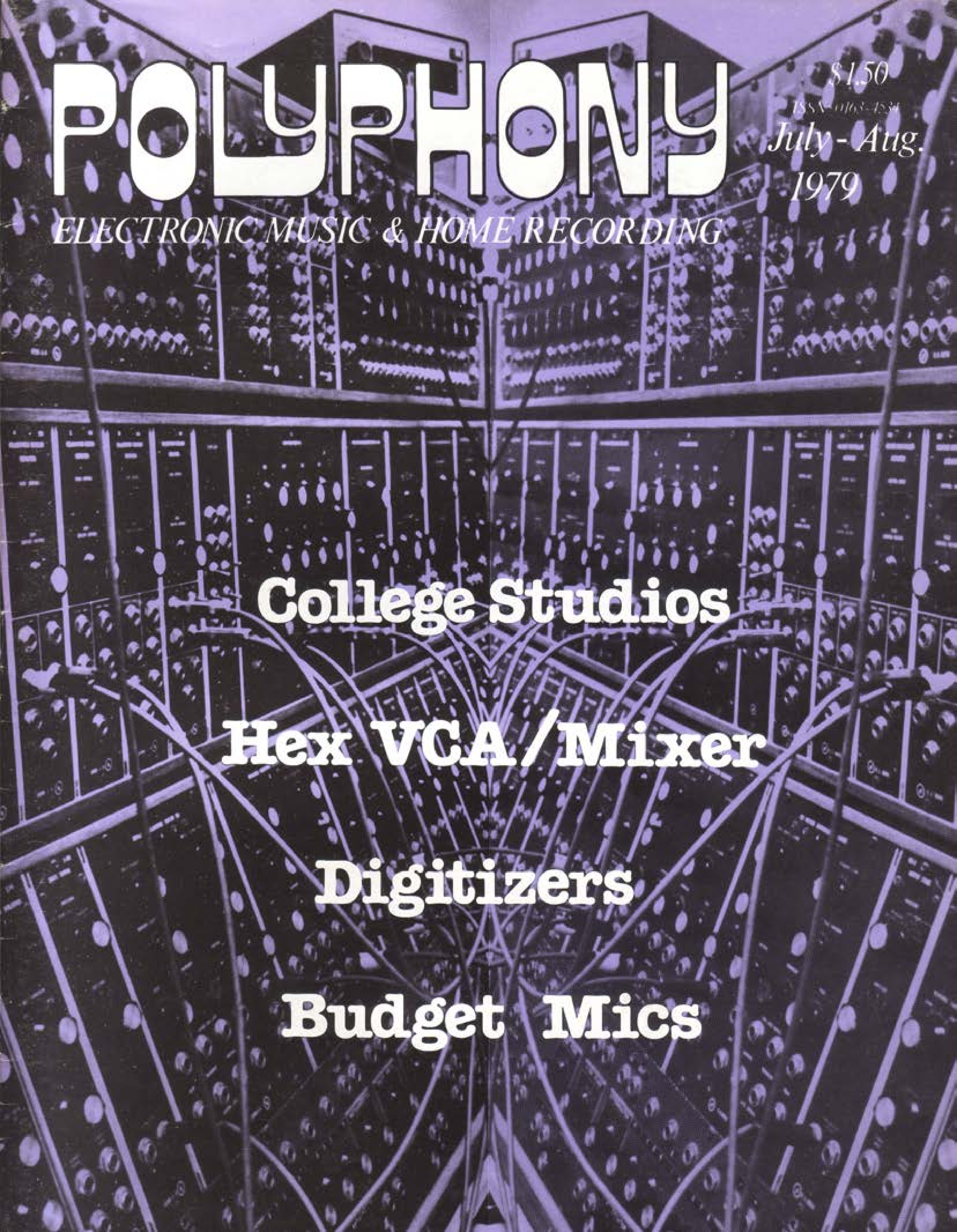

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Volume 5, Number 2
July/August 1979

FEATURES

CONSTRUCTION & MODIFICATION

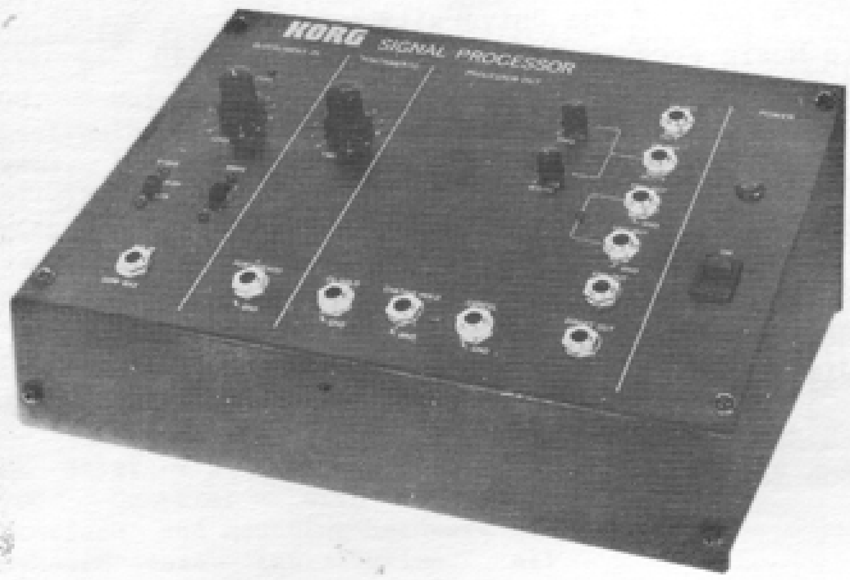
COLUMNS & REGULARS

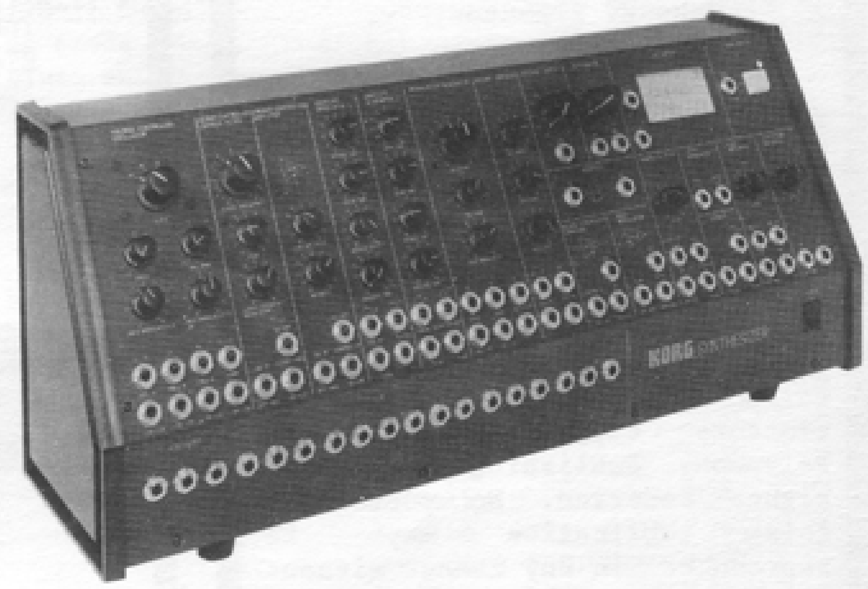
ON THE COVER: The electronic music lab at Roosevelt University in Chicago is a big system to begin with, but, when doubled back on itself, provides a sight which should make any synthesist drool.

Candy-O is a lifesaver.

Tell them you saw it in POLYPHONY

INDUSTRY REPORT:





Interface & Expand

The people at Unicord have been hard at work, and it looks as if they have come up with some real winners. The Korg MS-02 is a Universal Synthesizer Interface which is the first such device to appear on the market, and allows unlimited possibilities for advanced synthesists in interfacing equipment from all manufacturers. The unit provides two control voltage converters (1 Log Amp for linear to exponential conversion, and 1 Antilog Amp for exponential to linear conversion), and two trigger processors which will match two different trigger threshold values, invert triggers for interface between units using "S" triggers and "V" triggers, and buffer trigger sources for use in driving multiple trigger inputs. Also included are Adding Amps to amplify or attenuate control signals (continuously variable from 0 to 2:1 ratio) and add bias voltages for range transposition (variable from -10 volts to +10 volts). Jack multiples are also provided to facilitate inter-unit patching. Suggested list price is $225.

Also for advanced interfacing is the MS-03 Pitch to Voltage Converter. The MS-03 was designed specifically for guitar/ synthesizer interfacing, and the tracking accuracy and pitch stability are optimized for operation throughout the full range of the guitar. Major features include: portamento (a Korg MS-03 exclusive) with optional foot switching capabilities, envelope follower output, direct instrument sound output, mode switch to select triggering characteristics, linear and exponential control voltage outputs, and "S" and "V" trigger outputs. Facilities are provided for foot controlling the control voltage and trigger 'hold', control voltage 'hold', and synthesizer cancel. The suggested list price is $300. I have heard the P to V section of the MS-20 synthesizer, and it really works. Assuming that this unit uses the same circuitry, it should be well worth the cost.

Also new from Korg is the MS-50 Expander Module. This add-on contains a VCO with mixable waveforms, VCF, VCA/high pass filter, two envelope generators with normal and inverted outputs, multi-waveform LFO, adding amp, bias supply, signal inverter, integrator, second VCA, ring modulator, Sample/Hold, White/Pink noise source, waveform dividers, envelope follower/ trigger detector, AC/ DC meter, headphone amp, and jack multiples. The MS-50 should provide quite a bit of add-on versatility for the experimenting musician.

For more information contact Unicord at 75 Frost St., Westbury, NY 11590, or check your local music store.

New Software

E-mu Systems has announced the release of six new programs in their library of special software functions for their 4060 microprocessor based polyphonic keyboard and sequencer. Included in this release are UP1.1, MEMTEST, INTERVALS, PROGRAM MODE, SEQUENCER EDIT, and PAT'S UP1.3 .

UP1.1 is a revised version of the basic software for the 4060 and is available free to owners of the earlier version.

MEMTEST is a field service aid for testing the 4060 and any attached 4065 sequencer memory boards.

INTERVALS is a program that allows the user to redefine the tuning of the 4060 keyboard. Two modes are available. In the DEFINE INTERVAL mode, one can specify the interval between the keys. In the DEFINE KEY mode, one can specify the value for each individual key, thus allowing the user to define any arbitrarily intoned scale. In both cases, control voltage can be defined to an accuracy of .015 semitones over a ten octave range.

PROGRAM MODE is intended for the individual who wishes to write his own programs for the 4060. It allows one to enter programs and data using the 4060's keyboard, and to display addresses and data on the 16 gate indication lamps of the poly keyboard output section. A list of useful entry points and their functions is supplied with the program.

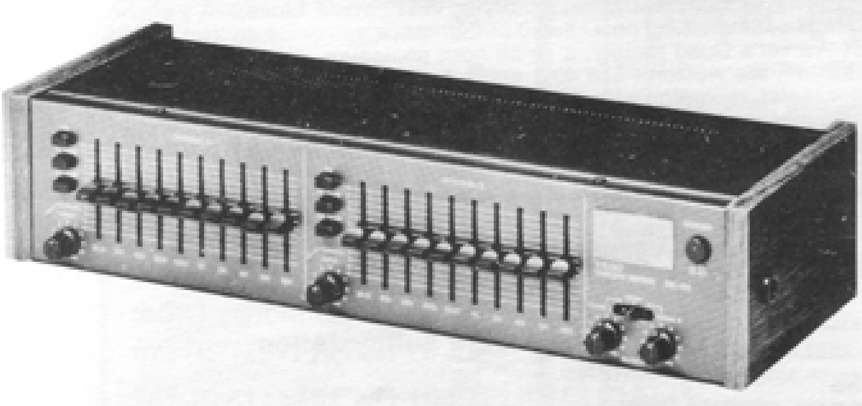
SEQUENCER EDIT is a version of the standard 4060 software modified for special real-time and sequencer editing functions. PUNCH OUT mode allows the repair of one or more notes in the middle of a complex sequence, while a modified STORE SEQUENCE function allows continuous playing while building up multi-layered sequences.

PAT'S UP1.3 is another enhanced version of the standard software. Originally developed to the specifications of composer/ synthesist Pat Gleeson of Different Fur Music, this program allows, in addition to all standard functions, both up and down sequence transposition, the storage of up to 81 individual sequences, the ability to inhibit the recall of selected sequencer channels, and the uninterrupted building of sequences in real time.

All programs are supplied on cassette tape and are loaded using standard "From Tape" procedure.

Programs range in price from $25 to $200. The 4060 Polyphonic Keyboard is $3000. For more information, contact: E-mu Systems, 417 Broadway, Santa Cruz, CA 95060.

Rack Mount EQ



The GE-20, TEAC's first entry in the equalizer field, is the latest in the company's system of multi-track components. The unit's specifications include: frequency response of 20 to 30K Hz $\pm$.5 dB, THD less than .03%, S/N ratio of 85 dB, input level of -10 dB (.3 volts), maximum output of +18 dB (8 volts), output load impedance greater than 10K ohms unbalanced, headroom 28 dB, and maximum boost/ cut of 12 dB. The 10 bands- one per octave- have center frequencies of 31.5, 63, 125, 250, and 500 Hz, and 1, 2, 4, 8, and 16 Khz.

The GE-20 has a high pass filter (12 dB/ octave) at 31.5 Hz and a low pass filter at 16K Hz. The filters can be initiated after completing equalization functions. The unit has input LEDs, input level controls, output level controls, and a switchable meter to prevent overloading the following unit in the audio chain. The equalizer is constructed with active op-amp filters rather than coils to reduce the possibility of RF hum and interference.

The GE-20 carries a suggested retail price of $350, and will be available for shipment in August. For more information, contact: TEAC Corporation of America, 7733 Telegraph Rd., Montebello, CA 90640, Ph. 213-726-0303.

The Drum

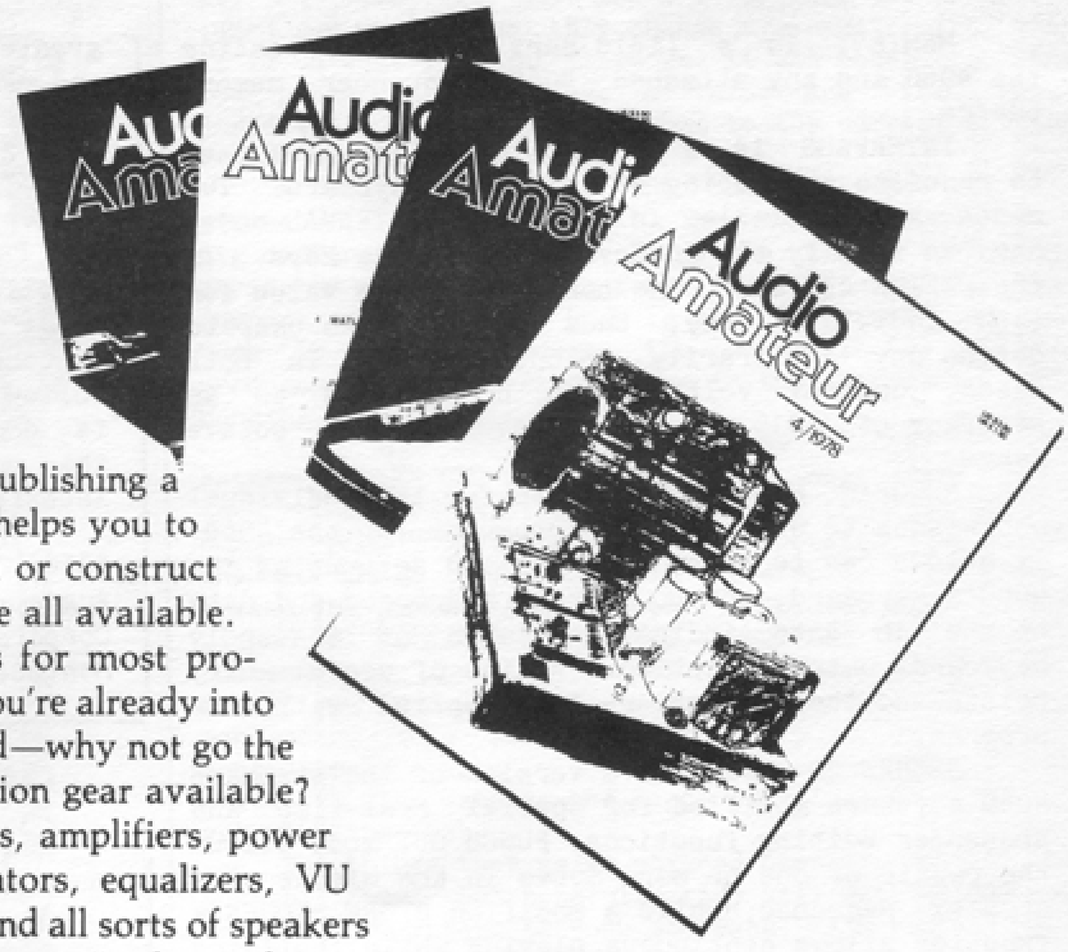
The latest addition to Paia's line is The DRUM (tm) percussion synthesizer. Keywords for the DRUM's design are versatility and price.

The DRUM provides true synthesizer versatility by using continuously variable controls for pitch modulation - up or down, oscillator waveform mix, noise filter frequency, and oscillator/ noise mix. Modularity means that you can configure an electronic drum set ranging in size from mini to monster, and numerous rear panel patching and control points allow multiple cards to be cascaded for an even wider range of effects.

The DRUM's encapsulated sensor can be mounted permanently or temporarily to an existing drum (an optional 'cancel' foot switch disables The DRUM when not needed) or can be used with practice pads available from Paia.

The prices start as low as $59.95 for a kit module. Factory assembled and tested units are available, as are complete kit or assembled four module systems with case and power supply included. For more information, contact Paia Electronics, PO Box 14359, Oklahoma City, OK 73114.

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.

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COMPOSER PROFILES
THE SOUND ENVIRONMENT PAINTINGS of ERNEST GARTHWAITE and DAVID ERNST

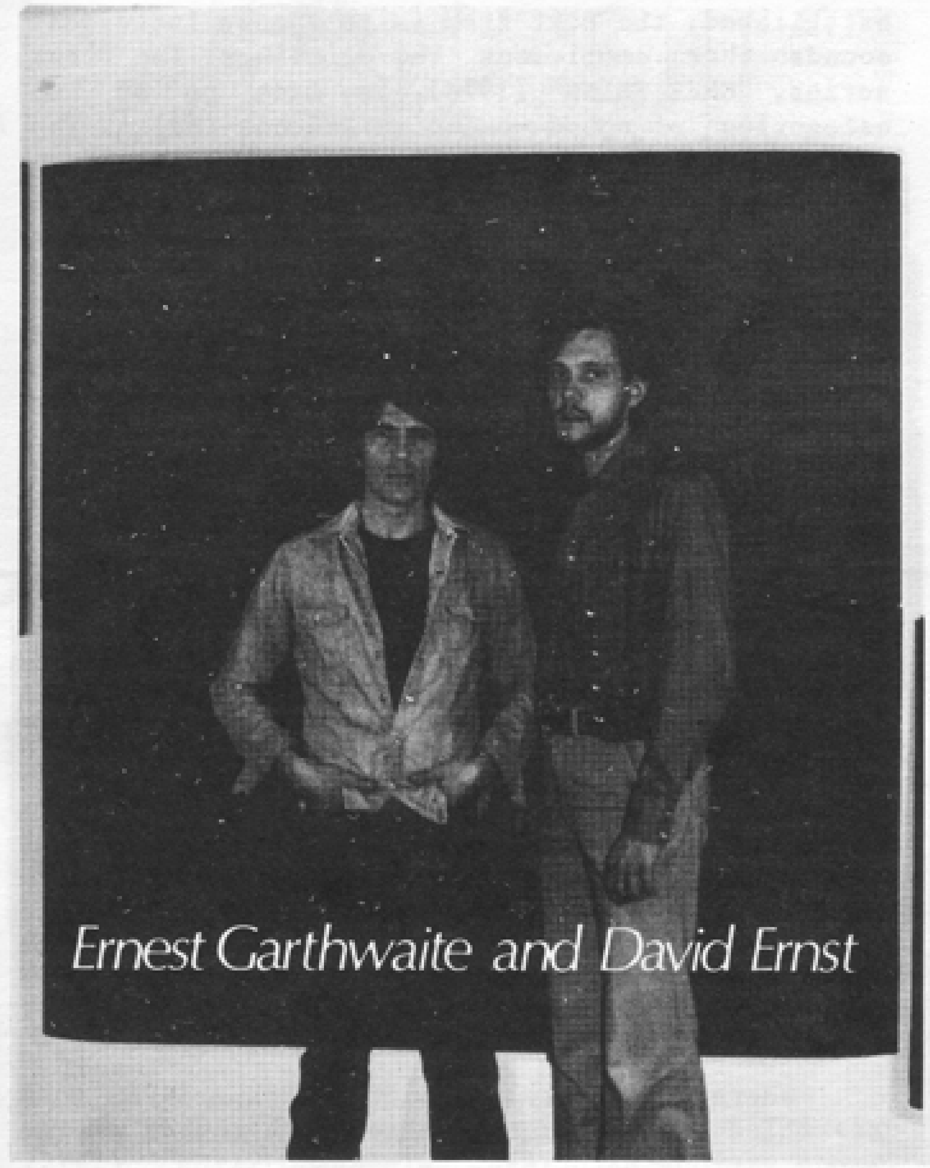
BY: DAVID ERNST

Ernest Garthwaite (b. 1940) was born in Saskatoon, Canada, and he is presently professor of art at York College of the City University of New York. His works have been shown throughout the country, and for the past two years we have been collaborating on a few series of sound-environment paintings ("CREE SKINS" and "WINNEBAGO SKINS"); these have been exhibited in New York City and Sarasota, Florida. The object of these works is to unite the visual and aural media to the extent that they become complementary, and the most obvious example is the method of presentation. A prerecorded tape is played over loudspeakers suspended behind individual paintings so that the paintings 'speak' and, in some instances, vibrate. Before we examine these interrelations more closely we will consider the visual and aural elements separately.

