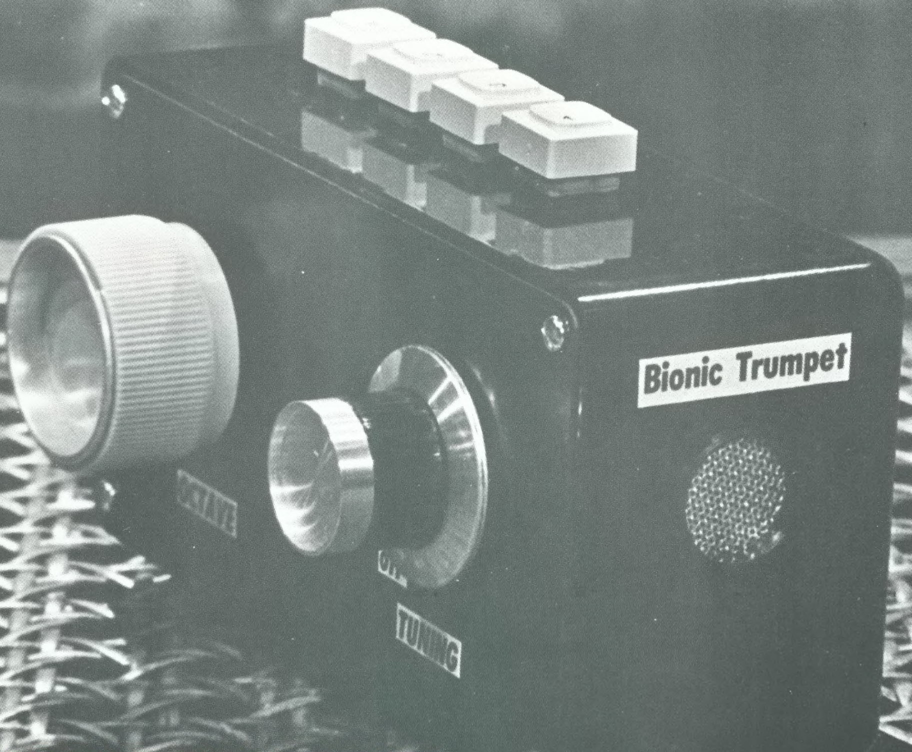
POLYPHONY

FEBRUARY 1978

A POLYPHONIC SYNTHESIZER

SOLVING STICKY CONTROL PROBLEMS



BIONIC TRUMPET

PATCHES MODIFICATIONS

STAFF

EDITOR:	Marvin Jones
ASSISTANT EDITOR:	Linda Kay Brumfield
CONTRIBUTING EDITOR:	John S. Simonton, Jr.
PRODUCTION:	Debbie Collins Shirley McConnell
PRINTING:	Reggie Terry
Published Quarterly by:	PAIA Electronics, Inc. 1020 W. Wilshire Blvd. Oklahoma City, OK 73116

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Write for Polyphony

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LETTERS:

WHAT'S NEXT?

Marvin et al,

Here's another Polyphony issue and once again I am amazed. How long have I been using your stuff and probing, asking questions, wondering what you'd do next, etc. I had begun to get the feeling you were going to "dead end" synthesizers as you had created virtually all necessary modules and adaptations. How many times has someone in OK expressed the sentiment "Well, we're really here for the experimenters...". And of course I saw Simonton going off the deep end into computers a few months back and I wondered where it would lead. What next, right? Being a musician, I was chiefly interested in live performance, and through your equipment, technical advice, and some of my own innovations I have never had to sacrifice performance in real time for portability, reliability, etc. Of course your stuff may take experimenting, adjustment, and calibration, but anyone too lazy to do that is going to be too lazy to play the synthesizer anyway. They'll have to opt and cop for "name brands" with labeled plastic overlays or a preset b--------d (organ, indeed).

Yet this issue shows me that in fact you guys are more alive than ever–your computer search is sending you not just to more attachments, but new available packaging resulting in more efficient PLAYING of synthesizers. Great! Obviously there is no end to what you can and will do for synthesists. Thank God someone is still trying to help us, not just taking our money.

Of course, I never doubted you for a minute.

John Deaton
Victoria, TX.

BACK TO MUSIC, PLEASE!

Polyphony,

I was pleased with your magazine until you started putting computer and gate articles in. It would be alright if the articles involved a direct interface to synthesizers or something like that. Leave the other part of computing (teaching etc.) to the computer magazines. The "Blinky Light" project in November was cute, but what did it have to do with synthesizers? Speaking of November, why did I get that issue in January?

Respectfully,
Ja Souter
Clovis, CA

..... continued on page 27

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ON THE COVER:

This strange looking instrument is the Bionic Trumpet; designed, constructed and photographed by Greg Leslie. Associated construction article on page 15.

POLYPHONY REVIEWS!

ASTRAL WARRIOR

by Lord Manuel
$1.50 ppd. Available only by mail order from:
Lord Manuel, BX 153, Arlington, MA 02174

Some of you may recall my editorial last issue talking about the "New Folk" music. Here is some more of it. And very good, too. Lord Manuel is definitely more of a musician and poet than a singer, but the enthusiasm and variety shown in his voice throughout the record masks any lack of training. The record is a 7 inch 33 RPM, with four songs on it. The pressing is very good, with few hot spots or pressing noise. "Astral Warrior" is an introductory instrumental, and is reminiscent of the combo organ music of the early sixties. Melody is overlayed by synthesizer, and in conjunction with the electronic drummer and some tape echo, the overall effect is a "happy space" sound. "Have a Little Faith in Me" has some very good string-type synthesis, but musically was not too interesting to me. The progressions are very repetitive, almost hypnotic, and serve primarily as a vehicle for the extensive lyrics. Side two starts with "Star Poacher", guaranteed to take you where no man has gone before. The tape manipulation on the drums is used well to represent the interstellar motion described in the song, while the vocals are mixed back quite a ways to give a floating lonliness to the sound. "Silver Fantasy" is the rockiest cut, and features a ponderous sinister synthesizer bass which works well for the song. Also the use of acoustic drums and cymbals along with the electronic drummer helps fill in the usual void generated by auto-drummers. The primary disappointments throughout the record were the excessive use of tape echo and panning effects, although the effects seemed to fit the songs in most cases. Lord Manuel's equipment consists of Fender Rhodes, Fender organ, Univox Mini-Korg, a Paia Oz-Gnome, and a Paia 2700. But, he said he sold his organ and piano in order to pay to have the record pressed. Now, how's that for dedication! This record is definitely a good example of "New Folk" music in every respect; get it if you can.

Marvin Jones

MAGIC IS A CHILD

By Nektar
Polydor PD-1-6115

The sound of Nektar on this album has reverted to the more basic sound of "Remember the Future" while retaining a bit of the electronics and intracacies developed during the "Recycled" period. Guitars and vocals now take a front seat to keyboard wizadry and studio effects. Most of the keyboard work consists of smooth piano and organ, with Larry Fast adding his touch of flowing polyphonic fill in the background. Personally, I quite enjoyed the thick interacting arrangements and extensive electronics on "Recycled", but the smooth rock sound they come across with on this album will probably take them a long way towards gaining the recognition they have deserved for several years. Of particular interest to the synthesizer oriented person will be the backwards drums in "Love to Share" and the hypnotic, tunneling, flanged guitar on the intro to "Eerie Lackawanna". The slow "Listen" would make a good prospect for a commercial hit, as would the Aerosmithish "Spread Your Wings" with its tracking guitar/bass riffs and intricate Kansas-like rhythm patterns.

If you liked earlier Nektar albums, get this one. If you liked their last one, "Recycled", then perhaps you should wait to see what they have in mind for the future.

Marvin Jones

INTERGALACTIC TOURING BAND

Passport PB9823

IGTB is a concept album, revolving around a group of musicians who travel around the galaxy in an attempt to find new friends and spread the good word of Earth. The theme is as trite on album as it is on paper, and the compositions are an electrified lesson in how to get a recording contract. All tunes were penned by Wil Malone and Danny Beckerman, who obviously aren't the best progressive musicians but who have an amazing ability to come up with an entire album worth of music which will have recording executives and teenagers pounding at their doors. The primary redeeming factor for this album is the finesse with which the multitude of performing artists pulled off the material they were presented with. I don't know the full story behind the production, but I know that several dozen artists appear on the album - a la Alan Parsons. Immediately recognizable vocals include Arthur Brown, who does his usual outstanding work on "Universal Zoo", and Annie Haslam (vocalist for Renaissance) on "Reaching Out". It was a disappointment to hear Annie's versatile voice in an orchestral context which could easily have been lifted from many of the Renaissance albums. Scoring her into another style, perhaps a heavier composition, would have considerably emphasized her appearance. Larry Fast's unmistakable sound is instrumental in filling in the required "spaciness" and pulling the rambling musical themes into a semi-coherent work.

Technical effects on the album are quite good. The droning engine sound on "Approach (Overture)" is quite good in imparting the loneliness of space travel. Unfortunately, "Approach" is badly marred by a Strauss (2001) ripoff. The Vocorder shows up at several places each demonstrating a good match of vocal text and signal on which the vocals are superimposed. Someone obviously took the time to make sure there were harmonics in the signal where vocal fricatives were required for inteligibility. "First Landing" contains some good mandolin synthesis, and the string parts are shared between an orchestra and synthesis, which is nice.

Overall, this is a fun album to listen to. But the fact that the execution is so much better than the concept leads me to hope that this batch of musicians will someday get together to produce their own ideas.

Marvin Jones

MISCELLANEOUS

There are a few records which have recently been released that contain a few isolated highlights that everyone into synthesis should know about. First, drum synthesizers. Have you all heard the Syndrum. This excellent instrument is being snatched up by drummers everywhere, and recorded examples range all the way from Linda Ronstadt's "Simple Dreams", to the new Boz Scaggs and Rod Stewart albums. Also, Joni Mitchell uses the drum synthesizer on a few cuts of "Don Juan's Reckless Daughter". Joni's backup band for this album is the incredible Weather Report. As usual, Joe Zawinul's synthesis refuses to be anything but superb. That guy's bass lines are unsurpassed. Keep your eyes on drum synthesis. It's the next wave of special purpose synthesizers.

There have been a few people wondering what polyphonics is all about. If you want to hear a few good examples of polyphonic systems, check out the following: the "dance band-ish" style saxophone section in "Byrdland" from Weather Report's "Heavy Weather" album was done on an Oberheim 8 voice, I believe, and the introductory Spanish guitar solo for "Day at the Dog Races" on Little Feat's "Time Loves A Hero" was similarly done on an Oberheim system. Full blown demos of poly-

...... continued on page 14

SAMPLE AND HOLD RESURECTION
what to do with your analog sample and hold once you've gone digital

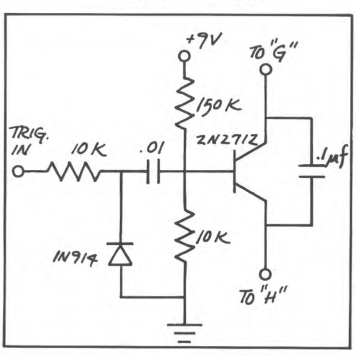
If you're like me, you've probably been working with PAIA for years. So, you began with the good old trimmer type keyboard. As PAIA grew, you've grown, and agreed with PAIA's better way of doing things. You now have a modern up to date Digital Keyboard. Now you probably think you have a useless -8 Sample and Hold, right? Wrong. This modification will change your line of thinking! This simple circuit will give you a very useful Sample and Hold module.

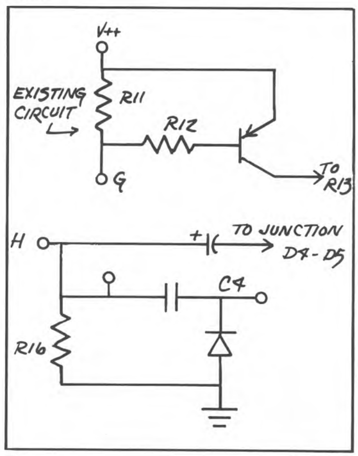
Start by constructing the trigger circuit on a small pc or perf board. Next, modify the Sample and Hold board by shorting out R15 in the old trigger-gate circuit. Then install the new circuit.

Put the voltage you want to Sample and Hold into "E", from any source (LFO, AR, ADSR, etc.), trigger it using the keyboard or any other source and you have a useful module!!

Have fun!!

Paul M. Gullotti





SPOTLIGHT EARLY EXPERIMENTS WITH COMPUTER MUSIC

By Marvin Jones

Robert Libbey has been experimenting with electronic music for quite a while. Starting early in his career with radio broadcasting and sound recording, his interest in electronic storage, processing, and generation of audio signals has continually grown into new technologies. It was inevitable, then, that Bob would end up interfacing one of today's electronic music synthesizers with one of tomorrows magic machines-- the RCA COSMAC microprocessor. Since the early seventies, Bob's job at RCA has been concentrating on video work, so his audio/music interests have become more of his avocation than they were previously. But, nevertheless, he never set aside his concept of a "better" home organ-- a family instrument that could generate a wide variety of voices, plus teach his children about the intricacies of music performance and composition. and provide a method of "recording" and composing a major musical work without the bulky tape recorders commonly required.

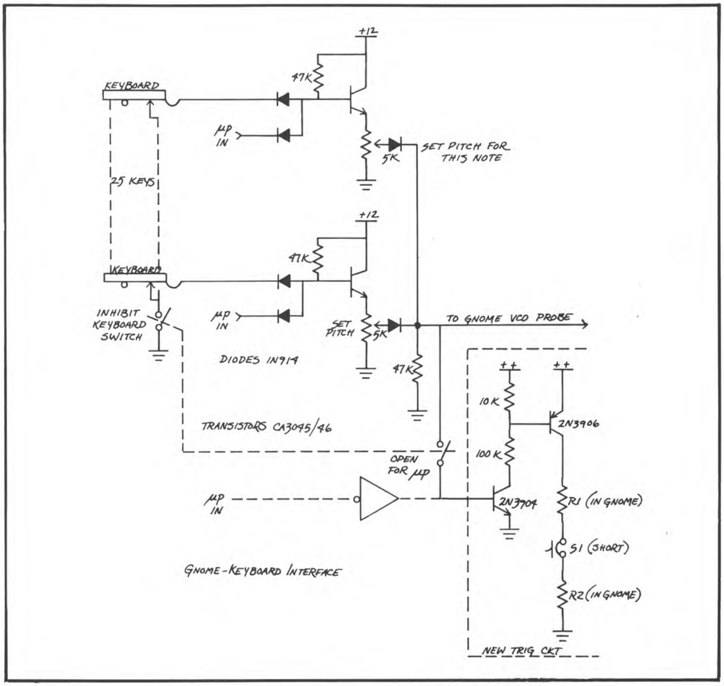
The high speed processing required for the video work Bob was doing increased his awareness and understanding of digital circuitry and the many applications it implied. Availability of microprocessors in the mid-seventies evoked some major breakthroughs in most every area of product development. Bob's job was no exception. And now, he had the tools necessary to start making his personal goals come to life. The work described in the following article was done in the latter part of 1975. When Bob showed the results to some of his colleagues at RCA, they published the results in the August/September 1976 RCA ENGINEER, an in-house magazine which keeps all the RCA employees up on what's happening. All of us here at Polyphony are very thankful to RCA for letting us reprint Bob's article so our readers can learn from his experience.

Since the Gnome/COSMAC interface, Bob has been working with the KIM system in an attempt to derive the most efficient programs for outputting monophonic and polyphonic signals. His goals during this period may have been somewhat different from those of others who have done similar work with the KIM or other microprocessors. He said that although he is not a professional musician, there are certain qualities he demands of the music he creates or listens to; one of those qualities is correct pitch, or tuning. Thus he did an in-depth study of various pitch systems, and how to create them with a microprocessor. Most literature seems to forget other tuning systems, especially physical which is used by many choral groups, in favor of the more popular equally tempered tunings. These experiments stretched through 1976 and led to his current hardware experiments. Bob has spent the recent months conjuring up an eight note, eight voice microprocessor controlled polyphonic synthesizer. Bob feels it has been an excellent exercise, and was amazed at the parallels between his experimental system and the recently announced Paia system.

Bob is beginning to get some ideas about how to tie his video knowledge in with his digital music systems. He would like to work with designing a microprocessor based system which would play a note selected by a musical keyboard, and simultaneously write the note in a musical staff on a video display. The 'processor' would remember the notes while more were added to build up a musical theme, or melody. During playback, the notes would be displayed, and upon command could be transposed to new keys or perhaps altered to play retrograde or inversion or other musical "tricks". All the while, the 'processor' would be producing a video display which would show the NEW, TRANSFIGURED themes as they are played. This would be a highly valuable teaching aid for music theory or keyboard technique classes.

And, as if all this weren't enough, Bob is also teaching a course on microprocessors two nights a week. Polyphony congratulates Bob on a great deal of contribution to the world of digital music, and thanks him and RCA for taking the time to tell us about his work. Paia is proud to be a part of it.

Along with the reprint of the original Gnome/COSMAC article, Bob sent the schematic of the interface circuitry he used. This may be of interest to those of you who wish to try similar experiments.



The ideal combo— a microprocessor and a music synthesizer

R.L. Libbey

A hobby is pursued for the fun it provides, but having fun does not exclude the educational benefits and intellectual stimulation that a hobby can offer. In this paper, a hobbyist describes the marriage of a music-making machine and its controller—the RCA COSMAC Microtutor.

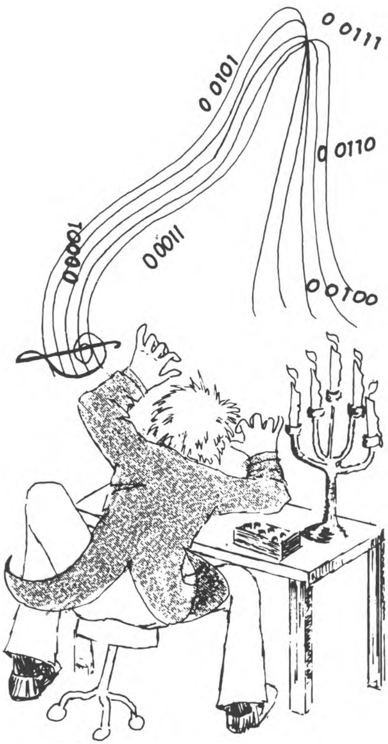
Robert L. Libbey, Broadcast Synchronizer and Time Base Corrector Group, Commercial Communications Systems Division, Camden, N.J., received the BSEE in 1950 and the MA in Dramatic Arts in 1952 from the University of Wyoming. Before joining RCA in 1952, Mr. Libbey had a diversified background in radio broadcasting and sound recording. Also, he has taught at Drexel University and the University of Wyoming. At RCA, he first did advanced development work in acoustics and magnetic recording in the Home Instruments Division. From 1958 to 1969, he was with the RCA Electron Microscope group. There, he specialized in high-voltage regulation, and was project engineer for the design and installation of the first one-million-volt electron microscope in the western hemisphere (developed for the U.S. Steel Corporation Laboratories in Monroeville, Pennsylvania). After joining the Television Terminal Group in 1969, he designed video switching and effects systems and is now involved in the design of digital television products. Mr. Libbey's hobbies include music, multi-channel sound recording, and now microprocessors.

