight collect ....

Name: __________

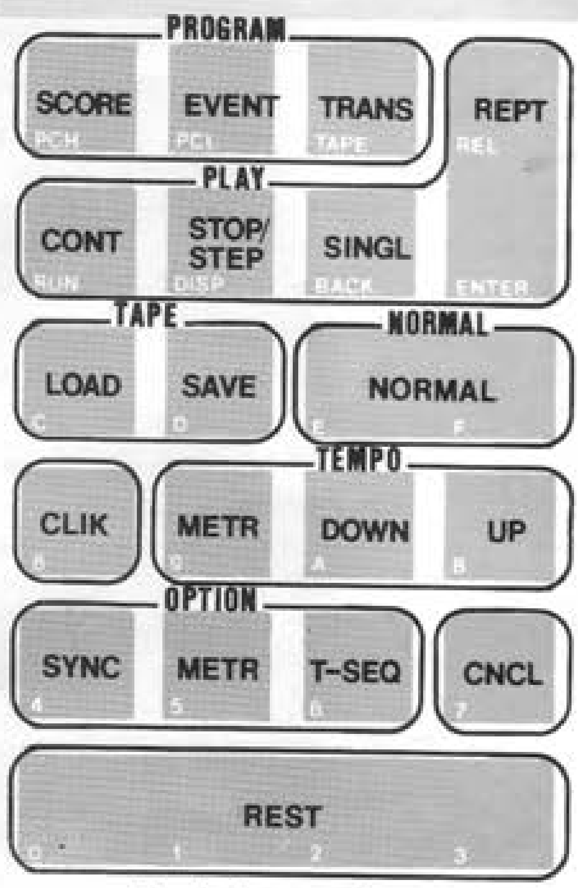
Address: __________

City: __________ State: __________ Zip: __________

Visa/BAC ____ Master Charge ____ Card No. __________

Expiration date: __________ __________

PAIA ELECTRONICS DEPT. 1020 W. WILSHIRE BLVD., OKLAHOMA CITY, OK 73116 (signature)



Typical control panel configuration

ONLY FROM: PAIA ELECTRONICS, INC.
1020 WEST WILSHIRE BLVD.
OKLAHOMA CITY, OKLAHOMA 73116
(405) 843-9626