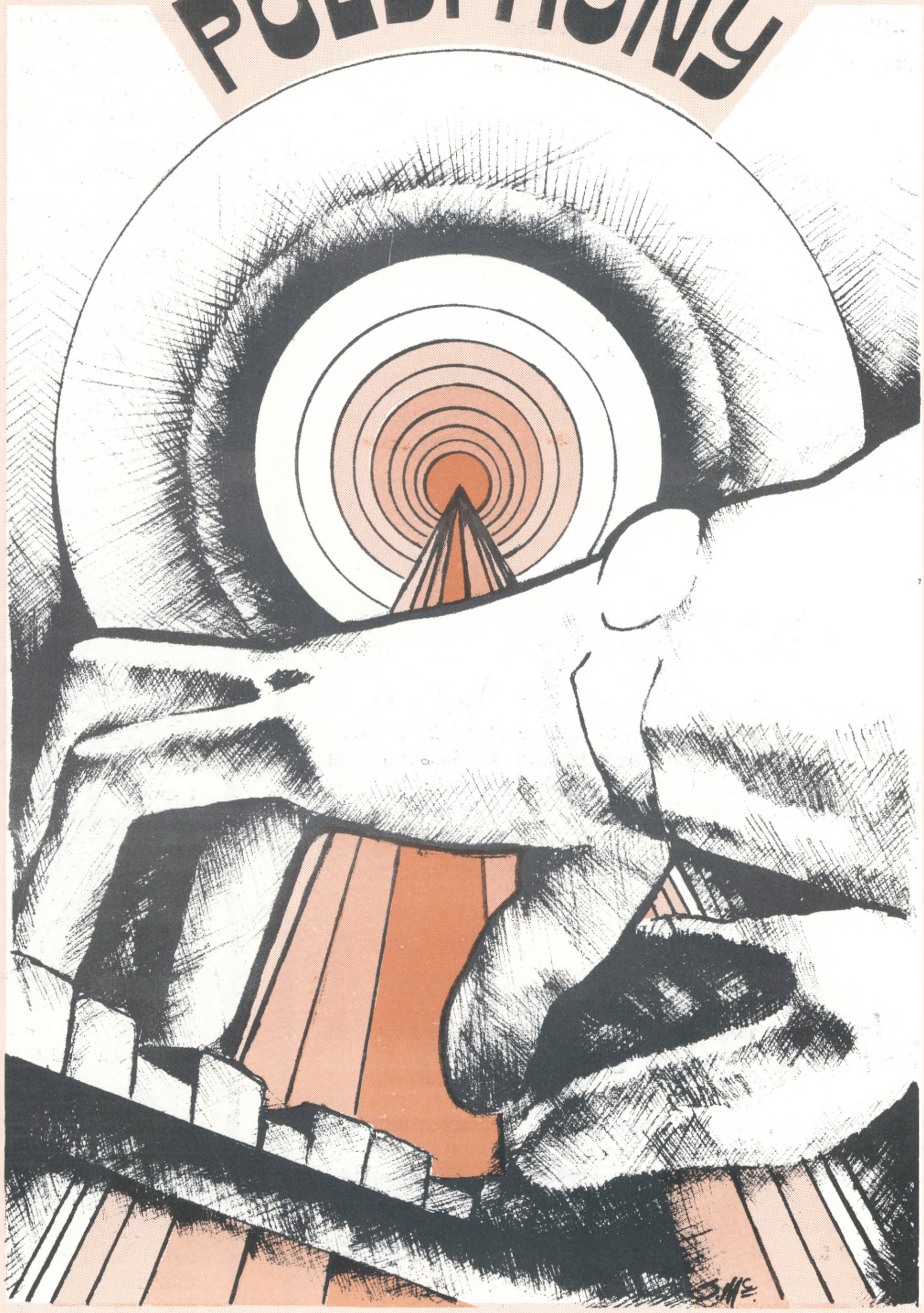
JULY 1977

POLYPHONY



Computer/Synthesizer product summary
A Quad Addressable Sample & Hold
Construction Projects * Patches
Eliminating Patch Cords

STAFF

EDITOR:	Marvin Jones
ASSISTANT EDITOR:	Linda Kay Brumfield
CONTRIBUTING EDITOR:	John S. Simonton, Jr.
PRODUCTION:	Ramona French Shirley McConnell
PRINTING:	Paul Scarborough

Published Quarterly by: PAIA Electronics, Inc.
1020 W. Wilshire Blvd.
Oklahoma City, OK 73116

Subscriptions

Subscription rates are $4.00 per year mailed in the USA; $5.00 per year mailed to Canada and Mexico; International subscriptions are $6.00 per year. Address Corrections requested. Subscriptions accepted on a one-year basis only. For change of address: be sure to send in your former address with zip code along with your current address.

Write for Polyphony

..... It's not something that you'll get rich at, but, nevertheless, we are paying at the rate of $25.00 per printed page for feature articles including such things as circuit modifications, new circuits, narratives, how to articles, etc. All material submitted must be original and never before published. We are also paying $5.00 for each patch accepted for publication. Payment is made upon publication. We cannot assume the responsibility for the return of manuscripts unless specifically requested and supplied with a stamped self-addressed envelope for its return. All accepted materials and associated copyrights become the sole property of Polyphony and may not be reproduced without written permission from the publisher.

LETTERS:

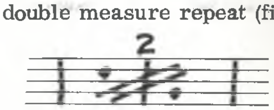
Comments & Corrections

Dear PAIA POLYPHONY PEOPLE;

As a music teacher I am impressed with Marvin Jones' concise pedagogy in the "Fundamental Music Notation" series, but have a few additions and corrections to suggest. First, in mentioning relative keys, you omitted telling how to find out which minor is relative to which major (the minor is a third lower). Perhaps people could figure it out from the two examples given, but it would have been nice if you could have told them. It would also have been helpful if you had told how to figure out key signatures simply (the next to the last flat is the name of the key, or the next note up from the last sharp). Also helpful to know is the order of flats BEADGCF and sharps (reverse the order of flats). In the section on TIME SIGNATURES you implied that C stands for "Common Time". This is a popular misconception. In truth it stems from the earliest days of notation when 3/4 was considered "perfect" time, (related symbolically to the trinity - remember the first guys to write music down were monks) and represented by the circle, the "perfect" geometric. Thus 4/4, or "imperfect" time was represented by a broken circle. I know that all this seems very academic, and is probably of interest only to music historians, but you don't want your magazine publishing falsehoods, now, do you?

You also implied that the TIE was used only between two or more measures. It is also used to represent time values within measures that can't be shown otherwise (i. e. two and a half beats = 𝅗𝅥‿♩). In addition to the single measure repeat there is a double measure repeat (figure

fig. N.



n) which is useful in notating longer ostinato passages. Finally, in regards to TEMPO markings, I prefer to use both a metronome marking and a verbal marking (not necessarily in Italian if your intentions can be expressed more explicitly in English) to avoid confusion. Metronome markings, whenever used, should be regarded ONLY as approximations, to be varied according to individual taste. I look forward to seeing the rest of the series - especially the final segment on electronic music notation where there is so much diversity.

I would also like to say a few words about a concept that hasn't been discussed in Polyphony: System design. While a

continued on page 11 ....

Editorial

Over the past year, there has been a flood of new keyboard instruments appearing on the market which have been called polyphonic synthesizers.

"Polyphonic" has become a very fashionable word in the keyboard industry, and many companies think that if an instrument can play more than one note at a time they have it made. I feel it takes a lot more than that to make a polyphonic instrument which represents any amount of musical or technical advancement.

Let's take a minute to look at the derivation of the words polyphonic and polytonic. The prefix "poly-" is generally taken to mean many or multiple. The Latin word "tonus" provides the basis for the word tone, or for our purposes "-tonic". Tonus means stretching or tension. Taken in musical context, this translates directly to pitch. Consider: stretching a guitar string increases the pitch; stretching a drum head increases its pitch. Our remaining section, "-phonic", seems to come from the Greek word for sound. Granted, sound is a word which covers a broad area and I don't want to categorize it too much, but, I think most of you would agree that the "sound" of an instrument has a lot more to it than just pitch -- it must also consider harmonic content, amplitude envelope, phase relationships of the harmonics, and on and on.

Putting our pieces back together again, we come up with polytonic meaning "many notes" and polyphonic meaning "many sounds" or, in musical context, "many voiced". If so desired, either one of these terms could be stretched and distorted to represent even the most simple electronic instruments. As an example, let's consider the PAIA Gnome Micro-Synthesizer. By varying the VCO range and using the controller strip, a virtually infinite variety of different notes can be played. True, you can't play several notes <u>at once</u>, but you CAN play several notes. So, why don't we start calling the Gnome a polytonic synthesizer? Likewise, the multiple VCO waveform outputs, noise source, VCF and VCA of the Gnome can all be combined in so many different settings that the Gnome will produce an infinite number of sounds or permutations. So, since we can get <u>many sounds</u> out of the Gnome, let's start calling it a polyphonic synthesizer! I can see you all rolling in the aisles with laughter. Yes, this is obviously carrying the words to the extremes of their meanings, but I think you can see that we could have used these advertising tactics if we wanted to. Similarly, a Mini-Moog or Arp Odyssey could have been promoted with these terms. Several

continued on page 4 ....

CONTENTS

Volume 3, No. 1 1977

ON THE COVER:

A pen and ink drawing by Shirley McConnell who is new to our production staff with this issue. We'll be seeing more of her work in the future.

Editorial

.....continued from page 3

years ago nobody would have noticed, but now we would all laugh. So what is it that has changed on the new synthesizers (the Polymoog, Arp Omni, and many string synthesizers) that impels the manufacturers to squeeze this ubiquitous term - polyphonic - into their advertising and promotion. The only thing I can see that has changed much is the ability to play more than one note at a time. You still have only one set of master controls to determine the voicing of the instrument, and whatever voicing you select is repeated for each note all the way across the keyboard. Does this description sound like I'm talking about an organ? Well that's all these units are -- super organs. The only thing that pushes them into the synthesizer category is the fact that they are using circuitry which proliferated during synthesizer development; active filtering instead of the older passive networks, VCA envelope generation instead of diode keying, and control voltage concepts to change many circuit parameters with a single front panel master control. If your friend showed you a Farfisa combo organ or a Hammond B3 and told you it was a polyphonic synthesizer, you may start to phone the white coat brigade. But, he wouldn't be that far wrong -- according to the definitions of the major manufacturers. The only difference would be that his wouldn't be quite as versatile and wouldn't have quite as many variables to play with. In my opinion, these types of instruments would more appropriately be called polytonic.

OK, so what is a polyphonic synthesizer? Define it. Where do we draw the line? Well, I'm not arguing the point that these machines could be called polyphonic. But, phrases like "...the first and only truly polyphonic synthesizer", and "completely polyphonic"????! Sorry, I won't go with that. Who's to say what a truly polyphonic unit is? I know we have the technology to do a LOT more than make super organs. I've seen it.

My concept of a true or complete polyphonic system stems from my vision of what synthesizers will be used for in the future. A composer should be able to sit down at a keyboard and realize real time performances of pieces originally written for several instruments. That implies playing up to ten keys and not only getting ten notes, but ten individual voices -- each completely different if you wish. As far as I know, the only commercial unit to do this is the Eu Systems keyboard. The Oberheim polyphonic system uses the same scheme (in fact, it is manufactured under a license from Eu). Also, there are other companies (ahem, I think you know who I'm talking about) who are releasing similar systems -- hopefully with even more versatility and power due to the use of microprocessor control.

In the end, you each have to decide for yourselves what degree of sophistication YOUR polyphonic system will need. Perhaps the polytonic super organs serve your needs, perhaps they are even more than you were expecting. But, somehow this all rubs me the wrong way.

Thanks for listening.

- Marvin Jones -

UPDATE GNOME/GUITAR INTERFACE

As often happens with an electronic device, once it gets out into the hands of creative people more uses are discovered. For example, Jim Riter (an electronically-oriented keyboard player) wrote me to say that the Gnome/Guitar Interface circuit, published in the 4/76 issue of Polyphony, will also allow a guitar to trigger ARP envelope generators directly. It can also trigger minimoog envelope generators via a reed relay; in this application, you will need a saturated transistor driver for the relay since the output of the Interface does not deliver enough current to drive this type of load directly. Jim also suggests possibly using a 4016 CMOS switch as an alternative to the relay, since it would not require a driver transistor.

Incidentally, this circuit also gives a reasonably good envelope follower output, which you can tap off through a buffer at the junction of the 47K resistor and .22 mfd. capacitor, at the output of the second stage of the Interface. Happy Experimenting!

- Craig Anderton -

WHAT THE COMPUTER DOES... AN INTRODUCTION

By John S. Simonton, Jr.

The computer in our system does not itself generate any sound. It is simply acting as a performer/composer assisting control system for a more or less normal synthesizer. Providing what amounts to an extra set (or several sets) of hands.

From a system standpoint, it fits between the keyboard and synthesizer like this:

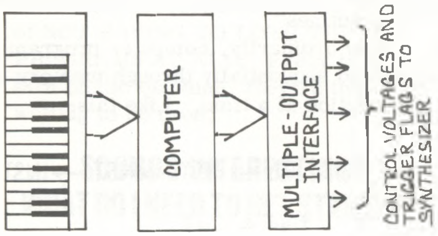


figure (a)

We said above, "more or less normal" synthesizer because there are three special elements involved in the synthesizer/computer interface:

1) a digitally encoded AGO keyboard
2) a Digital to Analog Converter
3) a multiple S/H circuit to allow several simultaneous outputs from the Digital to Analog converter.

The computer runs programs (either supplied by PAIA or user written) that receive data from the synthesizer keyboard and issue instructions to the D/A and multiple S/H which in turn control the synthesizer.

PROGRAMMING OVERVIEW

Just saying that the computer controls the synthesizer is hardly a satisfactory explanation of the system. Hardly satisfactory because it leaves out a

VERY IMPORTANT CONCEPT

which is that it is not really the computer that is controlling the synthesizer, it's the programs. In a very real sense, the computer is there only because it's a way to run the programs.

One of the programs (for example) "reads" the synthesizer keyboard and builds a table of what it finds there.

If the phrase "builds a table" is unfamiliar to you, it simply means that when the program finds that a given key is down on the keyboard it records in a special place (location or address) in memory which key it is. The next key that it finds down, it records in the next memory location; and so on. When the program has finished looking at the entire keyboard the result is a list or "table" of the keys that were down during that scan. If you were holding down a C chord for example, the table might look like this:

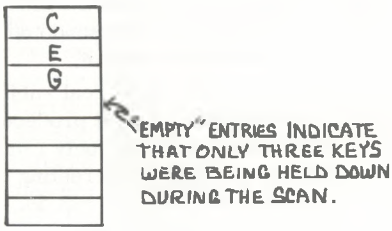


figure (b)

That's not really all there is to this program - there are some subtleties that would probably be confusing at this point. We'll get to them later. For right now, we'll just think of this program as a list-builder.

Also, so that I won't have to keep typing "the program that builds the list of keys that are down on the keyboard", we'll agree among ourselves that we'll call this program by the name "LOOK". From now on, when I say something like "we LOOK at the keyboard" you'll know that I mean we "execute" (run) this program.

And, while we're hanging labels on things, we may also just as well name the list that LOOK generates "key-table", or, since I'm a lazy typist, just KTABLE.

Got that? LOOK builds KTABLE. OK, next.

There is another program that we'll call NOTEOUT, because it takes care of outputting the notes.

Like LOOK, this one can be stated in simple terms: it reads the first entry from a table and causes the D/A to convert that key data to a control voltage which it then strobes into the first S/H. It then gets the second entry from the table, converts it to a control voltage and assigns it to the second S/H. Gets the third entry, etc.

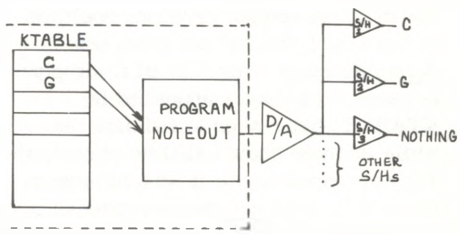
Also, like LOOK, there are subtleties that we'll look at later but the important point is that this routine works quickly. A block of 32 Sample and Holds can easily be refreshed and up-dated in about 16 ms. - more than fast enough.

The table that is read by NOTEOUT we will call the "note-table" or, simply NTABLE.

LOOK builds KTABLE and NOTEOUT reads NTABLE. Maybe you're wondering why two tables - why not just one.

Well, we could do it that way - if we did, a simplified diagram of the system should look like figure c.

You will recognize that we're still holding down that C chord. Now suppose we let the E go. On the next scan of the keyboard, LOOK up-dates KTABLE to reflect the fact that the E is no longer held down. KTABLE now looks like this:



THIS MIGHT NOT BE TOO BAD — MANY ORGANS DO NO MORE.

figure (d)

And when NOTEOUT reads this table and up-dates the S/H circuits, guess what? The G has moved to the loca-

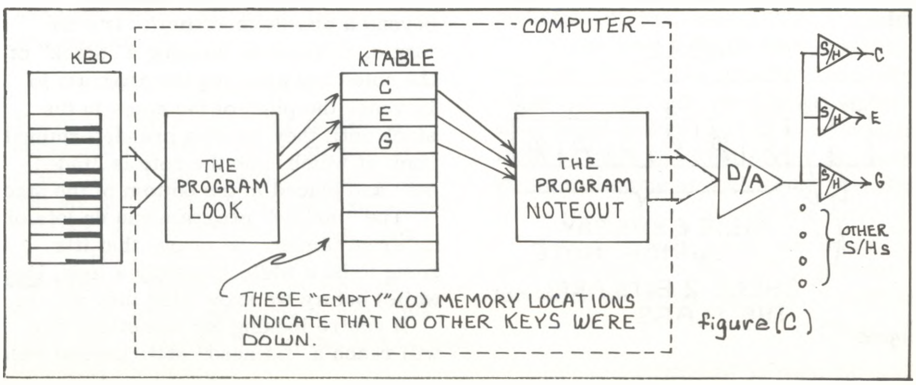


figure (C)

tion previously occupied by the E and from the S/H that previously was producing the control voltage for the G we now have .... nothing.

As if it weren't bad enough that the VCO which was previously producing an E is now playing a G (and we can hear when it makes this change), we can't do any decay processing on the E - the way a natural instrument would - because it's not there anymore.

Maybe this isn't too bad. A lot of organs produce results very similar to this - and all multiple output analog keyboards do this exact same "guess where the note's going to come out" trick. Still, it seems that there would be a more pleasing way to do it.

There is.

Because we're using two tables, we can generate a large (very large) family of programs that make decisions on how to transfer the information from KTABLE to NTABLE. This produces a machine which diagramaticly might look like this:

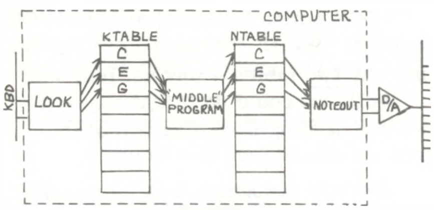


figure (e)

> How this new middle program makes transfers from KTABLE to NTABLE determines completely the "personality" of the instrument.

For instance, a better way to handle the multiple -output problem would be to have the "middle" program not delete an entry from NTABLE simply because it no longer appeared in KTABLE, but rather to indicate that while the note should still be played, the key corresponding to it was no longer being held down and decay processing should begin. This is where the concept of "flags" associated with each note comes in and while it is slightly out of sequence, we should examine this important feature now.

The data that goes out to the synthesizer interface is a collection of 8 binary digits (bits - "1" or "0"). Like this:

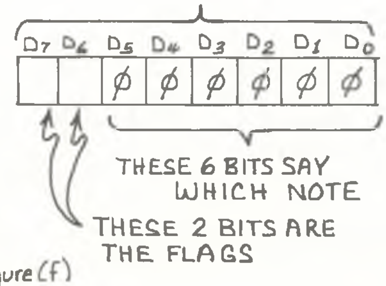


figure (f)

If we want to indicate to the synthesizer that the note that the data represents is one which currently corresponds to a key that is being held down on the keyboard, then we set bit #7 (D_6) to a "1". If the data does not correspond to a key that is currently down then this bit is a zero. As you can see, if you're already familiar with synthesizers, this flag bit corresponds to the "gate" signal that you get out of most synthesizer keyboards.

This leaves us with a "left-over" flag that can be used in a variety of ways. It can, for instance, be used simply as an independent gate signal allowing the processor to select between one of two patching arrangements that we've set up. Or, and I believe that this is the preferable use, it can be used as a GLIDE SELECT bit that turns glissando on and off - under computer control.