Author with his Combo. At left is the RCA COSMAC Microtutor; at center (front to rear) are the keyboard, interface, and attenuator; synthesizer at far right.

Ed. Note: Last fall, Bob Libbey of Commercial Communications Systems Division in Camden took the CEE course, "Microprocessors for Logic Design—C55." After the formal learning period, Bob considered ways of increasing his understanding of the RCA Microtutor used in the course. Because of his background and interest in music, Bob decided that mating a microprocessor and a music synthesizer would be a logical step for furthering his understanding of the Microtutor and its programming. Although Bob says it was "a first, crude attempt at programming," the system worked and delighted Bob and his music-loving family. Bob's next project is to build and program a system to play complete chords.

The programming portion of this paper, of necessity, assumes some familiarity with the RCA COSMAC Microtutor and its software techniques. References are also made to the COSMAC User's Manual and COSMAC Microtutor Manual, which are produced by the Solid State Division and are included in the course material of "Microprocessors for Logic Design."

If you are interested and want more detail on Bob's project, write to him at Bldg. 10-2, Camden, N.J.



THE COMBINATION of a microprocessor and a music synthesizer creates an instrument with outstanding artistic and educational potential. By using a microprocessor to control a sophisticated synthesizer, sounds, tempos, modulations, etc., can be produced that could only be dreamed a few years ago. Even a modest synthesizer makes an exemplary output device for learning programming and multiple input/output techniques. If the synthesizer used is somewhere between the simplest and the most complicated, rudimentary software and hardware skills can be expanded to include analog-to-digital-to-analog (we still listen for analog signals) techniques and, as one advances further, the Combo can be used to produce more complex musical sounds and harmonies, using (and learning about) digital filters.[1]

Combo system

Our first-attempt Combo consisted of the RCA COSMAC Microtutor and a PAIA Electronics "Gnome" micro-synthesizer.[2] It was decided that even this first attempt at processor-controlled music should have the ability to control the character of the tone (note), as well as its pitch and duration.[3] The Gnome synthesizer had previously been adapted to a surplus organ keyboard and could be tuned to play music using the familiar western world even-tempered scale.[4] A switch made it possible to control the synthesizer with either the keyboard or the microprocessor. Fig. 1 shows the system block diagram. The Microtutor and its interface create coded signals that choose notes in a manner similar to striking a key on the keyboard. In addition, the output

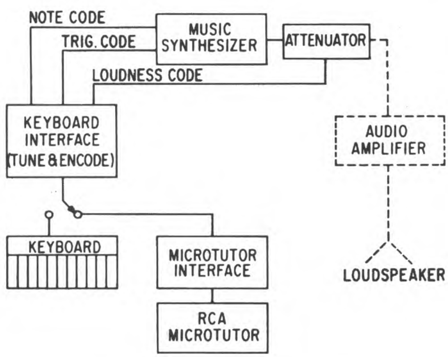


Fig. 1 — Combo system block diagram.

Reprint RE-22-2-21
Final manuscript received August 6, 1976.

codes can control the characteristic sound and loudness of each note. The loudness attenuator is a resistor-diode network (a simple digital-to-analog converter) that changes its attenuation when different parts of the network are grounded with a logic "LO". The amplifier and loudspeaker were part of a home hi-fi system.

The program software had a goal of performing five major functions. First, it should select the proper pitch and duration of the note (including "no note"—a rest). Similarly, it should change the character of the note by triggering the voltage-controlled amplifier and voltage-controlled filters in the synthesizer. Also, there should be a provision to select one of four loudness values. Finally, it would be desirable to change the key in which the music is played. This "key change" feature would be especially useful to the music student using the microprocessor-synthesizer Combo in composition studies.

The program included memory tables for pitch and duration. The duration tables included a time: T1, T2, T4, T8 and T16 for one-sixteenth, one-eighth, etc., through whole notes; and T1.5, T3, - T24 for "dotted" notes.[5] The duration value for each note depends upon the type of note, the tempo of the work being played, the programming technique, and the clock and instruction speed of the microprocessor. The Microtutor Manual and the COSMAC User's Manual describe the time calculations. The Microtutor has an adjustment on its clock frequency which provides for a convenient way to change the tempo.

The pitch code can be any system of binary outputs that will translate into the desired musical note. As an example, middle-C might be code "0 0001", C-sharp "0 0010", and D would be "0 0011", etc. Moreover, appropriate codes can be used for loudness values, key signatures, and for triggering the synthesizer.

Using the music sheet, each note could be given a separate code for pitch, duration, loudness, key and character (wave-shape—attack and delay).

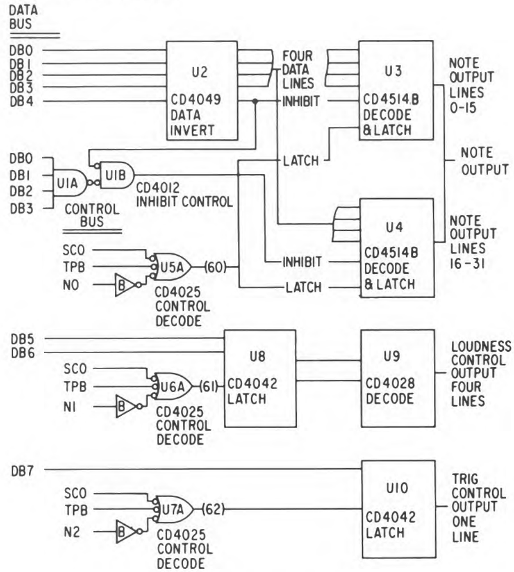


Fig. 2 — Interface circuits.

Hardware

The interface hardware for this first-attempt combo proved extremely interesting. The first paper design required *seventeen* interface chips as compared with thirteen chips for the complete Microtutor. The final design required ten CMOS interface ICs. Fig. 2 shows the interface hardware. This first-attempt system used a simplistic hardware design. Integrated circuit U2 inverted the sense of the output data to make it compatible with the decode/latch IC's U3 and U4. U3 will decode only the lower four data bits DB0 through DB3. A "1" on DB4 will inhibit U3 but, with the help of U1B, will enable U4. If all data bit lines are "0," all outputs of U3 will be "0" (high) and likewise U1A will cause all data output lines of U4 to be "0" (high).

The control lines SC0, TPB, and N0, acting with U5, will cause data to be latched in U3 and U4, and only in U3 and U4, when a "60" output instruction is executed. In a similar manner, the data on lines DB5 and DB6 only is latched in U8 and decoded in U9 when control lines SC0, TPB, and N1 enable U6 with a "61" instruction. Likewise, U7 and U10 use DB7, SC0, TPB, N2, and the "62" Microtutor instruction.

Software

The program was patterned after the "Table Driven Sequence" in the Microtutor Manual. Fig. 3 shows the software program flow chart. The COSMAC (Microtutor) microprocessor contains sixteen very powerful and convenient registers that, in this program, are used to count and point—in consecutive order—to different blocks of memory. The block (table) containing the addresses for the 32 note codes starts in memory location 223 and is pointed to by the register termed note pointer or "NP." The table for the loudness values starts at memory location 190 and uses pointer register "LP," etc. The counter register used in several parts of the program is "CR"; and "PC" is the program counter. The steps in the first column of the flowchart are preparatory steps—they input data, set up the tables, etc.; the steps in the second column make up the working program—they play the music.

Thus, for each note a pitch value was chosen, outputted, and latched. If loudness and triggering information was required, it was immediately chosen and sent out. The program would then go through an appropriate delay loop to hold the selected note the proper time. At the end of the delay, the next note and loudness and trigger values are selected Because of memory limitations, the key-

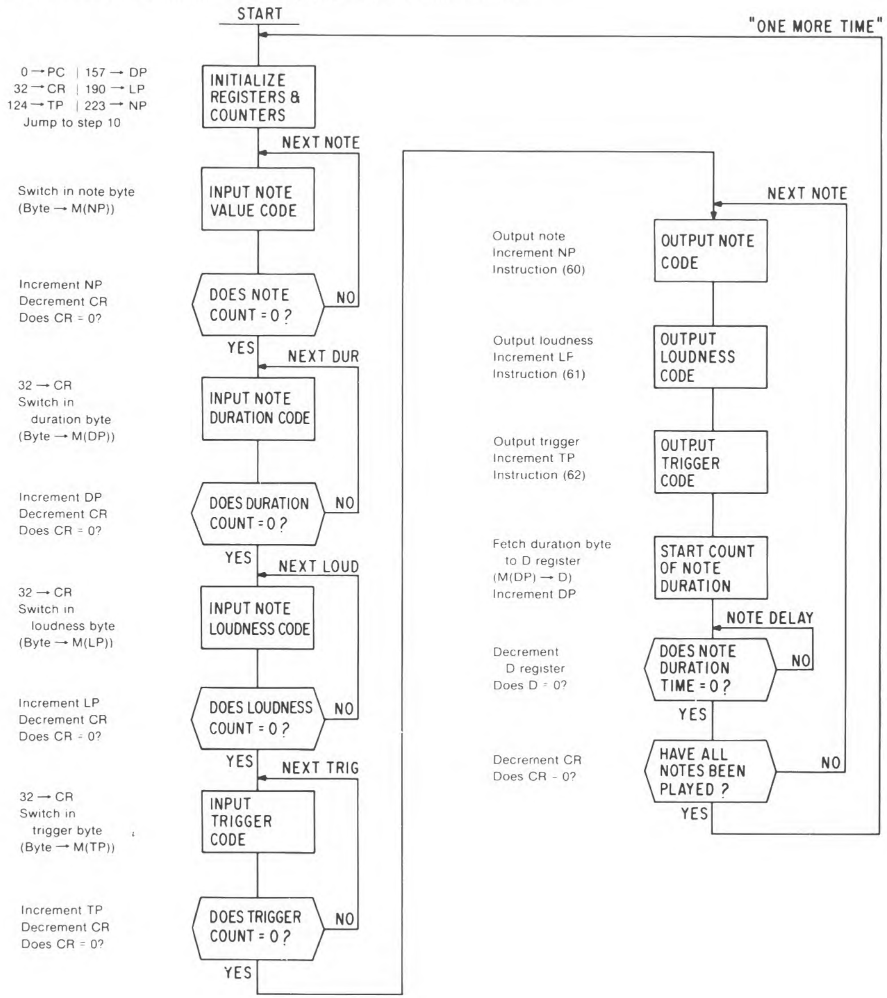


Fig. 3 — Combo flow chart.

change feature was not implemented in this first-attempt combo.

Conclusions

In keeping with the Bicentennial, my son—who was taking a high school course in fundamentals of music—transcribed the Revolutionary War marching song "Chester" for our microprocessor-synthesizer experiment. The program and sixteen measures of the music took all 256 bytes of memory.

References

1. Alles, H.G. (Bell Telephone Laboratories): "A Hardware Digital Music Synthesizer" *Proc. EASCON "75,"* pp. 217-A to 217-E.

2. Simonton, J.S. Jr.; "Build a Portable Synthesizer", *Radio-Electronics* (Nov 1975) pp. 37-101.

3. L. Bazin and D. Schneider of Broadcast Camera Engineering used the Microtutor to control a "Music Maker" with programmed pitch and rhythm generators, based upon "A Rhythm Section You Can Build," *Electronics Illustrated* (Nov 1968) pp. 57-66.

4. This was accomplished even though the Gnome manufacturer states that this can not be done. If the scale was not so tuned, programming the pitch would be much more difficult. In fact, tuning and keeping them in tune is a problem in many music synthesizers. Also, see Carl Helmer's "Add a Kluge Harp to Your Computer," *BYTE* (Oct 1975) pp. 14-18.

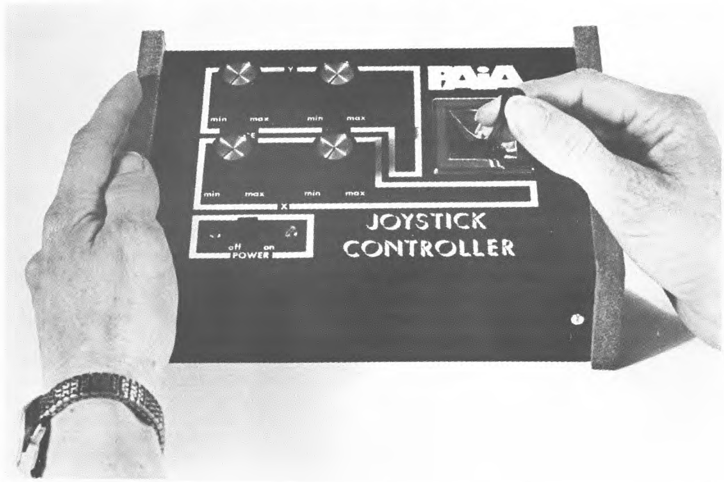
5. A dotted note is held 1.5 times as long as the note would be held without the dot.

Hobbyists...share your hobby interests with *RCA Engineer* readers! The "on-the-job/off-the-job" column offers you the opportunity to report on your experiences. The information should be related to technology and be of general interest to engineers. If you wish to contribute, contact Frank Strobl, Bldg. 204-2, Cherry Hill, N.J., ext. PY-4220.

a handy way to solve STICKY CONTROL PROBLEMS

by Marvin Jones

Electronic music has always been an attractive alternative for radical innovators in music composition or performance. One of the reasons is that the music circuitry is easily adapted to many different modes of control. Some synthesists still use the AGO keyboard, but others are die-hard anti-keyboardists, and their search involves use of push switch matrices, ribbon controllers, light sensors, computer controllers, and on and on. One of the earlier alternative controllers was the Joystick. Its popularity was no doubt founded on the fact that with one hand you could easily control two independent parameters. Later joysticks even added a third axis of control which was accomplished with rotation of the stick itself. The applications of a multi-parameter controller are many. It can be used by itself, or in conjunction with other controllers for a WIDE range of effects -- probably more so than any other type of controller (excepting a computer, I guess).



Before getting deeply involved with some sample applications for the Joystick, it is important to understand that the Joystick is a much more versatile and wide range controller than the more common AGO keyboard. The musical keyboard offers precision repeatability of a limited set of musical notes. But, the Joystick has a much wider range of outputs, with capability to generate an infinite number of outputs across its range. Additionally, it has the capability to generate TWO variable outputs PLUS a third trigger output. With all this increased capability comes an increase in operational subtlety and lack of applications comprehension. The point is that you must be prepared to experiment and PRACTICE in order to LEARN how to get the most from this new controller system, just like you had to LEARN how to play the AGO keyboard. The reward will be the ability to do some very complex control of a large assortment of synthesizer equipment from one efficient controller device.

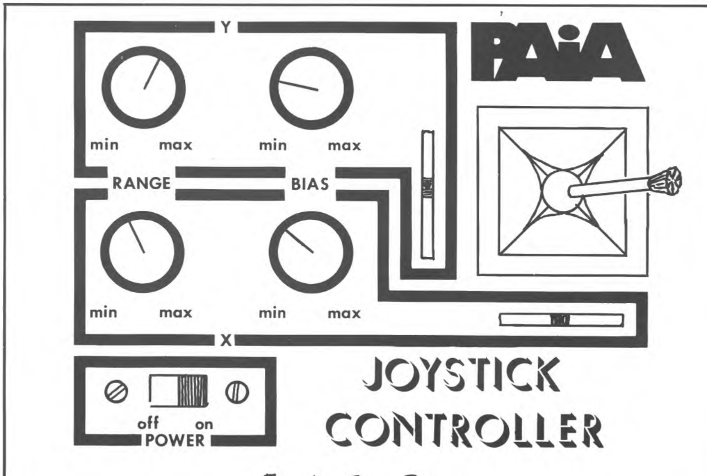


Fig. 1: Front Panel

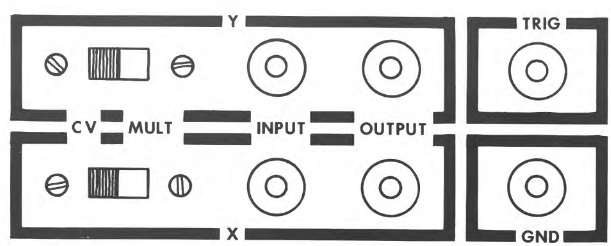


Fig. 2: Rear Panel

PANEL LAYOUT AND CONTROL OPERATION

Note that the front panel graphics (See Figure 1) are arranged to depict the interaction of the rotary controls with the Joystick. The "X" and "Y" designations are derived from standard notation for designating a point in a two-dimensional graph. The X axis refers to horizontal, or side-to-side Joystick motion. The Y axis refers to vertical, or up-and-down Joystick motion.

The controls in the X section operate exactly like the Y controls, except they affect a different axis of motion.

RANGE - This control determines the amount of output voltage change for a given amount of Joystick motion. Clockwise rotation will give increased output changes.

BIAS - This control determines the amount of initial voltage at the output; or in other words, the center voltage of the Joystick at rest. Any movement of the Joystick will then cause an increase or decrease in this center bias setting. Minimum settings of the BIAS controls represents an output of approximately -4 volts. Clockwise rotation will increase the output voltage to a maximum of about +4 volts.

TRIMMERS - The ribbed, thumbwheel sliders adjacent to the joystick mechanism acts as fine tuners for the setting of the Joystick.

POWER - Sliding this switch to the right applies power to the internal circuitry. Be sure to switch off the power when not in use to help prolong battery life.

On the rear panel (See Figure 2):

GND - This jack should be patched to a common ground point on the equipment which the Joystick is to be controlling. This connection is REQUIRED for proper operation of the Joystick.

TRIG - This output jack provides a trigger output of approximately +8 volts when the metal section of the Joystick shaft is touched.

As with the front panel controls, the operation of the rear panel is similar for both the X and Y sections.

OUTPUT - The voltage which is the resulting summation of Joystick location, control setting, and so on, appears at this jack. This jack will be patched to the module you wish to control with the Joystick.

INPUT - This jack allows for input of complex control waveforms which can then be selected, processed and outputted from the Joystick. This jack is used for some very special modes of operation, and will be discussed in detail in a later section.

