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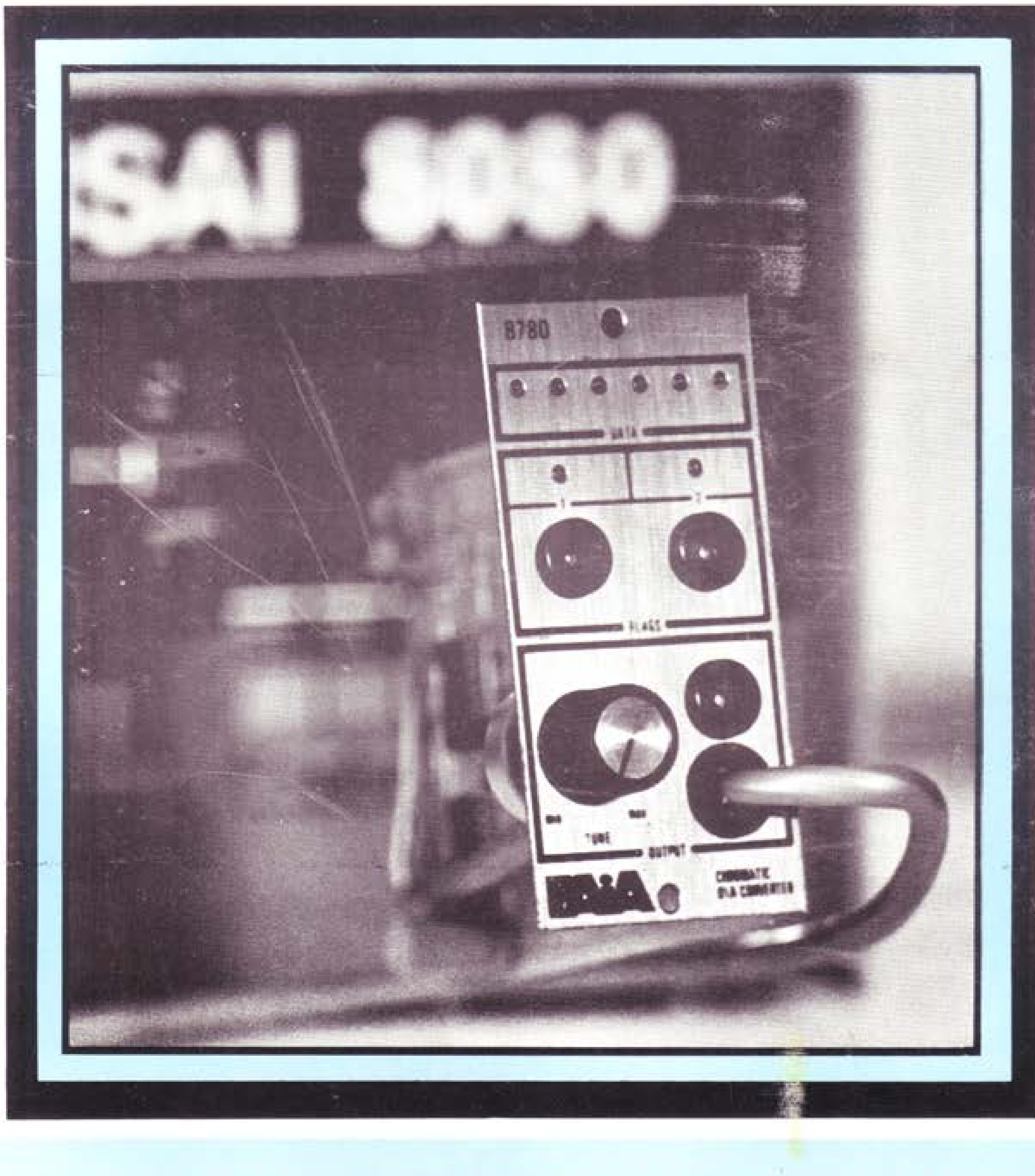
POLYPHONY 4/76



AN EQUALLY TEMPERED D/A CONVERTER
INTERFACING EXTERNAL SIGNALS TO THE GNOME
PROJECTS·PATCHES·MUSIC NOTATION

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Write for Polyphony

..... It's not something that you'll get rich at; but, nevertheless, beginning with the material published in this issue 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 writtem permission from the publisher.

ERRATA xxx

Most publications can only correct errors (and we all make them) in following issues. You'll find some corrections for the preceding issue in this issue. But, before you delve into this issue you might like to know that we have discovered some errors in this issue after the pages were printed. They are as follows:

Page 19, Figure 2... the equation should be:

$$2^{1/12}, 2^{2/12}, 2^{4/12}, 2^{8/12}, \ldots 2^{2^n/12}$$

Page 21, Figure 9 ....

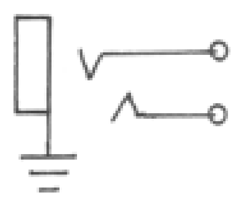
CA3401 will not work for IC5. It MUST be 4136 quad op-amp.

Page 14, Figure B .....

The point labeled GNOME INPUT... should be: GNOME OUTPUT

Page 14, Figure C ....

The added Trigger Jack should appear as shown below:



Page 16 .... A gremlin has struck, throwing in an extra line in the copy for the 8782 ENCODED KEYBOARD that reads "digital am" This is only in some issues others have a blank line. Has the gremlin struck your copy? Save it as a collector's item!

Also on page 16.... the weight of the Encoded keyboard is marked with an asterisk (*). The asterisk indicates that this item is shipped by FREIGHT, NOT that the offer on this unit expires on April 15th as does the free articulator with the purchase of a Pygmy Amp.

Editorial

Wow! Has it been a year already? We've covered a lot of things this year in Polyphony, but it seems like we've only scratched the surface. Electronic music is growing so fast -- not only in technology, but in popularity, application, and "hands-on" experience by the consumer. In trying to keep up with all this (and trying to keep you informed about it), you will notice a major change in this issue of Polyphony. Namely, a larger physical format. After that last issue, we realized that we couldn't get too much bigger before we wouldn't be able to print or staple the thing. So we hope the larger format will allow us to get more info to you in the most economical way possible.

Subscription renewals will be sent to you on an individual basis whenever you are due for renewal. Note that like everything else our price has gone up this year. I think that everyone will agree that last year you got a lot of information for our $2 subscription fee In fact, our prices for this year would even be a bargain for last years issues, but remember that we are expanding, and I think you will find your new subscription well worth it.

We've had a lot of people urging us to go to bi-monthly or monthly publication. Well, thanks for the compliment, but we have to run PAIA too, you know. If we didn't, we wouldn't have any new products or research to tell you about in Polyphony. (Isn't life a vicious circle?) If you do want us to publish more often, let me make a STRONG suggestion. Polyphony was originally conceived as an intercommunication media for you. We want you to share with us what you are doing with patches, recording, modifications, designs. In turn, we will share your ideas with others. Now, don't get me wrong. We aren't trying to get you guys to do all the work. We at PAIA will continue to write as many articles as we currently do; but if you send us more information, we can get an issue together faster and print 'em up more often. See? Out of our 2000 plus subscribers, we have a couple of dozen that send us information on a regular basis. Additionally, there were about a hundred or so that wrote us once, or maybe twice during the last year. Now, I know that there are more electro-maniacs out there than that!! But despite all this raving, we may be able to sneak out an extra issue this year. (Assuming we can get back on schedule. Keep your fingers crossed). What I have in mind is possibly an issue to "empty our files" of small tidbits you've sent us. These

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Volume: 2 No. 4 1976

On the cover: PAIA's new Equally tempered Digital to Analog Converter Module floats in front of an IMSAI 8080 Mini-Computer. But you don't need a Computer to use the D/A Converter. See the story on page 18 and Lab Notes, page 23.

SEE US AT THE FAIRE!

We'll be there to show all those West Coast Computer Freaks how to interface any computer to our electronic music synthesis equipment so they can program anything from simple tunes to full orchestrations ---- And, if all goes well, we'll be showing PAIA's own micro-computer prototype, designed for but not limited to electronic music applications (the price will knock you over).

We'll also have our full line of Electronic music devices there to demo so don't miss your chance to talk to Marvin and John and receive your personally conducted demonstration of PAIA equipment. Get the low down on the future of Electronic Music.

See us at the Faire.

Editorial

.... continued from page 3

would be things I was saving in hopes of working into a larger article, or comments that didn't seem to fit in with what we were trying to accomplish at the time, and so on. Perhaps we can just type them all up, print them and get them out to you so you could see if there was anything you might be able to use. What do you think? Would you be interested in one of these "free for all" issues?

I mentioned earlier about getting back on schedule. What can I say except "Sorry about the delays". I mean that" Prior to the last issue, our printing and production department was down while we were moving it to a larger more efficient location. Now, with this issue we are working on the changes to a larger format. Hopefully, we can go full steam on pumping out the issues for 1977. So bear with us.

Thank you all for your support in 1976, I hope to see you all again this year.

- Marvin Jones, Editor. -

LETTERS:

ECONOMY vs. QUALITY

To PAIA (especially the users),

From the way PAIA has appeared over the last few years, it looks as if they really are trying devilishly hard to produce quality at minimal cost. It is often the case easier to design and produce electronic equipment using pre-established "algorithms" for various circuits, than to try and achieve like results using less elaborate methods. The people at PAIA demonstrated this point with their VCO designs lately. VCO's in my opinion have to be one of the most critical components in a system. Tractional deviations from S/H circuits and VCO's can be readily heard by the average musician/purist.

But somehow I get the feeling that PAIA is playing the economy route too well. For people like myself who perform mostly in real time, little things like 3-octave keyboards can be irritating. I feel a larger keyboard, say 5 octaves, could eliminate reaching for the pitch knob for drastic scale changes. Polytonic keyboards would be interesting enough, but their total impact could be realized through a larger keyboard. By now, most of us have become comfortable tracking several oscillators and it would be ideal if this ability could still be incorporated into the polytonic system (rather than having two separate keyboards). This is where economy and versatility come to battle. On the one hand you have the people able to produce these electronic wonders, and on the other one users/buyers who may or may not be offended by the price of the final product. This applies not only to keyboards but for everything else as well. I feel that users shouldn't swoon when they see prices on individual kits that would normally accompany entire package systems.

Perhaps if enough users get demanding enough, PAIA will have no choice than to satisfy these perfectionist-related whims of incomparable versatility and quality.

Sincerely,
Michael N. Levesque

WANTED- BASIC ELECTRONICS

Dear Sir,

I read the third Polyphony the same day that it came and let me tell you that the whole thing is great. The format is better in every issue so keep the mag. on & out of the press.

I was thinking of something that I forgot to put in the questionnaire. I think if you can add a special section that deals with how those new keyboards & black boxes work, and perhaps a section like the one of music theory, but instead of music write about the ABC's

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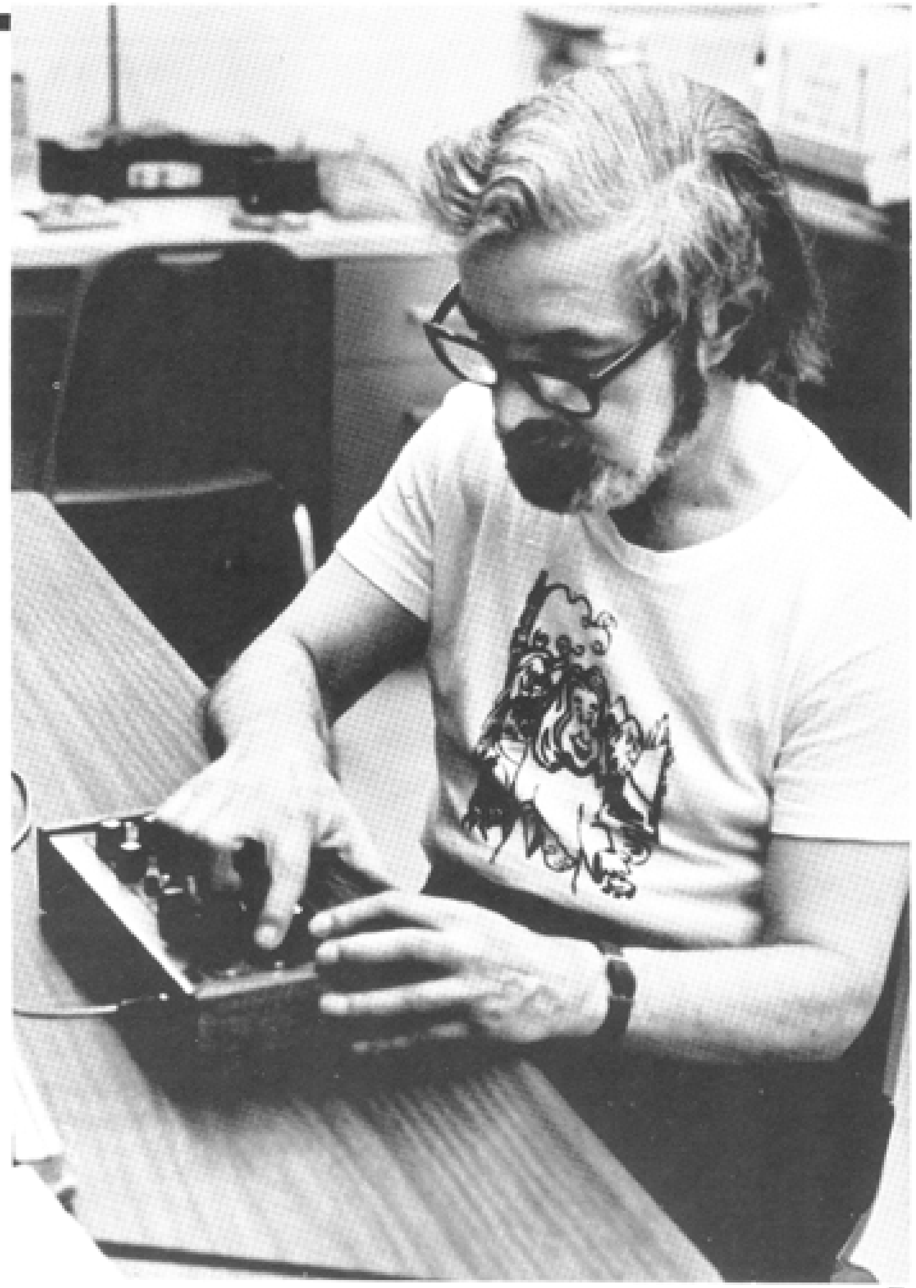
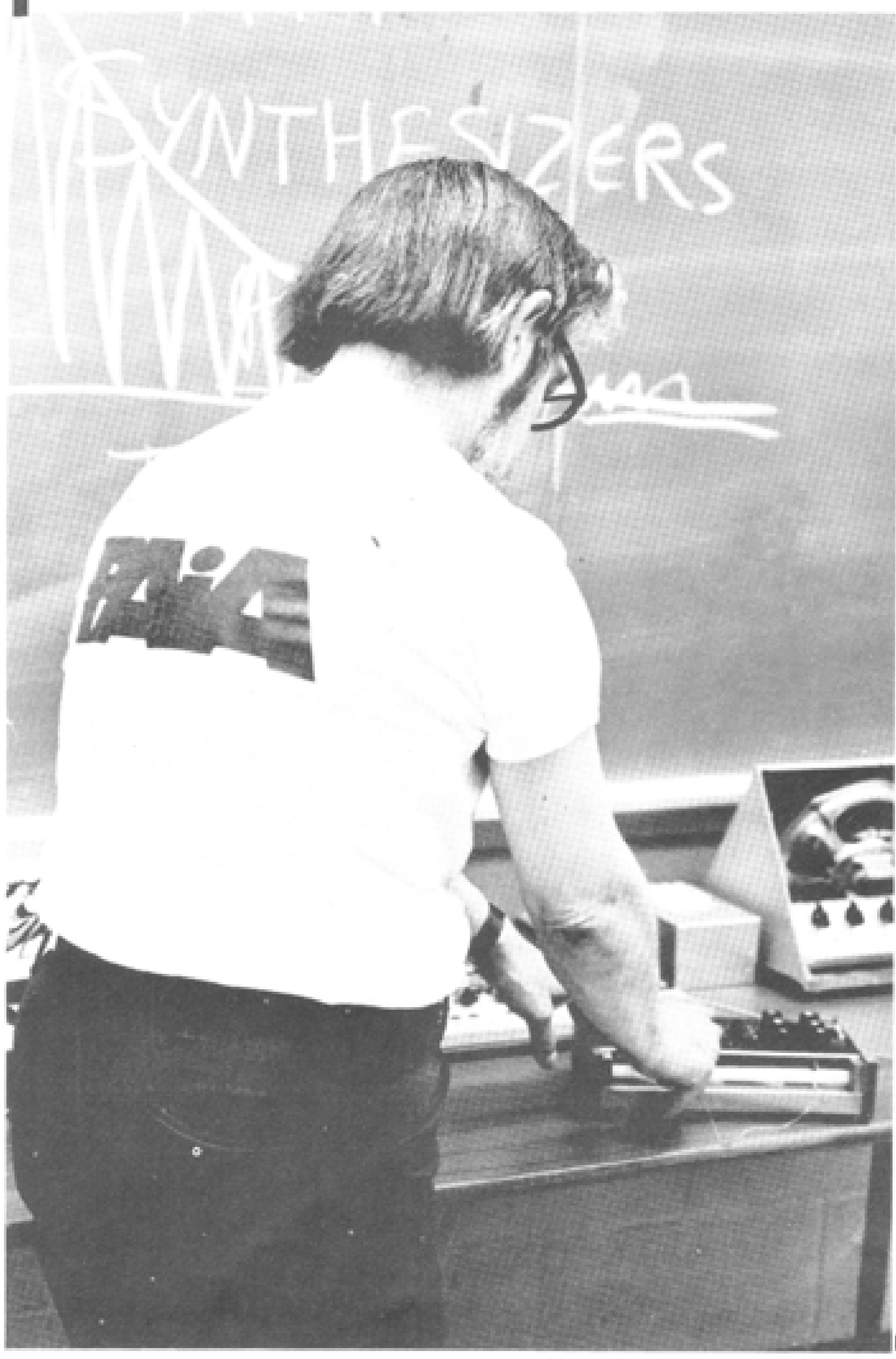
SPOTLIGHT



R. W. BURHANS

OHIO UNIVERSITY

ATHENS, OHIO



Ralph Burhans is a research engineer and lecturer at Ohio University's Dept. of Electrical Engineering - where he conducts one of the country's few courses on the design of electronic musical instruments. PAIA Gnome and 2720/A packages as well as equipment from ARP and other manufacturers are used as both "hands-on" instruments and design examples in this seminar.

Ralph's critiques of PAIA products have in the past been an invaluable asset in product improvement.

Readers of technical journals such as the Journal of the Audio Engineering Society will recognize Ralph as a constant contributor. His past work has included the development of the "Petite Clavichord" and timbre box; and amplification/filtering system that maintains the character of the distinctive clavichord sound while allowing it to produce sound levels consistent with its performance with other instruments.

His work in the digital generation of audio waveforms resulted in the DIGIVOX II, one of the few entirely digital musical instruments ever designed.

Ralph's current interests involve him in the design of extremely low cost electronic navigation systems ranging from Loran to Very Low Frequency Omega receivers.

CONVERT YOUR PYGMY INTO AN ELECTRO-LARYNX

By Marvin Jones & Jim Fleming

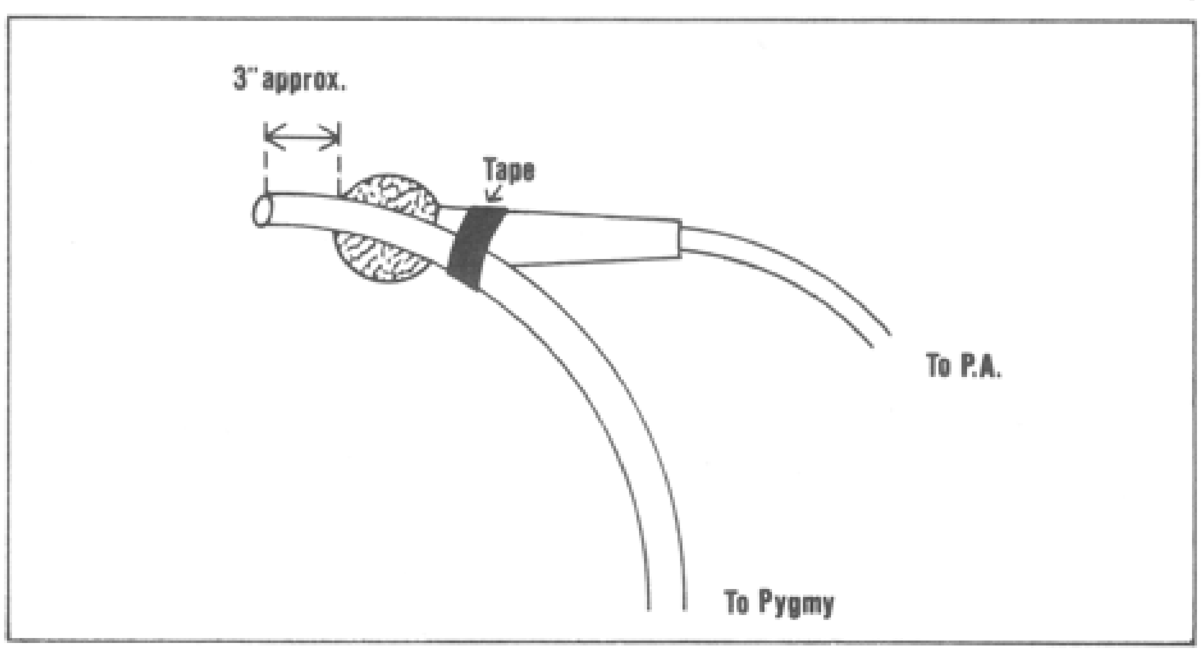
They've been around for quite a while now -- about six years. Iron Butterfly and Steppenwolf used them. Todd Rundgren used one on "Breathless" and "Range War". But the two people who finally sold the public on these were Joe Walsh ("Rocky Mountain Way") and Peter Frampton ("Show Me The Way" and "Do you Feel Like We Do"). Yes, I'm talking about those gizmos with a rubber tube that you stick in your mouth so your guitar (or other instrument) sound comes out of your mouth and you can say words with your instrument. Any of you who have been stricken with this fever and checked into prices on t these units probably changed your mind rather abruptly. But, yes, dear friends, PAIA has once again broken the price barrier with a $5. modification kit which can be installed on the PYGMY to convert it into a voice processor for electric instruments. The modification kit includes a replacement speaker bezel which concentrates the sound from the speaker and emits this higher pressure signal from a small hole. The sound pressure is fed into a 6 foot length of rubber hose which terminates in one's mouth. Thus, audio signals fed to the input of the Pygmy are accoustically deposited in the human resonant cavity most suited for formant processing.

