# POLYPHONY

T.M.

$2.00

March/April 1980

ELECTRONIC MUSIC & HOME RECORDING

ISSN:0163-4534

Computers In Music........

# ANALOG DELAY


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

### FEATURES

* TWO INDEPENDENT CHANNELS
* 3072 STAGES OF DELAY PER CHANNEL
* ADJUSTABLE INPUT AND OUTPUT LEVELS WITH INPUT OVERLOAD INDICATION
* INTERNAL OR EXTERNAL VOLTAGE CONTROLLED DELAY TIME
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If you haven't heard what analog delay can do for home music reproduction, you're missing something. Let's face it, stereo in your living room is flat and 2 dimensional. Without the mixture of direct and delayed sounds that a large hall provides, almost all music reproduced in the home is lifeless. Quadraphonics has not proved to be the solution to this problem. The recent developement of bucket-brigade semiconductor technology has made it possible to offer a reasonably priced delay unit that can transform your listening room into a concert hall. Using your present stereo system, the 2AS-A, and whatever you have in the way of 2 additional speakers and 2 channels of power amplification— you have all the parts to put together an ambience system that is capable of creating the kind of 'space' you enjoy music in. You don't need state-of-the-art componentry to enjoy an ambience system. The secondary power amplifiers and speakers can be of very modest performance and yet still serve to create strikingly realistic spaciousness in your listening room. If you don't have 2 extra power amp channels on hand, we offer several low cost, low power amps in kit form that would be ideal for this purpose.

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

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

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COLUMNS & REGULARS



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Elect Ronic Music in 1980!

# LETTERS

## More Thoughts on Records

I have been listening and reading on the current debate on new record companies for small artists with new music, and have organized my views on the subject. Considering that your magazine has been a forum for such issues, I am submitting it for possible publication.

In the last few issues of Polyphony, the idea of forming a private record company for alternative music has been debated. Although this concept has its merits for certain types of music, a common thread through many letters is an idea which could impede the growth of music, if accepted universally. Many musicians don't want to play games with the record companies any more, but what they fail to realize is that fragmentation of music is an inevitable result. The idiology of these musicians seems to be "I have a new musical idea, and I want to keep it pure, and free of commercialistic compromises." But, in actuality, these compromises, in the past, have resulted in most of the musical growth in the 20th century. The necessity of cross-fertilizing new ideas with accepted popular idiom has resulted in new types of music, a wider acceptance and knowledge of these new ideas, overall- an advance in music in general. If, through these splinter groups now forming, artists with new ideas do not try to bridge the freshness of the concepts with the popular musical idiom, musical progress will undoubtedly suffer. For this reason, therefore, it would behoove current artists to attempt to bring their ideas into a popular context. From the past, we can see that this compromise of ideas with pop idiom has not caused the artist to suffer a lack of integrity. Rather, it has resulted in most of the major musical advances of the twentieth century, along with the creation of new musical forms.

The Beatles, surely the single most influential group in modern history, were abound with totally new concepts, along with concepts borrowed from other pioneers. The use of the studio, the redefining of the pop album, use of synthesis and multi-track vocal and instrumental techniques, and changes from traditional song structure, both lyrically and musically, are just a fraction of the ideas they fostered. Yet, they mixed these new ideas with popular idiom to maintain their audience. Without this compromise, their music would not have had such long range influence. For example, the use of synthesis ideas in such folksy songs as "Here Comes the Sun" and "Maxwell's Silver Hammer" resulted in acceptance of synthesis by their listeners, while esoteric, "just synthesizer" songs would have turned many off. Their attitude about using all types of music in relation with new ideas to make a sound which can achieve mass acceptance can be seen best in works like Abbey Road and The White Album. The end of the latter work shows their respect of all types of music, from the esoteric to the totally safe.

The fruits of this compromise between new ideas and the popular line can be seen in modern artists like Earth, Wind, and Fire, Steely Dan, The Eagles, and The Cars, all of which have combined the idiom of the day with their ideas, with the upgrading of music as a general result. Thus, as a writer once said of Steely Dan, "warping pop conventions instead of breaking them" is the route to true creativity which touches a large number of people. If Steely Dan had taken their rather sophisticated musical and lyrical ideas and kept them "pure", we would all be the big losers, as we probably would not hear from them too often, and the lack of sales potential that would result would limit their studio time, resulting in an inferior product for their few remaining fans. Pink Floyd's new album also shows their shift to more accessable music, and yet Floyd freaks love it. They even have a single.

I'm not saying that commerciality is ipso facto musical success. Many artists, by not bringing new ideas to the pop idiom, have created boring music which have given pop a bad name. The Bee Gees, Barry Manilow, and 95% of disco is proof. Also, I'm not condemning efforts to form small labels. I'm just saying that compromise to bring music to more people and improve the state of music should be considered before assuming a "commercial music stinks" stance.

John Lazzaro
Phila., PA

## Good Vibes

Just a note along with my subscription renewal. You've got the best mag in the art and I really look forward to getting it. I've used your PolyMart ad a few times to purchase records and hope to get more. I have over 40 albums of electronic music and have a few singles from the New Wave electronics that's coming out of Britain (Throbbing Gristle, Orchestral Movement in the Dark, Robert Rental and Thomas Leer, John Foxx). John Foxx is really a very good album. Keep up the good work and I'll spread the word around that yours is the best magazine.

Alex Douglas

Thanks for the good comments, Alex. You and other readers may be interested to know that PolyMart will be stocking an increasing number of electronic and experimental albums. In many cases, these will be releases which are hard, if not impossible, to find locally. Additionally, the majority of the records we will be selling will be import pressings to assure the best possible quality and musical variety. Watch for Pat Gleeson's latest, Moebius, Eno, Cluster, Tangerine Dream, Jarre, and many more. PolyMart wants to be your 'one-stop' distributor for books, records, and ...?... Let us know what types of supplies you have trouble finding locally. We'll dig them up!

Marvin

## Article Ideas

I have enjoyed the development of Polyphony and, as evidenced by my subscription renewal, look forward to your continued success. However, I would welcome:

- articles which are longer, more in-depth when appropriate rather than truncating articles to allow more articles - a compromise between depth and breadth
- a column on technical, electronic, and artistic questions and answers
- a series of articles on programming the Paia computer: approaches, techniques, etc. along with current software presentations
- more software examples
- delivery of promised features and projects: interfacing computers to organ/ string ensembles, multiplexed digitized pots, and so on.

Terry Ward

Thanks for the recommendations for new Polyphony material. As always, we are working on expanding our format with more material and increased publication. The Q & A column sounds good; Polyphony has built up a number of contacts throughout the industry who would probably be willing to help us get the answers to your questions. We can get through to most of the designers, technicians, and product specialists for the leading manufacturers, as well as many well known artists, producers, and record labels. If you have questions, send them to us, along with the names of the people to whom you want the question directed (if applicable). We'll see what we can dig up for you.

Marvin

## Elements

On your center page (PolyMart) of the November/ December 1979 issue, you have mentioned that 'Elements' is self produced. These were not self-produced by Danna, but definitely the production of a publishing company, with the same involvement as that of any recording company. As a matter of fact, Danna is better off, because we are paying every cent put into production and in no way will we ever collect anything back from him, as do the record companies. This is an unusual move but one I feel worthy of a publisher of our present status.

(Mychael) has no basic home equipment and all of his works have been done through my company, at the studio. The last one in Toronto on 24 track, so you can see - 60 hours of 24 track studio time, plus everything else thrown in - this is certainly not self produced.

Record #2, which is named 'Mychael Danna', should be off the press within the next few weeks and I will send copies for you to sell and review at that time.

W. Ray Stephens
President
Frederick Harris Music
Oakville, Ontario

I wish there were more record companies who had the faith in their artists that you show. It couldn't help but nurture the musical state of the art a bit!

Marvin

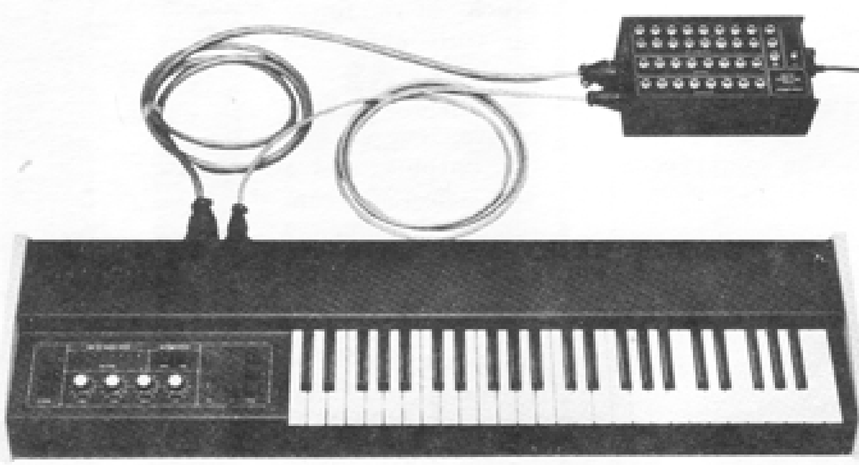
## Send Us Your Ideas!

for projects, interviews, columns, reviews, or ...?... Polyphony can provide all the material you need; give us the raw ideas - we'll find people with info! Our increasing scope and space for coverage make Polyphony your #1 source of info on experimental music technique.

# INDUSTRY REPORT

*'Tell Them You Saw It In Polyphony'*

POLYPHONY PUB.CO. MARCH/APRIL 1980 VOLUME 5/ISSUE 6



## POLY KEYBOARD

Polyfusion's new model 2058 Polyphonic Keyboard represents a major technological breakthrough in synthesizer controllers. All external controls are music related, the logic programming is done automatically inside. Although originally designed to control synthesizer voices in our large modular systems, the 2058 can control from one to eight synthesizers or expander modules of any type, through use of the 2068 interface. This powers the keyboard and provides the 8 pitch, gate, and velocity outputs as well as inputs for three different modulation signals. If fewer than eight voices are patched, the 2058's "auto logic" circuitry will adjust itself for proper control of the outputs being used.

Among the many special performance features are the UNISON mode which allows all voices to track one key, GLIDE in both UNISON and POLYPHONIC modes, and VELOCITY sensitive keys for highly responsive dynamic expression. The OCTAVE TRANSPOSE function provides a six octave playing range. Micro-Touch (TM) MODULATION SOURCE SELECTORs allow the choice of any of three modulation inputs. A totally new PITCH BENDER with spring return, positive detent, and zero friction, gives bi-directional control of all voices. Voice STATUS INDICATORS (a row of eight LEDs) inform the musician as to which voices are on. Footswitch inputs on the rear panel allow remote selection of UNISON mode, GLIDE, and OCTAVE TRANSPOSE features. Also on the rear are eight separate scale controls so that all voices can be tuned to track perfectly.

In addition, there are master controls for SCALE, RANGE, GLIDE (rate), VELOCITY (amount), and MODULATION (amount). All connections are made at the interface, which can be locarted up to 50 feet away from the keyboard.

The 2058 Polyphonic Keyboard comes complete with the 2068 Polyphonic Keyboard Interface, interconnect cables, footswitch, operator's manual, and a one year warranty. For more information, contact: Polyfusion, Inc., 160 Sugg Road, Buffalo, NY 14225.

## COMPUTER FESTIVAL

The Personal Computer Arts Festival of 1980 is a two day festival of talks and papers, films and graphics, demonstrations, and musical performances scheduled for August 23 and 24 in Philadelphia, PA. Daytime activities for the first day include tutorials and seminars with presentations by artists and amateurs. People interested in submitting papers or topics for presentations should send rough outlines and tapes of compositions or performances to PCAF before June 1, 1980. The PCAF-80 Computer Music Concert will take place on Saturday evening, with performances selected for their originality and artistic merit. Daytime activities for Sunday August 24 will present the Visual Arts Festival with still graphics, video and film works, and real time graphics generation.

The PCAF-80 is being held in conjunction with the Personal Computing '80 Show at the Philadelphia Civic Center. For more information, contact: PCAF 80, c/o Philadelphia Area Computer Society, Box 1954, Philadelphia, PA 19105.

## DIGITAL SYNTH

The Con Brio ADS 100 is an advanced digital synthesizer which utilizes 3 microprocessors to provide capabilities for 64 digital oscillators, each with independent amplitude and frequency control, and 128 separate and extended envelope generators. The full sized 8 inch floppy disc provides the speed and capacity the live performer needs for instant recall of previously developed patches, mix assignments, voice placements, or ensemble structures. In addition to storage of patches and programs, the discs can be used to store any activity from the two 5 octave keyboards or the digital command console. This includes performed musical material, keyboard voice assignment in predetirmined sequence, and alternative- not well tempered- tunings.

Operational modes include Synthesis 1, which is adjusted similarly to conventional analog synthesizers, or Synthesis 2, which allows any number of ascillators to be added together or fed into one another to produce sounds. Each oscillator is individually frequency and amplitude modulated by digital envelope generators which consist of 16 separately adjustable segments. Video display of the envelope under construction is provided. All information needed to recall any patch from either of the two Synthesis modes can be stored on the disc. Vast libraries of voices for the ADS 100 are available from Con Brio.

For more information, contact: Con Brio, 975 San Pasqual St., Suite 313, Pasadena, CA 91106, (213) 795-2192.

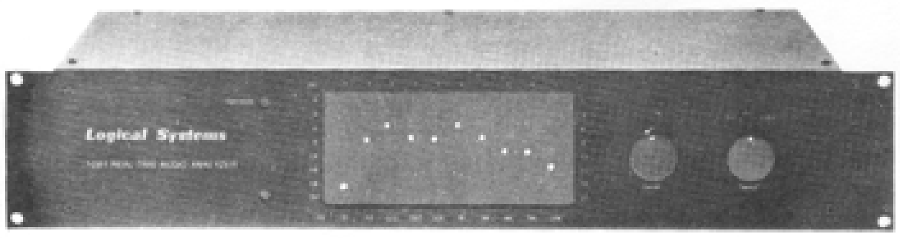
## SUMMER SCHOOL

The New England Conservatory of Music will hold a summer session from June 30 through August 8, 1980 featuring workshops, courses, and master classes. Among the numerous offerings of special interest is the Electronic Music Workshop with Robert Ceely, June 30 through July 3.

The workshop will be divided into two sessions. Mornings will include lectures and demonstrations of the hardware and software of Electronic Music and discussions of how and why synthesizers work and the theory behind their sounds. Video tape is used to give 'over the shoulder' views of a wide variety of synthesizers and related equipment. The afternoon sessions will be individual hands-on experience with the Arp, Moog, Buchla, and EML synthesizers. Studio technique covering multi-channel recording, sel-sync, overdubbing, reverberation, etc., sophisticated sequencing techniques, and unusual patch configurations will also be explored and demonstrated.

Robert Ceely, a member of the New England Conservatory's composition faculty, is also Director of the school's Electronic Music Studio. He received his B.M. from NEC, his M.A. from Mills College, and has studied extensively at Tanglewood, Princeton, and Darmstadt with such composers as Francis Judd Cooke, Darius Milhaud, Leon Krichner, Roger Sessions, and Milton Babbitt. Mr. Ceely's compositions include works for large and small instrumental ensembles and for magnetic tape.

For additional information on this or other offerings of the New England Conservatory, contact: Robert Annis, Director of Summer School, 290 Huntington Ave., Boston, MA 02115, (617) 262-1120 ext. 205.

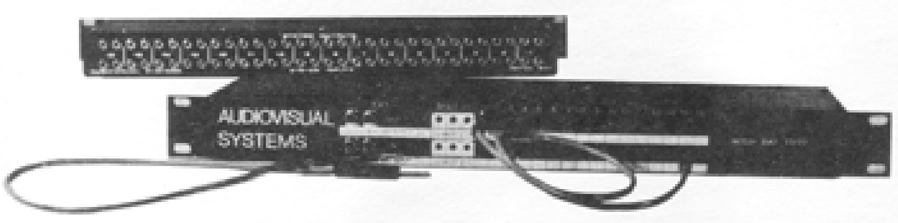



# ANALYZER KIT

Logical Systems has just announced their newest electronic kit - the 1081 Real Time Audio Analyzer. The 81 LEDs of the 1081 form a picture of any audio signal you wish to see. It displays a 21 dB range of energy in each of ten octaves. The octaves displayed match most graphic equalizers, as the filters are tuned to ISO standards.

The 1081 kit comes with a 3.5 inch tall black rack mount chassis with optional oak wood ends. The device may be hooked up to a receiver, preamp, mixing board, tape machine, or the audio jack of a video tape recorder. Input level is 1 volt nominal into 100K ohms; filter Q for each passband is 2.9; Q and frequency stability of filters is $\pm 5\%$; meter accuracy is 0.5 dB per step with 3 dB steps; connections include terminal block and stereo jacks. The features include a diagnostic sweep signal and an input jack for a dynamic microphone. All components are circuit board mounted for ease of construction. The kit takes approximately two evenings to assemble.

The 1081 Audio Analyzer kit sells for $179; it is also available assembled for $295. For more information, contact: Logical Systems, 3314 "H" Street, Vancouver, WA 98663, (206) 694-7905.




# PATCH BAY

AudioVisual Systems announces the release of the first studio quality patch bay system for the audiophile and home recordist. The patch bay features 16 stereo inputs and outputs (256 2 channel crosspoints), with 64 gold-plated RCA phono connectors on the rear panel and 3 conductor Bantam jacks on the front.

"Fully-normalled" design means that no patch cords are necessary for normal operation. Cords may be inserted to break normal connections and insert or rearrange components or signal-processing devices. Once set up, there should be no need for access to the rear of any equipment. Front panel phono connectors enable connection of external equipment anywhere in the system. The device uses a unique fully-shielded printed circuit design with no discrete wiring or active circuitry, has gold plating on all contact surfaces, uses no power, and is contained in a 1 3/4 inch EIA standard rack package. The patch bay will introduce no additional noise, crosstalk, or distortion to line-level signals, as all connections are "hard-wired".

The AudioVisual Systems Patch Bay comes with two patch cords, and detailed instructions, ready to use. The unit sells for $540, fob Los Angeles. For more information, contact: AudioVisual Systems, 725 Lorraine Boulevard, Los Angeles, CA 90005, (213) 934-3006.

# HAPPY BIRTHDAY!

On December 31, 1979, Oberheim Electronics celebrated its 10th anniversary. The original charter of Oberheim was to develop and manufacture products oriented toward the performing musician ("to bring the studio to the stage"). During their first decade, Oberheim has been the innovator of numerous products, many of which have become standards in the field, such as: first performance ring modulator (1970), first performance phase shifter (1972), first digital sequencer for performance (1973), first synthesizer expander module (1974), first polyphonic "true" synthesizer and first performance synth with built in sequencer (1975), and many other recent releases.

For more information on the growing Oberheim line, contact: Oberheim Electronics, 1549 Ninth Street, Santa Monica, CA 90401.

# EFFECTS SYSTEMS

Musicians who require specialized equipment to develop their sound will be happy to learn that Sounder Electronics (21 Madrona St., Mill Valley, CA 94941, (415) 383-5811) builds custom Portable Effects Systems. Each system is designed and built to the musician's exact requirements.

For example, the effects system pictured here includes a two channel, nine input mixer with a stereo octave equalizer. This system is programmed to produce up to eight special effects, yet it is surprisingly compact, weighing less than forty pounds. The entire system, including foot control board and cords, fits neatly into two cases for quick set-up and easy transportation. The special effects- which are engaged by touching a push button switch on the unit or by tapping a switch on the foot control board- include an octave divider, flanger, distortion, digital delay, reverberation spring, compressor, a voltage controlled filter, and wah-wah. LEDs on the foot switch board pulsate to tell you which effects are engaged.