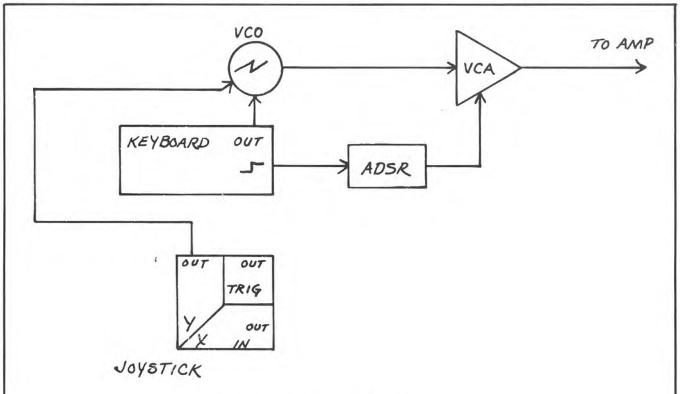


Fig. 3 : Pitch Bending Patch

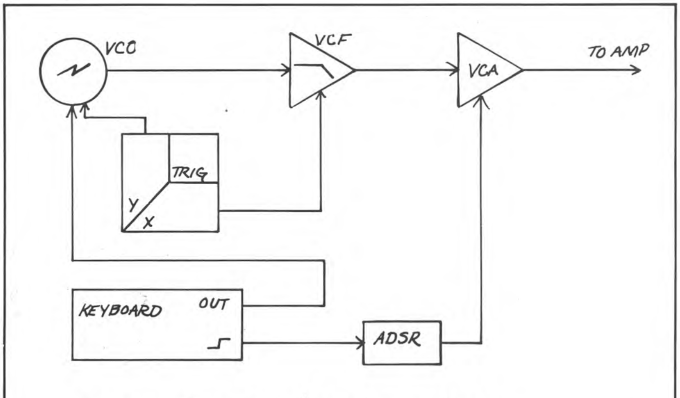


Fig. 4 : Pitch Bend/Tone Control Patch

CV/MULT - This switch selects the normal variable voltage mode of operation (CV), or the multiplier mode of operation (MULT) which uses the previously mentioned INPUT jacks.

One of the first things that comes to mind when you set a knob, lever, or thumbwheel next to a keyboard is pitch bending. Since you will be bending the pitch up and down, it makes sense to use the "Y" (vertical) axis for this application. First, MAKE SURE that you have a patch cord connected from the Joystick GROUND jack to a ground jack on your 4761 Wing (or to the chassis of the unit to be controlled, if it is from another system). The Y output should be patched into the VCO(s) to be controlled IN ADDITION TO the normal control voltage from the keyboard, sequencer, etc. (See Figure 3). The front panel Y RANGE should be set to minimum. We do not want any control voltage at the output when the Joystick is at its resting center position, so the Y BIAS should be set to approximate mid-position. While inserting and removing the patchcord from the output jack, make slight adjustments to the Y BIAS control until there is NO pitch offset when the Joystick is connected to the VCO. This is the point where there is 0 volts bias at the Joystick output. If you can get close with the BIAS control, then you can fine tune the zero setting with the Joystick trimmers. Now, you should be able to play up and down your keyboard and selectively add upwards or downwards pitch bends of up to a musical fifth or so. If you prefer more dramatic, wide-range pitch sweeps, the RANGE control can be advanced as desired. NOTE, though, that the BIAS control will need to be reset to provide the initial 0 volt output.

Those of you who are involved with experiments in microtonal or macrotonal tunings can use the front panel BIAS control to generate the positive (for microtones) or negative (for macrotones) voltage offset required. The wide range of the BIAS control should provide for a world of new tonalities to be explored.

Now let's take the basic patch a step further. Let's add variable filter sweep which is controlled by horizontal (X axis) motion of the Joystick. (See Figure 4). Use an additional patch cord to connect the X output to the control input of a VCF. For now, use the Joystick as the only input to the filter. With this patch, moving the Joystick up and down while playing the keyboard will produce pitch bends, while moving the Joystick from side to side will act as a tone control. When using this patch you will probably find it most useful to have the X RANGE set between midpoint and maximum for the widest range of tonal changes. This demonstrates a very important design concept - instead of using one set of master controls to adjust range and bias of both Joystick axes simultaneously, we chose to retain separate control for each axis to allow for the widest range of possible applications. On Joysticks which have only one master range control, you would never be able to execute the preceding patch without having either an extremely narrow (thus ineffective) filter sweep range or a very wide pitch bend range (thus difficult to use for bends of a few semitones only).

To add a third dimension of control to our expanding patch, connect the TRIGGER output to the input of an envelope generator. (See Figure 5) Then, patch the output of the envelope generator into the control input of the VCF which is being used as a tone control for the X axis. With these connections completed, you will be able to bend pitch, change the tone manually, AND touch the metal shaft of the Joystick to get an automatic sweep of the filter. NOTE that when using the touch TRIGGER function, all Joystick movement must be done by holding the end of the shaft.

From these basic applications, we can expand to using the Joystick trigger output to initiate a sequenced pattern, or better yet - to single step through a sequenced pattern. This way, you could have a number of preset voltages for a tonal sequence, a chord progression, or perhaps just use the sequencer gate outputs to enable various patches through a number of VCAs. Now, whenever you touch the Joystick shaft, the sequencer will advance one stage, and the next preset "patch" will be heard. By now your creative instincts should be getting a few ideas worth experimenting with. TRY THEM ALL. Experiment with having the Joystick control every module in your system. This will help you derive or stumble upon some good control combinations which you may not have thought of yet.

The Joystick can be used as a range determinant for control signals in several ways. Probably the most obvious is to use the Joystick as a bias voltage generator, with the front panel BIAS control set to the lower 50% of its range. This will cause a negative voltage output from the Joystick which, when summed with the control signal you wish to process, will provide enough negative bias to keep the other control signal from having any effect. However, as the Joystick is moved to sweep the output voltage up to 0 volts or greater, the other control signal will have more and more potential for forward biasing the module it has been attempting to control. Finally, when the Joystick is outputting 0 volts, it is as if there is no Joystick patched into the circuit, and the control signal will have its full sweep range applied to the module. Let's apply this technique in a situation where we want to have a variable amount of repeating sweep on a filter. (See Figure 6) Patch an oscillator signal through a low-pass or band-pass type VCF, and out to your

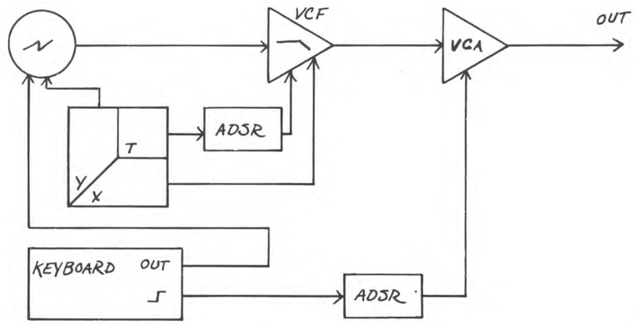


FIG. 5 : ADDING TRIGGER CONTROL

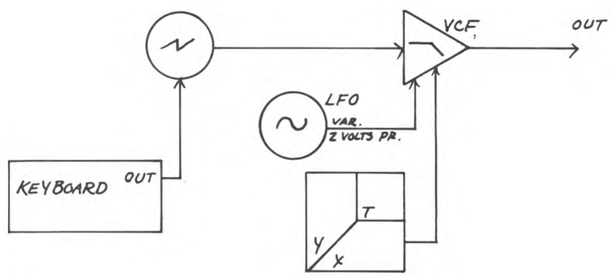


FIG. 6 : FILTER SWEEP RANGE CONTROL

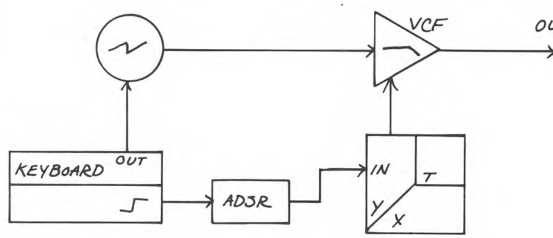


FIG. 7: EXTERNAL SIGNAL MULTIPLIER PATCH

amp. Select the variable output of a control oscillator, and patch into the control input of the VCF. Set the variable output of the control oscillator to produce about 2 to 3 volts output. Patch a Joystick output into the VCF along with the control oscillator. If you can hear the Low Frequency Oscillator (LFO) sweeping the filter a bit, decrease the setting of the Joystick Bias control to sum in a bit more negative voltage. Now, when the Joystick is moved, the sweeping oscillation of the filter should become increasingly predominant. With a bit of practice, this type of Joystick operation can add a significant amount of expression to your playing by providing fade-in vibrato, left hand modulation depth, or envelope range or emphasis control for doing accents, etc.

A slightly different method of control waveform range processing is through the use of the MULTIPLIER function which is accessible from the rear panel of the Joystick. In this mode of operation, the fixed joystick bias voltage is switched out, and an input is provided so the Joystick mechanism can be used to select variable amounts of a control waveform of your choice. This will provide a smoother variation in amplitude than the previous method, as we no longer need to bring the control waveform above a certain threshold before the control signal can take effect. Now, the Joystick will cause a true amplitude control of the input waveform.

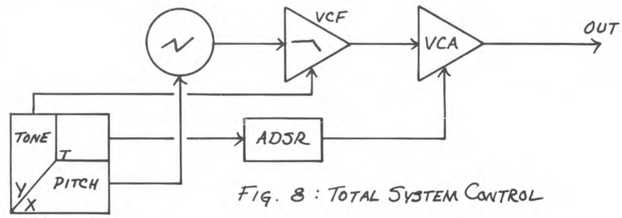


FIG. 8 : TOTAL SYSTEM CONTROL

Additionally, a bias offset can be added to the output, if desired, through the use of the front panel BIAS control. This could serve as an initial range setting before the Joystick introduces the various amounts of the modulating waveform. As an example, let's use the Joystick for envelope amplitude control. (See Figure 7) We will use a signal from an ADSR to be fed into the Joystick. Patch the ADSR output into the Y INPUT jack on the rear of the Joystick. Slide the Y selector switch to the MULT position. Patch the Y OUTPUT to the module you wish to control — let's use a filter again. In the MULTIPLIER mode of operation, you will probably find it easiest to have the RANGE control set in the 50% to 100% area. This will provide the widest range of Joystick action. Initially, set the Y BIAS control to approximate mid-position. While playing the keyboard, the Joystick can be moved up or down to cause an increase or decrease in the deflection range of the filter. Experiment with different RANGE and BIAS settings, as there are limitless possibilities with this mode of operation. How about having the Joystick trigger the module it is processing? In figure 7, remove the ADSR trigger from the keyboard and connect to the Joystick trigger output.

If you are ready to move up to some VERY subtle and tricky control configurations, you can try complete system control with the Joystick. At first, this type of patch is frustrating, as it is hard to get the degree of control required to produce "musical" sequences. A typical system control patch would be similar to that shown in figure 8. Once you have refined this type of Joystick operation, you will have gained a very powerful new tool for the expression of the extreme subtleties of your music.

POLYPHONY REVIEWS!

phonics can be heard on both of Pat Gleeson's albums, "Beyond the Sun" and "Star Wars". Both albums are ridiculously commercial, and still employ some track building, but the use of E-Mu's microprocessor system is displayed through the reproduction of some complex stored voicings and generation of unplayable time signatures and patterns.

Those of you into studio effects will like the extensive use of time delay and flanging on Robin Trower's "In City Dreams". They almost overdid it, but all examples are very tasty. Miking techniques and EQ are doing a lot of work for Camel on "Rain Dances". These guys do some outstanding composition, and the keyboardist is a real pro at voicings and left hand control when he cranks up the synth.

Following the "New Folk" trend, "French Kiss" by Bob Welch was recorded entirely by Bob and a drummer. There is a little bit of synthesis and studio effects, but most of the cuts are straight-out feel-good progressive rock. As I understand it, the first few tracks of most of the songs were actually done at home on a Teac. I hope more companies pick up on letting the creative people handle more of the creative process.

Rumor has it that Larry Fast did a session on the new Barbara Streisand album. Interesting. He's also working on Synergy III, so watch for it!

That's it for now. What albums have you been listening to recently? Why? Let us know.

SONGWRITER'S COOKBOOK & Creative Guide

by Robert Harvey
$2.95 from:
Sunshine Publishing Co.
P O Box 8044
Anaheim, CA 92802

Those of you who are more electronic hobbiests than musicians will welcome a book of this type. Although the book is only 24 pages long, Robert has managed to provide an overview of the most important concepts and musical details required for aspiring songwriters. If you know little or nothing of music theory, introductory material will help get you on your feet. For those who have music background but keep forgetting things, this book can serve as a valuable reference. The back of the book is filled with charts, glossaries, scales and lots of other goodies. It's hard to summarize or categorize what it takes years to learn and understand, but Robert has made a solid attempt in "Songwriters' Cookbook...".

Marvin Jones

THE EVOLUTION OF ELECTRONIC MUSIC

by David Ernst,
Schirmer Books, New York

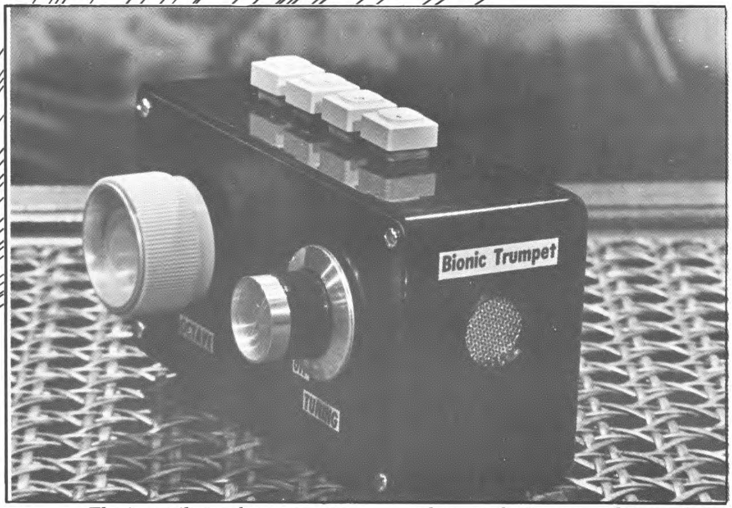
This is an amazingly comprehensive historical study of electronic music from WW II to the present. David Ernst has done a tremendous job of assimilating, categorizing, and correlating a very diverse assortment of information. The author divides electronic music into three categories: solo tape, performer with tape, and live electronics. In each category, Ernst discusses various compositions by small groups, so the continuity or evolutionary relationships between various composers, performers, or procedures can easily be seen. Some of the primary features of the book include:

- A chronological list of pre-48 events related to electronic music provides

..... continued on page 27

BIONIC TRUMPET

By:
Greg
Leslie



In the last Polyphony, I showed you my plans for a saxophone controller/tone source that was interfaceable with all PAIA gear, including the Gnome. This time, I continue my battle against synthesizers designed for keyboard players. This controller is very similar to the saxophone, but it is aimed at valve players: trumpets, french horns, tubas, baritones, etc. It can also be used by people who have never played valve instruments, as this controller requires no embouchure training, and fingerings are fairly easy to learn.

The circuit is shown in fig. 1. If you compare it to the Bionic Sax in the last issue, you'll find a lot of similarities. In fact, the only differences are the ways the keys or valves are encoded. Tone selection is made possible again by two 4051 multi-demultiplexer CMOS IC's. Eight tones are fed into each 4051, and one tone comes out of each pin 3. The tone that comes out is determined by the status of the address lines, (pins 9, 10, and 11). Valves 1, 2, and 3 (actually calculator switches) directly connect to the address lines and provide the tone selection codes. Valve 4 selects one of the two 4051 chips by connecting to the INHIBIT pins on each one. When the fourth valve is open, the top 4051 functions and the bottom one cuts off. When the valve is closed, the reverse occurs, and the tones come out of the bottom chip.

The tones themselves are produced by a 50240 top octave generator chip in about the same way as the OZ. A clock formed by two NOR gates is divided down by a 4024, then fed into the TOG to make nice, clean square waves in an equally - tempered relationship. The switch on the 4024 produces octaves - five, in this case. Then as mentioned above, the output tones are fed into the input lines of the 4051's. Notice the G and F# tones go to two inputs each - that's to provide alternate fingerings (see chart below).

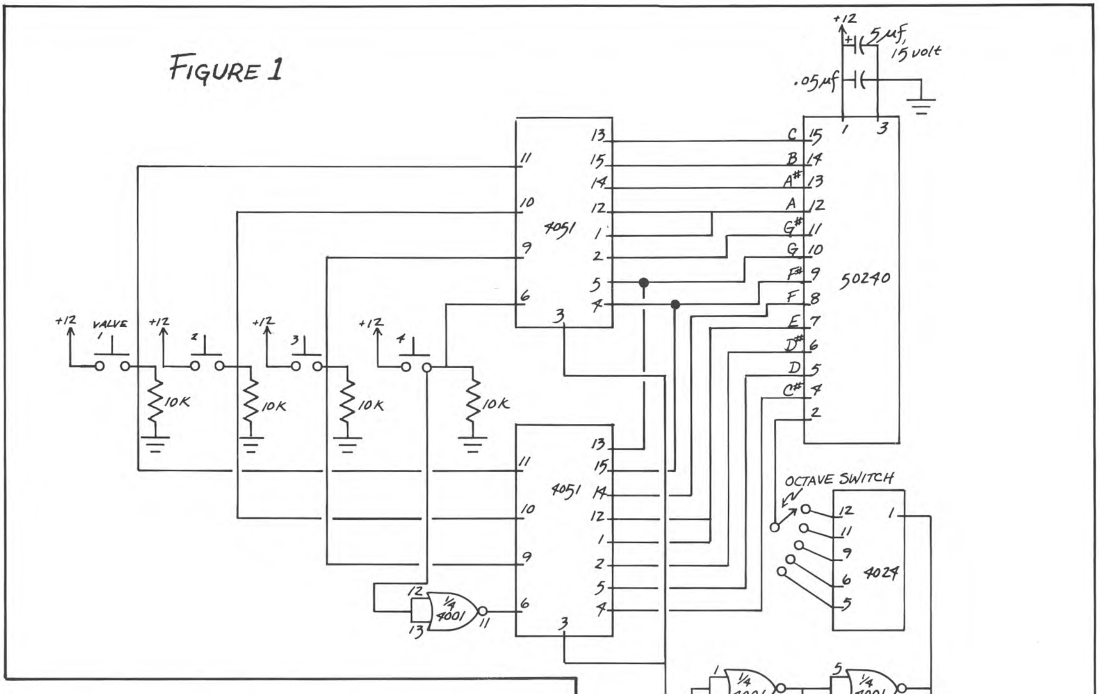
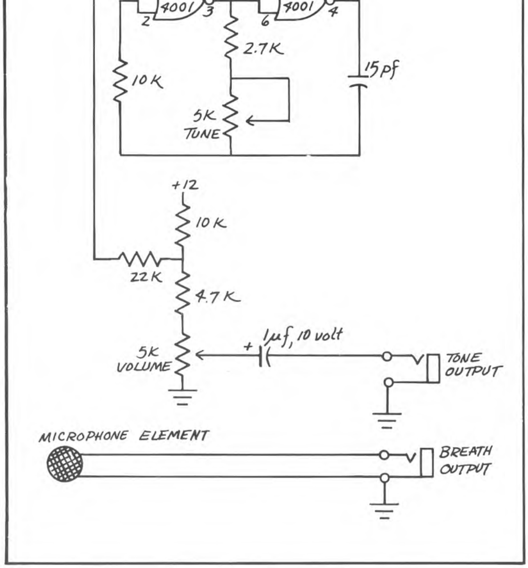
Construction is completely up to the individual: I used a wiring pencil (handy little gadget - I use it more than any other prototyping method now), but you may prefer perfboard or circuit board construction. For a case, I used a medium-sized bakelite box - one that "felt good" in my hand (I got some strange looks from the store manager, too), then installed the circuit board, the battery pack (8 penlight batteries - 12 VDC) and a 5-pin DIN jack (see photo). I mounted the valves on "top" - actually one of the long sides - and the tuning control and octave switch on the left side of the body. The on-off switch is on the tuning control. The DIN jack is underneath, out of the way.