FURTHERMORE, we have a special deal for those of you who haven't gotten a Pygmy yet. During February & March of 1977, if you buy a Pygmy we will include the Electro-Larynx modification free. Now really -- where else could the average George Leroy Tirebiter find such a fantastic deal on such a nifty development of modern space(d) technology?

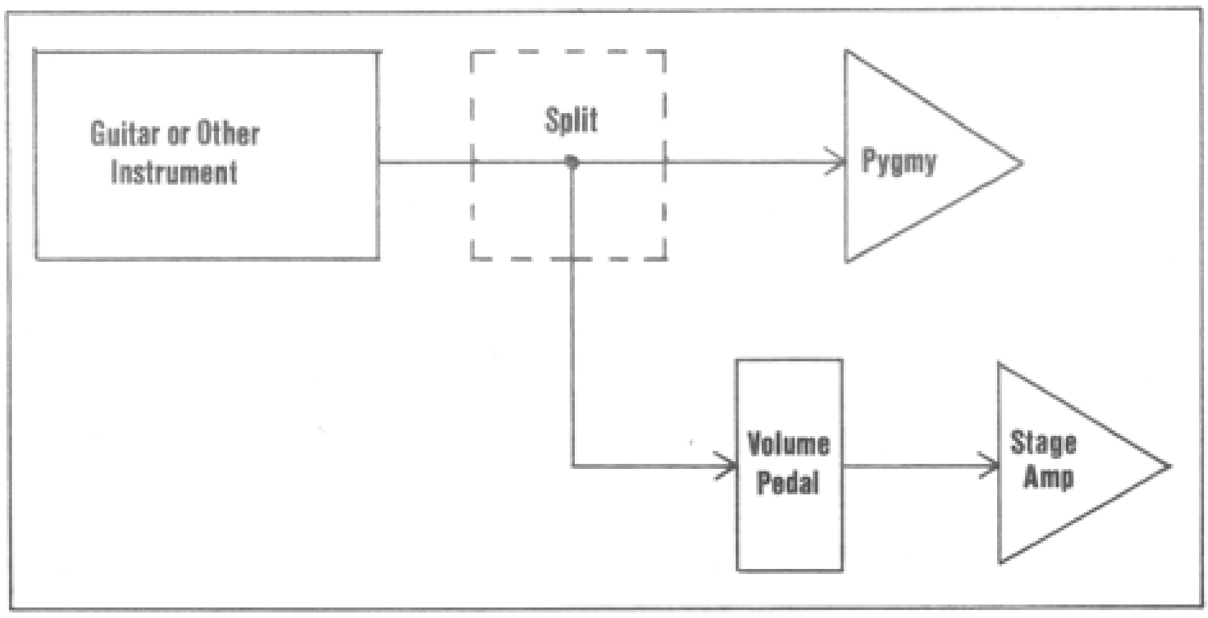
Seriously, this device is a lot of fun to work with and mechanically easy to operate. As with any type of unit that radically processes a signal, artistic control will be the hardest factor to master. Too much of an effect can ruin its impact.

In use, the free end of the rubber tube should be taped to a microphone such that the end of the tube is about 3 inches beyond the mike. When the talking effect is desired, the tube is placed inside the mouth, and the sound is picked up by the mike and amplified on the PA. Only certain phonetic sounds can be produced, and some sounds (like b, m, t, and l) will sound about the same. Try not to use your own vocal cords, and avoid using your breath for such sounds as S, P, or K. This is a good way to make your soundman an enemy. Work with the unit a lot to gain better understanding of what can and can't be done. Try saying as many words and sounds as possible. You'll soon get the hang of it!

When used in a performance situation,



your guitar signal will need to be "split" with a Y cord so it can be fed to both the Pygmy and the larger stage amp. The Pygmy should be set near maximum volume. This will produce a lot of distortion, but the high order harmonics of the distortion greatly increases the intelligibility of the talking effect. Also, most performers will want some method of eliminating the stage amplifier when the Pygmy Electro-Larynx is being used. This will avoid having the talking effect "masked" by the higher volume amp signal. The easiest way to achieve this is illustrated in figure B. After the guitar signal is split, the line going to the stage amp is first fed through a volume pedal such as the PAIA/De Armond #1600. In this set-up, sound will continually be fed from the tube on the Microphone, but the tube is pointing away from the mike and should cause no problem. When working with the talking effect, try running the instrument through various types of processing prior to the Pygmy Electro-Larynx. You may end up with a wa-wa voice, or an echo voice, or any number of combinations. Have fun with this one!



LETTERS

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of electronics. (My weak points are the transistors & CMOS, not to mention the TTL & micro-processors; I don't have anything about them - I mean books).

That's all friend,
Guillermo Santiago

COMMENTS/QUESTIONS

Dear Editor:

I have a bunch of opinions and speculations in response to John's column on digital keyboards in the 1/76 Issue of Polyphony.and some questions that I would like answered about PAIA synthesizers.

I am a computer programmer (for money) by day and a musician (for fun) by night, so for me the most interesting idea in John's column was the idea of programming a digital memory to play music. I have programmed a computer to play music, storing bits in magnetic core memory and picking up the Hertzian waves caused by magnetizing the cores on a radio. I used two words for each note: the first gave the pitch (number of machine cycles before a bit is stored) and the second gave the duration of the note (number of times through the pitch loop - which varied for different pitches for the same duration). The most tedious part was coming up with the numbers to put into those two words for each note; that is, programming the tune. If the computer were larger and had a console typewriter I could have written a routine to accept some sort of typewriter input and calculate the pitch and duration from that. But what sort of input?

The problem in programming will be to come up with an operating system analagous to a compiler (like Fortran) for a composer to use in entering a composition into memory. However, by the time Fortran was invented, mathematics had a very extensive, precise notation easily adaptable to computers; this is not the case with music.

For programming a composition we need at least the following parameters for each note: pitch, duration, articulation, and intensity; or, instead of intensity for each note, phrasing over an entire phrase, which would be easier for the composer to work with. This is the minimum for piano and organ works up to the beginning of the 20th century; other instruments have even more parameters (vibrato and others); modern music can accomodate all these parameters and more besides.

What would I like to see in programmable synthesizers? Well, for a start, a machine that can play existing pieces of music that are unplayable by humans (some of Paganini's violin music, for example) with the same "warmth", "expression", "dynamics" or whatever that a human would use in playing them if he could. (I know that the words in quotes are not defined or even completely understood, but hopefully we will discover how to give a reasonable facsimile of them as we go along). Such a machine would make it possible for composers to write anything at all, playable or unplayable without any more work than would go into writing a piece with conventional orchestration.

Music like "Switched-On Bach" is nice but impractical when you have to spend 400 hours to produce what a few instrumentalists could do better after 10 hours of rehearsals.

Anyway, if we can ever understand just what we want from an instrumentalist clearly enough to be able to notate it precisely on paper, the programmable synthesizer will do for composers what the computer did for engineers, scientists, and clerks.

Changing the subject sideways, slightly, here's what I would like to see in ordinary synthesizers (and I'm surprised that no one has built it yet): Mainly, a pedal keyboard for bass lines. Preferably a 32-note organ style keyboard. It would not have to be polytonal. However, it would be expensive. The Shober Organ Company of New York offers 32-note pedal clavier kit for $276.50 and that's just the wood parts, no electronics. In my opinion, the most practical real-time-oriented large scale synthesizer would be one modeled on the electric organ, with a pedal keyboard for bass lines and occasional melodies (when your right hand is busy changing patches), one or two five-octave monophonic keyboards and at least one polytonal keyboard for accompaniment. This would give the synthesist a start toward becoming a self-contained live performance musician (like a pianist), but he would still need: 1) something analagous to an organ's combination action for instantly switching from one patch to an-perhaps a matrix switchboard like the Putney synthesizers, 3) sequencers for automatic accompaniments, or a program memory for large-scale accompaniments, and (most important) 4) more technique than Virgil Fox just to be able to handle it all and make it sound like something you would want to hear.

Maybe I'm dreaming, but it's fun.
Mark Lutton

PROGRAMMABLE DRUM SET

John Simonton

It' s fascinating, the way the development of some products is a struggle from beginning to end, while others seem to almost design themselves.

This programmable drum unit is an example of the latter. In retrospect, it seems to be an amalgamation of effortless insights and serendipitous "discoveries" that all came together with such perfect timing that the product itself just sort of spontaneously appeared. No Sweat.

And, best of all, now that it's designed I discover new things that it can do every time I play with it.

But, this is getting ahead of myself. First I ought to tell you what it does - and more importantly; why it does it. You're really going to like this one.

As I see it (and I've designed these things before so I certainly have a right to an opinion) past percussion units have had three major failings:

1) They do the same thing again and again.
2) They do the same thing again and again
3) They have no provision for unusual time signatures.

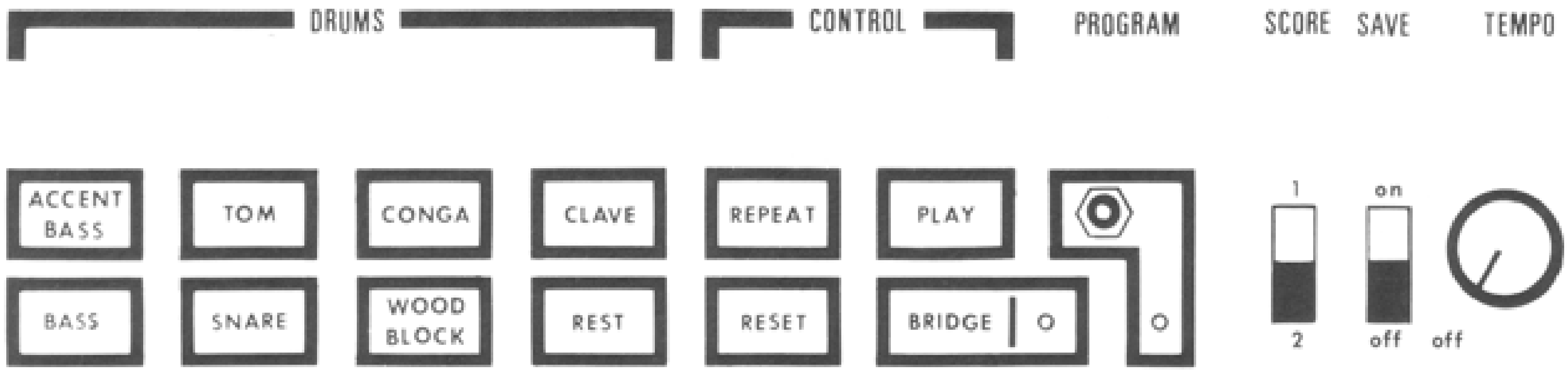
Are you surprised that the first two are the same? I was. I'm talking about two different "sames though. Let me explain.

The first same is that other than choosing very generalized and idealized "classic" rhythm patterns, you have no control over the percussion score that is generated. Every time you punch up "tango" it's going to be just like the last tango. Programmability is the obvious key here. You can produce as many different tango's as you can dream up.

The second same is that the same pattern repeats, at best, every couple of measures. Real music isn't played like this. In real music you've got intros and bridges and jazzy stuff like that. This unit takes care of that by giving you a BRIDGE key. When this control is activated, instead of repeating the same pattern again, the unit switches to a separate pattern and repeats it as long as the bridge key is activated. When the bridge key is released, the unit shifts back to the first pattern and begins playing it again. I should mention that the shifting from one pattern to another happens automaticly only at the repeat points (the end of the pattern - more details on this shortly), but it can be "forced" to happen anytime.

Every rhythm unit plays 2/4 and 4/4 times with no problems. The ones that are worth anything at all also include a couple of 3/4 or 6/8 things too. But, I don't know of a one that has provision for even 5/4 time - and as

The Controls Look Like This:



"unusual" time signatures go, that one's pretty common. This new unit can be programmed for any conceivable pattern, no matter how strange the time signature.

Earlier I said that serendipity was a major contributor to this product and here's what I had in mind. All of the labeled squares that you see on the control surface are non-mechanical touch switches. Even the slightest touch activates them (the touch switches will be analyzed in the next issue, but we're not going to be concerned with the circuitry now, only the operational details of the entire unit). The design of these switches was purest dumb luck.

Let's talk about concept for a moment.

Inside the unit are drum circuits, logic and 256 eight bit words of semiconductor read/write (RAM) memory. Each of these words represents an "event" that was, or will be, programmed. When the unit is playing a pattern back a clock whose speed is adjusted by the TEMPO control causes a counter to step through the events. Whatever is recorded at each event is going to happen. What's recorded can be one of three things: a drum beat, a rest, or an instruction that causes the circuitry to go back to the beginning of the pattern and repeat it.

256 events is a lot of memory for a thing like this. Too much, in fact, if it represented only a single pattern. It doesn't. It represents 4 different patterns, each of which can be up to 64 events long. The 4 different patterns can be further broken down into two 64 event "scores" each of which has an associated 64 event "bridge". The slide switch labeled SCORE 1/2 allows you to select either score for programming or playback.

Let's say that we want to program a pattern:

The first action would be to touch the RESET pad. This resets all of the internal circuitry and will in general be the first step in doing anything.

Select which of the two patterns (SCORE 1/2) you are going to record and set the slide switch accordingly.

Press the PROGRAM push-button and observe that the PROGRAM LED lights indicating that you are in the program mode.

Now the DRUM pads are activated in turn to produce the desired rhythm pattern. And remember that most musically interesting rhythm scores will have as many RESTs as drum sounds. When you reach the point at which you want the pattern to repeat, touch the REPEAT pad.

The setting of the TEMPO control during programming is not critical, but leaving your finger on any DRUM pad will cause that drum sound to repeat at the TEMPO rate. This also causes that drum sound to be recorded in successive events in memory.

Programming a BRIDGE is essentially the same as programming a SCORE, the only difference is that you have to get the unit into the BRIDGE mode (as indicated by the BRIDGE LED) by touching BRIDGE and RESET before pressing the PROGRAM button.

Playing back a score is simply a matter of setting the tempo desired and touching the PLAY pad. When you get ready to play the BRIDGE pattern, touch the BRIDGE pad and at the next REPEAT event the unit will automatically shift to the bridge, play it, and then shift back to the main pattern. If the BRIDGE pad is continuously touched, the bridge will play continually.

All of that would be plenty neat by itself, but there's more.........

You don't have to start playing with the main score, you can treat the bridge as an intro by putting the unit into the BRIDGE mode and touching PLAY. When the end of the bridge pattern is reached you will automaticly shift to the main score.

You don't have to let any pattern go to completion, touching the REPEAT pad at any time will cause the drum set to go back to the beginning. If the BRIDGE pad is touched at the same time as REPEAT, you will go to the beginning of the BRIDGE. This opens up enormous possibilities for having programmed rhythm patterns that seem to never repeat.

You don't have to play a score back at a fixed tempo. Touching the REST pad causes the drums to play at the TEMPO setting (without being "locked" into PLAY mode). Tapping the REST pad causes the drums to play at the tempo of the taps. The unit has an external Synch input, so it's tempo can easily be governed by an external clock (a sequencer, for example).

The two programmable scores that you have available don't have to be entirely different, they can be part of the same score and judicious use of the SCORE 1/2 slide switch will allow for switching back and forth between the two parts.

If you're beginning to get the idea that this unit is an extremely versatile device, I think you're right. It's not going to replace a living percussionist (a good one anyway) but it does come a lot closer than anything else that I know of.

The programmable drum unit is battery powered and runs off of 4 "AA" size pencells and a single 9 volt transistor battery. The nine volt battery supplies bias to the noise source in the snare circuit and should have essentially shelf life. The pencells should be good for about 20 hours of intermittent operation.

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FUNDAMENTAL MUSIC NOTATION

Part II ... TIMING

By Marvin Jones

In writing or determining the timing used for a musical piece, several factors are involved interactively. The physical shapes of the notes represent timing of the notes in relationship to each other. A time signature given at the beginning of a piece gives a full beat to one of the note shapes. And finally, a beat frequency is stated either by number of beats per minute or by a term or phrase representing a certain range of beats per minute.

PHYSICAL SHAPES OF NOTES AND RESTS are shown in figure A. Each of the notes or rests are equal to two of the units on the row below. For example, a half note is equal in duration to two quarter notes. Or, a quarter rest is equal in duration to two eighth rests, and so on.

DOTTED NOTES are 50% longer than standard notes. For example, a dotted half note is equal in duration to three quarter notes, or a half note and a quarter note. Dotted rests are not used. If you wanted a rest equal in duration to a dotted half note, you would use a half rest followed by a quarter rest.

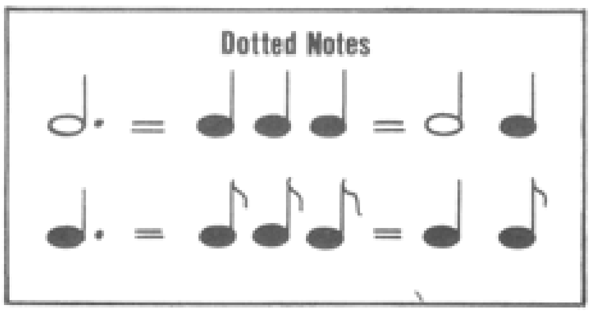


Figure B

A BEAT is a unit of musical time. When using a metronome, the ticking of the metronome represents the musical beat. Any of the note values can occupy a full beat, and this information is specified in the time signature at the beginning of a piece.

A BAR-LINE is a vertical line drawn on the staff to divide the music into measures.

A MEASURE is a fixed number of beats of music which occurs between two bar-lines. The fixed number of beats will be stated in the time signature of the piece.

A DOUBLE BAR-LINE is used to divide major sections of a composition (such as dividing themes, or compositional styles. A double bar could be used between an introductory fanfare and the beginning of a waltz, for example.) A double bar is generally used at the end of a song.

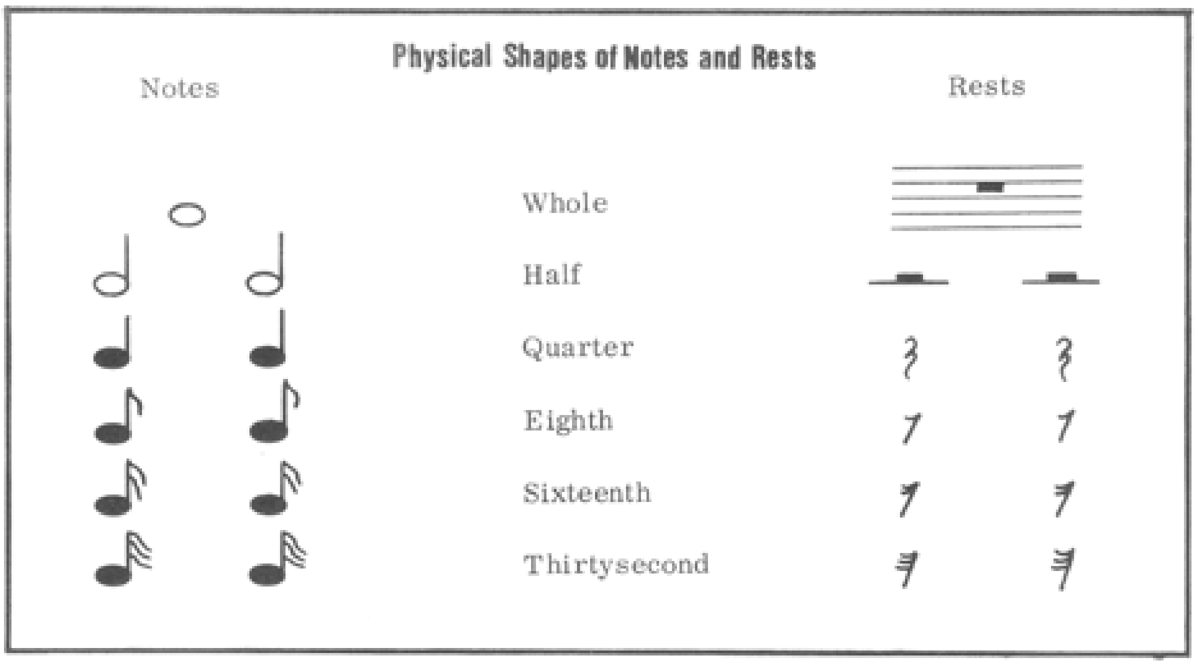


Figure A

TEMPO is the speed at which a composition is to be played. A specific frequency can be stated at the beginning of the song, written above the staff, such as ♩ = 60MM. This indicates that a quarter note will occur 60 times per minute. MM stands for Maelzel's Metronome.

The second method of tempo indication is slightly more popular, and in my opinion, more artistic. This method uses standard Italian terms to indicate approximate range of tempos. This allows for more expression or interpretation by the performing artist. The most standard terms are: (from slowest to fastes) largo (broad), lento (slow), adagio (at ease), andante (walking), moderato (moderate), allegro (fast, cheerful), presto (very fast), prestissimo (as fast as possible). These terms will normally be found written below the staff at the beginning of the piece, and throughout the song wherever speed changes are required.