"We like to work closely with each musician to build a system that will match their needs. When Paul Kantner of Jefferson Starship came in, he wanted a system that could do everything. We built it." said Hamilton Agnew of Sounder Electronics.

Sounder has a long history of designing and building custom equipment for such artists as thhe Grateful Dead, Jefferson Starship, Joni Mitchell, and members of the Oakland Symphony. For more information, contact Sounder Electronics at the above address.

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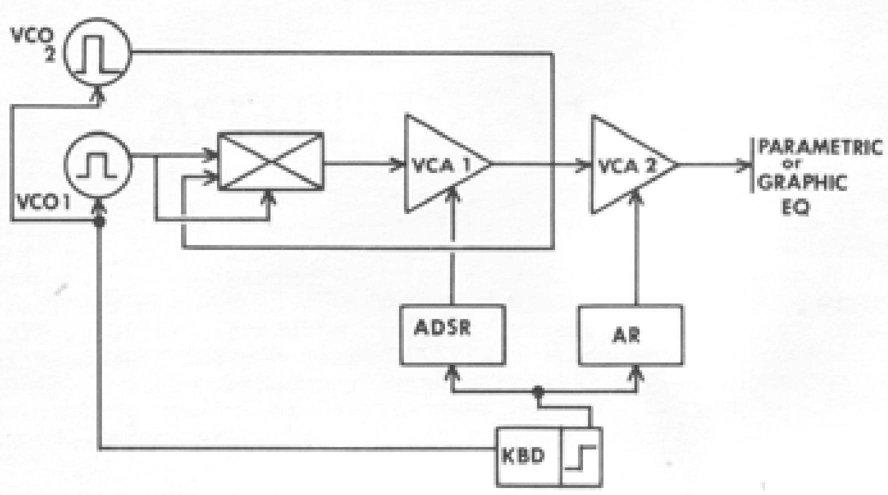
# Patches:



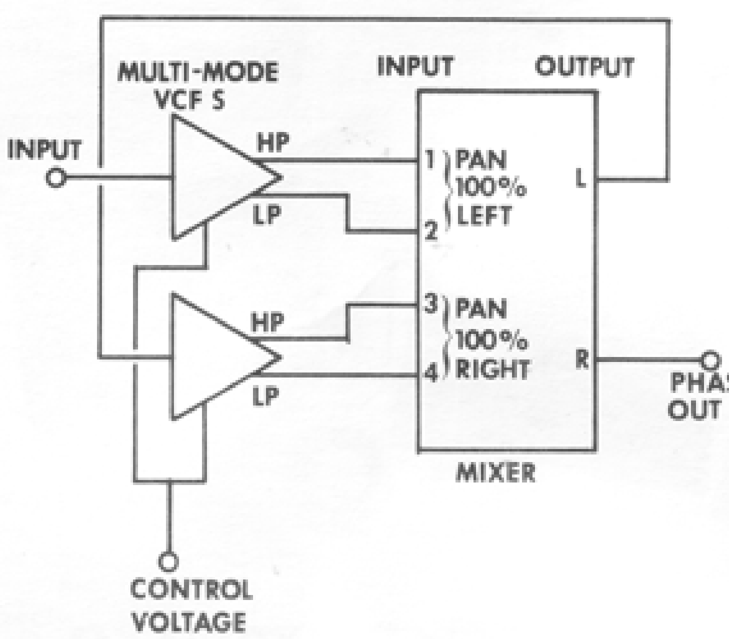

## Ring in the Fundamental

I like to make patches that are capable of a large variety of sounds, what I call a compositional patch. With this patch you can ride pitch, filter frequency, and LFO output with interesting effects. Reduction of AR parameters decreases crossover time. An ADSR with its S set to minimum could be inserted between the inverter output and balanced modulator control jack.

LFO: 8 Hz
VCOs: 50% pulse width (on filtered VCO)
25% pulse width (on enveloped VCO)
VCF: Q - 100%
Inverter: 5 volt offset
Reverb: 30%
ADSR: A - 100%
D - 100%
S - 100%
R - 100%
AR: A - 50%
R - 100%
Expanded Range

Variable output: Temporarily disconnect VCA plug from audio jack of balanced modulator. Press a key down and adjust AR output to balance the modulator for no signal out. If it doesn't balance out, leave it at maximum or sum in a little positive bias.

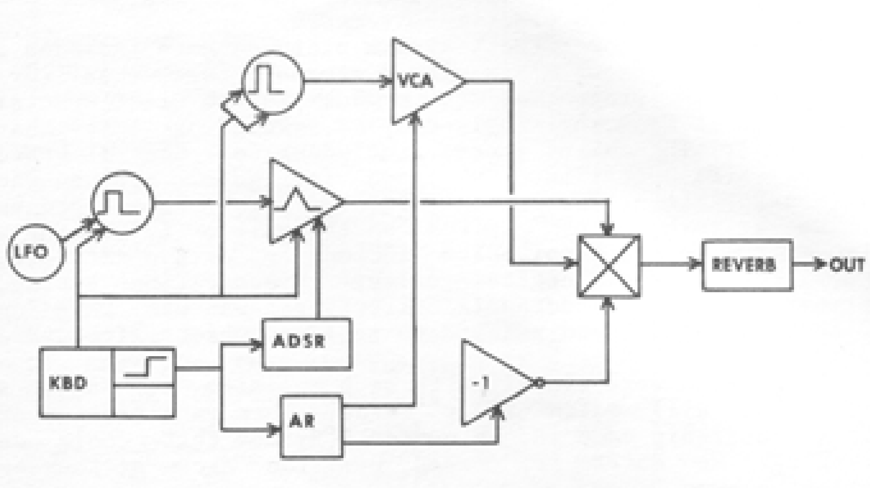
Jack Savage
Ogilvie, MN




## Voltage Controlled 4 Stage Phase Shifter

NOT an imitation, this patch provides a true 4 stage phase shifter response, easily voltage controlled by any 0 - 5 volt control source. Adjust mixer ins and outs for best effects. Q of each VCF can be varied to suit the application. Also, spacing of the notches is continuously user variable by adjusting the initial frequency controls of the VCFs. More VCFs and another mixer could be added in series for more complex phasing effects.

Tim Fluharty
Farmington, NM

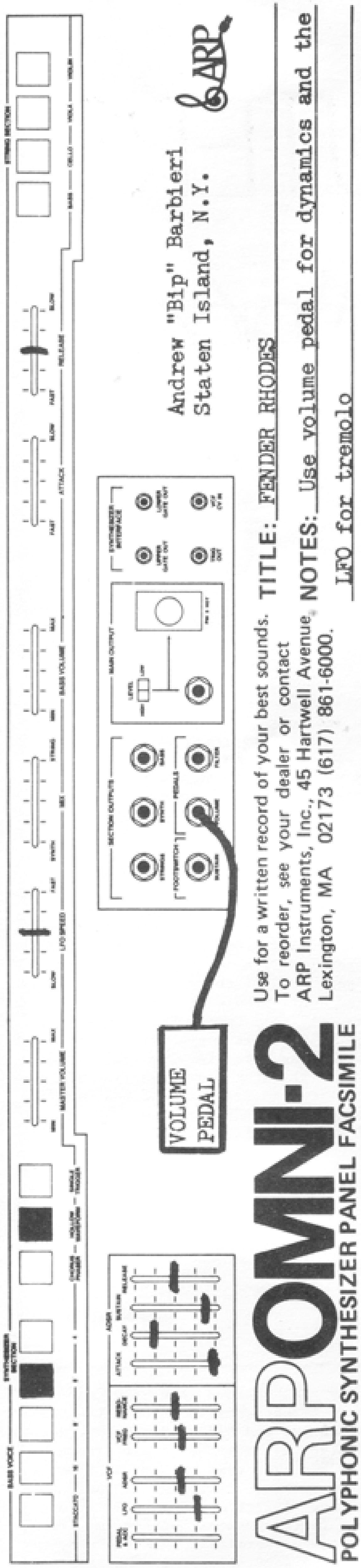


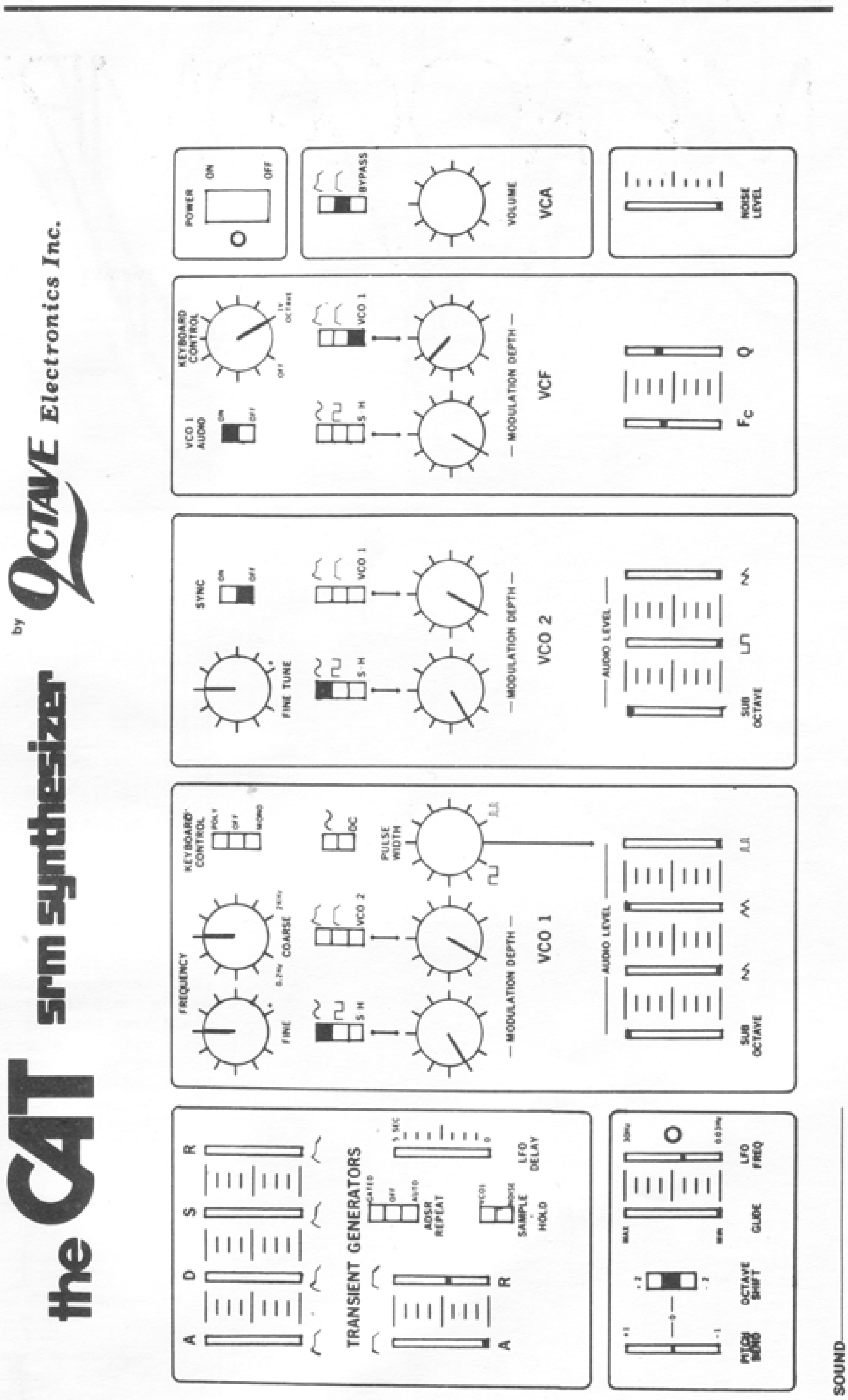

## Hot and Buzzy Filter Effect

Kbd: Try lowest range first
VCOs: Tune unison to a slow phase
30% duty cycle
Ring Modulator: Adjust input level for 100% modulation
ADSR: A - 0%
D - 0%
S - 100%
R - 0%
AR: A - 10%
R - 30% or less
Low timing range

To get the full benefit of the effect, your amp's controls should be set to maximum treble and bass and, if possible, run through an equalizer; I use a good 6 dB boost at 2 KHz, 3 dB at 6 KHz, and 6 dB at 280 Hz. This makes a good lead synthesizer patch - a real tribute to the virtues of abusing your VCA.

Bill Williams
Clarkston, MI




Serendipity One

Notes: Tune VCO1 slightly flat. In polyphonic mode, this patch sounds best when perfect intervals (fourths, fifths, octaves) are played on the keyboard.

Jim Johnson
Terre Haute, IN

# Part 1: HARDWARE

# MICROCOMPUTERS IN REAL TIME AUDIO

by Tony Lewis

By this time the power of a computer in the field of number handling is well known and exploited throughout industry; however, only recently have computers begun to be used in applications requiring the computer to interface to an analogue world. I have been doing audio research for several years and wondered if a digital computer might somehow be a useful tool in the so-far non-digital field of audio research.

After reviewing the limited amount of technical information on the subject of digital audio the following became apparent:

1. The only computers heretofore applied in the field of audio were the larger mainframe type and minicomputers.
2. Real time processing of an audio signal by computer had not yet been explored.
3. Computer generation of waveforms had been overlooked by many in the field of digital audio.

With the above realizations I decided that it would be worthwhile to attempt to develop uses for a digital computer in audio work with one important difference from the work done by those before me. Due to economic constraints the only type of computer I could afford for my laboratory was a microcomputer as opposed to a mainframe minicomputer. If I could develop several useful applications for a microcomputer in the audio labratory, I would have a truly cost effective general purpose audio tool.

Upon appraising my past projects in audio I set forth the following list of functions that would be an asset to have available to me in my laboratory. This paper deals with my attempts to realize these functions through use of a microcomputer.

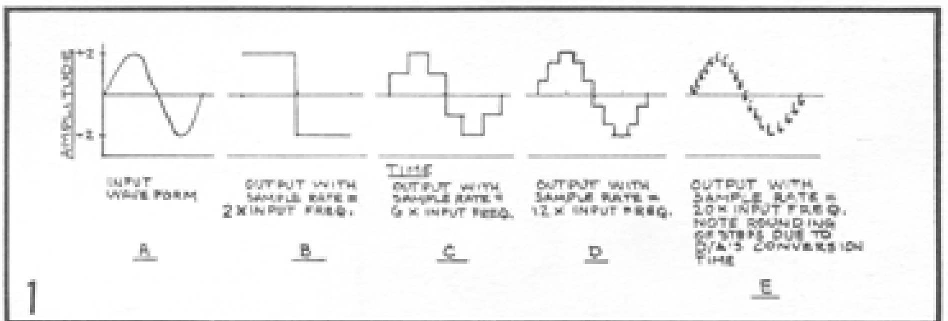
1. An automated polytonic pitch source.
2. A source of complex waveforms containing only those frequences that I specify in any combination or at any amplitude.
3. Waveform analysis to yield data on root frequency and harmonic content of a repetitive signal.
4. The ability to process a signal in real time without the use of extensive banks of analog filters.
5. The ability to record and redisplay by some means short term electrical pulses.

Within the last year I have developed hardware and software to meet each of these goals with varying degrees of success. In this paper for the sake of brevity I would like to concentrate on only one of these goals. For this paper I have selected what I feel is the most novel of the goals in the fact that, to my knowledge no one has ever applied a computer in quite the manner I have in my use of the computer as a real time audio processor.

I will first discuss pertinent background information and specifications I found necessary to set forth at the beginning of the project. I will then explain the functioning of the hardware that was developed to work with the computer and the software that was written to run the machinery. I will conclude with my results to present and my plans for future research in the area of digital audio.

To achieve real time computer processing of an analog signal it is first necessary to be able to convert an analog signal to a continous string of binary numbers with which the computer can work. After the computer has performed whatever operations were necessary to this string of data, the numbers must be converted back into discrete voltage levels to make up a waveform which can be amplified and fed to a speaker or an oscilloscope. The process I have just outlined requires two types of data transformation-from analog to digital and from digital to analog. (These are henceforth abbreviated A/D and D/A respectively). Before going into great detail concerning how the entire process is achieved it is important to examine one critical specification we must place on the process.

Each cycle through the process (A/D conversion, computer function, D/A conversion) requires a finite amount of time and since we are working a signal in real time the data is continuously changing. Therefore the entire process must repeat very rapidly to keep up with the changing data. In fact, the maximum rate at which the process can cycle determines the maximum rate at which the data can change. For example, suppose I have a sinewave that oscillates at a rate of ten kilohertz. It is obvious from basic sampling theory that the computer could determine frequency of waveform by taking only two samples per cycle of the wave. From this it would seem that the computer could handle the ten kHz sine wave with a sample rate of only twenty kHz; however, the computer has no way of knowing if it is a sine wave or any one of an infinite number of waveforms. It is for this reason that the sample rate (the rate at which the conversion process cycles) must be more than twice the frequency of the waveform with which the computer is to work. Figures 1A through 1D illustrate the effect of waveform distortion due to sampling rate.



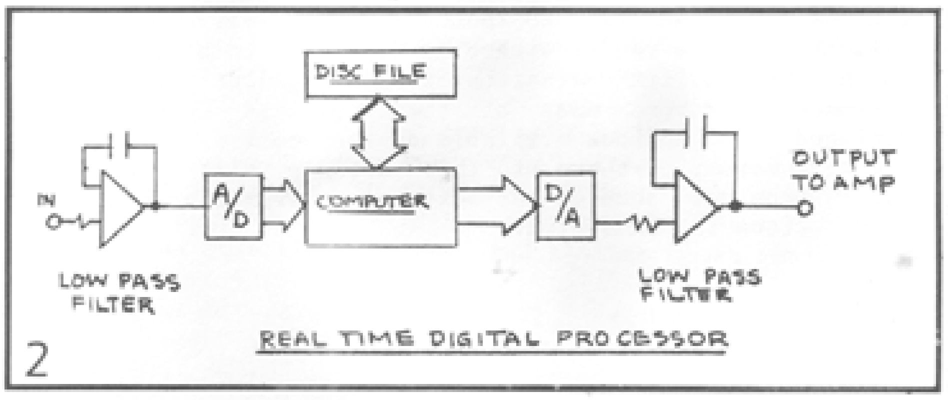

1

The only practical way to achieve the A/D and D/A conversion was by development of hardware to perform the task. I felt that I could design A/D and D/A hardware that would function rapidly enough to place my maximum input frequency well above the audio range. The real problem of time lay within the computer, for its machine cycle is a one full microsecond long. There are faster machines available but economic constraints place a faster machine outside my range of practicality. The fastest operation that the computer could perform would be to merely take data from the A/D and send it to the D/A. Obviously this is not a very purposeful operation on the data but it will serve to get an idea of the max. operating frequency of the system.

The "pipeline" operation just described requires thirteen machine cycles, therefore thirteen microseconds. A sample rate of thirteen microseconds corresponds to a frequency of 76.9 kHz (frequency = 1/sample rate). The assumption of 10 samples per cycle to reduce sampling arror would yield a max. input frequency of 76.9 kHz/10 samples per cycle = 7.69 kHz.