For breath control, as in the saxophone, I used a cheapie microphone element - actually, I tore up a microphone from a cassette recorder and used the top part of the mic body as a bushing to hold the element in place on the trumpet, then used epoxy putty to fasten the whole thing in place and ran a short piece of co-ax from the mic to the DIN jack, along with the output from the circuit (both pins 3).

For the valves, I heartily recommend calculator switches, because they have a good feel - much better than dime store pushbutton switches. To mount them, I drilled two small holes for each switch, one hole per switch lead. Then I soldered small wires to each lead and stuck the wires into the appropriate holes, pulled the switches down flush to the top of the case, bent the wires underneath out (like cinching a resistor on a circuit board), and mashed a glob of epoxy on each wire near the hole to make sure the keys stayed in place. The switches may be hard to mount, but I assure you it is worth it in the long run.

To use the trumpet, I made up a cable that has a DIN plug on one end and two mini-phone plugs on the other, one connected to the circuit, the other to the microphone.

Plug the circuit audio cable into a VCA, and the mic plug into an envelope follower/trigger. Use that trigger to control an envelope generator, which in turn controls the VCA. See patch diagram 1. Set up the envelope you want, turn on the trumpet, adjust the sensitivity on the envelope follower, and start practicing. Much more complex patches can be realized,

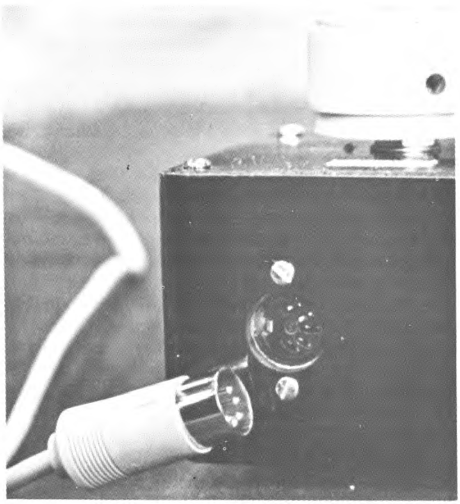


Photo #2 DIN plug and jack carries both the audio from the circuit and the audio from the mic.

like those in diagrams 2 and 3. Dynamic breath control is possible with the controller - let your imagination go wild!

If you have modified the Gnome to adapt to the OZ, you can also interface it with the trumpet, and use the Gnome's VCA, VCF and envelope generators to process the trumpet's audio.

It may take a bit of getting used to the fourth valve if you're already a valve player; some use the left hand to work the fourth valve as well as the octave switch. Whatever feels best is fine. You may even want to mount the fourth valve somewhere else on the horn. A pitchbend plate can be fashioned and mounted on the horn if desired, in fact, there's an awful lot of modifications that can be made, and you'll probably come up with some that will make playing easier for you. If so, drop me a line care of PAIA, and if there's enough interest, we'll print them up in a future Polyphony article. I'd also like other opinions on the subject of alternate controllers for synthesizers. Meanwhile, I'll be working on a clarinet controller . . .

FINGERING CHART

NOTE:	VALVES (1, 2, 3, 4):
C#	1111
D	1011
D#	0111
E	1101
F	1001
F#	0101
G	0001
G#	0110
A	1100
A#	1000
B	0100
C	0000

Note that a 1 indicates that a valve is to be depressed, while a 0 indicates that the valve is still up.

DATA:	NOTE SELECTED:
000	C
001	A#
010	B
011	A
100	A
101	G
110	G#
111	F#

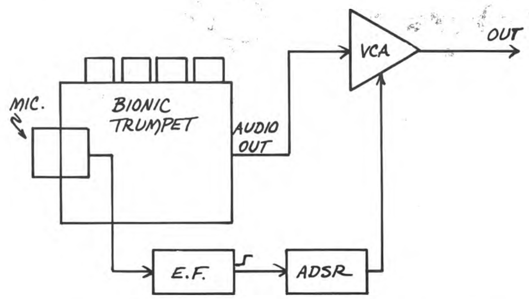
This chart will be helpful for finding alternate fingerings, or for troubleshooting purposes. The binary code given corresponds to the data fed from the valves (1, 2, 3) to the 4051 addressing inputs (AO, A1, and A2) which are found at IC pins 11, 10, and 9 respectively.

Figure IC1 Address Encoding

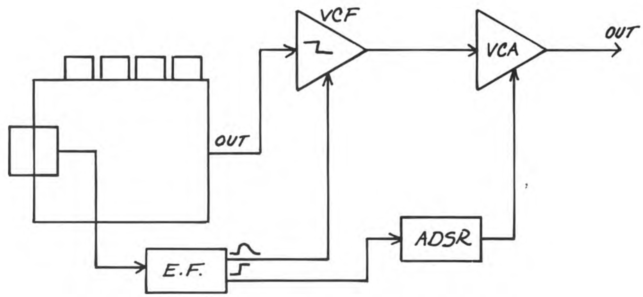
DATA:	NOTE SELECTED:
000	G
001	F
010	F#
011	E
100	E
101	D
110	D#
111	C#

Valve 4 must be depressed to disable IC1 and enable IC2. Otherwise, the data source and addressing pins are the same for IC2 as for IC1.

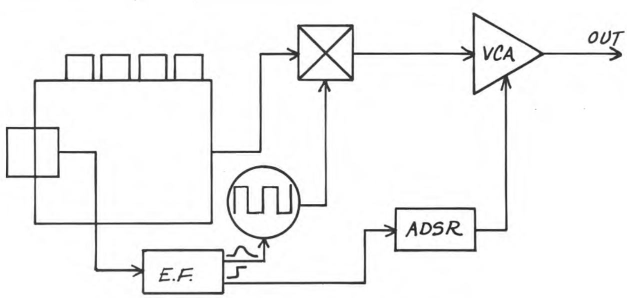
Figure IC2 Address Encoding



USING BREATH CONTROL AS AN ENVELOPE TRIGGER.
PATCH DIAGRAM 1



USING BREATH CONTROL AS A TIMBRE MODIFIER.
PATCH DIAGRAM 2

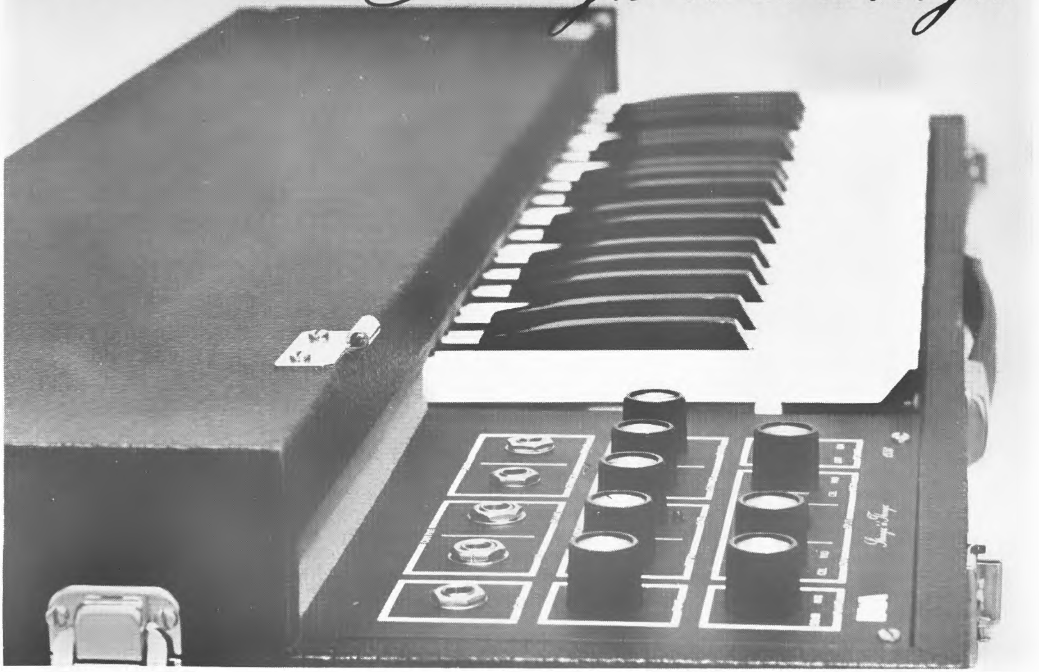


USING BREATH CONTROL FOR MODULATION SWEEPS.
PATCH DIAGRAM 3

BIO-1 KITS AVAILABLE

For those of you interested in working with the Bionic Trumpet, or similar type instruments, Paia has the BIO-1 Experimenters kit. The kit consists of a circuit board, all components which mount on the board, plus the volume and tuning control, rotary octave switch, and a crystal earphone for use as the breath sensor. User must supply the case, calculator switches, and output connectors. The BIO-1 is, at $24.95, an excellent way to begin your own series of experimental alternative controller devices. Polyphony would be most interested in hearing about your experiences.

HERE'S *Stringz 'n' Thingz*



Maybe we should have called it thingz 'n' stringz

by Marvin Jones

String synthesizers are pretty neat. They were one of the first "special purpose" synthesizers to come along. They represented a kind of accpetance, a kind of "growing up" of synthesizers. Strangely enough, the basic circuitry in a string synthesizer is more a development of combo organ technology than of synthesizer technology, but the rich moving sounds they produce are so powerful that most anyone (musician or not) gets a kick out of playing with them. One of the main complaints of the musicians is that the commercial units each have a slightly different sound to them, and none of them happen to sound like what they think strings should sound like..... may I be the first to introduce you to STRINGZ 'N' THINGZ.

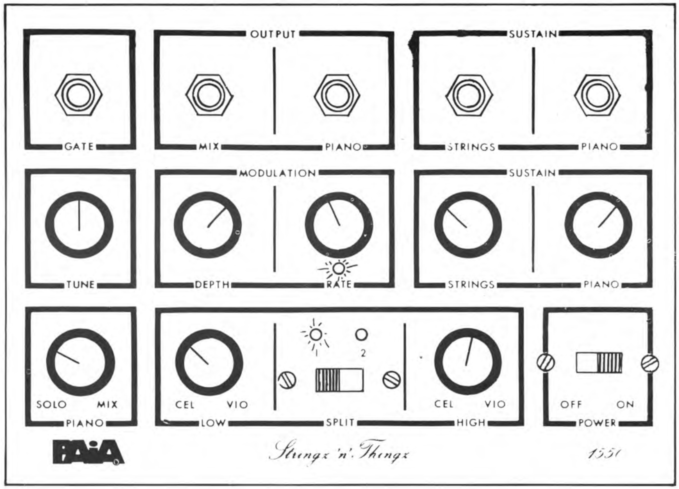
The most important part of this product is not the stringz. It's the thingz. Variable thingz. Optional thingz. Foot controlled thingz. Thingz that make this a FULLY VARIABLE string synthesizer so you can tailor the sound to fit your needs. Or even create sounds that you have been formerly using two or three extra keyboard instruments to obtain. Every effort has been made to make this instrument applicable to as many situations as possible.

A representation of the front panel of STRINGZ 'N' THINGZ is shown in figure 1. Let's take a look at some of the operating features. The GATE jack provides a voltage which steps from 0 to about +9 volts when any key is pressed. This allows interface and triggering of external generation or processing equipment. For example, when the chorusing circuitry is disabled, the normally "stringy" sound is changed to a straight combo organ "reedy" voice. When this bland voicing is processed through external synthesizer lowpass filters, the gate output can be used to trigger the external envelope generator which will be used to sweep the filters. Excellent polytonic synthesizer effects, or brass synthesis can be obtained in this way.

There are two signal outputs on the stock 1550. The first is the MIX output, or the master output. At this jack you will find the sustained, modulated string voicings PLUS a variable amount of the electric piano sound. The user can mix any proportion of piano signal, or none at all. The second output provides only the piano voicing. This output is useful if you wish to use separate mixing, EQ, or amplification for the piano effect. When using a standard 2 conductor 1/4" phone plug in the piano output jack, the piano signal is automatically disabled at the MIX output. This produces a definite separation from the string signal if desired. OR you can use a 3 conductor 1/4" phone plug which has no connection made to the ring section of the plug, and the piano signal will appear at BOTH the mix and piano outputs in whatever proportion YOU select with the piano mix control (explained later). So far, that's a LOT of possibilities for various types of operation. But there's more.

The sustain controls on the panel set the amount of time it takes for the signal to die out once the key is released. For those of you familiar with synthesizer equipment, this would actually relate tc the Release control on your ADSRs. Note that there are separate, variable sustain controls for BOTH strings and piano. Some commercial units have only a switch for long or short sustain, or no control at all. In addition to the panel variables, two jacks are provided for changing the sustain characteristics with a remote switch, control voltage, foot switch, or foot pedal. With the two conductors of the standard 1/4" plug shorted together, the panel control operates as usual to set the minimum sustain time. When the foot switch opens the two conductors, the sustain time goes to maximum (as if the panel control were advanced to maximum).

Various modes of operation can be obtained by using different types of foot switches. If you use a normally closed, momentary contact push button foot switch, the switch will operate exactly like the sustain pedal on a real piano. Press on the switch, long undamped sustain. Let up on the switch, sustain damped to whatever minimum time you designate with the front panel control. Alternately, you could use a push on/ push off type switch to turn the sustain on permanently, and then push again to damp the sustain. Internal design also allows for use of foot pedals (such as the PAiA/De Armond foot volume pedal) or 0 to 5 volt control voltages for remotely programming a variable sustain level, in case you want to get somewhere between the two extremes for a little expression. Like all the possibilities? Read on!



The TUNE control is standard. It provides enough range to not only tune to other instruments, but also to be able to raise or lower the tuning by an octave. With a little work, you can do orchestral glissandos, pitch bends and so on.

Perhaps some of the more powerful controls of the 1550 STRINGZ 'N' THINGZ are the two controls in the Modulation block. The DEPTH control determines the amount of vibrato/chorusing. This can be varied from completely off (for straight organ sounds or external processing) to a very deep rich chorusing effect to a near excessive pitch modulation of over a semitone. While the extremes of this controls operation are definitely out of the bounds of traditional string synthesizers, they are very convenient features to have if you are trying to use this one keyboard to fulfill the foll of several standard keyboard instruments.

The RATE control provides a variable modulation rate from nearly a standstill (good for the fat, rolling sound of a pipe organ), to a normal vibrato effect, and on to a fast quivering vibrato for special effects. The adjacent LED indicates the rate of one of the low frequency vibrato oscillators. It is useful in determining a relative vibrato rate adjustment without having to actually play (and hear) the keyboard. If you are a performing musician, this is a feature that you will appreciate. During a song you will be able to look at the Stringz and determine if it is ready to do a string passage, organ effects, or whatever.

The Piano SOLO/MIX control is used to send the piano voicing to either the master MIX output, or to the solo PIANO output. The control acts as a panning control, and the piano can be assigned to any blend of the two outputs. When using only the composite MIX output, the Piano control acts as a volume control for the amount of Piano present in the MIX output. When a standard 2 conductor 1/4" phone plug is used in the Piano output, the Piano voice is disabled from the MIX output, and the Piano control acts as a volume control for the Solo Piano output. When a 3 conductor 1/4" phon plug is used for the piano output and the ring connection is not used, the Piano Mix control will pan the piano voice between the outputs for some stereo imaging effects.

The large box of controls at at the bottom of the panel is used to design the string sound you desire. The SPLIT select switch is used to desiginate the point at which the keyboard is divided in half. In position #1, the keyboard is split at the first octave (between B and C). In the #2 position, the keyboard splits at the second octave (again between B and C). LED indicators show the selected split position. The LOW and HIGH MIX controls are used to tailor the mixture of violin and cello voices for <u>each</u> half of the keyboard. The low control operates below the selected split point; the HIGH control mixes the voices above the split. Cello voices are two octaves lower than the voilins, and have a slightly wider spaced harmonic series for a bit more raspy or nasal quality as in true string instruments of this type. The two string mix controls take on added power when the stereo string output modification is installed, as the infinite variety of mixes available take on a three dimensional effect.

The internal circuitry has several features built in which aren't apparent from the outside of the unit. By now it is a generally accepted fact that analog delay lines by themselves are noisy. Period. It is a characteristic of their circuitry and operation. But, several types of circuits can be used along with the delay lines to give improved signal to noise ratios, or to silence the chips when not being used. In the 1550 STRINGZ 'N' THINGZ, a noise gate is built into the master MIX output. The attack time of the gate is slightly less than the attack rate of the strings, and the release time is about equal to the maximum string sustain capability. Voila—this circuit in conjunction with the regulated power supply give you the wonderful world of silence between string passages. While the unit is in operation, the delay line chorusing noise is sufficiently masked by the string signal which is much louder than the chorus noise. Other inside goodies include terminal clips at all points where future modifications are to be attached so you don't need to remove any boards or wiring when updating to stereo outputs or digital interface. Just open the case and wire in the new boards. The power transformer used is sufficiently overrated to provide stable operation at a variety of AC line situations, PLUS reserve power for use with the various modification options available.

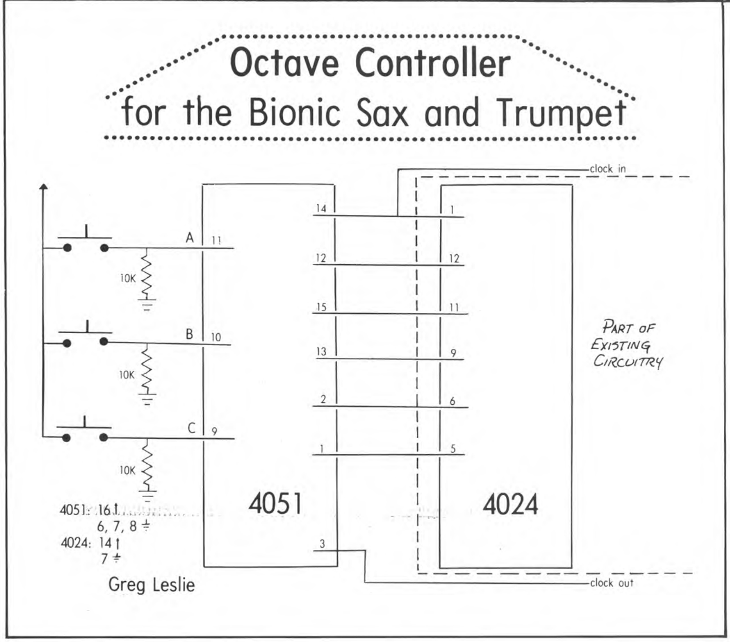
The 1551 Stereo String Output modification adds another set of chorus circuits and noise gate to the original 1550 to provide an independent audio output for part of the signal. In the tradition of the original 1550, here comes some more versatility. You can switch select between two modes of stereo operation. Although you can't see it on the basic 1550 kit, positions are provided on the front panel to mount the extra output jack required for stereo, and the slide switch required for stereo mode selection. The obvious mode of stereo operation is to take the split keyboard and send one half to one output, and the other half to the second output. Flip the switch, and you now send cellos to one side and violins to the other. The second mode tends to add a "wideness" to the sound that gives a tremendous illusion of an orchestra. In both stereo modes, the piano voice is applied to each side of the stereo when the Piano control is turned towards mix. The Solo Piano output works as usual, thus giving you capabilities for 3 simultaneous outputs from this one instrument.