The Indial tribal titles for these works suggest both their visual and sonorous orientation. Ernest Garthwaite depicts the culture and environment of the North American Indian in a variety of ways. The visual imagery is comprised chiefly of landscapes and facial portraits thoroughly developed to a high level of abstraction via layered application of thin washes of acrylic. This process often takes years and relates to the glazing techniques of the old masters, e.g. Rembrandt; the result is shapes, forms, and shadings as elusive as shadows. This effect is intensified by mounting the skin-like canvases on bow-shaped stretchers, whereby the perceptual illusion of movement is created.

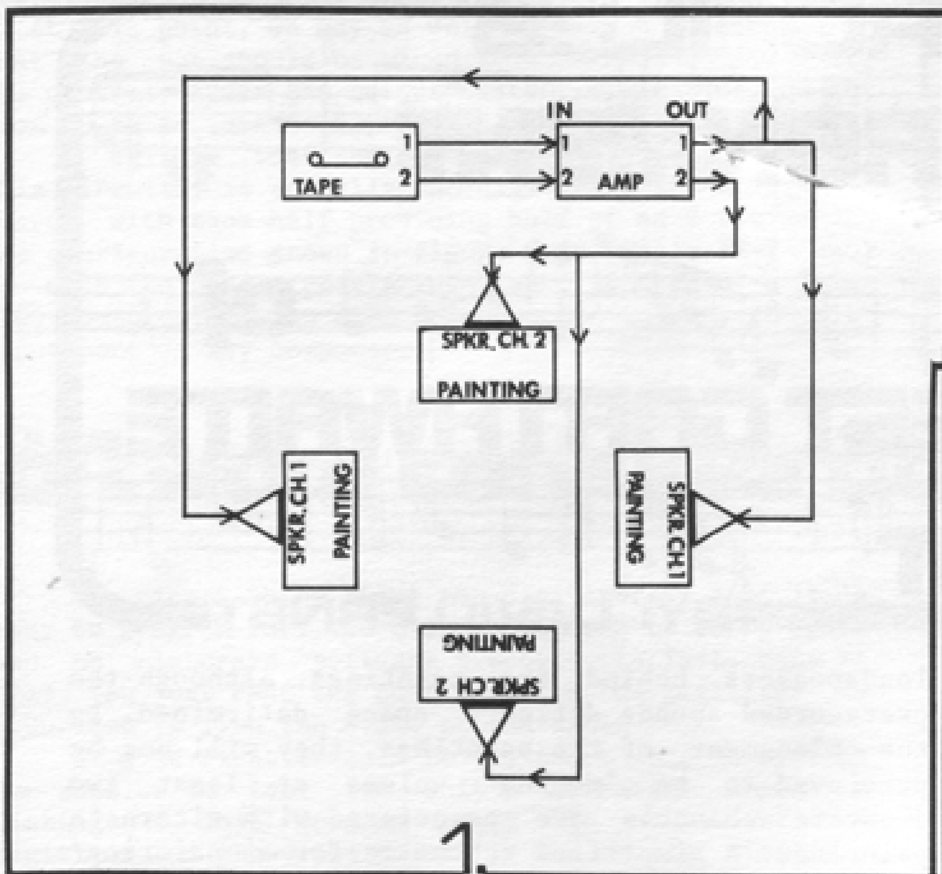
Similarly, the accompanying sounds are often derived from Indian sources. Tapes of tribunal chants and dances are modified by electro-acoustical means such as ring modulation, filtering, feedback loops, tape transposition, and reversal. In addition, some sonorities are produced directly on synthesizer. No attempt is made to isolate or to distinguish concrete sounds from those that are generated electronically. Consequently, specific compositional techniques are employed only as a means to approach the aforementioned unification of painting and sound.

Whereas sounds move in space, paintings are stationary. This disparity provoked us to search for a way to integrate our method of visual-aural presentation. A visual illusion of movement results from the shaped or sculptured canvases and stretchers, and this is enhanced by placement of loudspeakers behind the paintings. Although the prerecorded sounds define a space detirmined by the placement of the paintings, they will not be percieved to be 'moving' unless at least two discrete channels are associated with alternate paintings. A simplified schematic for a painting/ loudspeaker network is illustrated in figure 1. We are presently only working in stereo, but there is really no limit upon the choice of number of channels. We generally set up more elaborate networks with paintings in corridors, corners, etc., so that the sounds 'jump' across rather large areas. It should also be mentioned that the



Ernest Garthwaite and David Ernst

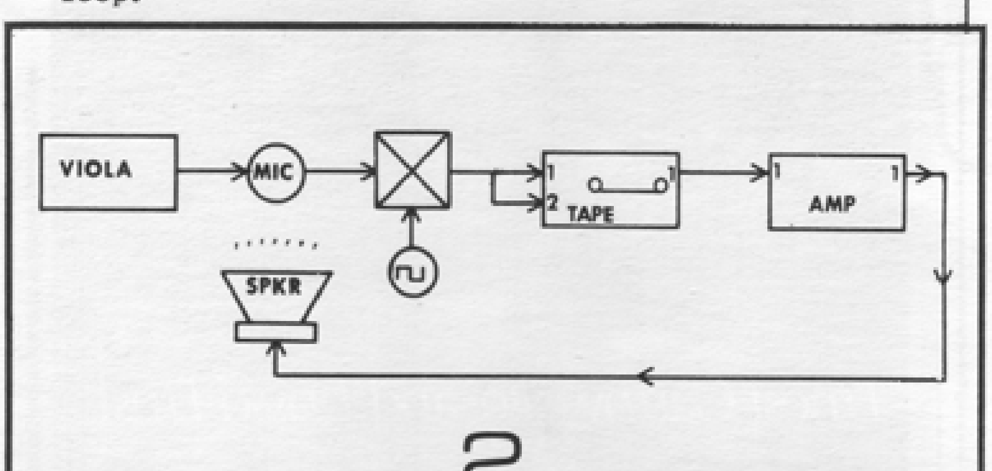
PHOTO BY: JONI DELL'AQUILA



1.

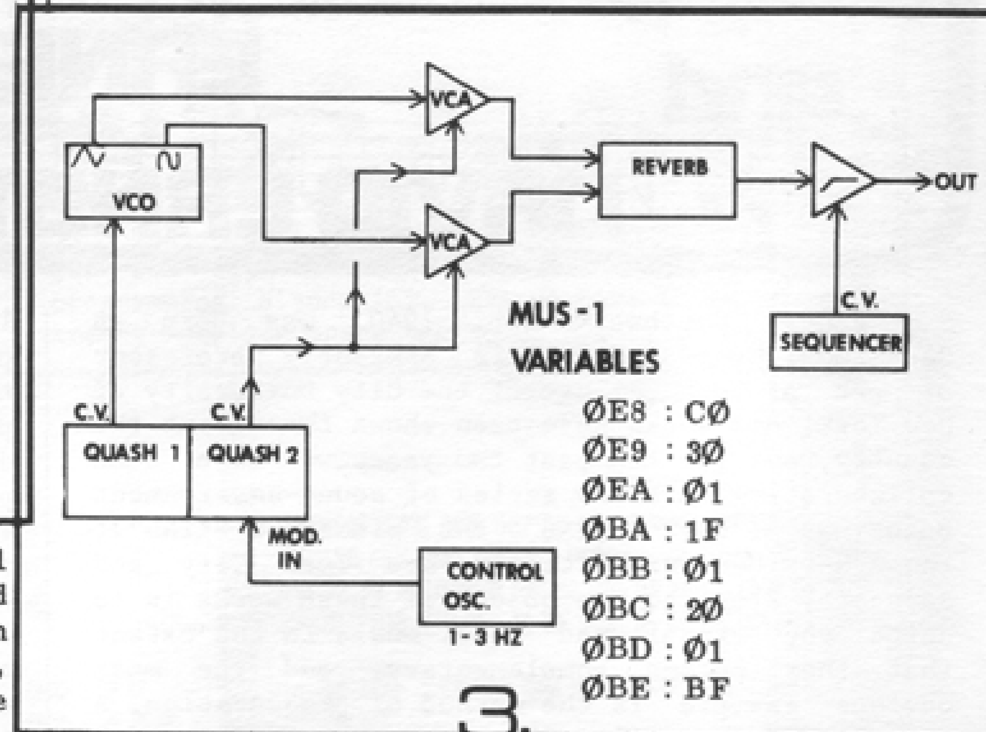
playback levels are critical, for many spatial effects are lost if the sounds are either too loud or too soft. Since this is dependent upon acoustical properties of individual rooms, amplitude levels are usually detirmined after the loudspeakers and paintings have been positioned.

After the sonorous space has been established, the next step is to choose individual sounds that complement the paintings. The first series, "CREE SKINS" (1978), is made up of two categories of sound -- Indian and non-Indian. The former includes authentic tapes of tribal chants and dances, and these are frequently altered by ring modulation, filtering, tape transposition, and reversal to establish a closer relation to the abstract nature of the paintings. The non-Indian sound, transformed viola, was chosen because its distinctive timbre helped to articulate the Indian material, yet its unique tone quality did not alienate it from the overall timbral context. Figure 2 illustrates the recording technique used for the viola sound--ring modulation and feedback loop.



2.

Whereas the aural portion of "CREE SKINS" was assembled in collage fashion with sounds moving among various paintings, "WINNEBAGO SKINS" (1979) exhibit strict structural delineations. Individual sections were constructed upon Indian rhythmic patterns electronically produced via the Paia computer drums interface. Upon this framework, additional tracks were layered, including the viola texture from "CREE SKINS" (to link both series together), an authentic fish chant (moderately filtered so that some intelligibility is retained), and an electronic analog of a wooden flute (played in the melodic idiom of the Winnebago Indians). The flute sonority was produced on a P4700J system using the MUS-1 software; specific details are given in figure 3.

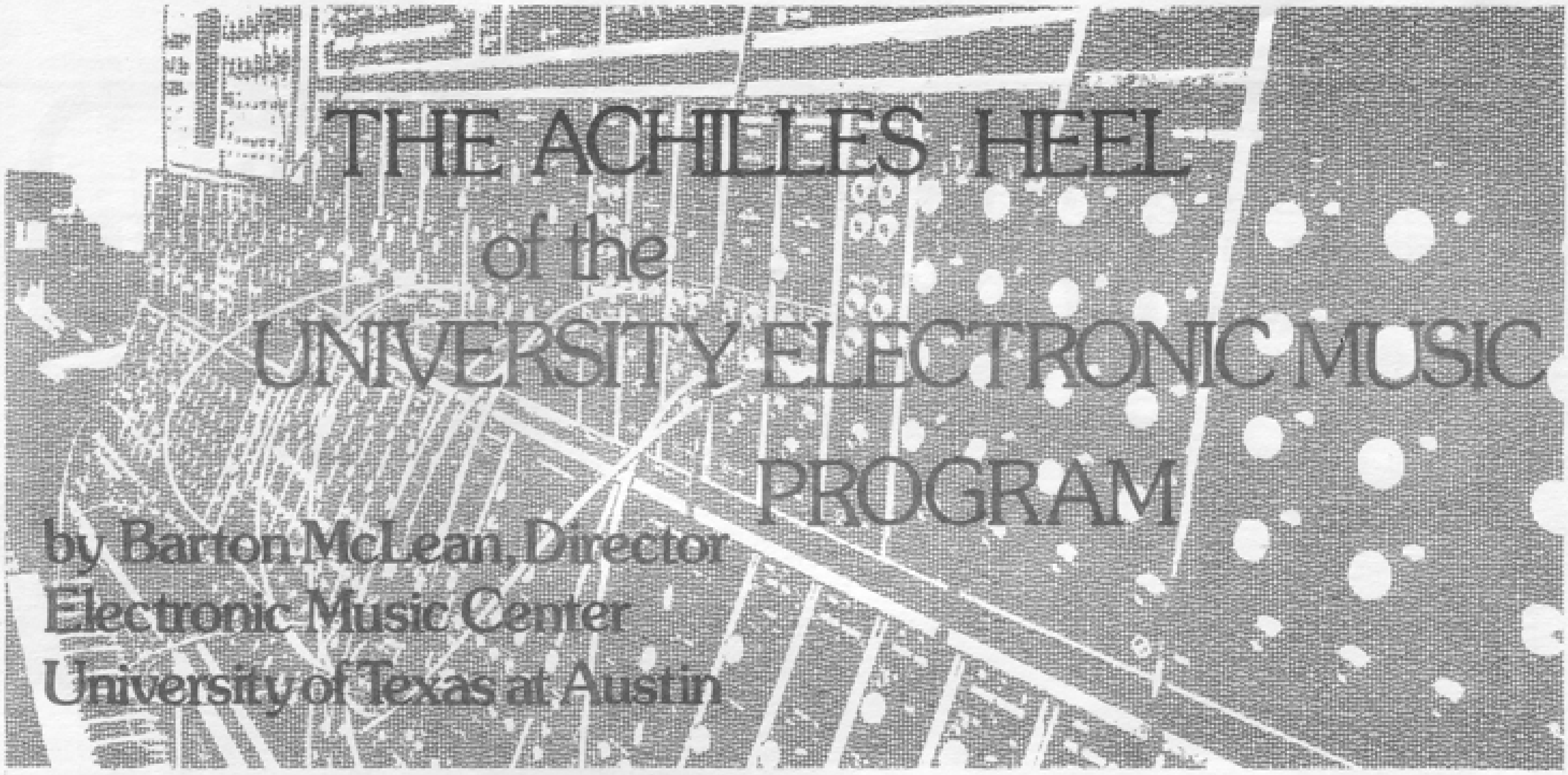


3.

Musical authenticity was a prime concern in both series. In "CREE SKINS" this was provided by extensive use of original American Indian tapes frequently modified by conventional synthesizer and recording techniques. Conversely, "WINNEBAGO SKINS" relies less upon Indian recordings in an attempt to create an appropriate sonorous environment via electronic means. An advantage of the latter method is exemplified in the rhythmic design of the subsections. A characteristic of this music is for a rapid series of drum beats to follow a slower drum pattern. Because of the ability to produce the drums electronically, we were able to chose timbres, tempi, patterns, etc., with great precision and relative ease, and to time accurately the length of each section.

Both "CREE SKINS" and "WINNEBAGO SKINS" establish a visual and sonorous atmosphere suggestive of the American Indian's culture and environment. The paintings are abstract, with concrete Indian imagery veiled by thin layeres of multi-shades of brown acrylic so that the paintings assume the form of stretched canvases. Lighting and the viewer's focal point detirmine the extent of image recognition, thereby releasing the illusion of visual motion. This subtle visual movement is enhanced by the accompanying tapes, and the result is a continually changing visual-sound environment wherein the music is designed to articulate and to define a physical space within which spectators move freely.

A cassette tape including a two channel mix of the sound tracks for "CREE SKINS" and "WINNEBAGO SKINS" is available for $5 from: David Ernst, 150-29 89th St., Howard Beach, NY 11414.



THE ACHILLES HEEL of the UNIVERSITY ELECTRONIC MUSIC PROGRAM

by Barton McLean, Director
Electronic Music Center
University of Texas at Austin

How many are the ways that a university electronic music program, looking good on paper, can fail to fulfill its potential due to problems with its "nuts and bolts" daily operation and lack of administrative support! It has become increasingly clear to me in my travels and correspondence that there is very little awareness of what constitutes the norms and the options in coping with these problems. All the sophisticated hardware in the world will not be productive if insufficient time is allowed for the composer to work, nor insufficient funds allocated for studio maintenance, support of electronic music concert programming, new equipment, teaching assistants, monitors, research and design, proper security arrangements, and adequate faculty load compensation. This article is designed to provide a sense of where we are in the US university in regard to these issues, points out some questions of basic studio philosophy, and is written to be of service to students who would like more searching criteria for the evaluation of electronic music courses of study and to faculty who have been grappling with these issues locally and who need objective information with which to evaluate and seek support for their own programs.

The basic data for this article was compiled from a survey of 12 electronic music programs across the US. They vary widely in size, philosophy, location, and production, although there is a bias toward the larger studios in major music departments in the Northeast, Midwest, and Southwest. My main criteria for selection of these particular twelve were personal knowledge of and repect for the directors as well as the wish to achieve variety and balance. Perhaps a measure of the importance these directors assigned to the topics raised can be found in the high percentage of those returning the questionnaires (12 out of 13) (see the end of the article for the results of the 13th survey which was recieved late) and in the care and thoughtfulness with which they approached the questions, for which we are grateful. (See Appendix for a listing of the schools and directors participating in the survey.)

Typical Size of the Programs

With an average music department size of 500 in the survey (ranging from 20 to 1500), totals of beginning, intermediate, and advanced students were tabulated. The average beginning class had 14 students, intermediate classes had 9, and advanced classes had 8 students (with only nine of the questionnaires indicating an intermediate category). The largest programs were found at SUNY- Stony Brook (25 + 12), North Texas State (18 + 18 + 15), and Northwestern (33 + 15) on a per semester basis.

Course Offerings

Generally, the smaller, private schools which deal only with the undergraduate level have an introductory electronic music course and perhaps an advanced studio composition course. As to the larger music departments, three emphasized acoustics, literature, and introductory electronic music undergraduate courses, reserving serious studio composition for the graduate level only. On the other hand, seven of the larger schools allowed all levels of students to take beginning and advanced studio work.

A distillation of the average course structure from all of this would resemble: Electronic Techniques I, Electronic Techniques II (sometimes), Advanced Studio Composition, with perhaps a computer seminar or course in acoustics or literature. One of the most exciting and complete programs was developed by Dr. Cleve Scott of Ball State University. His program leads to degrees in audio technology as well as composition, an unusual feature in an academic world where the usual procedure is to view electronic music as an adjunct to the compositional degree program.

Ball State's Program

Undergraduate Level:

M 225 Introduction to Recording Technology
M 226 Advanced Recording Technology
M 325 Compositional Practices in Electronic Music
M 425 Electronic Music Studio I
M 426 Electronic Music Studio II
M 427 Intro. to Computer Applications in Music
M 434 Composition in Electronic Music
M 429 Composition-Project Recital
M 491 Independent Study in Music Theory

Graduate Level:

M 525 Electric Music Studio I
M 526 Electric Music Studio II
M 529 Composition- Project Recital
M 591 Independent Study in Music Theory


photo by Michael Dunham
Computer Studio, Ball State University
David Waite, Electronics Technician

Elegibility Requirements for the Programs

What is electonic music? Is its essence to be found only in the compositional process, or does it constitute the total interrelated phenomena subsumed under its heading including literature, acoustics, engineering concepts, performance, recording/ film/ radio/ television concepts (or perhaps a combination of these)? Must the student approach its study exclusively through studio composition as the final goal, or can equally


photo by Michael Dunham
Main Analog Studio, Ball State University
Michael Rees, Graduate Ass't; Assoc. Studio Director, left
David White, Electronics Technician, right

legitimate case be made for similarly rigorous but non-compositional role? It is clear that the respondents were divided on this issue. Seven of them cited 90 - 100% of music majors (not necessarily composition majors) enrolled in the electronic music program, three of these being 100%. Five of this group required substantial music theory and composition training as a requirement for entrance into the program. In general these were the larger music departments who had plenty of "qualified" students to fill the class roles. On the other hand, four schools (two of these had large music departments also) had a range of only 20 - 60% of music majors in their programs. In lieu of specific prerequisites these latter schools rely on personal interviews as an eligibility requirement. Three of the four mentioned "weeding out dilettantes" during this interview.

Although practical considerations in large music departments often mitigate against a more open enrollment policy, I personally feel we must keep a flexible position designed to service the serious student in other disciplines, not only out of fairness, but also in the broader view of increasing the public awareness of our art. There is a dangerous tendency for us to become so insular, so safe and comfortable in our own little world that we further decrease the already miniscule audience that we have.

Studio Time: How Much is Enough ?

As one who shares a very large home studio with his wife (Priscilla McLean) and even then is barely able to secure enough studio time, I should say at the outset that under most circumstances I have seen, significant composition at any university or professional studio is impossible at best (although since it is highly subsidized, particularly in terms of teacher/student ratio and overhead, the university program in my opinion usually offers vastly superior benefits when compared to the professional facility, which must break even on its program). In any public situation the composer is unable to focus enough of his highest energy for a long enough duration to avoid making numerous comprimises (many of these becoming normal features of his technique) which, in total, force him to produce an inferior composition. As far as I am concerned, much of the really significant work is done outside the teaching studio. If we view this facility not as one designed for mature compositional activity but rather as one to develop conceptual and technical attributes we will be giving our students a more honest experience.