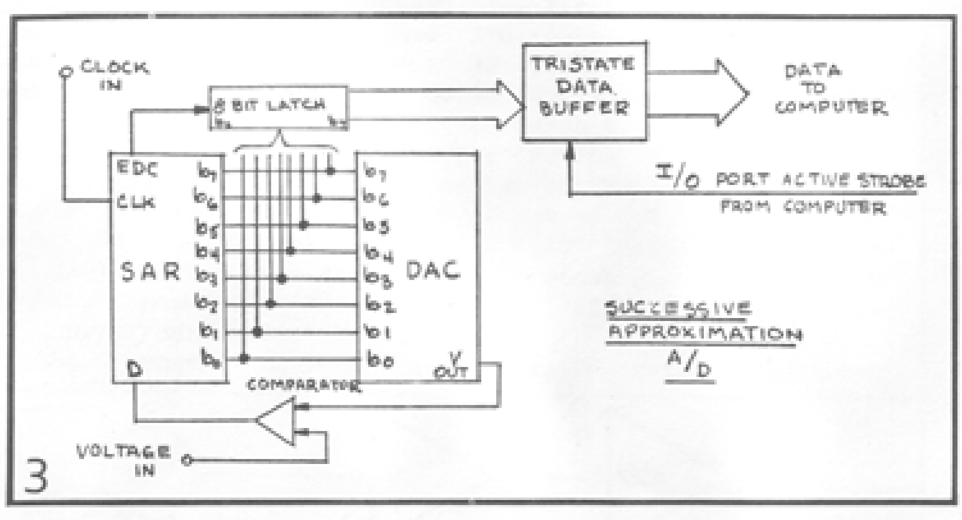
The purpose of the preceeding discussion was to alert the reader to the fact that even though a computer seems to operate infinitely fast it barely operates fast enough for my intended application. 7.69 kHz is cutting rather far into the upper end of the audio spectrum and as the complexity of the computer's operations on the data stream increases, the band width drops to approximately 5 kHz. While this limited bandwidth will degrade the sound quality to some degree, it will not be intolerable.

Now that the main limiting factor on the process has been discussed I will turn to a more detailed discussion of how the system as a whole and the separate pieces of hardware operate. See figure 2 for a block diagram of the system hardware. The input waveform first enters a low pass filter which acts to remove all frequencies in the waveform above the filter's cutoff point. From the preceeding descussion it is obvious that the system has a maximum frequency input limit and the low pass filter's cutoff point is set to this frequency. If frequencies above the system limit were sent to the A/D, a problem called aliasing would develop. When I report the problems I encountered in the system along with my results, I will further explain aliasing. The low pass filter which I designed for the input is a standard integrator type active filter.



2 REAL TIME DIGITAL PROCESSOR

Following the input filter is the A/D converter. There are several basic types of A/D converters. The two most commonly used types are called dual slope converters and successive approximation converters. Dual slope converters have less complex circuitry; however, they are quite a bit "slower" (i.e. require more time to convert a voltage level to a binary number) than the successive approximation type of A/D. For this reason I decided to use a successive approximation (abbreviated SA) converter in the system. Figure 3 illustrates a diagram of the SA converter I designed.

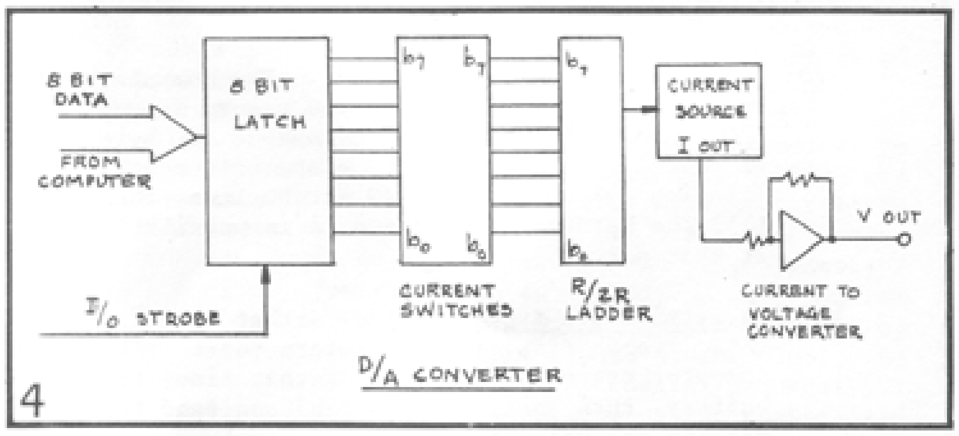


3 SUCCESSIVE APPROXIMATION A/D

The analog voltage to be converted is applied to the point labeled "voltage in" (in my system this point would be fed by the LP filter's output). The clock input is a continuous string of pulses that determine the time it takes to perform one conversion. During the first clock pulse the SAR (successive approximation register) causes it to produce an output voltage. The comparator then compares the DAC's output to the input voltage and pulls the D input of the SAR high if the DAC's output is greater than the input voltage. If the DAC's output is less than the input voltage, the comparator pulls the SAR's D input low (to the zero state). The SAR then examines its D input and if it is high (indicating DAC output is too high), it resets bit 7 to zero and sets bit 6. If D was low, the SAR would leave bit 7 set and preceed to bit 6. This process continues for each bit down to bit 0 at a rate of one bit per clock pulse. On the ninth clock pulse EOC (end of conversion) is pulsed which loads the data word on bits 7 through 0 into the eight bit latch. When the computer pulls the I/O strobe low, the buffers are turned on and the contents of the latch are transferred to the computer. Thus the data is latched and the A/D can be allowed to run freely. In this fashion the computer can read the data whenever necessary (as long as the data is read before the A/D writes new data into the latch). To ensure that this occurs, the A/D's clock pulse train is derived from the same oscillator that drives the computer's phase two clock signal. Although the A/D's clock is nine times as fast to offset its nine pulses/conversion, the net result is that the computer and A/D run in sync with each other so that no data is lost.

The next step in the system is the computer itself. For my experiments I utilized a Motorola 6800 processor with 12K of 450 nanosecond RAM, a 300/ 600 baud cassette interface, a Pertec FD200 minifloppy disc drive, and a 1200 baud ASCII terminal with full cursor control. The operation of the computer hardware would require far too complex a discussion for a paper of this brevity; however, the operation of the software will be explained in the next installment of this article.

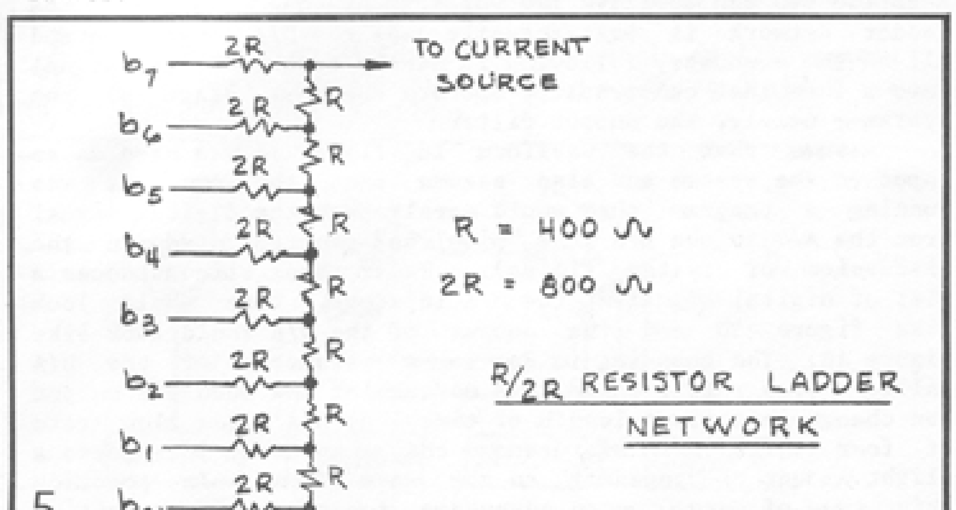
The computer's data output feeds the D/A converter (actually the second D/A converter in the system, for one D/A was part of the A/D). Refer to figure 4 for a diagram of the D/A.



4 D/A CONVERTER

When the computer outputs data to the D/A, the particular I/O strobe on that port goes low and the data is latched. The purpose of the latch is to hold the data so the computer can turn its attention elsewhere. In this manner the computer can send a number to the D/A which will be converted into a voltage and said voltage will remain at the D/A's output until the computer sends a new number to the D/A's data latch.

The output of the data latch feeds a set of eight current switches which serve to convert the digital voltage levels into very well defined current levels. The reason for this is the R/2R ladder. This ladder is a resistor network which divides and sums its eight input currents in a binary weighted fashion. For example, suppose an input of Y current on bit 7 of the ladder produced an output current of X. Then the same Y current applied to bit 6 would produce an output of X/2. Y on bit 5 would produce X/4 output current and so on down to bit 0. Figure 5 shows a schematic representation of the R/2R ladder.



5 R/2R RESISTOR LADDER NETWORK

# GLOSSARY

The following is a list of technical terms used in this article with which the reader might not be familiar. The words are listed in order of their first appearance in the article.

real time: computer processing data as the data is generated, as opposed to storing it on tape for later use.

sampling theory: the branch of mathematics which deals with taking a limited number of samples of a continuously changing set of variables.

machine cycle: a single machine operation performed by the computer. Each computer instruction consists of one or more machine cycles.

bit: a single binary digit. The computer used in this project worked with groups of eight bits (called one byte) designated b7 (bit 7) through b0.

high (also called set): when one bit is made to be a 'one' instead of a 'zero' in binary.

low (also called reset): when a bit is made to be a zero instead of a one in binary.

I/O: abbreviation for input/output. The computer used in this project employed memory mapped I/O, meaning that to input or output a byte it merely stores it in a specific memory location called a port. I/O strobe lines then tell the hardware that the data is available at that particular port.

buffers (tri-state type): hardware that isolates data lines from the computer until the computer requests the data. At that time, the buffers turn on (like a switch) and send the data to the computer.

software: computer programs.

monitor firmware: a computer program that is always present in the computer. The monitor is responsible for running the computer and all its associated equipment such as a disc drive, terminal, and tape recorders, etc.

The output of the ladder network drives a dual transistor current source whose output current (I out) varies between zero and two milliamps in direct proportion to the ladder's signal. The current source feeds an op amp stage which acts as a current-to-voltage converter whose output swings between negative two and positive two volts. As one can see, the ladder network is what actually does the D/A conversion and all of the circuitry following it merely transforms the signal into a form that can properly operate the next stage of the system-- namely, the output filter.

Assume that the waveform in figure 1A was used as an input to the system and also assume that the computer was running a program that would merely pass the digital signal from the A/D to the D/A (the "pipeline" program used in the discussion of system timing). Under these circumstances a plot of digital signal at the D/A input vs. time would look like figure 1D and the output of the D/A would look like figure 1E. The rounding is due to a parameter of the D/A called "slew rate", which is a measure of how much the output can change in a given length of time. My D/A has a slew rate of four volts per microsecond and, therefore, introduces a slight amount of "rounding" to the wave form. In practice this type of error has an advantage, for the rapid transition from one voltage level to the next in the D/A's output will contain high frequency harmonics which tend to add hiss and buzz to the sound output. This type of distortion is called quantization error and reducing this type of error by removing the high frequency harmonics is the output filter's job. The output filter is a low pass type and is similar to the input filter except that the output filter is a 3 stage design rather than two stages as used previously. As discussed earlier, the overall frequency limit of the system is dependant on the program that the computer is running. For this reason both filters have an adjustable cutoff point so their performance can be optimized for whatever function the system is performing.

The input and output filters were assembled on homemade printed circuit boards and the A/D was assembled using wire wrap techniques and perf board. The three aforementioned circuits were then mounted on a piece of plywood and all controls (filter cutoff and A/D calibration) were mounted on a steel front panel. Data, I/O port control, and clock signals were coupled to the computer through a twelve conductor cable. The D/A was assembled using wire wrap on a perf. board and sockets were mounted on the perf board so it could be plugged into the computer. The D/A's output is coupled to the outboard filter by means of shielded audio cable.

The only additional circuitry is an op amp type preamp feeding the input filter so that a microphone could be used directly with the system, and a two transistor audio amp coupled to the output in order to drive a pair of eight ohm dynamic headphones. This circuitry was also constructed on homemade printed boards and said boards were mounted on the plywood base.

In my description of the hardware I developed I tried to keep the discussion at a function block level thus avoiding the use of schematics to reduce the technical complexity. I feel that if the reader wished to duplicate this project, he could (with a little electronics knowledge) construct similar equipment. If not, most of the equipment used could be replaced with devices available on the commercial market. In the next issue, part two of this article will turn to an explanation of some of the software I have developed and how it functions in the system.

Not because we say so, but because they say so. . .

# "The *'New Yorker'* of audio magazines" —ESS, Input, Sacramento, CA

**Audio Amateur** is a magazine that continues a great American tradition—a tradition that loves tinkering and experimentation and embraces rather than eschews technology. Readers of this magazine, I suspect, don't simply discuss the latest heavily advertised "quantum leap" forward. **TAA** subscribers are impressed more by an interesting project they can build from scratch. They love to extract, by modification, the greatest possible perfection from classic and recently introduced audio products.

Like the **New Yorker,** the **Audio Amateur** publishes articles that are measured and thoughtful, articles that are beyond superlatives by the bushel basket found in most of the mass circulated audio magazines. The reasoned tone results in part from the considerable contributions made by English writers, including the late B.J. Webb. Edward T. Dell, Jr., the editor, almost always includes a thoughtful editorial that, alone, is worth the cost of admission. Unlike some of the little audiophile magazines, **TAA** is generally beyond clannish allegiance to a few manufacturers. Articles on projects to construct and modify appeal to the fondness of its readers for a wide range of projects.

**Audio Amateur** has served up a smorgasbord of projects over its ten year existence. How to properly adapt a Grace arm to an AR turntable, build a record cabinet, modify a Formula-4 tonearm to improve low frequency reproduction, or build a 10 dollar three-element Yagi antenna have all been offered as appetizers, projects that require some familiarity with tools and a few nights of your time. The main course offerings demand various degrees of more sophisticated electronic skill. If you've only assembled a one tube radio (twenty years ago), many of the electronic projects are going to more than you can chew. Numerous past articles have shown how to improve classic Dynaco products. Recently, Nelson Pass of the Threshold Corp. discussed how to build a 40 watt per channel class A amplifier. Electronic articles typically assume an ability to find the parts necessary to build the projects. Chances are you'll spend some time searching through parts catalogs and local surplus houses before you can begin to wade into the actual construction.

Sophisticated articles that examine specific audio problems but do not involve building projects also abound. Walt Jung, contributing editor, has discussed slewing induced distortion in amplifiers in a series of articles. How we actually perceive sound and how many speakers may be necessary to recreate the closest possible approximation of the live event has also been discussed.

If speaker building is your forte, past articles have dealt with horn loaded and transmission line designs. Instructions on how to build electrostatic transducers from scratch, and box fabrication for sub-woofers with an accompanying active crossover have also been features. It's a measure of **TAA** contributor ingenuity that a complex driver like the Heil air-motion transformer has been built by an amateur — complete instructions on how to build a home version of the large Heil appeared in the magazine in 1977.

An excellent analysis of recently introduced audio kits is a regular feature. Kit reviews are technically very thorough and are often more objective than you find elsewhere. A regular feature, "Audio Aids," offers all kinds of informative hints from readers. A letter section from readers comments on past articles and present concerns and lends a thoughtful and inquiring tone to the magazine. Advertisements, themselves, are often helpful to the reader since many of the ads list parts that are vital for project construction. Most of the better kit manufacturers also advertise in **Audio Amateur.**

If you are already an audio craftsman, or would like to become one, **Audio Amateur** is an excellent touchstone. For less than the price of a good meal and a movie ticket, you can receive four issues a year.

—**George Hortin,** Staff Writer
**INPUT,** published by **ESS, Inc.**
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"...the sequencer comes of age..."

# ACT: an Automatic Composition Translator

by Roger Powell & Mark Styles

ACT, an acronym for "Automatic Composition Translator", is an interactive program designed to generate harmonic and percussive note patterns in real time. The composer can easily build and manipulate sequences to create a score which can be stored in computer memory. These scores can be saved on cassette for future processing or performance.

Diagram #1 is an overview of this system. ACT utilizes an IMSAI 8080 microcomputer, Cromemco's seven channel A/D - D/A, and Processor Technology peripheral cards. Photo #1 is a close up of the IMSAI 8080. The whole system was built from standard kits. ACT is written entirely in 8080 assembly language and occupies approximately 2K of memory.

A standard crt monitor is used for display purposes and assorted synthesizers generate the sound. The record switches and LED byte display are part of the IMSAI's front panel. Presently the system is configured for four simultaneous voices but could be expanded to seven voices before more hardware would be required.

Photos #2 and #3 show some of the equipment used in the studio. We have made up three junction boxes for connecting the IMSAI to the synthesizers. These boxes have DB - 25


PHOTO #1 - Close up of the IMSAI's internal workings. The set of eight switches to the left function as our record switches. The LEDs above these switches served as a visual reference when calibrating the synthesizers.

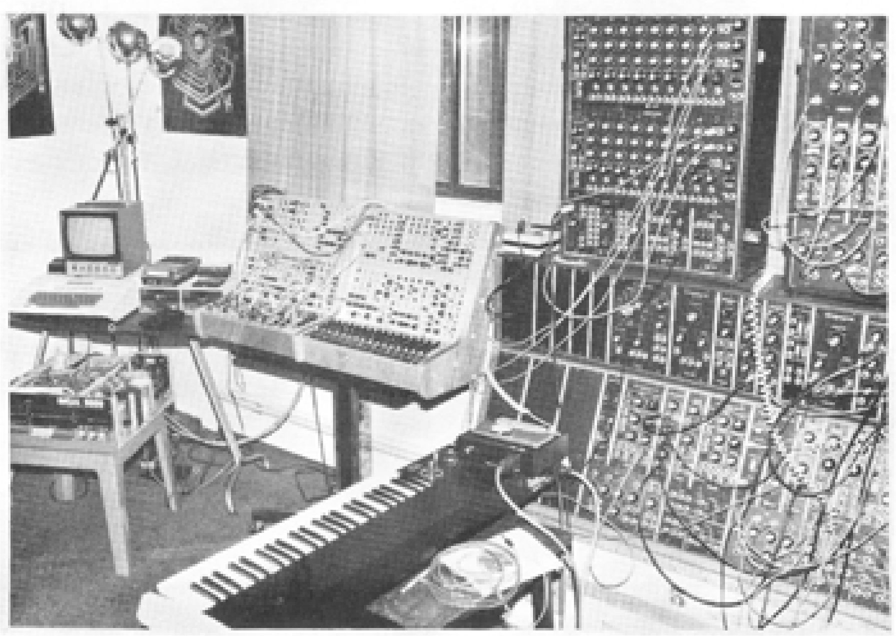


PHOTO #2 - Shown clockwise from left; IMSAI 8080, Apple II Plus with two disk drives, GBC - ITC monitor, Serge six panel custom synthesizer, Moog 55, Moog Sequencer, Moog 15 and a Polyfusion 2054 keyboard.

connectors and both phone and banana jacks to accomodate the synthesizers. Presently we are in the proccess of adapting ACT for use with the Apple.

A unique feature of the system is a series of seven real time pointers. In the computer, pointers are address values which "point" to tables in memory containing pitch and rhythm information. These pointers are directed by a series of pots on the synthesizer which attenuate the control voltages input to the A/D.

All four voices are tied to one pitch row pointer and can be set to read any one of sixteen sequences. Each voice has its own gate pointer however and can point to any one of sixteen rows. This allows different voices to play different rhythms. Since the voices are all drawing from the same pitch information, the result is an integrated, harmonic effect without the predictability one has come to expect from sequencers.

The element pointer is a sequentially incremented offset factor that is added to the pitch row pointer for each voice. With a zero offset, all voices will read in unison. With the element set to 3, the first voice will read pitch one, the second voice will read pitch four, the third voice will read pitch seven, etc. By setting the element to 2 and reading a Dorian mode sequence, one can easily block out major seventh chords in virtually any rhythm desirable.