Figure C

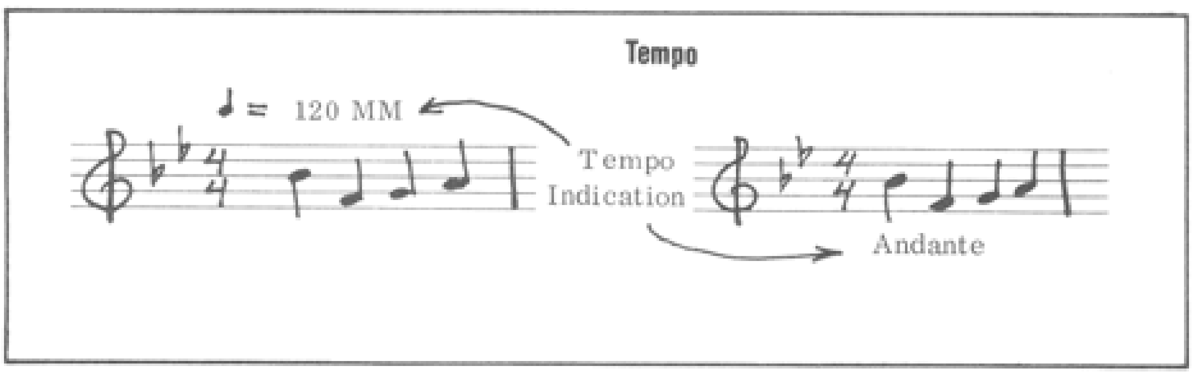


Figure D

TIME SIGNATURES are composed of two numerals stacked one above the other, and written directly on the staff after the key signature at the beginning of the song. The upper number states the number of beats in a measure, while the lower number signifies which note style occupies a full beat. 2/4 represents two beats per measure with a quarter note lasting one full beat. 3/8 represents three beats/measure an eighth note getting a full beat. Two abbreviations are commonly used in time signatures. One is C which represents 4/4 timing (often called Common time). The other is ¢ which means cut time (or Alla breve) and represents a 2/2 time signature.

Examples of several time signatures are shown in figure E. Several measures of music with the beats indicated by the arrows under the staff are illustrated.

When you finally get down to writing out your music, there are several points which may make your job easier.

FLAGS OR BARS? When several consecutive notes have flags (eighth, sixteenth, etc.), those notes within full beats can be barred to eliminate drawing a lot of flags. Also, this allows the performer to more easily see what notes fall in what beats.

STEMS - UP OR DOWN? To avoid wasting manuscript space or interfering with adjacent staves, notes written above the middle of a staff should have their stems turned down and connected to the left side of the note body. The major exception to this rule is that when a group of barred notes contains both high and low notes, all stems should be in the direction that the majority of the notes would normally use.

WHOLE RESTS can be used to signify a whole measure of rest regardless of the number of beats per measure. For example, a whole rest in 3/4 time occupies 3 beats while a whole rest in 7/8 time can last 7 beats.

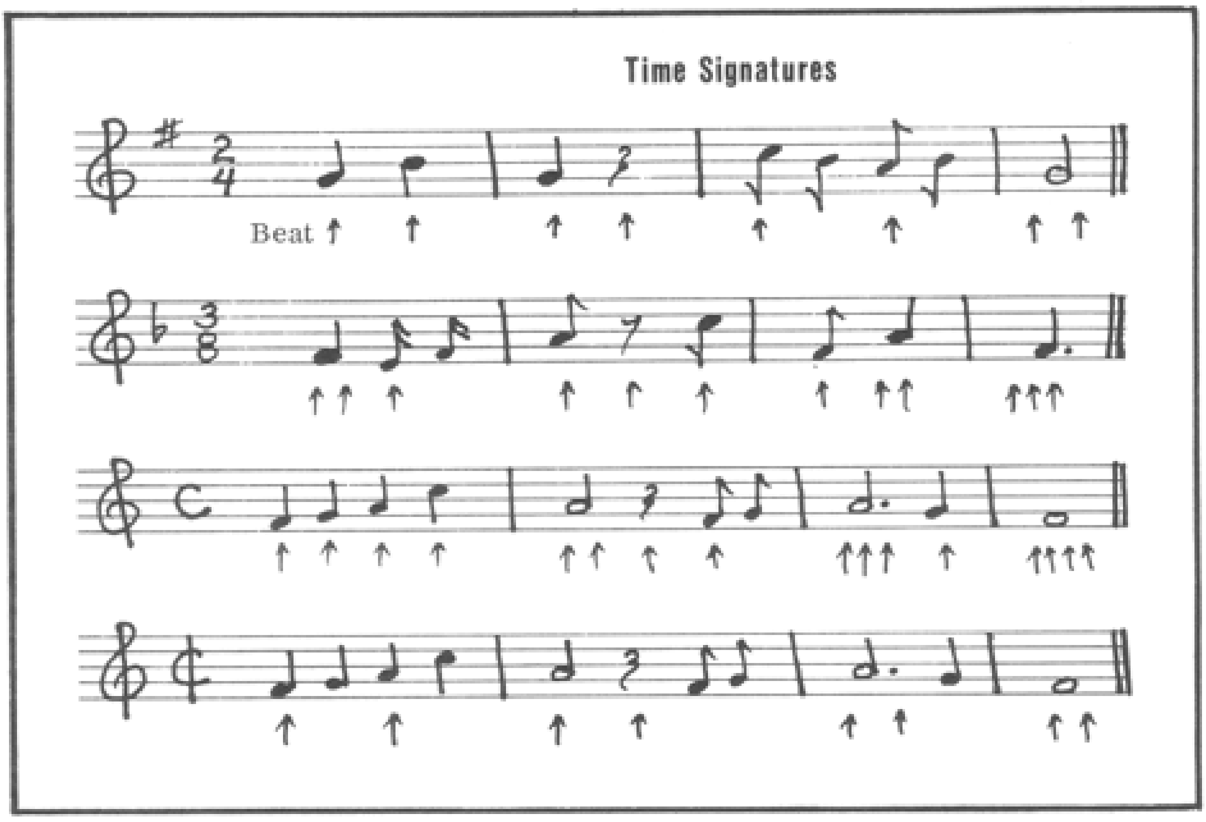


Figure E

Figure F

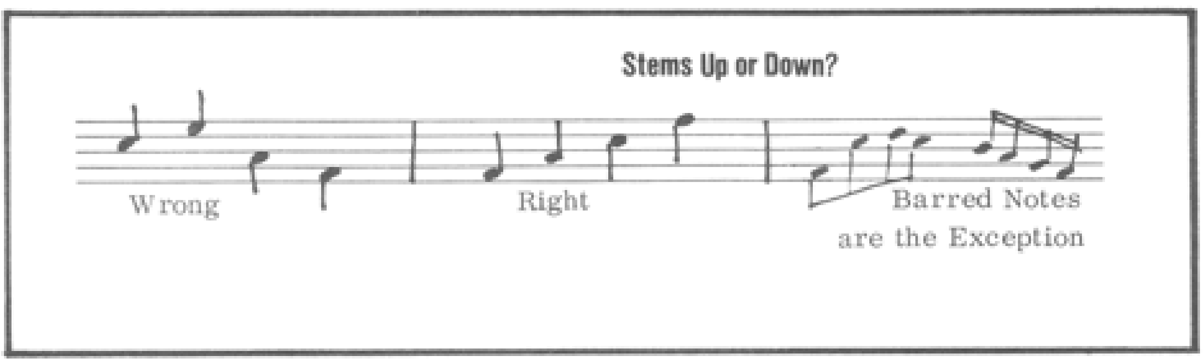


Figure G

Figure H

EXTENDED RESTS - When writing out separate musical parts for separate performers, it is not uncommon for one instrument to have several measures of rest while the rest of the performers continue. In this case, rather than writing measure after measure of whole rests, the total number of measures rest can be indicated in one measure to save space.

continued . . . . . page 12 . . . . . .

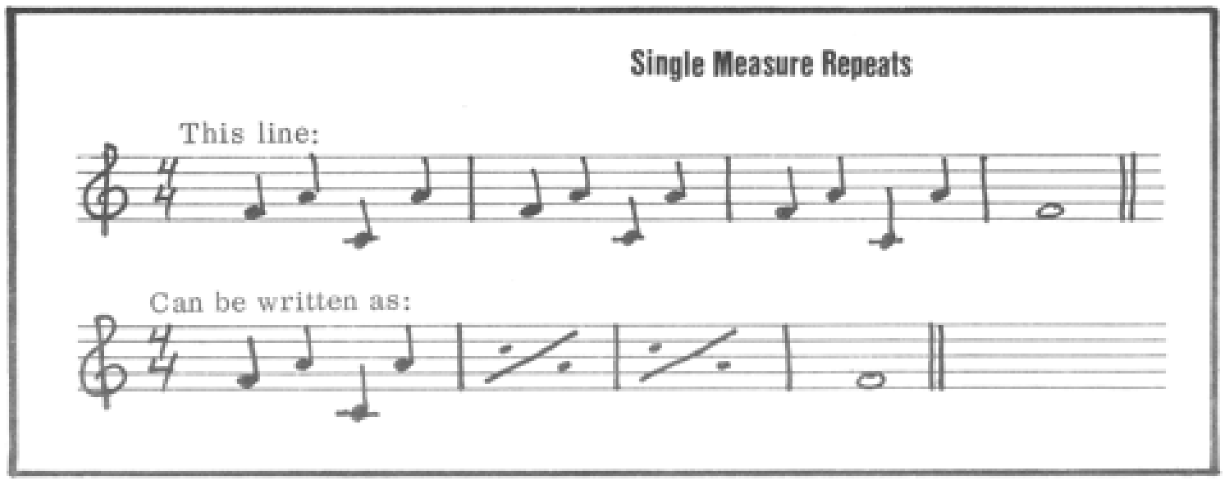


Figure J

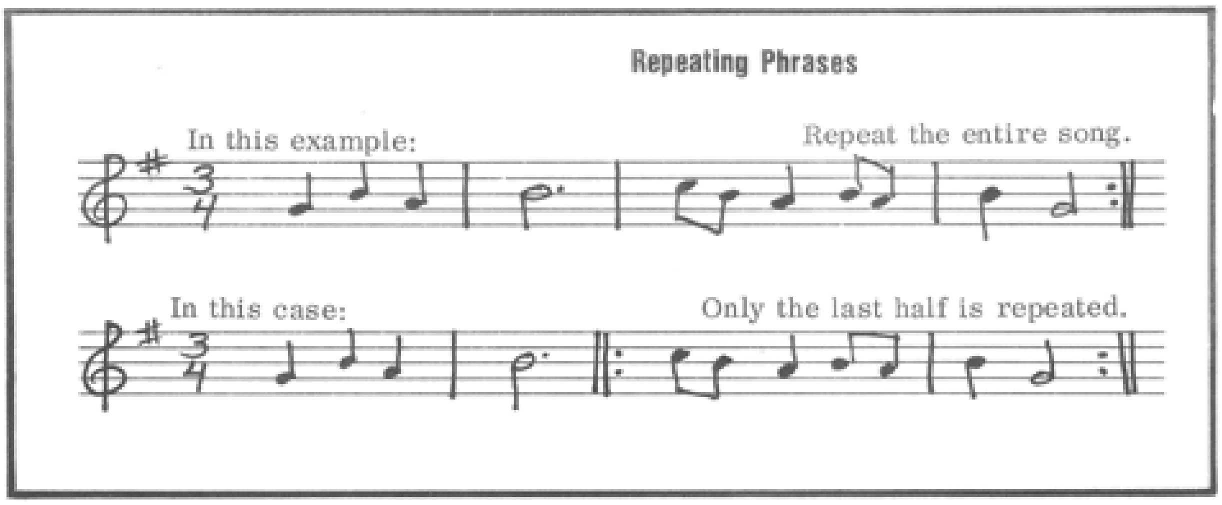


Figure K

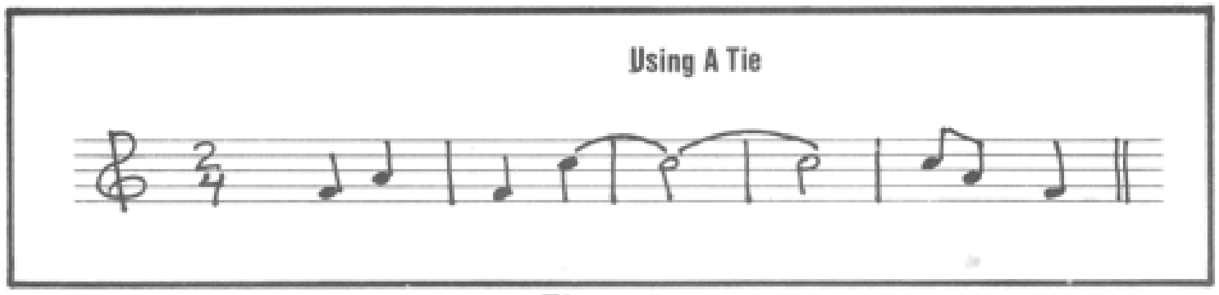


Figure L

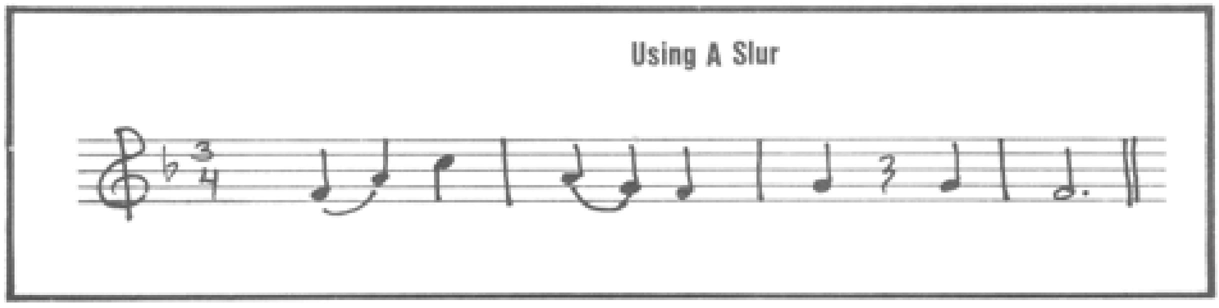


Figure M

MUSIC NOTATION.........

SINGLE MEASURE REPEATS are useful where a specific note pattern in one measure is repeated in two or more consecutive measures. Only the first measure of the pattern need be written out. Single measure repeats are then used until the pattern is to be changed. The "Oom-Pa, Oom-Pa" bass line shown below is a perfect example.

REPEATING PHRASES is signified by colons inside double bar lines. There should be a double bar prior to and following the phrase to be repeated, unless the repeat is to start at the beginning of the piece.

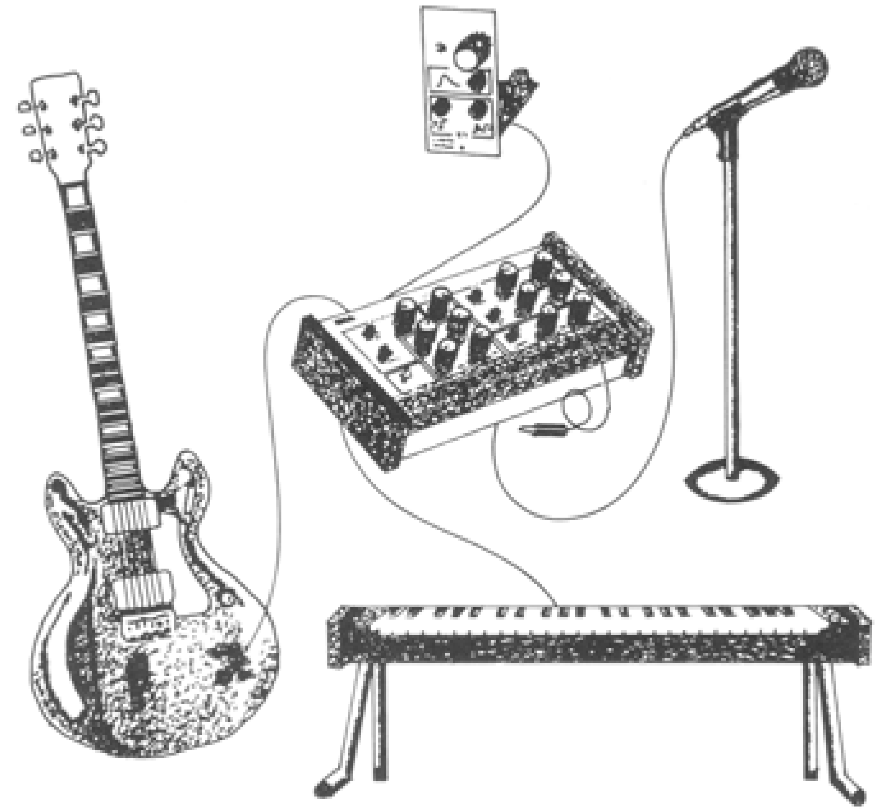
A TIE is used to show that a note is to continue longer than allowed in any one measure. For example, if a note is to be held for five beats, but you are only allowed two beats per measure, you will need to tie two half notes and a quarter note together as shown.

A SLUR is used when you want to tie two different notes to fall within the same envelope. The first note is played as usual and at the proper time the pitch is shifted to the next note without rearticulation or any silence between the two. This should not be confused with portamento or glide, as the pitch change is an instant one. On a synthesizer, this would be accomplished by pressing the second note slightly before the first key is released, thus preventing the envelope shaping from resetting and refiring.

By now, you should be able to fairly accurately notate your music -- at least well enough to jot down your musical concepts until you have a chance to fully develop or record them. In future articles we will "put the frosting on the cake" by discussing common methods of indicating expression and variables in your music.

-30-

Interfacing External Signals with the Gnome Micro-Synthesizer



Many people are interested in low cost methods of processing signals from standard musical instruments, microphones or recorded signals. As well as serving as a low cost introduction to the principles of synthesizers, the Gnome can easily be modified to become a versatile processing center for external signals. Here are some ways to accomplish this type of interface, each suited to slightly different circumstances. Perhaps you' ll find one here that is just what you' ve been looking for.

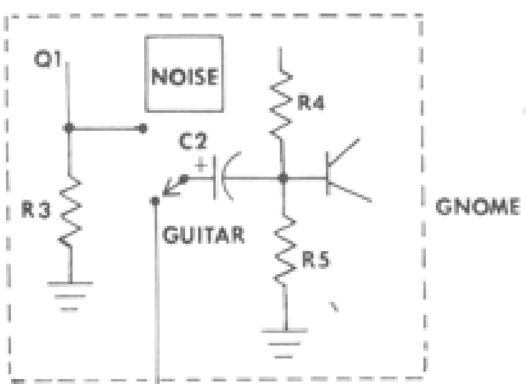
ed.

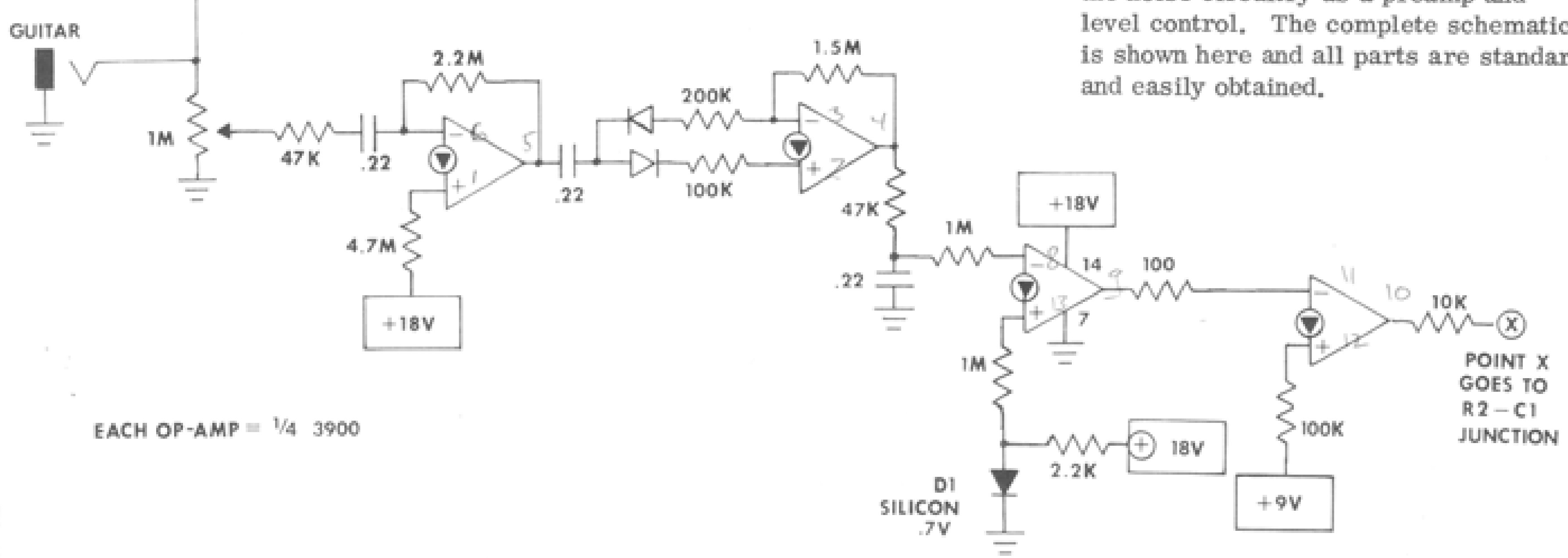
Guitar/Gnome Interface

—— by: Craig Anderton ——

This circuit serves as an automatic trigger for the GNOME. A suitable jack for an external signal input can be mounted on the rear panel. The external signal is applied to a 1 meg sensitivity control which can also be mounted on the rear panel. The first stage of the LM3900 is a high gain audio amp. The output from this stage is then fed to a rectifier built around the second amp of the IC. The rectified signal is filtered by the 47K/.22 mfd. combination. The voltage across this capacitor is present only when an input signal is present. Thus, the comparator output is connected to the point which will cause both of the Gnome's envelope generators to begin sweeping. In other words, the applied external signal will produce the same effect as pressing the trigger button. The original input signal is simultaneously applied to one side of a SPDT switch mounted on the GNOME case. The positive side of C2 in the Gnome is lifted from the circuit board, and an extention wire connects it to the wiper of the switch. The remaining side of the switch is connected by a wire to the unused positive mounting hole for C2. This switch will allow the user to select noise generation or external signal generation.

Another nice feature of this circuit is that your external signal will now use the noise circuitry as a preamp and level control. The complete schematic is shown here and all parts are standard and easily obtained.





Gnome/Instrument Interface

Using the 2720-11 Envelope Follower

——— by: Marvin Jones ———

For those of you who like to avoid making your own circuit boards and tracking down the parts for your projects, we present an alternative. You will still need to install an external input jack, and this can be connected to the Gnome's noise circuit as shown in Craig's circuit or it can be connected directly to the Gnome's audio buss (Ts-2, lug 3) through a 2.2 mfd. electrolytic capacitor as shown in figure B. Note that the positive lead of the capacitor will go to the audio buss, while the negative lead goes to the tip connection of the external input jack.