But, to get back to the real subject at hand, the polyphonic output procedure described above is not the only (or, in my opinion, the most) interesting thing that the "middle" program can do.

It can examine the entries in KTABLE and if they are lower than a given note on the keyboard assign them to one group of outputs and if they are higher assign them to a second group of outputs, Which has the effect of "splitting" the keyboard into two different voices - one for low keys and a second for high keys.

The "middle" program can take notes from the keyboard and not only play them immediately, but also store them in another permanent table in the machine's memory for playback again later.

The "middle" program can take notes from the permanent table mentioned above, assign them to outputs and simultaneously assign current keyboard activity to other outputs - so that you can play along with something that was previously "recorded".

These same programs can allow independent recording and simultaneous playback of multiple "tracks". Like a multi-track recording studio only without the hassle of tape splicing, editing and (worst of all) over-dubbing noise.

The "middle" program can do tricks like making a chord played on the keyboard seem to be rising in pitch, constantly, without ever actually going beyond a pre-defined limit. It's not magic, it involves forming a "stack" of the notes and allowing the program to increase the pitch of the notes in the stack until they reach a pre-determined limit at which time the note is "faded out" and placed in the bottom of the stack.

The "middle" program can do lots of different things. So many, that it's going to be a while (possibly a long, long while) before we know what they all are.

If you're looking for something that will reach a "finished" state beyond which there is nothing further to do, this isn't the product for you.

SO MANY "DIFFERENT" PROGRAMS

One thing that you may notice in the discussion above is that all of these very different "resource allocation" schemes have in common the fact that they all use LOOK and NOTEOUT. We could make these two routines a part of each of the larger programs if we wished - there wouldn't be any problems with that - except that they are long-ish and would take a while to "load" into the machine's memory. Particularly if you're not using the computer's optional cassette interface. I think there's a much better way.

We can write the LOOK and NOTEOUT programs so that they're what's known as "subroutines".

Now ordinarily, computer programs proceed sequentially through memory an instruction at a time. Like this:

figure (g)

But a subroutine allows a block of programming to be stored out of sequence in the machine so that when you "call" or "jump to" a subroutine it's like this:

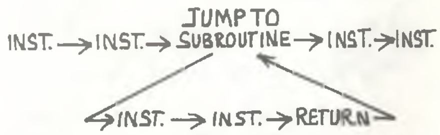


Figure (h)

The "return" causes the computer to go back to the place that it was before the subroutine was called and continue executing the main program.

Maybe the "subroutine" concept confuses you (though after such a terrific explanation it's hard to imagine how). If it does, here's another way that you can think of them:

SOFTWARE MODULES

You're certainly used to synthesizer "hardware" modules by now - all those little processing elements (VCO's, VCF's etc.) that we tie together with patch cords to produce different sounds or effects.

Here we have their equivalent in computer instructions - little modules of programming that are patched together (not with wire, of course, with more programming) which, depending on how they're tied together, produce different effects.

LOOK and NOTEOUT are not the only software modules that are useful, others include SAVE (the "recording" module, SREPRO (the "playback" module), DELAY (a time delay routine), POLY (a useful polytonic resource allocation algorithm), and others.

These various modules are available in a number of different forms. They're available just as program listings (which can be manually entered into the computer - very tedious but about as cheap as you can get) or they're also available on cassette tape that can be loaded into the computer using the optional cassette interface.

First choice for a place to save these universally useful programs, though, is Read Only Memory.

This is the most expensive alternative (ROMs have to go for about $20/each - one would be filled by the programs mentioned above) but it has the advantage of NOT HAVING TO LOAD THE PROGRAMS AT ALL. Every time you turn on the machine, they're there, waiting to be used.

SOUNDS INTERESTING WHAT DO I NEED TO GET STARTED?

If you already have some PAIA synthesis equipment, you're well on the way, but you need to convert to the new digital format. We've tried to make that as easy and inexpensive as possible by providing a retro-fit kit to digitally encode your present PAIA keyboard, the EK-3 Keyboard Encoder Kit mentioned in POLYPHONY "Lab Notes" 4/76.

This encoder is primarily designed to fit 4700 series keyboards, but will of course fit 2720 series equipment as well. It is one of our experimenter's kit series and does not include step-by-step instructions.

If you want to start over with a new keyboard, we have the 8782 Encoded Keyboard - one of our full kits with complete instructions.

If you already have an organ and would like to use that keyboard for either synthesizer or synthesizer/computer interface, we have the EK-4 Organ Keyboard Encoder which we will examine in next issue's "Lab Notes".

The advantage to this is that the keyboard already in the organ may be used for both synthesizer/computer and organ - all at the same time. Even if there are no "spare" contacts on the keyboard.

BUT I DON'T HAVE A SYNTHESIZER!

Looking back over the text to this point I notice an important point that has not been prominently mentioned. This system - because of the properties of the D/A - will work only with low-cost LINEAR synthesizer modules. Synthesis modules whose characteristics are exponential cannot be used (though it is an easy matter to substitute another D/A for ours).

It is difficult to tell someone what the configuration of their synthesizer should be. Particularly with modular equipment like our current line. The modules that make up the system are so much a function of the use to which the system is to be put.

Never the less, we have two systems configured as starting points. "Starting points" because it has been our experience that most people add and make changes to their system as time goes on. Customizing it to their application.

These two packages are the 4700/C (primarily a monotonic system) and the 4700/J (suitable for polyphonic work, limited multi-track recording, etc.). These are both systems that we originally put together to take to shows. Each for its intended purpose, they have proven to be reliable and versatile; each capable (by design) of turning someone from an "I don't like synthesizers" person into a "I never realized they could do that" person. Maximum usefulness and versatility within minimum "waste" capacity.

The 4700/C is a minimal, useable system. It has roughly the capabilities of the "mini" this and that that you see advertised. It's made for people who find synthesis interesting but aren't really sure that they're going to get into it in a big way. It is (briefly) an ideal place to start. And since all of our gear is modular and available separately, it is a system which will easily grow as your interest grows.

The 4700/J is by the standards of the industry a good-sized system. It's difficult to make comparisons, since some of the modules (particularly those that are the computer interface) aren't available from other manufacturers; but, if these modules were available and you purchased them assembled through the normal distribution chain the '/J would be on the order of $2, 500 to $3, 500 worth of equipment. And, again, it's not a dead-end system, but one that can grow.

One final comment in this section is in order, and it may seem strange for someone who is, after all, trying to sell you equipment:

DON'T OVER-BUY

There are two reasons for making a statement like this - both imminently practical; 1) our experience has been that you will probably like the equipment a lot and will be a customer for many years, but if you don't (and aren't) you don't have a bunch of money sunk in something you're not going to use. We won't have someone wandering around bad-mouthing the gear.

2) Without committing to anything in print, development goes on all the time - to the practical synthesist, the versatility of modular equipment makes it desirable to have some of it around (ask anyone seriously involved in electronic music synthesis). But, well, look at any issue of POLYPHONY - development goes on and you never can tell what's just around the corner.

WHICH COMPUTER?

This one is almost as bad as which synthesizer. For the same reasons - the decisions are very personal and user related. Also like the "which synthesizer?" though, we have suggestions.

Our first, and strongest, suggestion is our own 8700 Computer/Controller. High on the list of compelling reasons to select this machine should be the fact that it will have our fullest software support (all of the programs mentioned earlier are available now), it is physically designed to fit into a space that has been kept free in our 4700 and 8700 series keyboards and is a machine designed to the PAIA ideal of "maximum impact for minimum bucks".

The 8700 is based on a 6503 processor (a fully software compatible version of the increasingly popular 6502) and has features as described in the product summary. This processor was chosen over others which were - at the time that the decision was made - more popular for a variety of reasons, but by far the biggest was that it is an easy machine to use. Even if you're programming in machine language (and don't kid yourself, the day will probably come that you will want to do something completely different - something not available either from us or from the independent user's group program exchange - and the only way to do it will be to write the code yourself, it's easier than it looks).

But let's suppose that you already have a computer. If that computer happens to be something like a KIM-1, you're in great shape. We will shortly have a complete KIM-1 package showing how to interface and almost as complete a selection of programs as for our own machine (we like the KIM series stuff - and since it, too, uses a 6502......)

If you have a SWTP 6800 system, the 8780 and 8782 instructions already outline using one of their MP-L's for interfacing (sorry, no software support from us right now, but surely the user's group will come up with some - Southwest has a really nice, popular system).

Coincidentally, there are other machines that use the 6502 processor for which all of our software is written;

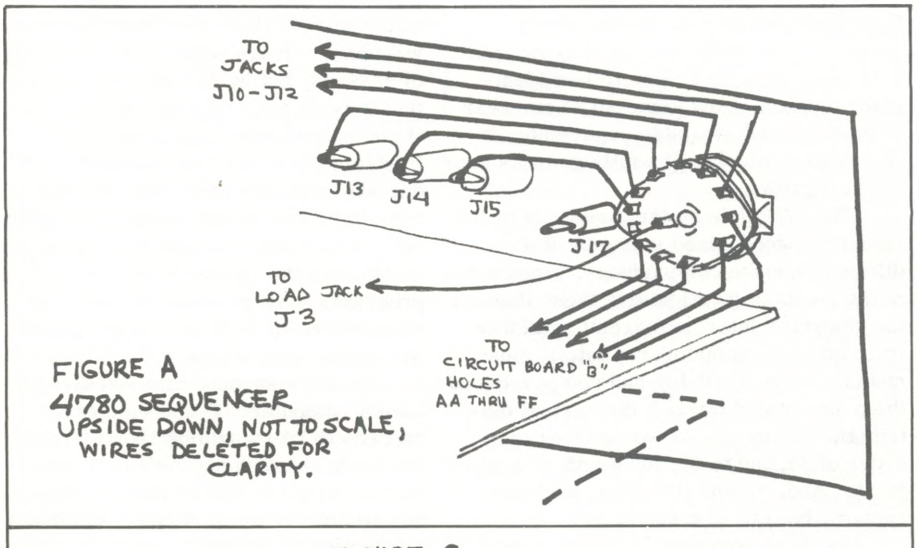
continued on page 28.....

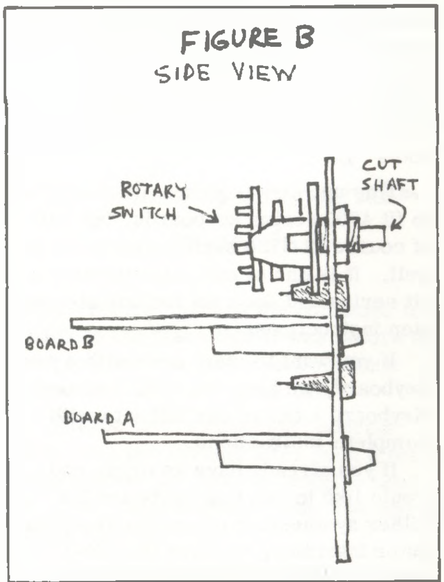
4780 SEQUENCER MODIFICATION

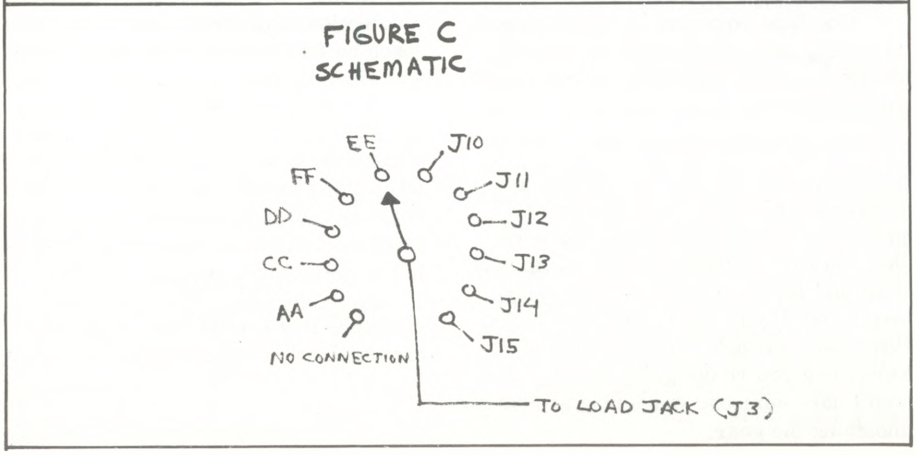
Allows you to set and rapidly change the sequence length without patch cords.

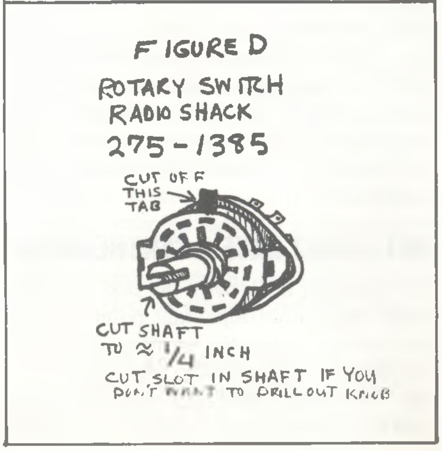
by Bob Yannes

I prefer to use as few patch cords as possible with my synthesizer and have therefore made extensive modifications to my 4700/S. One of the simpler and most useful modifications was the addition of a rotary switch to the 4780 sequencer which allows me to set and rapidly change the sequence length without the use of patchcords. Radio Shack sells a 12 - position single-pole rotary switch (#275-1385) which is ideal. There are no electrical modifications to the 4780 and only a few physical modifications. The switch fits in the lower right-hand corner of the sequencer (obliterating the word "out"). This hole can be drilled even after the module has been assembled. (my 4780 was already installed in the cabinet when I got this brainstorm), but you have to be careful, Drilling a small "pilot" hole is manditory. Figure F shows the approximate location of the hole. Figure D shows the rotary switch. I recommend that you trim off the small tab on the front and cut the shaft down to about 1/4 inch. If the knobs you're using don't have set screws (such as PAIA knobs), you might try sawing a small slot in the shaft and squeezing the shaft together, then jamming the knob on (you could epoxy it, but then you'll never get it off). Figures A & B show the mounting configuration. In figure A the wiring layout is also shown - NOTE: the wiring layout in figure A is just to give an idea of what it looks like, the actual wiring order is in figure C. Basically, the switch takes the place of a patch cord coming from each stage's output jack. You could wire all the switch wires directly to the jacks, but it's much easier to wire the lines that would go to Jacks J5 through J9, to the appropriate points on circuit board B (holes AA, CC, DD, FF, EE), besides, if your 4780 is already assembled, it will be impossible to get at jacks J4 thru J9. You may nave noticed that one position of the switch is not connected to anything (if you want to connect it, you'd connect it to hole BB on circuit board B). The reason for this is that BB is connected to jack J4, the first stage output jack. Now, if you want a one event sequence, that's up to you, but leaving the first position in the switch unconnected allows you to switch to this position and have the sequencer operate just as it did before you added the switch. In particular, with the RUN/STOP switch set to conditional run, the sequence will end and the clock will stop after the last stage. If you don't provide an unconnected switch position on the rotary switch, you'll never be able to get a non-repeating sequence. Although there is no interference between the rotary switch and the 4780; unfortunately, there is interference between the rotary switch and the metal bar on which the modules mount (in the wing cabinet). Be sure to leave enough room between the 4780 front panel and the front disc of the rotary switch, the mounting bar will fit into this gap (or you could cut away that portion of the mounting bar that interferes). See figure E at the bottom of page 9.





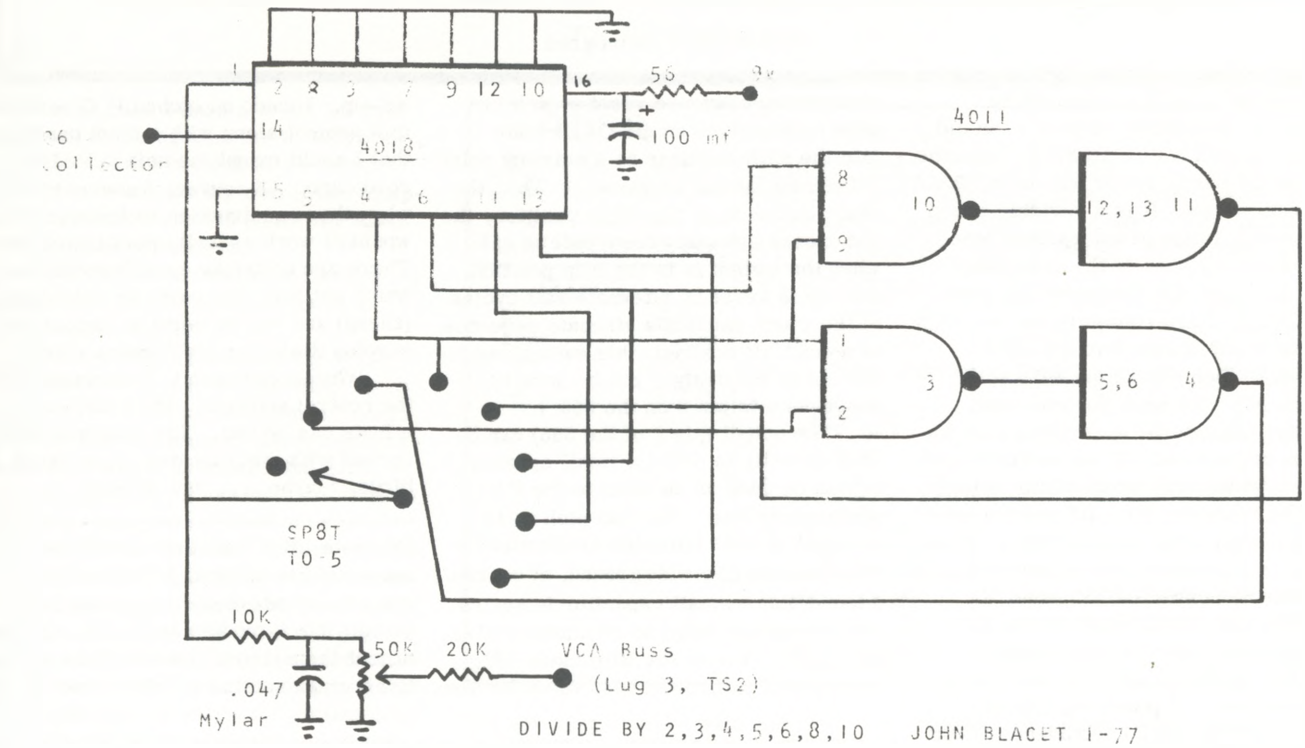




Frequency Divider

TURNS GNOME INTO "SUPER-GNOME"

By: John Blacet



Here's the schematic for the frequency divider that I've been using with my Gnome.

This is a fairly straightforward approach and the diagram shows inter-connection with points in the Gnome.

Make note of the 10K, .047 Low Pass on the output for a "mellow" sound.

The 50K level pot has to be a miniature type to fit in the Gnome case. The rotary switch is likewise miniature and has a small rubber grommet for a knob.

The only real challenge is getting everything to fit into the Gnome case. Mine ended up having the control about where the "Gnome" logo was on the case. Assembly was accomplished via perf board, wiring pencil and epoxy.

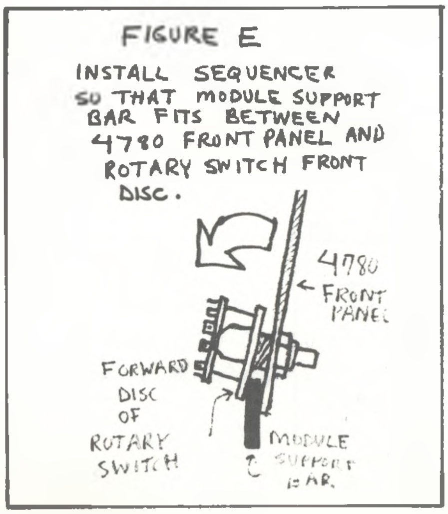
EDITOR'S NOTE:

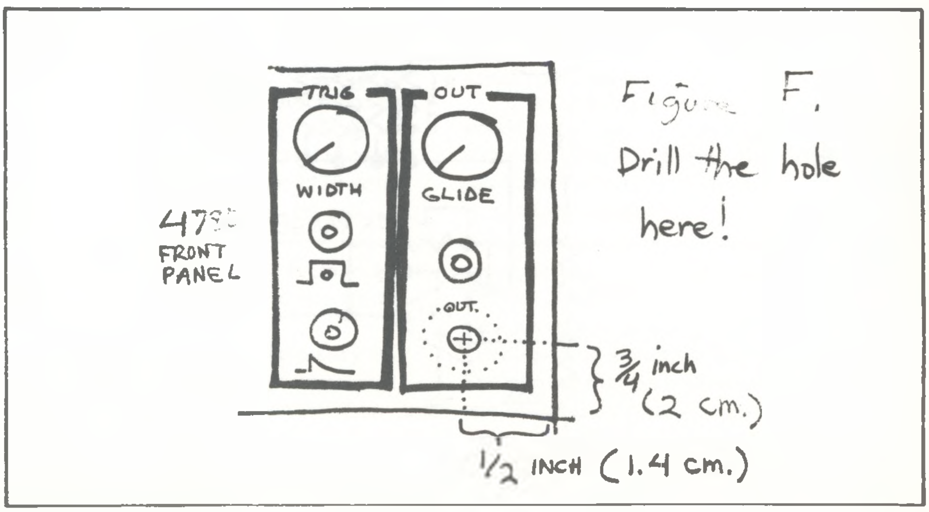
Some of you may recognize John's name. He has published additional items on electronic music in other magazines.