The transpose feature is implemented from the AGO keyboard during the record and input mode. Middle "C" is located at octave 2 and may be transposed up or down two and a half octaves.

# Commands

ACT is controlled by the following sixteen commands:

| | |
|---|---|
| SYSTEM | C: Calibrate |
| | ?: Display Measure Length |
| | Phrase length |
| | Sync/Async Mode |
| | I: Input current A/D values |
| LOAD | G: (A - P)(Data)(CR) - Gate Load |
| | V: (A - P)(Data)(CR) - Voltage Load |
| SET | M: Set Measure Length |
| | L: Set Phrase Length |
| | S: Sync Voltages to Gates |
| | A: Async |
| MEMORY | X: Switch Score Memory - Main/Jump |
| | >: Increment Score Pointer |
| | <: Decrement Score Pointer |
| | O: Reset Score Pointer to zero |
| RUN | R: Record |
| | P: Play |
| | (Space Bar): Stop Execution |

# System Commands

The system commands are essentially initializing and monitoring procedures. They facilitate tuning and keep a constantly updated display of the score pointers.

C: CALIBRATE The calibrate command will gate all voices on and output the current keyboard voltage to all of the D/A's. The first step in calibrating the keyboard to the computer is by adjusting the keyboard range and scale controls. The keyboard voltage input to the A/D is quantized by masking the two least significant bits. The IMSAI's front panel LEDs will show a binary display value which increments by four for each chromatic note.

The VCOs are then calibrated to the computer. If the one volt per octave VCO frequency input is accurate, tuning the VCOs to unison should be the only adjustment required.

?: DISPLAY If "?" is typed, the computer will respond by displaying measure length, phrase length and whether the sync or async mode is currently active. Measure and phrase length both have default values. The sync/async status must be assigned by the user however prior to execution of a score.

I: INPUT When input command "I" is entered, the computer will display the continuously updated values of the pointers. The user can then adjust the control pots that set the pointers to the desired values. This command is primarily used to match the current score pointers to previously recorded values.

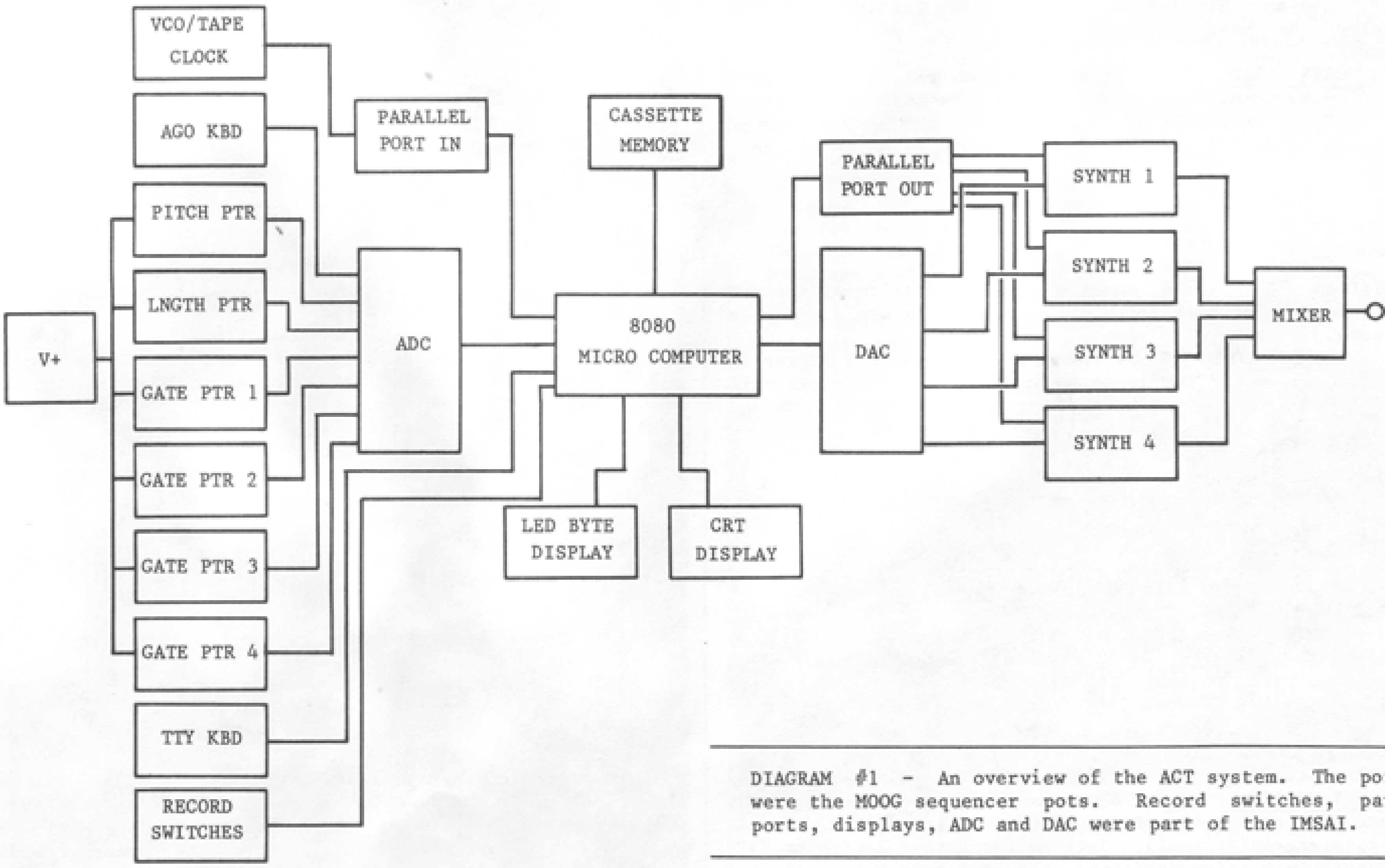



DIAGRAM #1 - An overview of the ACT system. The pointers were the MOOG sequencer pots. Record switches, parallel ports, displays, ADC and DAC were part of the IMSAI.

PHOTO #3 - Shown here is an Echoplex, Tapco and Techniques equalizers, Teac mixer, Eventide Instant Flanger, Teac 80 - 8, Sony TC - 880 two track, DBX noise reduction and miscellaneous test gear. A Yamaha 12 input mixer is also used.

## Load Commands

Once the system has been calibrated, the gate and voltage tables are loaded with data by entering a series of sequences. Each sequence may be up to sixteen events and up to sixteen sequences may be stored in each table. The gate table is loaded from the TTY keyboard and the voltage table is loaded from the AGO keyboard. The contents of these tables will greatly influence the personality of a piece.

For our experimentation, a fairly standard set of gate rhythms worked well.

Row A = a measure of rest
Row B = half notes
Row C = quarter notes
Row D = eighth notes
Row E = sixteenths

The remaining rows contained more exotic rhythm patterns.

The voltage rows contained different modal scales (Dorian, Lydian, Mixolydian, etc.). The transpose feature eliminated the need to load scales in different key signatures.

G: GATE LOAD "G" is entered by the user, followed by a letter (A - P). Anyone of sixteen gate rows may be written. Values of 0, 1, or 2 will be accepted until the length command value is reached. To quit before the length value is reached, hit Return.

0 = no gate
1 = gate
2 = hold gate till next count
(Carriage Return) = exit

Sixteen "0"s would equal a measure of silence, sixteen ones = a measure of 16th notes and sixteen "2"s would equal a whole note.

V: VOLTAGE LOAD "V" is entered, followed by a letter (A -P), signifying which row is to be written. The computer will accept values from the AGO keyboard until the length command value is reached. As each note is entered, it's hex equivalent is displayed on the crt. To exit voltage load, hit Return.

## Set Commands

The SET commands, like the load commands, determine the structural boundaries of a piece. They control the length of pitch and rhythm sequences plus the frequency with which the score is updated. The set commands also determine the assignment of pitch to gates for each voice.

M: SET MEASURE LENGTH The M command controls how often the score is being recorded. This parameter is the number of master clock counts that will occur before the next set of score values are written. The default value is 16 (hex 10).

L: SET PHRASE LENGTH The L command defines the length of pitch and gate rows being read. Unless specified, the default value is sixteen (decimal). Setting the phrase length to "1" would result in a voice repeating the same note continuously. Setting the phrase length to "16" would allow all sixteen positions in a pitch row to be read before repeating.

S: SYNC MODE This command will assign the next pitch only when the next gate for that voice occurs. This feature is similiar to the "multiple triggering" mode on synthesizer keyboards.

A: ASYNC MODE Typing "A" enters the async mode. In this mode the pitch pointer is incremented with each clock count regardless of whether a gate is present. This will result in several notes being heard if a long envelope is used. This is similar to the "single triggering" mode of commercial synthesizers.

## Memory Commands

There are two areas of memory set aside for score storage-- main and jump. Main is the actual score memory which is used to record and store the piece. Jump memory is a practice area where the user can experiment with pointer settings. The memory commands enable the composer to easily manipulate the score pointer and thus edit the score.

Memory commands increment, decrement and reset are operational only in main memory.

X: EXCHANGE The X command exchanges score and jump memory. Consecutive "X" commands will toggle back and forth between main and jump memory.

>: INCREMENT This will increment the score pointer one measure each time the right arrow is struck.

<: DECREMENT The left arrow will decrement the score pointer one measure each time it is struck.

0: RESET Typing "0" will reset the score pointer to zero.

## Run Commands

The run commands are used after the pitch and gate tables have been loaded. The program will produce music only when run or record has been entered and a clock is being received by the computer. Execution will begin at the current score pointer. The clock may be either a VCO or a taped click track.

R: RECORD Record will update all channels which have their record switches activated (located on IMSAI front panel). All pointer values present on the crt display are stored while in the record mode. Channels not in record will play from the score.

P: PLAY Play will execute whatever is in memory assigning the appropriate gate and voltage row to each voice for all channels. The record switches are inoperative during the play mode.

(SPACE BAR): STOP The space bar will stop execution of either record or play and reset the score pointer to the beginning of the current measure.

## Keeping Track

ACT uses an external click track to drive it's counters. At the computer, the clock input is one bit of a parallel port. The signal presented to this input should conform to standard logic levels of 0 and 5 volts. Because ACT processes data during both the negative and positive clock periods, a square wave (50% duty cycle) is ideal. A reliable method is to gate an audio sine wave (1Khz) by a low frequency square wave (refer to John Duesenbury's article on "Prerecorded Timing Signals", Polyphony, Volume 5 Issue 1, May/June 1979). The frequency of the square wave can be determined by the following formula:

$$(\text{Tempo} \times N) / 60 \text{ sec.}$$

where N = number of increments per quarter note. For example:

1 quarter note = 120 beats per measure
N = 4 sixteenth notes
(120 X 4) / 60 sec = 8 Hz

To record the synthesizer audio parts on multitrack tape, the clock must be recorded on one track of the tape for the full length of the piece. The taped clock is converted back to a DC square wave and sent to the computer during recording. Further "passes" can thus be synchronized.

ACT utilizes eight independent record switches, one for each parameter on the display. This allows the composer to make modifications to individual parameters in the score quickly and easily. With ACT, it is a simple matter to drastically alter the structure of a composition without wasting hours of recording time.

A score is written whenever the record mode is entered and the clock is being received. Once each measure all score paramaters, displayed on the screen, are stored. This score, as well as pitch and gate table information, may be dumped on tape or sent to a printer for documentation.

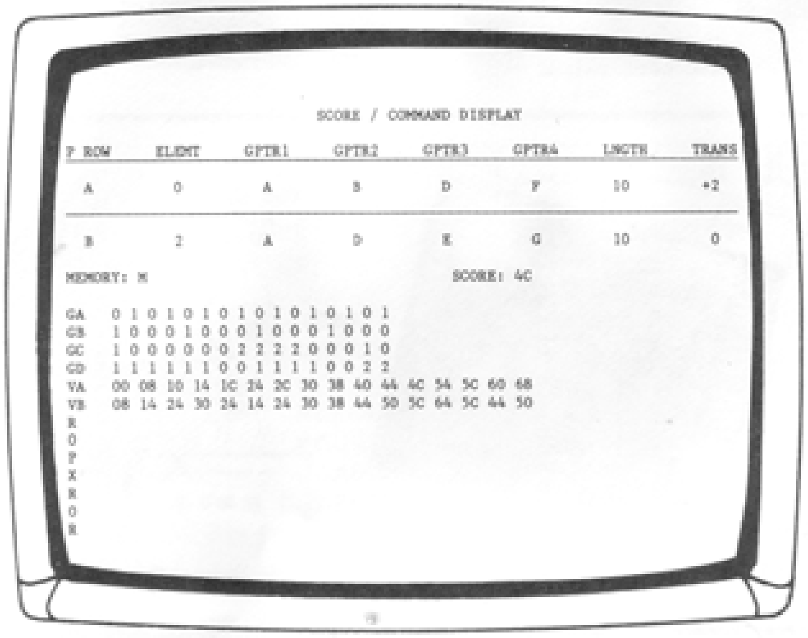



DIAGRAM #2 - A representation of the crt display. The lower half of the screen will display commands entered by the user.

Diagram #2 is a representation of the crt. The recorded score values are displayed on the second row and the third row shows the values of the pointers at the beginning of the

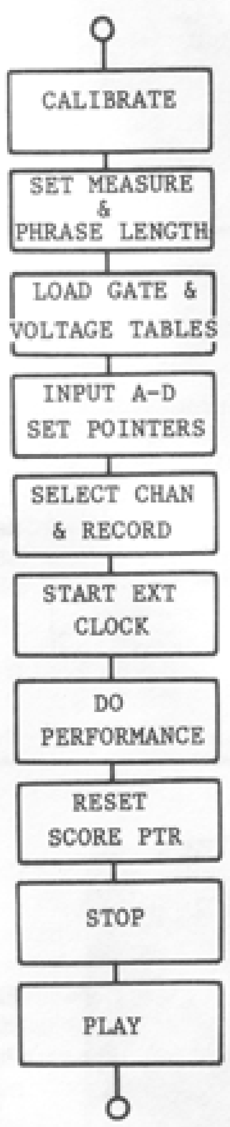



DIAGRAM #3 - Typical procedure for utilizing the ACT system.

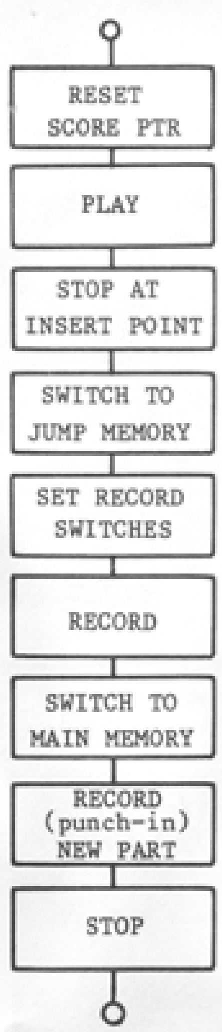



DIAGRAM #4 - Procedure used to overdub (punch-in) a new part.

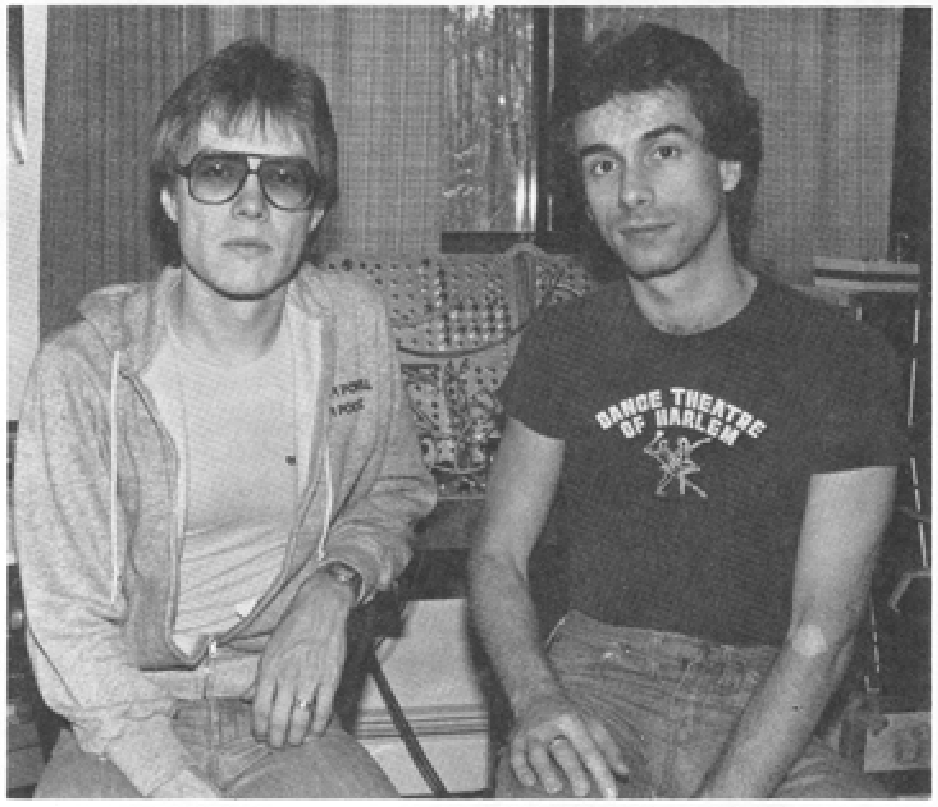


PHOTO #4 - Roger Powell and Mark Styles

current measure. Values are updated only at the beginning of a measure. The frequency with which these values are updated is indicated by the measure length.

Illustrated in Diagram #3 is the procedure generally used in running ACT. First the system is calibrated, then the measure and phrase lengths are set. Next the gate and voltage tables are loaded. The pointers are set by invoking the Input command. After the record channels have been selected and record is entered, start the external clock and begin playing. Stop the program by hitting the TTY space bar, reset the score pointer and execute the Run command to play back.

Diagram #4 demonstrates the procedure for overdubbing a channel. Once the desired score location is reached, switch to jump memory (X Command). Activate the appropriate record switches and begin recording. When the pointers have been adjusted to produce the desired results, switch back to main memory and record the new part.

piece, stop the score at the appropriate spot, and type "I" (input). Now the third row will reflect real time pointer values as they are being changed via the control pots. Simply match these values to the second row score values. When satisfactory settings have been acheived, increment the score pointer to the desired measure location and punch in the new part.

As one becomes familiar with ACT, the far reaching implications become more evident. One could easily create and store compositional "music beds" or ideas for future development. Major alterations no longer mean rerecording a piece. Also new insight can be gained on the structure of music itself.

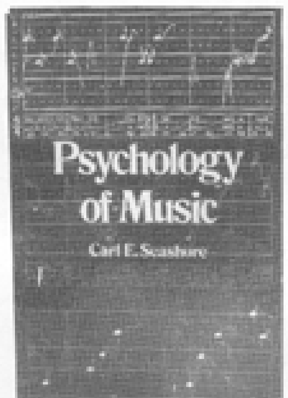
Using ACT, one could compose the basic tracks for a whole album at home, taking as much time as neccessary to work out harmonic and rhythmic parts. In the studio, once the desired synthesizer voices have been patched, the basic tracks for all the tunes could be recorded.

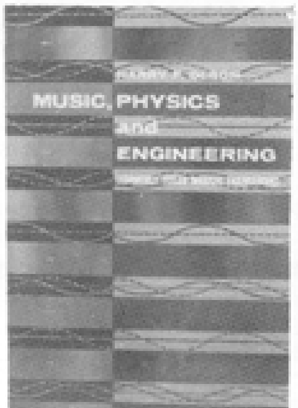
Since the control tracks have already been stored in computer memory, one could use the entire synthesizer for the generation of one layer of sound if desired. Thus by recording one track at a time, truly sophisticated and subtle results could be obtained.