What about the faculty composer who cannot afford his own studio? A solution to this problem has been found at North Texas State University where Merrill Ellis has designated one of his several studios as a "faculty research" facility. Nine of the other respondents have at least two studios, one for advanced composition and the other for teaching purposes, providing for a more efficient use of studio time.

The average weekly private studio time for beginners is 4.5 hours, with the best situations at Smith College (8 hours) and the University of Iowa (10 hours). All twelve studios provide some individual studio time for beginners each week, a crucial and encouraging sign. The average individual weekly time for intermediate students is 6.5 hours, and for advanced composition- 11.75 hours per week. The University of Texas at Austin, New England Conservatory, and Western Michigan University all listed "unlimited hours" for their advanced students. This figure was arbitrarily set at 20 hours per week for purposes of averaging with the others in the survey. Four of the schools surveyed gave their students less than 8 hours per week for advanced composition.

Security Arrangements

Security problems have a tremendous impact on the ability of the program to function effectively, since a studio that has been broken into with resultant damage or loss not only has that limiting factor but may also impose severe restrictions on the student's ability to work alone. A variety of solutions were offered toward increasing independent studio time while minimizing security risks. Beginners are allowed to work alone immediately in four of the twelve programs and almost immediately in three others. In three additional situations a paid teaching assistant is present during the first semester of studio experience. In one other case a student monitor was present, and one program maintained a full faculty presence. On the advanced compositional level, ten of the twelve programs allow students to work independently and alone, while one utilizes a monitor/teaching assistant combination and one a faculty member. Students working alone gain access in a variety of ways. Three programs allow each qualified student to sign out a key for the semester. Three others sign out keys from guards, secretaries, or librarians for short periods. Two other programs have a system similar to the above, with a double key system where only the director or his designate has the key to the second lock which is secured during nights, vacations, etc. Finally, two studios admit their composers by paid staff who retain the keys.

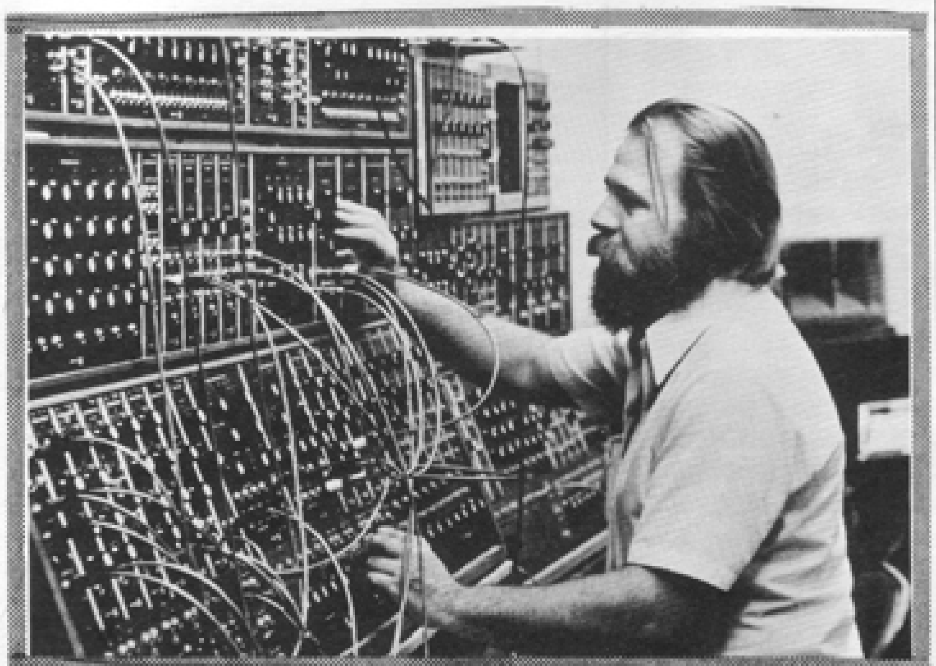
photo by Ann Shanahan

Director Ron Perera (center) with students in the Smith College electronic music studio.

Repair and Maintenance

An electronic music program is only as good as its ability to maintain equipment. A wide range of levels of support was found. The University of Texas at Austin, University of Iowa, North Texas State, and Ball State University have their work done by resident engineers, and routine maintenance done on a more or less regular schedule. At least two of these schools employ full time tech engineers for the electronic music program (and related areas) exclusively. Three other programs hired an outside engineer to accomplish a reduced version of the above, with small equipment often sent out to local repair shops (if possible). In the five remaining situations, whatever repair and maintenance was accomplished was done by the director or his students as "extra duty". Again a question of student philosophy: Is the director to be academically- or engineering-oriented, or both? Should he be expected to repair his own equipment? I think that all would agree that the teacher who must do this is forced to sacrifice his effectiveness, either as a teacher or a composer, if he is not granted faculty compensation or load time specifically for repair. After all, do the pianists have to repair and tune their instruments?

The average annual repair budget, including salaries and hourly wages prorated to show actual time spent on studio work, and supplies, was $3,218. Five of the twelve schools had new equipment budgets which averaged $2500 per year. Five of the others had no new equipment money and relied on Title VI funds, acts of God, etc.



Don Malone
Roosevelt University, Chicago

Concert Support

The ultimate goal of much studio activity being performance, it is heartening to see the support given to this area by the majority of schools. Ten of the twelve schools provided paid setup crews for electronic music concerts. Seven schools provided funds for the rental or purchase of tapes and performance materials, a necessary item for any department wishing to break from the insularity of only local compositions on its electronic music programs. There is absolutely no reason why a music department which provides for rental of orchestral, chamber, and opera performance materials can not do the same for electronic tapes and scores. To do less is to discriminate against the program and to deny its students the broad outlook they need.

Main Problem

The question was asked: "What do you see as the main problem in operating your program?" The largest number - six in all - cited equipment repair problems. Three mentioned lack of new equipment budget; two said "space and money", and one had as its biggest headache studio security.

Teaching Load

An average of 55% of the teaching load of the twelve respondents was directed specifically toward electronic music.

Conclusion

The common point through much of this is our need to develop an awareness of what it means, in terms of program design and administrative support, to operate a successful electronic music program. There is perhaps no other area in music colleges so dependent on the whims of the budget and on sympathetic understanding of requirements for a smooth running program. I feel we are moving toward standards in this area, such as the full or part time studio tech positions, proper support of teaching assistants, monitors, a rational approach toward upgrading and replacing old equipment, concert programming, and others. I have been encouraged to see most of these necessities becoming realities in a growing number of programs. Further progress will be tied directly to the degree to which we educate the 'purse string powers' to the self-defeating results of the "Achilles Heel Syndrome" as it relates to electronic music program support.

Author's note: The 13th response (of 13) was recieved from Scott Wyatt of the Experimental Music Studios of the University of Illinois at Urbana (School of Music of 900 enrollment), having been delayed in the mails. Interesting features of the program are: stationing of a monitor or teaching assistant through beginning and intermediate private studio work, a strict key policy, a half time tech engineer in residence, and a high percentage (98%) of music majors with theory/composition backgrounds enrolled.

Appendix

List of all Directors and Studios replying (equipment lists are in the June 78 Contemporary Keyboard, along with program and studio access information for over 100 studios in the US.).

Robert Ceely: New England Conservatory of Music, 290 Huntington Av., Boston, MA 02115
Merrill Ellis: North Texas State University, Denton, TX 76203
Hubert S. Howe, Jr.: Queens College, CUNY, Flushing, NY 11367
Peter Tod Lewis: Univ. of IA, Iowa City, IA 52242
Don Malone: Roosevelt Univ., 430 S. Michigan Ave., Chicago, IL
Barton McLean: University of Texas at Austin, Austin, TX 78712
Ronald Perera: Smith College, Northampton, MA
Cleve Scott: Ball State Univer., Muncie, IN 47302
Stephen Scott: The Colorado College, Colorado Springs, CO 80903
Daria Semegen: State University at Stoney Brook, Stoney Brook, Long Island, NY 11794
Stephen Syverud: NoWstn. Univ., Evanston, IL 60201
Scott Wyatt: Univer. of Illinois, Urbana, IL 61801
Ramon Zupko: Western Michigan University, Kalamazoo, MI 49008

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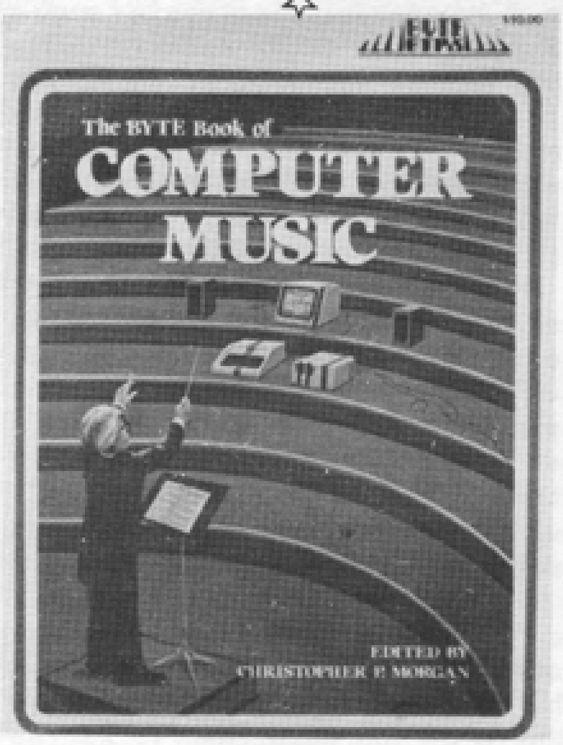
reviews

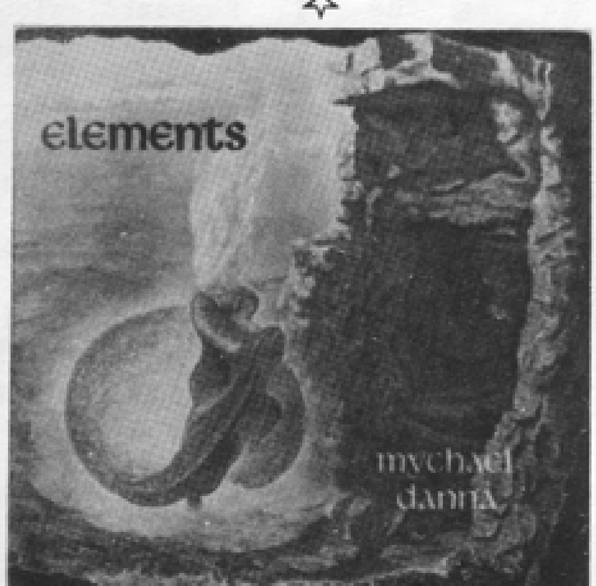


(available for $10 plus shipping from Polyphony's Bookpage)

Computer music is moving fast. As an artform, it is growing daily as new hardware and techniques become available for circumventing the problems encountered in yesterdays technology. To the musicians who aren't too interested in technology anyway, and to the musicians who have spent years on learning and understanding the analog techniques of sound synthesis, digital just presents what seems to be an unfair barrier to their mastery of the art. What most of these people need is an introduction to the field which surveys the practical applications which are available now, rather than texts that pile on the theory and experimental applications for the future. Byte's book fills the bill very well.

"...Computer Music" provides reprints of 17 articles from past issues of BYTE magazine which offer both theory and construction projects for the aspiring computer musician. Several articles are presented which cover the interface of computers to pipe organs and commercial electronic music equipment, as well as several articles with hardware output devices to generate sound directly out of the computer. Systems for which these articles are written vary from small KIM type machines up to mainframe systems like the SWTPC 6800 and Apple II. For the more experienced computer musicians, there are articles on the theory of Walsh functions and Fast Fourier Transforms (complete with programs for doing real time FFTs on a 6800 or Z80). For the really experimental avante-garde musicians out there, construction plans are included for a "Terrain Reader" which converts data on topographical maps for use in composing music.

Everyone should find several things of interest in this book. For the more experienced, the book serves well as a reference manual on digital techniques which can be kept near your bench, computer, or studio. And for those who are into sharing ideas and experiences with other experimenters, the book thoughtfully contains a listing of all the authors addresses at the back of the book. Until this field slows down a bit (Ha!), it is going to be hard to find textbooks which aren't outdated by the time they are released. "The BYTE Book of Computer Music" is the type of publication we need in order to get maximum application of new technology with a minimum amount of bookwork.



ELEMENTS
by Mychael Danna
FHR 791

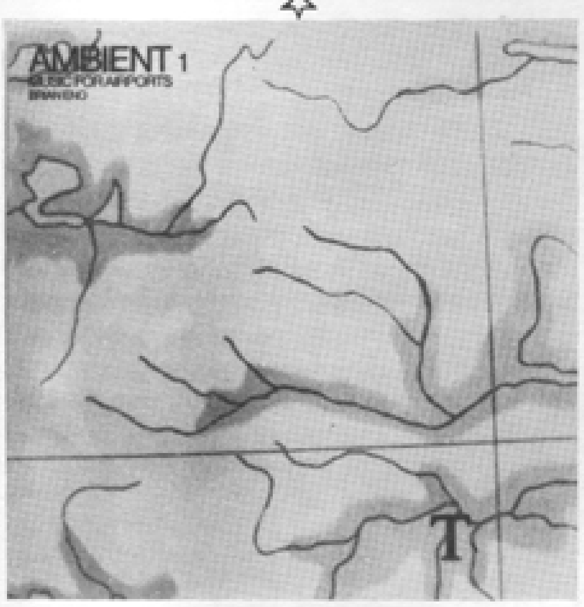
$6.95 from Frederick Harris Music Co. Ltd., 529 Speers Rd., Oakville, Ontario, Canada L6K 2G4.

Mychael Danna is a relatively young synthesist whose list of accomplishments is really quite impressive-- several piano solos published, film scores, radio and TV commercials, and a successful rock band in Canada. Upon listening to "Elements", his varied background shows up as smooth production, a clean and balanced mix, excellent composition, smooth cooperative textures in the synthesis, and just the overall feeling that this guy has worked hard and knows what he is doing. Mychael's background is obviously very heavily classical. There is a definite strength and accomplishment in his keyboard technique, and a refined and organized style in his composition. All compositions are strictly tonal and "related to the music of the past, as Danna believes in the natural importance of form and tonality. ELEMENTS definitely removes itself from the 'experimental' electronic albums which so often consist of noise effects, plus noticeable lack of continuity." (Quoted from the press release.) The album is mechanically very clean, with good mastering and pressing.

Side one contains the compositions "Fire" and "Air". The listener is initially pulled into the album with a Jarre type fade-in of strings and Mellotron choirs. The section which follows is much more electronic in texture with the synthesis taking on a Synergy type punch, but with more classical or Carlos type influences in the composition and patch development. The segue to "Air" is accomplished with one of the best explosions I've ever heard. This explosion is not any louder than the program material which precedes it; in fact, it is several dB below the level of "Fire". But the power which is implied in the explosion is due solely to the patch derivation and stereo processing used. Very impressive. "Air" is primarilly a solo piano piece with a synthesized 'chamber ensemble' and flute chorus accompaniments.

On side two, "Water" starts the program which is followed by "Earth" and "Ether". "Water" would make a good movie soundtrack, with its powerful orchestral synthesis and moody arrangement with sound effects. In the middle of the composition is a strange synthesized sound similar to a baby crying in the distance which progresses into the albums first evident use of sequencing. A strong brass fanfare closes the cut. "Earth" consists mostly of strings, flutes, and brass in a context which lacks a great deal of repetitive rhythm structure, thus producing a more liquid composition. The opening of "Ether" sounds like a raging forest fire or volcanic eruption, which drops the listener right into a brass/woodwind choir doing a grandiose baroque style finale.

Mychael Danna's "Elements" is an impressive entry into the solo album market. Regardless of whether you enjoy classically influenced music or not, I think most of you would enjoy this album due to the many layers of sound, the wide variety of patches, and the smoothness of the production.



MUSIC FOR AIRPORTS
by Brian Eno
PVC 7908

"Music For Airports" is a difficult album to review. In a sense, the album can NOT be reviewed. What needs to be reviewed is the effect of listening to the album. Huh?

What Eno has done is analyze the position of background music, or Muzak, in our society and set out to compose specifically for that medium. Theoretically, this sounds like a good idea with the possibilities for great success. After all, the people who brought us Muzak were very calculating about the whole thing in the first place. They knew Muzak must be very soft, so the music is almost subconsciously existing. They also knew that the music must appear and disappear at irregular intervals in order to become a part of a changing environment, yet there must be no repetitive or predominant rhythm patterns in the music to lock the listener's routine into any outside cycles. One thing the original Muzak failed to recognize is that everyone has different musical tastes and lifestyles, therefore generating different requirements for environmental music. Eno addresses this problem specifically, and recognizes the need for each individual to generate (or accumulate) his own personal catalog of ambient pieces.

"Music For Airports" contains four pieces which, by themselves, are not so interesting to sit and listen to (or review). The first is based on loosely structured piano lines by Robert Wyatt, with synthesized bass lines and harmonic reinforcements, and occasional introduction of second themes by chimes, synthesizers, and choir-like timbres. The second cut is entirely done with four 'a cappella' female voices. The voices seem to be processed with envelope manipulation, most often to provide sequenced fade-ins. The extreme stability of the voices sounds as if tape loops were made for each voice singing several notes from a scale or chord, the loops then being manipulated and mixed to yield the final ambient piece.

Side two first presents a cut which uses piano lines with vocal accompaniment, not unlike a cross pollination of the two pieces of the first side. The remaining piece is primarilly synthesized, with woody and brass type timbres in slow swelling envelopes.

After several listenings, and after playing the album while I was in the process of doing other things, I think Eno has a workable idea. I personally did not mesh with some of the environments he created on this album, but that is to be expected. If additional releases of ambient music are produced, I will definitely pick them up, try them out, and eventually assemble (along with original work) a personal collection of environment music. Give it a try!

GALACTIC FANTASY/EASTERN REFLECTIONS
by William Hoskins

Spectrum SR-106

(Available for $6.95 ppd from: Uni-Pro Recordings, Inc., Harriman, NY 10926.)

William Hoskins is director of the electronic music studio at Jacksonville University in Florida. His list of credits on the back liner are extensive, but the synthesis presented on this album breaks little ground.

The "Galactic Fantasy" on the first side contains five movements- Overture: Stars are Suns, Intermezzo: Interplanetary Communique, Star Nocturne, Scherzo: Comets, Beyond Beyond: An Entropy Study. The themes are cliche', and most of the synthesis work comes off equally as dry. There seems to be a fascination with processed noise and square wave modulation of audio oscillators throughout the side. The dynamic range is used quite well, and highlights the quiet recording and pressing job.

"Eastern Reflections" on the second side shows more planning, study, and development. "Eastern Reflections" is subtitled 'A Suite for Imaginary Orchestra', and consists of six movements- Prolog: Introduction, Theme, and Variant; Lower Heterophonie; Song: Open Skies; Drum Chime; Upper Heterophonie; and Epilog: Processional. The composition throughout this work is based on a view of the future where multiple Eastern musical influences have been melded into a common style, along with influences from Western music. The 'orchestra' which Hoskins has created consists of simulations of existing types of instruments (wooden and metal flutes, oboes, gongs and drums) plus instruments which do not now exist, except through the synthesizer (bass oboe, rattle chime, bowed alto drone, etc.). Interesting patches and tonal colors occur throughout the suite, and are accented by performance in a number of modes and temperaments. Much moreso than "Galactic Fantasy", "Eastern Reflections" offers an interesting application for synthesis and composition.

MODIFY MODIFY MODIFY the SEM for LFO TRILLING

by: Bob Yannes

I happen to feel that the Oberheim Synthesizer Expander Module (SEM) is one of the finest performance synthesizers on the market. It is not only inexpensive (without case and power supply) but it provides features not usually associated with smaller performance synthesizers, such as a multimode filter, inverted modulation, and the availability of virtually every input and output at the rear panel molex connectors (including some that aren't even available on the front panel!). Obviously the SEM was designed as a modular system and then normalized -- observe that there is an LFO trigger input for gating the LFO which is available at the rear panel, but isn't even mentioned on the front panel. Both oscillators have sync inputs and outputs, despite the fact that only one oscillator is sychronizable via the front panel. Those are some of the hidden treasures in the SEM, however, anyone with a SEM manual knows that they are available inside. There is, however, more treasure that is deeply buried amid the jewel-like circuitry.

The SEM uses a very common LFO design based on two op-amps configured as a Schmitt trigger within an integrator loop. The integrator produces a linear slope from the Schmitt trigger output until it reaches the Schmitt trigger threshold, causing the output of the Schmitt trigger to change state and reverse the charge slope of (or discharge) the integrator until it hits the lower threshold of the Schmitt trigger. This is a continuing process and the result is a triangle wave from the integrator and a square wave from the Schmitt trigger. Oberheim takes the triangle output and uses it to overdrive a differential input amplifier (CA3080) which produces a nice smooth "sine" wave for LFO modulation purposes. The frequency of the circuit can be varied over a wide range using a single pot. The circuit is very popular and is used by ARP and others, and has appeared in Electronotes and other hobby magazines.