When we were in San Francisco for the Computer Faire, John dropped by the booth and whipped his trusty Gnome out of his backpack. He showed me this nifty "divide by N" circuit which allows you to do pseudo-tracking VCO sounds. The switch positions corresponding to "divide by 2, 4, and 8" will, of course, provide octaves below the original Gnome signal range. However, the "divide by 3, 5, 6 or 10" provides a really full bodied harmony which you somehow don't expect from a little Gnome.

This circuit is definitely worth the 5 bucks or so for the parts. Hopefully, we'll have more goodies from John in the future.

- M.J. -





Random Tone Generator

By Kenneth J. Winograd

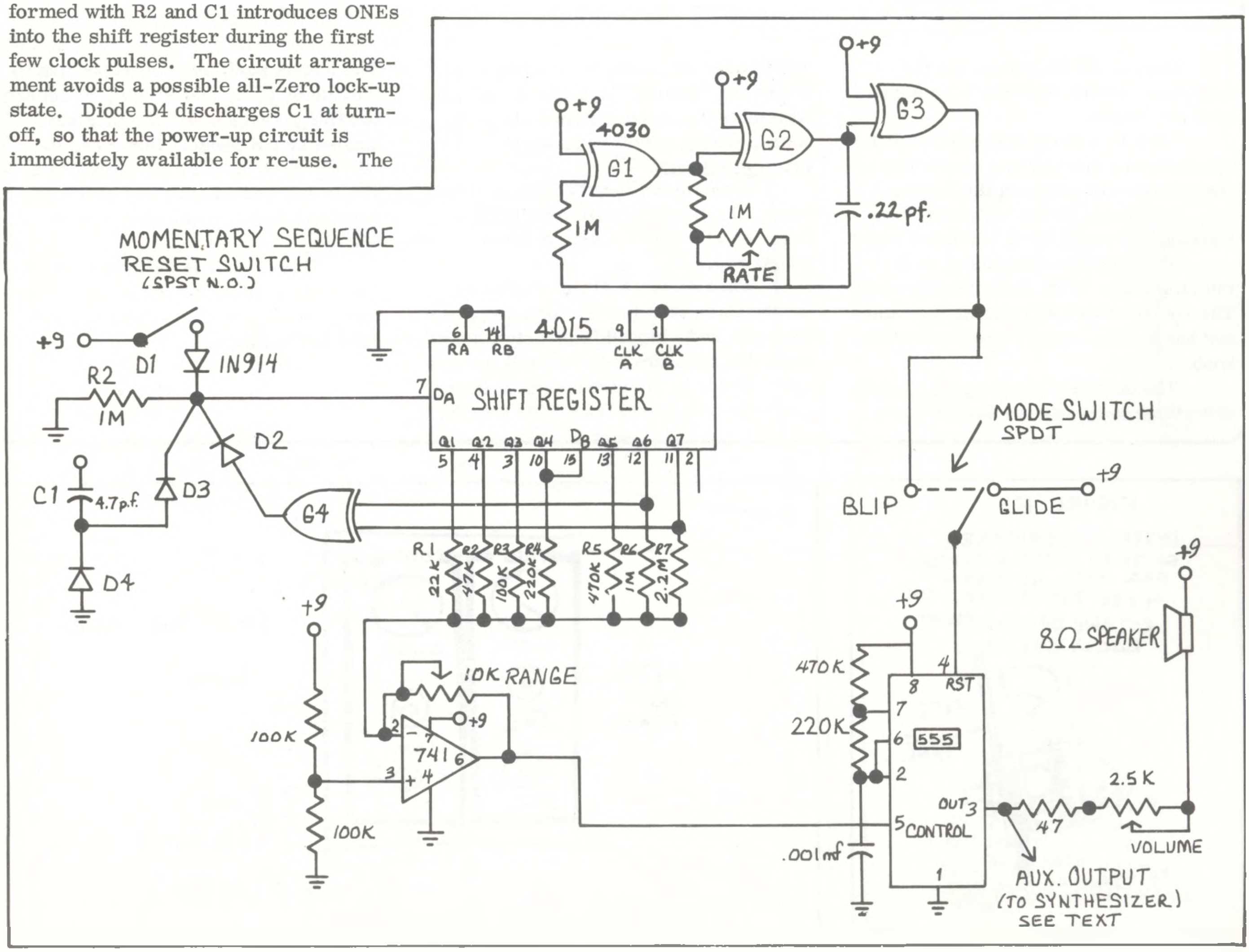
I recently found an interesting circuit, by Michael S. McNatt in an old issue of ELECTRONIC DESIGN Magazine, that I would like to share with other PAIA equipment users. It is a variable speed and variable range pseudorandom tone sequencer. As seen in the schematic shown in figure (a), exclusive OR gates G1, G2 and G3 are configured as an oscillator (adjustable through about 2-20 Hz.) which clocks the 4015 shift register IC. The shift register and exclusive OR gate G4 generate a pseudo-random sequence of 127 seven bit binary numbers which are decoded into voltage levels by resistors R1 - R7 and the op-amp (a simple D to A converter.) These voltage levels control the output frequency of an astable connected 555 timer IC. The range pot, which controls the gain of the op amp, adjusts the spread between the minimum and maximum frequency desired. A power-up circuit formed with R2 and C1 introduces ONEs into the shift register during the first few clock pulses. The circuit arrangement avoids a possible all-Zero lock-up state. Diode D4 discharges C1 at turn-off, so that the power-up circuit is immediately available for re-use. The momentary sequence reset switch is used to introduce an all-ONEs state into the shift register as a starting point reference for the sequence." When the Mode Switch is in the Glide position, the 555 moves smoothly from note to note; when the switch is in the Blip position, the 555 is reset on alternate half cycles of the clock causing a staccato sequence of notes. (If desired, this switch can be left out of the design, but be sure to short pin 4 to pin 8 on the 555.)

The output (pin 3 of the 555) can be used directly to drive a small speaker, or can be used as an input to the PAIA synthesizer line. You can pick up the required 9 volts from the synthesizer's 9 volt bus rod (likewise ground, of course.) I found that a small capacitor in series with the output helps to eliminate clicks, especially when in the Blip mode ...... experiment to find the best value for your set-up. I made an etched P. C.board for this project along with a front panel so that I could mount the unit in PAIA's road case. Of course, I see no reason why other construction techniques wouldn't work as well, perf-board, etc. There are no tricky considerations to watch out for. All parts are standard (cheap) and can be found at almost any surplus dealer or electronics store.

When constructed, experiment with the control settings. Many different effects can be had. Try a narrow tone spread with fast clocking ... a jungle of birds, maybe .... who knows!

Ed. note: Ken said that he will sell copies of his original PC layout to anyone who is interested in making this circuit into a module as he did. Contact him at the address listed in the Local Happenings section of this issue.

-M. J. -



Normalizing Synthesizer Controls

By: Garry Miller

There hasn't been a whole lot said about normalizing thus far in Polyphony, so, since I have organized a central switching board for my own PAIA equipment, I thought I would share my ideas with Polyphony readers.

My reasons for switch-orienting my synthesizer can be summed up in a single word; Speed!

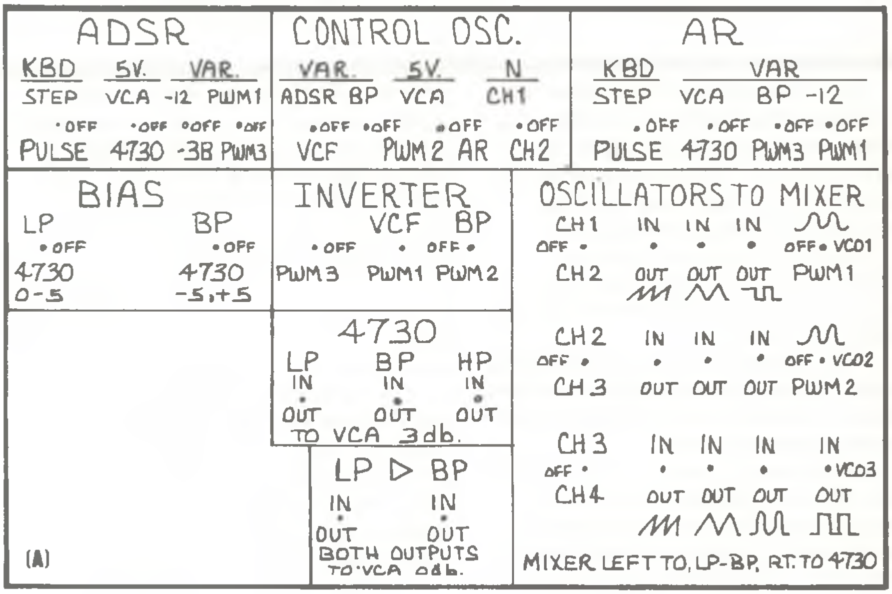
When I first assembled my PAIA equipment and got it all working, I was amazed at the variety of sounds I could get with the multitude of patch combinations available. Even with the 2720 equipment alone, if you take the time to set things up, you can have a very versatile axe. However - fast, it's not.

If you don't have a lot of equipment, and only a couple of keyboards, and don't have to sing all the time, you can possibly get by with patching your synthesizer. If you're anything like our group, though, you would have to settle for getting only a couple of different sounds out of your unit per set.

Although we use other synthesizers in our act, I now use my PAIA synthesizer extensively and with a minimum of effort for bass, as a lead instrument and also for special effects. The best part about it is; I can still patch my synthesizer any way I want - for studio work, "special" patches, or just experimenting.

What is involved is procuring about 35 or 40 switches and sitting down with your soldering iron and a roll of wire to the tune of about four or five hours of work.

My switch panel is set up on a one eighth inch aluminum panel which is the size of three single width module panels, or four by six inches. The panel is drilled out to accomodate 36 mini-toggle switches. I use two different types; single throw and dual throw. Dual throws are much more expensive than single, so should only be used when necessary. Actually, many different types of switches can be used, selector, push-button, lever-type, etc. I use mini-toggles because



they are easy to work with and are extremely fast.

So far, I have fifteen SPST and twenty-one DPDT (center off) switches in use, and will expand my panel as I add more effects and modules. They are used to route both control voltages and audio signals, and, in effect, take the place of patch cords directly.

I have permanently wired my keyboard "out" to my transposer. In turn, I have one "master" VCO permanently wired to KBD out, with two other VCO's permanently wired to one each of the transposer outputs. The signals from the oscillators are wired to the patch panel, each signal being either on or off to a dual throw switch which sends it to one of two possible mixer channels or off. The mixer, in turn, is routed to two sets of filters (permanently), right channel going to the input of a 4730, with the left channel going to both a low pass and high pass filter simultaneously. Whenever any of the filter switches are in the "in" position they are sent directly to the inputs of the VCA, the 4730 going to the 3 db. input, and the -3B and -3L to the 0 db. input. The output of the VCA goes directly to the audio amplifier.

Signalwise, that's pretty much it, except for the noise output which I have sent either to channel one or channel two of the mixer. (Note, that, when the oscillator signal switches are in the out position, the mixer can accomodate any other signal, sending it directly to the filters.)

The voltage control outputs are similarly routed to the control inputs via a number of dual throw switches acting as selectors. (see figure (a))

Obviously, there are as many possible switch combinations as there are patches available, and each synthesist would want to tailor his system according to his needs. Agreed, many owners of PAIA equipment would not need nearly so many permanent set-ups, and there are probably those who would need more.

I've found that my system works wonders for me. I hope that I've helped someone else out there. Now all that you have to do is find room for a four by six inch panel!

LETTERS:

............continued from page 4

lot has been said about the design of individual modules, nothing is mentioned about putting them together to form a well planned synthesizer system. Although your equipment would be functional no matter what the arrangement of modules, certain configurations lend themselves to a more orderly thought process when making patches.

To begin with, it is necessary to classify modules into basic functional groups: Sound sources, Controllers, and Processors. Synthesizer sound sources are VCO's and Noise Generators (unfortunately, the PAIA Noise Generator is associated with the Control Oscillator which is is a controller, fouling up the logical set-up somewhat). Other controllers are the keyboard, Sequencer, and Envelope and Function Generators. Everything else is a processor (the Envelope follower/Trigger is sort of a special case since it processes a signal source for use as either a signal or a control).

continued on page 29 ....

Eliminating Patch Cords Without Eliminating Capability... A Practical Approach

By: Gary Bannister

In working with the PAIA synthesizer, I have found that the main drawback of studio synthesizers is still present -- PATCH CORDS! Those little demons are either just a little too short, or three times too long for the patch.

Totally prepatched is not the answer, either. Great on the road, but get them in a studio and everything sounds the same.

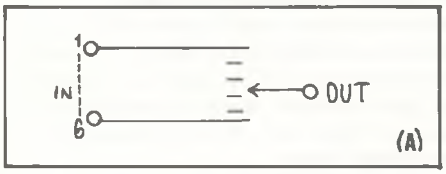
The ultimate would be a combination of both. Certain patches quickly available but easily modified and/or removed for more progressive work.

Unfortunately, only one such synthesizer with these capabilities exists. Several have made some attempts to interface a unit with the outside world, but the ARP 2600 still reigns supreme.

I've played and worked, laughed and cursed, created and destroyed, and some ideas are coming to a head. I would like to share them.

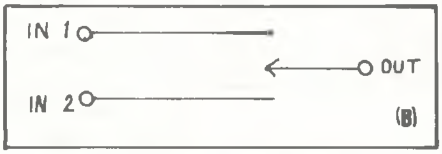
My first system was a PAIA 2720/A. Quickly there followed additions and wing cabinets. One of my first concerns was quickly switching audio sources into and out of processing modules. Plugging patch cords from sawtooth to square to triangle took too long, and plugging from one output to another caused pops in the amplifier.

The circuit below was the first that came to mind. Simple construction with audio jacks and rotary switch in a bakelite box, all parts readily available.



Note that this is a six-position switch. This would accommodate square, sawtooth, triangle, sine, modulated pulse, and noise. This switch, being a passive unit, is bi-directional; that is, inputs could be outputs and vice-versa. I built two such switches, and they serve me very well.

In any system, it seems that there are more control cords than audio. I guess this is as it should be. In any case, more need arises for switching controls than audio. The problem here is that in many cases controls may have to be changed (like the above audio example), modified, or shut off completely. Each change can be effected with the output level controls on many of the modules, but at the expense of accurately repeating the patch. The following sketch is the germ idea.

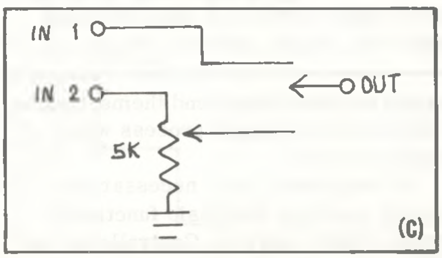


A simple SPDT switch and three pin jacks are all that's required. I suggest a toggle type switch. They give you direct visual indication of which input is selected (but so do the square slide type) and you don't have to drill square holes.

Digressing for a moment, a strong point should be made about the visual feedback. It is extremely important that all switch positions be accurately labeled. One should be able to simply look at your switches and tell what function they serve. In the previous examples it is not so important to know which jack connects to what function, but which switch position connects to which jack. The particular input to any one jack may change from patch to patch, but a given switch position should produce a given response in any one patch.

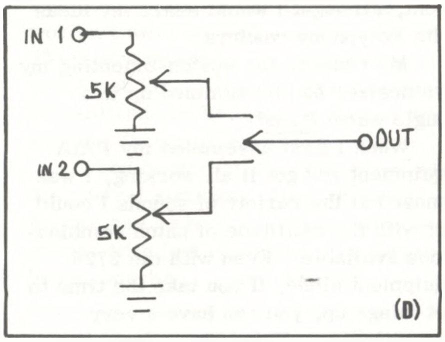
Back to control switching:

The SPDT set up works nice for two different inputs, but what if you want to simply change one input? A good example would be a different amount of filter sweep from the same envelope generator. Obviously this can be done with the variable output jack, but each change reduces the possibility of exactly repeating the others. A slight change here takes care of the problem:



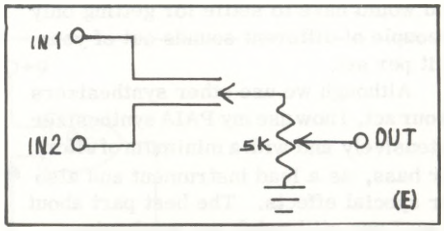
Patching the variable output into (1) and the fixed output into (2) allows for two totally variable outputs which are instantly and exactly repeatable.

But why not go one better:



This would allow the same effect as the first, but with even more versatility.

One more variation, now. I have one of these, but I haven't found much use for it. Maybe you can.

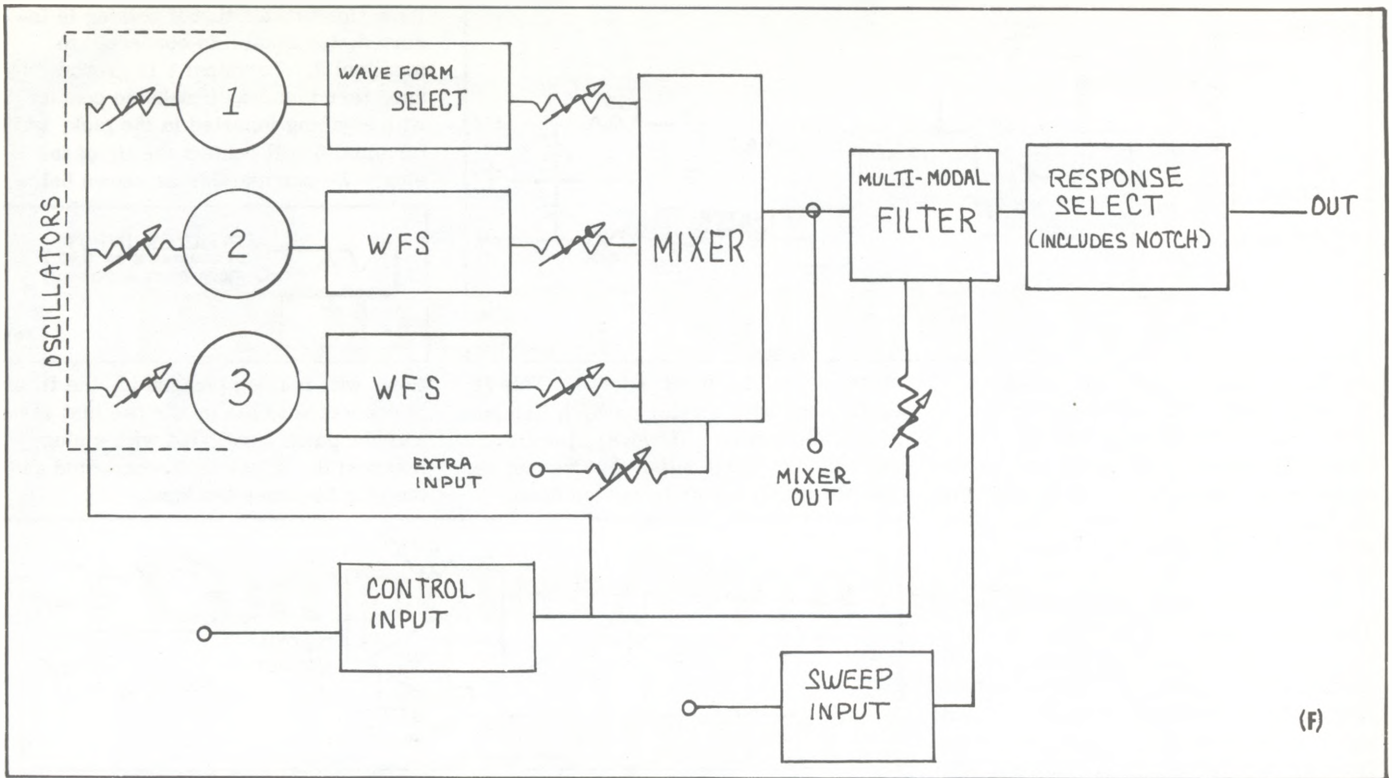


One important thing to note. Each of these circuits require a ground reference. I built two of the audio switches and two of the control switches in the same box. I always use shielded cable, so all of the grounds (including audio) can be tied in through the shield. If you build the control switches alone, it will be necessary to make provisions for a separate ground input jack.