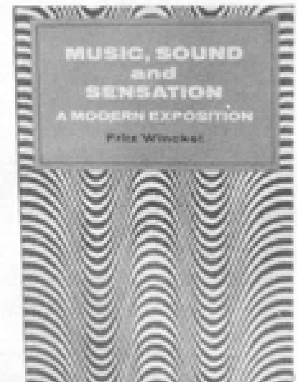
ACT offers the composer/synthesist a useful system to create and document pieces. It is centered around analog sequencer techniques that have been expanded to include the resources of the microcomputer. The entire system could now be purchased for approximately $2000 - $3000, assembled, depending on amount of memory and choice of peripherals.

Aside from being a powerful compositional tool, ACT's hardware also is a sophisticated micro computer system. The applications and uses of this system outside of the music realm are almost unlimited. ACT is a definite consideration for the synthesist who wishes to keep up with the fast moving technology of today.

# POLYMART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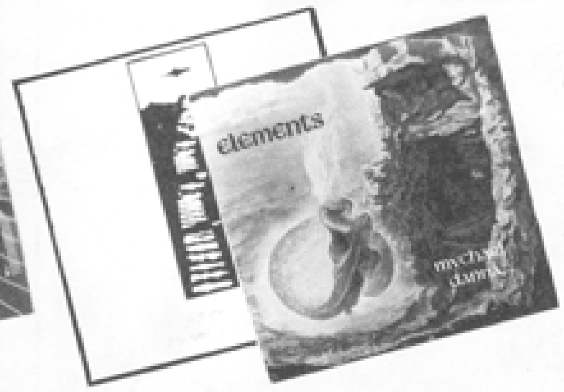



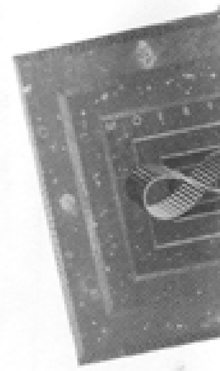

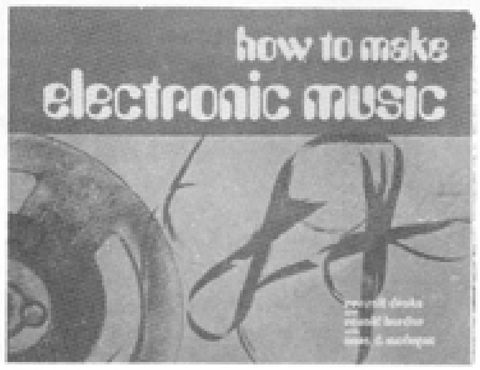



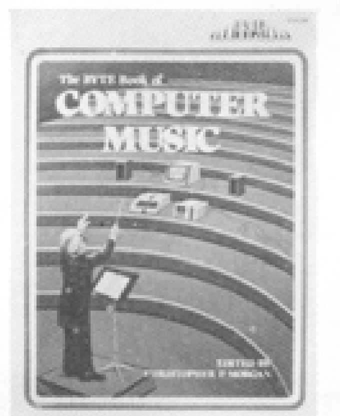


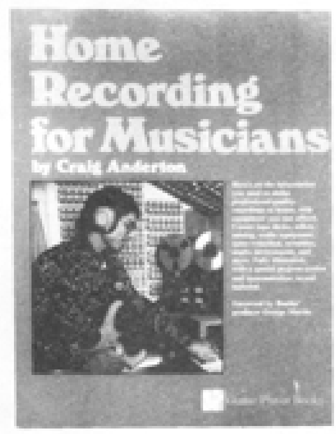


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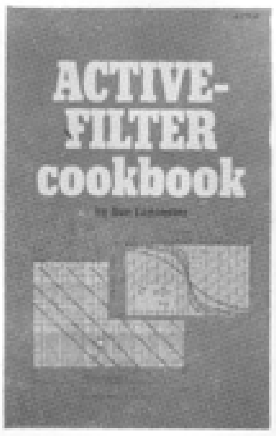


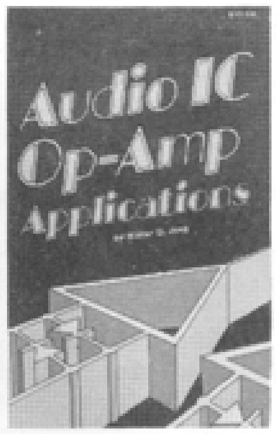


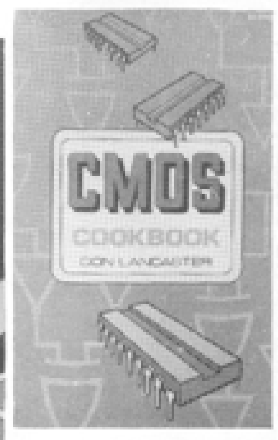



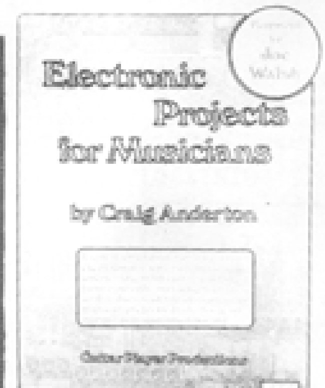


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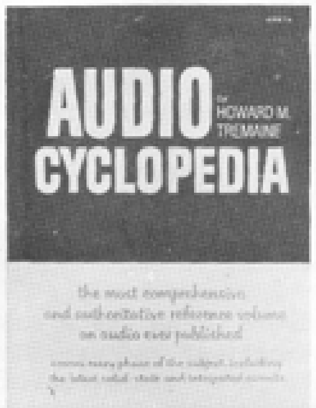


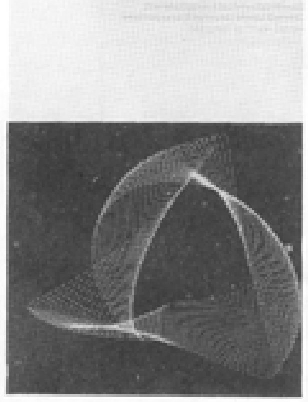

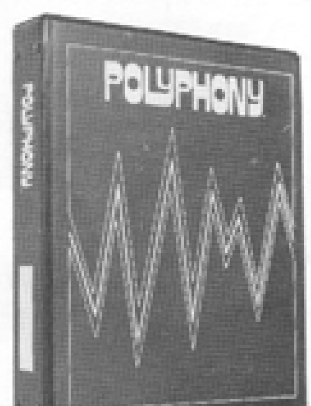


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☐#0302: November 77: The Sensuous Envelope Follower, digital gates, LED wall art, build a bionic sax, data to music peripheral project, Apple II as a music controller, using the NE566 as a VCO, patches.

☐#0303: February 78: computer controlled Gnome, using joysticks, build a bionic trumpet, octave controllers for bionic sax and trumpet, ultra-VCO modifications, voltage control the Mu-Tron Bi-Phase, oral joystick, patches.

☐#0304: April/May 78: Minimoog modifications, non-keyboard module use, phasing and flanging (theory and circuits), memory expansion for programmable drums, digitally addressed transposer project, polyphonic software (with software transient generators), patches, Vol. 3 index.

☐#0401: July/August 78: analog delay lines (theory and projects), composing for electronic music, note to frequency (and vice versa) conversions, build a trigger delay, software for computer composition, low cost VCO circuit, patches.

☐#0402: September/October 78: electronic music notation, notes on the recording of "Cords" by Larry Fast, sequencer software- part one, rhythmic control of analog sequencers, touch switch projects, modular vocoder techniques, PET as a music controller, patches.

☐#0403: November/December 78: multi-purpose keyboard software, Sohler keyboard and notation system, voice frequency to voltage converter project, proposals for tape exchange, VCA project, sequencer software- part two, frequency balancing in recording, Barton and Priscilla McLean.

☐#0404: January-March 79: add-ons for vocal F to V converter, shorthand patch notation, more on note to frequency conversions, graphic monitor project, George Russell, super VCA circuit, echo software, Vol. 4 index.

☐#0501 May/June 79: using click tracks, PET music software, clockable sample/hold and noise source project, voice processing patches, VCF circuits, profile of John Cage, linear DAC.

☐#0502 July/August 79: hex VCA/mixer project, electronic music schools and studios, modify the Oberheim Expander Module, profile of Ernest Garthwaite, budget microphones, digitizer projects and software, bar graph ICs.

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# INDUSTRY REPORT:

*'Tell Them You Saw It In Polyphony'*

## PANEL LIGHTS

Littlites by Custom Audio Electronics are now available for illumination of control panels and work spaces in dimly lit areas. These slim and stylish gooseneck lamps are available in black finish in 6, 12, and 18 inch lengths. Complete with bayonet type buld and 360 degree swivel base, the lamps are durable and detach easily from base connector for storage. Various mounting brackets, power supplies, and dimmer controls are available to facilitate the addition of these devices to new and existing equipment. A kit including lamp, base, dimmer, power supply, cable, and mounting hardware is also offered.

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# SYNTHESIS RESOURCES

ELECTRONIC MUSIC: Resources for Performance Groups and General Music Classes is a reference book containing a vast listing (over 50 pages) of electronic music resources. The book is divided into four main sections: A Comprehensive Discography; Electronic Music Scores for Live Performance (includes addresses of publishers); A Bibliography of Books, Periodicals, Articles, Papers, Student Textbook Sources, and Multi-media Resources; and a Supplement that includes titles of periodicals in which electronic music articles frequently appear. Considering that this book has a copyright date of 1979, the listings are far from complete or up to date, but the preface states plans to update the listing periodically and welcomes corrections and additions from readers. If this is followed through, future editions should promise to be a bit more useful. Nevertheless, for the low cost there is a lot of information listed here which should keep avid synthesists busy for many a night.

"Electronic Music" is available for $3.75 per copy from: Cross Creek Press, PO Box 12533, St. Louis, MO 63141. JCF

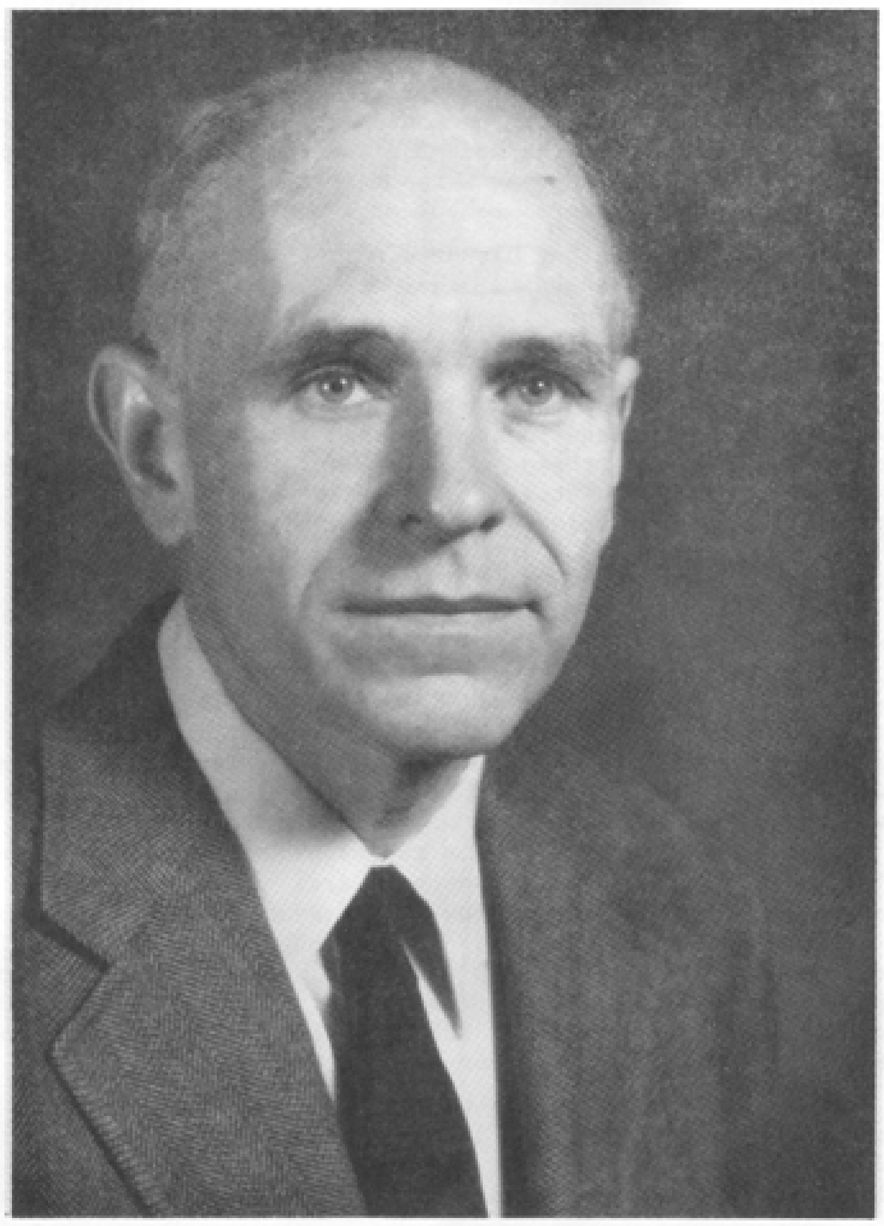
# MAX MATHEWS
# Computer Music From Bell Labs

## by Don Slepian

Computer music was first heard in a tiny room lost amidst the giant research complex of Bell Telephone Laboratories in Murray Hill, N.J. In attendance were a small group of engineers, who considered it "an interesting byproduct" of their more serious endeavors. Like any infant, the first cry was primitive. The year was 1957, when most electronic music composers were splicing tape and coaxing music from test oscillators. Today most new synthesizers are made with a computer, usually a microprocessor, that either stores and remembers patches or actually generates control voltages, gates, and triggers. They are hybrids, analog synthesizers under digital control. After 23 years computer music, that is music made using only a computer and a digital to analog converter, is still in its infancy. During the course of my work I took the opportunity to interview Dr. Max V. Mathews. Dr. Mathews wrote the original computer music programs, and continues to guide and inspire others in this field.

djs: What were the steps between the digital speech encoding work you were doing in the mid 1950's and the first computer music program?

mvm: Well, once we had made the equipment to put speech into a computer and take speech out of a computer we knew that we could write a program to create music. We had always felt that existing instruments were difficult to play, and that it would be nice to make an instrument that was both easier to play and more powerful in many senses than existing instruments, so that hopefully one could make better music. This music would be more valuable to the listener and more expressive to the performer and composer. The particular incident that started it was listening to a piano concert with my colleague John Pierce. We heard some Schoenberg, which wasn't bad, and some other modern pieces which we thought were pretty terrible. During intermission we decided that the computer ought to be able to do better than this. I wrote a program, "Music I", which had very limited capabilities. It created a monophonic instrument with a triangualar waveform, and offered a choice of pitch and note duration. A very brave psychologist, Dr. Newman Guttman, wrote a piece for Music I, and we saw the limitations and deficiencies of that particular way of making music. The limitations were overcomable, and further efforts led to Music II, III, IV, and V. The original programs only ran on large, expensive computers. People could only get their hands on one if they worked for some institution that was using it for some other purpose, and then they had to come up with a fair amount of money or "funny money" (bureaucratic juggling) to pay for the computation time. Despite all these obstacles some interesting music was made.



djs: What led up to that 1959 record, "Music From Mathematics"?

mvm: There were actually two records called "Music From Mathematics": a ten-inch disk issued privately by Bell Labs to show that the computer could be used for this purpose - they developed the computer for the speech technology but this was such an interesting byproduct that they wanted to demonstrate it - and a regular commercial disk issued by Decca (note 1) containing some additional pieces. The purpose was to document and make available to the public the initial computer music.

"The particular incident that started it was listening to a piano concert- During intermission we decided that the computer ought to be able to do better than this."

djs: I got my first copy when I was 8, and I wore it out from constant use. I still enjoy your "Numerology"; I grew up with that record. What program were you using at that time?

mvm: We were using Music III, which had almost all the potential of the present day program, with a little less convenience. One of the real problems was to give the composer great freedom to do whatever he wanted to do but at the same time help him in doing this so that he did not have to start from scratch in writing his instruments and other programs to create his music. It is difficult to help a person without constraining him. The way I attempted to do this in the music programs was to make building blocks, called unit generators, that the composer could link together in any way he chose to form his instruments. I tried to give him building blocks he would be somewhat familar with, such as oscillatiors, mixers, and filters, so that he would have some idea of what they might do. I did not make predefined instruments that he would have to use in making his composition. Looking back on it I feel the building block approach was a very good idea, and most of the other music programs have followed this concept.

djs: What role do you see for Music V today, now that we have real-time digital synthesis?

mvm: Originally I thought that the real-time equipment would completely supplant programs such as Music V, but now I feel otherwise. Playing things in real time is both a satisfaction and a limitation. It's nice to be able to express yourself, but great demands are made upon you as a performer, both in terms of virtuosity (wiggling your fingers fast enough) and control of timbre, dynamics, and all the other aspects of composition. It is easy to imagine a compostion which saturates the ability of a single performer to create in real time. Most interesting music has always been the creation of groups of performers.

djs: Or an individual using multi-track tape...

mvm: Right. An alternative to coordinating groups of people is to work in non real-time where you can fabricate each sound individually and store it in digital memory, which is a more facile recording medium than multi-track tape. You have lots of time to agonize over each nuance of the sound, and this kind of careful work really pays off.