Those of you who are familiar with the SEM know that no provision has been made for square wave LFO modulation, but the square wave is available inside nonetheless. This waveform is best tapped off the 33K feedback resistor on the Schmitt trigger. As viewed from the bottom of the "A" circuit board, this particular resistor is the 33K resistor, color code orange-orange-orange, directly to the left of yellow connector "F". The resistor is parallel to connector "F", directly beneath a 741 op amp and directly above a 390K resistor and 'board to board' connector "S". See figure 1. Be sure to take your tap from the side of the resistor which is closest to connector "F".

Before we can use this signal for modulation purposes, we must "condition" it. The square wave generated by the Schmitt trigger switches between voltage levels that approach positive and negative supply levels. You may think that in order to use this waveform we would need to attenuate the level (as a ten volt peak-to-peak modulation waveform corresponds to ten octaves of modulation depth on a one-volt-per-octave synthesizer like the SEM). You may need to attenuate the waveform for some purposes, like triggering the envelope generators or modulating the filter. However, if you are interested in modulating the oscillators for

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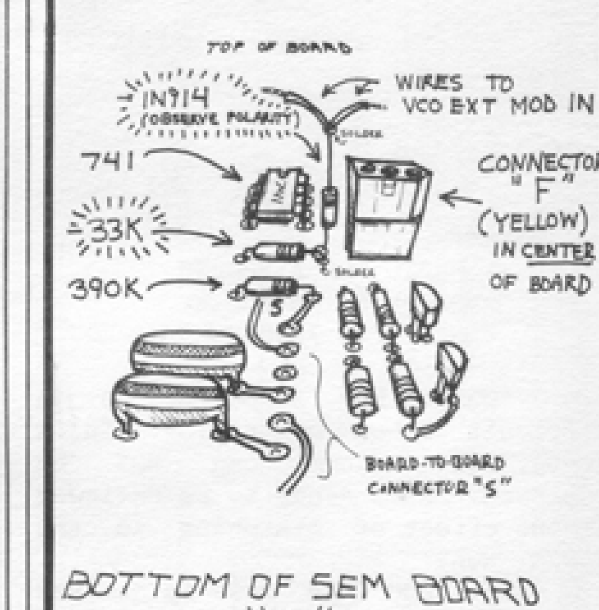

SEM MOD FOR SQUAREWAVE LFO

LFO controlled trills, you don't need any attenuation at all, as the inputs to the oscillators are already highly attenuated. One thing you do have to do is clip off the negative half cycle of the square wave with a diode. This is necessary to put the baseline of the effect always at zero -- if you don't clip the waveform, as you increased the depth of the effect, the high end of the trill would go higher <u>and</u> the low end would go lower (since the waveform would expand symmetrically around zero). By clipping the waveform, we can increase the depth of the effect on an oscillator without throwing it out of tune with respect to the non-modulated oscillators. As the effect is increased, the high note will increase in pitch, but the base note will stay the same. Actually, those of you familiar with diodes know that the baseline will be about .7 volts below ground, however the heavy attenuation of the VCO modulation inputs will make this value negligible. I recommend that you attach a 1N914, 1N4148, or any common signal diode with its unbanded end connected directly to the aforementioned end of the 33K resistor, and then run wires from the banded end to whatever locations you want -- modulation inputs for the filter, gate inputs for the envelope generators or, as on mine, to the external modulation inputs of the VCOs (connector pin A1 for VCO1, and pin E1 for VCO2).

If you elect to run the wires to the VCOs, you will be able to get square wave modulation by switching the Modulation Select switch to "EXT" and adjusting the depth for whatever trilling interval you desire. Note that in this configuration, the depth control allows maximum trilling range of less than an octave. This may be enough for you but, if not, you will need to change the input resistor on the Modulation Depth control. These are the 22K resistors (color code red-red-orange) found to the right of each MOD pot on the SEM front panel circuit board. Alternatively, you could change the value of the input summing resistor from the MOD pots to the VCO voltage summation bus. This would be the 1 Meg resistor (brown-black-green) found at the top of board "A", near the VCOs, connected to board-to-board connector pin N1 for VCO1, and pin O8 for VCO2.

You should realize that performing this modification will no doubt void your warranty (I never had a warranty as I didn't purchase the SEMs with the intention of putting them in an Oberheim polyphonic synthesizer -- they are the voices for my homebrew 8700 controlled poly system), however those of you reading this magazine probably are more interested in expanding the capabilities of your synthesizer than in your warranty. If you've never looked inside a SEM, you don't know what you are missing. It is positively immaculate inside; it is unquestionably the cleanest synthesizer I have ever worked on, with a beautifully compact parts layout, and no point-to-point wiring. The entire synthesizer is housed on a 7" X 9" card! Other modifications I have planned include variable waveshaping and a full patch panel.

If you have another brand of synthesizer, try to get the schematics and look them over carefully. Who knows what 'gold' lies hidden 'neath the front panel?

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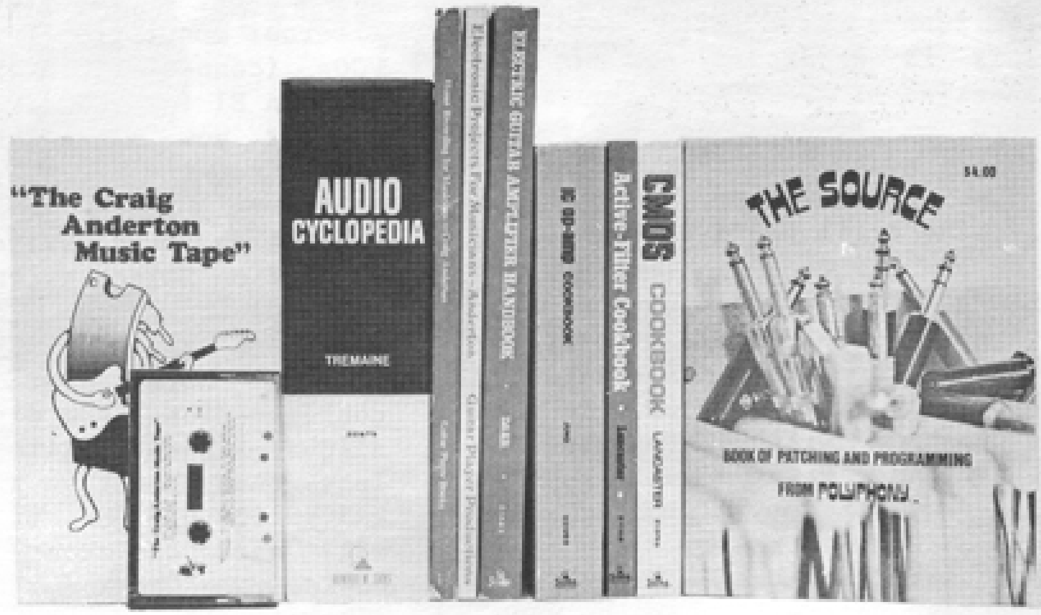

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Craig has prepared two fine books to serve as continuous reference manuals in building, and using musical electronics. Electronic Projects for Musicians is a perfect introductory manual for the musician with no previous experience in electronics. Home 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.

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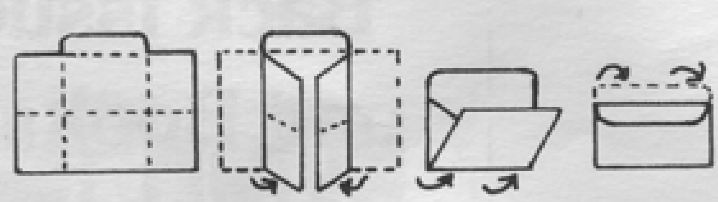

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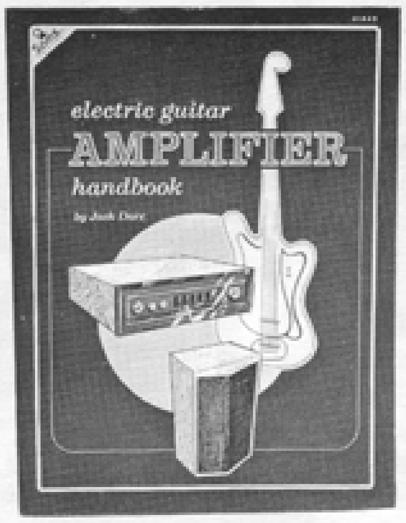

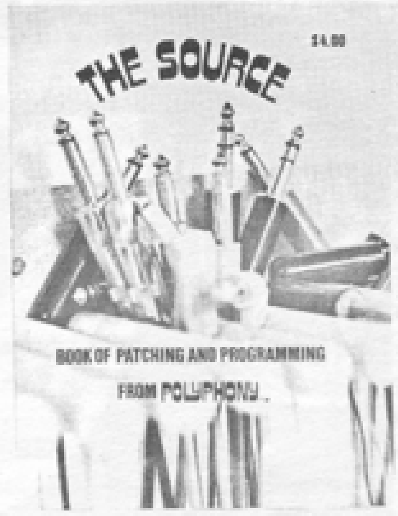

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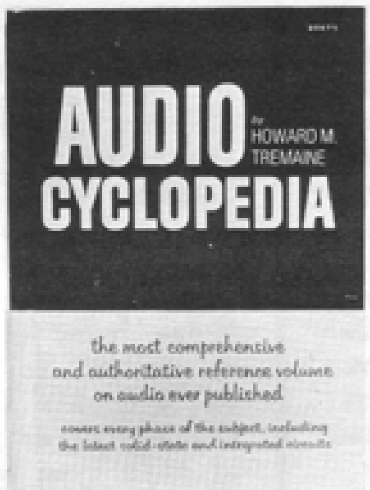

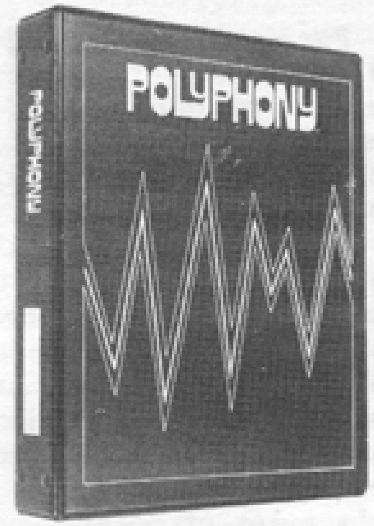

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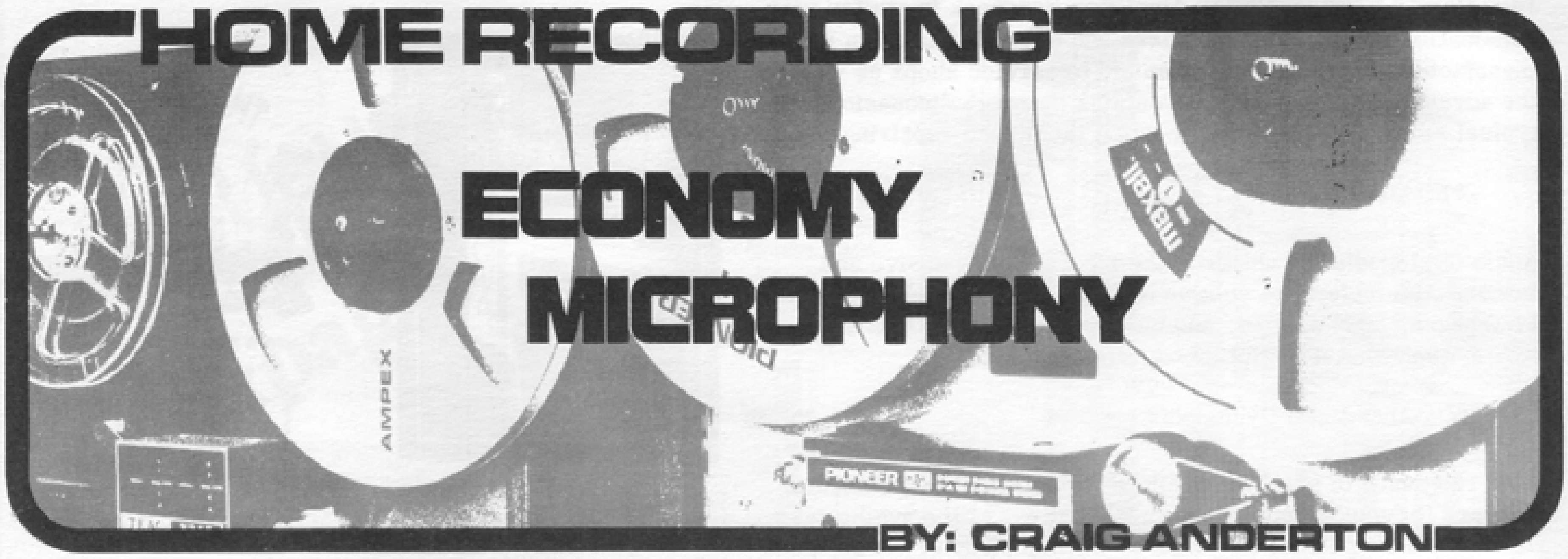


ECONOMY MICROPHONY

BY: CRAIG ANDERTON

For those on a budget, it's hard enough to get enough bucks together to build a decent home recording studio. But after that hurdle is finally cleared, insult gets added to injury -- you find out that you need all kinds of accessories, and the cost of adding these goodies mounts up pretty fast. This is especially true of microphones; it's not uncommon to see microphones with price tags of $50, $70, and lots more. If you need 5 or 6 mics, that's a lot of money.

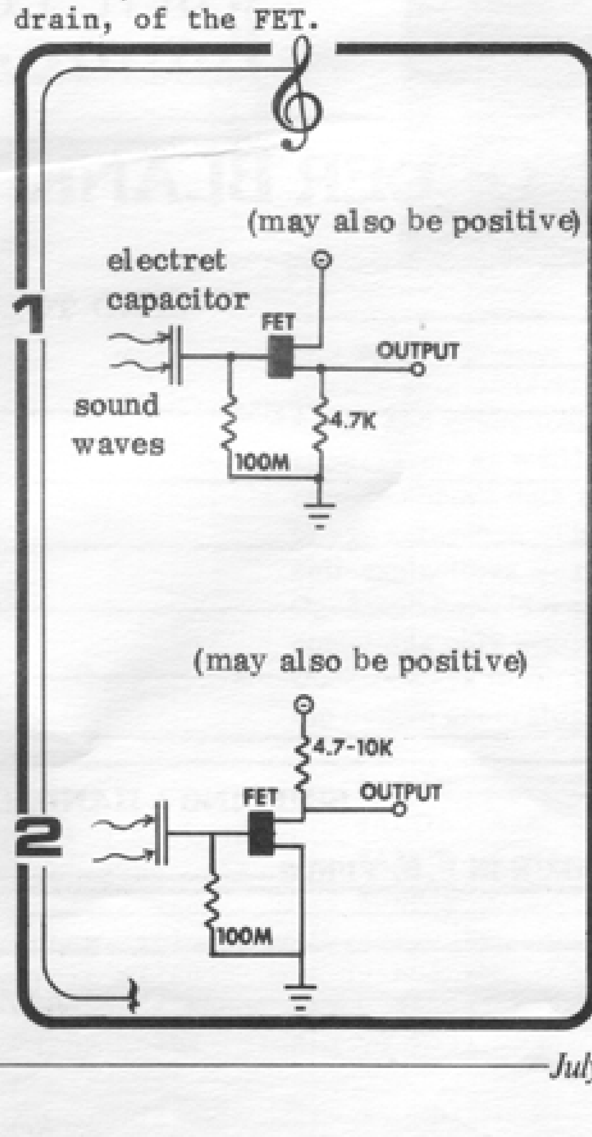
The purpose of this column is to pass along a few tips on how to build up your microphone stash <u>very</u> inexpensively. What's more, you don't have to settle for an inferior sound; we're talking about microphones with respectable specs.

the electret mic

In the last few years, electret (condenser) microphones have been perfected to the point where inexpensive electret mics offer very good performance, and even an electret costing as little as $25 or $30 can sound amazingly good. In my experiences with using a variety of studio quality condenser mics, it seemed (at least to this set of ears) that the difference between big bucks and cheap electret mics was far, far smaller than the difference between a good dynamic and a cheap dynamic. After a while, I started getting the feeling that it was almost impossible for a company NOT to turn out a decent electret microphone.

The basic schematic of an electret microphone element looks like figure 1. The electret element acts like a capacitor; sound waves striking the diaphragm cause minute voltage changes at the gate of the FET. These tiny signals are amplified by the FET to bring the sound up to a usable level. Note that the FET amplifier is an integral part of this type of mic; therefore, some type of power source is mandatory.

An alternate type of electret format is shown in figure 2. Note that in this case, the output is taken from the source, rather than from the drain, of the FET.



obtaining electret elements

Now we get to the good stuff. If you keep your eyes peeled as you read the electronics ads in the back of magazines like <u>Radio-Electronics</u>, <u>73</u>, and <u>Popular Electronics</u>, you will often see ads for "electret microphone elements" for around $3 to $8. These are, in many cases, high spec mic elements which became surplus for one reason or another. Poly Paks offers mic elements quite regularly; I also got a <u>super</u> good mic element from M + M Electronics in Seattle for only $7 (unfortunately, this item is no longer in stock -- I guess people know a good thing when they see it!).

However, obtaining a mic element doesn't mean you have a microphone .... you're only half way there, as we need to provide some kind of output connector, a power supply, and put the whole thing in a case. We also need to do some modifications not mentioned in the average data sheet.

from element to mic

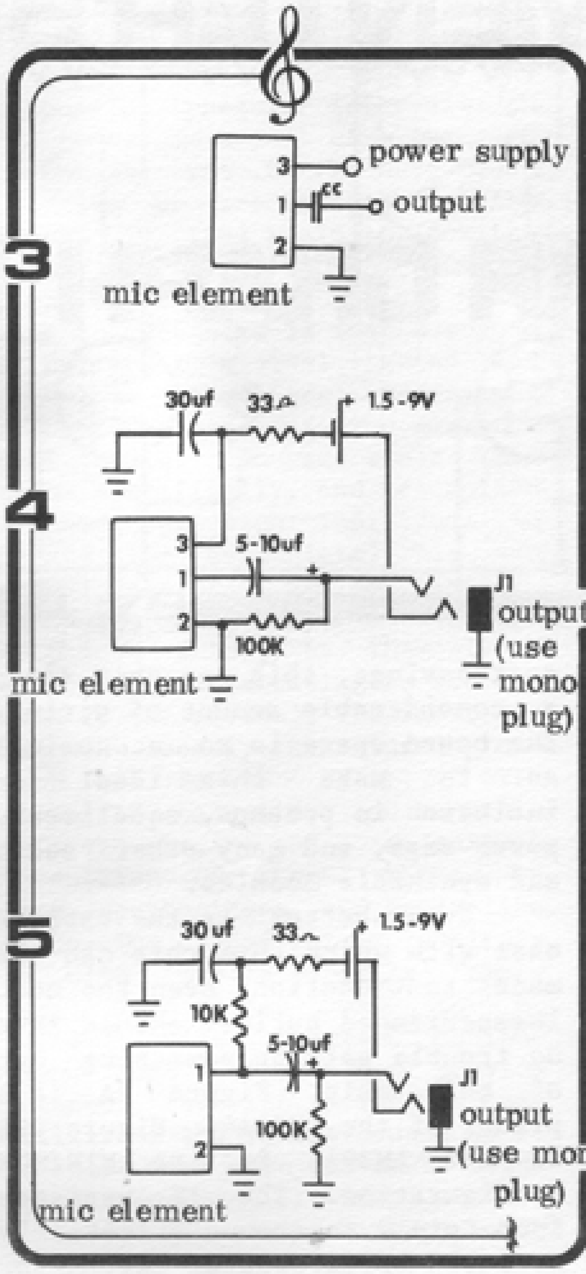
I've basically run across two electret mic types, which we will identify as "three terminal" and "two terminal" types. One example of a 3 terminal type is the Poly Paks 92CU5365 electret mic element. The data sheet identifies the three terminals, and recommends the hookup shown in figure 3. Coupling capacitor CC is required to block a DC offset coming out of the mic. However, it's not indicated that supply bypassing is desperately needed to cut out noise if you're using an AC supply, or lower the supply line impedance if you're using a battery. Also, coupling capacitor CC should have its output pulled towards ground so you don't get nasty pops when you plug the mic into a live microphone input. Finally, its nice to have some type of automatic on-off switching so you don't leave the battery on accidentally when the mic is unplugged.

The revised schematic in figure 4 shows a complete microphone based on this or similar three terminal elements. Note that J1 is a stereo jack, so plugging in a mono cord connects the + end of the battery to ground, thus completing the power supply feed to the microphone. By the way, some of the microphone elements I've seen recommend using a 1.5 V penlight battery, but I have yet to find a mic that didn't perform better with about 9 volts. Luckily I haven't blown up any mic elements, so 9 volts is probably OK.

This whole thing also has to go into some kind of box. I used a metal enclosure just big enough to hold the element, battery, and phone jack. Next I found a large grommet to fit around the mic element, and drilled a hole in the box to hold the mic-grommet combination. Voila, -- instant mic

One problem remains, although it is mercifully simple to solve. Since the electret element has no kind of protective screen or pop filter, if you attempt to use your new mic you're going to get pops on your "P" and "T" sounds like you wouldn't believe .... not to mention the blast of air that happens whenever you breathe close to the mic.