The circuits shown to this point did solve one of the problems concerned with here: quick and accurate selection of a patch. However, the other problem, length and number of patch cords, has been multiplied. To select one of two controls now requires three cords instead of one, and to select one of four oscillator outputs requires five cords.

In my search for innovative new sounds, I found that I required more oscillators and filters, and that all must track the same source, the keyboard mostly. I purchased three 2720-2A oscillators and one 4730 filter. Any one can see how many patch cords this would require! There are a couple of hidden problems, though.

How do you construct a chord with the three oscillators and then process



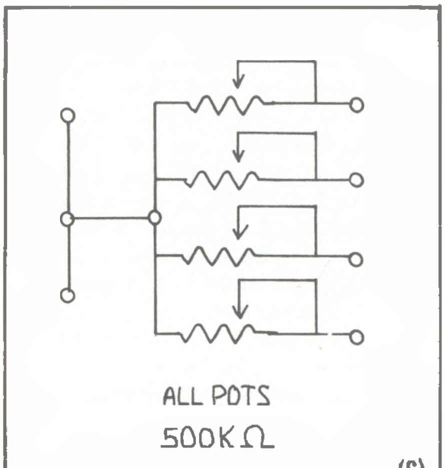
(F)

it with the filter? First off, the filter and all three oscillators must track. A manual transposer handles this well. Now you need a mixer between the oscillators and the filter. Before you are done you have about $175 tied up in a "voice module". But take heart!

Let us look at a block diagram of what we need. (see figure F)

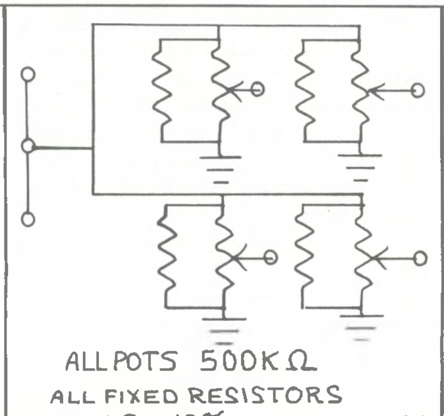
Some rules to follow: Do the above with a minimum of patch cords, and don't lose any of the versatility of modular systems.

I ran into one problem right away. It is a function of the Manual Transposer, whose schematic follows:



(G)

The effect is to put more resistance into the summing networks of the oscillators and filter. This works great, but it also increases the input impedance of the modules, making them more susceptible to hum pickup. I changed the circuit as follows:

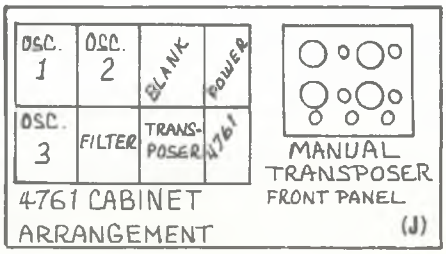


(H)

The equivalent impedance of each pot is now approximately 50K ohms, and the total load on the input circuit is about 12K ohms. This produces negligible loading effects and no susceptibility to hum.

Note that three oscillators, a filter, one power supply (all that's necessary is ± 9v.), the manual transposer, and the wing cabinet panel take up almost all of the toom in a 4761 Road cabinet. In fact only two single widths are left, I'll use them later.

Now keep in mind the importance of visual feedback. I decided to arrange my cabinet like so:



(J)

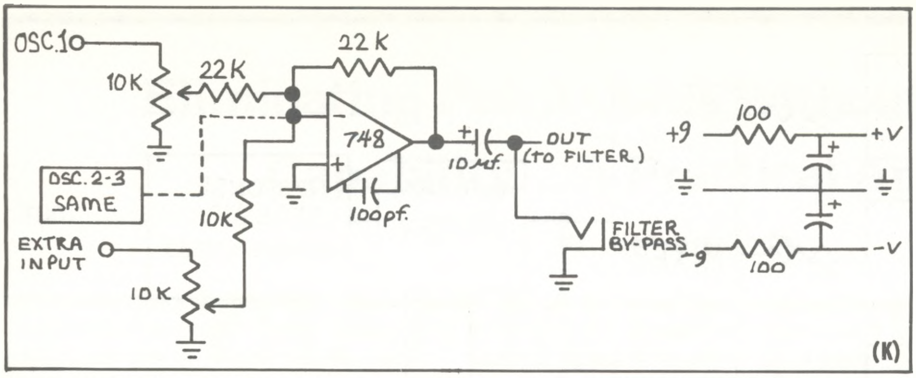
Note also the layout of the Manual Transposer front panel, and how it corresponds to the layout of the oscillators and filter. Here is a prime example of visual feedback!

Now, how do we get rid of patch cords? All prepatched synthesizers use the same idea, hardwire (solder connections) BEHIND the front panel. We will use the same idea here. Connect a wire from each oscillator to its corresponding control on the manual transposer.

Each oscillator and the filter have three control inputs. I connected to the first (leftmost) of these in each case. Remember that while tracking all three oscillators from the same source that the first pin jack cannot be used for an external input. However, the other two can still be used for vibrato, etc. The same applies to the filter. Now, playing the oscillators is a simple matter of connecting ONE control voltage to the Manual Transposer (and ground of course) and tuning each oscillator and filter with its corresponding pot on the Transposer. (I also have the RANGE controls mounted on the front panel, but that's another story). Note that this requires ONLY ONE patch cord for four units, and still the other inputs are available.

We've solved the control problem, but we still have to mix the signals before we get to the filter. Another 4711 Mixer adds ultimate versatility, but it also adds more cost for features we don't need (like panning and stereo). The circuit shown in figure (k) is no surprise, and serves the purpose very well.

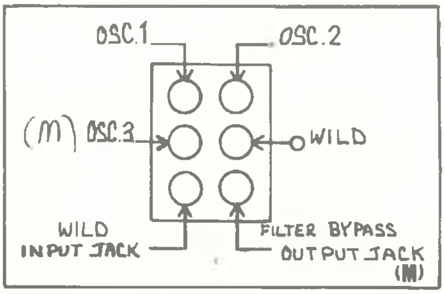
Each oscillator has a separate level control. Maximum volume is unity gain



(K)

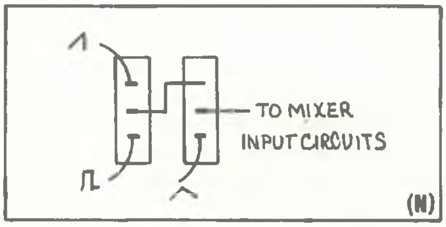
for the oscillators and 2 for the extra (or wild) input. The wild input is for adding another oscillator, noise, or cascading filters. The extra gain makes up for some signal loss. Be careful that the total input does not exceed that allowable for the filter, or distortion will result.

Also included is a filter bypass output. This simply parallels the filter so that the mixer output may be processed by another filter in parallel, or the included filter may be bypassed completely. This unit may be breadboarded and suspended from the back of the panel. Here is a front panel layout. This mixer will take up one section of the blank space in the 4761 cabinet.



(M)

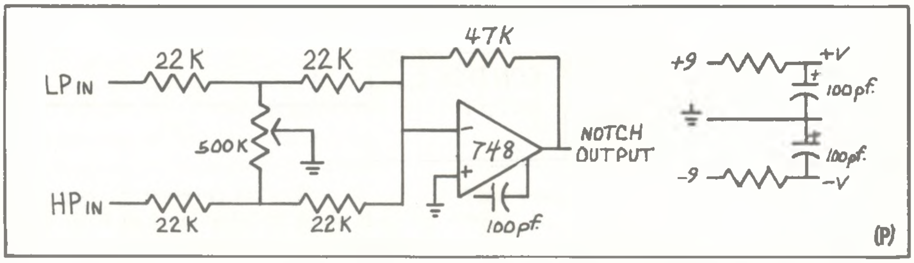
We still have one problem. How do we select which waveform goes into the mixer? This will require some modification to the oscillator panels. A simple Or type of switching is used, and constructed from two SPDT switches. Each oscillator will require one such switching circuit.



(N)

I mounted these switches just below the Pulse Width control knob, and included appropriate labels. One switch selects either the "hard" or "soft" waveforms. When wiring these switches, wire directly to the proper point on the output jacks of the oscillators. Here is versatility again, since the oscillators may be mixed, yet each output is separately available.

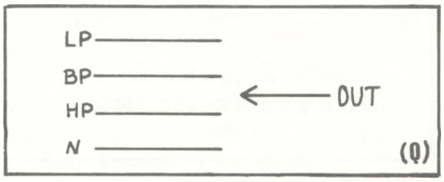
The only thing left is to quickly select one of the filter outputs. This is easily done with a rotary switch like our first audio switch. However, the filter has a hidden notch output that we can use. Another small mixer is needed here.



(P)

This circuit produces an output that is continuously variable from lowpass through notch to high pass. I must admit I have found this tuning ability to be of little use, so if it is not desired simply remove the 500K pot and replace the two 22K resistors in each leg with one 47K resistor.

Now the filter outputs may be selected like this:

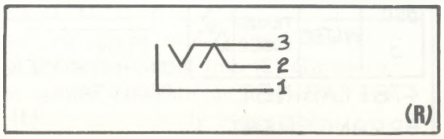


(Q)

Again, wire directly from the output jacks so that each output still remains individually accessible. Build this unit into the remaining single blank panel space in the 4761 cabinet.

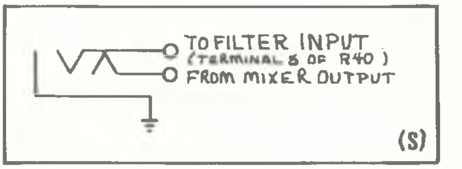
We still have one problem of "preventative medicine". We must take care that we do not have two inputs to the filter. If the mixer output is wired directly to the filter input jack (like it should be) provisions must be made that something cannot be plugged in the jack at the same time. We could remove the jack, but this would decrease the usefulness of the module. Therefore, we must change the input jack.

A closed circuit type should be used to replace the filter input jack. It is drawn like so:



(R)

Its action is such that if nothing is inserted, terminal 2 is connected to terminal 3. Terminal 1 is ground. In use, terminal 2 will make no contact with anything inserted in the jack, while terminal 3 will contact the tip of the plug. We can use this as shown below.



(S)

Now, with nothing plugged in, the filter is connected to the mixer (we just saved another patch cord), but with a plug inserted the mixer is disconnected and the plug becomes the input.

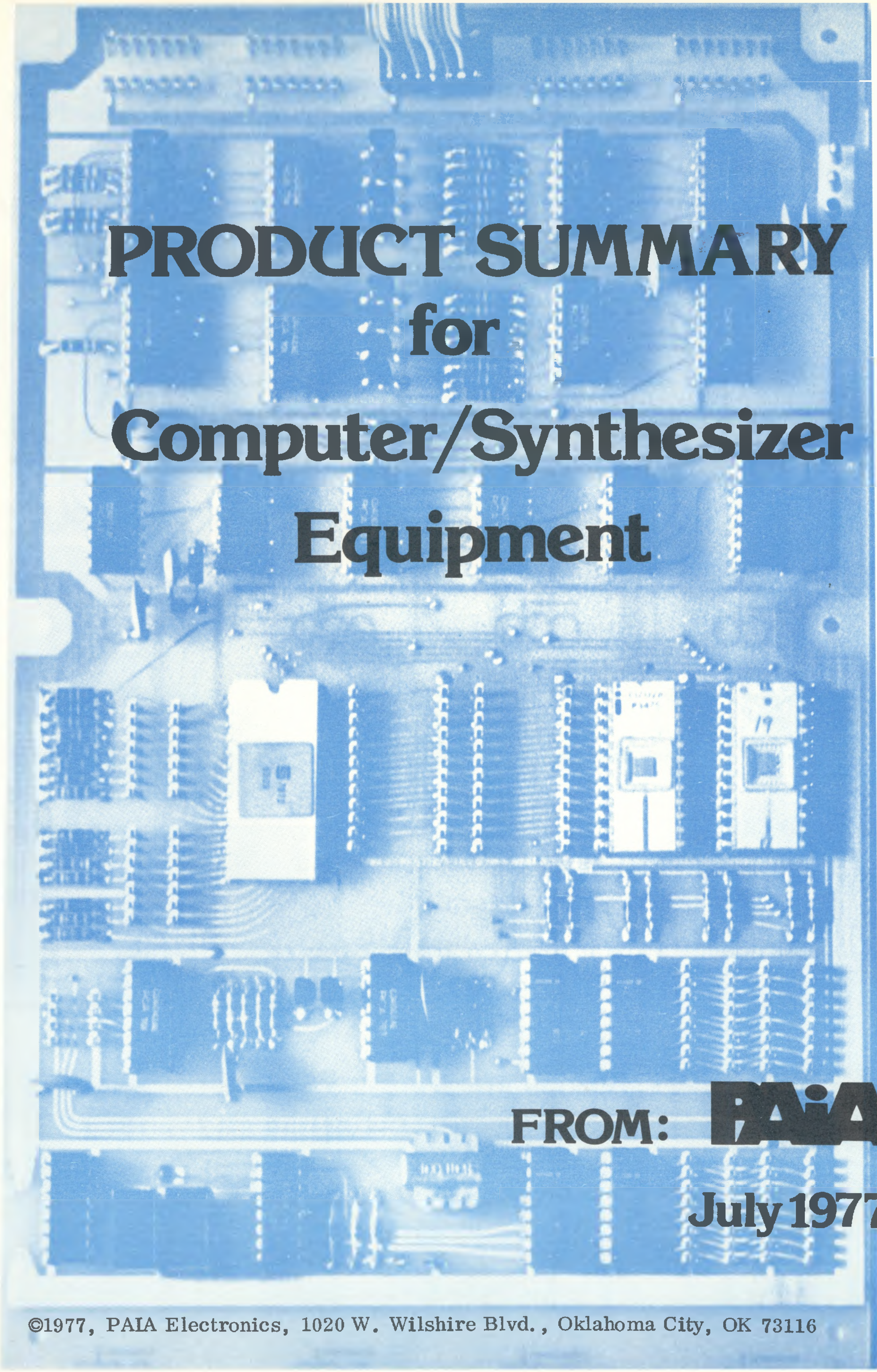
With assembly completed, let's see what we have. (1) Control voltage input requires one cord into the unit. (2) Audio output from filter requires one cord. (3) Ground requires one cord -- WHICH MAY BE ELIMINATED WITH SHIELDED AUDIO CABLE! This means that this unit requires only THREE patch cords to run all units. A fourth will be necessary in order to sweep the filter, and none will need to be disconnected, eliminating pops, thumps, and time-consuming fumbles. In addition, ALL INPUTS AND OUTPUTS ARE INDIVIDUALLY ACCESSIBLE.

I think that a few remaining notes on the uses of this unit may be in order. One of the most useful features of this is the wild input for the mixer. Obviously, another oscillator could be added for a bigger chord, or noise mixed in. But what about a 24 db/octave filter?

For this effect, two 4730 filters must be SERIES CONNECTED. Apply an audio signal to the first filter (not in the pre-patched unit). Take the low pass output and route it to the wild input of the mixer. Adjust all oscillators to minimum and the wild input to suit.

Now, run a control voltage into the Transposer. Using the jack on the transposer that corresponds to the filter, take the control voltage to the first filter. Now, carefully tune each filter to the same cut-off frequency. This is best accomplished by using a little Q on each filter (about half way around) and tune for maximum resonance. Note that this also can produce a very high Q filter, so take care that no distortion results as the Q is advanced. Now, the cut-off of the filter set up may be controlled by

...........continued on page 30



PRODUCT SUMMARY for Computer/Synthesizer Equipment

FROM: PAiA

July 1977

COMPUTER AIDED SYNTHESIS
new real-time performance capabilities
For the Musician/Composer

4700/J

By anyone's standards this is a BIG synthesizer, as you can see by reviewing the module complement.

Like our other packages, it may be used without a computer as a normal monotonic synthesizer.

With a computer in the loop, you are ready to do polyphonic instruments, multi-track recording work, and innumerable composer and performer assisting functions that are only possible with a computer/synthesizer combination.

The 4700/J module complement consists of: the 8782 Encoded Keyboard, digital to analog converter, QuASH, two 4710 Balanced Modulator/VCA's three 4720 Voltage Controlled Oscillators, 2720-5 Control Oscillator/Noise Source, three Watt Block Power Supplies, two Envelope Generators, Reverb and two Wing Cabinets. Included are step-by-step instructions and using manual. **Also includes a 4730 Multi-modal filter and one 4711 Stereo Mixer.**

#4700/J ......SYNTHESIZER KIT $549.00
(shipped freight collect)

8782

An n key roll-over scanning matrix encoder tied to a 37 note AGO keyboard provides 6 bits of data and both STROBE and STROBE control outputs. Input control lines to the encoder include SCAN (starts and stops the encoder clock), RESET, START and RANDOM making the keyboard universally applicable to all computer/processors from the very largest to the very smallest.

Housed in a trim and sturdy vinyl covered road case, the kit consists of all parts including keyboard, power supply and detailed assembly instructions; software overview for computer applications and detailed instructions for digital sample and hold.

#8782 .. ENCODED KEYBOARD KIT ....$109.95 ....32 lbs.
(shipped freight collect)

Up date PAIA (or other)keyboards to the new digital format. Consists of encoder circuit board and all parts required to modify your present keyboard for use with computers and/or the 8780 D/A. Off-print instructions from "Computer Music Without The Computer", Polyphony 4/76.

#EK-3 .. KEYBOARD ENCODER EXPERIMENTER'S KIT $14.95
(plus $1.00 postage)

8700

An exceptional price on an applications oriented 6503 based microprocessor system featuring: 1K bytes RAM locations (512 bytes supplied), 1K bytes ROM locations (256 byte monitor included), two 8 bit input ports, two 8 bit output ports, one latched and one buffered.

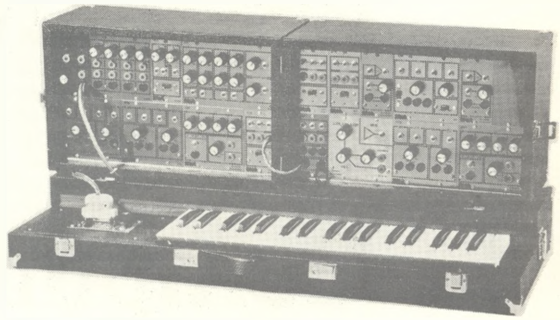
A 24 key touch operated keypad is used by the monitor to allow entry and execution of user programs and is also user definable. Two latched seven segment displays are used by the monitor to display memory location and contents, also easily user programmed.

An optional Cassette Interface is available ($22.50) that fits entirely on the processor board allowing the use of any audio cassette recorder for program storage.

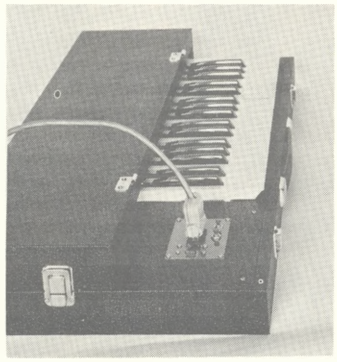
The 8700 fits in a space reserved in the 8782 Encoded Keyboard's case. PAIA software support available for Electronic Music Synthesizer interface.