"It is pretty clear that digital equipment is going to supplant the analog, since it is more reliable easier to maintain, and won't cost any more than today's methods."

djs: Where are computer music programs currently in operation?

mvm: There are enough places now so that I can't remember them all. Stanford of course runs Music 10 (an advanced version of Music V that runs on the PDP 10), and there are well developed programs at MIT, Colgate, Princeton, and Columbia. Many schools and universities have set up their own programs, and this is getting easier to do as these programs can effectivly run on small computers.

djs: Is it possible to run a computer music program on a micro system such as the Apple, Pet, etc?

mvm: I have already heard Laurie Spiegal (New York City electronic music composer) play interesting music on her Apple computer. I don't think you could do any Music V kind of program on a microproccessor alone. You need to have a digital storage medium that can play back the samples fast enough to convert them to sound at the normal sampling rate. Hard disks are coming out that are not much more expensive than floppies; I would think that you could make music with a micro system and a hard disk.

djs: What sort of micro system would be most suitable?

mvm: I don't think it's worthwhile running music programs on less than 16 bit computers, especially now that 16 bit micros such as the Motorola 68000 are cheap and widely available (note 2). The 16 bit DAC (digital-to-analog converter) is currently the standard of excellence.

djs: How did you first become involved with IRCAM?

mvm: Jean-Claud Risset (talented music composer) came here and worked with me, and when he went back to France he became the head of one of IRCAM's divisions. He suggested to Pierre Boulez that I be recruited as Scientific Advisor. Most of my work was in planning the technical facilities, and I've been influencial in bringing a lot more computers and digital technology into the project. They have a large PDP 10 as their main Music V machine, and a number of smaller PDP 11's for sound analysis and real-time programs. They have a very heavy program in the construction of digital instruments. Luciano Berio (electronic music composer), who is also in charge of an IRCAM department, originally wanted a very large analog synthesizer with a thousand oscillators! I persuaded him that he should have digital oscillators, where a small amount of equipment could make the signals that would come from a thousand analog oscillators. We recruited a physicist, Pepino di Giugno, who is continuing to design and construct this sort of equipment.

djs: How would you compare the real-time digital synthesizer produced at IRCAM with the one here at Bell Labs? (note 3)

mvm: Pepino di Giugno and Hal Alles (Bell Labs engineer) worked together for a month or so and produced the basic building block for both machines, a digital oscillator time-multiplexed 64 ways (i.e. 64 digital oscillators) on a single 8" X 10½" card. (note 4) Subsequently Alles returned here and produced the design that is being marketed by the Crumar company (Music Technology, Inc.). While the IRCAM machine is more flexible and programmable, both machines are far from any sort of ultimate. They produce their envelopes by a series of linear ramps, and the control computer has to initiate a new ramp each time an old one ends. This is a lot of work for the computer and a lot of work for the composer who has to instruct the computer, as you well know. The machines that have been produced so far are harder to control than we had hoped they would be.

djs: In many cases I've had to approximate an exponential envelope with a series of ramps. The end of each ramp creates an interrupt (signal to the computer), so in situations when I am playing chords against the sequencer I've created enough interrupts to swamp the software and slow down the sequence! This problem will disappear with faster software. (note 5) What about the composers' school at IRCAM?

mvm: To my mind that has been pretty successful. They run three or four classes a year of 6 to 8 composers as a full-time school. The composers get plenty of time with the equipment, and they start from the basics of computer music score writing and manipulation. They get lots of work with timbres and compositional algorithms, and they get some time with the real-time equipment. I heard some of the student compositions from these classes, and they were just great, ...real musical innovations. They had obviously learned a great deal in that relatively short amount of time.

djs: How would one go about getting into this program?

mvm: You would write to IRCAM, and go through their application process. There are a lot of people who find the computer glamorous, and would be happy to play with it, but in most cases these people don't have enough background, either in technical directions or in music, to justify their work. The serious people who are already competant musicians and have had some background in both programming and electronic technology, ...well, there aren't many of those.

djs: There seems to be two main thrusts to the present "state of the art" synthesizers: real-time digital synthesizers such as the Crumar, the Synclavier (note 6), and the Fairlight CMI (note 7), and computer-controlled analog equipment, either program patched such as the Oberheim polyphonic and Roland's "Jupiter 4" or micro-controlled such as Sequential Circuit's "Prophet", E-Mu modular, and Paia's P-4700/J. What trends do you see for the future?

mvm: It is pretty clear that digital equipment is going to supplant the analog, since it is more reliable, easier to maintain, and won't cost any more than today's methods. I look forward to real-time digital synthesizers that are capable of being controlled by score programs such as Music V. I feel a score, that is a moment-by-moment description of the music in the computer memory, will greatly simplify the production of complex, rapidly changing live electronic music performance.

djs: A performer invoking and interacting with an electronic music score live in concert?

mvm: Yes, I think that will be the next big step.

NOTES

1. Decca DL9103 "Music From Mathematics".
2. This is refering to programs that actually generate sound. Eight bit micros are well suited to controlling analog synthesizers.
3. See Computer Music Journal, Nov. 1977, pg 5 for a description of the Bell Labs machine. Computer Music Journal is published by People's Computer Company, Box E, 1263 El Camino Real, Menlo Park, CA 94025.
4. Ibid, pg.7
5. The Crumar Machine has this improved software and many hardware improvements. Such problems are typical of a research machine, not a commercial product. The Crumar is manufactured by Music Technology, Inc., Long Island.
6. The Synclavier (tm) is manufactured by the New England Digital Corp., P.O. Box 3065, Norwich VT 05055 (802) 649-5183.
7. The Fairlight CMI (Computer Musical Instrument) is manufactured by Fairlight Instruments Pty. Limited, 15 Boundary Street, Rushcutters Bay, Sydney, Australia 2011. The US tech. rep is Bruce Jackson, P.O. Box 401, Lititz, PA 17543, write for information.

# QUA4:

## QUA4: An Advanced STG Control System

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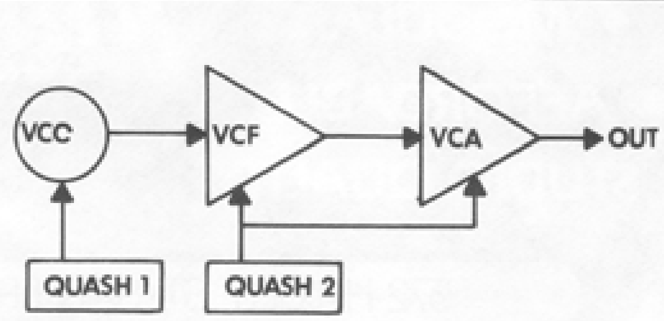
by David Ernst

"QUA4," a modification of "MUS1," provides multiple Quash control channels for each voice, and it may be used either as a monophonic or polyphonic control program. This article is divided into three parts---software, hardware, and program listing---so that interested readers will be able to follow the logic and hopefully begin to write their own programs. Before writing your own programs, however, you must know how to program the 6502 microprocessor in machine language, in addition to understanding the operation of "MUS1." This is not as complex as it may appear, and some helpful reference books are listed in the bibliography. The ability to program your computer---to make it do what you want it to do---is just as necessary as programming your conventional synthesizer. Computers and synthesizers are simply 'tools of the trade' that must be mastered in order to obtain 'musical' results.

## Software

Upon using "MUS1" in compositions I found that I liked to employ the STG's quite frequently, but I was also sending the

1



control voltage from Quash 2 to a multiple so that it could activate both a VCA and a VCF (see Figure 1). By attempting to alleviate this minor inconvenience via software I developed the algorithm for "QUA4," which consists of two sections. The first part simply reassigns the sequence of Quash channels with respect to NTBL (NTBL. ODF), and affects the POLY/NWKY (OD71) routine of "MUS1." The output from Quash 1 (NTBL, ODF) is retained as a pitch control voltage, and the output from Quash 2 (NTBL, ODE) is retained as a STG control voltage. But, according to "MUS 1" the outputs from Quash 3 (NTBL, ODD) and Quash 4 (NTBL, ODC) repeat the pitch-STG control voltage sequence, so this is the point at which program modification begins. I decided to reassign the Quash channels so that Quash 3 (NTBL, ODD) and Quash 4 (NTBL, ODC) are reserved for operations are slightly more complex than those for POLY/NWKY. First of all, after the initial STG value has been loaded into Quash 2 (PARM, ODE; identical to NTBL, ODE) it must be saved in the stack via the PHA command. The complementing operation then takes place (EOR $FF), followed by DEX to reach the appropriate PARM (ODD) location for the output of Quash 3. This is followed by a brief check of the System Control Word (OE8) to verify that bit D7 has been set. The output of Quash 3 is then modified (EOR, $AA), followed by DEX to reach the appropriate PARM (ODC) location for the output of Quash 4. Finally, the original STG value (Quash 2) is retrieved from the stack by the PLA command, and the PARM counter is decremented twice to insure the correct location (ODA) for the STG value of Quash 6.

2

| | | | | | | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| | voice 1 | | voice 2 | | voice 3 | | voice 4 | | voice 5 | | voice 6 | | voice 7 | | voice 8 | |
| MUS1 | pitch | STG | pitch | STG | pitch | STG | pitch | STG | pitch | STG | pitch | STG | pitch | STG | pitch | STG |
| QUASH | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | 9 | 10 | 11 | 12 | 13 | 14 | 15 | 16 |
| NTBL | ODF | DE | DD | DC | DB | DA | D9 | D8 | D7 | D6 | D5 | D4 | D3 | D2 | D1 | D0 |
| QUA4 | pitch | STG | STG | STG | pitch | STG | STG | STG | pitch | STG | STG | STG | pitch | STG | STG | STG |
| | voice 1 | | | | voice 2 | | | | voice 3 | | | | voice 4 | | | |

additional STG control voltages (achieved in the second part of the algorithm). A comparison between "MUS 1" and "QUA4" with respect to Quash channels is illustrated in Figure 2. Keep in mind that bit D7 of CTRL (Systems Control Word, OE8) must be set, i.e., OE8 must be loaded with either $CO or $80. Furthermore, the usefulness of "QUA4" increases with the availability of additional Quash channels: four Quash channels for one voice, eight Quash channels for two voices, twelve Quash channels for three voices, and sixteen Quash channels for four voices.

The program alteration required to reassign the Quash channels in this manner is minimal. With respect to POLY (071), two DEX commands must be inserted within the LP1 routine. Similarly, two DEX commands must be inserted within the NK3 routine of NWKY (ODA2). Because of these additional commands POLY/NWKY must be reassembled, and the JSR POLY instruction (OD03) of OPTN must be changed to include the new address of POLY. Therefore, OPTN (OD00) of "MUS1" must also be modified. I chose to place "QUA4" in page 01, so that the altered POLY/NWKY routine begins at 0121. Comparison of the POLY routines of "MUS1" and "QUA4" should clarify this matter.

Now that two additional NTBL locations for STG control voltages have been reserved, the second part of the algorithm of "QUA4" will be considered. The first choice to be made concerns the nature of the new STG outputs. Rather than simply duplicating the output of Quash 2 by loading it successively into Quash 3 and Quash 4, I complemented the output of Quash 2 and loaded it into Quash 3. To complement a number is analogous to inverting a control voltage, and this is acomplished on the 6502 microprocessor by the EOR operation with $FF, i.e.1111 1111. For the output of Quash 4 I executed another EOR operation with $AA, i.e. 1010 1010, and applied this modification to the output of Quash 3. Since the STG values are computed in the TRGN (ODC3) routine of "MUS1", the NEXT (ODF7) routine must be modified.

The program changes necessary to carry out these

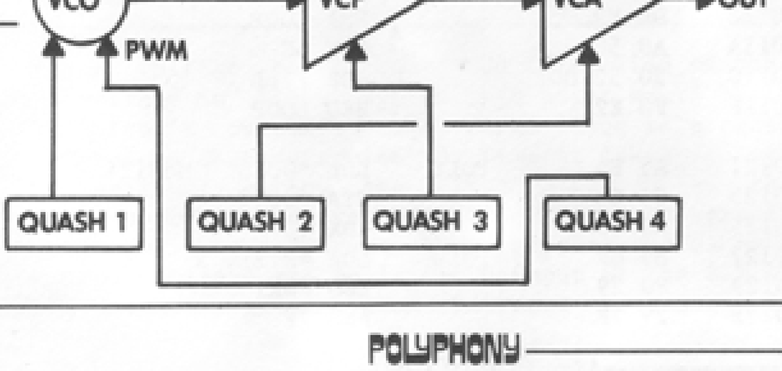
After the reassembly of TRGN, the JSR TRGN instruction (OD06) of OPTN must be changed to contain the new address of TRGN; in this instance TRGN's new address is 0177. "QUA4" runs at 0100, and since it is built upon "MUS1" it employs the same variables. The System Control Word (OE8) must set D7 to activate the transient generators; the Output Delay (OE9) may be the same as used in "MUS1"; the number of Output control channels (OEA) operates as in "MUS1," but in "QUA4" each output includes four Quash channels; and the transient parameters (OBA-OBE) function as in "MUS1." The control voltages available at Quash 3 and Quash 4 may be altered by changing the values at memory locations 01AD and/or 01B6. If "QUA4" is to be stored on cassette, the loading sequence is: 0100 01BF, followed by the appropriate file number and Load/Dump command.

# Hardware

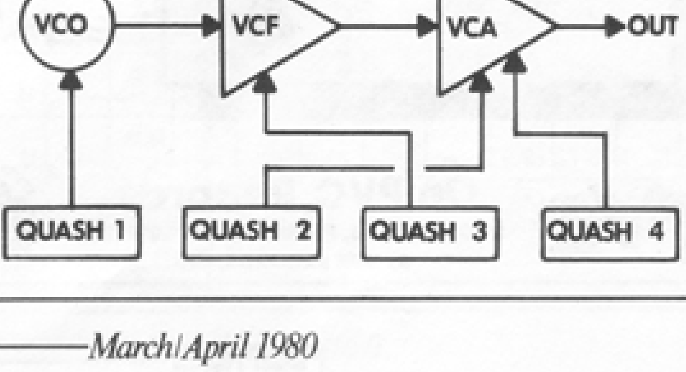
If the preceding software discussion is understood throughly there should be no problem implementing "QUA4." Since only transient generators are used there is no need for hardware envelope generators, but it must be remembered that the outputs of Quash 3, 7, 11, and 15 produce inverted voltages. A basic monophonic patch, where OUTS (OEA) is set to 01, is given in Figure 3. Quash 1 and 2 supply control voltages to a VCO and VCA respectively---the standard format of "MUS1." The two additional control channels regulate the VCF, (Quash 3) and the pulse-width modulation (Quash 4). "QUA4" provides the user with a way of working completely under computer control, eliminating hardware-supplied control voltages.

A slight variation of the previous example is illustrated in Figure 4, where the transient ouputs of Quash 2 and 4 are added to produce a 'new' envelope.

3

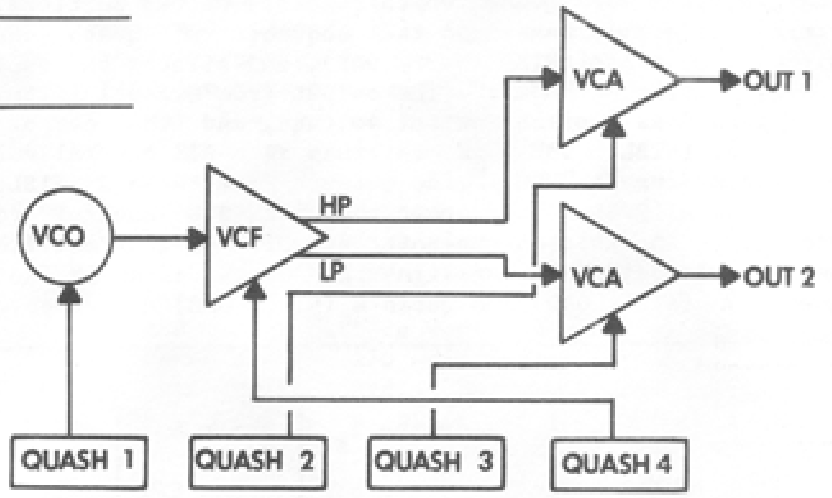



4



The inverted voltage present at Quash 3 is a source of some useful effects, one of which is shown in Figure 5. Quash

5




1 serves the VCO and Quash 4 controls the VCF. Two differently filtered signals, obtained at the VCF, are sent to separate VCAs. In this example, the Hi-Pass output goes to VCA 1, and the Lo-Pass output goes to VCA 2; Quash 2 controls VCA 1, and Quash 3 controls VCA 2. Since Quash 3 produces the inversion of the voltage at Quash 2, depression of a key on the AGO keyboard causes Quash 2 to activate VCA 1. But when the AGO key is released Quash 3 activates VCA 2, producing a ping-pong effect if recorded on two separate channels.

These are just a few of the possibilities afforded by "QUA4." Experimentation is the only way to discover its usefulness. Careful study of the following program listing should provide a few hints as to the possibility of utilizing the enormous capacity of "MUS1."

The following books are recommended to those who wish to learn more about the 6502 microprocessor. Together with the information available from Paia and POLYPHONY, writing original programs is possible.

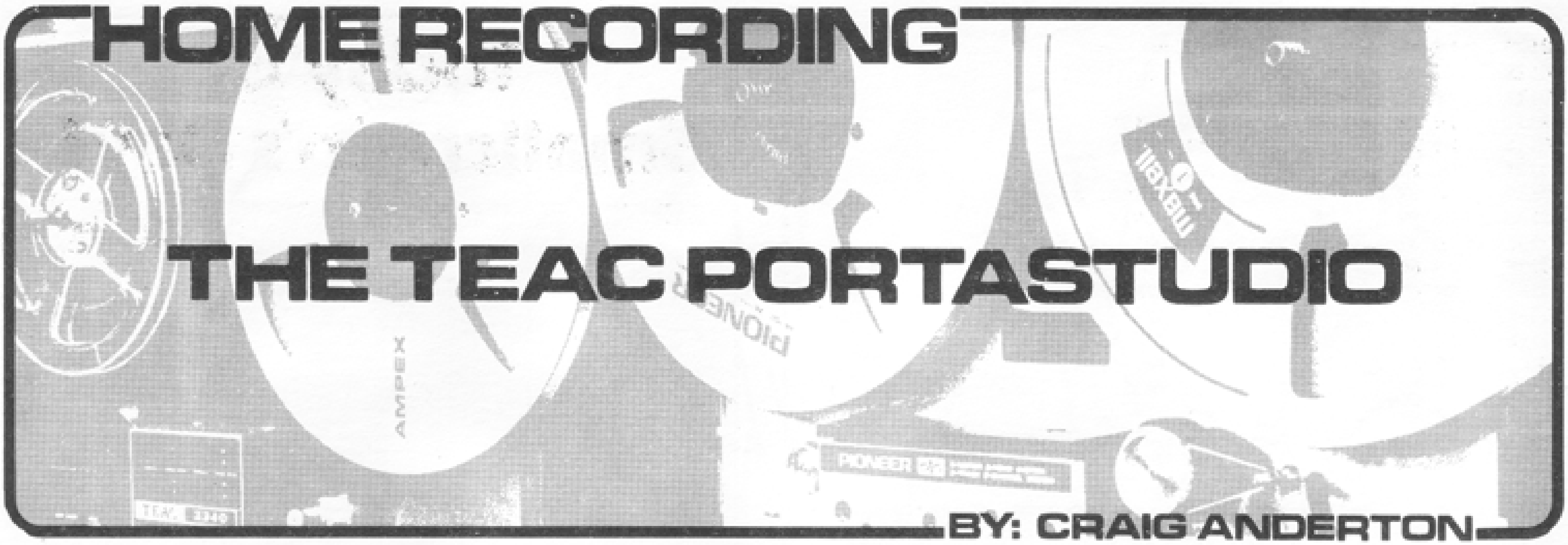

Barden, William. How to Program Microcomputers. Indianapolis: Howard Sams & Co., 1977.
Foster, Caxton. Programming a Microcomputer: 6502. Reading, Mass.: Addison-Wesley Co., 1978.
Zaks, Rodnay. Programming the 6502. Berkeley: Sybex, 1978.
__________. 6502 Applications Book. Berkeley: Sybex, 1979.