An inelegant, but practical, solution is to simply cut out a square piece of foam rubber and hold it in place on the box, directly over the mic element, with a rubber band. A somewhat more novel solution is to obtain a "Nerf" ball and use that for a screen. Cut out just enough of



the ball with an X-Acto knife or scalpel to hold the mic box in place (see photo). Of course, acoustical foam is more "correct" for the job, but if you sing close to the Nerf ball you won't notice much (if any) treble loss.

The Poly Paks 92CU3178 mic is an example of the two terminal type of element mentioned earlier. If you run into an element like this, try hooking it up as shown in figure 5. Make sure the battery polarity is correct -- usually the data provided with the mic element will provide some type of clue.

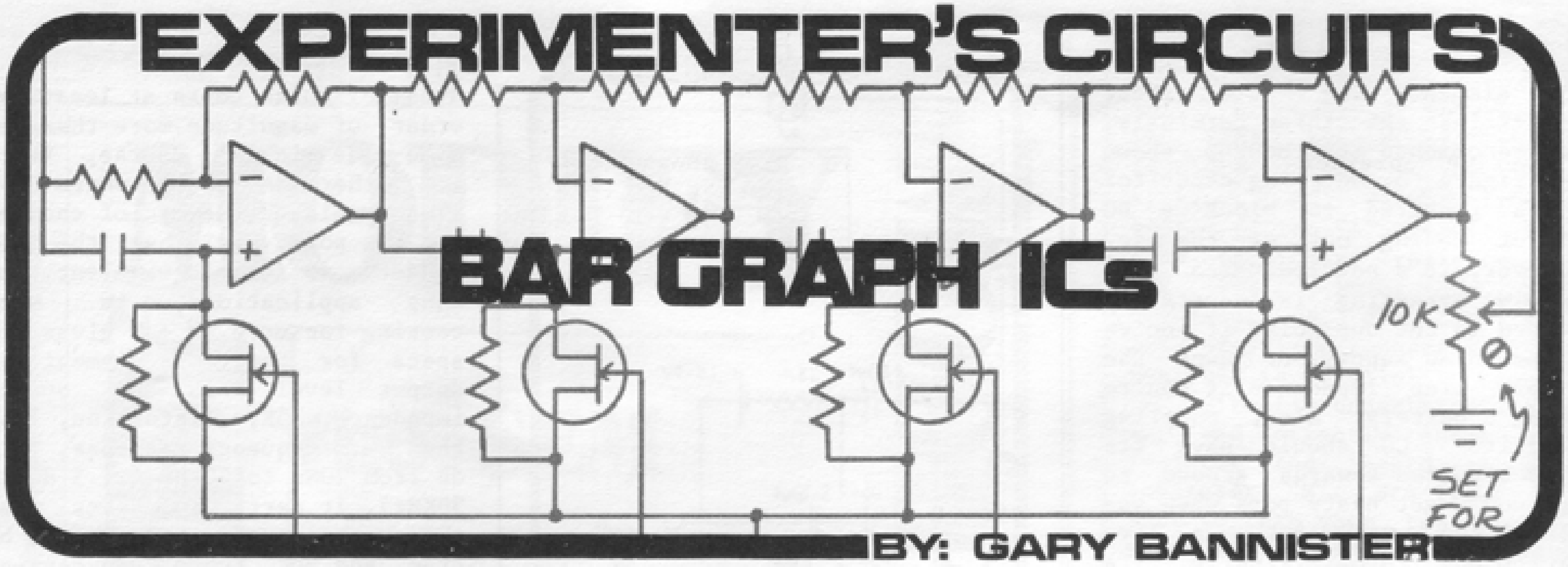
user evaluation

My favorite mic element so far is the one I got from M + M, although the Poly Paks ones are OK too, and Radio Shack sells a good electret element for about \$3. My budget mic actually gives me a better vocal sound than what I get from my Electro-Voice RE-16 dynamic, which costs at least an order of magnitude more than the home-made mic. Of course, there are other instances where the RE-16 would be the mic of choice, but the point is that the "\$7 Wonder" is fully equivalent, for many applications, to mics costing far more. M + M gives the specs for their mic element as: output level, -55 dB; output impedance, 5K; distortion, less than 2%; frequency response, ±0.5 dB from 20Hz to 20KHz (-1.5 dB at 30KHz). It certainly seems that these specs are too good to be true, and yet, there's no denying that it does have a very good sound.

conclusion

Keep your eyes open, and take the chance on a surplus mic element from time to time. You'll save a lot of bucks by building your own mic, and the performance may just surprise you.

EXPERIMENTER'S CIRCUITS



BAR GRAPH ICs

BY: GARY BANNISTER

I've always liked blinky lights. Rampant sequencers get me mildly excited, and I don's sleep much because of my binary clock. It's no wonder I think LED VUs are the best thing since sliced bread, and when I saw that solid state spectrograph at a local recording studio ....

I've got to have them all! Reasonably, of course, and that's why I don't have any. Figure 1 is a typical schematic of a voltage sensor with an LED display. Just look at all the parts! For a ten stage unit (illustrated) you need twenty resistors, ten op amps, ten LEDs, and untold support components for doing reference voltages, gain, etc. Even using quad op amps (which are not the best choice for things like this) this is a BIG project. If you want a VU type response, the resistors in the reference divider must be oddball, hand matched values, consuming even more time and money.

the LM3914

Several new chips are becoming available to the consumer which can reduce these linear display jobs to mere busywork. In fact, one is available as close as your nearest Radio Shack. There are several different units available, but I'm going to focus this article on the National Semiconductor LM3914 -- the one from Radio Shack.

What is so neat is that this one chip replaces figure 1 completely -- except the LEDs, of course. While not necessarily a cost savings, this one chip saves a considerable amount of wiring. The board space is so economical as to make this ideal for inclusion in preamps, equalizers, power amps, and many other audio and synthesis modules.

Even better is the extreme ease with which this chip can be made to function. Even the most inexperienced builder should have no trouble getting something out of this chip. Figure 2A is a pinout of the LM3914. Figure 2B is the LM3914 in its MINIMUM configuration. The 1K resistor from pin 7 to ground effectively sets the brightness of the LEDs. 1K is a practical value (minimum), and may be made greater for a dimmer display.

Switch 1 may simply be a jumper from pin 9 to either of the two points. It selects either a BAR MODE or a DOT MODE, which we'll discuss later. The other two resistors form a divider to produce a + reference voltage. In special cases, the + reference voltage can come from pin 7, so the two resistors may not be necessary.

What this means is, with the + reference voltage coming from pin 7, and the MODE (pin 9) unswitched, this project needs 1 IC, 10 LEDs, and <u>only one other part</u>! How simple can you get?

Actually, this is only good for a brief turn-on. I found it boring after a couple days. Let's discuss it further and try to get

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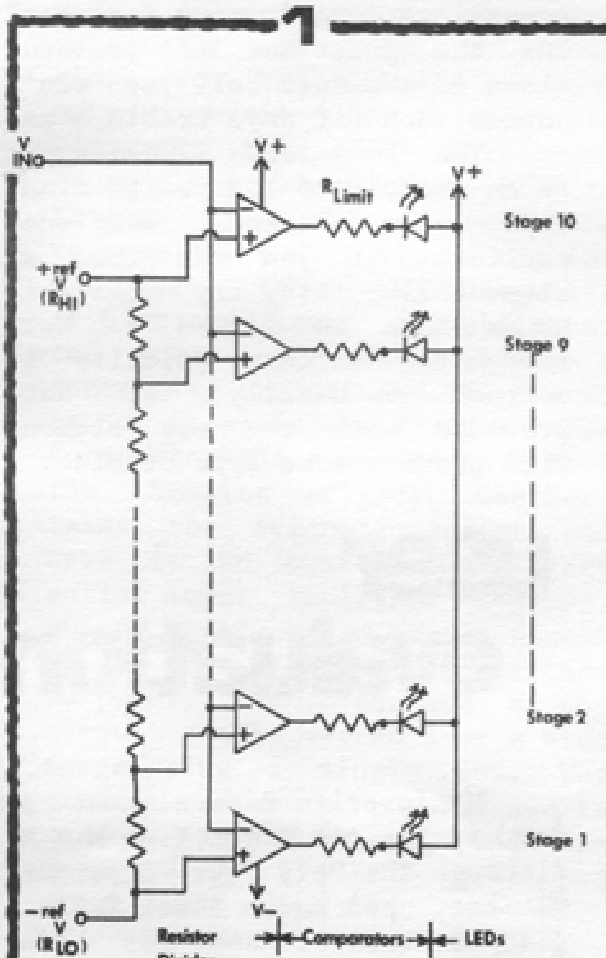

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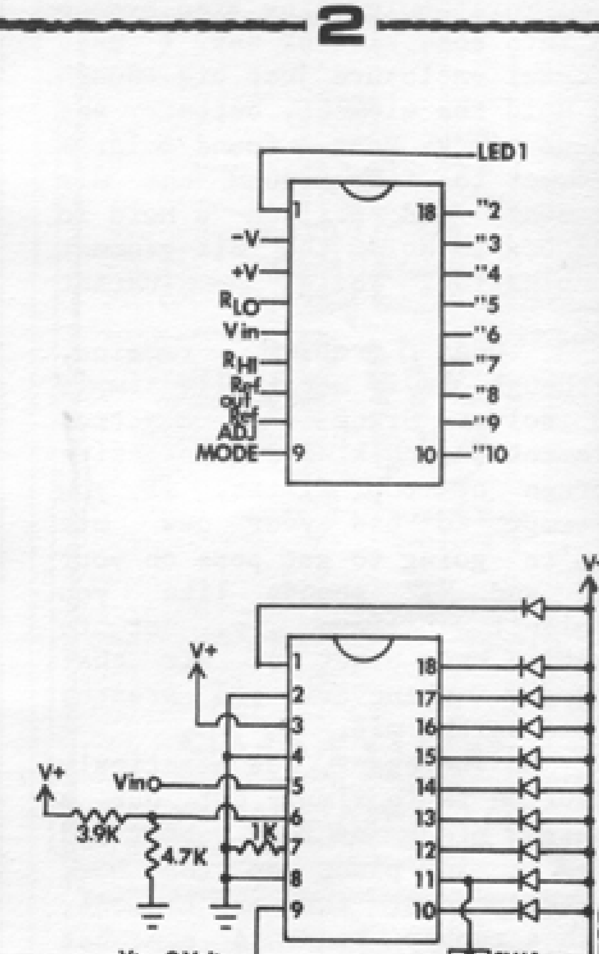


something useful out of this chip.

Refering back to figure 1, we find that this IC contains a resistive divider chain, 10 comparators, and the R_{limit} resistors for the LEDs. Actually, I don't think the R_{limits} are really there. The 1K resistor from pin 7 "programs" the current in the LEDs, eliminating the need for R_{limit}. This is good, as we'll do some tricks with the outputs later.

The resistive divider chain is LINEAR; that is, all the resistors are equal in value, and matched as well as IC technology will allow. The spec sheet claims better than 2% accuracy, but I measured better than 1% on the chips I have. Unfortunately, all resistors ARE equal in value, meaning that this chip is initially unsuited for use as an LED VU meter, but we'll try to get around this later.

The + reference voltage and the - reference voltage bear discussion. The + reference is easy to understand. It is a positive voltage applied to pin 6. It sets the range over which the input voltages (pin 5) are compared, and the resolution, or how small a voltage can be accurately compared.

EXAMPLE: (see figure 1) The - reference is usually ground. Let's apply 5 volts to the + reference -- pin 6. This means that the RANGE of input voltages can be anywhere from 0 to 5 volts. The divider string gives us ten equal increments of .5 volts each. The resolution is .5 volts, or voltage changes no smaller than .5 volt can be accurately detected. If we change the + reference to 2 volts, the total range is 2 volts and the resolution is .2 volt.

One of the hidden features of the LM3914 is the - reference. It is usually listed as ground, but can actually be anything from ground to + reference. Let's reuse the previous example, but now make the - reference equal to 4 volts. Now the range is from 4 to 5 volts, and the resolution is .1 volts (5 volts minus 4 volts, divided by ten steps).

One last point about the divider string. It is almost completely independent of the supply voltages to the chip (pins 2 and 3). There are only two requirements -- + reference must not be MORE than V+, and the - reference must not be LESS than V-. This is going to come in handy later.

There are a few requirements about the comparator section. The most important is that the total supply voltage must not exceed 25 volts. That is, the potential between pins 2 and 3 must not exceed 25 volts. Nothing is said about running the chip on bipolar supplies, but it does work. In this form, you can go no more than ±12 volts. For many synthesizers and audio devices using supplies of ±15 volts a dropping circuit of resistors and zeners will need to be used (we'll see how later).

Since the R_{limits} don't seem to be there, all we'll say is that we don't have to worry about them when we interface to other chips.

applications

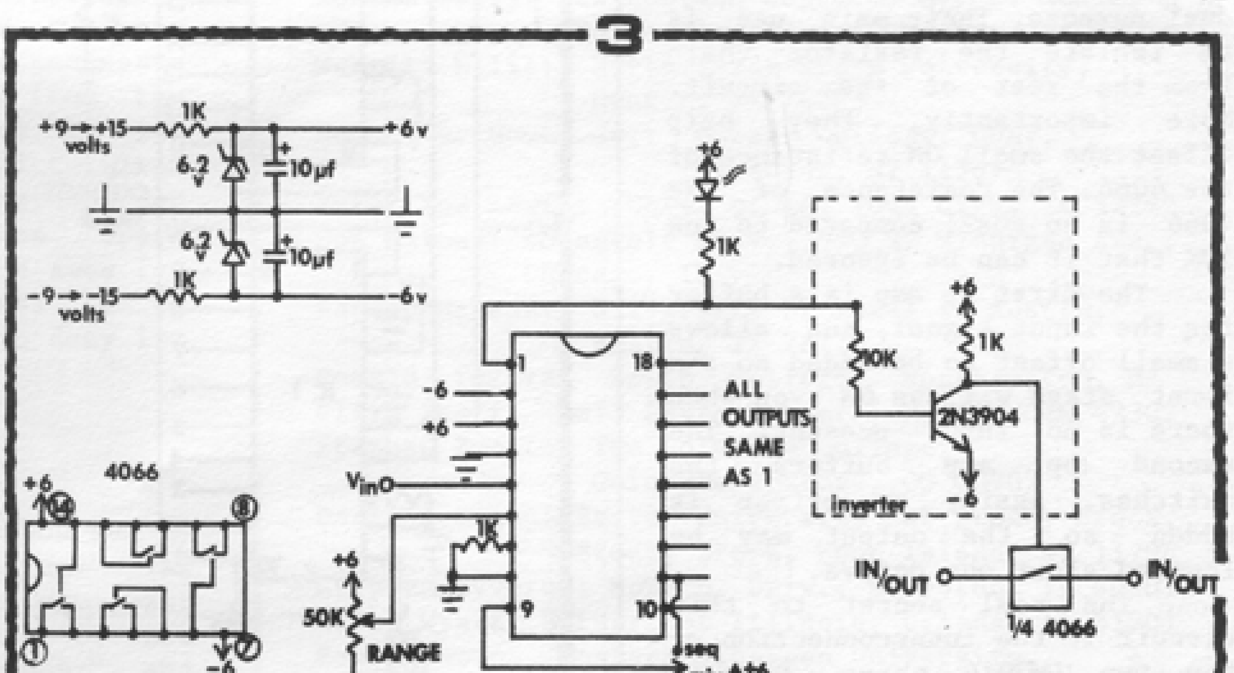
Now that you have all the information, let's do something. One idea I've liked is a level sensing switch. Similar in action to an ARP or MOOG sequential switch, this unit responds to voltages rather than triggers. The voltage can come from an LFO, ADSR, Envelope Follower, or any control source. As the voltage increases, successive switches are turned on to pass controls, triggers, or audio. With the LM3914, one can select a MIX or SEQUENCE mode. In the MIX mode, a switch stays on until the control voltage drops. In the SEQUENCE mode, only one switch at a time can be enabled.

Figure 3 is such a unit. First note the zeners necessary in the power supply. Actually, this is more for the CMOS 4066s than for the LM3914. The 4066s cannot have more than ±7.5 volts applied. Allowing a safety factor, ±6 volts should work well. Note the LED arrangement. We said that R_{limit} was not necessary. Actually, the 1K resistor is there to make the output more compatible with the rest of the circuit. The transistor circuit is a simple inverter. Without it the switch would be ON when it should be OFF.

During construction, be sure to connect the positive and negative supplies to the 4066s correctly. In addition, when you wire the front panel, make sure that you don't hook up the output of one switch to the input of another. A pinout of the 4066 is provided to guide you. Just to avoid confusion, the schematic shows only ONE such switch. There is, of course, one switch for each output -- ten in all.

There is one requirement for this circuit. The input to the switches cannot be greater than ±6 volts. This is usually safe for audio level signals and 0 to 5 volt control signals, but the square and saw waveforms of some synthesizers (like Aries and Arp) may need biasing to swing plus-and-minus. This is usually described in the owners manuals. With proper biasing, this circuit can easily handle signals up to 10 volts peak-to-peak (±5 volts).

Also note that the switch inputs and outputs can be interchanged. The switches are truly bi-directional. The RANGE control allows the number of



switches to be matched to any input -- especially useful with Envelope Followers. The MIX / SEQ switch has already been discussed.

Quite frankly, I've found little use for this circuit. At least, as it sets. Let's develop it a step further. Suppose we add another resistive divider string that contains 19 1K 1% resistors. The bottom of the string goes to ground, the top connects to a simple constant current source. At each resistor junction let's add one of the level sensing switches we discussed earlier. We have twenty junctions (counting both ends), so we'll need to series connect two LM3914s (easy with this chip). Now, set the LM3914s for the DOT mode, and sum the output of the switches.

For those of you who have trouble making schematics out of words, figure 4 is such an animal. Some of you may recognize the resistor string as the beginnings of a keyboard. If we feed this circuit with the output of a sequencer, the output will not be a smooth transition during tuning, but will move in a step-wise fashion. With proper adjustment, the divider will produce proper 1 volt/ octave (or whatever LINEAR response your synthesizer needs) outputs. What we have here is a 'quantizer' that will work with most ARP, MOOG, ARIES, etc.

There is a little more here than meets the eye, so we'll discuss it section by section. The resistor divider is so simple as to not need discussion. Just be sure to use 1% resistors to avoid tuning problems. More expensive, but necessary.

The 10K 1% resistors serve a dual purpose. Their main use is to isolate the resistor chain from the rest of the circuit. More importantly, they help offset the small ON resistance of the 4066. The resistance of the 4066 is so small compared to the 10K that it can be ignored.

The first op amp is a buffer for the input signal, and allows a small offset to be added so the first stage will be ON even when there is no input present. The second op amp buffers the switches. Again, an offset is added so the output may be lowered about one octave.

The real secret to this circuit is the interconnection of the two LM3914 chips. Here is where the independence of the resistive divider comes in handy. As I said, the - reference does not HAVE to be grounded. To stack these chips, all that is necessary is to connect the - reference of one chip to the + reference of the next lower chip. Also make a special note of how the MODE pins are connected. The MODE of the lower chip is connected to the first output of the next higher chip. The highest chip is connected normally. If you follow this, you can easily expand the string to 10 chips -- 100 LEDs!!