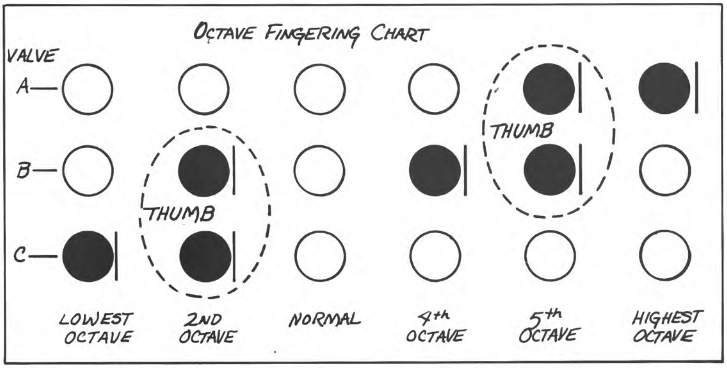
The digital interface is in the midst of production development at the time

.... continued on the bottom of page 20

AN ALTERNATIVE OCTAVE CONTROLLER FOR THE BIONIC TRUMPET & SAX

By Greg Leslie





One of the big drawbacks of the Bionic Sax and Trumpet (Polyphonys, Vol. 3, nos. 2 and 3) is the selection of octaves. In my prototypes, I just used a rotary switch. But this proved extremely difficult to control with any amount of finesse, so I took away the switch, added an IC and a few calculator-type switches, and came up with the circuit below.

The circuit replaces the octave selector in the previous sax and trumpet schematic. The clock is fed in from the oscillator/tuning circuit and the output goes to the input of the top-octave generator.

On the Bionic Sax, the three switches are mounted in a vertical line on the back of the body and are operated with the left thumb. I included a thumb rest alongside the center switch. On the trumpet, experiment to find the point that is easiest to reach and mount the switches there.

Now, when you use the instrument, it will play in a moderate-to-low octave with no octave keys depressed. To get the next lowest octave, press the lower two keys simultaneously (and a well-placed thumb can hold both down if the keys are mounted close together). Pressing the bottom key along gives the lowest octave. Higher octaves involve the upper two switches. For the octave above the "normal octave", hit the center switch. Pressing both the top and center switches simultaneously gives the next highest octave, and pressing the top switch alone gives the top octave.

After getting used to the configuration, this octave controller works well and sure beats the heck out of a rotary switch. If anyone has an alternative idea for controlling octaves, let me know in care of Polyphony.

Here's STRINGZ 'N' THINGZ .....
...... continued from page 19

of this writing, but it's primary features can be discussed so you'll know what to expect. Digitally encoding a keyboard which uses standard diode keying (this includes most string synthesizers, combo organs, and so on) is a fairly easy task, and once accomplished the same keyboard can be used to feed a microprocessor for purposes of composition storage or controlling a polyphonic synthesizer. Think of it-- you already have a keyboard that will do stereo strings, electric piano, and can be processed externally for synthesizer-type effects. Add a circuit board and the same keyboard can control a compliment of modules to produce up to 16 voice polyphonic work.

Here's more:
Data can be fed into the string synthesizer and actually play the strings for you. You could store the string or piano parts of your composition in the memory of a microprocessor, and have the computer play your arrangements on command. At any speed you desire. In any key you desire. Even backwards if you want, and more and more...

A few of you have known for quite a while that we were working on a string synthesizer. It's been a long time. Sorry 'bout that. But, I think you'll agree that it's been worth a wait. You're gonna' love it!

AN ULTRA VCO FROM THE 4720

By: Gary Bannister

The 4720 Voltage-Controlled Oscillator does a super job as it comes from the factory. While guaranteed to go from 16 Hz to 16KHz, they typically perform much better. Several units I have tested ran from 4Hz to 20KHz. This is not really amazing, considering the design of the unit. With regulated power supplies, the combination of precision current source, and temperature-compensated integrator produce an oscillator of exceptional range and stability.

However, PAIA does not yet produce a voltage-controlled CONTROL oscillator. Requirements of such a unit would be low frequency (typically 0.5Hz to 50Hz), DC coupling (no blocking capacitors), and compatible signal level (0 to 5 volts). A super unit would include both positive-going and negative-going signals useful for panning or dual filter sweeps.

The obvious difference between this and the 4720 is the super low frequencies. The design of the 4720 makes it easy to produce such a response by simply replacing capacitor C2. If the original capacitor and the new one are placed on a single pole double throw switch, each range may be easily selected. (See Fig. 1) The value of the new capacitor is selected to give the desired range. Changing the value by a factor of ten (to .1 uf) will drop the frequency for a given voltage by ten, also. Using a 1 uf cap will drop the frequency to 100. At this rate, a typical oscillator will produce a lowest frequency of .04Hz, or ONE CYCLE EVERY 25 SECONDS! Be sure to use good quality capacitors, preferably mylars and NOT electrolytics. Maybe a tantalum would be good for VERY LOW frequencies.

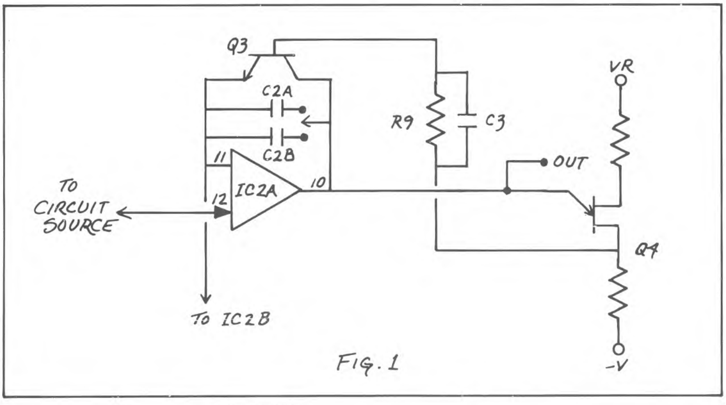


FIG. 1

With a range switch suitably installed, a new problem develops. The new low frequency waveforms viewed on an oscilloscope begin to get a peculiar distortion, especially when both the control voltage and range pot are at maximum. (See Fig. 2) In fact, this same distortion appears with the normal capacitor in place as well.

This is obvoiusly a form of harmonic distortion-the waves are not pure. This distortion is also a sign of linearity error. This means that the oscillator goes slightly out of tune with the range pot maximum. We don't notice this as much when the oscillator is in the audio range because the harmonic distortion is too high to hear, and the slight detuning actually produces sometimes pleasing "phasing" effects. (Please note that this distortion appears only above about 10KHz in the audio range.)

However, it should be easy to see (hear?) what this distortion would do if we were to use the sine output for vibrato. The results would be less than musically pleasing.

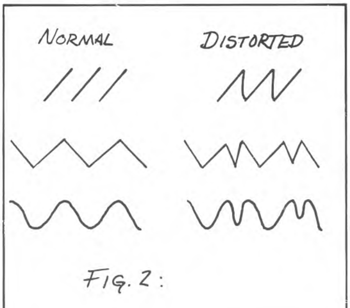


FIG. 2:

Since we also find that this distortion appears in the low range (but at a frequency about 10 times lower-approximately 1000Hz), we have a clue to its cause, and therefore its remedy.

Without repeating the operation of an integrator (which is covered in the 4720 Owners Manual), let it be said that if a large current flows into the integrator, a suitable large current must flow out of the integrator. The problem lies in the fact that Q3 cannot short out this current fast enough. This means that the fall time of the sawtooth is no longer infinitely short, but now is a good portion of the total waveform. Not only this, but the amplitude becomes less, and a DC offset is produced which ultimately affects the triangle and sine, and makes the pulse width unstable.

The remedy for this problem is to find a transistor which CAN short this current fast enough. What makes a transistor suitable for this purpose is known as High Frequency Cut-Off (f_T in transistor substitution manuals). A good suitable replacement is 2N3641. This has an f_T of about 300MHz (300,000,000Hz!). A special aircraft transmitter transistor was also tried with more than excellent results. Its f_T is better than 1 Gigaherz-1,000,000,000Hz! Also of slight importance is the combination of R9 and C3 (See Fig. 1). Some improvement is made by removing C3 and changing R9 to 10K.

Now, with some minor realignment of the Pitch Range and Triangle Symmetry controls, the 4720 is capable of a frequency range of about 20 OCTAVES in two overlapping ranges. As a fringe benefit the oscillator stays better in tune through its total range.

These low frequencies are not really of much use since we can't hear them. At least, if we can't hear them, then we should be able to hear their effects. Vibrato is obvious, but a good low frequency sine wave is great for sweeping a Phlanger.

To get these effects we need an oscillator with 0-to-5 volt swings. The normal output of the 4720 is .5 volts, but there is an easier way.

When building the 4720 the set-up is to adjust the sawtooth for a 4.5-volt swing. VOILA!! Here is one output that needs no amplifying, only simple buffering. By measurement, we find that there is a pulse wave available that is also 5 volts. Further measurement finds a 2.5-volt triangle and a 2-volt sine. Fig. 3 shows where in the circuit these are available.

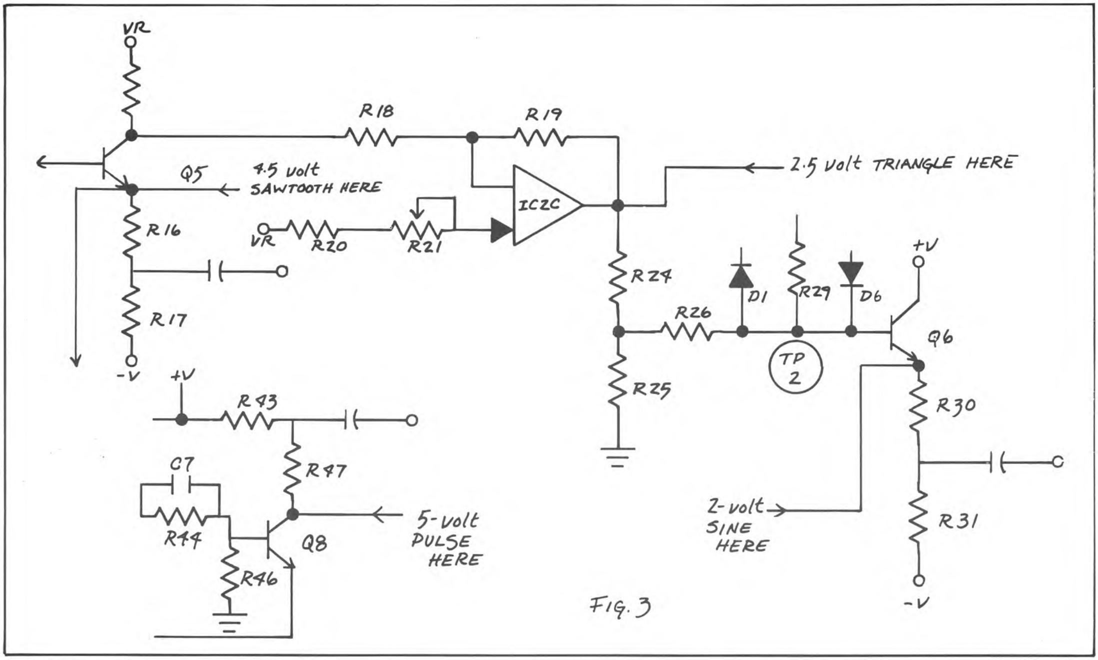


FIG. 3

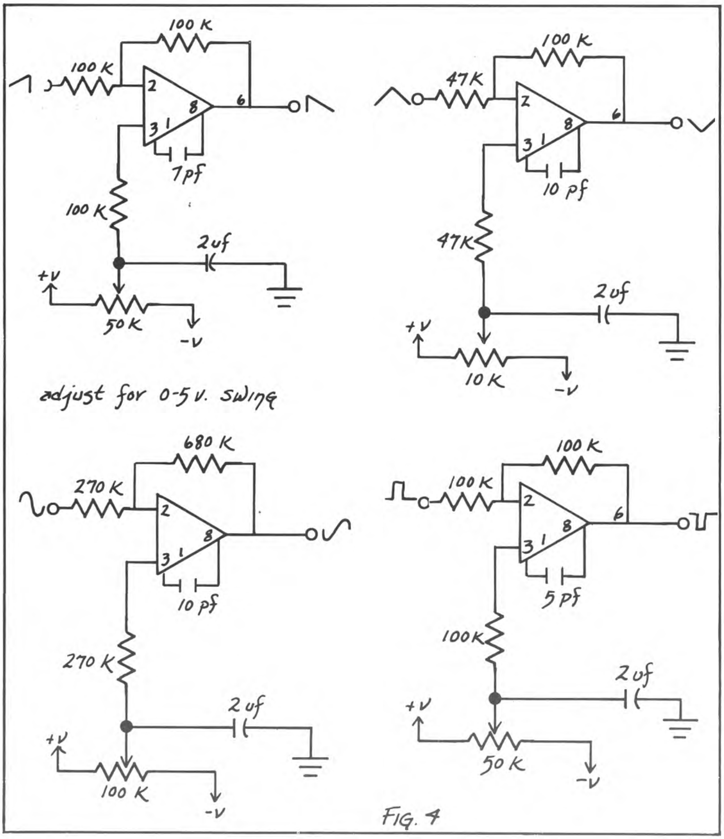


FIG. 4

Now, all we need is some simple amplification to get the 5-volt swings, and some level shifting to get the swing to go from 0 to 5 volts. Fig. 4 shows four such circuits. Note that all op-amps are of the 748 variety. Some non-standard frequency compensation is used to keep the fall times short. When finished, each unit should be adjusted so that the waveform swings from 0 to 5 volts. The 2uf Capacitors are optional, but may be necessary to keep any hum in the power supplies from getting in and being amplified.

Notice that all the buffers in Fig. 4 are inverters. This is the easiest form to work in with this use, and the upside-down waveform is of little importance.

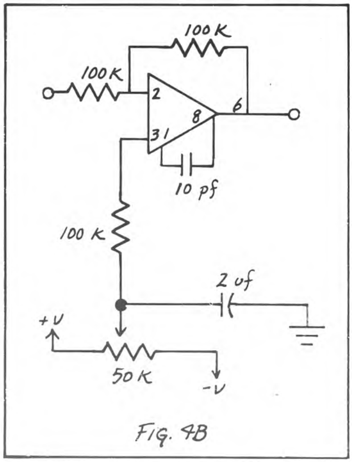


FIG. 4B

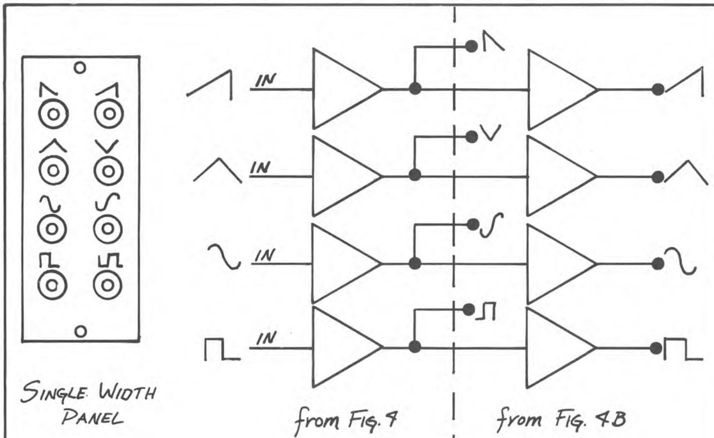


Fig. 5A

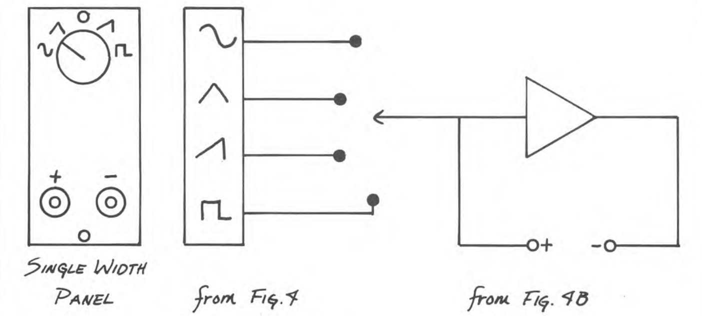


Fig. 5B

If the original form of the wave is necessary, it is a simple matter to add another inverter. Fig. 4B is such a unit. Any waveform may be put in, and the offset pot adjusted for 0 to 5 volt output (without it the swing would be 0 to -5 volts). A quad op-amp such as a 4136 would be useful in this application.

Now, our goal is reached. We have a super oscillator with a 20-octave range, control voltage outputs, and inversions available (known as differential outputs). All this and we can still use all four AUDIO outputs at the same time.

How do we use it? Several additional modules suggest themselves, all of them similar to the Sine converter for the 2720-2A. Fig. 5A is a control voltage module. It needs 8 op-amps, but all waves and their inversions are available at the same time. Fig. 5B has a rotary switch to select one wave and its inversion. This needs 5 op-amps. In either case there is no provision for variable output, but a 5Kohm pot to ground is a simple chore.

Other units suggest themselves. I am using a separate container and power supply for a super unit. Audio, control, and inversions are available. A bias supply is included so that the oscillator may be run by itself. Shut off the bias and the normal VCO response is available. The outputs are variable, and an offset switch is planned so that the outputs will swing 0 to 5, 2.5 to 2.5, or 0 to -5.

Uses of this VCO are varied. Easily done is a vibrato that increases as the played pitch increases. A super-slow sweep for the Phlanger is possible, and may also be applied to the filters as well. How about sweeping the filter at an AUDIO rate? Unbelievable phasing, rolling timbres are possible if the oscillators track each other at a harmonic interval. Gongs and chimes do not detune through the ring modulator. By producing vibrato at an audio rate that tracks the pitch oscillator, even thicker chimes may be produced. The thickness of the chime depends partly on how much the pitch is modulated. With two of these oscillators some ultrasonic effects may be possible through the ring modulator, giving rise to eerie shortwave sounds.

Truly, the 4720 is a super oscillator.

SOME QUICK THOUGHTS: Can you computer musicians see a way to digitally select the range capacitor? If so, how about computer selection of octaves by using several capacitors and range trimmers for R8?