To provide the signal detection and triggering function, you can use the PAIA 2720-11 Envelope Follower module. The external signal source will be fed to both the 2720-11 and to the Gnome external input. The desired trigger (step or pulse) output of the 2720-11 is then patched to the external input of the Gnome (the black pin jack on the rear apron of the case).

Be sure to have a common ground between the Envelope Follower and the Gnome in order for the trigger signal to have effect.

As an extra bonus, the 2720-11 has an envelope control voltage output. The voltage at this jack is a DC voltage which varies from 0 to 7 volts and is directly proportional to the amplitude of the signal applied to the input of the 2720-11. The wiper probe of the Gnome can be inserted into this envelope output and you can sweep the frequency of the Gnome's VCO or VCF each time a signal is generated by your external source. This particular set up can yield a great variety of effects.

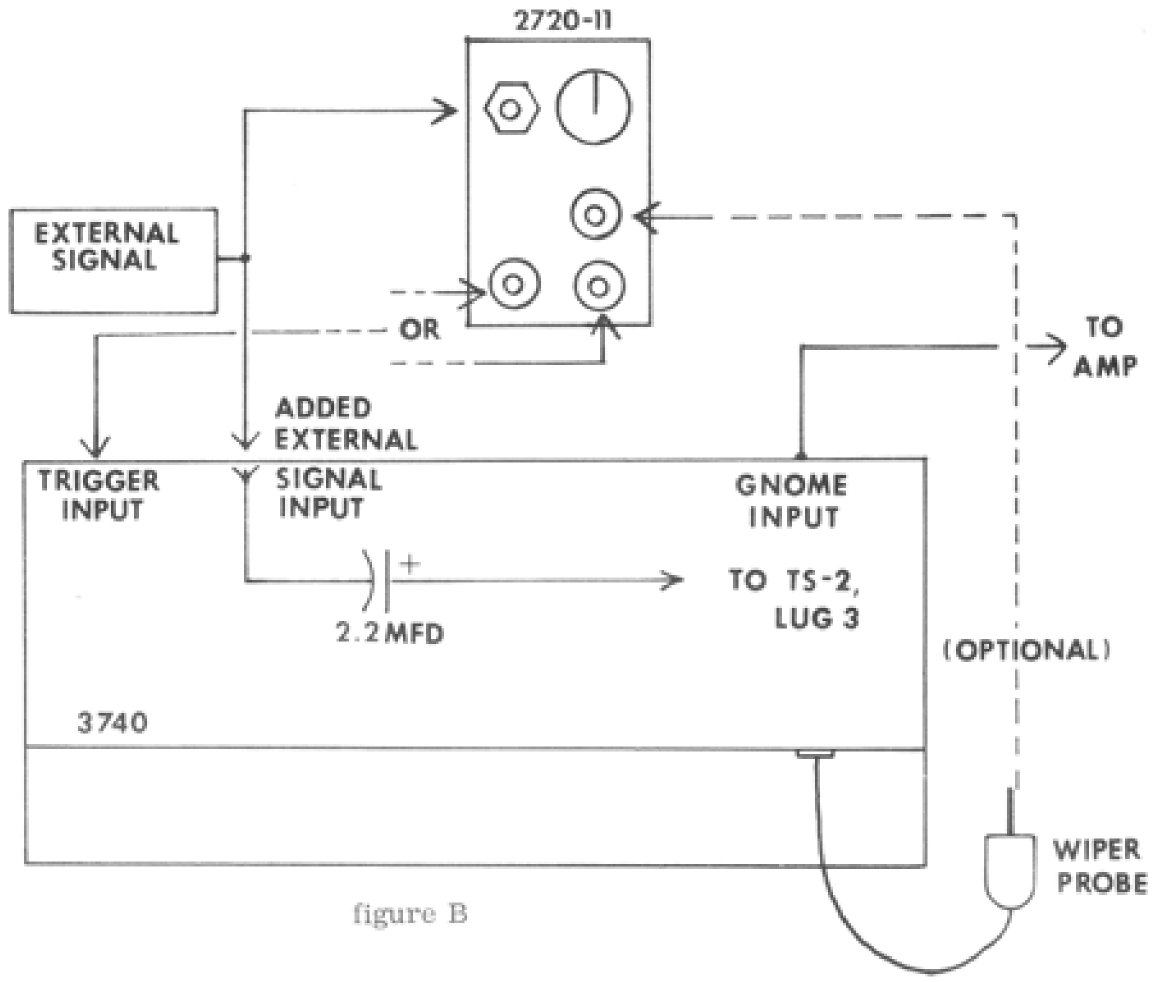


figure B

Electronic Piano/Gnome Interface

——— by: Dana Lee ———

Recently I've done some slightly different things with the Gnome Micro-Synthesizer. It involved modification of the trigger jack and the addition of the external input jack for the Gnome.

The black pin jack for the trigger was removed and in its place a stereo phone jack was inserted. The tip and ring were connected to each side of the trigger button on the front panel - the shield was automatically connected to ground. This means that if I wish to plug in an external trigger footswitch, it can be done using the tip and ring connections. If I want to use an external trigger voltage, I use the tip connection and ground with the ring connection unused (see figure C).

For some time I wanted to interface my electronic piano with the Gnome.

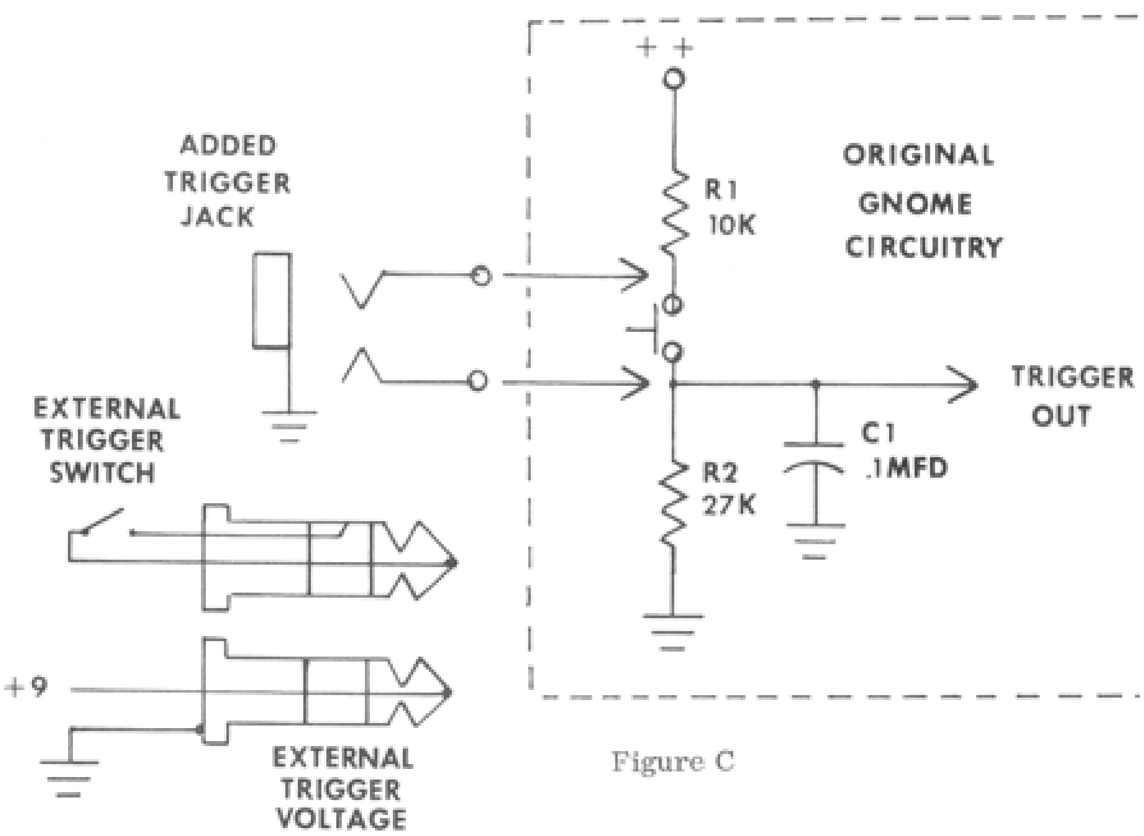
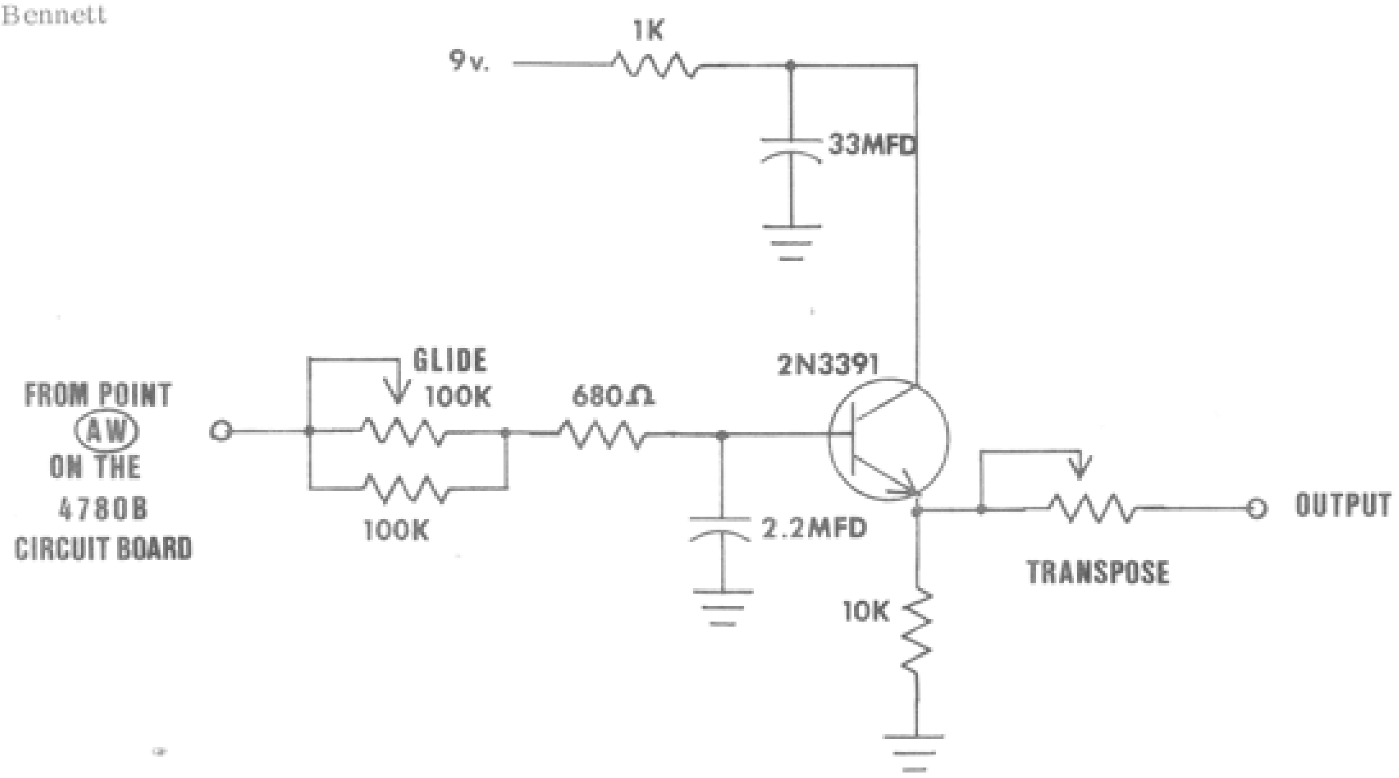


Figure C

continued...... page 31 ....

ADDING AN INDEPENDANT TRACKING OUTPUT TO THE 4780 SEQUENCER

By: Brad Bennett



I have added an extra stage to the sequencer which allows two VCO's to be used with a constant pitch offset on one of the VCO's, This is similar to using a 4781 Transposer on the output of the sequencer, except with this modification individual glide times can be set for each oscillator. This can be used to cause the oscillators to detune when changing from note to note, and after a short time they will resolve to the properly tuned interval.

The circuitry is actually a duplication of the original output stage of the sequencer. I don't believe that several of these output stages would overload the sequencer. I mounted my extra circuitry behind the blank panel of the 4700/A Sequencer Pac. The blank panel can be used to mount the glide and transpose controls and the output jack.

POLYPHONY REVIEWS!

THE LIBERATION OF SOUND

By: Herbert Russcol, $10.00
Prentice-Hall Publishers
Englewood Cliffs, New Jersey

As in any area of study, it is important to know the technical developments and major personalities responsible for bringing synthesis to its present status. Many people would think that electronic music is too young to warrant a complete book devoted to the subject, but in actuality electronic music has shown a steady development since Edison gave us a device for reporduction of sound. In The Liberation of Sound, Russcol starts his history lesson around 1900 when musicians were looking for change and resorted to radical experimentation with tonality and performance methods. Then, as electromechanical technology begins to provide composers with answers to their search, Russcol goes into great detail about the long period of "Musique Concrete" and development of tape techniques. Interesting "behind the scenes" material is provided throughout the historical section of the book, making for light, enjoyable reading.

The second section of the book deals with the more familiar names of the sixties - the decade that electronic music really came of age. Excellent biographies are given for people such as Morton Subotnick, Milton Babbit, John Cage and Karlheinz Stockhausen. The book continues with discussions on the impact of the computer on electronic music, and subjects like calculated (or theoretical) compositional techniques. The final chapter gives numerous reviews and listings of records available in the electronic music field.

Excellent appendices and bibliographies at the rear of the book provide comments from other writers about computer music, a glossary of electronic music terms, and an extended bibliography which should keep you bookworms happy for some time. Even if you aren't a history fan, you will enjoy this book because of the heavy emphasis of technology, and the extremely broad range of the book makes it a frequently used reference source as well.

Marvin Jones

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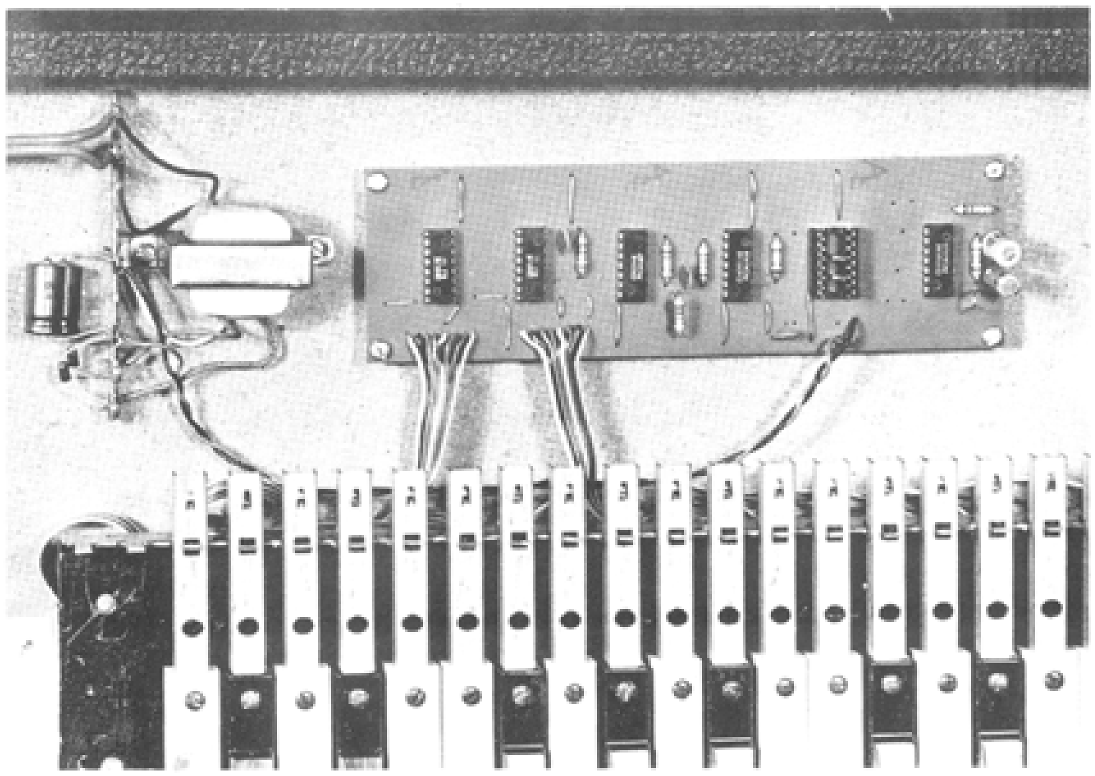

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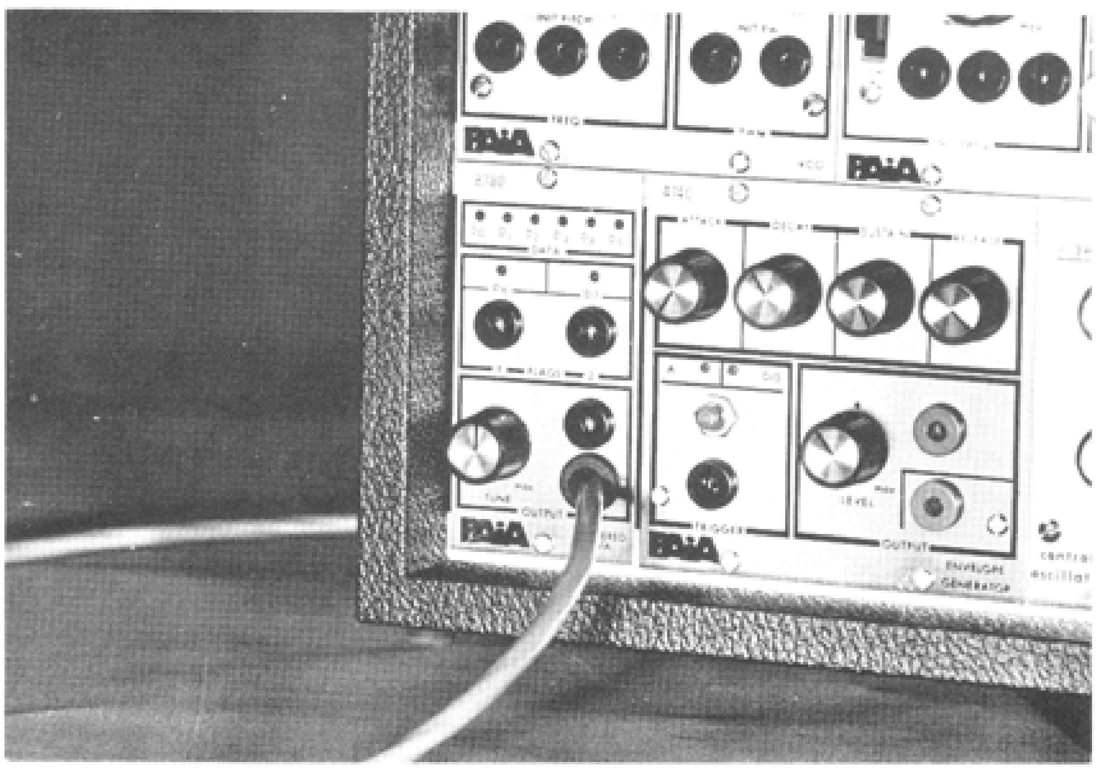

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W NEW NEW NEW NEW

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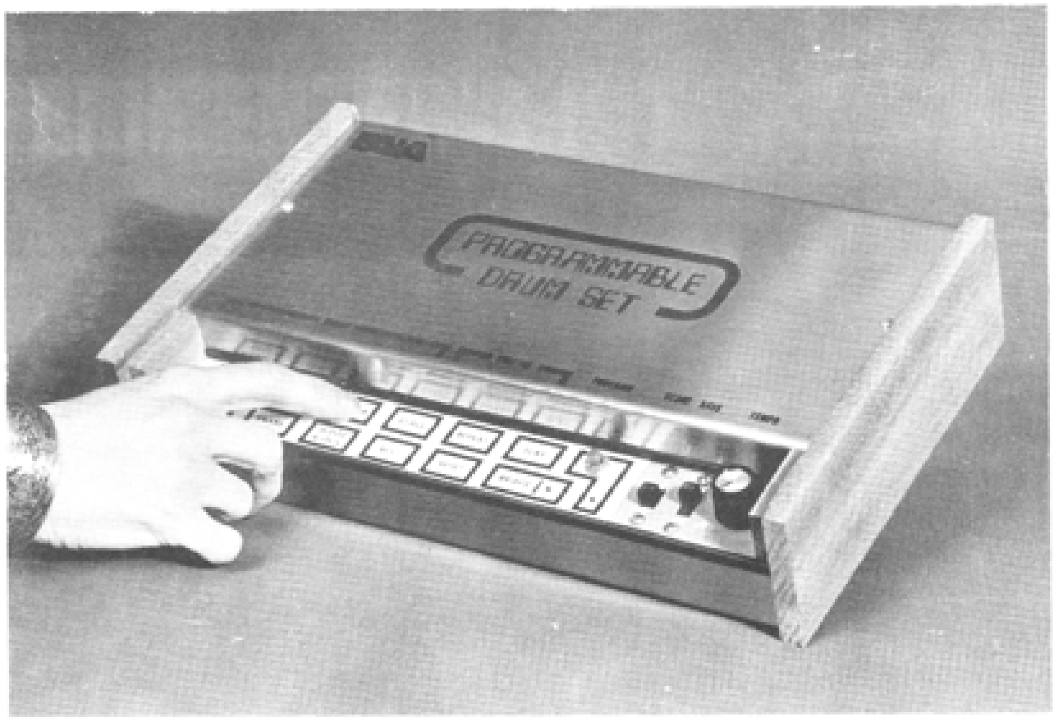

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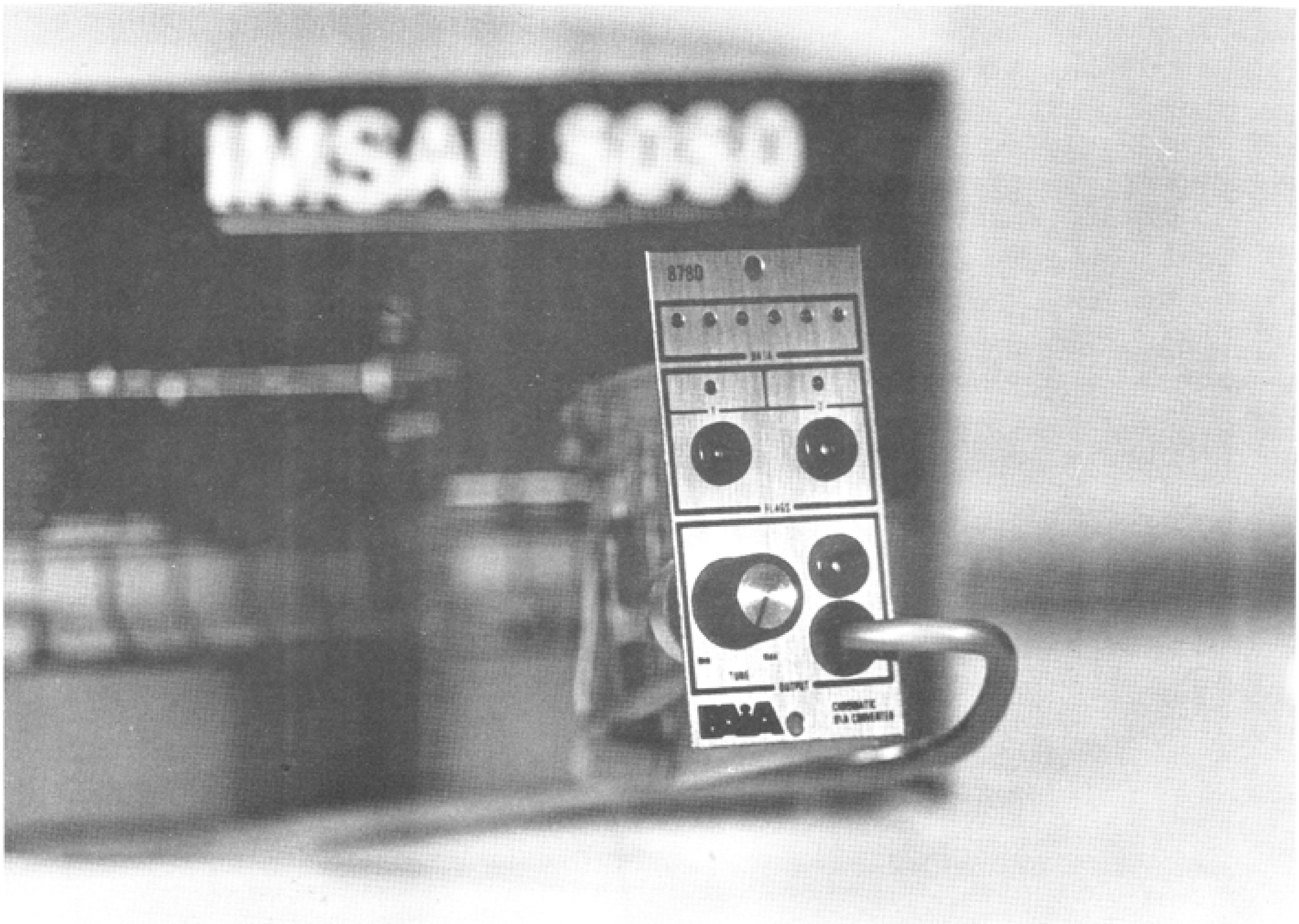


Equally Tempered Digital to Analog Converter

By: John S. Simonton, Jr.