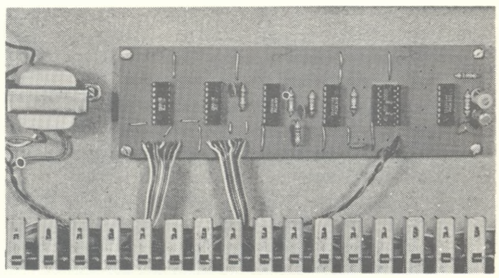
#8700 .. COMPUTER/CONTROLLER KIT $149.95
(plus $3.00 postage)

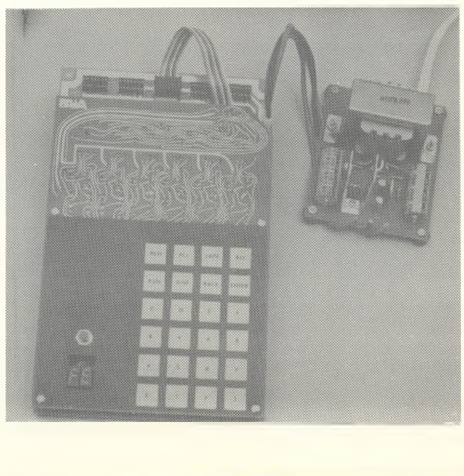
EXPANDABL



MODULAR···TO SUIT YO



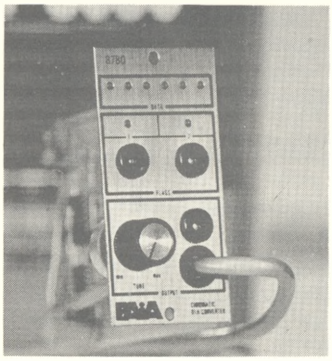


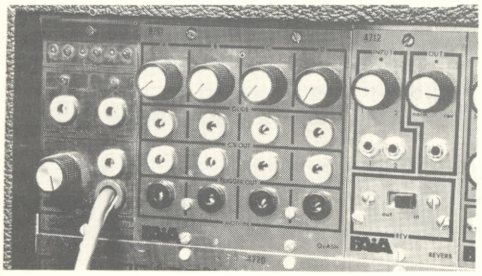


COMPUTER MUSIC that sounds not just good but great! For the Computer-oriented Hobbyist/Professional

PACKAGES

R CUSTOMIZING NEEDS





4700/C

The ideal monotonic starter system. Can be used without a computer/ processor as a conventional electronic music synthesizer.

By simply unplugging the synthesizer head from the keyboard, a computer can be put into the loop to provide power and versatility never before possible for synthesizers of any cost.

The 4700/C module complement consists of: the 8782 Encoded Keyboard, Digital to Analog Converter, 2720-5 Control Oscillator/Noise Source, 4710 Balanced Modulator/Voltage Controlled Amplifier, Reverb, Voltage Controlled Oscillator, Voltage Controlled Filter, Envelope Generator, two Watt Block Power Supplies and a 4761 Wing Cabinet. Complete step-by-step instructions and using manual.

#4700/C .....SYNTHESIZER KIT $325.00
(shipped freight collect)

8780

Unlike more conventional R-2R ladder type digital to analog converters, the PAIA 8780 kit is based on a multiplying principle that allows the module to generate the exact exponential stair-step function required to mave even the simplest linear response oscillators and filters produce equally tempered musical intervals. The 8780 uses only 6 bits of data to generate over five octaves of control voltage. In an 8 bit system, the remaining two bits are ordinarily reserved for trigger flags, but may be used to provide micro-tonal tunings.

The module is physically and electrically compatible with the complete line of PAIA music synthesizer modules and is easily interfaced to any micro-processor with or without hand-shaking logic.

#8780 .. D/A CONVERTER KIT ...... $34.95.....(plus $1.00 postage)

8781

The least expensive way to provide multiple Control Voltage channels in a computer based synthesizer system is to multiplex the output of the D/A using multiple, computer addressable, Sample and Hold circuits. The QuASH provides 4 of these outputs in a single module.

Other features of the module include: adjustable glide rate for each channel with the glide selected or de-selected under computer control, individual trigger "gate" signal associated with each output channel and individual modulation input for each channel that is non-interactive with other channels.

On-board address decoding allows up to 4 QuASH to be bussed together in a single system without external logic and Bank Select input allows external logic expansion beyond this point.

NOTE: THE QuASH MUST BE OPERATED UNDER COMPUTER/PROCESSOR CONTROL.

#8781 .. Quad Addressable Sample & Hold $34.95
(plus $1.00 shipping)

next ?

NOT A DEAD END

But continuing development keep in touch

COMPUTER/SYNTHESIZER SPECIALS

To really get you started in the right direction, we're offering the following special packages:

P-4700/C MONOTONIC Package

KIT		REGULAR PRICE	SPECIAL PACKAGE PRICE
4700/C	Synthesizer	$325.00	$325.00
8700	Computer Controller	$149.95	$149.95
PS-87	Power Supply	24.95	free
CS-87	Cassette Interface	22.50	free
PMUS	Software Cassette	5.00	free
		$527.40	$474.95

(shipped freight collect)

P-4700/J POLYPHONIC Package

KIT		REGULAR PRICE	SPECIAL PACKAGE PRICE
4700/J	Synthesizer	$549.00	$549.00
8700	Computer/Controller	$149.95	$149.95
PS-87	Power Supply	24.95	free
CS-87	Cassette Interface	22.50	free
MUS 1	Music firmware prom	20.00	free
PMUS	Software Cassette	5.00	free
Additional	512 Bytes RAM	7.00	free
		$778.40	$698.95

(shipped freight collect)

** OTHER STUFF **

PS-87 ····· Computer/Controller Power Supply ···················· $24.95 ···· (plus postage 3 lbs.)
CS-87 ····· Cassette Interface Option ································ $22.50 ···· (plus $1.00 postage)
MUS 1 ····· Music firmware prom ···································· $20.00 ···· (plus $1.00 postage)
PMUS ······ Software Cassette ·· $ 5.00 ···· (plus $1.00 postage)

** Attention KIM owners **

We have several products available to owners of MOS Technology's KIM series of micro-computer products.

KIM Synthesizer Package

Detailed instructions on interfacing KIM-1's to the PAIA Synthesizer equipment. Package includes software cassette with multi-track recording/playback, compositional (computer "composes" music) and several polyphonic routines.

KIM MUSIC PACKAGE #KMUS $4.95 (plus $1.00 postage)

[The KMUS package is FREE for the asking when you purchase a 4700/C or 4700/J Synthesizer package.]

** WANT TO KNOW MORE? **

An information package comprised of Polyphony offprints, using manuals and more is available to explain in detail how it all works and the unlimited applications possible with computer technology applied to music synthesis.

I-COMP (Friendly Stories About Computer Music)..........$1.00 postpaid

PAIA ELECTRONICS, INC. • 1020 W. Wilshire Blvd. • Oklahoma City, OK 73116

LAB NOTES

IN PURSUIT OF THE WILD QuASH

by John S. Simonton, Jr.

Now that we have a way to interface our synthesizers to computers - the 8780 D/A - we can begin thinking of ways to independently control large numbers of musical elements simultaneously. Lots of VCO s, lots of VCFs.

The first time that you think of this your reaction may be something like:

WOW! - ALL THOSE D/As.

Multiple D/As (one for each control "channel") would be a possible way to go. An expensive way - at $35.00 each, controlling just 4 VCOs means almost $150 worth of just D/As.

There's a much cheaper way.

You may find this a little circuitous, after working so hard at our digital interface, but we're going back to analog Sample and Hold circuits.

Now wait, don't panic. These S/H's are nothing like the ones that we're accustomed to. They don't have to hold a voltage steady for a long period of time - only a few milli-seconds. Long before even that short time has passed we will have used the computer to come back and re-write the correct voltage into the circuit. Computer re-freshed S/Hs.

Magic!

When you're designing a S/H to be good for only a fractional part of a second it gets really easy. Like this:

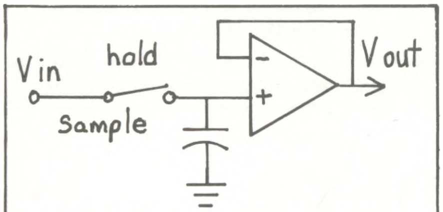


figure (a)

I'm sure that we've all seen this kind of thing before. It's an op-amp used as a unity gain voltage follower.

When it comes time to take a sample, the switch closes causing the capacitor to charge up to the input voltage. The output of the voltage follower "follows" this voltage (what else?), and when the switch opens again, the capacitor "remembers" the voltage.

One of the characteristics of this circuit is that the "+" input represents a very high input impedance to any load that it sees. A relatively small capacitor can accurately hold a voltage for almost a second.

Now, we're not going to use a mechanical switch here. Last time, we looked at the 4051 multiplexer and decided that we would be using it a lot. And we are, just not this time.

This time, we're going to use a very close relative of the 4051 - the 4052 (I defy you to get any closer than that). The 4052 looks like this:

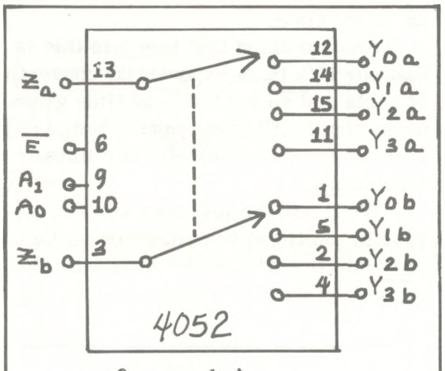


figure (b)

and whereas the 4051 was an electronic equivalent of a Single Pole Eight Throw switch, the 4052 is like a Double Pole Four Throw one.

Which pairs of switches are to close is specified by the two address lines (A_0 & A_1). The switches actually close when the $\overline{E}$ pin goes to ground.

Using 1/2 of one of these devices we can come up with a Quad Addressable Sample and Hold (QuASH?) that looks like figure C, and it works about the way that it looks. An address applied to the A_0 and A_1 pins sets up one of the four switches and when the $\overline{E}$ pin is taken to ground that switch closes connecting the output of the D/A to the selected S/H. Simple.

That takes care of our control voltage output - but there are still other things to think about. For instance, we need a trigger flag (gate signal) to go along with each of the control voltages to take care of things like triggering envelope generators. *(1)

An easy way to handle this is to use the other 1/2 of the 4052 to route one of the two trigger flags available from the D/A to an output corresponding to the control voltage output. And since we're time sharing the D/A we also need some way to hold the status of that flag during the times that other control

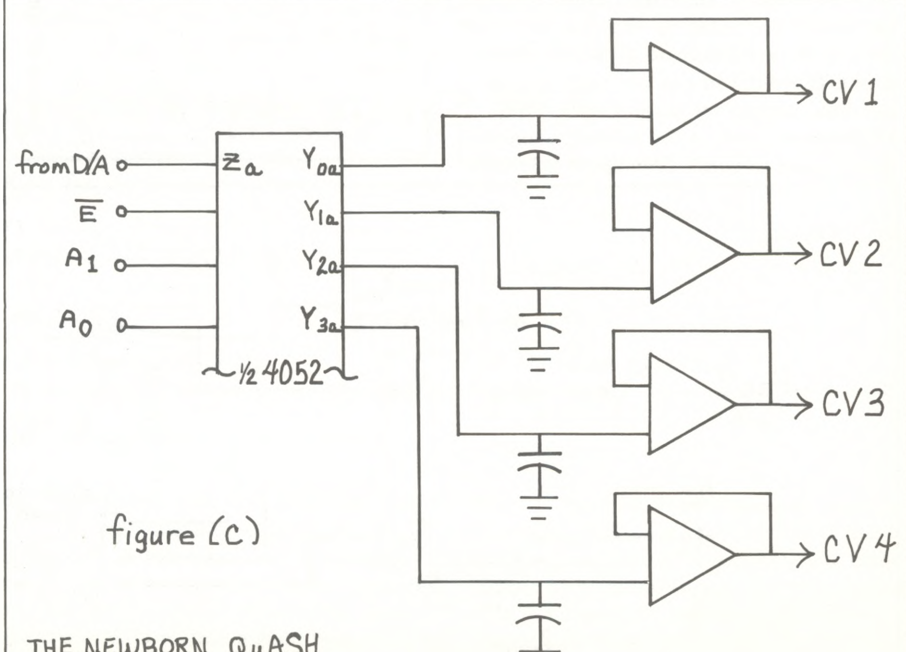


figure (c)

THE NEWBORN QuASH

channels are being addressed. Do latches come to mind? Forget them - in this application they're going to be far too expensive and complex by the time we get them to act the way we want.

Instead, we'll use a small capacitor and a CMOS inverter like this:

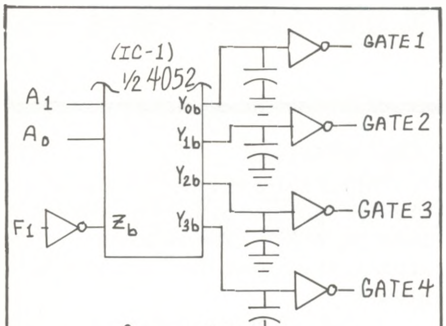


figure (d)

This is a little S/H in its own right - but it doesn't hold an analog voltage, only a "1" (output high) or "0" (output low).

Oh, yes - since we are buffering the condition of the capacitor with an inverter, we need to also invert the trigger line going into the 4052 so that everything comes out right. That's why that other inverter goes between the trigger flag line from the D/A (F_1) and the Z pin of the 4052.

But, there are two trigger flags available from the D/A - and here we are only using one of them. Waste, ugh.

Let's do something neat with the left over flag, something really sexy. Let's use it to:

SELECT GLIDE

(tah-dah)

You may think that because we're time sharing the D/A we've eliminated the possibility of doing things like this, but we haven't. In a functioning system the S/H's are being up-dated so fast that we can in fact generate glide the same way that we did in our old pure analog system, simply by placing a variable resistor in front of the holding capacitor. We'll use a regular 4066 Quad Bilateral Switch to turn the glide off by shorting out the resistor (so that the glide is on when the switch is off), and to latch the status of this glide bit we'll use the same capacitor/inverter trick that we used on the other flag. One section of this circuitry looks like figure e.

For programming reasons, it will be handy to have the glide select bit (which is now flag 2) be a "1" when the glide is enabled and that requires a second inversion - between the trigger output of the D/A and the Z pin of this new 4052.

And now here we are with 1/2 of a 4052 left over.

Don't you believe it.

Since we will frequently have more than a single synthesizer module controlled from one of our control voltage outputs (two VCO's or a VCO and VCF would be two typical cases), it will be handy to have a modulation input associated with each control channel so that all modules driven from that channel will experience the modulation at the same time.

Another thing that ties into this is that our D/a is an exponential converter of sorts and so for the first time gives us the opportunity to do equally tempered vibrato (for example) with our linear oscillators.

We'll use the left over section of 4052 to multiplex a modulation voltage back into the D/A in the same way that we multiplexed the control voltage out. Like this:

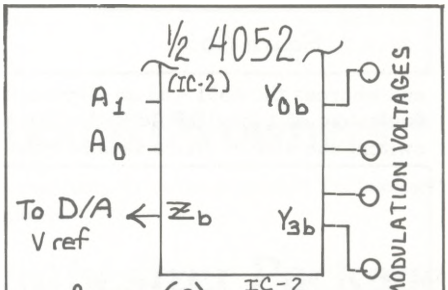


figure (f)

Because the modulation voltage corresponding to a given control channel is applied to the D/A only when that channel is re-freshed, you may think you will be able to hear the modulating influence as a series of steps. But you don't for the same reason that the glide doesn't appear to be a series of steps. Everything is just happening too fast.

One last detail and we're done with the design of this circuit.

Addressing (selecting) one S/H out of the four on this card is of course handled by the address pins of the 4052's. But, many systems will not stop at just 4 outputs; some folks, I'm sure, will want to take the system to the limit (in practical terms about 32 outputs) - which implies that more than one of these cards may (and probably will) be used in a system. We need a way to be able to select not only one of the four outputs on this card, but also a means of selecting one card from many.

Here's the address decoding scheme we'll be using:

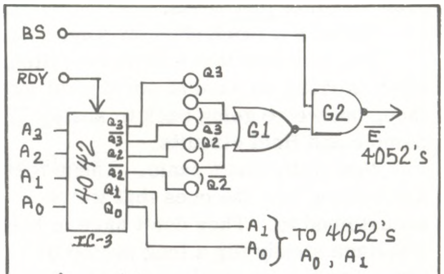


figure (g)

The 4042 Quad Latch is an old friend - here we're using it to latch the computer's 4 least significant address bits at the same time that data is put out to the D/A (the $\overline{RDY}$ line on this card is connected the same as the corresponding line on the D/A).

We want to latch these address lines because the WRITE cycle of any computer we come up with is going to be much shorter than the time required for settling of the D/A and S/Hs. Latching the address lines allows us to output

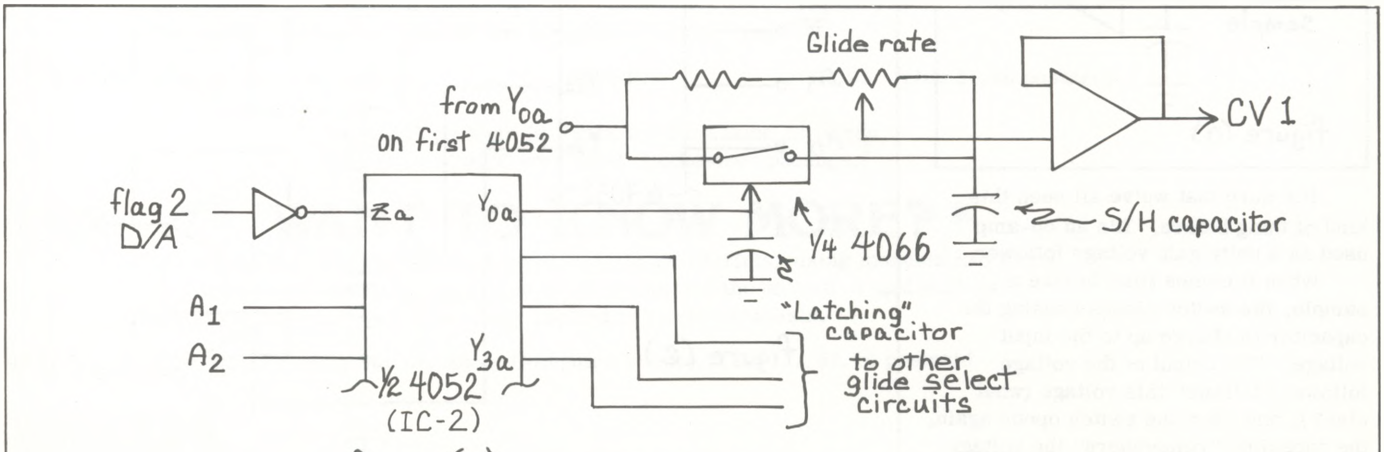


figure (e)

data and then wait (or do something else) while these analog circuits get to where they're supposed to be. *(2)

Notice that the Q_0 and Q_1 outputs of the latch - corresponding to the two least significant address bits - go directly to the 4052's where they serve to select one of the four outputs.

Notice also that Q_2 and its complement $\overline{Q_2}$ as well as Q_3 and $\overline{Q_3}$ from the 4042 come out to pads on the circuit board. By jumpering these outputs to the inputs of the NOR gate G1 we can determine which group of addresses the card we're working with represents.

For example, if we connect the inputs of G1 to $\overline{Q_2}$ and $\overline{Q_3}$ then this block of four S/H's occupies the addresses 00XX in binary where XX represents the bits that select one of the four S/H. Address 0000 corresponds to the first S/H, 0001 to the second, and so on. By connecting the inputs of G1 to Q_2 and Q_3, the S/H's occupy the address 01XX. The first S/H is 0100, the second 0101, and like that. This scheme allows us to easily use up to four of these expanders (16 outputs) in a system without needing to do anything but set the jumpers properly.

You will notice that there is another line coming out of this decoding circuit which is labeled "BS". This is not my opinion of this whole mess, it's a means by which we may expand the system beyond even four expander modules - BS stands for "Bank Select" and as long as this line is held at a logical "1" level the system operates as described to this point.

But, when the BS line is pulled low one input of the NAND gate G2 is not fulfilled resulting in its output being high which in turn holds the 4052's enabling input $(\overline{E})$ high - which means that none of the switches in the multiplexer will close (even if addressed otherwise) and none of the S/H's will be selected.

External decoding circuitry is required to drive the BS input, naturally, but we would begin to need external circuitry at about this point anyway to buffer address lines. The decoding required here will be covered in the instruction manual for this kit.

When we tie all of these bits and pieces together, we come up with a thing that looks like figure H, our complete QuASH. And in the interest of saving space and time, we will from this point forward represent it with the symbol shown (at least until we can come up with something more abbreviated). The knobs in the output "boxes", by the way, represent the glide rate controls associated with each output channel.