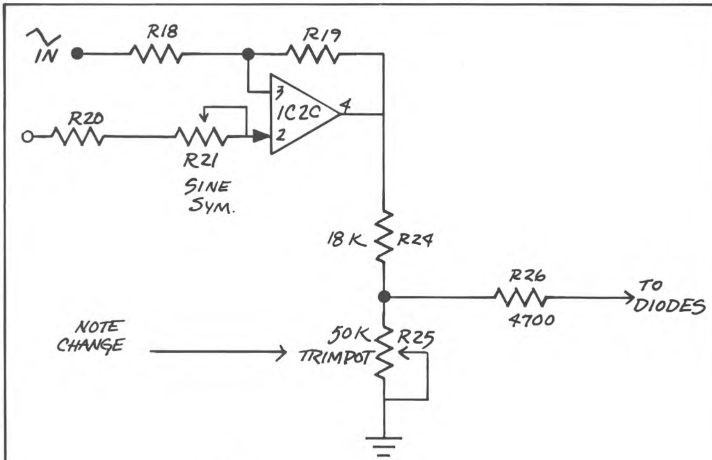
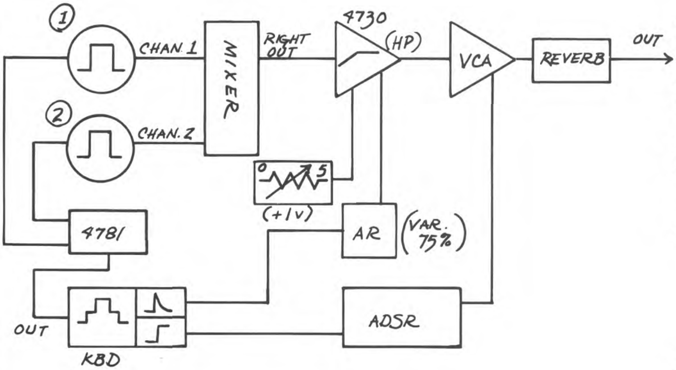


Fig. 6

PATCHES

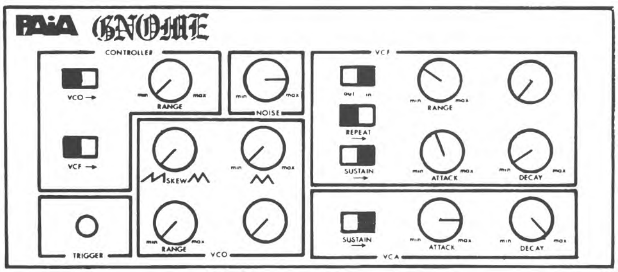
SOUND USED DURING CARROUSEL FROM "LOGAN'S RUN"



Keyboard:	Tune to middle octave range.
VCO 1 & 2:	Tune to unison or near unison with wide square waves.
Bias Supply:	1v (approx).
Mixer:	Both inputs equal amplitude.
Filter:	High pass output.
Mode:	Track "Q" at 90% (approx).
Init. Freq.:	High at 90%.
AR:	Attack - 50% Decay - 75% Var - 75%
ADSR:	Attack - 25% Decay - 50% Sustain - 75% - 100% (controls overall loudness) Release - 100%

Submitted By: Dave Smith
VAFB, CA

ON THE ROCKS



COMMENTS:
Set VCF Range for a just-discernable sweep.
Set VCF Decay for a just-noticable "crack" of the surf on the rocks.
Experiment with timing by varying both Attacks and both Decays.

Submitted By: Terry Fitzpatrick
Independence, Kentucky

PATCHING THE 3750 DRUMMER FOR A 6/8 SWING

MAIN PATTERN

Prep: Reset, Program.

Data: AB, R, R, S, R,
Co, B, R, R, Co,
T, Cl, AB, R, S,
Co, R, S, B, S,
R, R, R, Cl, Repeat,
Reset.

BRIDGE PATTERN

Prep: Reset/Bridge, Program

Data: R, S, S, Cl, T,
R, R, S, R, T,
Co, B, Repeat,
Reset.
. . . and then - Play.

COMMENTS:

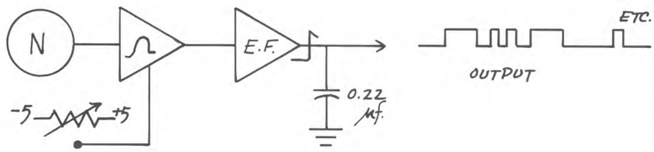
The drum data abbreviations are as follows:

AB - Accent Bass
B - Bass
S - Snare
T - Tom
Co - Conga
Cl - Clave
WB - Wood Block
R - Rest.

Other programming steps are self-explanatory.

PATCHES

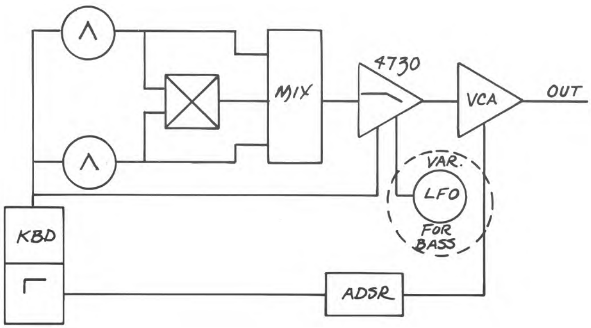
RANDOM PULSE GENERATOR



Experiment with bias setting for various random characteristics.
A random output from the E. Follower.
The .22 cap eliminates clicks.
Filter - "Q" max.
Follower - min. sensitivity.

Submitted By: Josef Wittman
Bennington, Vermont

FRENCH HORN OR BASS BEING PLUCKED



<u>For Horn</u>

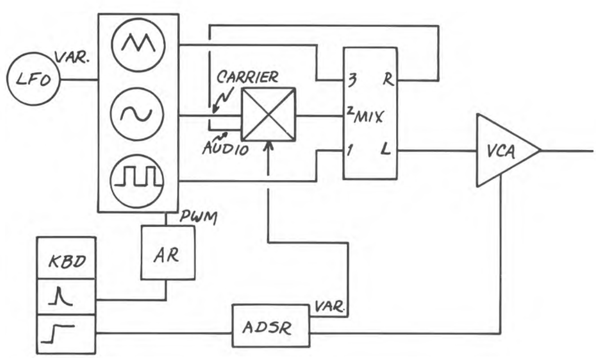
VCOs:	One octave apart.
MIX:	Equal.
4730:	LP, Init. Freq. 75%, Q 60%, Range - High, Track.
ADSR:	A 25%, D 40%, S 75%, R45%, Best in mid octaves.

<u>For Bass</u>

4730:	70% Init. Freq, 100% Q.
ADSR:	A 10%, D 40%, S 50%, R 75%.
LFO:	7 Hz., very slight Low octave only.

Submitted By: Kenneth Libby
Roseville, MI

SOLO VIOLIN, CELLO, & BASS



AR:	Exp. off, A 60%, D 60%.
ADSR:	A 40%, D 40%, S 70%, R 70%, var. 60%.
MIX:	1 max. 75% left 2 max. left 3 max. 80% left Right 80% Left 70%
LFO:	7 Hz., very slight Higher octaves violin, Lower cello, Lowest Bass. (Perhaps Bass should be run through an LP filter.)

Submitted By: Kenneth Libby
Roseville, MI

MODIFYING THE MU-TRON BI-PHASE FOR CONTROL VOLTAGE INTERFACES

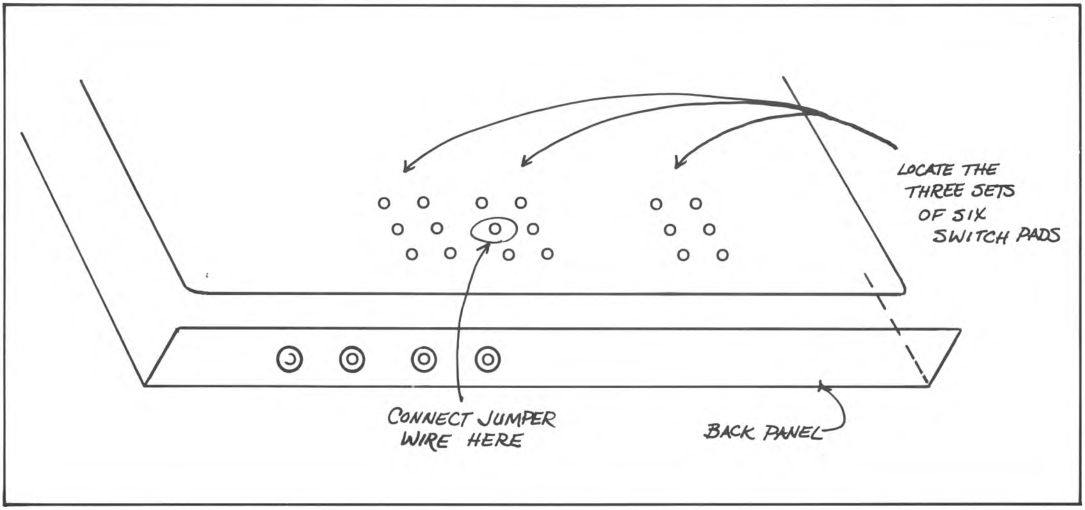
By: Jim Riter

Those of you who keep up with the various new processing items that are available commercially will recognize the Bi-Phase which is manufactured by Musitronics of Rosemont, New Jersey. This is one of the more elaborate of the hundreds of phasors which have been put on the market over the past two or three years. It features two independent phasors which can be run in series or parallel, and swept with your choice of waveforms. Despite the physical size and array of front panel controls, this unit is easy to modify so you can tap into the internal sweep oscillators or feed your own sweep signal into the phase shift line. There are even two unused holes in the rear panel of the Bi-Phase which are the perfect size for mounting the 1/4" jacks required for the interface!

Mu-tron has a voltage producing photocell pedal as an optional accessory that can be plugged into the footswitch jack (5 pin DIN type connector). It puts out 0-5 volts peak-to-peak, and it will provide higher center sweep (if the Bi-Phase's switches are so set) or more rapid LFO oscillation (see owners manual for getting these effects) as voltage is increased. The trick is to take advantage of the fact that the DIN plug on back has both a voltage send and recieve terminal for that pedal, and tap off that control voltage input and wire it to a jack instead. The reason you can't just "plug in" to that DIN jack is because both of the footswitches provided stock with the Bi-Phase must be plugged in and "on" (so the LED's are "on") for the phasor to operate. So leave the DIN jack alone, you'll still need it.

Before beginning the modification, remember that altering a unit such as this will most always void your warranty from the manufacturer. If that's OK with you, then here we go. Internal access is made by removing the 3 phillips head screws in the front lip of the case, the 4 screws on the top (front panel), and the 4 screws on the rear panel. The green wire in the DIN plug is the control voltage input for the phasor. One can verify this by experimentation if you wish to double check it. Solder a wire to this DIN jack terminal, and connect the other end to the "hot" connection on a 1/4" phone jack. Mount the phone jack in one of the available rear panel holes. This jack will be grounded by virtue of the fact that the case is already at ground. Thus, no ground wire will need to be connected to the jack unless you wish. In this case, the power supply is in a seperate section, and is not far from the rear panel. A good solid grounding point can be found there.

The next step involves taking a jumper off a trace that has the LFO signal on it, and running it to the second interface jack. If you look at the front of the panel you will notice the "shape" switches that choose a sine or square wave for modulating the phase shift lines. The Bi-Phase should now be turned over, and the rear of the case should be facing you. The circuit board pads which connect to the three rocker switches are easily visible as three groups of SIX pads each. The switch pad which is in the middle of the LEFT row of three for the center set of switch pads is the specific pad which we will be using. (See drawing) Connect a wire from this point to the "hot" connection of your second 1/4" interface jack. The modification is complete.



A wild effect is to have an ADSR control the rate of Bi-Phase sweep speed. Also, try a quick attack and decay controlling the phasor's center setting. The Bi-Phase's LFO gives you a beautiful sine wave for modifying flangers, vibrato, or filter sweeps. The square wave can be used to trigger sequencers, envelope generators, trill VCOs and bunches more. I've had a blast with one of these interfaced to my PAiA system, and I bet you will too!

The first jack you installed is the Sweep Control Voltage Input. The second jack is the Internal Sweep Oscillator Output.

LETTERS!

..... continued from page 3

ADDRESSES & SUBSCRIPTIONS

Friends!! (romans & countrymen even!)

I have a few minor suggestions with regard to 'Polyphony'. I look forward to getting it whenever it does come, as it always seems to spark new and fresh ideas that send me running back to my 2720/R, just when the thing was getting to be a wee bit . . . dare I say it? . . . boring!!

However, since the July issue, two things have cropped up to nag me. One has to do with the articles that are sent to you. It would be a great help if on all 'how to' or 'try this' type stuff, you print the Full address of the writer!! Why? (you may ask?) Because sometimes people like me, who have just enough electronics know-how to get by, sometimes run into problems. For instance, the article by John Blacet and his Frequency divider. I constructed this circuit three (count 'em) times, and have yet to get it to work properly! I don't know if it's an error on my part, a printing error on your part, or a design error on his part! If I could only write him and perhaps ask for advice! Another example. Ken Winograds Random Tone Generator. His address was printed and luckily I wrote him. He points out that there were errors in the published version, and sent the corrections! (Great guy, this Kenny!) I'm sure it saved me a lot of headache's!

Item #2 is more on a personal 'hey you guys' level. Namely, that I'm not certain if I'm getting all the Polyphonies published!? Before the July issue I recieved the #4-1976 issue. Then for six or seven months, nothing. Now the July issue. However I subscribed to Polyphony in Jan, 1977! (The 1976 issue was a courtesy issue I assume!) Have I or have I not missed an issue (or two). If so . . . why?! If not, aren't you a little behind?

Believe me, knowing the answer to this will help me sleep better nights. Who can stand being left out of all the PAIA latest ideas? Take what I have written into consideration okay! And perhaps even publish an answer in Polyphony. I'm sure there are others who have the same ideas as myself!

Till Then
Ron Jones

RESPONSE

Ron (and other Polyphony readers),

Your letter brings up a couple very valid questions. There is a specific reason we do not publish the full addresses of some of the people whose names are mentioned in Polyphony. Namely--respect for their privacy. Some people don't care to have their vital statistics publicly announced. For those that do, we have the Local Happenings column. At any rate, we will NOT print a readers full address unless he specifically asks us to.

With regard to problems you encounter when working on ideas or schematics printed in Polyphony, it is common courtesy for most magazines (including Polyphony) to aid their readers in successful completion of published projects. This means, if you are having problems, contact us here at the Polyphony offices. In most cases we will be able to answer your questions; if not, we will forward your query to the author in hopes that he may be able to help. To specifically answer your question concerning the Blacet Frequency Divider, I know the design and printing were OK for this circuit. We have built several or them around here, and I have heard from several readers who have built the circuit with very good results.

Now . . . concerning subscriptions; don't worry. Yes, we are running a bit behind, but the forecast and Farmer's Almanac say that we should be back on schedule by the first quarter of next year. Keep your fingers crossed, or better yet-- send us your favorite patches, circuit modifications, computer programs, or anything you want to get off your chest. We'll print it, and the more material we have to select from-- the faster we can put an issue together. At any rate, you won't lose any money from your subscription, because we consider a subscription to be for FOUR ISSUES. You won't be asked to renew until our computer says you have been mailed four issues. OK?

Marvin

POLYPHONY REVIEWS!

..... continued from page 14

an historical background.
- It is the first non-technical examination of specific categories of electronic music based on sound sources and compositional techniques.
- Ernst analyzes major compositions by Stockhausen, Berio, Ligeti, Reich and the Paris school of music concrete.
- Graphic Illustrations of selected works aid the reader in listening to these compositions. Additionally, many of the composers or performers techniques are illustrated for an understanding of the hardware manipulations involved.
- The appearance of voice in electronic music is documented.
- The use of electronics in jazz and rock is explored.
- Over 90% of all electronic compositions that are available as recordings are discussed, and each chapter has an extensive discography of albums discussed in or relating to the chapter.
- The extensive bibliography presents readings in acoustics, computer music, available scores, and general electronic music.

"The Evolution of Electronic Music" is by far the most comprehensive, yet specific, book I have seen concerning the development of electronic music compositional and performance techniques. It really helps tie our predecessors work to what we are doing now.

Marvin Jones

Local Happenings

The PAIA users club originally started by Joe Lattanzi of Heggins, PA, is being reorganized by Dan Mocsny, 707 Park Ave., Springdale, OH 45246, phone (513) 851-2155. Those interested in more information about the club should contact Dan directly.

LOOKING FOR SERVICE OR CONSULTATION? We recommend contacting the following local representatives:

Doug Slocum
Synthetic Sound Labs
1 Gale Rd. (NEW ADDRESS)
Bricktown, NJ 08723
Phone (201) 477-3319 (NEW PHONE)

Gary Bannister
7205 New Augusta Rd.
Indianapolis, IN 46268

....continued on page 35

THE POLYPHONIC SYNTHESIZER

By
John S. Simonton, Jr.

LAB NOTES

We've come a long way over the last year in terms of developing a series of digitally interfaced modules that will allow computer control of music synthesizers. I suppose that the time has come to look at tieing them all together, with the computer, and begin doing interesting things.

I had wanted to start with "the ultimate sequencer programs" but am not completely happy with them yet. They still need a little polishing.

Instead, we'll start with what should be another popular system:

THE POLYPHONIC SYNTHESIZER
Which is a much simpler job than the ultimate sequencer.

I would like to go through the system showing specific ways to do things for a variety of manufacturers equipment but that just isn't practical. Instead, we'll look at a completely PAIA based system and assume that if you are using different equipment you are familiar enough with it to make whatever changes are necessary.

Oh, one more thing before we begin, be sure that you understand that there are a wide variety of ways to do polyphonic synthesizers. This is only one of them. I hope that the algorhythm used here works for you. It's one of many, some with sort of special quirks that make them useful in certain situations but difficult to work with generally - This seems to be good general purpose way. Ready? We have lots to do and little space and time; here we go.

THE HARDWARE

Most of the hardware that we'll be using has been described here over the last year (or so). For the controller portion of this system we'll need:

1) AN ENCODED KEYBOARD
8782 or EK-3 retro-fitted equivalent
2) A COMPUTER
An 8700 in it's minimum configuration will run the programs that we'll list. A cassette interface system is useful to the point of being almost mandatory. We'll show some new panels and stuff to make it all pretty.
3) DIGITAL/ANALOG CONVERTER AND SAMPLE AND HOLDS
the 8780/8781 system.

And, of course, we'll also need as much synthesizer as we think is necessary.

With all of the items listed, various wiring schedules have been mentioned for doing various non-computer things. We now need to establish some standards for this new use, a computer based polyphonic system.