Many experts will tell you that in order to interface a computer to an electronic music synthesizer, you must use exponential response voltage controlled elements (oscillators, filters, amplifiers, etc.).

Here's why:

Computer control of synthesizers requires a Digital to Analog converter to change the numbers that the computer puts out into an analog control voltage that the modules can use.

By far the most common type of D/A (so common that many seem to think it's the only kind) is known as an "R/2R ladder". I don't want to get into the design details of this circuit. If you are interested, there is plenty of information available from text-books, manufacturers literature, etc. But we do need to examine a functional aspect of this circuit.

Any analog to digital converter works by accepting at its input a digital quantity (we will call this data) and generating at its output an analog voltage that is a unique representation of that data. Most of the D/A's that I'm familiar with accept the data as binary digits - a bunch of 1's and 0's that appear simultaneously on a group of wires going into the converter.

In a R/2R ladder converter, a unique weighting is assigned to each bit in the data coming in. When the time comes for a conversion to be made, the circuitry adds together the weightings corresponding to the bits in the data that are in an "on" state (for our purposes, a 1; through not always) and ignores the weighting represented by the bits that are "off" - equivalent to adding in a zero.

If we assume that we are going to be using exponential response oscillators, the R/2R ladder converter works quite well. We can assign weightings to the bits that are integral multiples of 1/12 volt; the same incremental voltage change that keyboards designed to operate exponential oscillators produce, and when we do we come up with a series of weightings which - progressing from the Least Significant Bit (LSB) to the Most Significant Bit (MSB) - Looks like this:

LSB … **MSB**

$$1/12,\ 2/12,\ 4/12,\ 8/12,\ \ldots\ldots\ n/12$$

Figure 1

Where n is, of course, the number of bits that the converter can accept as data.

Let's watch four bits "count" into this type of converter and observe the resulting output voltages.

DATA	MEANS	OUTPUT
0 0 0 0	$0+0+0+0$	0
0 0 0 1	$0+0+0+1/12$	$1/12$
0 0 1 0	$0+0+2/12+0$	$2/12$
0 0 1 1	$0+0+2/12+1/12$	$3/12$

------		
1 1 1 1	$8/12+4/12+2/12+1/12=15/12$	

Table 1

If I had made the "word" (collection of 1's and 0's going into the converter) 6 or 8 bits long instead of just 4, the resulting series of output voltages would still increase 1/12 volt for every unit increment of the data and the only effect would be to increase the range of the output voltage.

Unfortunately, while the distinguishing feature of an exponential oscillator is that equal incremental voltage changes will cause it to generate a series of equally tempered pitches, this is not the case for linear response oscillators. A linear oscillator requires constantly increasing voltage increments to produce equally tempered semi-tones.

While this increasing voltage requirement doesn't make the application of R/2R converters to linear oscillators impossible, it certainly makes it cumbersome.

Cumbersome because we have to make the incremental change from the converter small enough to guarantee that there will be some pattern of 1's and 0's that defines a control voltage reasonably close to what we're really after.

Very small voltage increments - there are three things "wrong" with this:

1) We're going to need a "bigger" converter - one with greater resolution and consequently greater word size. Whereas 6 bits of data will provide a little more than 5 octaves of control voltage to an exponentail oscillator; the same 5 octaves from a linear oscillator will require 12 data bits. Now, if that doesn't offend you by its notable lack of elegance, it's cost certainly should. A 12 bit D/A is going to set you back about $100.00; then you've got to put it on a pc board, add controls - front panel, etc.

2) As if to add insult to injury, there will be lots of combinations of bits that represent the intervals between adjacent semi-tones, but notice that they are not equally tempered intervals and therefore next to useless even for micro-tonal tunings. We're paying out our hard earned bucks for words that we're never going to use, but must have to fill up the "cracks".

3) We've turned the determination of what data to output from a relatively simple matter of counting the keys and using that as the data into a process that at best is going to require a look-up table (where the machine says "Aha - key number 12, that's note 0001110010100001") or some such similar computer calisthenics. Not particularly complicated, perhaps, but why bother with it if we don't have to.

And that, friends, is the point of all this. We don't have to. For the simple reason that an R/2R ladder converter is not the only kind that we have to work with. There are other kinds. One of the other kinds is called a Multiplying D/A (or just MD/A, I guess).

While the most important operational feature of the R/2R ladder converters was that it added things to arrive at the output, the dominant feature of an MD/A is that (you're ahead of me, right?)

IT MULTIPLIES.

Far out.

If you're up on your basic music theory, a responsive chord (if you'll pardon the expression) should be struck here. The determination of the frequency of the pitches in equally tempered tunings is itself a multiplication process. The frequency of each semi-tone in the series is greater than the frequency of the preceding semi-tone by a factor of $2^{1/12}$ - the infamous "twelfth root of two" ($2^{1/12} = \sqrt[12]{2} = 1.059$). Intuitively, it would seem that this type of D/A would be more appropriate for our purposes.

In fact, this is true. We assign weightings to the bits (starting with the LSB) according to this series:

LSB **MSB**

$$2^{1/12},\ 2^{2/12},\ 2^{4/12},\ 2^{8/12} \ldots\ldots 2^{2n/12}$$

Figure 2

Where again, n is the number of bits of data that the converter will accept.

Now, we count into this converter the same way that we did in the R/2R ladder type. Remember that bits that are "off" here are not included in the total (only now this is equivalent to multiplying by 1) and that the product that results from the condition of the data will be multiplied by some internal reference voltage.

DATA		OUTPUT
0 0 0 0	1·1·1·1·Vref	Vref
0 0 0 1	$1\cdot1\cdot1\cdot2^{1/12}\cdot$Vref	$2^{1/12}$ Vref
0 0 1 0	$1\cdot1\cdot2^{2/12}\cdot1\cdot$Vref	$2^{2/12}$ Vref
0 0 1 1	$1\cdot1\cdot2^{2/12}\cdot2^{1/12}\cdot$Vref	$2^{3/12}$ Vref
-------		
-------		
1 1 1 1	$2^{8/12}\cdot2^{4/12}\cdot2^{2/12}\cdot2^{1/12}\cdot$Vref	$2^{5/12}$ Vref

Table 2 *

* Multiplying a base number raised to various powers (exponents) is accomplished by adding the exponents. That's how a slide rule works - remember slide rules?

You may recognize this as an equally tempered series (if not you'll just have to take my word; it is). All we have to do now is design a circuit that does this.

Let's do that.

Here's a simple unity gain buffer amplifier:

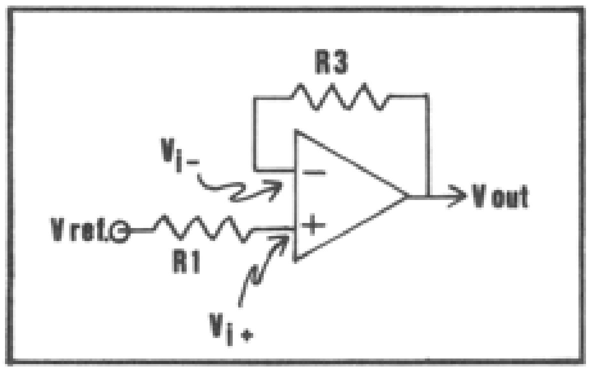


Figure 3

You may not be used to seeing it in this form because ordinarily the resistances that are shown would be replaced with direct connections. But having the resistances there doesn't matter simply because for any practical case, they are going to be much smaller than the equivalent resistance from either of the operational amplifier's inputs to ground. I should mention here that for any linear operational amplifier circuit the voltages at the inverting and non-inverting unputs are equal ($V_{i+} = V_{i-}$; this is the key to op-amp design, but that's another story). Of the circuit in figure 3 we can say:

(a) $$V_{out} = V_{ref}.$$

An excellent beginning. Here's another circuit:

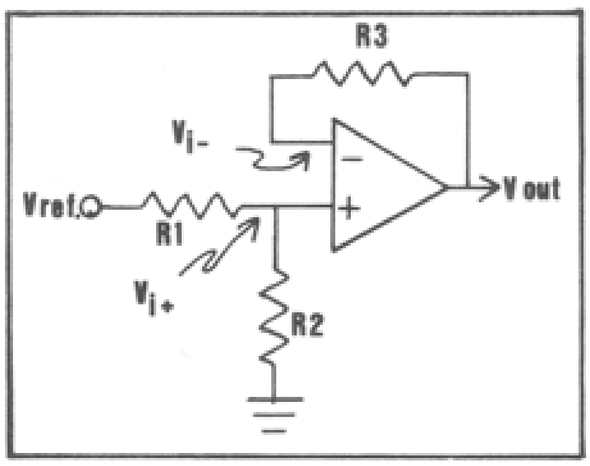


Figure 4

Adding R2 to the circuit has produced a voltage divider at the + input of the op-amp and because of this we can say:

(b) $$V_{out} = \left(\frac{R2}{R2+R1}\right) V_{ref}.$$

Fantastic. Now we change the circuit again so that it looks like this:

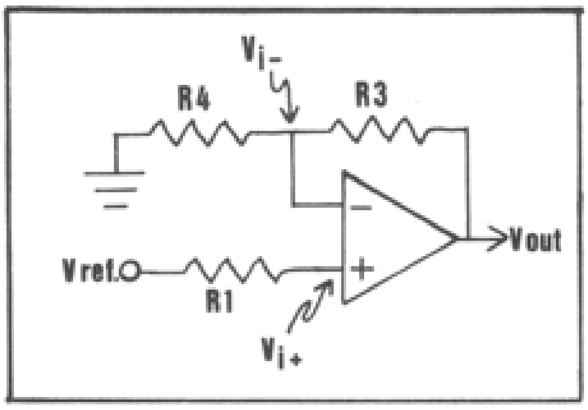


Figure 5

and instead of a voltage divider at the + input we now have one at the - input. This means that:

$$\text{(c)} \quad V_{out} = \left(\frac{R3+R4}{R4}\right) V_{ref.}$$

All together now:

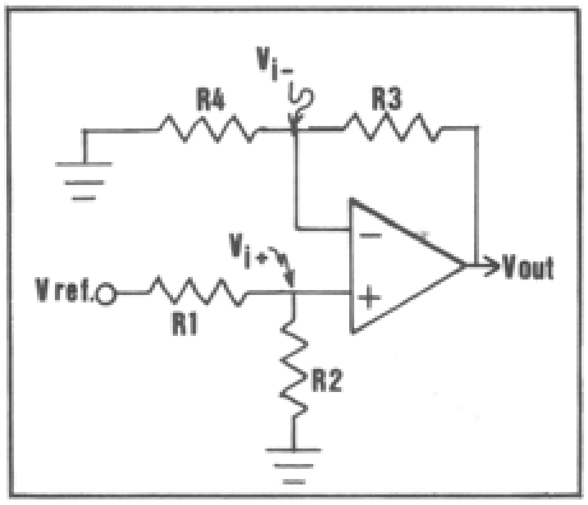


Figure 6

And for this configuration:

$$\text{(d)} \quad V_{out} = \left(\frac{R2}{R1+R2}\right)\left(\frac{R3+R4}{R4}\right) V_{ref.}$$

Do the four equations from (a) - (d) look familiar? No? Look back at Table 2. Now do they look familiar? Still no? Then let's say:

$$\text{(e)} \quad \frac{R2}{R1+R2} = 2^{1/12}; \quad \frac{R3+R4}{R4} = 2^{2/12}$$

and then by making these substitutions and putting the equations together:

Ⓐ

$$\text{(a)} \quad V_{out} = V_{ref.}$$

$$\text{(b)} \quad V_{out} = 2^{1/12} \cdot V_{ref.}$$

$$\text{(c)} \quad V_{out} = 2^{2/12} \cdot V_{ref.}$$

$$\text{(d)} \quad V_{out} = 2^{1/12} \cdot 2^{2/12} \cdot V_{ref.} = 2^{3/12} V_{ref.}$$

Now you must certainly recognize them - it's the same series as the first four entries in Table 2. Putting the resistors R2 and R4 into the circuit and removing them is simply a matter of putting switches (either mechanical or electronic) in series with them and when we do the whole circuit looks like this:

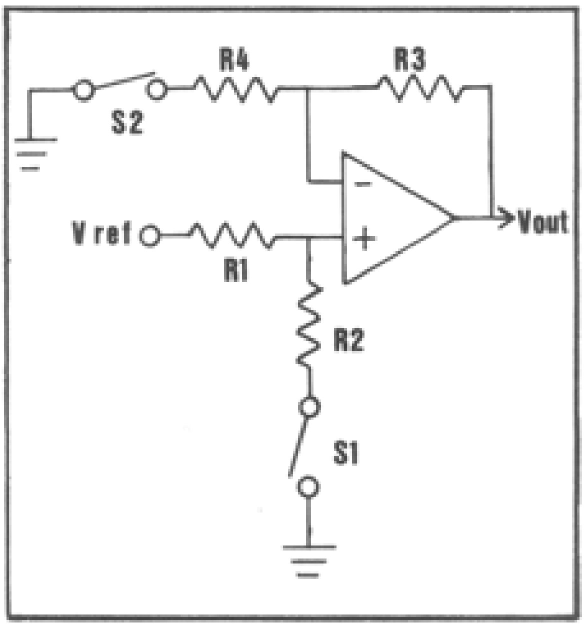


Figure 7

The switches S1 and S2 here are, respectively, the Least Significant and Most Significant data inputs to the converter; and I will avoid the obvious comment about this being a 2 bit D/A.

Oh, but there's one thing that I forgot to tell you:

$$\text{(f)} \quad 2^{1/12} \neq \frac{R2}{R1+R2}$$

Why? Because $2^{1/12}$ is a number greater than 1 and the only way that the ratio of a number to itself plus something else can be greater than 1 is if the something else is negative - which in our case, it's not (yes, there is such a thing as negative resistance, but the concept is not applicable here).

Happily, we have an alternative to negative resistance and that is to make:

$$\text{(g)} \quad \frac{R2}{R1+R2} = 2^{-1/12}$$

Making the exponent negative is equivalent to taking the reciprocal of the number.

At this point I'm afraid that in the interest of brevity I must make a gigantic leap and say that -- because we're using the reciprocal of the weighting, we must also complement the bit representing that weighting. In the instruction manual for this module, we will cover why. But there's not enough space to do it here. And, in any case, any of you who really want to can figure it out for yourselves. It's easy, honest.

Expanding this D/A out to handle greater word lengths is simply a matter of cascading several of these sections. When we do this and replace the mechanical switches that we had earlier with 4066 type Quad Bi-lateral CMOS switches we come up with a thing that looks like the circuit shown in figure 8.

Notice that the complemented bits that we require are indicated bv the overbar on like $\overline{D_0}$ for instance. This is read "not D_0" and by custom indicates that the low (0) state is considered to be "on".

You are probably also wondering about those R_a's, R_b's, etc. The values of these resistors are determined by solving equations (a) through (d) and they produce some strange values that need to be exact. 5%'ers won't get it here. In order to meet the necessary precision and stability requirements, we've had "one of the nation's leading resistor manufacturers" (at least that's what they say) make up some custom Cermet resistor networks. They look about like any 16 pin DIP IC (except that they're a beautiful robin's egg blue), but inside are resistors instead of other stuff. Once manufactured, they're trimmed by LASER to be exactly the right ratios (Laser, yet - how about that!).

I really don't expect that to impress you too much, but this should:

THERE ARE NO ADJUSTMENTS TO THIS MODULE

You just put it together and it "plays" (which is the computer people's phrase for works).

Do you realize that this gets rid of all those trimmers from our old '-8 keyboard - it even gets rid of the zero pot. I really like it.

But we're really not through yet, we need to completely dress the design by adding input latches (so that the input information can be stored), and some kind of indicators so that we will know what's going on (LED's - they wink, they blink, they twinkle like stars in the night; anybody can look at this thing and know that it's got something to do with computers). This part of the circuit is shown in figure 9.

The 4042's are the latches and one of their features is that they have both Q and $\overline{Q}$ (the complement of Q) outputs- since we needed some complemented data bits, this is nice. Q9 - Q14 are level converters. We need to have the "on" resistance of the 4066 switches in the converter circuitry working at as high a supply voltage as possible in order to achieve predictable low "on" resistances and this means that they operate from the +9v. synthesizer supply rather than the +5v. logic supply.

That's the design. Let's take a few minutes to review what we've got here.

continued . . . . . page 21. . . . . .

We've got a new synthesizer module that does at least one thing that many people thought couldn't be done: a 6 bit Digital to Analog converter that will provide slightly more than 5 octaves of equally tempered control voltage to linear response voltage controlled synthesizer elements.

The front panel PITCH control allows the module's output to be chromatically transposed over another octave, so the total range of output voltage available is a little more than 6 octaves (compared to typicly 4 octaves for a #4782 keyboard).

We have two trigger flags available, either of which can be set or re-set independently (very handy). As we will see in a future issue, these flags can also be used to select micro-tonal intervals.

The status of the 8 bits of data coming into the module is displayed on the front panel LED's, six of which indicate the note that the module is producing and two of which indicate the status of the trigger flags.

To make the module easy to interface to anyone's computer (or simply keyboard encoders - see LAB NOTES) we have an input terminal marked $\overline{RDY}$ (not ready). When this terminal is grounded, the latches that are provided on the data lines are in a "pass" condition and any changes of the data on the data input lines will be reflected as changes in the module's output voltage. When the $\overline{RDY}$ line is taken to a high logic state, the last data that appeared on the input lines is stored in the latches and further changes on the data bus will not produce any change in the output voltage (this is about like the action of the SAMPLE inputs of clockable sample and holds).

The road to applying the processing and control power of the computer to electronic music synthesis is not a short one - but it is certainly a trip worth taking. The Equally Tempered D/A is only a first step.

As first steps go, though, this is a good one - like walking in seven league boots.