# QUA4: An Advanced STG Control System Program Listing

```
Initial Pointers:                       0ED -- FF
                                        0FE -- FF
                                        0FF -- 00

MUS1 Variables :                        0E8 -- C0 or 80
                                        0E9 -- 20
                                        0EA -- OUTS

                                        0BA -- ATCK
                                        0BB -- DCY
                                        0BC -- SUST
                                        0BD -- RELS
                                        0BE -- PEAK

0100     20 21 0D     OPTN     JSR INIT     MUS1
0103     20 21 01     LOOP     JSR POLY     Modified version
0106     20 77 01              JSR TRGN     Modified version
0109     20 2B 0D              JSR NOTE     MUS1 Continued
010C     A5 BF                 LDA *CLCK
010E     8D 20 08              STA DISP
0111     20 00 0F              JSR DECD
0114     C9 01                 CMP 01
0116     30 E8                 BMI OPTN
0118     D0 E9                 BNE LOOP
011A     A0 5C                 LDY 5C
011C     20 52 0D              JSR FILL
011F     F0 E2                 BEQ LOOP

0121     A5 EA        POLY     LDA *OUTS    Modified version
0123     85 EB                 STA *OUTT
0125     A2 10                 LDX 10
0127     B5 CF        POLO     LDA *NTBL, X
0129     F0 29                 BEQ NWKY
012B     29 7F                 AND 7F
```

```
012D  09 40              ORA 40
012F  A0 09              LDY 09
0131  88          LP0    DEY
0132  F0 12              BEQ NEXT
0134  D9 DF 00           CMP KTBL, Y
0137  D0 F8              BNE LP0
0139  95 CF              STA *NTBL, X
013B  C6 EB              DEC *OUTT
013D  F0 37              BEQ OUT
013F  A9 00              LDA 00
0141  99 DF 00           STA KTBL, Y
0144  F0 04              BEQ LP1
0146  29 BF       NEXT   AND 0BF
0148  95 CF              STA *NTBL, X
014A  24 E8       LP1    BIT *CTRL
014C  10 28              BPL OUT
014E  CA                 DEX         Decrement counter
014F  CA                 DEX         to provide 4
0150  CA                 DEX         successive Quash
0151  CA                 DEX         channels per voice.
0152  D0 D3              BNE POL0
0154  A2 10       NWKY   LDX 10
0156  A0 09              LDY 09
0158  A9 40       NK1    LDA 40
015A  35 CF              AND *NTBL, X
015C  D0 0E              BNE NK3
015E  88          NK2    DEY
015F  F0 15              BEQ OUT
0161  B9 DF 00           LDA KTBL, Y
0164  F0 F8              BEQ NK2
0166  95 CF              STA *NTBL, X
0168  C6 E8              DEC *OUTT
016A  F0 0A              BEQ OUT
016C  24 E8       NK3    BIT *CTRL
016E  10 06              BPL OUT
0170  CA                 DEX         Decrement counter
0171  CA                 DEX         to provide 4
0172  CA                 DEX         successive Quash
0173  CA                 DEX         channels per voice.
0174  D0 E2              BNE NK1
0176  60          OUT    RTS

0177  A2 10       TRGN   LDX 10      Modified version
0179  A0 40       ADSR   LDY 40
017B  B5 CF              LDA *NTBL, X
017D  2A                 ROL
017E  2A                 ROL
017F  B5 CE              LDA *PARM, X
0181  90 19              BCC RELS
0183  10 0B              BPL DS
0185  18          ATTK   CLC
0186  65 BA              ADC *ATCK
0188  C9 BF              CMP 0BF
018A  90 1B              BCC NEXT
018C  A5 BE              LDA *PEAK
018E  D0 17              BNE NEXT
0190  A0 C0       DS     LDY 0C0
0192  E5 BB              SBC *DCY
0194  C5 BC              CMP *SUST
0196  10 0F              BPL NEXT
0198  A5 BC              LDA *SUST
019A  10 0B              BPL NEXT
019C  38          RELS   SEC
019D  09 80              ORA 80
019F  E5 BD              SBC *RLS
01A1  30 04              BMI NEXT
01A3  A0 00              LDY 00
01A5  A9 80              LDA 80
01A7  94 BE       NEXT   STY *CWRD, X
01A9  95 CE              STA *PARM, X
01AB  48                 PHA         Save accum in stack.
01AC  49 FF              EOR FF      Complement Quash #2.
01AE  CA                 DEX         Output, decrement
01AF  95 CE              STA *PARM,X   counter, and store
01B1  24 E8              BIT *CTRL     in Quash #3 output.
01B3  10 0A              BPL OUT     Check for D7 set.
01B5  49 AA              EOR AA      EOR Quash #3 output,
01B7  CA                 DEX           decrement counter,
01B8  95 CE              STA *PARM,X   store in Quash #4.
01BA  68                 PLA         Retrieve accumulator.
01BB  CA                 DEX
01BC  CA                 DEX
01BD  D0 BA              BNE ADSR
01BF  60          OUT    RTS
```

# HOME RECORDING




# THE TEAC PORTASTUDIO

BY: CRAIG ANDERTON

There has been an amazing amount of interest in TEAC/Tascam's latest contribution to the home recording market, the Model 144 Portastudio. From what I hear, people are snapping these units up as fast as TEAC can make them. Considering that for $1100 list (usually available for less) you get a complete, self-contained, highly portable 4 track studio with tape transport, mixer, and patch points, it seems only logical that this is the type of product that would get musicians very excited.

The Portastudio continues the trend of the democratization of recording. Until the early 70's, recording was the domain of the music industry elite, partly due to the cost, and partly due to the complexity, of the equipment involved. While machines such as the 3340, 80-8, and Otari MX-5050 series helped open up the world of recording to an ever-expanding number of people, the realization of a truly portable, truly low cost means of providing 4 channel recording has been elusive--until now. The TEAC Portastudio achieves that goal in an admirable fashion, and will no doubt help to swell the ranks of recording fanatics.

The purpose of this column is twofold. One, I want to point out the features, advantages, and disadvantages inherent in the design of a unit such as this. Armed with this info, the reader will then be able to decide whether the Portastudio is indeed applicable to his or her uses. Second, I'd like to get a little into the applications end of things, since the Portastudio opens up many avenues of audio exploration not easily attainable with conventional recording setups.

THE TRANSPORT: We'll begin by looking at the machine's transport. This transport is designed to use standard cassettes, however, neither the speed nor the head configuration is standard. The cassette runs at 3-3/4 ips, which dramatically increases the fidelity at the expense of half the running time per cassette, and incompatibility with other standard cassette machines. While some people consider this a disadvantages ("What? I can't play back my cassettes on the Portastudio?"), the fact is that <u>any</u> multitrack machine does not have a format compatible with consumer tape recorders....I mean, you can't play back a 3340 tape on a cassette deck either, but who cares?

Because of the 4 track format, if you record on all four tracks you are recording over the full width of the tape. As a result, you cannot flip the tape over to gain more recording time. Between the doubled tape speed and inability to flip sides, a C-90 turns into a C-22½; when you throw in the fact that you also need to use premium cassettes such as TDK-SA types, your tape costs might come as a shock if you're used to having a regular cassette deck. However, bear in mind that when recording, you want quality and that quality has to come from somewhere. I feel TEAC has made the right tradeoffs with this machine; I don't begrudge the extra tape cost considering the extra performance achieved through these tradeoffs.

The tape motion controls are all-electronic, which is great; it's impossible to jam or spill the tape by hitting the wrong buttons. In addition to the standard tape motion controls, there is also a pause control, a memory rewind, and a button labeled "door", which opens up the plastic dust cover that lays over the cassette when the Portastudio is in use. The final transport control is the pitch control, which provides a wide variable speed range. Variable speed is tremendously useful technique that can only be appreciated when you have such a control on your machine (I hope to do a column soon on variable speed techniques). About the only negative thing I can say about the transport is that, as with some other memory rewind systems, the location counter seems to creep backwards at a slow rate as your session progresses. So, if you record one song right after another, you might find yourself bumping into the end of the first song after memory rewinding back to the beginning of the second song. It's no big deal, but it is something to look out for. Of course, simply resetting the location counter to 0 from time to time will solve the problem.

MIXER: The mixer is a 5 input, 2 output type of mixer. Four of the inputs include the following controls:

1. Mic-line/tape switch. Chooses the corresponding tape channel output or an auxiliary input (patch points are located along the rear panel).
2. Preamp. With the above switch in the mic-line position, a single control sets the preamp gain. Fully clockwise gives maximum gain (mic level); full counterclockwise gives minimum gain (line level). You can plug instruments like guitar directly into an input, and adjust the gain for a suitable recording level--no other preamps or buffers are required.
3. Aux send. This sends part of the signal to an auxiliary buss output, which can connect to something like a reverb or echo unit. The output of the reverb, echo, or what-have-you returns back into the machine via the 5th input, which is simply a knob labeled "Aux Rcv".
4. Equalizer. This is a simple bass/treble, boost/cut type of device for general purpose equalization.
5. Panpot. Pans the input between the left and right output busses.
6. Fader. Controls the overall channel level.

There is also a master fader that sets the overall volume level going on to the stereo output buss.

MONITOR SECTION: The Portastudio has pretty complete monitoring facilities, which is one reason why it is so easy to use. In addition to line outputs for feeding conventional stereo monitor amps, there is also a built-in stereo headphone amp.

The monitoring mode is determined by two interlocking pushbuttons, one labeled <u>cue</u> and the other <u>remix</u>. In the remix position, you will <u>hear</u> whatever is happening on the stereo output buss (the level in your phones is set by the buss monitor control). There is also a four control, mono cue buss which is enabled when the pushbutton is in the <u>cue</u> position. The cue section listens to the tape outputs of each of the 4 channels, as will as the stereo output buss (whose level is again controlled by the buss monitor control). As an example of how you would use all this, let's say you've already recorded on tracks 1-3 on the tape and want to overdub four vocalists in channel 4. By turning up the cue controls for channels 1-3, you will hear what's happening on the tape. You then plug the four vocal mics into the mixer, and send them into channel 4 by depressing channel 4's record button and adjusting the appropriate mixer controls. Turning up the buss monitor control allows you to hear the mix of the four vocalistis along with the cue signals from the first three channels.

If all of this sounds complicated, remember that it's a lot easier to understand all this with the machine sitting in front of you! In the meantime, let the complexity assure you that the Portastudio has enough built-in flexibility to handle a wide variety of recording situations.

The monitor section also includes four VU meters, which monitor the tape outputs when you're not recording, and monitor the stereo output buss when you are recording. Because TEAC has provided a number of patch points at the back, you can play many tricks with the Portastudio such as having two cue busses, an extra effects buss, and so on.

RECORD SELECT MATRIX: There are 6 pushbuttons that select the channels on which you will record; four of these correspond to the four individual channels and are arranged in two pairs, while two of these are "record off" switches that cancel the recording function for each pair. When you're set up to record, a small red light flashes in the transport area. When you actually start recording (by hitting record and play at the same time), the light glows red continuously. The flashing red light is a convenient reminder that one of the channels is ready to record, and helps minimize mistakes ("Gee, I forgot the record switch was on for track 2...I guess we'll have to record the drum track over again").

OTHER FUNCTIONS: As mentioned earlier, there are a number of patch points on the back. The four main mixer channel inputs use ¼" phone jack inputs, while the jacks for cue buss out, aux out, line out, aux in, aux send, etc. are RCA phono types.

The preceeding description of the controls and jacks is perhaps overly brief; many more words could be spent on the above functions and features, as well as how to use them. However, if you're interested in the Portastudio, your best bet is to go to your local Tascam dealer and take a look at the thing, paying particular attention to the instruction manual (which is quite thorough). This will give you a greater insight into the functions of the various controls, features, and jacks. Rather that spend any more time on these, then, let's get into a more subjective evalutation of how the Portastudio performs.

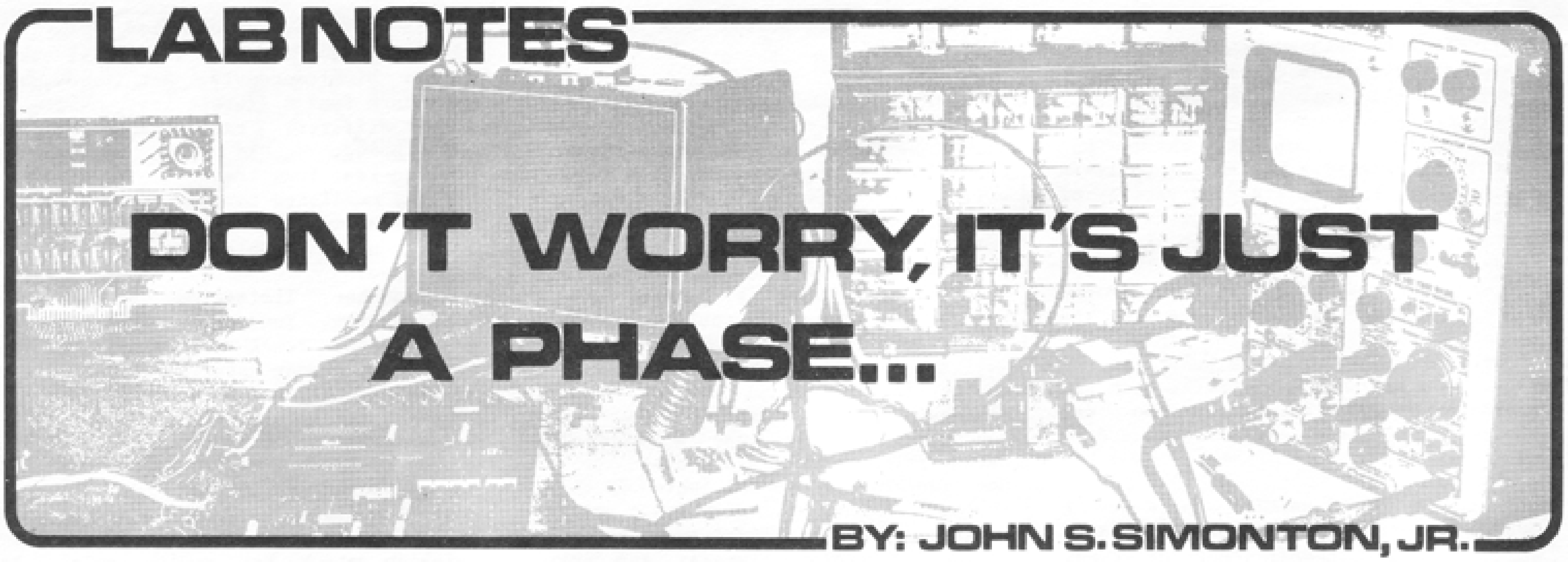
LIMITATIONS: There are several limitations associated with the Portastudio's operation. For one thing, you can only record on 2 tape tracks at any one time; you cannot record on all four simultaneously, although you can of course record a cumulative total of four tracks. Whether this is a problem or not is up to you to decide. The recording work I do is all overdub-oriented, so I've never needed to record on all four tracks simultaneously. But if you want to record four different instruments on four different tape tracks without overdubbing, you had better dig out a reel-to-reel four track.

As mentioned towards the beginning, the cassette format is non-standard. This is not a problem as far as I'm concerned.

Another limitation occurs with ping-ponging. You cannot ping-pong to an adjacent track--for example, if you want to ping-pong material in channels 1 and 2 over to track 3, you can't do it because track 3 is adjacent to track 2. However, you could transfer tracks 1 and 2 over to track 4 since this track is not adjacent to either track 1 or 2. What this means is when ping-ponging, you have to spend a little more time thinking about how you're going to assign your tracks. This is a minor inconvinience, but certainly nothing too terrible.

If you're used to reel-to-reel recording, you'll note some other Portastudio idiosynchrosies. For one thing, you cannot record real hot levels on the the machine without getting considerable distortion. Whereas with a ¼" reel-to-reel going at 15 IPS you can keep you meters in the red and still get acceptable sound quality, that is a no-no with the Portastudio...I pretty much treat the 0 VU point with respect. Also, funny things happen to the Dolby

continued on page 32....



# LAB NOTES

# DON'T WORRY, IT'S JUST A PHASE...

BY: JOHN S. SIMONTON, JR.

Any waveform, no matter how angular its appearance, can be produced by summing together sine waves of the proper frequency, amplitude, and phase relationships. A square wave, for example, can be produced by adding to some fundamental frequency the components which are odd integral multiples of the fundamental (odd order harmonics) and whose relative amplitude is the reciprocal of the order number.

The series of illustrations in figure 1 begin by showing the result of adding to a fundamental a 1/3 amplitude third harmonic. The next illustration adds a 1/5 amplitude fifth harmonic. The next shows a 1/7 amplitude seventh harmonic. As you can see, at each step the walls of the resulting waveform become slightly steeper and the top slightly flatter. It's not hard to extrapolate and realize that eventually, as more and more harmonics are added, the final result will be square.

Once this concept becomes part of your nature (when it doesn't seem strange anymore) you are well on the way to really understanding synthesis, because you are beginning to see that the timbre of any sound is the result of its harmonic structure.

So, in light of this, here's a little quiz. Three waveforms are pictured in figure 2. One is obviously a ramp and the others are obviously not. By looking at the scope traces, can you imagine what these sound like?

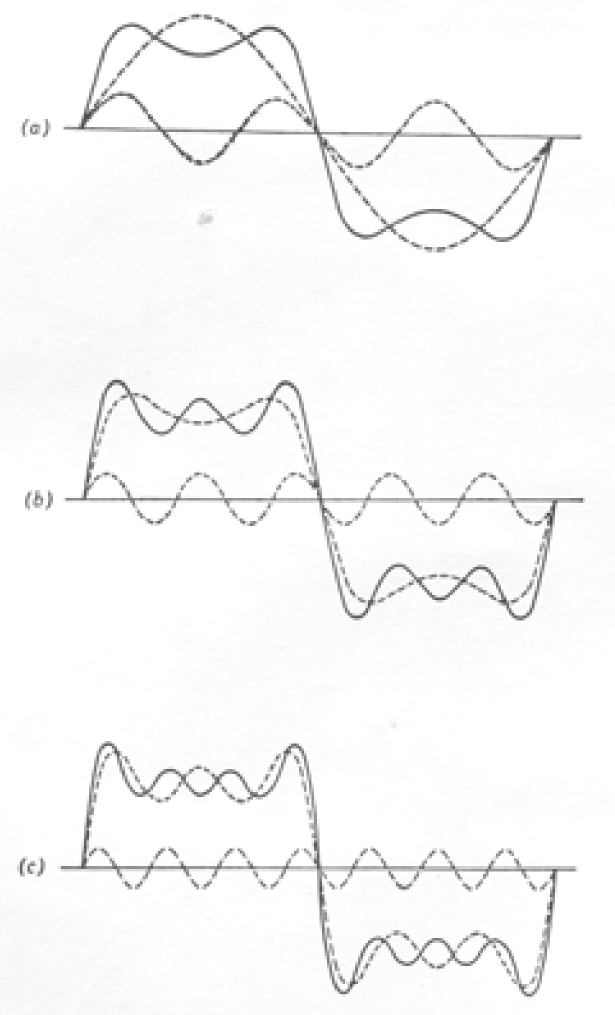
Well, it's a trick question. The answer is that even though these three waveforms appear to be entirely different from one another they all sound the same. Then what's the difference? Phase.

The constructed square wave in the earlier illustration had all harmonics "in phase"; by that I mean that all the component frequencies crossed zero and began increasing at the same time, at the beginning of the square wave.

Another case is illustrated in figure 3, that in which the third harmonic does not begin increasing at the same time that the fundamental does. Here the third harmonic is 180 degrees out of phase with its equivalent component in the square wave. As you can see, it makes a big difference in the appearance of the trace, but it doesn't make any difference in the way that these two would sound if they were applied to an amplifier and you listened to the result. Phase doesn't count.

Now realize that when I make the sweeping comment that "phase doesn't count" I'm talking about relatively specific cases, monophonic signals whose frequency is well within the audible spectrum. In the broader sense phase matters because it is the primary mechanism by which we are able to locate sound sources in space. The separation between your ears causes the sound from a source at your right side to reach your right ear before it reaches your left and this time (phase) difference is decoded to tell you that the source is to your right.

one




Similarly, at subaudible frequencies (as with control voltages) phase matters. The only difference between an upward and downward ramp is the phase relationships of the components, but if you use them both as pitch modulating control voltage into a VCO, you are obviously going to notice a big difference. But if the ramps themselves are of a high enough frequency to be audible and you listen to both of them, I can guarantee that you can't tell them apart.

This is a really hard thing to buy, it seems so intuitively wrong, so I invite you to prove it to yourself.