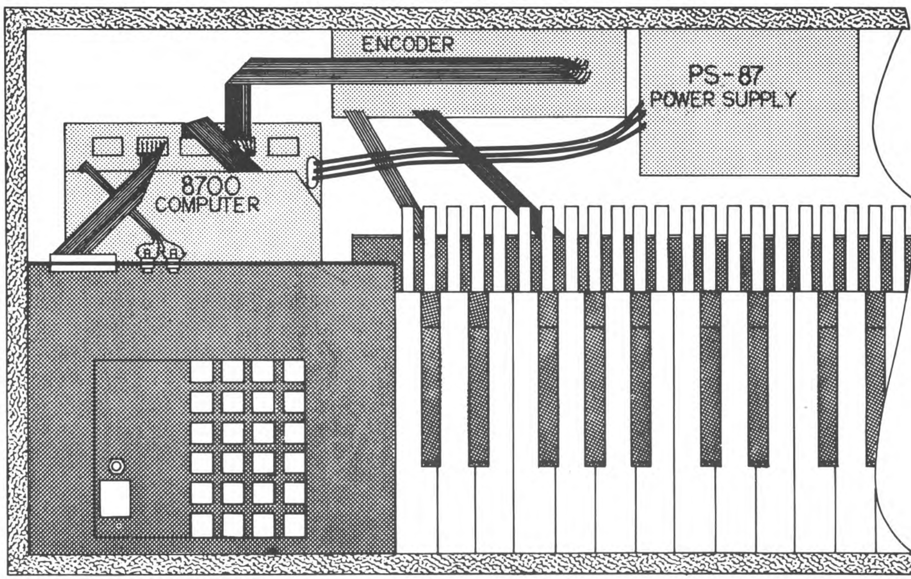
If we choose wisely, we should come up with a standard that has plenty of room for future growth. Some consideration has gone into the system which follows and I believe that it will serve our needs for some time to come.

Many of you will already have much of this wiring done, as much of it is simply an extension of what we've done before. Check carefully to be sure your wiring is to this new standard.

THE KEYBOARD

Let's go ahead and configure this system from the beginning so that the computer fits in the synthesizer cases that we've been using. All of the parts will fit in the case like this:

figure 1. computer/synthesizer sub-module placement.



PAIA 8700 COMPUTER, POWER SUPPLY AND KEYBOARD ENCODER RETRO-FIT TO 4700 OR 8700 SERIES KEYBOARD.

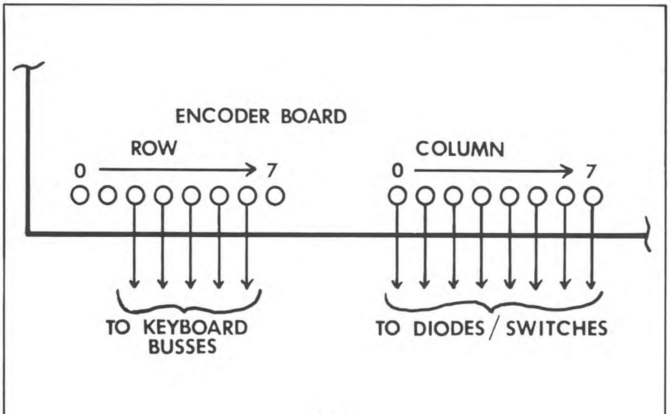


FIG. 2

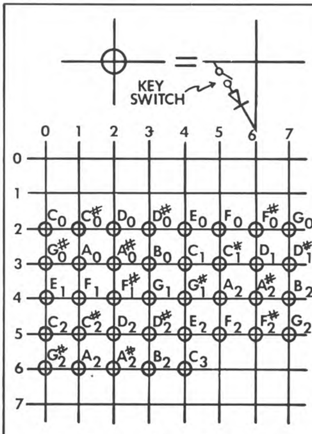


FIG.3: The Keyboard as Matrixed

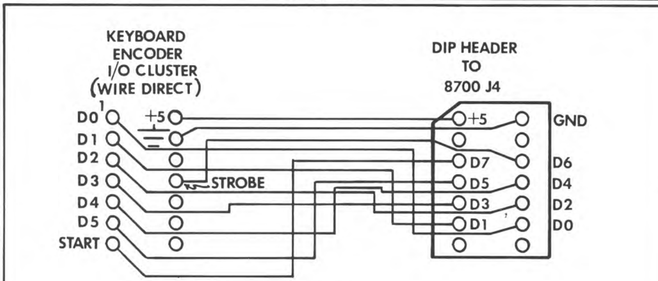


FIG. 4

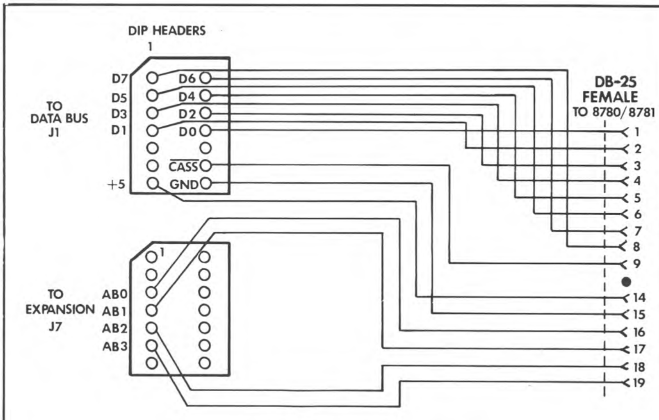


FIG.5: COMPUTER TO 8780/8781

At this stage you may have more dis-assembly to do than assembly. Particularly, the old control panel of the keyboard is removed to make room for the computer and any unregulated supply that was powering your keyboard encoder is replaced with a PS-87 which supplies all digital power for the entire system. This is going to give you a few parts for your "bench stock", the old power supply components and a couple of push-buttons, but some of the parts we will be re-using. Don't throw anything away.

KEYBOARD TO ENCODER CONNECTIONS

Maximum useability of the system would seem at first to depend on where the AGO keyboard switches appear in the key matrix. We want them in the middle so that we have as much room to transpose down in pitch as we do for up-scale transpositions. Some 8782 instructions had the keyboard placed 8 switch positions below where it should be for this ideal. The "column" connections are fine, but the "row" connections on these keyboards will need to be "slid up one" so that they conform to the configuration as shown in figure 2.

This will place the keyboard more or less in the middle of the matrix as shown in figure 3. This is really a fine point, and the system will work OK in most applications almost no matter where in the matrix the keys are, but go ahead and change now so that you won't be limited in the future.

ENCODER MODIFICATIONS

We don't need any of the "trick" things that we used when we didn't have a computer (the orgasmatronic glide circuit, etc.), just the bare-bones encoder. You may remove all push-buttons slide switches, pots etc.; most of these will come out when you remove the old front panel.

ENCODER TO COMPUTER

If your system previously had a DB-25 female connector tied to the output of the encoder, desolder it (carefully - whistling may make the job seem easier). In place of the DB-25 connector, we now need to terminate the output of the encoder in a DIP header that will mate with the INPUT PORT #1 (J4) connector on the rear edge of the 8700 computer board.

These connections should be made as in figure 4.

These connections should also be made carefully and the DIP header pins well heat-sinked to prevent melting the plastic header. NOTE that while many of the non-computer applications used the STROBE line to trigger the D/A, here we ignore this line and instead use the STROBE as the seventh data bit (D6) of the interface.

Similarly, the encoder's START line becomes the 8th data bit (D7).

Also, you will notice that power to

the encoder is picked up through this connection from the 8700 itself.

COMPUTER TO SYNTHESIZER HEAD

So that our resulting system can be easily broken down into two separate units (computer/keyboard and synthesizer head), this is the place to use the DB-25 connector that was salvaged from the old keyboard front panel.

Connections should be made between the female DB-25 connector and a pair of DIP headers like those in figure 5.

NOTE that the first header (P2) provides data lines and the $\overline{CASS}$ select signal (our 8780/8781 shares this output structure) while the second connector (P3) provides the address lines required by the QuASH.

8780/8781 WIRING

The male DB-25 connector that terminates the cable to the 8781 is wired in what is essentially an expanded version of our previous standard so that here you are faced more with adding wires than re-arranging them.

Connect these elements together as in figure 6.

This wiring schedule is examined in detail in the 8781 QuASH assembly manual. An important thing to notice here is the way the grounds are handled. Note that the ⏚ (ground) pin on the rear of the 8780 board serves as the central ground for both analog (synthesizer) and digital power distribution. This grounding scheme is important to prevent ground loop problems and should be followed exactly. This entire 8780/8781 assembly should be mounted in the synthesizer head cabinet.

FINAL ASSEMBLY

Finally, make arrangements for physically mounting the computer in the keyboard case by first mounting the computer to a suitable front panel as shown. (See figure 7)*

And don't forget to provide a socket at the 8700's expansion connector (J7) or to mate P3 with this socket before assembling the computer/front panel. If the cassette interface is being used, terminate the input and output lines in miniature phone jacks as shown in figure 8.

Plug all the connectors together and you should be ready to load a program.

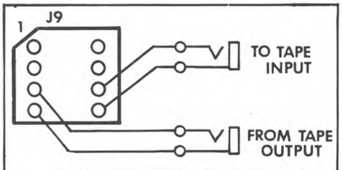


FIG.8: CASSETTE CONNECTIONS

THE PROGRAM

The polyphonic program that we'll be using is called simply:

POLY 1.0

This program supports up to 8 output channels the way that it is written and can be easily modified to provide for more.

POLY 1.0 allocates synthesizer resources to keyboard requirements using this algorhthm:

1) Output all notes appearing in the output buffer area (NTABLE) after adding the corresponding transposing figure from TTABLE. Go to 2.

2) Wait for keyboard scan to start and place a list of all keys currently being held down in the input buffer area (KTABLE). When buffer full or scan complete go to 3.

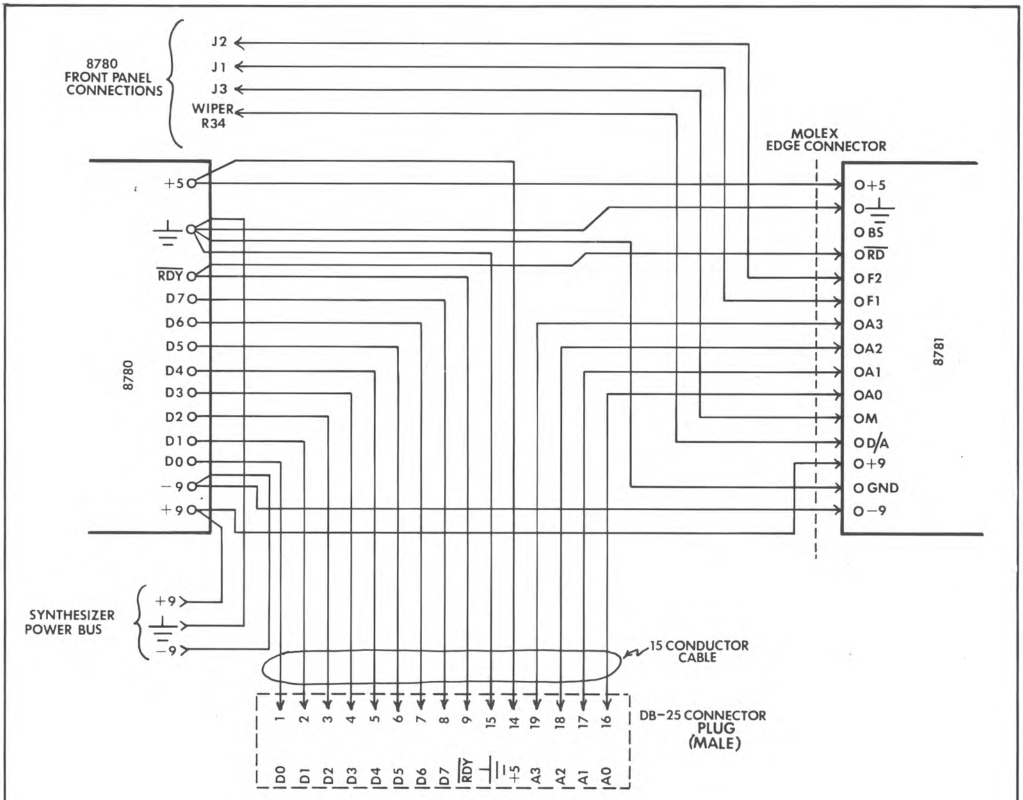


FIG.6

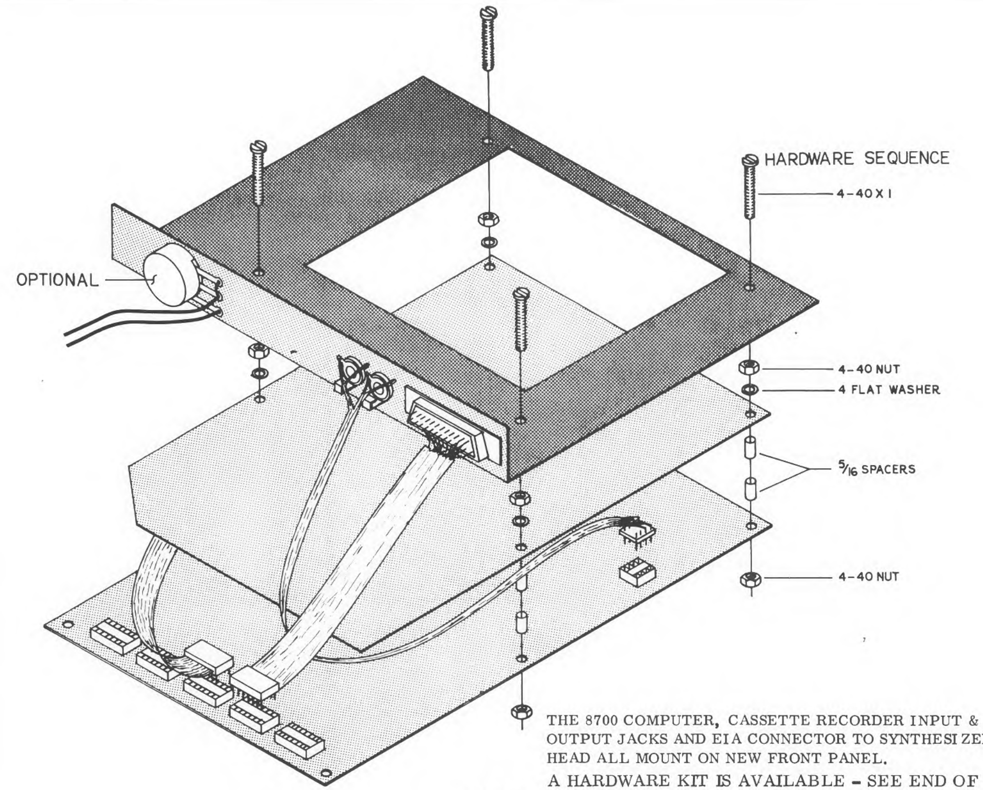


FIG. 7

THE 8700 COMPUTER, CASSETTE RECORDER INPUT & OUTPUT JACKS AND EIA CONNECTOR TO SYNTHESIZER HEAD ALL MOUNT ON NEW FRONT PANEL.
A HARDWARE KIT IS AVAILABLE - SEE END OF ARTICLE

3) Clear the trigger flags (D6) of all notes in NTABLE (the output buffer).
4) Compare each entry in the input buffer (KTABLE) to each entry in the output buffer (NTABLE). If they are the same, set the trigger bit of the NTABLE entry and eliminate (zero) the entry from KTABLE. If all available outputs are used, or if all keys down find a home go to 1.
5) Place the remaining input buffer entries in output buffer locations which do not currently correspond to a down key (those in which D6 is cleared). When all input data has been placed or all channels available have been used go to 1.

There are a number of subtle implications here and unfortunately not enough space to cover them all.

A couple of really important ones are that if we think of "new" notes as ones corresponding to keys that were just pressed, this method tries to place those new notes in output channels which at some point in the past were already producing those notes.

This prevents a string of identical eighth notes (for example) from being assigned to different outputs each time they're used. Notes, once assigned, tend to stay assigned regardless of other keyboard activity - they don't move around in a totally unpredictable fashion as with some analog multi-note keyboards.

It also means that once the number of output channels available is "used up" by down keys that need to be placed, all other keys that are down are simply ignored (this is exactly what you want).

One important aspect of the above is that the program must "know" how many output channels are available to it, otherwise there is the possibility that notes may be assigned to non-existant channels (ones that have no corresponding hardware, not too bad in itself) and further (the really bad part) future activations of the note will be assigned again to these non-existant outputs - producing "dead" synthesizer keys that seem not to be doing anything.

Memory location $00EA contains the number of synthesizer channels available, more on this shortly.

THE PROGRAM

Shown on the next page is a disassembled listing of POLY 1.0.

Because, again, of space limitation we cannot re-print a fully documented version of POLY 1.0. It is supplied with the assembly and using manuals for the 8781 QuASH.

POLY 1.0 is also available in 8700 compatible cassette-tape form for $4.00.

LOADING AND INITIALIZING POLY 1.0

If you have a cassette interface on your 8700 and the POLY 1.0 tape, loading is simply a matter of connecting your tape recorder to the cassette input connectors on the 8700 and loading the tape using the following entry sequence:

0-0-0-0-0-0-F-F-0-0-1-1-TAPE

If you don't have the CS-87 option, you must enter the code manually from the 8700 keyboard.

The cassette version of this program loads all of page zero of memory (its total requirement) and in the process initializes a couple of things that you will need to care for manually if the cassette is not available. When entering manually, be sure to set the number of outputs to correspond to the number you have available. For example, assuming that you have a system with a single QuASH, the number of channels available should be set to 4 using the following computer keyboard sequence:

RESET-0-0-E-A-DISP
0-4-ENTER

The tape version initializes the number of outputs at the most likely number of 4. If you want to use less channels (because of lack of modules, say) or have a system with more, do it as was shown above.

When entering the program manually, make sure the decimal mode flag in the status register is cleared by using this sequence:

RESET-0-0-F-F-DISP
0-0-ENTER

This is automatically taken care of when the tape version is loaded.

USING POLY 1.0

With everything connected, loaded and initialized, we're ready to begin making music. Go to the beginning of the program and begin running it.

RESET-0-0-0-6-RUN

If everything is working properly, we will see the 8700 displays counting quickly, incrementing by one for each scan of the keyboard. All of the QuASH outputs should be at a very low output voltage (the program initializes them as zero) and the trigger flags for each channel should be cleared.

As we press synthesizer keys, QuASH channels should "come alive" and produce control voltages corresponding to the keys that POLY 1.0 has assigned to them. The trigger flags should be set if the key corresponding to the channel is currently down and clear when the key is released.

TWO MORE FEATURES OF POLY 1.0

While POLY is running, touching any of the keys from 0-3 on the 8700 keyboard (the first row of keys) causes the system to clear all QuASH channels to zero and wait for new data to be assigned. You'll figure out what this is good for as you become familiar with the system.

Maybe more importantly, touching any of the keys 4-7 (the second row on the 8700) provides a tuning function and causes all QuASH channels to produce the same note with the trigger flags set, allowing all oscillators to be set to the same pitch. The note produced corresponds to the 2nd C on a standard configuration 3 octave keyboard. THE CHANNELS MUST BE CLEARED AFTER TUNING by touching the first row of 8700 keys.

THE SYNTHESIZER

There are an almost unlimited number of ways to use the multiple control voltage produced by the QuASH and POLY 1.0. You may want to use multiple VCO's mixed into a single voicing circuit, (See figure 9), or what amounts to a complete synthesizer for each control channel or anything in between, (See figure 10).