Note that we don't use any LEDs. We don't need them in this application, so they are replaced with suitable resistors. The 4049 chip replaces the transistor in figure 3. With proper connection of the supply pins, the 4049 could be used in figure 3 as well.

calibration

Calibration is easy. Connect the output of the sequencer to the IN jack, and temporarily add the 'Calibration Only' jumper on the RANGE pot. Manually advance the sequencer to the first stage and attach your voltmeter to test point 1 (TP1). Adjust the

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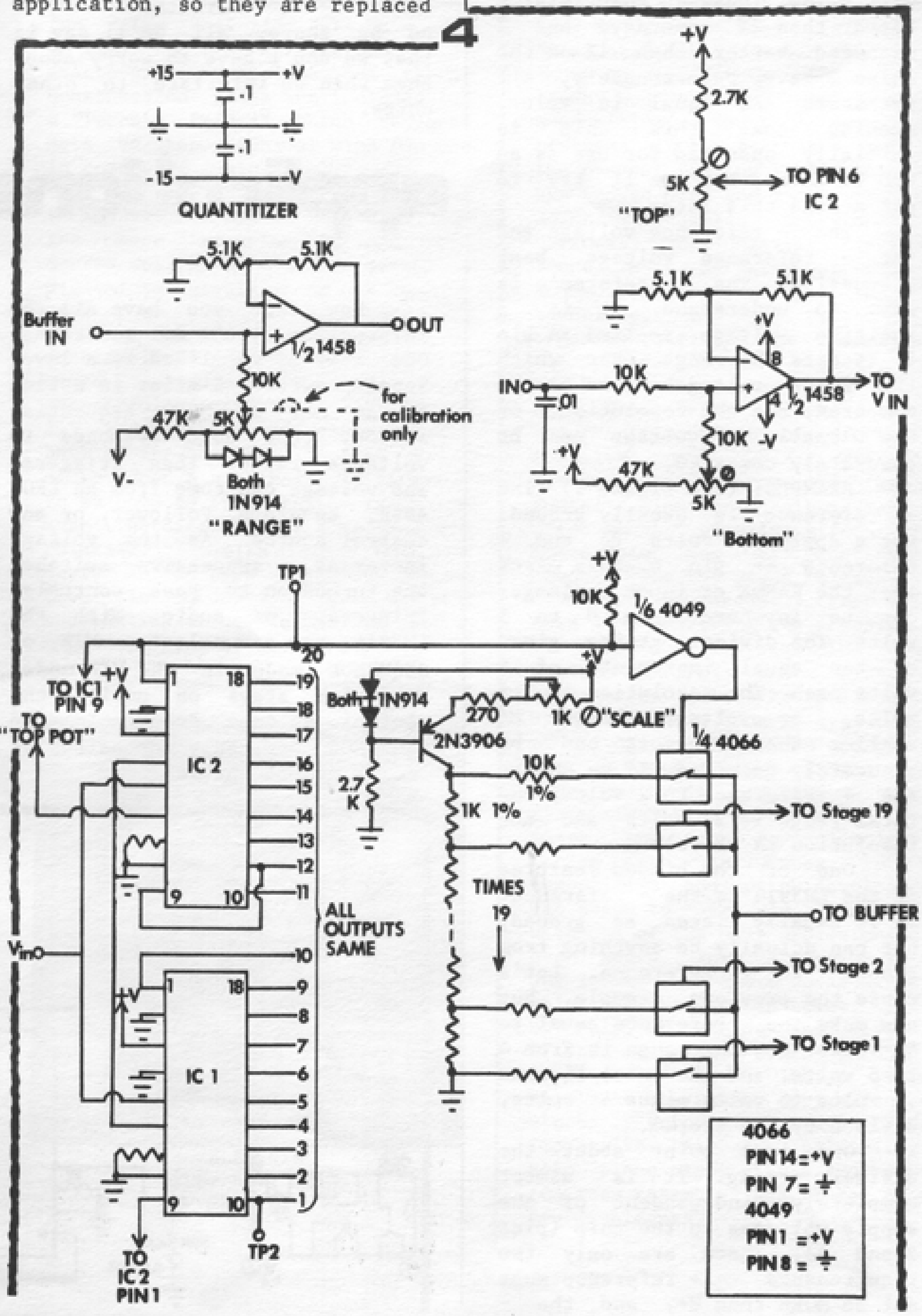


sequencer control to minimum, and verify that TP1 reads almost full +V. Turn the sequencer control to maximum, and adjust the TOP trimmer until the voltmeter suddenly reads very close to zero. Turn the sequencer control back and forth, and confirm that the voltmeter reads zero only when the sequencer tuning is at maximum.

Attach the voltmeter to TP2. Adjust the sequencer control to minimum, and adjust the BOTTOM control until the voltmeter just switches to zero. Play with the BOTTOM control until you find the point at which the voltmeter just switches. Now play with the sequencer tuning control and confirm that when the control is advanced only a small way the voltage at TP2 switches to almost +V.

Attach an (exponential response) oscillator to the OUTPUT jack of the 'quantizer'. Tune the sequencer control to minimum, and tune the oscillator by hand to some reference tone. Advance the sequencer control to maximum and adjust the SCALE control until the oscillator is exactly an octave and a fifth above the reference. Repeat this procedure several times to minimize errors. Slowly turn the sequencer control and confirm that the output moves in a stepwise rather than smooth fashion. The range should still be 1.5 octaves. Remove the CALIBRATION ONLY jumper from the RANGE control and verify that it lowers the pitch somewhat more than an octave.

The quantizer is now functional. Any tuning errors can almost always be traced to bad matches in the 1K 1% resistor divider string. In rare cases a 4066 might be bad. Note that the quantizer will do its thing on any voltage. Sample/Hold circuits and LFOs make good possibilities. In use, the RANGE control can be used to center the output around some 'home' pitch, so that sequences may vary up and down from this pitch.

While writing the final draft of this column, I started studying the LM3915 which is functionally similar to the LM3914 except the divider string is logarithmic (as opposed to linear). This is truly the 'one chip' VU meter, and should provide some interesting projects in future columns.

Construct

A HEX VCA MODULE

The main disadvantage to most polyphonic synthesizer systems is that you need a VCO and VCA for every voice, and when you want lots of voices this means lots of money and lots of space. Here is a circuit which will go a long way towards alleviating the VCA problem in terms of both cost and space.

The HEX VCA circuit contains six VCAs and fits behind about 4" X 4" worth of panel space. Parts cost for this circuit should be under $30, which is a lot less than six separate VCAs. The only 'disadvantage' to this circuit is that the six individual outputs are mixed to one master output. In most applications, this is what you would be doing anyway, but for complex multi- voicing or applications requiring a stereo mix this could be a problem. However, this feature does leave the mixer free to perform other duties and you will most likely already have two or three VCAs to do the other special voices.

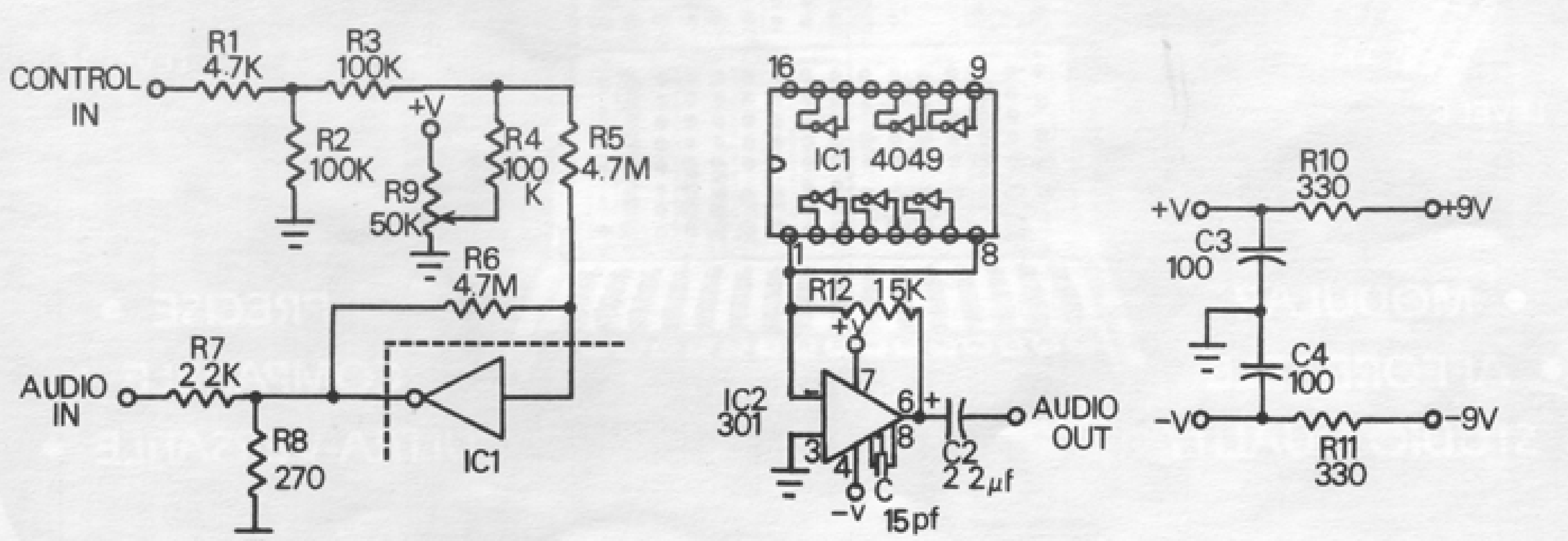
CIRCUIT DETAILS

As you've probably read in many places, an N-channel FET with some drain to gate feedback resistance will make a good voltage controlled resistor with fairly linear characteristics. The heart of the HEX VCA circuit, the 4049 CMOS hex inverter, contains six N-channel MOS transistors which are used for just this purpose.

The circuit diagrams appear in Figure 1. Only one of the six VCAs is shown, as all of them are identical. The voltage divider formed by resistors R7 and R8 attenuates the audio input signal to a level acceptable by the MOS transistors and also provides an acceptable input impedance for the transistor. Resistor R6 is the drain to gate feedback resistor and R5 is the gate resistor. Resistors R4 and R9 serve to bias the transistor just below the linear control voltage range and resistors R1, R2, and R3 attenuate the input control voltage so the control voltage for the transistor does not go above the linear operating range. The outputs of the six transistors are summed together and amplified by op-amp IC2. The output of the circuit is capacitively coupled so as to be compatible with any type of following circuitry.

Care must be taken when selecting the 4049 IC, as not all will work. This is most likely due to variations on internal diode arrangements by different manufacturers. The 4049s by RCA and Motorola will work, and I have found that the ones sold by Radio Shack will also work (they appear to be manufactured by Motorola, but Radio Shack's supplier for this part may change periodically). The drain to gate feedback resistor may be anything from 3.3 Meg to 5.1 Meg without affecting the operation of the circuit, but R5 and R6 should be the same value.

FIG.1



Several modifications can be made for special applications. If master control of the output level of the HEX VCA is desired, replacing R12 with a 1.5 K pot in series with a 500 ohm resistor should provide ample control. Similarly, if throughput for one channel with 5 volts control applied is less than unity gain, the value of R12 can be increased to raise the output signal to the desired level. Also note that the audio inputs offer a fairly low input impedance which may load some sources. In certain applications, it may be desirable to add a high gain transistor (2N3391 or similar) as an emitter follower buffer for each audio input. Individual level controls can be added for each input if you wish the unit to be configured more like a VC-Mixer. As usual, the possibilities are numerous.

CONSTRUCTION & CALIBRATION

Figure 2 shows one possible PC board layout for this circuit. Control voltage inputs are labelled "C" followed by a small letter subscript to designate which of the six VCAs (lettered 'a' through 'f') the input will control. Audio inputs are labelled "A" and use identical subscripts. The board is designed to fit easily behind a panel which is at least 4 inches wide or tall. Figure 3 shows the parts placement from the top of the board. The parts for each channel are designated by a lettered subscript corresponding to that channel. The basic set of nine resistors is repeated six times, each set with a different subscript character. Parts common to all channels carry no subscripts. Fixed resistors should be soldered in place first. There are seven jumper wires indicated on the parts placement by straight lines which can be installed using excess resistor lead clippings. Following this, install the four capacitors, six trimmer pots, and finally the ICs. I strongly recommend sockets for both ICs, as this will reduce the possibility of damage and allow easy replacement if the IC you have selected does not operate properly for this circuit, or if it becomes damaged in the future.

Calibration is simple but it _must_ be done with the _envelope generator connected to the control input_. Start by rotating all the trimmers towards the rear of the board. Apply power to the circuit and connect the control input of channel A to an envelope generator. Connect the A audio input to the output of a VCO, and the output of the HEX VCA to an amplifier. Apply a control voltage to the VCO, but do not trigger the envelope generator. Advance the trimmer for channel A towards the front of the board until a tone is heard from the amplifier. For this test it may be helpful to turn up the volume on your external amp. Back off the trimmer slightly until the tone just disappears. Channel A is now calibrated. Remember to decrease the volume on your external amp before triggering the envelope generator or using the HEX VCA. Repeat this procedure for the remaining five channels and the HEX VCA is ready to be put to use.

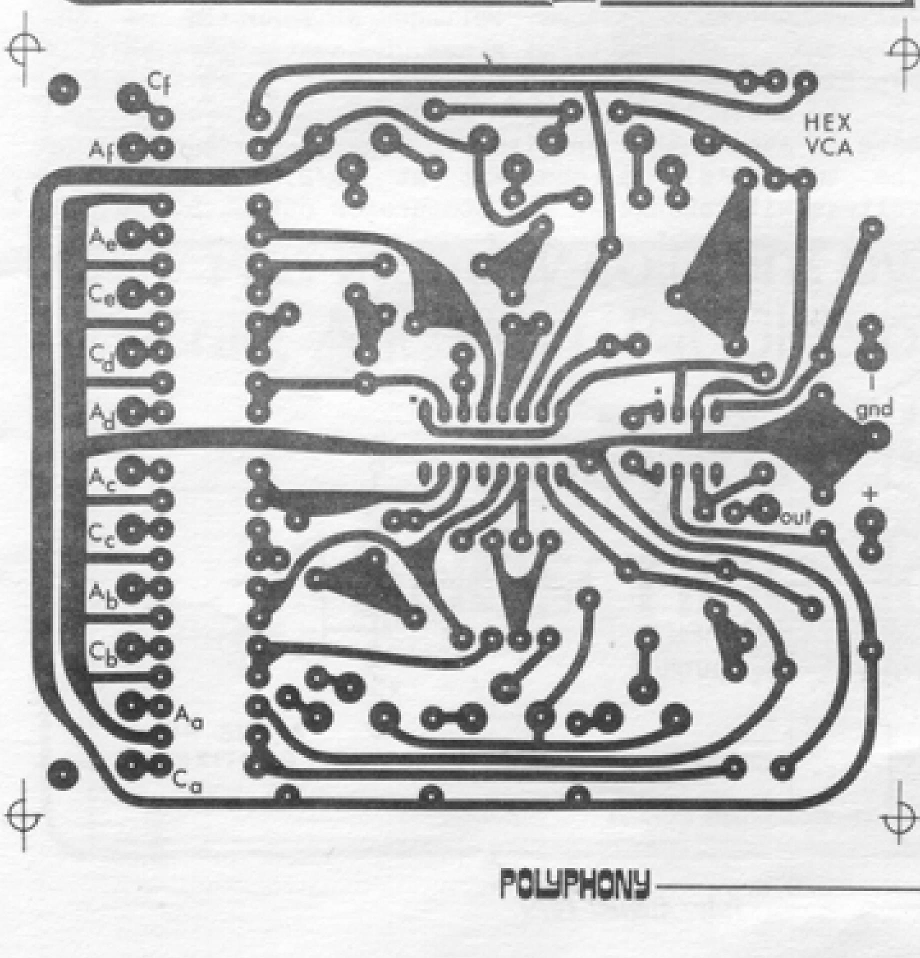
PARTS REQUIRED

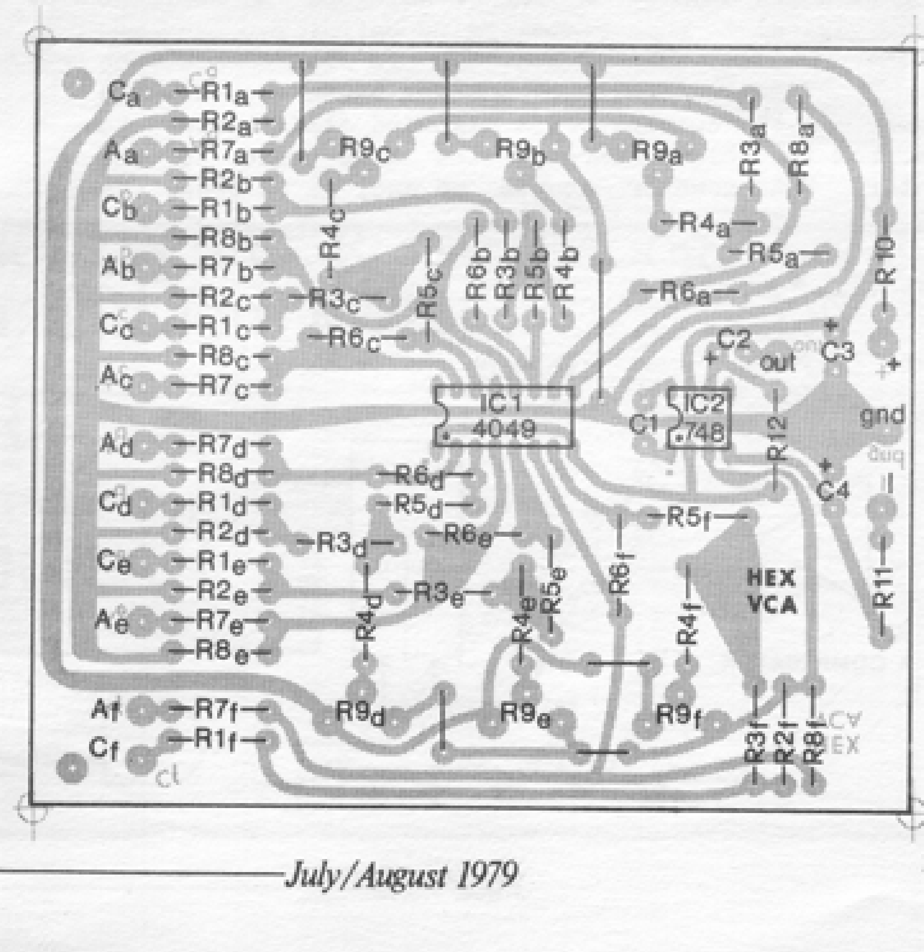
1- 4049 hex inverter (see text)
1- 301 or 748 type op amp
18- 100K (1/4 watt, 5%)
12- 4.7M (or 3.3M to 5.1M)
6- 4.7K
6- 2.2K
6- 270 ohm
2- 330 ohm
1- 1.5K (fixed or pot, see text)
6- 50K trimmer
1- 15 pf ceramic capacitor
1- 2.2 uf, 10 V electrolytic
2- 100 uf, 15 V electrolytic
1- 16 pin IC socket
1- 8 pin IC socket
6- control connectors
7- audio jacks
Front panel, PC Board, wire, etc.

**We have arranged for a board and kit to be offered by Paia Electronics, Box 14359, Okla. City, OK 73114.

PC Board (etched, drilled, and screened) -- #EK8PC, $8.95 ppd.
Kit with board and on-board parts _only_ -- #EK8, $15.95 ppd.

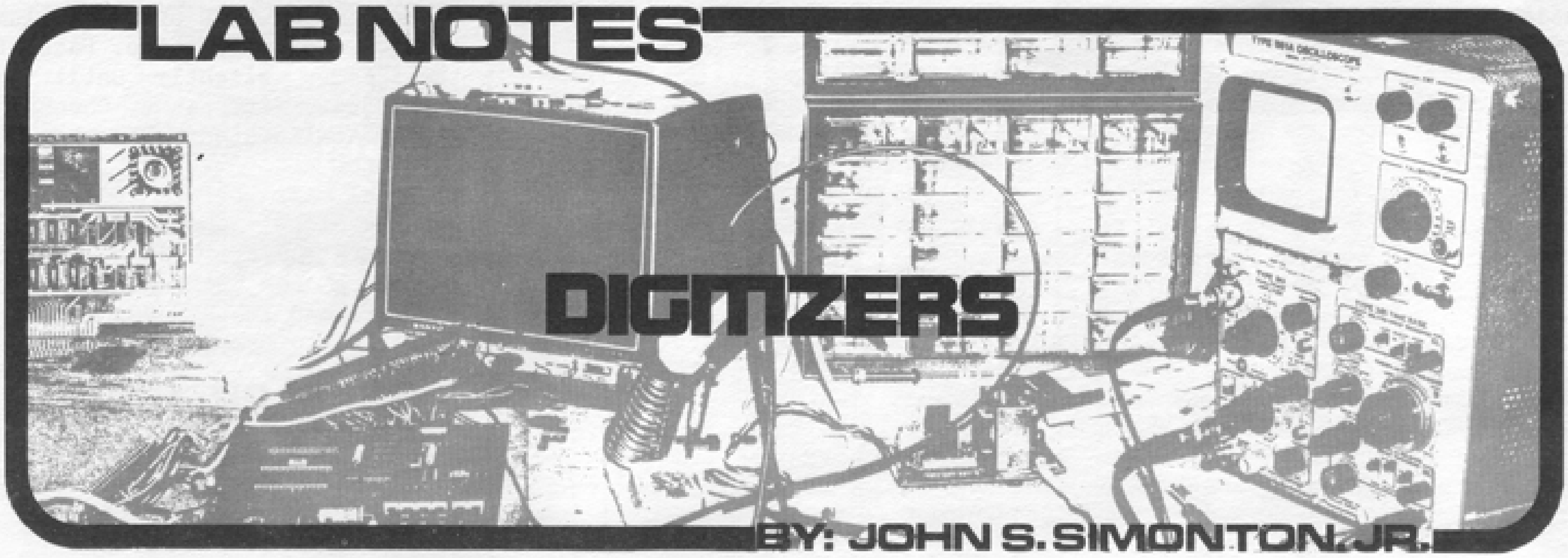
2 FIG. 3





LAB NOTES

DIGITIZERS



BY: JOHN S. SIMONTON, JR.

There are plenty of times when a switch is a great way to control things- like when you want to turn something on and off, or select a preset. But when you're just playing around looking for the right sound, there's nothing quite like a knob. Unless it's a joystick.

Knob or joystick, either one- we need some way to digitize it's position so a computer can read, save and manipulate the data various ways. And preferably it should be a cheap and simple way.

We need something we'll call a digitizer. It's an analog-to-digital converter, really; the only reason I don't think we should call it an ADC is that we reserve that term for something more elaborate than what we are getting into. This is really simple.

In every electronic scheme that I know of to convert an analog parameter to a digital one there is a thing called a comparator. See figure 1. The thing it compares are the voltages at its "+" and "-" inputs. If the voltage at the "+" input is greater, the output is at a high voltage. If the "-" input is greater, the output is driven to a low voltage.