You're going to need some way to control the phase relationships of a signal's component frequencies. Most university level Electrical Engineering laboratories have machines that can do this. These days they are probably electronic, but when I was in school they were much like the tone generators of "electric" organs, tone wheels which spun underneath magnetic transducers. In the ones that I've seen, the position of the transducers could be altered to change the phase relationships and attenuators were provided to control component amplitude. If you're able to lay your hands on something like this you can easily spend more time than you can afford playing with it to see if you can produce the various basic waveforms. And of course you can change the phase relationships of the components and see whether you can hear the difference.

If you don't have access to this admittedly rather special purpose piece of equipment you can perform similar experiments with the "all-pass" configuration provided by most high quality synthesizer filters (such as Paia's EKx-20).

An all-pass filter may itself seem a contradiction in terms (I, for one, think it is) because a filter is a device which in one way or another eliminates or separates something. But if you accept the name at all, it's accurate because an all-pass filter doesn't eliminate anything. It's a phase shifter.

As the voltage which controls the frequency of interest of the filter is increased, higher frequencies which are components of the input waveform undergo a phase shift. A typical all-pass filter is capable of 720 degrees of shift and if it's a good unit the phase is the only characteristic of the input which will change. If it's not so good a filter, it may also color the sound by changing the amplitudes of components and this will of course invalidate the results of the experiments we are going to run.

Apply some nice harmonically rich wave (ramp or square) to the input of the filter and set the frequency of the waveform so that it's within the operating range of the filter (maybe 1 KHz or so). Monitor the output of the filter with a scope (you want to see what you're listening to) and also pump it into an amp.

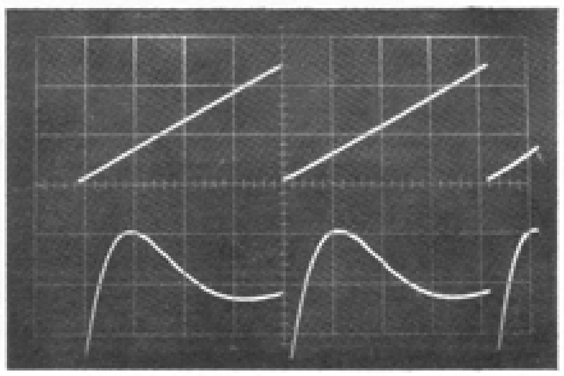
Begin by setting the frequency of interest (Fc control voltage) well below the fundamental frequency of the exciting waveform - with most filters this will require a negative control voltage. And make sure that the filter has no resonance (ground the control voltage input for this parameter) since resonance will tend to alter the amplitude of components at the frequency of interest and thereby produce timbral changes. At the output of the filter you should see essentially the same waveform as is being applied to the input.

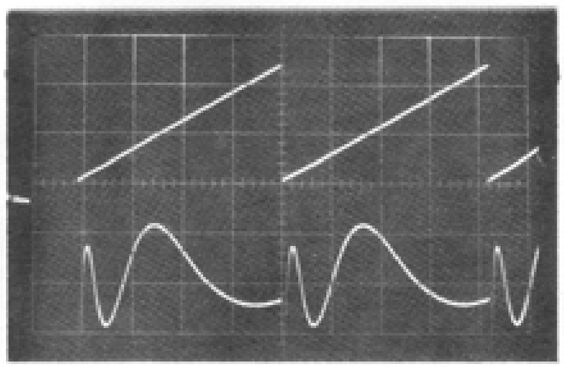
Now slowly (we'll see why this is important in a moment) increase the Fc control voltage and observe both the appearance and sound of the resulting output. To me, it's incredible to watch the way the output bends around into things that look for all the world like what ought to be the familiar ringing tone of a resonant filter, but aren't. In fact, there is no sonic change at all.

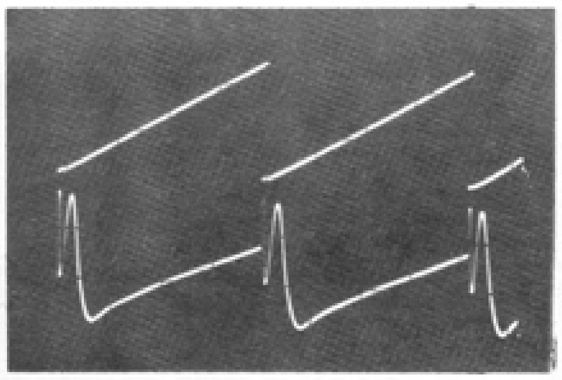
What a handy thing (perhaps you're thinking), a device that changes the appearance of a waveform without changing it's timbre. Just what you need for music.

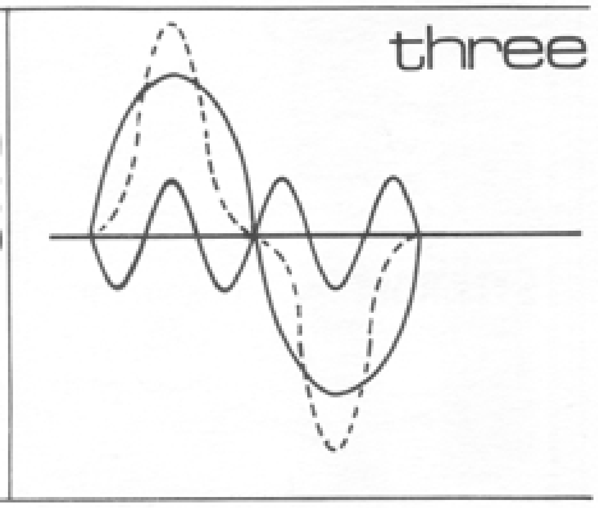
But the application of an all pass filter goes somewhat beyond the specialized use that we're making of it here.

For example, we've just seen that constant phase difference (within the bounds discussed) don't produce timbral changes, but changing phase relationships - that's another case. For one thing, in order for the phase of a sine wave to change it must temporarily increase or decrease in frequency. If phase is going to advance, the frequency must (during that time that the phase is changing) increase slightly. Obviously, since pitch detection is where it's at as far as the physiology of our ears is concerned, we can hear the frequency shifts. They're perceived as a vibrato of sorts.










Possibly you heard them in the above experiment if you changed the voltage too quickly. If you didn't, do that now. Quickly raise and lower the control voltage of the filter and observe that there is definitely audible effect. To make the effect even more noticeable, bring in a periodic control voltage (a low frequency sine wave - a

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# PORTASTUDIO

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circuitry when you use extreme amounts of variable speed. Again, no big deal, but something to remember.

Frankly, I don't think you're going to hear of too many gold records being cut on Portastudios; there is more noise and distortion than reel-to-reel recording. But I don't think it's fair to expect something that lists for $1100 to offer equivalent performance to a reel-to-reel based 4 track studio that costs a minimum of $1500 to $2000. The real strength of the Portastudio lies in its abilities as a "sketch pad", a rough draft machine, a chance to make your mistakes-- and learn your arrangements-- before going into a "real" (i.e. expensive!) studio.

APPLICATIONS: I hope the above section has gotten across two important points: 1) the Portastudio soes not skimp on flexibility or ease of use; and 2) it does not offer performance equivalent to ¼" reel-to-reel machines. Now, let's talk about why I'm so enthusiastic about the Portastudio.

For one thing, the low cost means that every serious musician with about $1000 of discretionary income can now learn recording techiniques at home and realize a variety of musical dreams. Because the Portastudio is designed pretty much for foolproof operation, people who are all thumbs when it comes to threading tape and can't figure out what's happening with mixers should welcome the functionality of the Portastudio and its extreme ease of use. Additionally, it's easy to set up for recording. Got an idea for a song? Just plug your guitar or mic into an input, press the record button, and you're ready to go. Want to do an overdub? Change a knob or two, push a different record button, and turn up the cue control for the first track--instant overdub. Unless you're doing something radical, there's no need for re-patching or extensive control changes that could slow you down.

Many times I'll be playing guitar in my living room late at night, and come up with a nifty idea for a song. Usually, it's just too much bother to go into my "big" studio, turn on all the equipment, set up mics, etc; and it used to be that most of the time, my nifty ideas would never turn into anything. Now that I have access to the Portastudio, I just plug in and record, and do a couple of overdubs. Since punch-in and punch-out are also easy with the Portastudio, I can do things like record a verse on a track, listen back to the verse on the cue buss, and then at the right moment, punch the record button and continue recording the chorus, next verse, and so on. For composers and songwriters, the Portastudio is a dream come true. Its ease of use and rapid set up time actually encourage the creative impulse, as opposed to more complex studios where you can forget a melody by the time you've go everything set up. Want to see what your voice sounds like at different speeds? Use the variable speed control. Want to experiment with harmony lines? The overdub capability easily allows for that.

Actually, I think <u>every</u> studio should have a Portastudio handy to serve the role of an instant "Studio B". Suppose you've got a group in a mammoth, 24 track, computerized studio and it's time to do overdubs. You can make a real rough mix of the tracks you already have on the 24 track master on to one or two tracks of the Portastudio. Then, let's say the vocalist wants to lay down some vocal overdubs, and the guitarist wants to lay down some harmony parts. While the vocalist is cutting tracks in the main room, the guitarist can tuck the Portastudio under his arm, go out into the hallway, and practice the part (even adding overdubs). After the vocalist has finished, the guitarist can come in and lay down the overdubs in no time flat because they've already been extensively rehearsed on the Portastudio. I had a situation recently where a sax player wanted to add a part to one of my arrangements, but I wasn't sure about spending additional time in the studio to see if the part would work out or not. So, I mixed my basic track down on to two tracks of the Portastudio, and let the sax player fool around with putting down different parts for a couple of hours. After she got the parts she wanted, she played back the Portastudio cassette for me with the new parts and I could tell that, indeed, the parts were real good and would fit perfectly into my arrangement.

The more I use the Portastudio, the more uses I discover for it. The portablilty is another big plus--you're not tied down to a specific room or location. If you've got a jam happening somewhere, you can easily take the Portastudio along with (try taking a 4 track, reel-to-reel based studio around under your arm!).

CONCLUSION: I feel the Portastudio is one of the most significant developments in home recording since the birth of the 3340. It has opened up creative channels for me that would not be opened otherwise, saved wear and tear on my "big" machine, allowed me to work out parts and write songs at any time of the day or night, and overall, has made the process of recording just that much easier and more accessible.

However, you also have to realize that you're not going to get full-blown, reel-to-reel performance out of the thing; this particular device has been optimized for convenience, portability, and ease of application rather than high performance. In this respect, I feel that TEAC deserves high marks for the conservative promotion they've been giving the Portastudio--they tell you in front that it does have certain limitations, rather than promising the sky and the moon and then failing to deliver. But when all is said and done, the performance is definitely more than adequate to fulfill the function that the Portastudio was designed to fulfill--an audio "etch-a-sketch" that would allow songwriters, composers, and musicians greater creative freedom. When looked at from this perspective, the Portastudio is a complete and total success. I'm glad I have one to play with.

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## Literature

HOW TO START YOUR OWN RECORDING BUSINESS PART TIME. Our guide discusses planning, business deductions, selecting equipment, types of services, recording for demos, churches, schools, etc., advertising, and many more suggestions to make a profit using your own equipment. Send $3.95 to HSC, PO Box 4870-P, Austin, TX 78765.

## Services

CUSTOM ELECTRONIC MUSIC equipment consulting service. Anything from just the design to complete fabrication. Also custom modification of existing equipment. Other items soon to be available. For more information or estimate, write Jim Patchell, 533 San Blas Pl., Santa Barbara, CA 93111.

## Music equipment

360 SYSTEMS Spectre OEM circuit board (interface section only). Fully trimmed and tested, includes power supply, PVC, string select circuitry, hex fuzz, octave and transpose selects, hex pickup and documentation. $625. Kevin Monahan, 2360 7th Ave., Santa Cruz, CA 95062, (408) 476-5630.

HEXAPHONIC SYNTHESIZER- expanded Paia P4700J includes computer, 12 channels S/H, 6 VCOs, 4 VCFs, 8 VCAs, 2 LFOs, 2 reverbs, 8 channel mixer, and 3 custom modules. $900. (303) 722-0508, 6 PM MT.

ORCHESTRON (Laser Optics Disc) includes Hammond B-3, violin, flute, and choir discs. Needs some work. Cost new: $2500. Best offer. Steven Halpern, 620 Taylor Way #14, Belmont, CA 94002, (415) 592-3300.

4700C, assembled, factory calibrated. $350. William Quiroz, 510 S. Breed St., Los Angeles, CA 90033, (213) 269-3541.

ARP AXXE, mint condition, $450. Everett Rantanen, 2829 So. 56th Ct., Milwaukee, WI 53219, (414) 327-4393.

MODULAR SYSTEM with Paia 2720-1, -2A, -3B, -3L, -4, -11, 4710, 4720, 4750, and 8780 D/A; all in two hinged road cabinets. Also 8782 Keyboard. Fully assembled and calibrated with patch cords. Must sacrifice for $450. Tony Roide, 4909 St. Thomas Dr., Fair Oaks, CA 95628, (916) 966-1963.

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POLYPHONY

# DON'T WORRY, IT'S JUST A PHASE...

....continued from page 31

effect even more noticeable, bring in a periodic control voltage (a low frequency sine wave - a few Hz.) and notice that with this phase modulation going, the vibrato effect is quite obvious. But realize that you're not hearing the phase difference, you're hearing the phase change and the only reason you're hearing that is because of the slight pitch variations that result from it.

There are also other instances in which phase is important, specifically those cases where a phase shifted signal is recombined with the original waveform. But here again, it's not the phase shift you hear; it's an artifact of the phase shift. In this case, components in the filter output which are out of phase with their counterparts in the original signal are altered in amplitude. A component which is 180 degrees out of phase but of equal amplitude cancels, eliminating that frequency from the final output. You can easily observe this classical "phase" phenomenon by simply summing the input and the output of the filter together with a pot. And notice that you can not only hear differences as the frequency control voltage is changed, you can also stop at any point and the timbral difference will remain.

# VOLUME - 5 INDEX

# the CATSTICK synthesizer controller

## THE BETTER ALTERNATIVE TO WHEELS, RIBBONS, PRESSURE PADS, KNOBS, SLIDERS AND PEDALS, FOR ANY PATCHABLE SYNTHESIZER

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### HOW DO YOU USE IT?

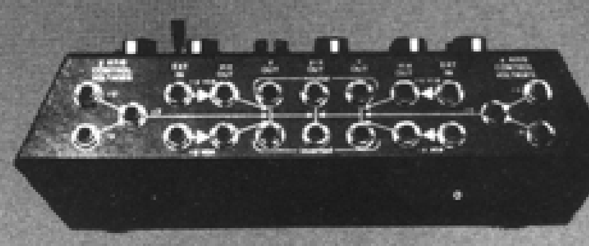
For portable synthesizers, like the CAT, Odyssey, or Minimoog, you can connect the CATSTICK outputs to the VCO, VCF or VCA inputs normally intended for footpedal controls. This lets you use the CATSTICK LFO's and control voltages to modulate the synthesizer as the joystick is moved. In patchable systems like the ARP 2600 or Modular Moogs, you can connect the CATSTICK VCA's in series with patchcords to allow real-time control of synthesizer patches.

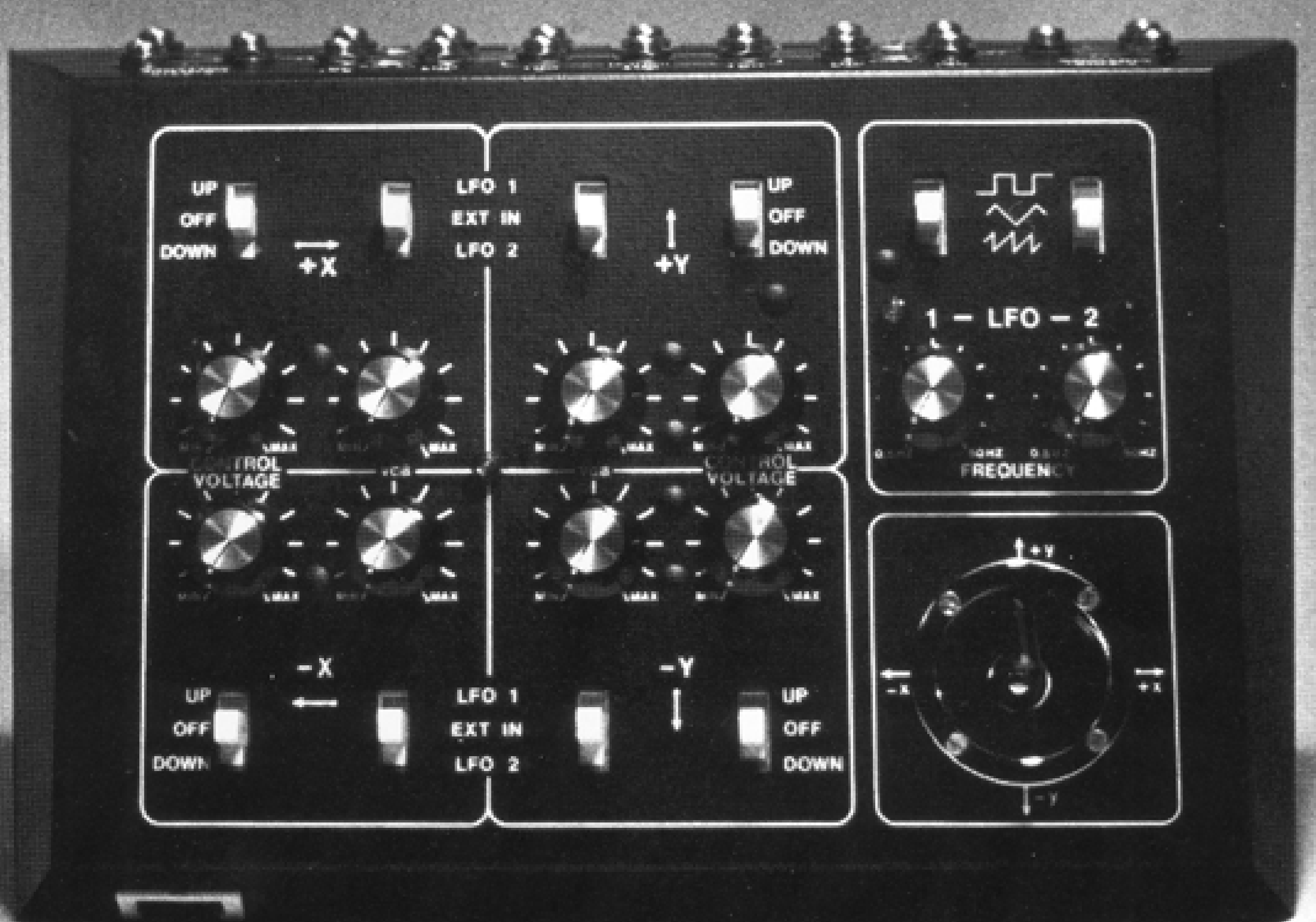
### WHO CAN USE IT?

If you own a MINIMOOG, CAT, KITTEN, ODYSSEY, 2600, OBERHEIM, MODULAR SYSTEM or any other synthesizer with control voltage inputs, you can use a CATSTICK. And, if you don't have control voltage inputs or want more, we'll show you how to modify your instrument or do it for you at a very modest cost. We can also modify your synthesizer for "single cable" connection to the Catstick outputs.

### PATCHING VERSATILITY

Included are four VCA's (each externally accessible), two wide-range LFO's with rate monitors, and a complete internal voltage processing system. The twenty-jack rear panel patch bay allows access to all of the internal control voltage signals and makes the CATSTICK a versatile addition for both performance-oriented and studio synthesis systems.



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- Liberal patch over hardwire points on rear panel
- Absolute minimum point to point wiring for easy kit assembly and high reliability.
- 2 exponential VCO's; sine, triangle, ramp, square and modulated pulse waveforms.
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