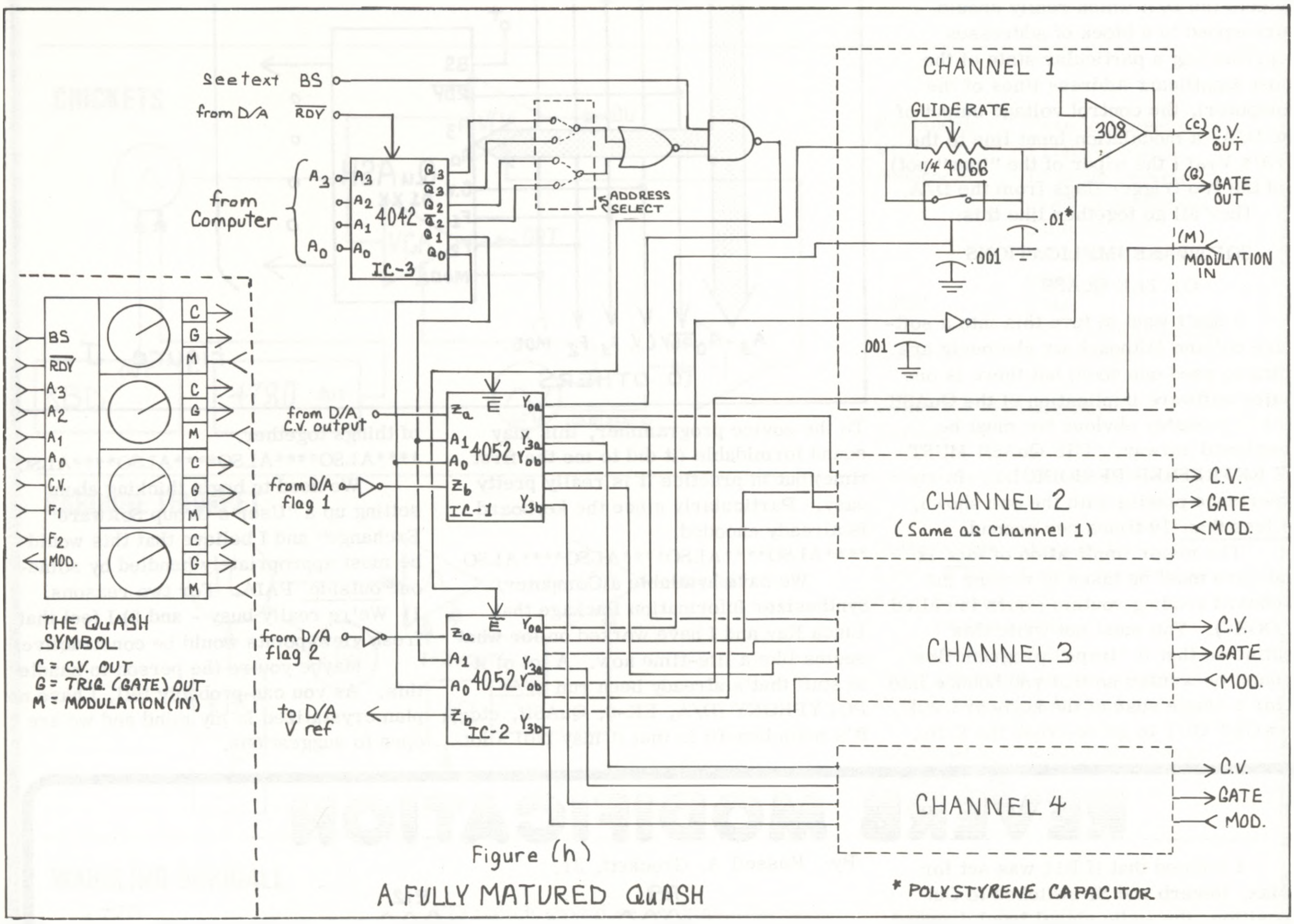


Figure (h)

A FULLY MATURED QuASH

NOTES:

*(1) Those of you who have been thinking about this stuff for a while will, of course, recognize the imminent demise of the ADSR. Providing Attack, Decay, Sustain and Release parameters is one of the easier tasks to turn over to the computer entirely. On the other hand, I've played with this some and can testify that varying the position of a knob is handier - in this case - than changing parameters in the memory of the machine. Some Hardware ADSRs mixed with some Software ADSRs seems a good compromise.

*(2) This off-hand statement is not meant to imply a wait in human terms (major fractions of a second), but rather a wait in machine terms - micro-seconds. You don't have to wait for a GLIDE to finish (for instance) before doing something else.

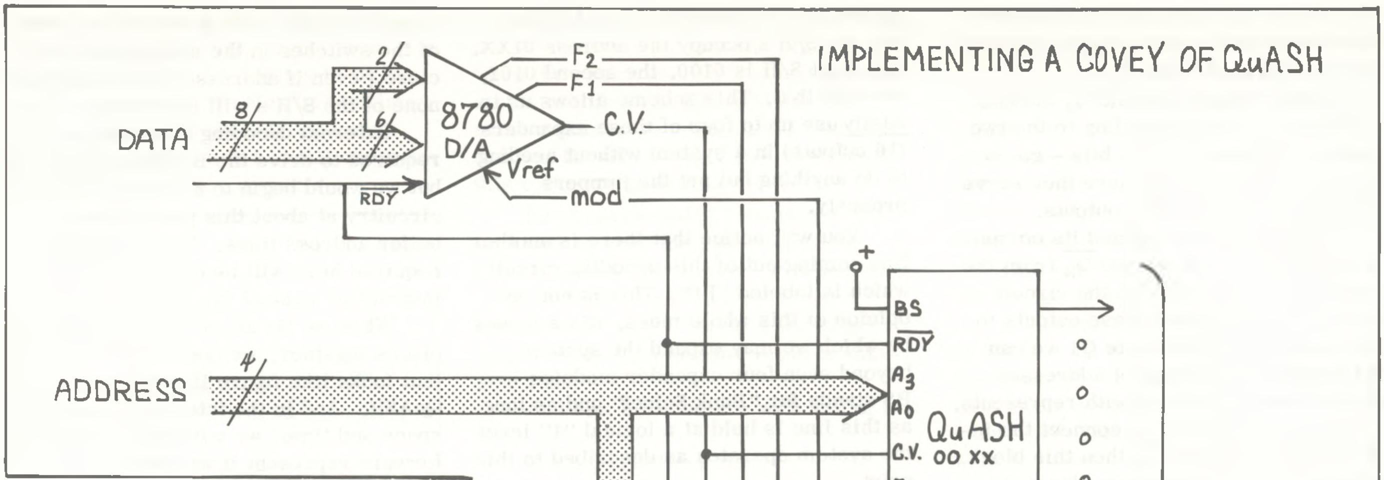
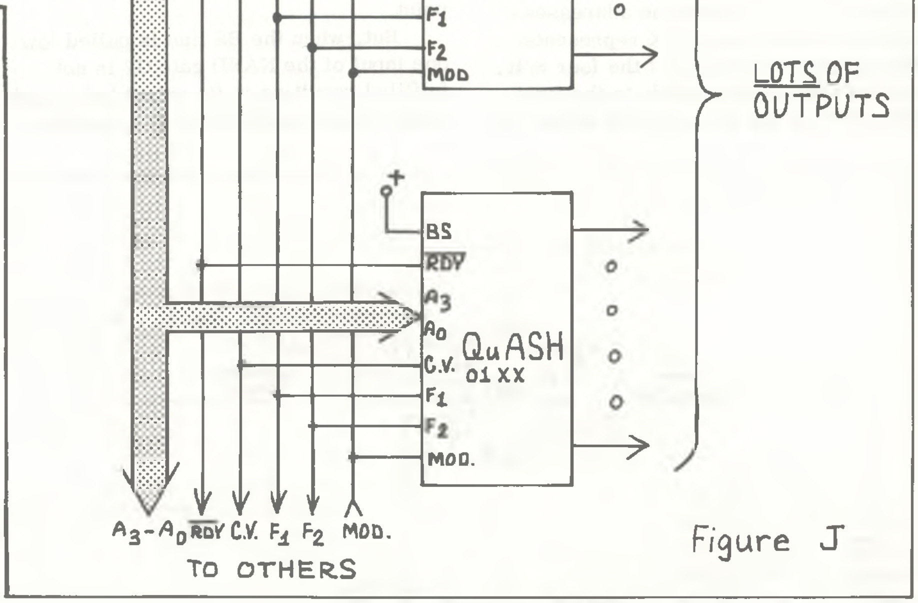



Figure J

If we're going to use more than one of these things, they all tie to a bus made up of four address bits (the least significant four from the computer probably), a $\overline{RDY}$ line (the same $\overline{RDY}$ line that selects the D/A which really should correspond to a block of addresses representing a particular state of the Most Significant Address lines of the computer), the control voltage output of the D/A, a modulation input line to the D/A's V_{ref} (the wiper of the "tune" pot), and the two trigger flags from the D/A.

They all go together like this:

SOFTWARE IMPLICATIONS OF THE QuASH

I don't want to turn this into a software column (although we obviously are going to need one soon) but there is one major software implication of the QuASH that is probably obvious but must be mentioned anyway. THE QuASH MUST BE REFRESHED PERIODICLY. In my experience playing with the one shown, no less than 10 times per second.

The major implication of this is that care must be taken in writing the keyboard reading routine (ours is called "LOOK"). You must not write this routine so that it "traps" program flow. It must be written so that you bounce into it for a single scan of the keyboard and then GET OUT to go re-fresh the S/Hs. To the novice programmer, this may sound formidable (it did to me the first time) but in practice it is really pretty easy. Particularly since the keyboard is already encoded.

****ALSO****ALSO****ALSO****ALSO

We have available a Computer/Synthesizer Information Package that Linda Kay and I have worked on for what seems like a life-time now. A lot of it is stuff that's already been run in POLYPHONY (D/A, EK-3, QuASH, etc.). It's main benefit is that it may pull a lot of things together.

****ALSO****ALSO****ALSO****ALSO

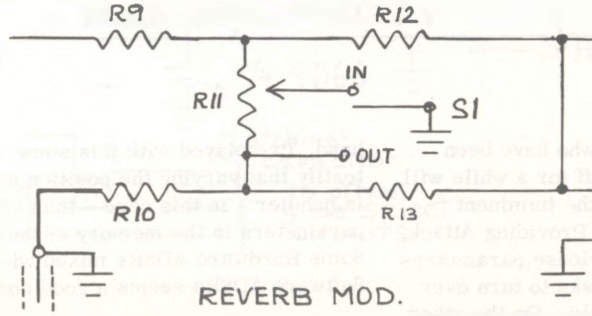
We need to begin thinking about setting up a "User's Group Software Exchange" and I believe that this would be most appropriately handled by someone outside PAIA. For two reasons. 1) We're really busy - and 2) I feel that fresh view-points would be constructive.

Maybe you're the person to handle this. As you can probably tell, I have no plan crystalized in my mind and we are open to suggestions.

REVERB MODIFICATION

By: Russell A. Grockett, Jr.

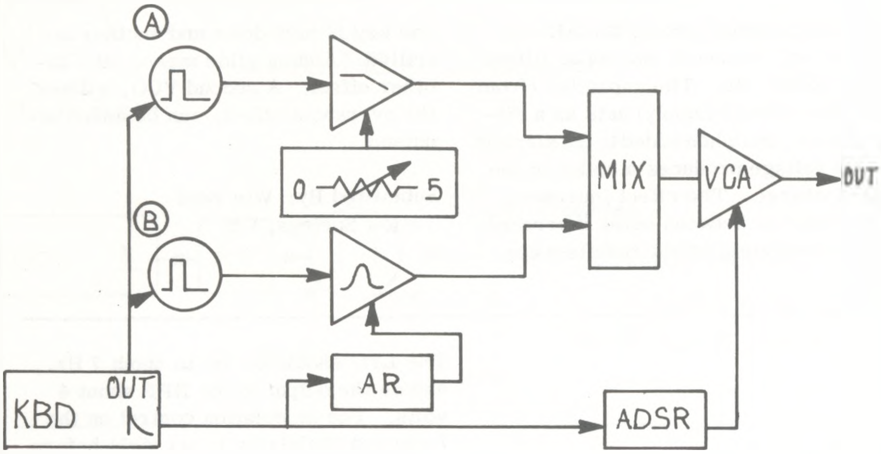
I noticed that if R11 was set for Max. Reverb and S1 switched to Out position, the audio signal level dropped considerably. This can be easily corrected by replacing S1 with a SPDT switch and connecting it as shown in the accompanying diagram. When S1 is in the IN poistion R11 works normally, when switched to OUT, the ground is lifted from R11 rendering it functionless and grounding the R10 - R13 junction thereby killing the reverb effects.



REVERB MOD.

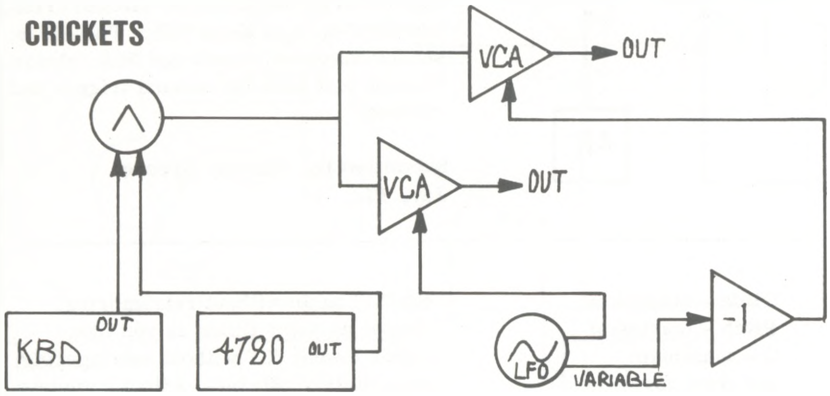
PATCHES

ELECTRONIC HARPSICHORD



VCO B tuned one octave lower than VCO A. Both VCO outputs are narrow width pulses.
Bias Supply - +5 volts.
Mixer - VCO B mixed at 25% of VCO A.
AR - Expand-on
Release-30%
Attack-minimum
ADSR - Attack-10%
Decay-20%
Sustain-50%
Release-100%
Submitted By: Allen Fairfield
Wakefield, MA

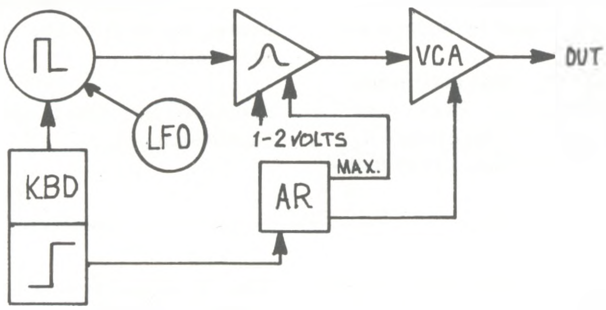
CRICKETS



Keyboard:	Tune to lowest octave.
VCO:	Triangle output.
Sequencer:	12 stage recycle clock at maximum Glide at 75% All stages tuned close together.
LFO:	3 Hz. 75% variable output
Inverter:	5 volt offset

Submitted by: Dale Naylor
Parma, OH

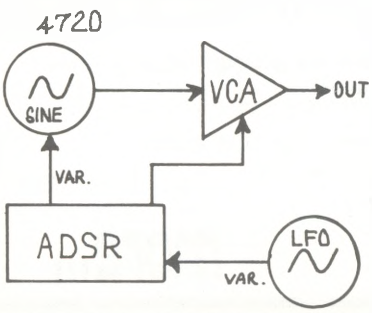
BAROQUE TRUMPET



Pulse Width - about 20%.
LFO - 9 Hz., Level - very slight.
AR - Expand - off
Attack - 40%
Decay - 20%
Use a generous amount of reverb.
Play keyboard in highest range.

Submitted By: Robert Matarazzo
Brooklyn, NY

WARBLING BIRDCALL



VCO pitch should be set high. Increase LFO output to trigger ADSR at proper rate. Set ADSR variable output to obtain proper frequency change. Work with one hand on the initial pitch knob and the other on the ADSR attack knob. Decreasing attack time gives a warble effect. Pressing the manual trigger stops sound well.

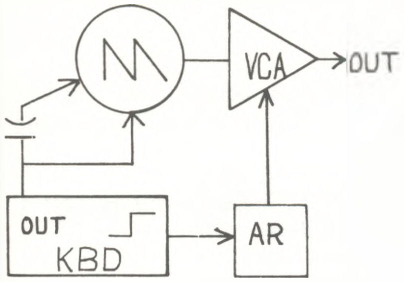
ADSR:	A - 15%
	D - 20%
	S - 0%
	R - 0%
LFO:	8 Hz.

Submitted By: B.R. Revor
Palos Heights, IL

more...

PATCHES

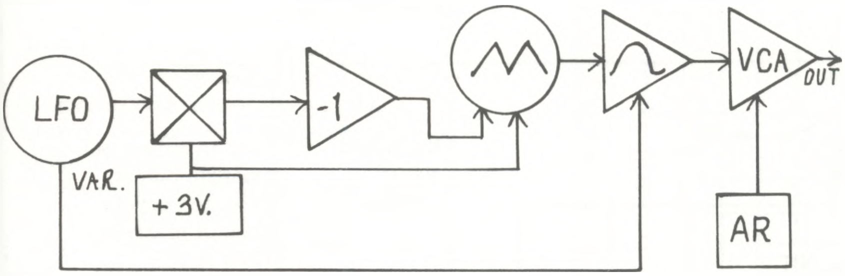
OVERSHOOT OSCILLATOR



Obviously a simple patch, the AR can be set to any standard envelope, filters can be added, etc. The capacitor (from the 2720-7 Power Supply) acts as a differentiator, and when added to the straight control voltage produces a spike on the normal change. The effect increases as the interval between notes increases, and an interesting effect results when one key is held down and another is trilled. Adding glide makes still another effect. A second VCO, without the overshoot effect, can be added at unison.

Submitted By: Win Bent
Yellow Springs, OH

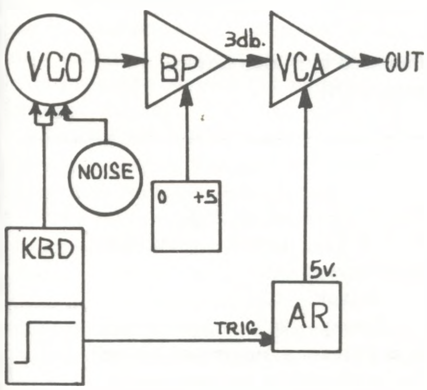
SEAGULL



The LFO should be set to about 7 Hz., and on the output to the BPF about 4 volts. The modulation control on the Balanced Modulator is set right before the LED comes on. The 5 volt offset is to the 5 volt setting. The bias voltage can be varied for different pitched cries. The BPF-Q is at about 75%. The AR is set for minimum attack and 50% release. To play just push the manual trigger and release.

Submitted By: Sammy Greene
Jay, FL

DRILL



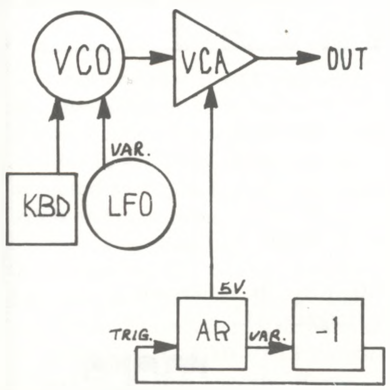
KBD:	Glide - maximum
	Pitch - maximum
BP:	Q - maximum
VCO:	any wave form
AR:	Expand - off
	Attack - 0%
	Release - 100%
Bias:	Set to resonance point of note played by highest key.

Press highest key, wait until pitch reaches highest point. Press next "C" down without retriggering AR, wait as long as you like, press highest "C" again without retriggering. Repeat as many times as you like. Press lowest "C" without retriggering, immediately release. Effect complete.

This can also be used as a buzz saw sound. Fiddle with the AR and glide and bias settings. If the glide is lessened, the drill can be played as an instrument.

Submitted By: Ken Keeler
Charleston, SC

REALLY AWFUL SOUND



VCO:	whatever wave sounds worst to you
KBD:	pitch and glide generally worst at maximum, any key is useable
LFO:	about 9 Hz., or whatever you hate. Output - 50%
AR:	Expand - On
	Attack - 30%
	Release - 5%
	Variable Output - 75 - 100%
INV:	5v. offset

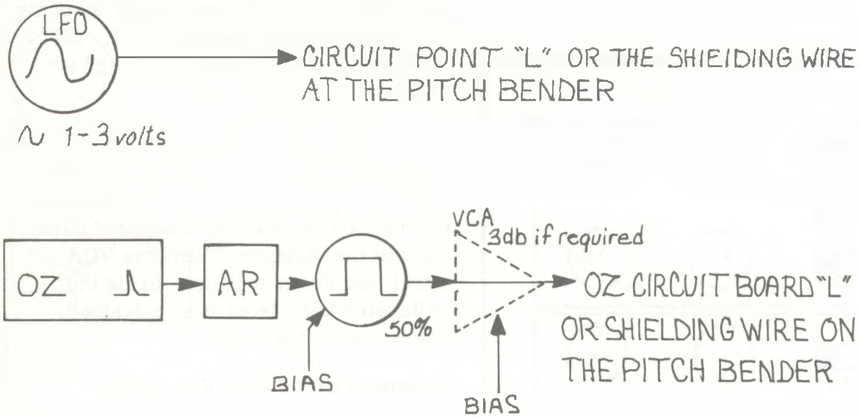
This is the worst sound I've found yet. Now it's not gross or disgusting, it's torturous, and it's all the better because it's automatic. If you like the sound that you get with the given settings, you should change them. This patch is great for getting people to leave you alone.