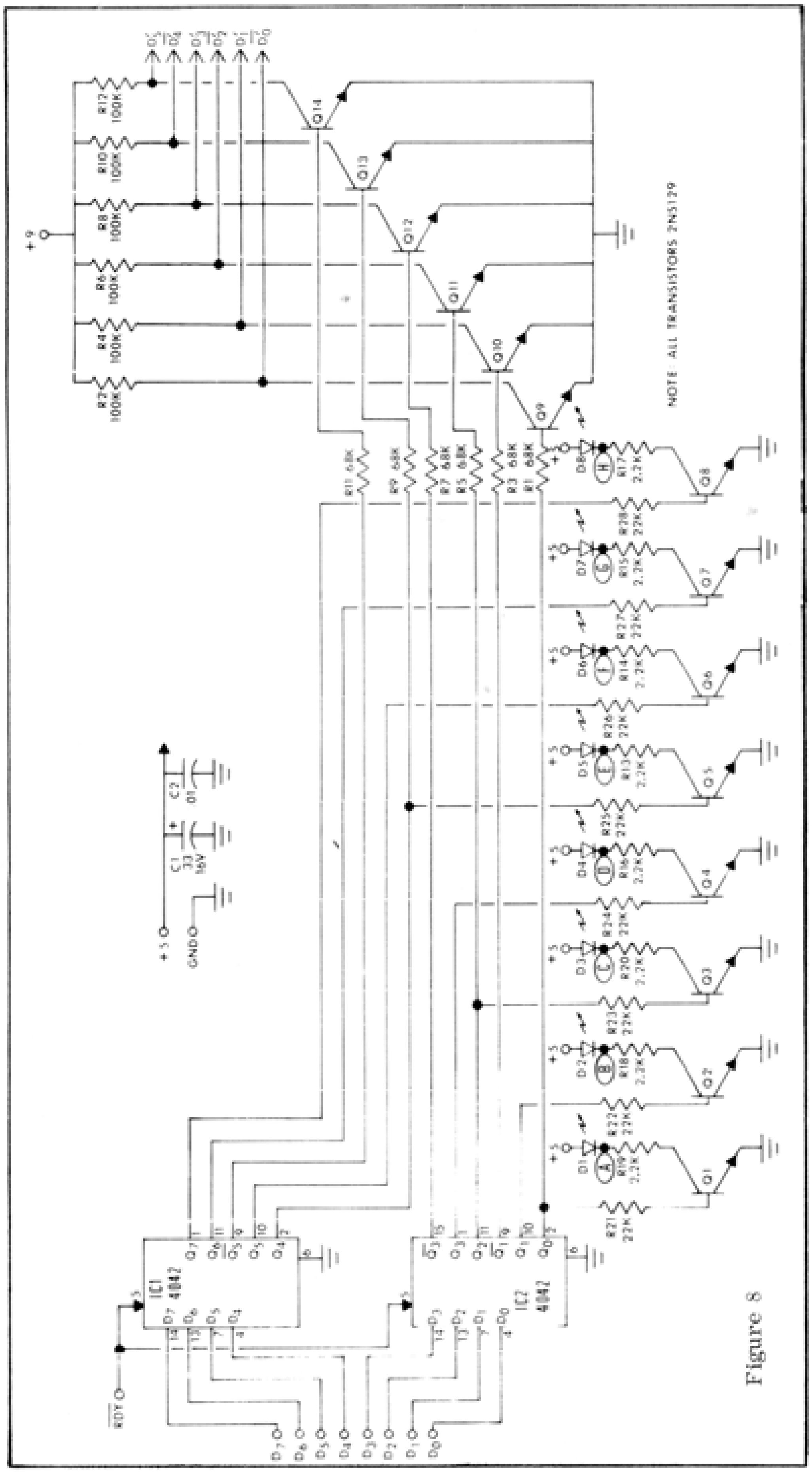


Figure 8

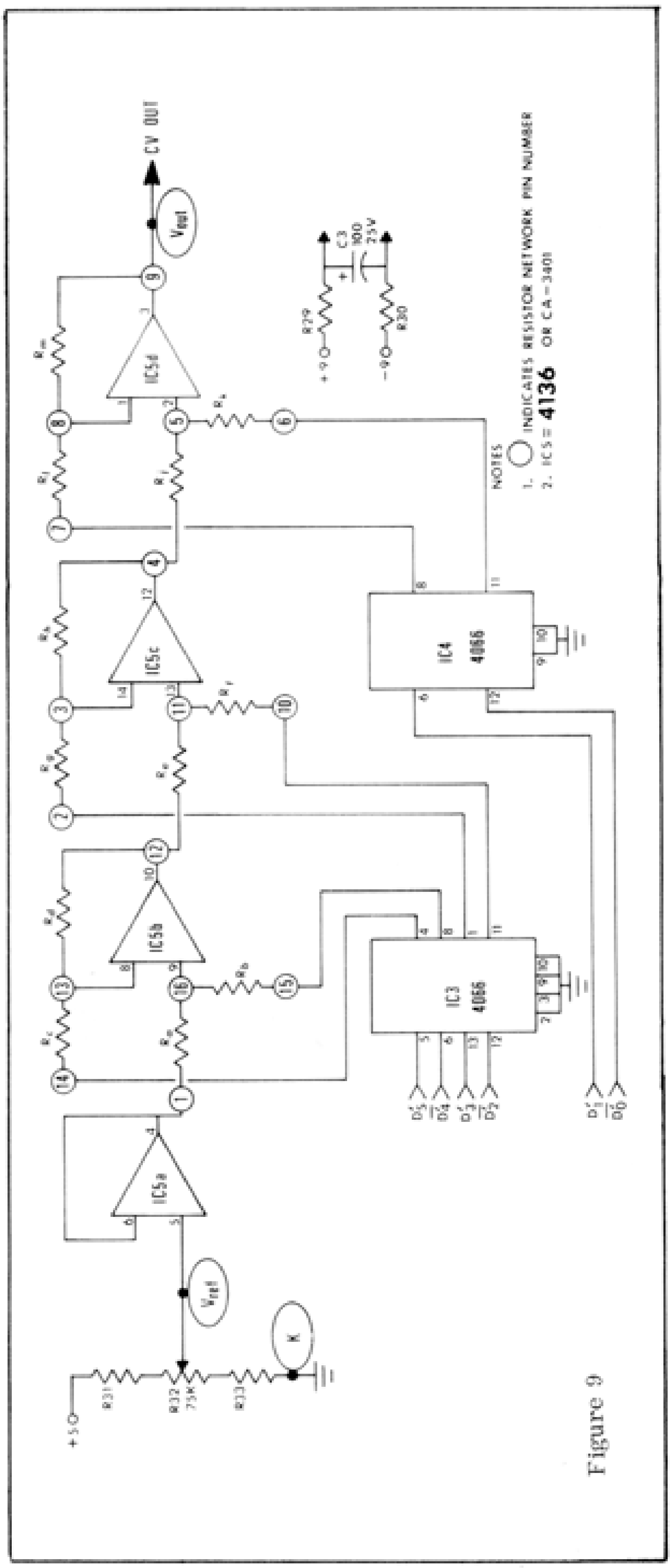


Figure 9

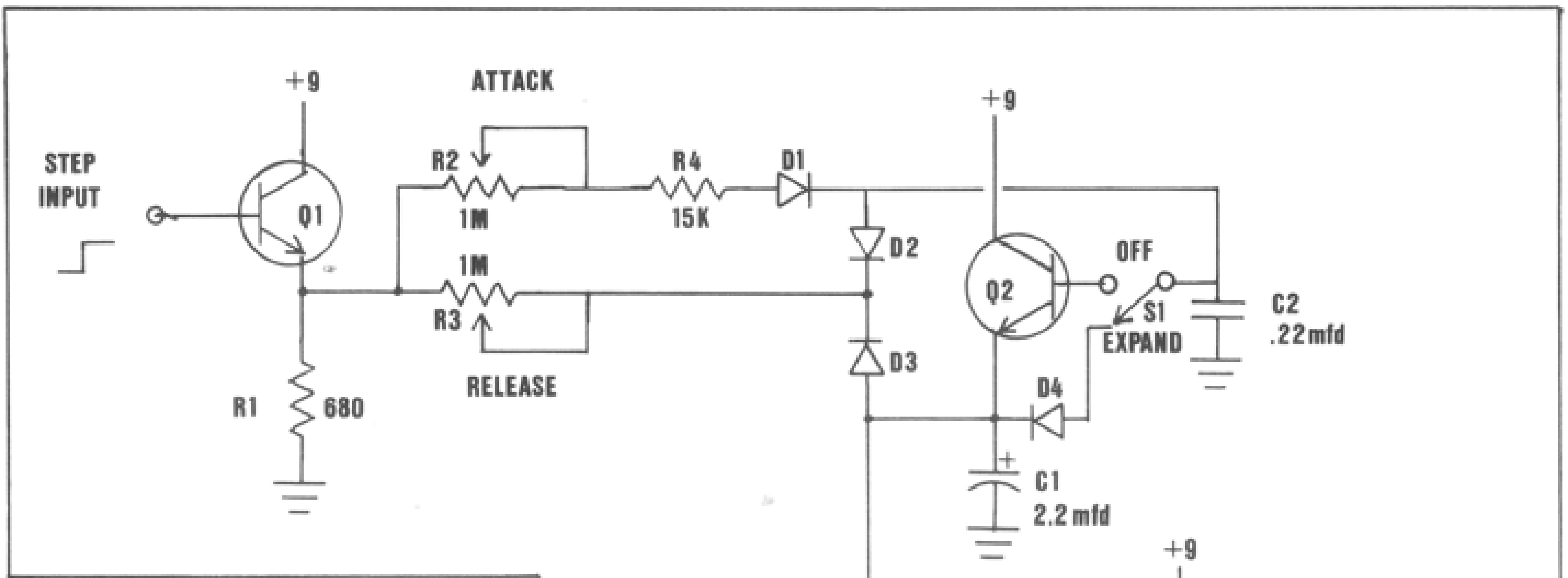
A LOW COST, SPECIAL PURPOSE AR GENERATOR

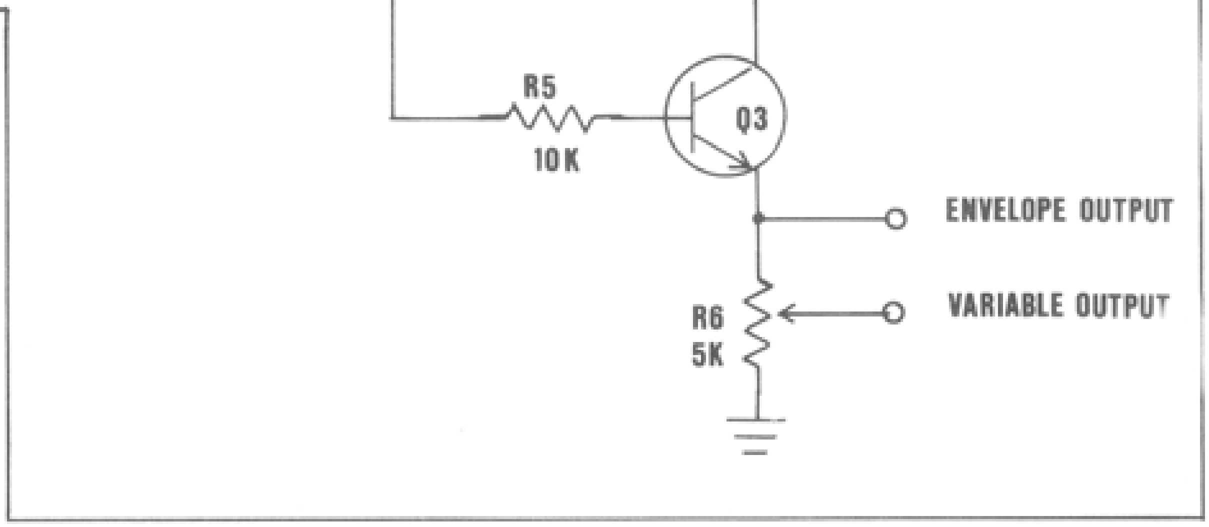
By Harry Woodell, Springfield, VA

Being a poor student, I have added several "homebrew" circuits to my 2720 in an attempt to derive as much versatility as possible for minimum cost. One of my most used units is a stripped down version of the 2720-4 Envelope Generator. The 2720 User's manual showed how to generate an ADSR envelope using two 2720-4's. However, since one is triggered by the keyboard step trigger output, the input flip-flop and UJT reset circuitry are not needed. Thus the circuit shown will replace the standard 2720-4 in this application. This circuit can also be used wherever a step or gate trigger is available and only a simple AR envelope is required: sequencers, envelope followers, or even the repeating trigger source shown on page 6 of issue #1/76 of Polyphony.

The circuit, as shown below, is quite straightforward and uses common parts. All transistors are 2N2712, all diodes are 1N914 or similar. C2 should be a mylar capacitor, and C1 will be electrolytic.

-30-





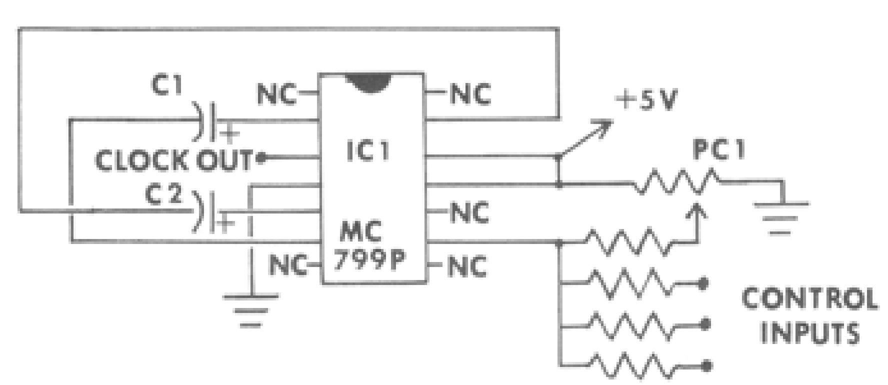
PHLANGER UPDATE

All models of the PAIA PHLANGER currently being shipped are the new 1500A. This model incorportates several minor design changes to eliminate a small amount of circuit noise which went undetected in our early prototypes. Those people who have the earlier model 1500 can receive a sheet detailing the modification procedure and the few substittute parts required. For further information, contact: PAIA Tech Service, 1020 W. Wilshire Blvd., Oklahoma City, OK 73116 or phone (405) 843-9626.

We apologize for any inconvenience this may cause.

CORRECTION

It seems that John Mitchell included a few errors in the schematic he sent us for his Voltage Controlled Clock (3/76 Polyphony). The correct schematic for the clock is shown to the right. Also, the unlabeled terminal above the 7404 in the original article should have been labeled "From clock out terminal of IC1".



VOLTAGE CONTROLLED CLOCK SCHEMATIC

LAB NOTES

COMPUTER MUSIC, WITHOUT THE COMPUTER.
or: What to do 'til your processor arrives.

By: John S. Simonton, Jr.

I realize that a lot of you will respond to the introduction of the 8780 Equally Tempered D/A with a frustrated, "But, I don't HAVE a computer."

Here's a little surprise. You don't really need a computer to do some very interesting and useful things with the 8780. You are going to need some additional hardware, as we'll see in a moment, but it's not only inexpensive, it's also equipment that you'll need for processor interfacing later on anyway. You're not building something that will be scrapped when your computer arrives, just getting a head start. Getting READY:_, so to speak.

Let's shift our mental gears for a minute, and instead of thinking of the 8780 as a computer peripheral, we'll consider it in terms of being a digital sample and hold.

Our analog S/H circuits are acceptable, but they will always drift because they store information by charging a capacitor. Even if we were able to miraculously devise a capacitor with no leakage, we still have to measure the charge on the capacitor; and whatever circuit we use to do that will itself eventually drain away all the charge (I think that a Mr. Heisenberg had something to say about this, but I'm not certain). With a digital S/H, we don't have that problem, because we're storing the information as a pattern of 1's and 0's.

To use our new digital S/H we need some way to provide it with the 1's and 0's it needs to decide what voltage to produce. We need some way to "encode" our AGO keyboards.

There are lots of ways to do this, including the simple expedient shown in figure 1.

This is frequently referred to as a "brute force" encoder. When a switch closes, any diode connected to the switch line forward biases, causing a 1 to appear on the data line connected to it. The diodes are there in the first place to prevent "sneak" current paths back through the matrix. This is an acceptable encoder as long as you assume that only one key is going to be down at a time. But, when two keys are pressed

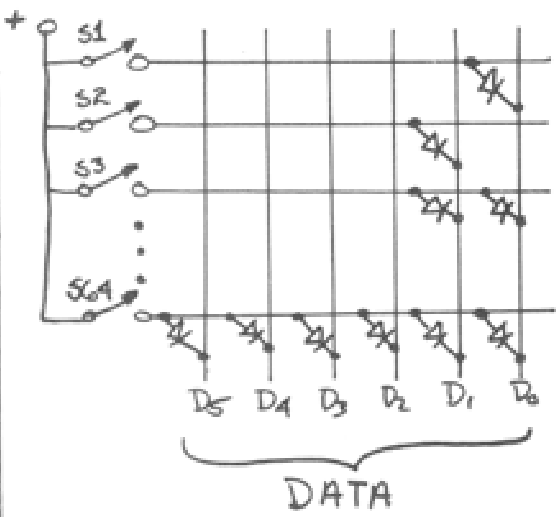


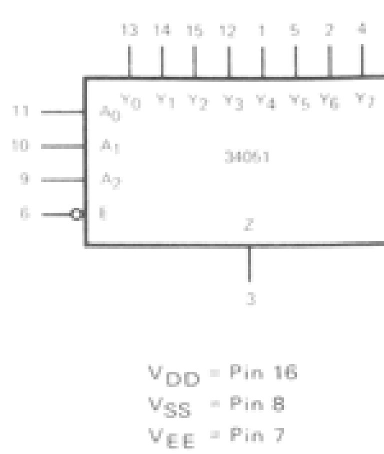
Figure 1 - Brute Force Encoder

simultaneously, the diodes act like OR gates and the data that comes out may or may not (most probably not) represent those keys. If, for example, we were to press the first two keys down at the same time, data lines D_0 and D_1 would both go high. Exactly the same situation that we had defined in figure 1 as being an indication that key 3 was down:

BUMMER

A more popular approach (because it works better) is to "scan" the keyboard a switch at a time to see if any are closed. There are LSI chips that do this with a single integrated circuit package; but, while saving design time is a great temptation, we're not going to use them. They're too expensive, and worse yet, not versatile enough to do all the things that I have in mind.

So that you can follow the design that I prefer, let me turn you on to a new part:



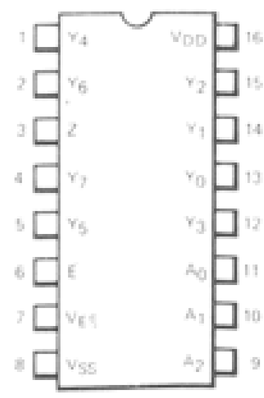


Figure 2

This is called a "4051 8 channel analog multiplexer/demultiplexer". Or, just 4051. Inside the package are 8 bilateral CMOS switches. While one side of each of these switches is tied to one of the pins Y_0 - Y_7, the other side of all the switches are commoned and connect to pin Z. In mechanical terms, it looks like this:

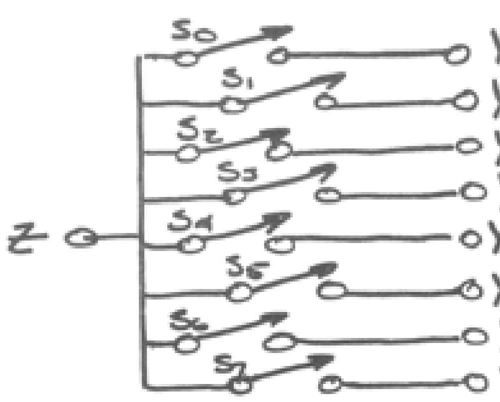


Figure 3 -
Mechanical Equivalent of 4051

One of the neater things about the 4051 is that each of those switches is individually "addressable" from the pins marked A0 - A2. If I put the binary number 000 into the address pins, switch S_0 will "close". 001 causes switch S_1 to be activated, and so on to 111 which addresses S_7.

You will also notice a pin labeled E. This is an enable pin that sort of says "GO" to the rest of the circuitry in the package. As long as this pin is held at a high voltage, all of the switches will be "off", but when the E pin is grounded, the switch specified by the address currently on the A pins will close.

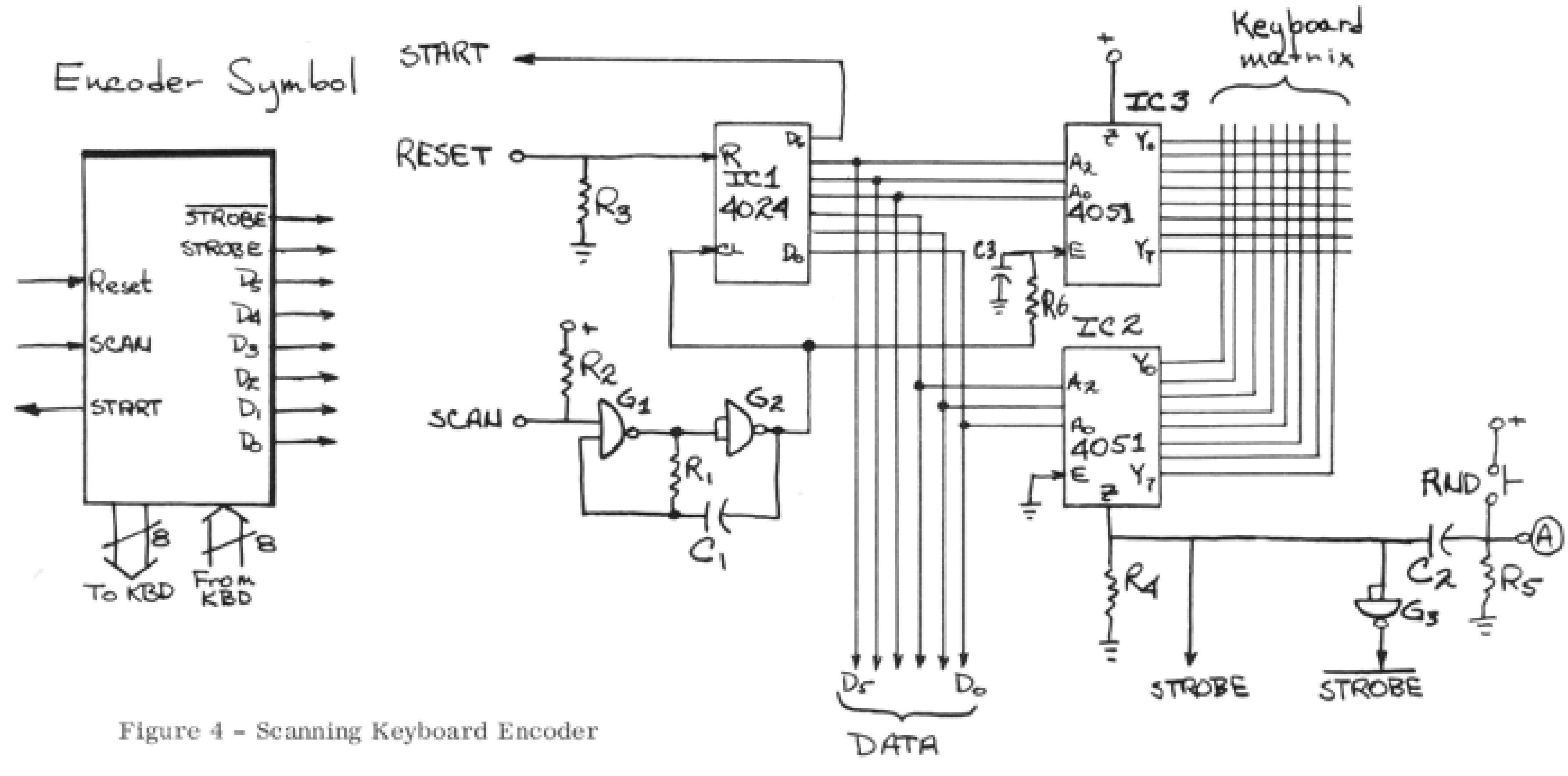


Figure 4 - Scanning Keyboard Encoder

What a terrific part. We really need to spend some time soon looking at all of the potential applications for this device. Not today, though. Today we have too many other things to do.