The elaborate ADC's use the comparator as only a small part of a larger circuit that will probably look something like figure 2. When it's time to quantize the voltage to be measured, the counter is reset and its digital output goes to zero. Because of this, the D/A puts out a low voltage (in this scheme you must first have a digital to analog conversion before you can have the reverse). The output of the D/A will probably be lower than the voltage that is being measured, so the output of the comparator is high and allows pulses to pass from the clock through the NAND gate to the counter. The counter counts up and, as it does, the output of the D/A increases. When the output of the D/A exceeds the voltage to be measured, the comparators output goes low and clock pulses can no longer pass through the gate to the counter. At that point, the counter's output is a digital representation of the analog voltage being measured.

There are a number of variations on this design that have to do with the way the counter works, and in a computer based system it is common to replace both the clock and counter with software. Unfortunately, the common features of all these variations are modest complexity and/or relatively slow conversion rate.

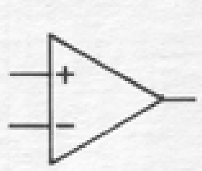
härdwara

Now, for a really simple digitizer, take a look at figure 3. Since the resistors in the divider that detirmines the reference voltage (Vr) are equal, the digital output is a 1 (high) if the voltage is greater than 1/2 the supply voltage and 0 if the input is less then Vs/2. I know what you're thinking, and you're right. A one bit digitizer isn't exactly an improvement over a switch in most cases.

OK, let's add another stage. Only on this one, let's make the reference voltage a function of the output state of the first stage. Schematically, this is represented in figure 4.

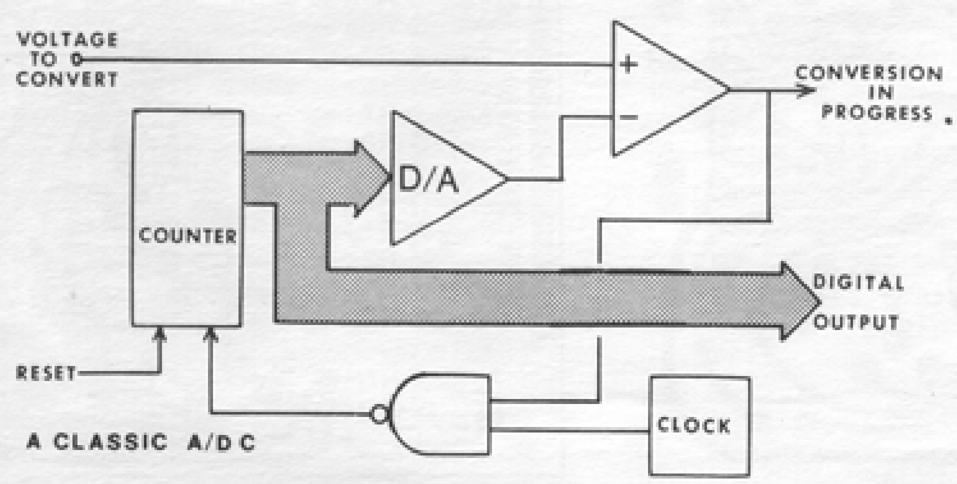
In order to easily see how this circuit works, you have to assume that Vr1 (the voltage at the junction of the two R1's) is constant at Vs/2. In fact, this voltage will change as the comparator output D1 changes

1



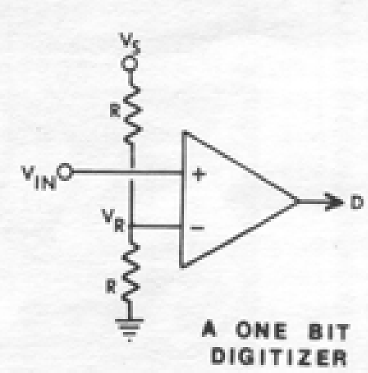
A COMPARATOR

2



A CLASSIC A/D C

3



A ONE BIT DIGITIZER

and alternately sinks or sources current through the two resistors, R2. But as long as the value R1 is kept much lower than the value R2 (the lower the better, at least 1/10), the change in Vr1 will not be too significant.

Imagine that a voltage which is increasing from ground to supply is applied to the input of the digitizer. When at ground, the voltage is less than Vs/2, so D1 is low (ground). The two R2's now form a voltage divider at the junction of which is a voltage equal to 1/2 of Vs/2, or 1/4 of the supply voltage (Vs/4). This voltage (Vs/4) is the reference voltage for the new stage. Since we said that our input voltage was initially at ground (which is less than Vs/4), the output of the new stage is also low. In binary, the output of the two stages is 00. An equivalent circuit would look like figure 5.

Now we increase the input voltage and, as it exceeds Vs/4, the output of the new stage changes from low to high. That's all that happens; the binary output of the two stages is now 01.

We continue to increase the input voltage and, as it exceeds Vs/2, the output of the first stage goes high. But, that's not all, because with the output high (at Vs), an equivalent circuit of the voltage divider that forms the reference for the new stage looks like figure 6. Since the input voltage is less than 3/4 of the supply voltage, the new stage changes state back to low and all is once again stable with a binary output of 10.

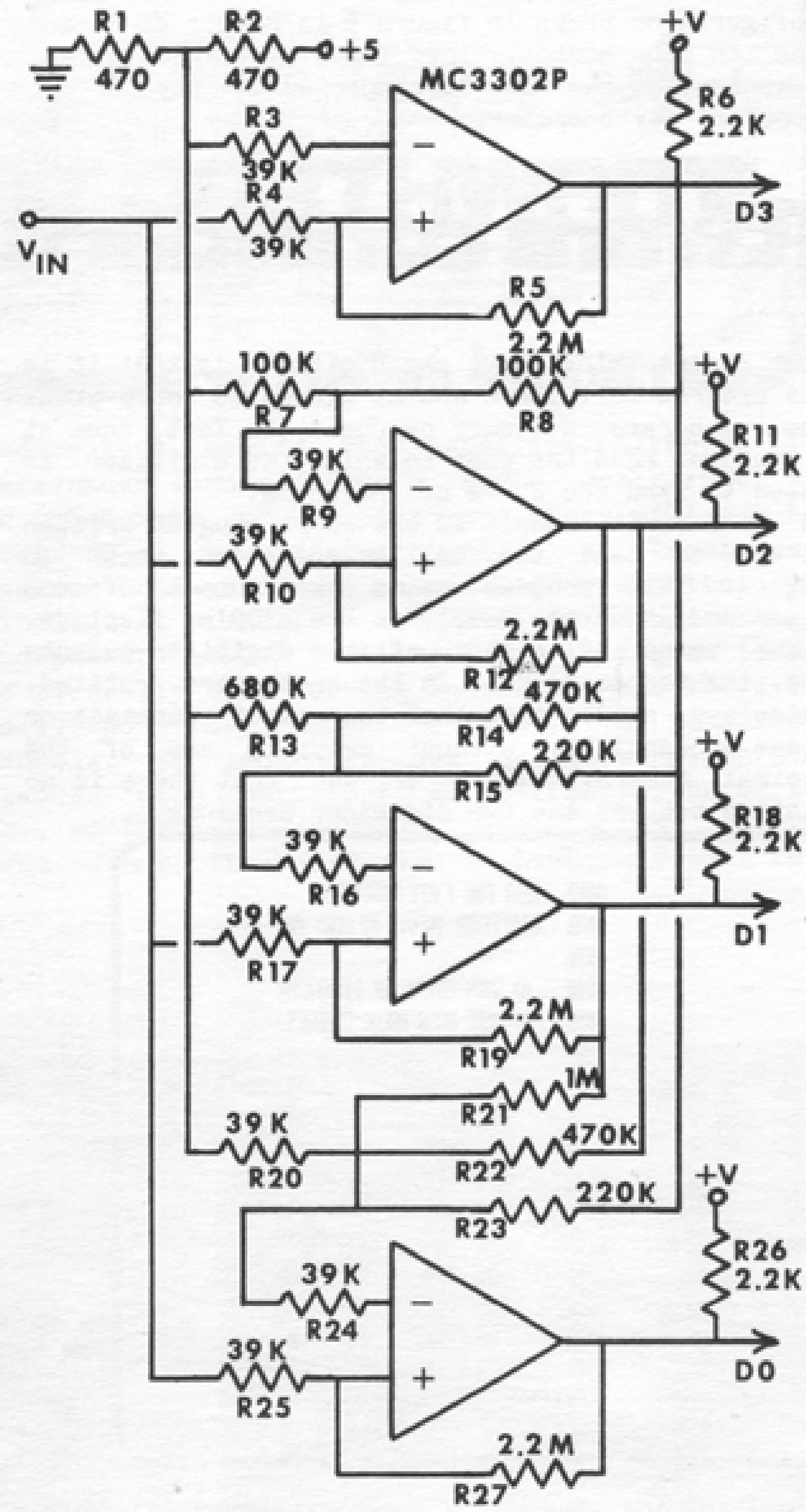
Increasing the input voltage further will exceed 3Vs/4. The new stage again changes to a high state and the binary output of the two stages reads 11.

Additional stages can be added in much the same way we just added the second stage. Each new stage becomes the least significant bit of the digitizer and its reference voltage is a weighted sum of the outputs of the more significant stages. Using 5% resistors, the scheme can be carried to 5 bits. 1%'ers would probably take us up to 6 bit resolution; 7 or 8 bit resolution should be realizable by going to active summing amps instead of the passive summing we've used. But, then you're back to complicated again.

Instead, we'll stop at an easily obtainable 4 bits with the design shown in figure 7. Since the MC3302P is a quad comparator, only one IC is used in this circuit. Like I said, it's simple. Resistors R5, R12, R19, and R27 have been added to give just the slightest hysteresis (positive feedback) to each stage to help overcome any uncertainty at input voltages that correspond exactly to 'change of state' points. When powered from a computer's 5 volt supply, the range of input voltages is also 0 to 5 volts and the pot to be digitized is hung across the supply as laboriously

7

A FOUR BIT DIGITIZER



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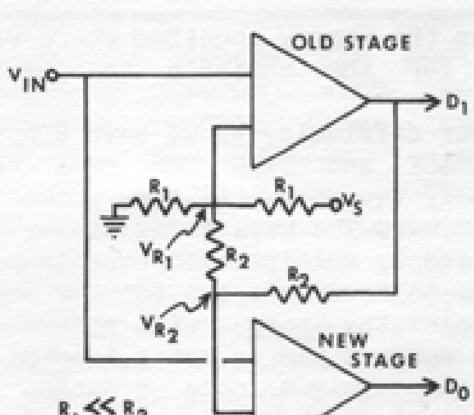

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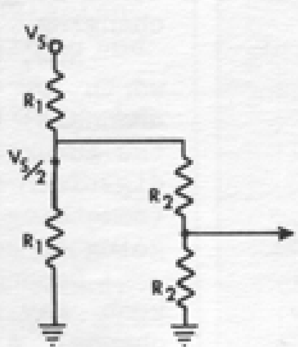
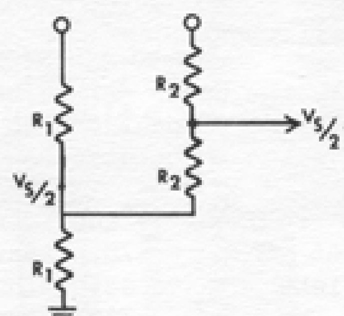


$\frac{V_S/2}{2} = V_S/4$

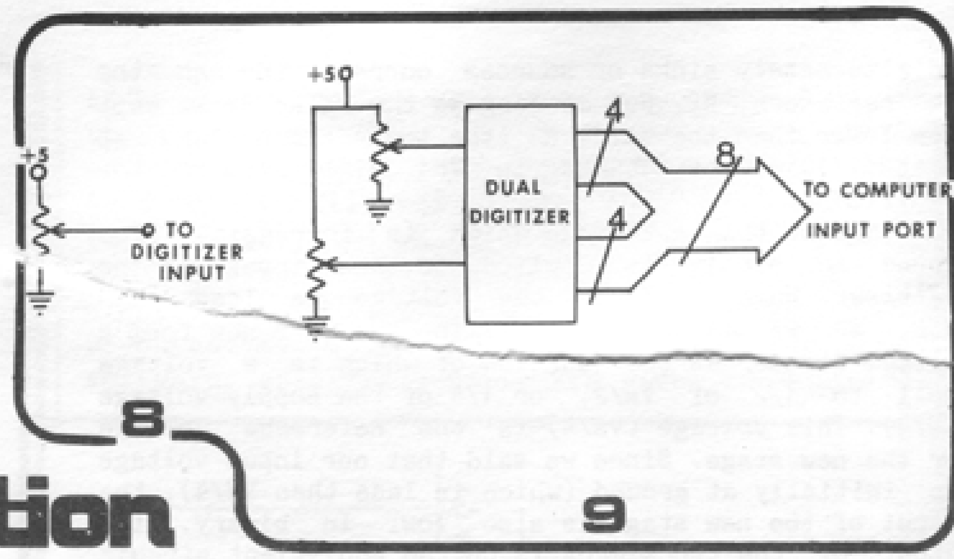
6



$V_S/2 + \frac{V_S/2}{2} = \frac{3V_S}{4}$

depicted in the formidable technical drawing of figure 8. At this point, we may as well establish the standard that the pot should be wired so clockwise rotation of the control causes the output of the digitizer to go from $0 to $F (see test program).

I believe that the most useful configuration for this circuitry is actually two digitizers on a single board, with each half providing half of an 8 bit word. The configuration shown in figure 9 is Paia's EK-7 and is made to plug directly into input port #2 of a Paia 8700 computer. It can also be connected to any 8 bit input port of any computer.



software consideration

The nicest thing about the digitizer is that it is easy to program for. There are no clocks to worry about and no elaborate software overhead (in fact, none at all). You just read the port to which the digitizer is connected to find the state of the knobs.

A good first example is the short program written for an 8700 to test the unit's operation shown in Listing 1. This program reads the output of the digitizer and shows the result in the 8700's displays. When the value of either of the digitizer outputs changes, the beeper sounds. As the knobs are rotated, the displays should show that the output increases or decreases sequentially without skipping any of the hexadecimal numerals $0 - $F and that there is no interaction between the two digitizer sections.

```
                :TEST FOR 4 BIT DIGITIZER
                :DIGITIZER INPUTS TO PORT #2
                :
                : A) SHOW OUTPUT OF DIGITIZER
                : B) BEEP WHEN VALUE CHANGES
                :
1000-  AD 08 08  STAR LDA INPT    :GET DIGIT
1003-  29 0F          AND 0F      :MASK HIGH BITS
1005-  C5 00          CMP *TEMP   :SAME AS LAST?
1007-  F0 04          BEQ LP1     :YES-BRANCH
1009-  18             CLC         :PREPARE
100A-  20 22 0F       JSR BEEP    :AND BEEP
100D-  85 00     LP1  STA *TEMP   :SAVE VALUE
100F-  8D 20 08       STA DISP    :SHOW VALUE
1012-  4C 00 10       JMP STAR    :AND SO ON...
                :
```

list1

The fact that there are two digitizer sections on the EK-7, one contributing the upper half-byte and the other the lower half-byte is going to be of great significance in some future software and hardware that we'll be doing.

For now, we'll use the PINK TUNES software (Polyphony July/August 78, pp. 22-26) as an example. When you review PINK TUNES, you'll notice that the statistical properties of the note durations (half notes, quarter notes, dotted notes, etc.) are controlled by the upper half-byte (UHB) and lower half byte (LHB) of memory locations we call MASK and TIME. We don't have the space here to duplicate the detailed explanation of how these variables interact which appeared in Polyphony, and is reprinted in "Friendly Stories About Computers/Synthesizers"; but briefly, both UHBs interact to determine the probability of a dotted note. The LHB of TIME sets the minimum note duration that will occur, while the MASK's LHB controls the range of possible note durations.

These dual half-byte control words are just right for use with a dual half-byte digitizer. From a programming standpoint, all we have to do is read the memory location where the knobs are ($808 on an 8700)

list 2

```
                              KNOBS FOR PINK TUNES
                :
                       .OR 1066
                :
                :BEFORE WE BEGIN, NOTE THAT THE FOLLOWING SECTION REPLACES PART OF THE
                :EXISTING PINK TUNES PROGRAM.  PRIMARILY, WE CHANGE THE BRANCH DESTINATION
                :FOR THE BNE AT LOCATION #066 SO THAT THE BRANCH IS TO THE TESTS WHICH
                :FOLLOW RATHER THAN BACK TO THE START OF THE PROGRAM AS IT WAS ORIGINALLY.
                :
1066-  D0 07          BNE TSTS      :RATHER THAN TO LP0 AS ORIGINALLY WRITTEN
1068-  20 71 11       JSR SET
106B-  20 2B 1D       JSR NOTE
106E-  00             BRK
                :
                :AS WE JOIN OUR PROGRAM, TEST HAVE ALREADY BEEN MADE TO SEE IF COMMAND
                :FROM KEYBOARD WAS FOR SCRAMBLE, TUNE, OR STOP.  NOW WE ADD TESTS FOR
                :CHANGE TEMPO OR TIME AND MASK PARAMETERS
                :
106F-  C9 0C     TSTS CMP 0C        :IS THERE A COMMAND AT ALL?
1071-  B0 9B          BCS LP0       :NO, JUST GO AHEAD AND BRANCH TO KEEP ON TRUCKIN'
1073-  E9 03          SBC 03        :NORMALIZE COMMAND FOR POINTER USE (CARRY WAS CLEAR)
1075-  08             PHP           :SAVE THE + OR - STATUS OF THE SUBTRACTION FOR LATER
1076-  AA             TAX           :AND TRANSFER THE RESULT TO POINTER, MAY USE
1077-  AD 08 18       LDA DGIT      :GET THE DIGITIZER OUTPUT
107A-  28             PLP           :NOW RECOVER THE + OR - STATUS OF THAT SUBRACTION
107B-  10 06          BPL TST6      :IF THE POINTER IS =>0 BRANCH TO CHANGE MASK OR TIME
107D-  09 F0          ORA 0F0       :TEMPO CHANGE, SET ALL UHB BITS TO 1'S WITH THIS MASK
107F-  85 A9          STA *TMPO     :THEN SAVE RESULT AS TEMPO CHANGE
1081-  D0 88          BNE LP0       :AND BRANCH ALWAYS TO CONTINUE
1083-  95 08     TST6 STA *TIME,X   :CHANGE TIME OR MASK PARAMETERS
1085-  10 87          BPL LP0       :BRANCH ALWAYS TO KEEP ON
                :
                       .EN
```

and put the result in the memory location where PINK TUNES is going to look for the variable that we're changing.

Now, there's a minor difficulty as we have 8 bytes worth of variables (MASK and TIME for each of 4 channels) to set with only two knobs. At some point in the future we'll look at hardware ways to multiplex our digitizer so it can be fed by multiple addressable pots (something like QuASHes in reverse), but for now we're going to actually multiplex the knobs - with software.

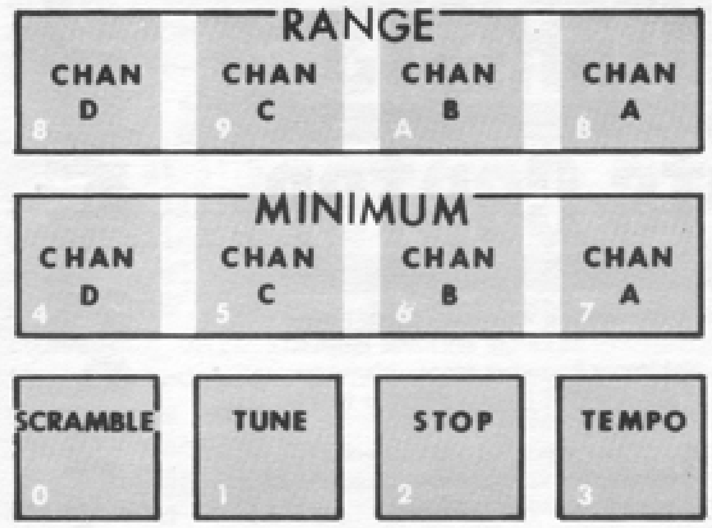
Depending on which command pad is being touched, a knob may be controlling minimum note duration on channel A, or the range of durations on channel C, or any of the other possibilities. The program shown in Listing 2 can be added to PINK TUNES to make all this happen. With the software running, the first 12 pads of the 8700's command keyboard take on the responsibilities depicted in figure 10.

The first three keys on the computer serve the same function they did in the un-altered PINK TUNES, but from there on it's all new. When TEMPO is touched, the knob corresponding to the LHB of the digitizer provides a coarse control of tempo; the other control has no effect. Touching one of the pads $4 - $B causes the selected parameter for the selected channel to be read from the pots. By the way, thinking of the pots as being labeled as shown in figure 11 will help you keep their functions straight in your mind (particularly if you remember that TEMPO is a duration function).

Yep, the knobs are definitely a plus for PINK TUNES. You can really try things out fast without having to shut everything down and scratch your head each time you want to change a channel from quarter to half notes, and so on. Also, the first program is a good example of how to program the knobs when you're setting variables that are organized as 4 bits each, two to the byte.