Submitted By: Ken Keeler
Charleston, SC

more...

PATCHES

ADDING FREQUENCY MODULATION TO THE OZ



If an amplified waveform from a VCO is used a variety of voltage controlled vibrato effects may be produced.

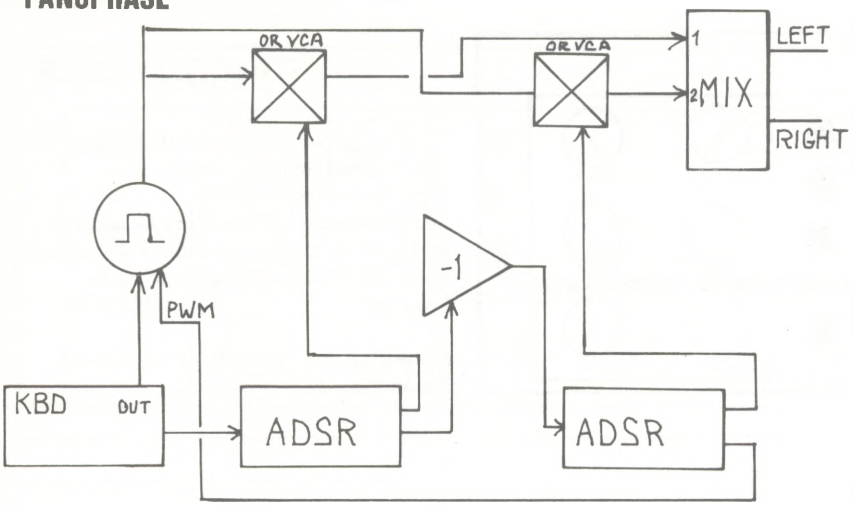
AR: Attack - minimum
Release - variable

Bias in enough voltage to VCO to produce desired speed of vibrato after decay. Amplifying waveform may be necessary.

By using the AR to control the VCA instead of the VCO you can have a fading vibrato. NOTE: Be sure to ground OZ to the synthesizer.

Submitted By: James Vicari
Dallas, TX

PANOPHASE



ADSR 1 & 2: A - 100%
D - 100%
S - 100%
R - 100%

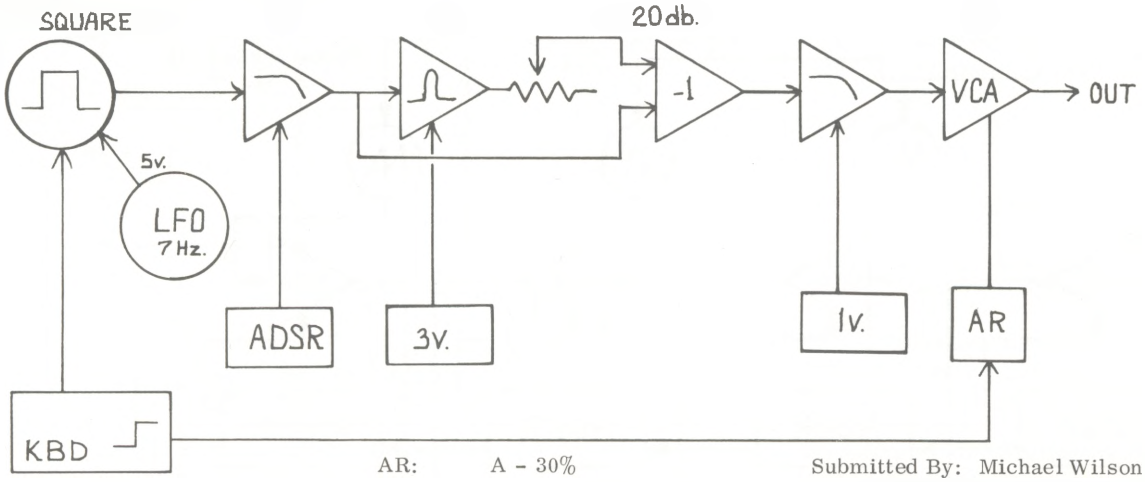
Mixer:
Channel 1 - full left
Channel 2 - full right

VCO: Initial P.W. - CCW
Frequency - whatever

Inverter: 5 volt offset

Submitted By: Miles Flewitt
Minto, Manitoba, Canada

BASSOON



ADSR: A - 20%
D - 40%
S - 50%
R - 60%

AR: A - 30%
R - 5%
Expand - off

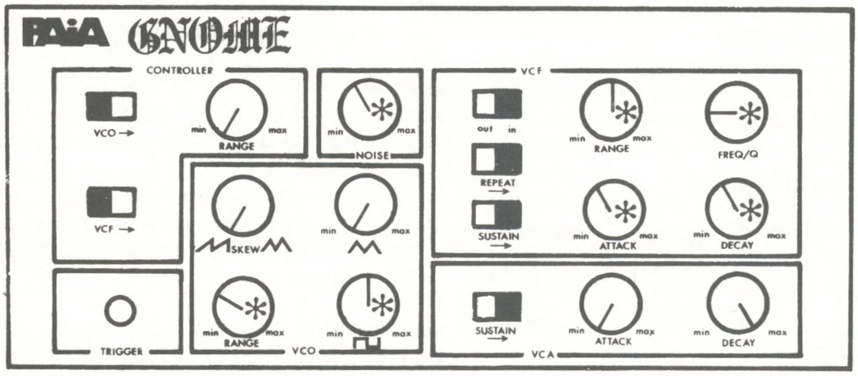
Control voltage to notch filter may require adjustment to produce reedy sound.

Submitted By: Michael Wilson
Ottawa, Ontario, Canada

more...

PATCHES

HELICOPTER

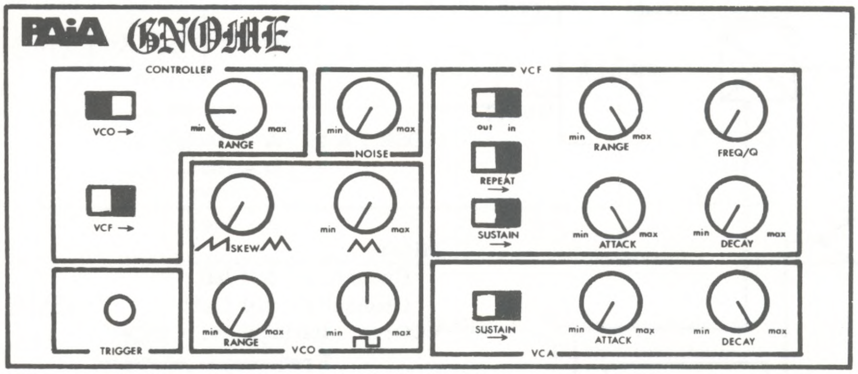


Set VCF controls for the whirling blades. Mix NOISE and VCO for proper amounts of blade noise and engine sound. Set VCO Range for subsonic "putting" of engine.

By varying NOISE, VCO, Q/FREQ, and VCF Range, the engine starts, idles, picks up speed, and takes off! Release the TRIGGER, and the "chopper" flies off into the distance. Reverse VCA attack and decay settings, bring the helicopter back from it's "mission", land it, and shut down.

Submitted By: Terry Fitzpatrick
Erlanger, KY

SUBTERRAINEAN DRIP

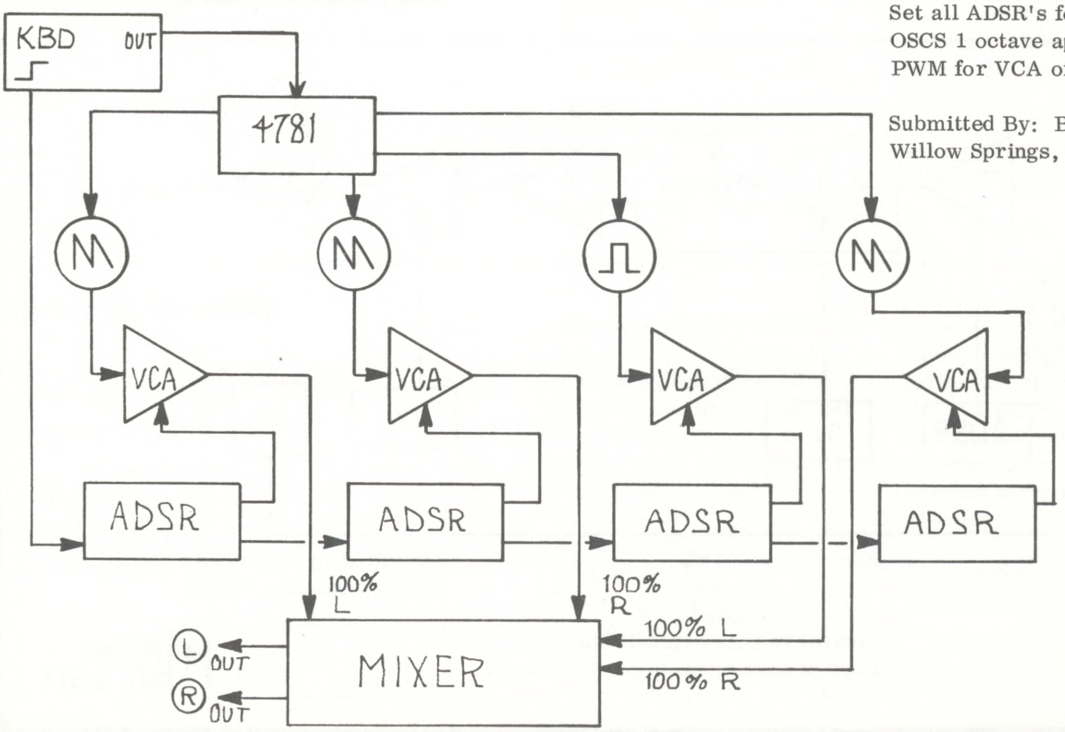


Very Important: Adjust VCO Range for low pitched dripping sound.

The frequency of the drips is adjusted by the VCF attack (rotating the control counter-clockwise makes it more frequent.

Submitted By: Johnathan R. Merril
Rockville, MD

SEQUENTIAL HARMONIC ENVELOPES



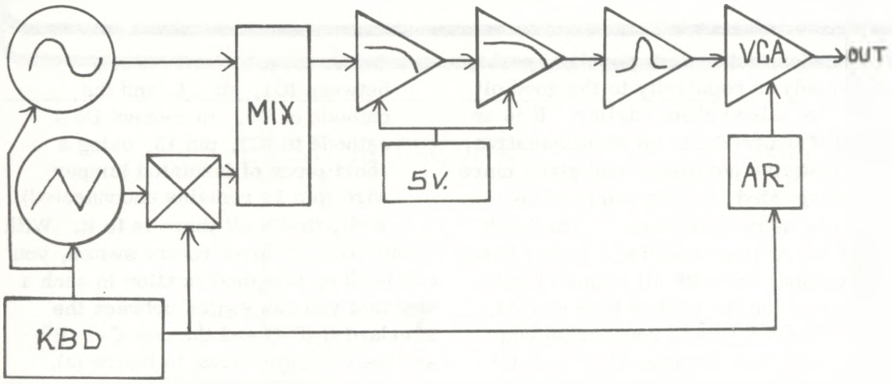
Set all ADSR's for full attack. Offest OSCS 1 octave apart. Try substituting PWM for VCA or VCF for VCA.

Submitted By: Bruce Wojak
Willow Springs, IL

more...

PATCHES

MARIMBA



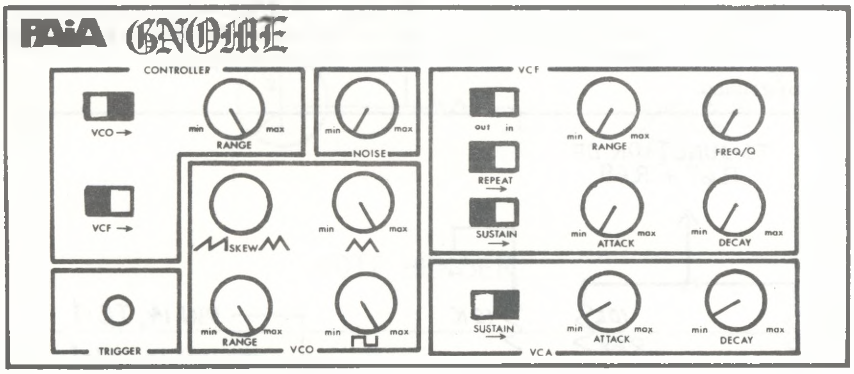
ADSR: A - 5%
D - 5%
S - 0
R - 0

AR: A - 5%
R - 50%

The triangle waveform should be adjusted to be an octave and a fifth above the sine.

Submitted By: Michael Wilson
Ottawa, Ontario, Canada

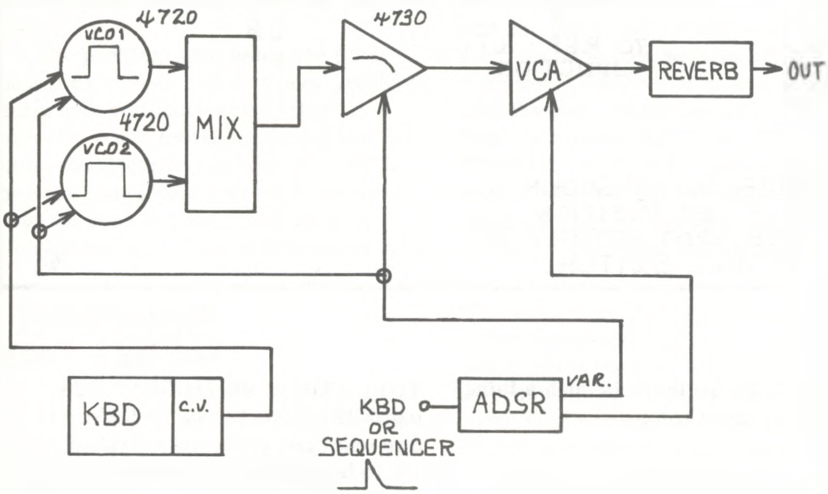
HUMAN ALARM



To make the "Human Alarm" operate, you must have two people. One person must take the probe and touch it to his or her palm. (Make sure you apply a lot of pressure to insure contact.) Have the second person but his or her finger on the up-most part of the controller strip. Also, have this person push and hold down the Trigger. Now, when the first person touches the other person's skin, the alarm will go off. Notice that different parts of the body make different sounds. (Hint: The least distance the voltage control has to pass through the bodies, the higher the pitch).

Submitted By: Bill DePatie
Newington, CT

SMALL DOG OR LAUGHING HYENA OR CHEEPING BIRD OR ?



VCO 1: Frequency - *
Pulse Width - 50%

VCO 2: Frequency - *
Pulse Width - minimum

MIX: Both Equal

VCF: Frequency - 85%
Q - 60%

ADSR: A - 25%
D - 25%
S - 25%
R - minimum
Variable Output - approx. 25%

KBD: Pitch - minimum
Reverb - slight

*VCO frequencies should be tuned for whatever "animal" you want. High frequency for birds, medium for dogs, sweep VCO by hand for hyena. Almost any tuning of VCO's produces some kind of animal (unison; a few beats apart, etc.) Keyboard can be used to change pitch of "bark". Use sequencer for repeating bark (especially hyena!).

Submitted By: Bob Yannes
Media, PA

more next issue...

HOW TO MAKE AN EGG MOD

BY Craig Anderton

Most of you are probably familiar with PAIA's Chord Egg, a neat little device that both randomly changes chords and also randomly fades the individual notes of the chord selected in and out. The chords selected for the EGG's progression are C, F, and G major --- a pretty good choice, when you consider that chord progressions based on these three chords are found in music from Bach, to blues, to country, to "Louie, Louie". It has a nice major feel, and has a generally cheerful effect on listeners.

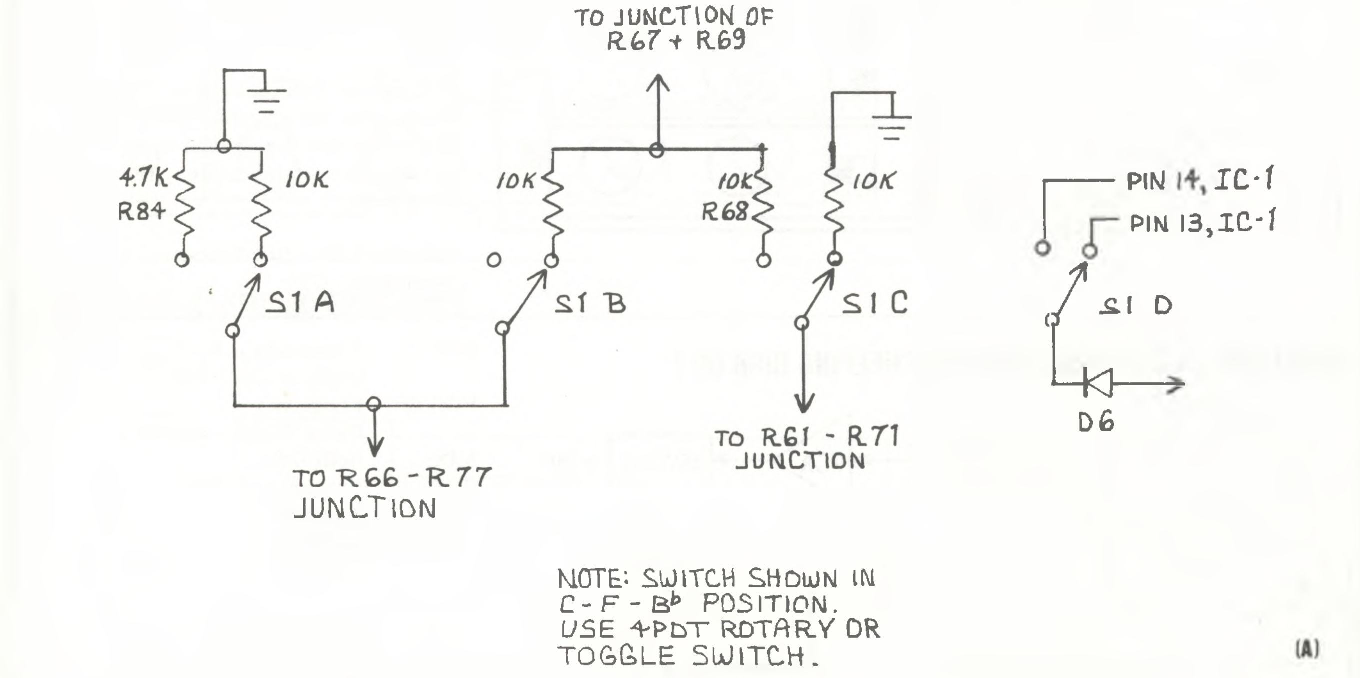
However, there are other progressions. You certainly wouldn't want to use minor chords (take my word for it; listen to a minor chord progression repeated endlessly and you'll wonder what got you feeling so depressed!). But by modifying your EGG to give a B flat major chord instead of a G, you come up with a chord progression that is very popular, perhaps second only in popularity to the good ol' C-F-G we talked about earlier. It is an excellent progression for contemplative and meditative purposes, and gives more of a "suspended" feeling whereas the C-F-G is more "grounded". The trick is that we're implementing a minor based progression, but with all major chords. This gives you the best of both worlds.

So if you'd like to reprogram your EGG to alternate between C, F and B^b major, here are the steps to take:

1. Change R84 from 4.7K to 10K.
2. Remove R68.
3. Add a 10K resistor from the junction of R67 - R69 to the junction of R66 - R84 - R77.
4. Add another 10K resistor from the junction of R71 - R61 - R76 to ground.
5. Break the p.c. board trace between IC1, pin 14, and the cathode of D6. Reconnect D6's cathode to IC1, pin 13, using a short piece of insulated jumper wire (pin 14 remains unconnected).

Well, that's all there is to it. With a four pole, 2 throw rotary switch, you can hook up this modification in such a way that you can switch between the standard C-F-G and the new C-F-B^b; see the diagram shown in figure (a).

One final hint: It seems to me that the C-F-G progression is the ideal daytime environment, and the C-F-B^b progression is more suited to night time. If you come up with any interesting thoughts about the connection between music and mood, I'd like to hear about it. Write me c/o Polyphony.