You're already familiar with the 4024 CMOS seven stage divider, we've used it before in other applications. Now we're going to use it again in a circuit that looks like figure 4.

This is our keyboard encoder. As far as parts go, there's not a lot to it. But it does a lot, watch.

Gates G1 and G2 along with R1 and C1 form an astable clock that feeds the seven bit counter IC1. Notice that I can start and stop the clock by raising or lowering, respectively, the line labeled SCAN. If I'm not using this line, I can simply leave it disconnected and the pull up resistor R2 will keep the clock running.

Notice that the three LSB's from the counter (D0 - D2) are connected to the address pins of IC2 while the next three MSB's connect to the address pins on IC3 (we are going to temporarily forget about the seventh bit). Assuming that the counter starts counting at 000000, both IC2 and IC3's Z pins are connected to their Y0 pins. If these two Y0 lines are isolated from one another another nothing happens; but, if they are shorted together (by a switch at the point at which they cross in the matrix, for instance) then a current flows from the Z pin of IC3 to the Z pin of IC2 through R4 which is tied to the ground. The resulting voltage rise across R4 appears on the line labeled STROBE as a logical 1, which we can interpret as an indicator that a key is down.

When the clock cycles and the counter advances to 000001, it has no effect on IC3, but IC2's Z pin is now connected to it's Y1 pin. If those points in the matrix are isolated - nothing; if they're connected, we get a 1 on the strobe line. As you can see, each clock cycle advances the counter, which will have the effect of looking at each cross point in the matrix, one at a time. A STROBE results if the cross points are connected.

At any instant in time, the six bit number appearing on the data line is the number of the key being examined - in binary, and the status of the STROBE line will tell us whether that key is up or down.

It will also be handy at times to have a line that goes low when a down key is found, so G3 is used as an inverter to provide the complement of STROBE - $\overline{\text{STROBE}}$. (I'm tempted to say son-of-strobe, but actually NOT strobe.)

One subtle point about the 4051's that we overlooked above: the line from the clock also connects to the E pin of IC3. The effect of this is to allow a STROBE to occur only during negative half-cycles of the clock (immediately after the counter changes state) like this:

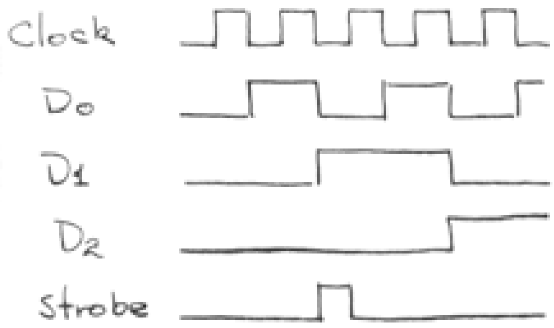


Figure 5 - Encoder Timing

which assumes that key 000010 is down. This, as well as the lag network R6 and C3, is done for timing considerations.

Also, getting back to the counter again for a moment, we have a reset available; and while I can't think of a use for it right now, one may come up later. I bring it out on a line with a pull down resistor, R3, and label the line RESET. Raising this line to a 1 will reset the counter. Also, that seventh bit that we conveniently forgot, we can now bring out on a line labeled START. In computer application this line will serve as an indication that a scan is just starting or ending.

So, that's our all-purpose, super-gee-whiz keyboard encoder. In all of the drawings, I've shown it operating from a 5 volt supply because in computer applications we're going to be tapping power from the processor; but we're using CMOS logic here and the big re... is that it likes all different kinds of supply voltages - anywhere from 3 to 15 volts. If we retro-fit this stuff into a 4782 Road Keyboard (which as you might expect, I highly recommend) we can easily use the +9v. part of the supply that's already there to power both the encoder and the D/A.

The encoder can handle up to 64 switches (the number of cross-points in the matrix) and it will obviously work with a 5 octave keyboard. We really want to concentrate on a 37 note unit, though, since this is our standard.

No matter whose keyboards we are going to use, we are probably going to have to make some changes in the switch busses first. I'll show you on one of ours. If yours is different, I'm sure you can figure out something.

PAIA keyboards (and most others, too)

have two busses; one of which boils down to a single switch that is closed as long as any key is down. With analog S/H's, this is a signal to the circuitry to do its stuff. We don't need this anymore.

The second buss is really 37 switches, with one side of each switch tied to a common connection. We could represent it like this:

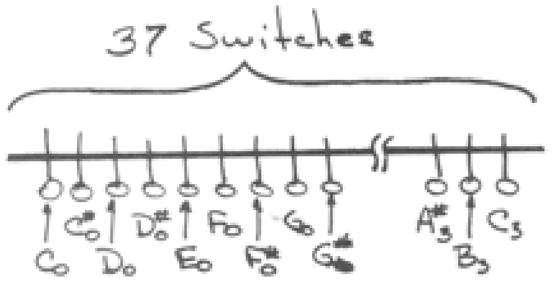


Figure 6 - An Unmodified KBD

The switch contacts that are not commoned would ordinarily go to the voltage divider board in an analog system.

We need to break these switches 'own into groups of 8 (giving us 4 such groups with a group of 5 keys left over) by cutting small sections (about 1/8 inch or so) out of the buss rod that runs the length of the keyboard. When you do this, don't forget that you have the keyboard upside down. Be sure that the first cut is between the first G and G# on the keyboard. I ran into structural problems after cutting the buss rod; one section of it was supported at only a single point. An easy fix for this problem was to slip short sections of clear tubing (spagetti) over the adjacent ends of the cuts, providing both insulation of the buss section and mechanical rigidity. When you're finished, what you have could be represented by figure 7.

Now we buss together the individual key switches from each group by connecting together all of the first keys in each group, all the second keys in each group, etc. Notice that again to prevent sneak current paths which could generate "phantom" keys if multiple switches were closed, we've added a diode in series with each key. When we're done, we have what's shown in figure 8.

If we now redraw what we've got and superimpose it on the matrix, we have what's shown in figure 9.

You probably noticed that the first key does not begin at note 000000, but rather picks up from row 2 of the matrix; equivalent to making it key number 010000 from the encoder's standpoint, and transposing the keyboard 16 semi-tones up-scale from the D/A's point of view.

IT DOESN'T MATTER WHERE THE FIRST KEY STARTS.

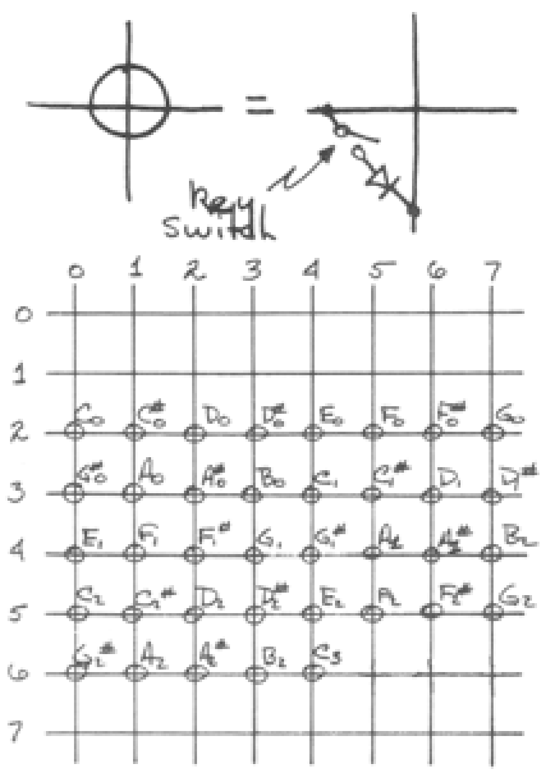


Figure 9 - The Keyboard As Matrixed

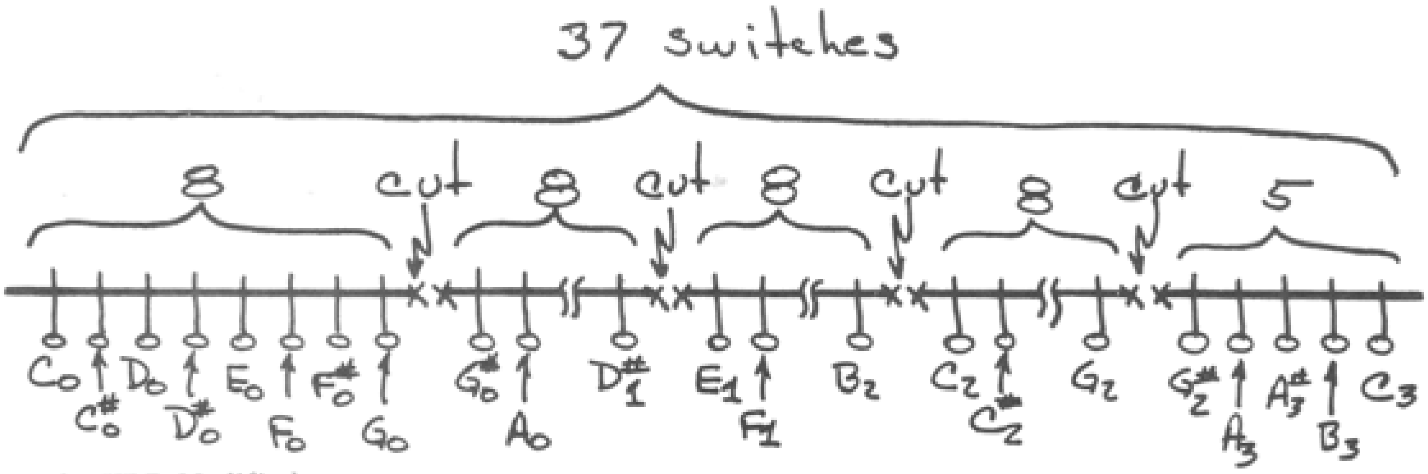


Figure 7 - The KBD Modified

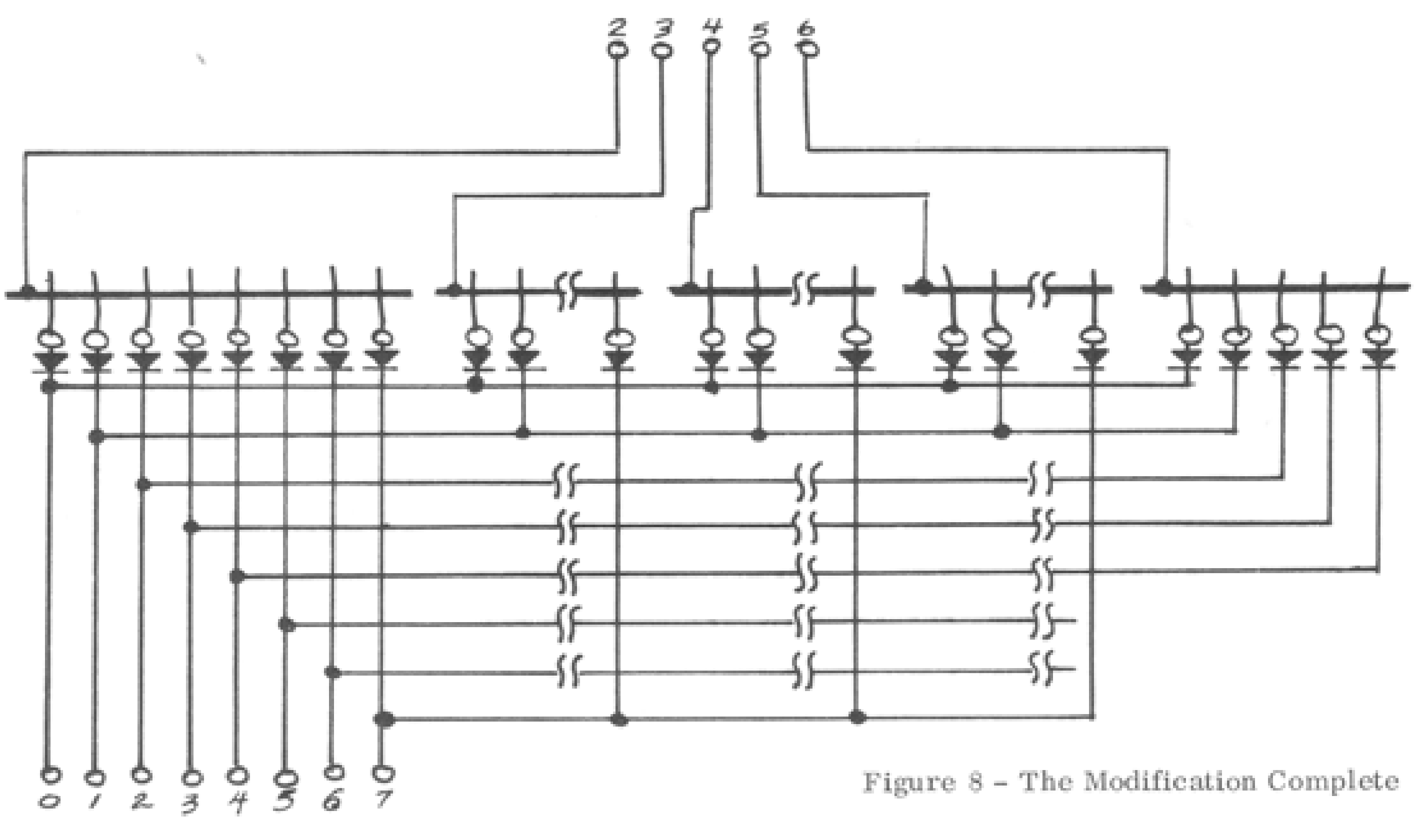


Figure 8 - The Modification Complete

Between the pitch knobs on our oscillators and the one on the D/A, we will be able to "put" the oscillator in any pitch range we want anyway.

There are a couple of good reasons for starting with key number 010000: First, I have a few computer things in mind for keys 000000 through 000111, and I want to hold them in reserve. Also, one of the things that our computer is eventually going to do for us is take care of transpositions into a new key signature, which will simply be a matter of adding to, or subtracting from, the note data the number of semi-tones by which we want to transpose. If my first key is 000000, I'm going to have a hard time transposing it down scale.

Now that I have the keyboard connected to the encoder, I'm ready to start doing things. Like replacing my old analog S/H with this shiny new digital model. There are lots of ways that I can do this. One is shown in figure 10.

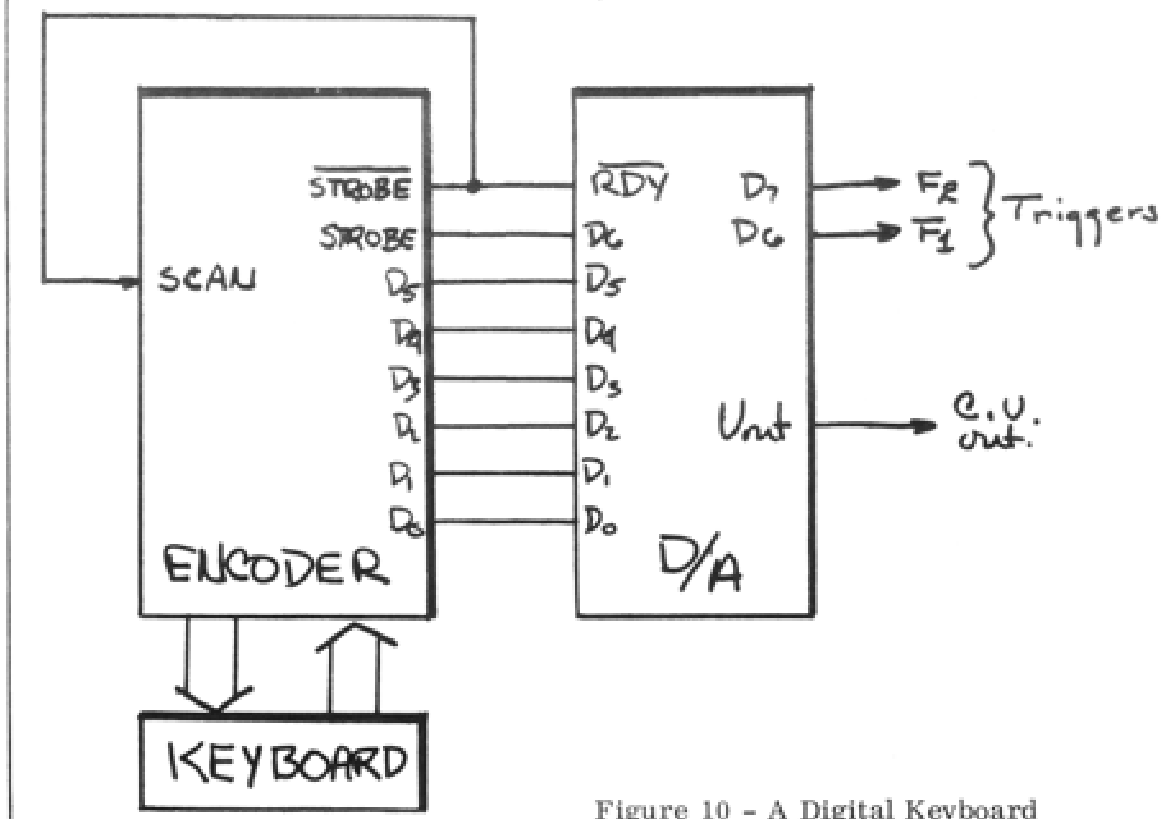


Figure 10 - A Digital Keyboard

Assuming that no keys are down, the encoder's STROBE line is at a 0 and $\overline{\text{STROBE}}$ is at a 1, making the $\overline{\text{RDY}}$ on the D/A high. The 8780's input latches are in a holding state and the activity on the data lines D_0-D_6 is invisible to the converter. This is fortunate, since the data lines are "counting" as the encoder continually looks at the keyboard.

Now, we push down a key. For the purpose of illustration let's say that it's the first key, number 010000. When the data lines next reach the state 010000, the encoder finds that that key is down, and because of that, the $\overline{\text{STROBE}}$ line goes low which both stops the encoder's clock and takes the D/A's $\overline{\text{RDY}}$ line to a 0 putting the D/A's latches in a pass state. The new note data (010000) is strobed into the converter and a control voltage representing that key appears at the control voltage output of the D/A. The STROBE line from the encoder also connects to the D_6 input of the D/A, which appears at the D/A output panel as the first trigger flag (F1), so we have a trigger showing that a key is down. And this trigger is used the same way we would a trigger from the analog system.

As long as the key is down, the system is going to sit in this configuration. But, when I release the key, new things happen. Almost simultaneously STROBE goes low and removes the trigger flag D_6 (which indicates that the key is now up) and $\overline{\text{STROBE}}$ goes high. The encoder's clock starts again (looking for the next key down) and the $\overline{\text{RDY}}$ line on the D/A is forced high putting the latches in a holding condition - and what they're holding is data on the last key that was down.

This is behaving exactly like the old analog system that we had, except, as I already mentioned, it doesn't drift. AND it gets rid of that annoying "in between" note that we had with the old keyboard if two notes were pressed at the same time (since the clock stops, the encoder can "see" only one down key at a time). AND, it doesn't have 37 adjustments to tune it; now there are none.

Let me show you something that this keyboard can do that our others can't.

Suppose that we remove the wire connection between the encoder's $\overline{\text{STROBE}}$ output and SCAN input. You will remember that this was the thing that caused the clock to stop when a key was found down. If we replace the wire with a capacitor, say about .22 mfd. or so, we have generated a little time delay in this loop. The clock will stop when a key is found down, but only temporarily - until the capacitor discharges - then it is going to go looking for the next key down. If, in the process of searching, the encoder finds another key down, it will strobe it into the latches, hold for the time delay, and then go searching again. With this arrangement, if two keys are held down, the output of the D/A will alternate between the two, and what we will hear is a trilling between these two notes. If three keys are held down, each note will be heard in turn and while this is not polytonic by any stretch of the imagination, it can certainly sound that way.

Can you imagine what the effect of pushing down a large number of keys will be? I call it the "orgasmatronic glide" but everyone here thinks that's a terrible name.

Anyway, the arpeggiation gimmick is slick and if you wish it can be left in place and bypassed with a switch when not used as shown in figure 11.

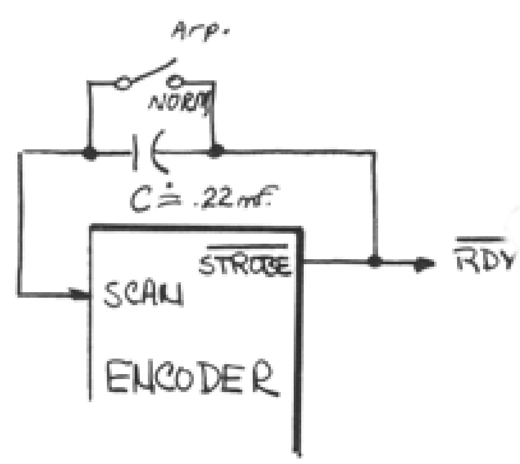


Figure 11 -
The "arpeggiation" Modification

Here's another one.

You may have noticed that there is a switch in the encoder that I hadn't mentioned; the switch labeled (innocously) RND. This switch activates the STROBE line briefly whether or not any keys are down, and that produces an interesting effect.

Every time that I press this button, whatever number happens to be on the data buss at that instant will be strobed into the D/A. Since the encoder clock is working very fast, there is no way to know in advance what the number on the data lines will be. As you've probably guessed, RND stands for RANDOM, and that is the effect of this button. It causes a random note to be strobed into the D/A. If I apply a low speed square wave clock to the point in the circuit marked A the encoder will throw out random notes at the tempo of the slow clock. We can make a suitable clock as shown in figure 12. There is an extra package of gates on the encoder board just for things like this.