But there are other ways that the knobs can be programmed. For example, some parameters simply require more resolution than the 16 quantizing levels that 4 bits provide. An obvious answer is to think of the two knobs as both controlling one value, in which case the UHB knob can be thought of as a coarse range control while the LHB knob is fine tuning (our first test program can be thought of as acting this way). We'll look at another way that resolution can be extended in a moment.

In some cases the 16 quantizing levels provided by a single digitizer "channel" is sufficient resolution, but the resulting parameter must have a greater range than 4 bits allow. A brute force method of dealing with this is to use the output of the digitizer as a pointer to a table of parameter values like the code in Listing 3. This program reads a value from the table based on



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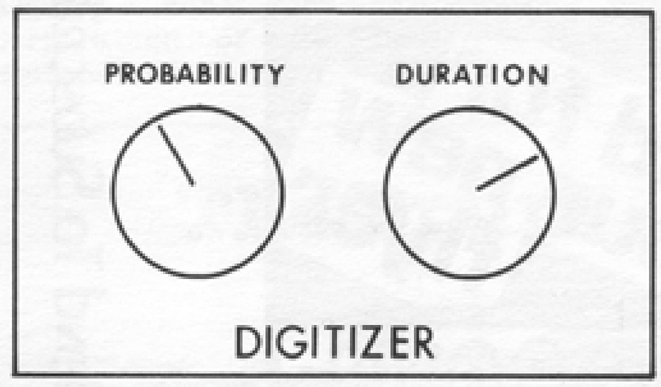

11

the setting of the LHB knob and shows it in the displays. In this case the table is an approximation of 1/4 cycle of a sine wave, but it could be anything.

In some cases the digitizer's output can be used in some way to calculate the parameter value.

One of the difficulties with software multiplexing of the knobs is that unless you're one of those people blessed with eidetic memory you have little way to know what the position of the knob was the last time you set

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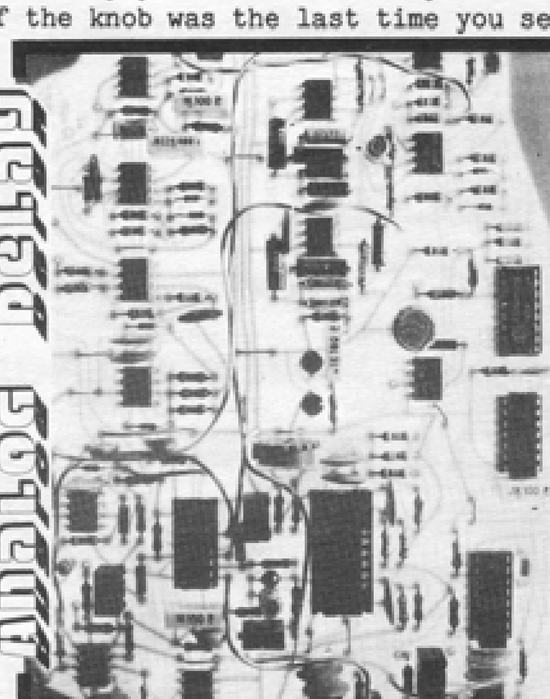

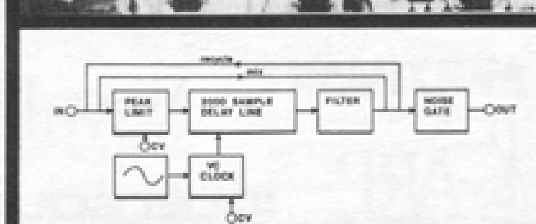

it. In some cases this isn't important, but in others (when you want to smoothly change a parameter from what it is to what you want it to be) it can cause problems. You punch in to change a value and the value immediately jumps to correspond to the current setting of the control. Glitch-ville.

A solution is to use the knob not to set the parameter, but to change it. That may not sound like a big difference, but it is. Using the knob to change the parameter means that when a function is punched in, the current setting of the knob is not important. As the knob is turned, though, the change in its position produces a corresponding change in the parameter. Try running the software in Listing 4.

With the code operating, any changes in the setting of the LHB knob are ignored completely until the parameter change is called for by touching the "0" command pad. Then, as the knob is rotated clockwise, the parameter (as shown in the displays) increases. Unlike the other code that we've examined, when the end of control rotation is reached, you can release the command pad, turn the knob fully counterclockwise, touch the pad again and continue increasing the parameter. This technique not only provides smooth control over a value without having to know its current state, it also extends the range of values that can be set with the knob.

The things that we've covered here are not all the possibilities, but hopefully they will get you started in adding variables to your software. It's really hard to beat a knob.

The following are available from Paia Electronics, Inc., 1020 W. Wilshire, Oklahoma City, OK 73116:
** EK-7 Dual Digitizer kit, with PC board and all parts (including pots, knobs, and sockets); $14.95 + $1 postage and handling.
** "Friendly Stories About Computers/Synthesizers", a compendium of past Lab Notes from Polyphony; $5.00 ppd.

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```
!L
0010 :                         TABLE LOOK-UP DEMO
0020 :
0030 :THE DIGITIZER IS READ AND THE LHB IS MASKED OFF WITH AN 'AND'.  THE
0040 :RESULT IS PLACED IN THE X REGISTER FOR USE AS A POINTER TO THE TABLE OF
0045 :VALUES WHICH OCCUPIES THE MEMORY IMMEDIATELY FOLLOWING THE PROGRAM.
0050 :THE VALUE CORRESPONDING TO THE KNOB POSITION IS FETCHED AND SHOWN IN
0060 :THE DISPLAYS.
0070 :
1000-   AD 00 18    0080 STAR LDA DGIT     :READ THE DIGITIZER
1003-   29 0F       0090      AND 0F       :'AND' WITH MASK (00001111) TO MAKE UHB ZERO
1005-   AA          0100      TAX          :THEN PUT TO X REGISTER TO USE AS POINTER
1006-   B5 0E       0110      LDA *TABL,X  :GET THE PARAMETER VALUE FOR THIS KNOB SETTING
1008-   8D 20 18    0120      STA DISP     :SHOW THE PARAMETER VALUE
100B-   4C 00 10    0130      JMP STAR     :THEN LOOP FOR MORE
                    0140 :
                    0150 TABL .HS 0019324A62788EA2B4C5D4E1ECF4FAFE
                    0160 :
                    0170      .EN
```

list 3

list 4

```
0010 :                         DELTA TUNE DEMO
0020 :
0030 :AFTER A SHORT DELAY GENERATED BY CALLING THE MUS 1.0 SUBROUTINE LOOK,
0040 :WE READ THE COMMAND KEYBOARD BY CALLING THE MONITOR SUBROUTINE DECODE.
0050 :ON RETURNING FROM THIS SUBROUTINE THE ACCUMULATOR AND Y REGISTER CONTAIN
0060 :THE NUMBER OF THE LOWEST KEY THAT WAS PRESSED ($18 FOR NO KEY). THE
0070 :CARRY FLAG IS CLEARED BY DECODE IF THE KEY WAS TOUCHED THIS SCAN BUT NOT
0080 :THE LAST, I.E. IF THE KEY WAS JUST TOUCHED
0090 :
1000-   20 4E 1D    0100 DLTA JSR LOOK     :THE CAPACITIVE KEYBOARD REQUIRES A DELAY BETWEEN SCANS
1003-   20 00 1F    0110      JSR DECD     :READ THE COMMAND KEYBOARD
1006-   D0 F8       0120      BNE DLTA     :IF ZERO KEY NOT TOUCHED, LOOP
1008-   AD 00 18    0130      LDA DGIT     :CHANGE COMMAND ASSERTED, SO GET THE DIGITIZER OUTPUT
                    0140 :
                    0150 :NOW THE ACCUMULATOR HAS THE DIGITIZED KNOB POSITION.  WE'RE REALLY ONLY
                    0160 :INTERESTED IN THE LHB, SO WE MAKE THE UHB ZERO WITH AN 'AND'.  IF THE
                    0170 :COMMAND WAS JUST ASSERTED, WE SKIP THE CALCULATION OF CHANGE IN SETTING
                    0180 :AND SIMPLY SAVE THE CURRENT SETTING AS THE STARTING VALUE
                    0190 :
100B-   29 0F       0200      AND 0F       :'AND' WITH MASK TO MAKE UHB ZERO
100D-   90 17       0210      BCC DNX0     :IF COMMAND JUST ASSERTED, SKIP CALCULATING CHANGE
100F-   48          0220      PHA          :SAVE KNOB POSITION ON THE STACK FOR USE LATER
1010-   38          0230      SEC          :PREPARE FOR SUBTRACTION TO FOLLOW
                    0240 :
                    0250 :IT'S TIME TO SEE HOW THE KNOB HAS CHANGED. CURRENT SETTING IS SUBTRACTED
                    0260 :FROM PREVIOUS SETTING AND THE DIFFERENCE (MAY BE + OR -) IS ADDED TO THE
                    0270 :CURRENT VALUE OF THE PARAMETER. TESTS ARE MADE TO SEE THAT WE'RE WITHIN
                    0280 :THE ARBITRARY RANGE $00-$3F AND IF OUT OF RANGE THE LIMIT IS SUBSTITUTED
                    0290 :FOR THE CURRENT PARAMETER
                    0300 :
1011-   E5 00       0310      SBC *TEMP    :TEMP IS THE POSITION OF THE KNOB THE LAST TIME THROUGH
1013-   18          0320      CLC          :NOW PREPARE FOR ADDITION
1014-   65 01       0330      ADC *PARM    :ADD THE DIFFERENCE BETWEEN NOW AND LAST TIME TO VALUE
1016-   10 02       0340      BPL DNX1     :IF GREATER THAN ZERO, SKIP THE NEXT INSTRUCTION
1018-   A9 00       0350      LDA 00       :IF WE'RE HERE, WE'RE UNDER-RANGE.  MAKE PARAMETER ZERO
101A-   C9 3F       0360 DNX1 CMP 3F       :ARE WE GREATER THAN THE MAX ALLOWABLE FOR PARAMETER?
101C-   90 02       0370      BCC DNX2     :NO, SO BRANCH TO SKIP NEXT INSTRUCTION
101E-   A9 3F       0380      LDA 3F       :OVER-RANGE, MAKE PARAMETER EQUAL TO MAX LIMIT
1020-   85 01       0390 DNX2 STA *PARM    :SAVE THE NEW VALUE OF THE PARAMETER
1022-   8D 20 18    0400      STA DISP     :AND SHOW IT IN THE DISPLAYS
                    0410 :
                    0420 :NOW WE GET READY FOR THE NEXT PASS BY SAVING THE CURRENT KNOB POSITION
                    0430 :
1025-   68          0440      PLA          :PULL THE DIGITIZER OUTPUT FROM THE STACK
1026-   85 00       0450 DNX0 STA *TEMP    :AND SAVE IT TO DETERMINE CHANGE IN SETTING NEXT TIME
1028-   4C 00 10    0460      JMP DLTA     :THEN JUMP TO START TO CONTINUE
                    0470 :
                    0480      .EN
```

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For Sale: Complete system: 2720-1, -2A, -3B, -3L, -4, -5, -7, -11, and -12 in a 4761 wing. Has been used as a guitar processor. Great condition, $175 ppd. Heit Bass guitar, excellent condition, $150 or best offer. Both for $300. Donald Poole, 17 Eagle Lane, Lutz, FL 33549.

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For Sale: 2720-7 Power Supply in great condition. Adjustment pots added for bias calibration. $20 or offer. Mark Carroll, 1550 Beverly Pl., Berkeley, CA 94706.

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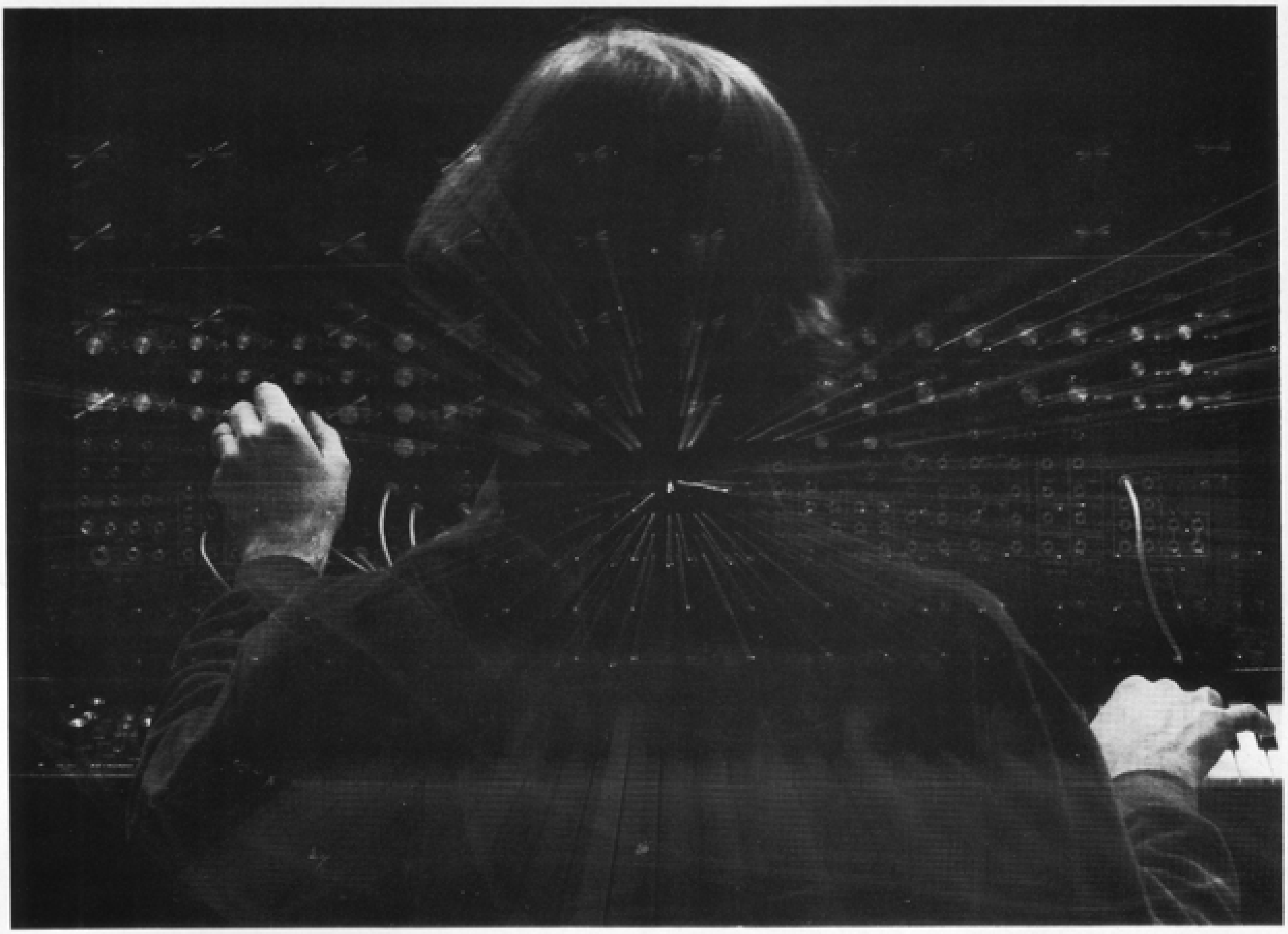

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Aries Music makes a complete line of modular synthesizers of unparalleled quality and performance at a price you can afford. Our music systems are available factory-wired or in easy-to-build kit form.

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:

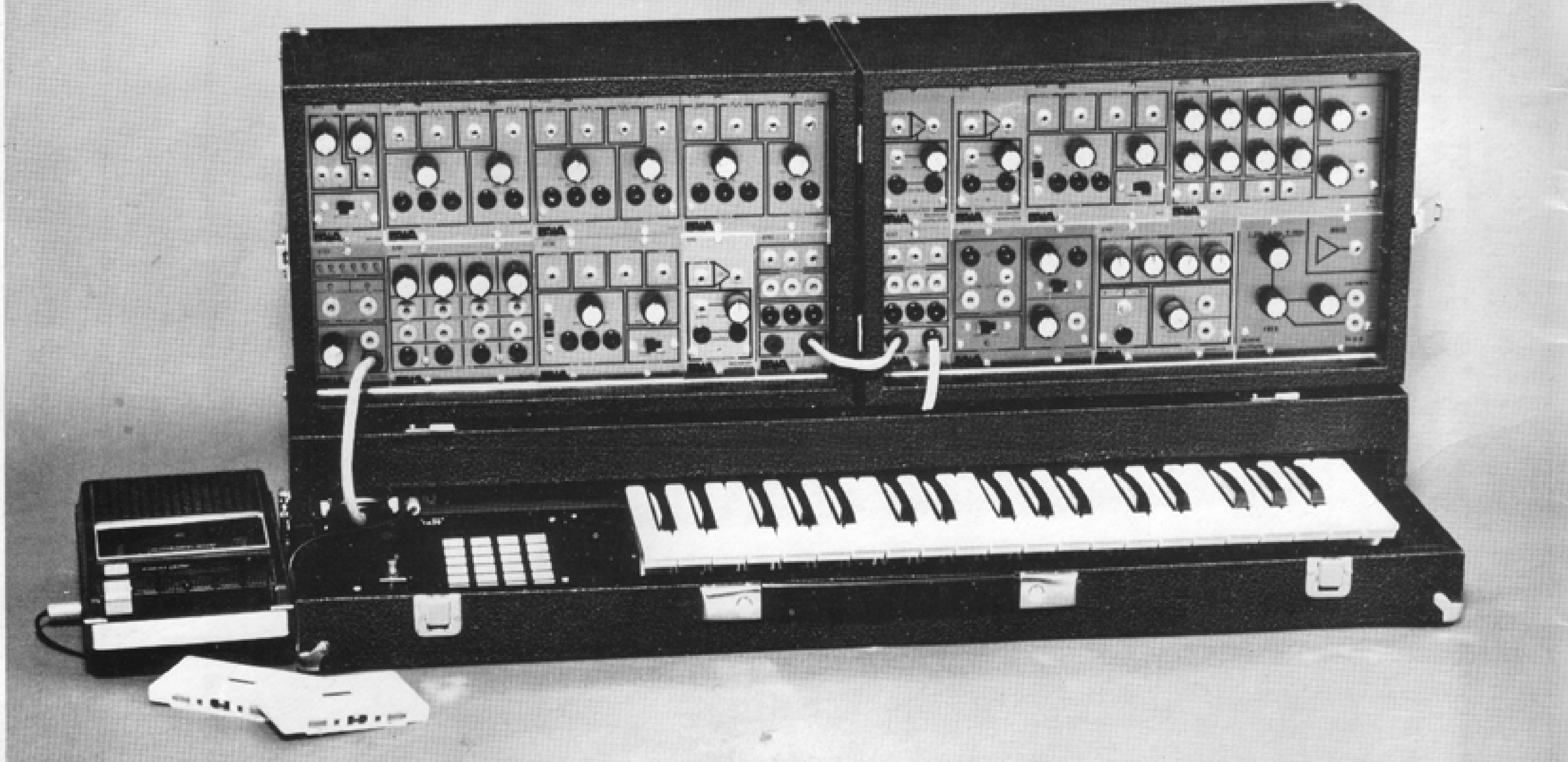
- phase-modulated sync
- dual-phase pulse waveforms
- voltage-controlled 10-stage phaser
- VCA sensitivity pan control
- hard and soft sync
- simultaneous linear and exponential fm
- 3 types of electronic switching

ARIES MUSIC

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

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 ....

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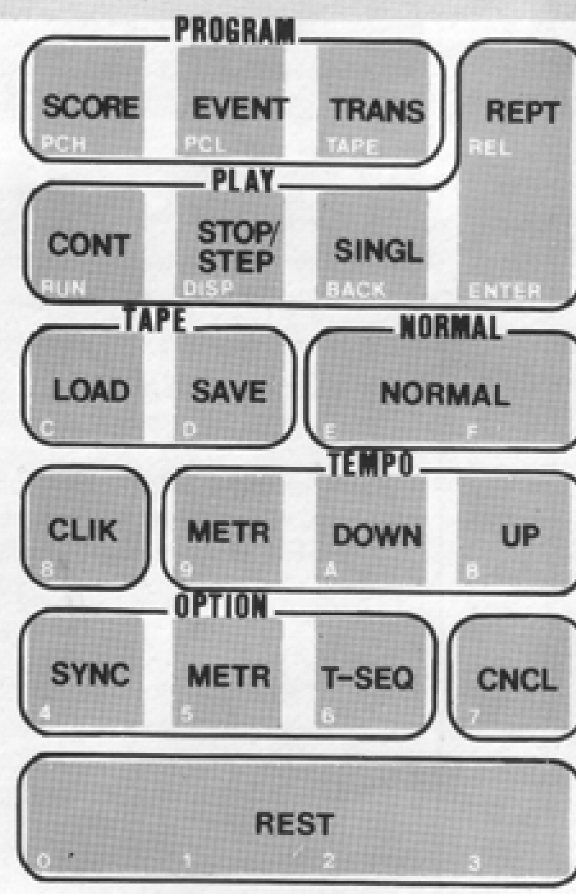
PAIA ELECTRONICS DEPT. 1020 W. WILSHIRE BLVD., OKLAHOMA CITY, OK 73116 (signature)



Typical control panel configuration

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