(A)

WHAT THE COMPUTER DOES... AN INTRODUCTION

.........continued from page 7

if you haven't heard of them yet, you will.

They are:

Commodore's PET (personal electronic transactor) which looks at this point like it will sell in the $600.00 range. Certainly you're all familiar with Commodore - they're an old-line (if there is such a thing) calculator company.

and

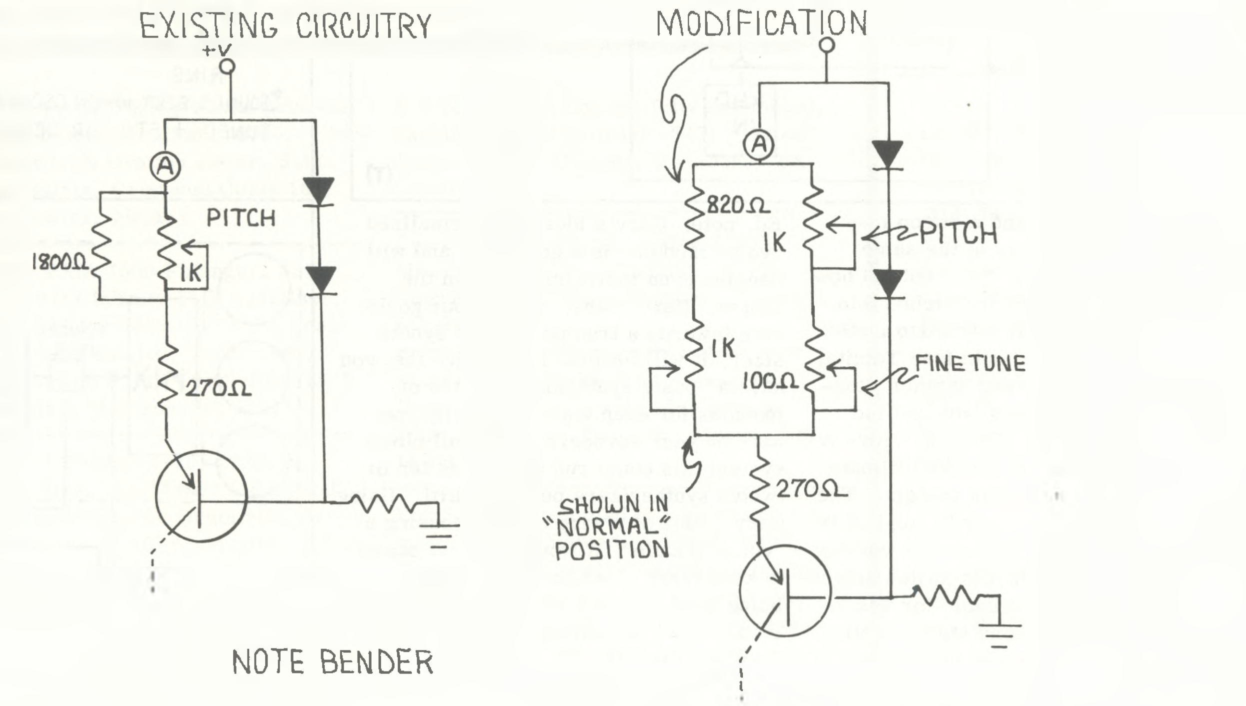
Apple Computer Company's APPLE II

We like the APPLE II machine a lot and probably a single glance will tell you why. It not only looks nice and can grow up to be a VERY LARGE system, but it has all the bells and whistles including FULL-COLOR VIDEO GRAPHICS capabilities (vectored, no less). I own one (one of the very first, I'm led to believe) and I can tell you - it's a very impressive system. You will be seeing a system available from PAIA (by October, we hope) based on this true "appliance" computer.

The PAIA/APPLE system is not yet fully configured, but target price is approximately $2,500.

SAMPLE & HOLD MODIFICATION
Provides Note Bender

By Barry Nesmith



A modification of the Sample and Hold circuit in the Keyboard Controller of my synthesizer provides a note bender. This effect is used extensively by Jan Hammer (with Jeff Beck), Patrick Moraz (Yes) and Roger Powell (Utopia). It is used the same way as a guitarist bends a note.

The problem with using the pitch control as a bender is that once you have bent a note you must again "find" the correctly tuned location. Using the fine tuning modification, Polyphony 3/76, does not completely solve the problem. It can be used to bend notes down in pitch but not up (zero resistance is the least you can go). This means that you would have to compensate for a "full on" position on the fine tune control by raising the coarse adjust. This greatly diminishes the range of the keyboard.

To get around this I replaced the 1800 ohm resistor that is in parallel with the pitch control with a 1000 ohm slide pot in series with an 820 ohm resistor. Together they are in parallel with the pot. When the pot is "full on" there is around 1800 ohm (close enough). However, as the pot is moved toward zero resistance the series combination approaches 820 ohms thus allowing more current. I've found that when using the keyboard on the middle pitch range (16') that it allows a bend of about 2 half steps which is perfect for most applications. It also is fun to use.

Raising the pitch control to the next octave diminishes the range of the bender since the pitch control itself is approaching zero resistance. But on the 16' range it is a great effect.

A rotary pot would work but I used a slide pot since it's easier to "feel the bend" and return to normal after the bend. One disadvantage to a slide pot, though, is cutting the panel.

LETTERS

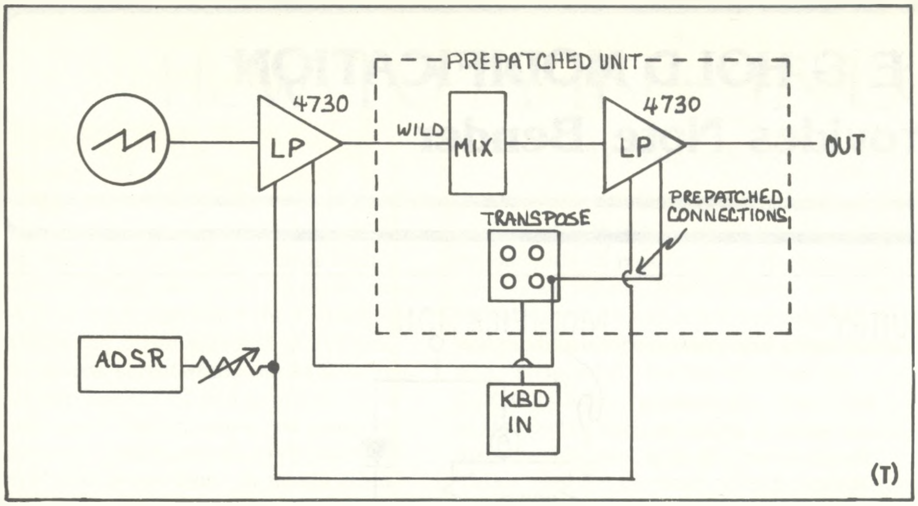
........continued from page 11

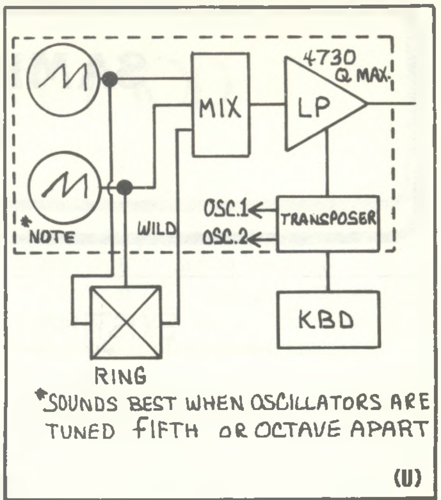
O.K., so far pretty basic, but the point is to use these classifications in setting up a system. What happens in most sounds that are created is that a sound source gets controlled and processed (the processors may also be controlled). In signal flow charts (patch diagrams) signal flow is from left to right and control is from bottom to top. The implications for system design are obvious. Wherever possible controllers should be below sound sources and processors, while sound sources should be to the left of processors. This, of course, assumes more than one row of modules. If there is only one row as in the basic 2720 packages it should be set up left to right - controllers, sound sources, processors.

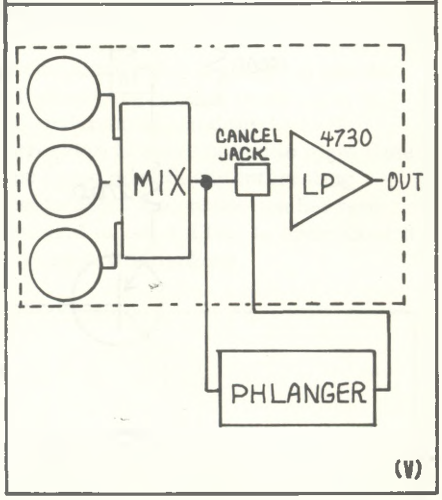
Positions within these groupings may vary, but logic should be applied to the decision making process. Within the controller group the control oscillators should be placed nearest the sound sources because of the noise generator, mentioned earlier (incidentally, I strongly disagree with the reasoning in the manual that the control oscillator isn't "likely to be duplicated within any single system".) Some of the most interesting effects can be achieved by having multiple oscillators controlling a VCO or a VCF or triggering an Envelope Generator. The variety of effects available by changing the relative frequencies and amplitude of the Control Oscillators is endless. On the other hand I can see no reason to ever duplicate a noise source (except as a possible spare

continued on page 30 ....

continued from page 14







the transposer knob, and a sweep may be applied to both filters at the same time. (see figure (t).) Take note of how the keyboard voltage is prepatched into one filter and externally patched to another.

Yet another patch, alla Isao Tomita, is shown in figure (u), and another illustrating the filter by-pass jack and the switching input jack is shown in figure (v).

I hope that you can see the ultimate ease and usefulness of this system. The ideas presented herein may be applied to other units as well. I'm already working on a unit with ALL controls switch selectable, yet externally available for use in both live performance and studio applications. Watch for details on this unit in the future.

Ed. note: Gary's idea of a normalized "voice module" is a good one, and will become even more important in the future. Here's why. When your goals turn towards a true polyphonic synthesizer, it will become imperative that you have a "basic synthesizer" worth of modules for each voice capability you have on your keyboard. In a full blown system this could run as high as ten or twelve synthesizers per keyboard. Using Gary's idea of a wing cabinet housing a "voice module", it would be a lot easier to keep track of which modules are being used for each keyboard voice --- let alone all the savings in patch cords. Food for thought.

-M.J.-

LETTERS:

.... continued from page 29

in case one fails) since the only way to vary the noise is through processing, and the signal from one source can be split to be processed as many ways as desired. Perhaps when the 4700 series control oscillator arrives this problem will be rectified; but enough of this. There should be nothing to decide on concerning arrangement of VCO's, unless you have 2720 and 4700 series modules mixed, in which case decisions would be esthetic, based on front panel finish. With processors, the Mixer should probably be the last module, although if you have multiple VCO's you may want one between them and the other processors. The VCA should probably be next to last. Filters seem to work best as one of the first processors.

In my own system the original keyboard/case unit is devoted to controllers: a Sequencer, two control oscillators and a Function Generator plus the keyboard and power supply. Atop that is a Road cabinet with Six VCO's (admittedly six is a luxury - but I couldn't resist the 4720's when they came out). To their right is a Wing cabinet with Band-pass, Low pass and Multiple function VCF's, a VCA and a Sine converter/PWM. Still awaiting adequate financial support are a mixer and additional processors, but you get the idea. I'm not saying that my design is the only way to go. Each system should reflect the owner's unique intended use, but I do think planning module placement to represent anticipated flow is a concept that should be used by anyone with any modular type synthesizer.

I'm glad that I finally got around to writing to you folks about my ideas. I've sure appreciated your magazine and the chance to share other users' thoughts. Keep up the good work.

Synthetically,
Joseph Adams

Musician vs. Machine?

Dear Polyphony;

It doesn't take much foresight or imagination to see one potential application of "Computer Drums". With a lot of memory and an easily programmable microprocessor based system (such as SWTP 6800), or even a cascading of the new PAIA Programmable Drum Set (and they're inexpensive enough to do just this) a keyboard musician who kicks bass (on a synthesizer of course) and sings as well as plays leads could book himself some nice gigs. Oh, I'm not talking about some cafeteria where some organist is pumping out "Tea for Two" with a tango beat. I'm referring to the very real potential for a multiple keyboardist to program a whole set of complicated jazz, rock or even disco and have the machine "Go To" whatever tune, whenever, with complete bridges, intro's, jams, etc. Disco's watch out! The live performance element that is impossible to replace could be had for a fraction of the cost. And, how interesting the whole thing would be!

One question, though - how are the unions going to take this? A lot of my friends are drummers. Will it put them out of work? The answer is, I think, an emphatic NO. String synthesizers aren't putting string players out of business... no machine will ever approach the subtleties of a good human performance. But, appreciated for what it is, an interesting application of technology to the arts, I know that I for one, find it fascinating.

Peace,
Jim Riter

continued on page 31 ....

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 2720R, works perfectly according to specifications, $300. Electric hollow body Ovation guitar, $100. Nataneal Salais, Cristobal Colon 166, Nogales Sonora, Mexico.

For Sale: PAIA 1500 Phlanger, $50. and PAIA 9711 Infinity Plus sustainer, $17.50. Will take $65. for both, postpaid USA. Assembled, never used, factory checked. Phil Corriveau, 2873 Kingston Dr., Madison, WI 53713

For Sale or Trade: 4780, 2720-14. Both work fine. Will trade for a good 1500 Phlanger or 3750 Drummer, or whatever trade or offer you have. Alvin Kawamoto, 940 17th Ave., Honolulu, HI 96816

For Sale: (2) PAIA 1500 Phlangers. Fully assembled, factory tested. $90. each. Mark Chelsvig, c/o Steve Seronko, RR #1, Story City, IA 50248, or leave message at (515)292-6023.

For Sale: PAIA 2720R and 1702 Synthespin. Fully assembled and operational. $400. for both. Charles Kimble, RR 1, Box 73, Sedona, AZ 86336 (602)282-3128.

Assembly - If you want to avoid assembling, calibrating and scoping (even tuning) and just want to start synthesizing, we're for you. We will build your kits for $5 per module and will calibrate kits for $1 and tune your keyboard for $1. We also guarantee our work for one year. Keyboards and sequencers are slightly more for assembly. Spock Enterprises, 7236 Whitson Dr., Springfield, VA 22153

For Sale: PAIA Gnome, fully assembled, excellent working order, $45. Gerald Jackson, 2501 Halkeis, Pasadena, TX, 77502

For Sale: Arp Omni, asking $1895. Minimoog, $1395. Clavinet, $870. Doug Keithley, 3411 S. Sundown, Spokane, WA 99206.

For Sale: (2) PAIA 1500 Phlangers. Assembled, $60. each, plus postage. John Bloom, 114 Cook St., Ithaca, NY 14850

For Sale: Assembled and working PAIA modules in homemade cabinet. VCO, Power supply, control oscillator/noise source, two bandpass VCF, two VCA, two ADSR, two AR, envelope follower, stereo mixer, and an extra power supply panel (with multiples, attenuator, etc.) One ADSR and control oscillator needs work. Best offer over $200. D. Genovese, 2130 Metcalf St., Honolulu, HI 96822

For Sale or Trade: Fender Palomino (acoustic) guitar. Excellent condition, rosewood back and sides. Also, old Gibson 20 watt tube type amp. Prefer to trade either or both for OZ, 2720 or 4700 modules, keyboards, etc. Or will sell to best offer. Terry Fitzpatrick, 92 Sherwood Dr., Independence, KY 41051

For Sale: (2) PAIA 2720-7, $15. each. (2) 4770, $8. each. (1) 1702 Synthespin, $15. All units assembled and fully functional. John Deaton, 401 Main, Parkville, MO 64152 Phone: (816)741-8761

Local Happenings

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

Lon W. O'Bannon III
11895 North Ranch Dr.
Florrisant, MO 63033
(314) 741-1664

Guy Hanks
8274 Gladewood Dr.
Baton Rouge, LA 70806

Robert Laughton
2851 Oleander
Merced, CA 95340
(209) 722-3220

Brian Rich
7331 Toulouse Dr., Apt. 3
Huntington Beach, CA 92647

Val Wyszynski
Box 188
Las Cruces, NM 88001

Karsten Lyngholm, E-Division
USS Durham LKA 114
FPO San Francisco, CA 96601

Kenneth Winograd
Seven Greenbriar Dr., Apt. 306
N. Reading, MA 01864

Daniel H. Ade
5821 Newberry Rd.
Durand, MI 48429

David Bleser
503 Anastasia, #8
Coral Gables, FL 33134

LETTERS ..... continued

Music Languages Exist

To the Editor;

In Polyphony 4/76 Mark Lutton was dreaming of a "music compiler" to enable a composer to feed a composition to a computer. Actually, several such music languages exist. Among them is a "Music System for the PDP-10" (not exactly a microcomputer!) by P. R. Samson and R. Clements of MIT project MAC and Digital Equipment Corporation. The system of notation used is described in a write-up available to members of the Digital Equipment Computer Users Society, for zilch the last time I checked, and possibly to ordinary mortals as well. It's worth a try. DECUS write-up 10-9; the address is DECUS, Maynard, MA 01754.

Sincerely,
Richard S. Holmes

Each day's mail brings more favorable comments on the

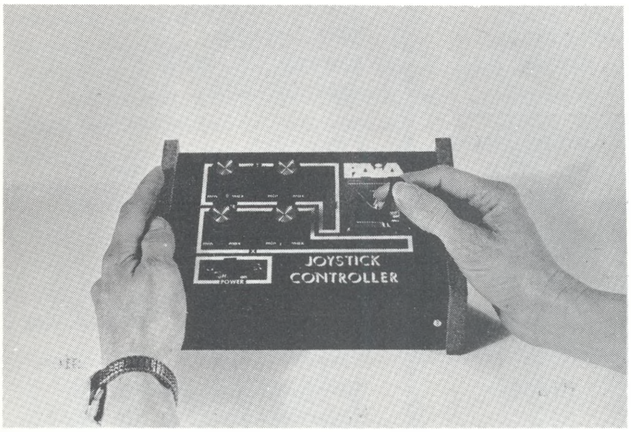
PROGRAMMABLE DRUM SET

While most electronic rhythm units offer only a limited choice of pre-determined rhythm patterns, the PAIA Programmable Drum Set allows the user to tailor pattern, time signature and drum sounds to each application. Among the unique features provided by the unit are touch sensitive electronic controls and the provision for an independently structured bridge rhythm.

Battery powered, the drum set includes a "memory save" switch which provides a lowered "keep-alive" voltage to the drum set's 256 byte memory.

Kit includes circuit board, all parts, completely finished case and well-illustrated, setp-by-step instructions.

#3750 PROGRAMMABLE DRUM SET KIT .....$79.95..(+ $3.00 postage)



WHERE CAN YOU FIND JOY for less than $40?

From PAIA of course!

What a shame that a joy stick is the missing control element on most American synthesizers.

A joy stick gives you total, infinitely variable control of two parameters simultaneously.

Our Joy Stick Controller works in either of two modes: As a CONTROL VOLTAGE SOURCE with your exponential front-end equipment. Use with any processing system/device that has provision for voltage control. As a CONTROLLABLE MULTIPLIER for linear response elements. All this and a touch-activated "gate" trigger that activates with the slightest touch of the JOY STICK's metal shaft. No buttons to search for. And at a typical "down-to-earth" PAIA price.

#4783 JOY STICK CONTROLLER KIT .....$39.95..(+$3.00 postage)

AVAILABLE NOW FROM PAIA 1020 W. WILSHIRE BLVD. OKLAHOMA CITY, OK 73116

cut out with scissors

MOVING?

When writing to change the address on your POLYPHONY subscription it is important that you enclose the mailing label. Our computer cannot locate your name on the subscription list without it.

ATTACH OLD LABEL WHERE INDICATED AND PRINT YOUR NEW ADDRESS IN THE SPACE PROVIDED.

Mail to: PAIA/Polyphony
P. O. Box 14359
Oklahoma City, OK 73114

★ use tape or glue to affix old label here ★

NEW ADDRESS HERE

Name ____________________

Address ____________________

City________ State________ Zip______

See Us In ATLANTIC CITY at PERSONAL COMPUTING 77
August 27 & 28
at the Shelburne Hotel
(on the Boardwalk)

Marvin, John and the crew from PAIA will be on hand to demonstrate the new line of Computer/Synthesizer Equipment and Recording artist/synthesist, Larry Fast will be present both days to lend advise on systems and applications.

Of course there's more to see there than just PAIA. There'll be two special seminar sessions relating computers to music applications plus a whole range of technology to absorb. It's all shaping up to be a real educational experience. Don't miss Personal Computing 77 and be sure not to miss PAIA at booth 408.