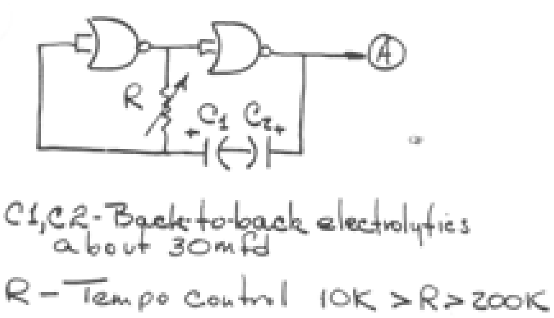


Figure 12 - A Simple Slow Clock

Out of space and out of time, again. And so much left to do. It will have to wait for next time.

Speaking of next time - here are some things that we're going to do:

We're going to look at a memory add-on for the encoder, D/A combination that will allow you to do some terrific digital sequencer things. We're also going to look at an expansion system that will convert what we've done so far into a polytonic (phonic) keyboard. Also we'll have a story on a touch keyboard - the easy way, and will look at ways that this kind of thing can be tied into our encoder, D/A set-up.

And, I think, our computer will be ready. We've put a lot of time into configuring it for maximum usefulness either as a stand-alone micro-processor trainer or for use with the music stuff. I believe that the time has been well spent. When you see all of the things that this system will do for you it's going to:

BLOW YOU AWAY

No kidding.

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.

Michael Wilson
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K2G 0Y7

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Columbus, OH 43224
(614) 267-4075

Andy Moore
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Shreveport, LA 71129

Stan Zielinski
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Trenton, NJ 08638

Steven Sande
310 Arnett
Boulder, CO 80310

Hank Jones
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Joe Lattanzi
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James Cox
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Brooklyn, NY 11225

Richard Bellnier
226 Hardenburgh Ave.
Auburn, NY 13021

Al Trautman, Jr.
824 Bank Ave.
New Iberia, LA 70560

Rick Hample
Route 11, Box 144-A
Roanoke, VA 24019

Chris Barrish
P.O. Box 113
Perry, OH 44081

Doug Maddox
1462 E. 3rd St.
Duarte, CA 91010

Michael Fenelor
646 W. 11th St.
Dubuque, IA 52001

R. Bruce Elder
Dept. of Photo Arts
Ryerson Polytech Institute
Toronto, Ontario, Canada, M4V 1S8
(Uses PAIA modules for 16mm film soundtracks & video synthesis)

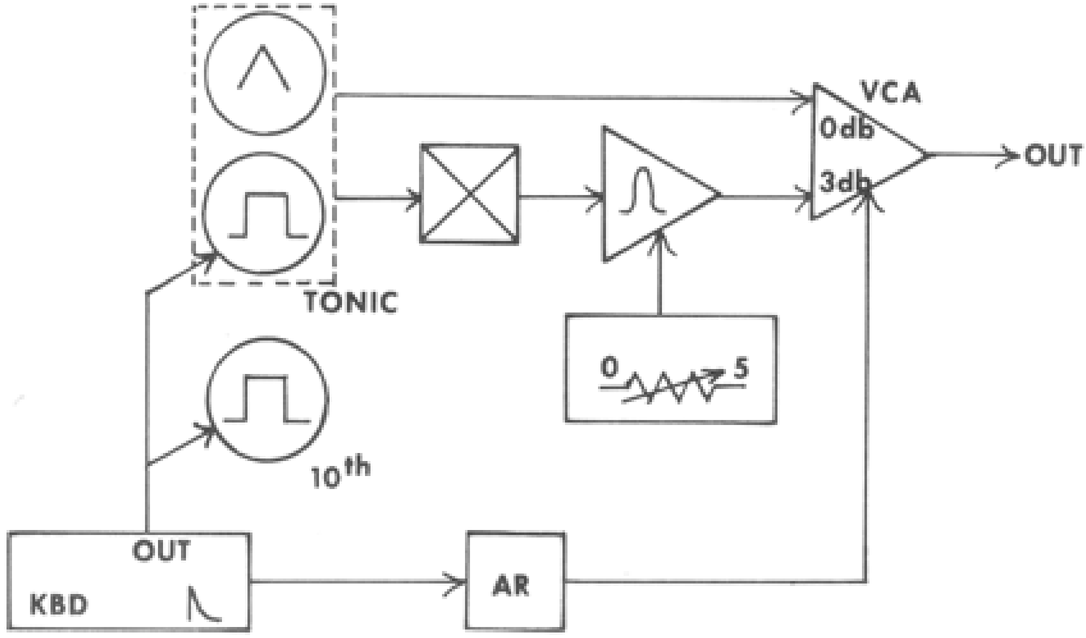
Anyone interested in a patch chart exchange, contact:
Eric Stumbaugh
45 Lacken Dr.
Watsonville, CA 95076

Peter Kaminski
P.O. Box 26
D-4619 - Bergk. - Runthe
West Germany
(Peter wishes to contact synthesists from German speaking countries)

Two of our Polyphony readers have informed us that they have started electronic music clubs. The Union College Electronic Music Club was organized by Jim McConkey. Members need not be students at the college, and Jim invites prospective members to contact him at (518) 370-6667, or write: Box 1731, Union College, Schenectady, NY 12308.

A club is in the works, being organized by Joe Lattanzi which will be exclusively for PAIA users. Charter membership is $1. 00 and Joe asks that you send a photo, list of equipment, and description of how you use it. For more info, contact Joe at P. O. Box 73, Hegins, PA 17938. Phone: (717) 682-9770.

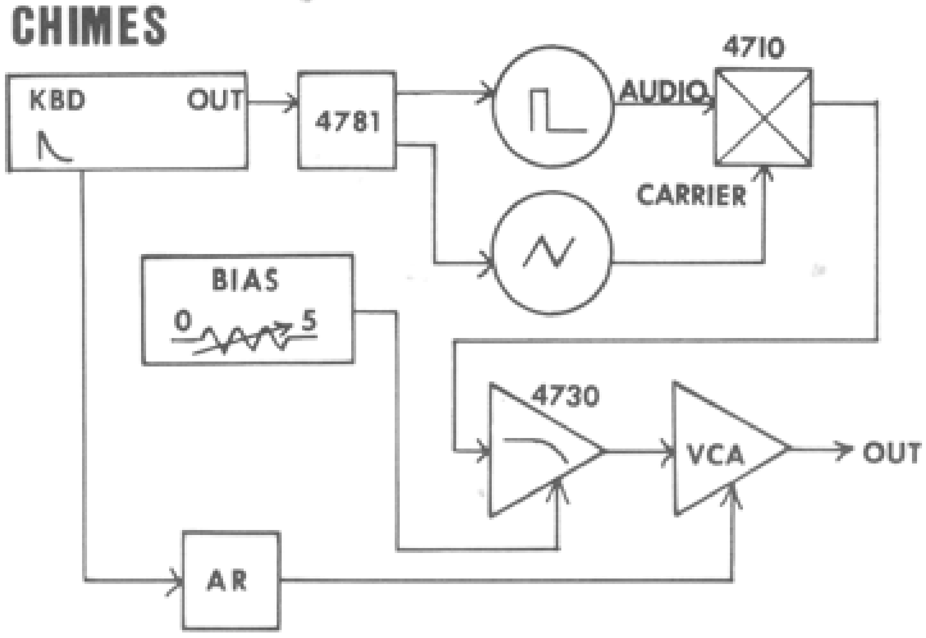
PIANO EFFECTS



VCO's:	Tune one VCO near the bottom of it's range. Tune the other to a harmonic 10th higher (slightly below the equally tempered 10th). This tuning is critical and must be tuned for minimum resonance. The usable range will be around the bottom two octaves.
Bias:	About 1.5 volts
Initial PWM:	About 50%
Envelope:	Attack - minimum
	Decay - 75%
Filter Q:	80%

Submitted by Sheila Thornton, New York, NY.

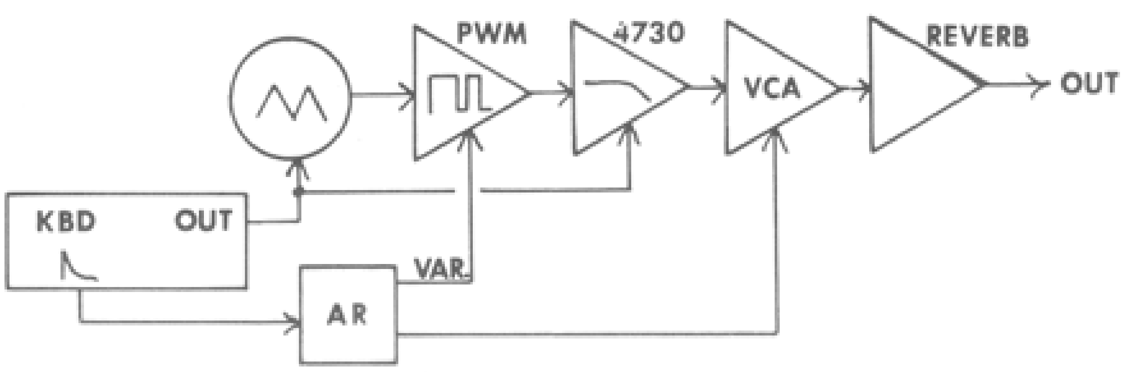
CHIMES



KBD:	High Octave, No Glide
AR:	Expand - Off
	Attack - 0%
	Decay - 75%
4730:	Range - Maximum
	Bias - 1 to 2 VDC set to just higher than the highest strong harmonic.
4781:	Tune triangle to an augmented fourth above the pulse.
Pulse VCO:	Pulse width approximately 30 to 40%.

CHOIR EFFECT

as used by Isao Tomita in Promenade of Pictures at an Exhibition

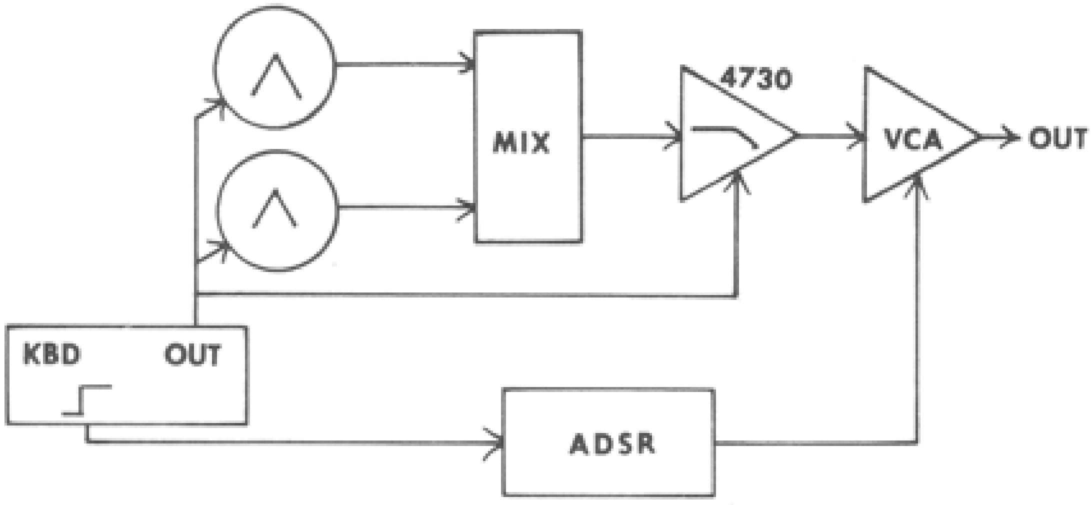


KBD:	Low Octave
4730:	Lowpass Output
	Initial Freq. - maximum
	Q Control - 80%
Envelope Generator:	Attack - 10%
	Release - 25%
	Variable Output - 75%
Reverb:	Output Mix - 80%

Submitted by Tim Fluharty, Lodi, OH.

more...

"LUCKY MAN" OR "MINOTAUR" PATCH

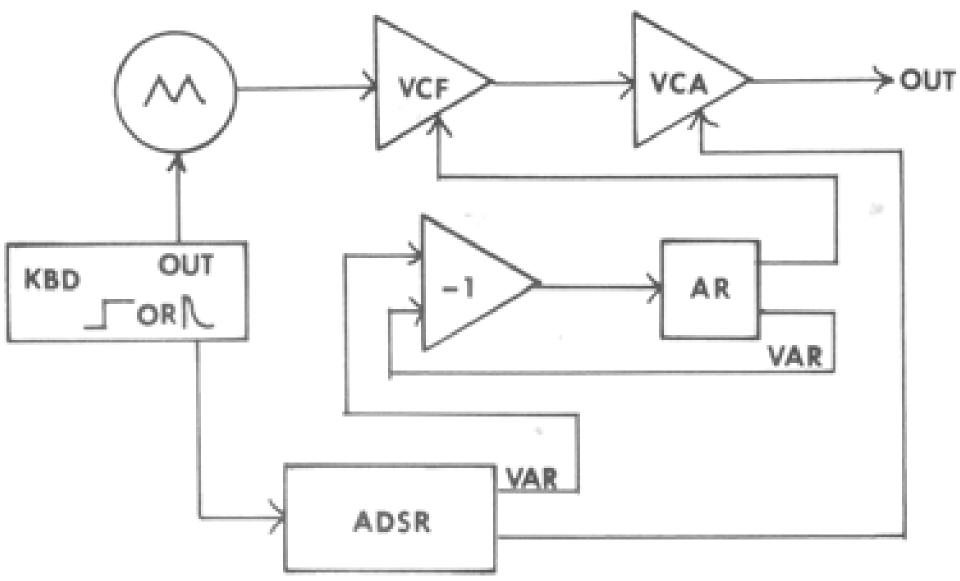


VCO's:	Tuned near unison. Triangle outputs used.
VCF:	Lowpass output used. Initial Freq. - maximum Q set at maximum.
KBD:	Any pitch range with a slight glide.
ADSR:	Set for long release and medium attack, with full sustain.

This readily recognizable synthesizer sound is perhaps the pioneering sound for synthesizers in pop music. The "hollowness" of the sound of this patch is equally applicable to Dick Hyman's "Minotaur" or Keith Emerson's "Lucky Man" or "Aquatarkus" as performed on "Welcome Back My Friends". Manually sweeping the filter's range setting can add a lot of expression to this patch. It seems to sound best when used for bass and low register work.

Submitted by Jim Riter, Richardson, TX.

RAPID REPEATING FILTER SWEEPS

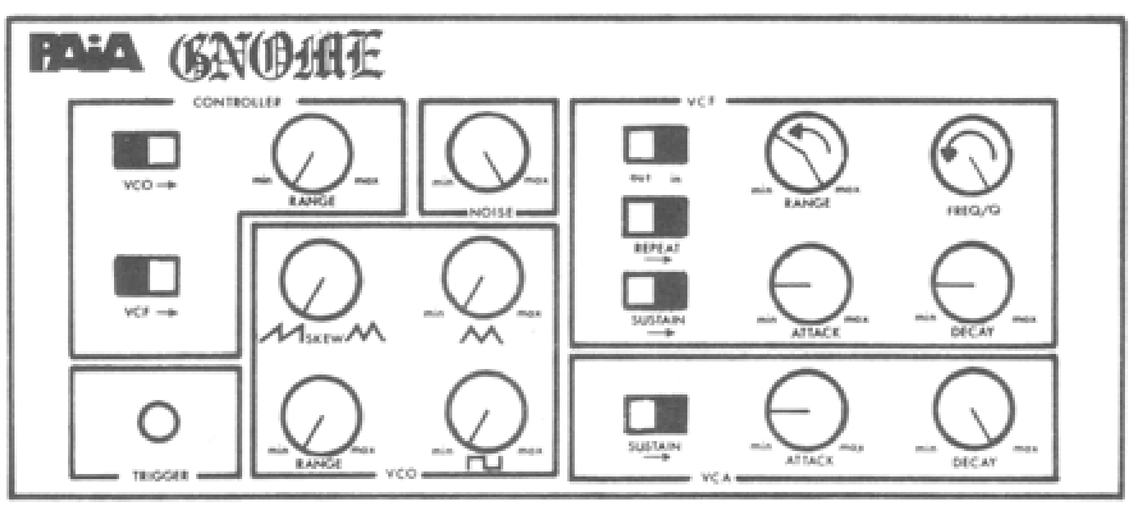


KBD:	Either pulse or step triggers may be used.
ADSR:	Attack - 100%
	Sustain - 100%
	Release - 40 to 50%
	Variable Output - 40%
AR:	Attack - 10%
	Decay - 20%
	Variable Output - 50 to 100%
VCO (or other signal source):	For best results should be harmonically rich.
Inverter:	Offset - 5 volts
Filter:	Any section may be used. Vary range and Q settings for numerous effects.

For more rapid repeating effects, decrease decay setting of AR generator. Almost every setting in this patch can be varied to produce more effects -- play with it.

Submitted by Robert Graves, Columbus, OH.

"SWOOSH" EFFECT FROM "FRANKENSTEIN"



Set VCF "attack" and "decay" for proper speed use; hit trigger. Rotate VCF "Freq/Q" slowly counter-clockwise. When fully CCW, begin rotating VCF "range" slowly CCW. When rotated about halfway, release trigger. Effect complete.

Submitted by Mike Weiblen, W. Hyattsville, MD.

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Polyphony

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Equipment Exchange

Equipment Exchange allows a place for our subscribers 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: EMS Synthi AKS with presto-patch, matrix patching and built in 256 event digital sequencer. Runs fine. Also looking for Pyschtone, JBL 075, and adapter to match an EV threaded driver to a JBL 1-inch throat. Alan Rowoth, 179 E. Genesee St., Auburn, NY 13021

FOR SALE: PAIA 2720/R with 2720/B, 4700/I, 4740 and 4710, works perfectly except keyboard needs retuning. $299. Also, a perfect GNOME for $30. Norm Nealy, 11 Rundle Ave., Mays Landing, NJ 08330, Phone: (609) 625-0138

FOR SALE: PAIA 2720/R with glide. Sample/Hold needs work. $175. Ed Arszyla, 320 Palmetto Rd., Bridgeport, CT 06606, (203) 374-3498

FOR SALE: 5 octave keyboard with case, interface, and portamento. $155. Also, dual LFO, S & H/Clock/Noise Generator, and VCA. Write for details. Joe Tardo, 817 Elmeer Ave., Metairie, LA 70005

FOR SALE: TEAC 3340 4-track recorder with sync. Includes Echo/Sound-on-Sound unit and mixdown panel. All mint condition, less than 6 mos. use. $700. Bob Graves, 4746 Nugent Dr., Columbus, OH 43220, (614) 459-0463.

FOR SALE: PAIA 4700/S, plus 4730. Sequencer needs minor repair. All for $500. Dennis J. Mauricio, 3610 Collier Ave., San Diego, CA 92116, (714) 282-2874

FOR SALE: PAIA 2720-1, -3B, -3L, -5, -7, -8, -9, -11, -12, 4710, 4711, (2)4740, 4770, 4712, (3)4720, 4761. Assembled, some perfect, some need repair. Also includes (1)4770 in kit form - $400. plus postage. Also for sale: 4760 Wing kit, never used - $10. plus shipping. Gurdy Leete, 37 Prince Royal Passage, Corte Madera, CA 94925, (415) 924-9729.

FOR SALE: 2720/R, works perfectly and to specs. $450 or best offer. Athan Pasadis, 5 Lenox, Kensington, CA 94707

FOR SALE: 4782 with -9, (2)4761 with (3)4720, 4781, (2)2720-5, (2)4740, (2) 4770, 2720-1, 4710, 4711, 4730, 4712. Slightly modified, good condition. $550 or best offer. Also one each: -2, -2A, -3L, -3B, -7, -12. Need slight repair. $5 each. Chuck McCallion, 14 Harbor Rd., Levittown, PA 19056

FOR SALE: PAIA 2720-1, -7, (3)4770, 4740, 4720 with manuals. Will sell to best prices. All working perfectly. Seth Zirin, 34-35 76th St., Jackson Hts., NY 11372, (212) 457-8649

FOR SALE: 25 used preamp boards from older solid state Fender amplifiers. Write for specs and schematics of the circuit. $2 each or $40 for all. Also, and 8 in/ 2 out mixer using these preamps: with high/low Z inputs, panpots, monitor mix, etc. $100. George Rigney, 7911 4th Place, Downey, CA 90241

Next Issue:

* MORE THAN YOU EVER WANTED TO KNOW ABOUT TOUCH SWITCHES
* MEMORY ADD-ON
* POLYTONIC EXPANDER FOR DIGITAL KEYBOARD
* A DIGITAL RANDOM NOTE GENERATOR
* MORE MODIFICATIONS
* MORE PATCHES
* MUSIC NOTATION, PART III - EXPRESSION

Programmable Drum Set

..... continued from page 9

There is still one switch key that we haven't looked at, the SAVE switch.

The Programmable Drum Set uses 2112 type memories. They're volatile, which means that if you turn off the power to them they forget everything that was programmed into them. The SAVE switch allows you to turn off power to all of the drum circuits, logic, etc., while maintaining a reduced voltage to the memory chips. It allows you to save a pattern without leaving the unit on all the time and draining the batteries. A nice touch.

INTERFACING THE GNOME

..... continued from page 14

This was not difficult, as I merely installed a regular mono phone jack with the shaft connection automatically to ground and the tip connection to where the outputs of the VCO are bussed. (Evidently, Dana's piano is capacitively coupled at the output. To be safe, you should install a coupling capacitor as described earlier. Once again, the audio bus is at TS-2, lug 3. -- ed.)

This now means that I can use the VCF and VCA for my electronic piano. Some of the effects are weird, such as when the Gnome's VCF is set for the fastest possible repeated sweep. This will give the most spectacular effects, but pleasant effects are obtained by adjustment of any of the filter controls. Also, I have found that I can add slower attacks to my piano using the VCA and the external footswitch trigger that I've made.

That's about it, but it allows for a phenomenal amount of experimenting. Just a few cents worth of jacks, and it's done. Just thought I'd pass the idea along to share with others in case they're into this type of thing.

Our thanks to Craig Anderton and Dana Lee for their contributions. We've had many requests from our readers concerning external inputs for the Gnome. Perhaps you can find the answer to your special needs here. Good luck to all you wire freaks out in Gnome Mans Land!

— ed. —