A word of advice: in your beginning stages of learning to use this system, you should try to stick to configuration in which all of the channels are producing the same "type" of sound - as close to identical as possible. As your skills progress and you develope a feel for how POLY 1.0 is going to massage data you can work up to using some output channels to set VCO pitches while

POLY 1.0

By John S. Simonton, Jr.


```
06-   A9 00      LDA   #$00
08-   A2 18      LDX   #$18
0A-   95 CF      STA   $CF,X
0C-   CA         DEX
0D-   D0 FB      BNE   $080A
0F-   A2 08      LDX   #$08
11-   B5 D7      LDA   $D7,X
13-   18         CLC
14-   75 DF      ADC   $DF,X
16-   8D 00 09   STA   $0900
19-   9D F7 09   STA   $09F7,X
1C-   A0 04      LDY   #$04
1E-   88         DEY
1F-   D0 FD      BNE   $081E
21-   CA         DEX
22-   D0 ED      BNE   $0811
24-   A2 08      LDX   #$08
26-   A9 00      LDA   #$00
28-   95 CF      STA   $CF,X
2A-   CA         DEX
2B-   D0 FB      BNE   $0828
2D-   A2 08      LDX   #$08
2F-   2C 10 08   BIT   $0810
32-   30 FB      BMI   $082F
34-   2C 10 08   BIT   $0810
37-   30 0F      BMI   $0848
39-   50 F9      BVC   $0834
3B-   AD 10 08   LDA   $0810
3E-   95 CF      STA   $CF,X
40-   CD 10 08   CMP   $0810
43-   F0 FB      BEQ   $0840
45-   CA         DEX
46-   D0 EC      BNE   $0834
48-   E6 E8      INC   $E8
4A-   A5 E8      LDA   $E8
4C-   8D 20 08   STA   $0820
4F-   EA         NOP
50-   EA         NOP
51-   EA         NOP
52-   A5 EA      LDA   $EA
54-   85 EB      STA   $EB
56-   A2 08      LDX   #$08
58-   A9 BF      LDA   #$BF
5A-   35 D7      AND   $D7,X
5C-   95 D7      STA   $D7,X
5E-   CA         DEX
5F-   D0 F7      BNE   $0858
61-   A9 09      LDA   #$09
63-   85 E9      STA   $E9
65-   C6 E9      DEC   $E9
67-   F0 23      BEQ   $088C
69-   A6 E9      LDX   $E9
6B-   B4 CF      LDY   $CF,X
6D-   F0 1D      BEQ   $088C
6F-   A2 09      LDX   #$09
71-   CA         DEX
72-   F0 F1      BEQ   $0865
74-   98         TYA
75-   55 D7      EOR   $D7,X
77-   0A         ASL
78-   0A         ASL
79-   D0 F6      BNE   $0871
7B-   98         TYA
7C-   15 D7      ORA   $D7,X
7E-   95 D7      STA   $D7,X
80-   C6 EB      DEC   $EB
82-   F0 31      BEQ   $08B5
84-   A6 E9      LDX   $E9
86-   A9 00      LDA   #$00
88-   95 CF      STA   $CF,X
8A-   F0 D9      BEQ   $0865
8C-   A9 00      LDA   #$00
8E-   A2 09      LDX   #$09
90-   CA         DEX
91-   F0 22      BEQ   $08B5
93-   B4 CF      LDY   $CF,X
95-   F0 F9      BEQ   $0890
97-   95 CF      STA   $CF,X
99-   A2 09      LDX   #$09
9B-   CA         DEX
9C-   F0 17      BEQ   $08B5
9E-   A9 40      LDA   #$40
A0-   35 D7      AND   $D7,X
A2-   D0 F7      BNE   $089B
A4-   A9 80      LDA   #$80
A6-   35 D7      AND   $D7,X
A8-   95 D7      STA   $D7,X
AA-   98         TYA
AB-   15 D7      ORA   $D7,X
AD-   95 D7      STA   $D7,X
AF-   C6 EB      DEC   $EB
B1-   F0 02      BEQ   $08B5
B3-   D0 D7      BNE   $088C
B5-   20 00 FF   JSR   $FF00
B8-   C9 04      CMP   #$04
BA-   B0 03      BCS   $08BF
BC-   4C 06 00   JMP   $0006
BF-   C9 08      CMP   #$08
C1-   B0 05      BCS   $08C8
C3-   A9 2E      LDA   #$2E
C5-   4C 08 00   JMP   $0008
C8-   4C 0F 00   JMP   $000F
```

others control filter parameters (just an example - the number of possible combinations is extraordinarily large).

POLY 2.0 is under developement and features the use of some QuASH channels as software controlled envelope generators, reducing the need for lots of these hardware modules.

POLY 3.0 provides for computer storage of sequences of chordes or notes.

ONLY POLY 1.0 IS AVAILABLE NOW. The others are still a couple of months away. I mention them only because I want to make sure that we all understand that the nature of this new musical tool is a function of the program that is running and not so much of the hardware that it uses.

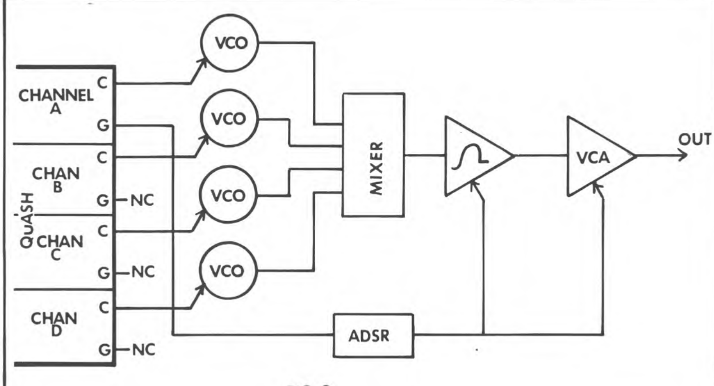


FIG. 9

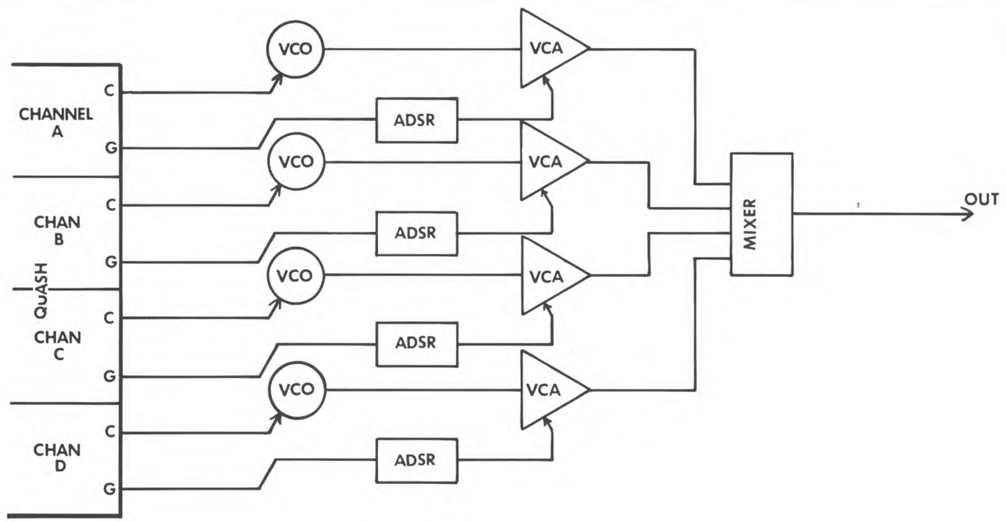


FIG. 10

FOR THE EXPERIMENTER

The following are available from PAIA for those of you who are experimenting with building this system from scratch:

POLY 1.0 CASSETTE tape and documented listing. $5.00

HARDWARE KIT
DB-25 connectors and 2 single ended DIP headers.
Satin black finished
Computer mounting bezel.
plus hardware 24.95 ppd.

WE'LL BE THERE

SECOND WEST COAST COMPUTER FAIRE

March 3 - 4 - 5, 1978

See us at Booths 410/412

It happens again this year but in a different location. The Second West Coast Computer Faire will be held March 3rd thru 5th at the San Jose Convention Center.

If you've hesitated about attending because your interest is primarily music, don't let the Computer Faire title keep you away. Last year there were half a dozen music exhibits and this year promises to bring more.

We'll have our latest product developments on display and be there to answer questions and personally demonstrate the new equipment.

HOPE WE'LL SEE YOU THERE!

- Linda Kay -

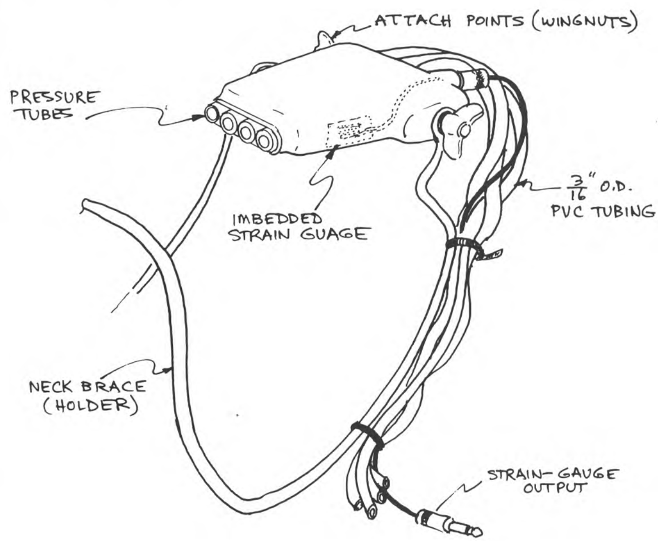
from the food for thought dept.

IS THIS THE ULTIMATE INTERFACE

by Jeffrey Noble

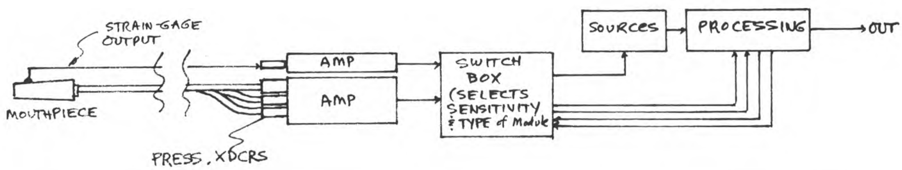
Background: A keyboard musician is constantly struggling to fill the vacuum left by the lead and rhythm instruments. In the studio he can spend more and more time to plug fewer holes and fewer gaps by adding tracks and patches (using the same equipment,) but time is something the performing musician does not have! He is constantly working to "fill-in", punching buttons and presets and changing patches and moving from keyboard to keyboard several times during the same song. Nightmares abound in which he has to grow 3 or 4 new hands and feet just to keep up. An alternate method of voltage control (in addition to his hands and feet which are already fully occupied) would, in effect, be that 3rd or 4th hand. At the risk of sounding a little perveted, I call this alternate device my ORAL JOYSTICK. It consists of a tube or tubes encased in a semi-rigid plastic mouthpiece. Breath pressure through the tubes puts out a proportional voltage, as does biting down on the mouthpiece. Again, this doesn't particularly help the studio musician. The performing musician, however, will immediately grasp the potential—not only for greater control—but for adding a dimension to a solo line that has been almost completely lacking in all other types of electronic keyboards, be they organs, synthesizers, or whatever. That dimension is FEELING. Tell an organ player to put expression into his work and he pushes down the only control he's got: the "expression" (volume) pedal. Someone blowing winds or brass or strings, on the other hand, has an almost limitless range of expression (the ultimate in expression, of course, is the human voice). By varying the tightness and position of his facial muscles, tongue, nasal passages, breath control, etc., a sax man can, for instance, start a solo with a "harsh, angry" attack, stretch it to a "wail", soften down to a "warm, sensuous" bridge, and take it out with a "half-bitter, half-sorrowful" lament. His fingers play the notes, sure 'nuff, but the subtle changes in pitch, harmonic content, attack, etc. are what makes the music come alive.

A mouthpiece "joystick" can do the same for the synthesist. One (or more) mouthpieces attached to a neck brace (just like Dylan uses for harmonicas) can vary voltage (and hence, tone envelope, pitch, and other designated parameters) to the ADSR, VCO, VCF, & other processors and sources. Better yet is to go through the computer which could be programmed to change several parameters simultaneously. I think you can see the possibilities.



ORAL JOYSTICK or MOUTHPIECE VOLTAGE CONTROL (MVC)

Example: When the musician "BITES DOWN", the strain gage transmits the voltage proportional to the deformation of the mouthpiece. This voltage is amplified and changes the processing. A heavy "bite" could be set up to increase attack amplitude, increase square wave component, decrease sawtooth component, decrease LFO vibrato modulation, etc. At the same time, expelling forcefully (couldn't bring myself to say blow hard) through the tubes could raise the pitch (typical adjustment would be for ± 1/2 step). Commercially available pressure xducers and strain gauges are too costly (they have to be accurate and repeatable within a very narrow range). Home built will suffice here since we already have inserted into the works a self-correcting device (wild-hippy-musician).



Equipment Exchange

Equipment Exchange allows a place for our subcribers to offer for sale or trade equipment related to music and electronics. If there is equipment you are looking for, we will list it, also. Please keep the listings as brief as possible. Persons responding to ads should write directly to the other party. DO NOT write to PAIA. PAIA is not responsible for any claims made in the ads or results of any transactions. PAIA has the right to edit or refuse any ads submitted.

For Sale: Paia Gnome, needs a little work. $25 or reasonable offer. George Smyth, Rt. 11 Box 424, Morgantown, WV 26505, Phone (304) 599-2472.

For Sale: 2720A, works perfectly and to specs. $235. Don Daugherty, 7313 Inzer St., Springfield, VA 22151, (703) 256-6148 evenings.

For Sale: 4782 Road Keyboard, two 4761 wings with 2720-1, -4, -3B, -5, -7, 4711, 4740, two 4712, 4720, and all patch cords. $400 new, asking $250 or best offer. D. Norris, Box 2672, USAF Academy, CO 80841

GRAND OPENING SPECIAL: Personal Vibracon Globes - discounted to 800 drimes. Limit one to a customer. Remember - these are pure globe. Hyperspace it out here and see our other techno-tainment items. SLASCN VIBRA DEVICES, orbital shopping post 97, sector 3- cove 5. OPEN during bioclock periods 4, 7, 9, and 0.

For Sale: Paia 4782 - $120, 4780 - $55. Both completely assembled and fully functional. David Kaleita, 2300 Poland, Hamtrack, Mich. 48212 Phone: (313) 875-7885.

FOR SALE: Assembled PAIA modules. Digital Keyboard, 8780 D/A, 3-4720 VCOs, 4740 ADSR, 4720 RING MOD, 3-4700s, 2-4761s, 4780 sequencer, 4730 filter, 1500A Phlanger. $531.00 value, will sell for $500. All modules work to specs. Jeff Markham, P O Box 408, Lynchburg, VA 24505.

For Sale: Gnome, assembled, working, including all literature. $40. Phil Mandel, PO Box 263, Burlington, MA 01802.

For Sale: Paia keyboard in road case. Fully assembled and tuned. Includes custom installed Joystick and control panel with glide and footswitch provision. $150. Excellent condition. Eric Hanson, 213 Cedar Lane, Seabrook, TX 77586, (713) 334-5920.

For Sale: EML Poly-Box, never used. $375 or offer. Paia 8782 Digital Keyboard, new, $90. Prices include postage. Everett Rantanen, 312 E. Schiller St., Milwaukee, Wisc. 53207.

For Sale: Phlanger 1500A, $45. OZ, $80. Gnome, $40. Heathkit TA16 Amp, $100. Prices exclusive of shipping. All units used very little, and like new. Robert Eggli, 163 West 935 So., Garland, Utah 84312.

MUST SELL: Arp Odyssey in perfect condition. With case, pedals, and manuals (owners and service). Will sell for best reasonable offer. Terry Ward, Rt. 2, Box 579, Durham, NC 27705. Ph. (919) 383-6140.

For Sale: PAIA 2720-2A VCO, A-1 condition $20.00; 2720-3B Band Pass Filter $10.00; 2720-3L Low Pass Filter $8.00. Tom Henry, 2306 M Ginger Lane, Charlotte, NC 28213.

Local Happenings

..... continued from page 27

From north of the border, we have news of a new Canadian electronic music club, the Ontario Polyphonic Society. Those of you interested in membership or more information should contact John A. Orpin, 6 Conrad Avenue, Toronto, M6G 3G5 Canada. Sounds interesting! Be sure to keep us informed of your club activities so we can share them!

If you live near any of these people, contact them. They are anxious to talk with other synthesists, organize ensembles and exchange information.

David Kaleita
2300 Poland
Hamtrack, Mich. 48212
Ph: (313) 875-7885

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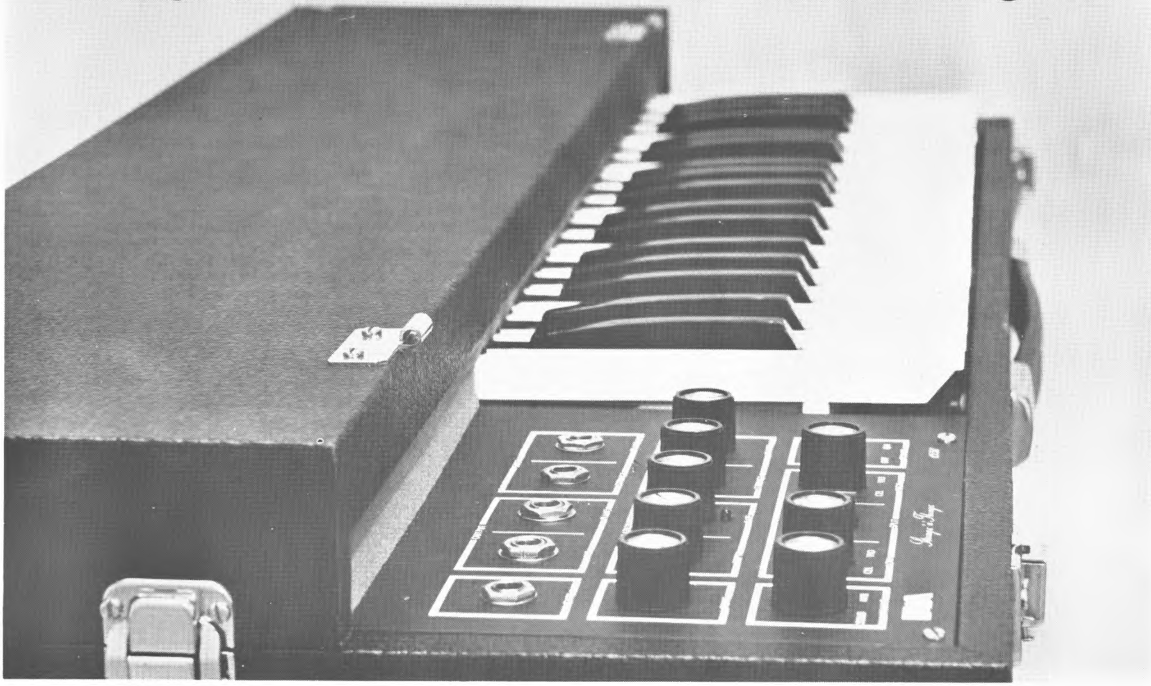

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