

POLYPHONY 3/76

Patches

Music Notation

Computer Drums

Construction Projects

Sound Systems for Synthesizer



Editorial

As promised, here is a brief review of what we learned from our three month flood of Polyphony questionnaires. I must say, I was surprised at the high rate of response, as nearly every questionnaire sent out was returned. Unfortunately, we weren't too clear about how we wanted you to rate new product ideas and Polyphony material. Therefore, some people used numbers twice, left spaces blank, and so on. This would throw off the statistics if I went down the line and added up the individual scores, so I will give a very generalized overview of the trends seen in reading the responses (yes, believe it or not, about four of us have read every questionnaire received).

First, concerning Polyphony, the overwhelming favorite was technical articles, including Lab Notes. A close second was new product reports. Perhaps in the future we can get other companies interested in sending us reports on their new products. I would like to see this, and I would have done it already if someone would have sent me something. We want to print anything we can get our hands on that deals with electronic music in any way. Ranking about middle was the Patches section. I was surprised that people didn't rate this higher, having thought from the beginning that this was one of the more popular sections. After that, the other sections all rated about the same: Random Noise, Spotlight, Reviews, Advertising, etc.

New product ideas weren't as clearly divided (at first, anyway). After we had received 400 or 500 replies, I went back through them to try to decide what you guys wanted. Then I saw what was happening. The dividing line wasn't between certain products, but groups, or types, of products. When reading the new product ratings in conjunction with "How you use the equipment" and "Other equipment owned", I could see that our customers are divided into musicians and electronic hobbyists. In the performance oriented group, equal weight fell on units such as string synthesizers, drum synthesizers, mixing consoles, normalized synthesizers, and so on. The electronic madmen were interested in the more obscure products: computer interfaces, human voice synthesizers, etc. If there was an overlapping area, it was the digital polyphonic keyboard/controller. I suppose the hobbyists are more interested in this for the newer and more complex technologies involved, while the musicians are most interested in how easy and versatile the technology will make their job.

In summation, you can be watching for: an increasing amount of Polyphony being devoted to technical articles, digital work (including programming), and some new products along the lines of a flanger, solid state echo units, string synthesizers, computer peripherals for music, and perhaps (but this would take a while) a modular mixing console and different types of modules for it. Polyphony will be the first place we mention most of our new products, so keep scanning our pages if you are interested in getting an early announcement about some of these things. And continue to send us your ideas for new products.

From the "Last Minute Department"- Does anyone out there have an interest in video synthesizers, or stage lighting and special effects (like lasers)? Just curious.

-Marvin Jones-

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Dear PAIA,

A column devoted to answering your questions about PAIA, electronic music or any other area which might be of interest to our readers.

Dear PAIA,

Kraftwerk (the group that recorded the song "Autobahn") has a machine that produces a synthesizer sound, but it's total configuration is the farthest thing from an average keyboard controlled synthesizer. It consists of an array of rings and plates, and when the performer wants to make a sound, he touches the rings and plates with a metal rod. I want to know if this is anything like the old-fashioned Martinot or Ondioline, and if I could do something similar with PAIA equipment. Not that they had any sounds that a PAIA synthesizer could not produce, but they had such a far-out appearance that I would like to construct one.

Lionel Cassin
Brooklyn, CT

Lionel,

The apparatus you saw Kraftwerk using is a trigger source for percussion sounds. I suspect they made the setup themselves, as they have built and modified much of the equipment they use. The metal rods the performer holds are used much like drumsticks, and have a DC trigger voltage applied to them. The array of rings and metal plates are connected to the trigger inputs of several envelope generators, sequencers, and so on. Each different trigger pad has a different patch associated with it. Thus, the percussion player can select the type of sound he wants, and trigger it at the proper time by hitting the pad with one of the drumsticks. The voltage applied by the drumstick initiates the sound in the same manner as the keyboard trigger output would work if keyboard control were being used.

A unit could easily be made that would work as theirs does. Metal tubing, rods, and plates are commonly found in hardware stores, or you may be able to find what you need in someone's junk pile or garage. A five to nine volt DC level can be applied to the sticks from a power supply or bias supply on the synthesizer. Be sure to insulate the sticks where they will be held. This will eliminate the 60 cycle hum from the performer's body, which would be superimposed on the control voltage causing possible mistriggers and so on. The trigger plates are wired to the trigger inputs of particular modules you wish to control. Shielded cable should be used with the shield being grounded at the synthesizer end but no connection at the trigger plate end. The plates can be mounted individually on old music stands, or microphone stands. If this is done, the plate should be insulated from the stand by mounting on a piece of plastic or wood, and then mounting the insulator to the stand. As I recall, Kraftwerk has all trigger plates mounted in a fixed configuration on one big fold-up stand. This would make it easier for setup at a job site, but would lack the versatility of a modular system.

I hope this spurs your creativity, Lionel. If you, or any of our other readers, DO build a setup such as this, PLEASE send us a picture of it (B & W glossy, of course) so we can print it in Polyphony.

Marvin Jones



THEATRICAL SOUND EFFECTS

(SYNTHESIZED, OF COURSE!)

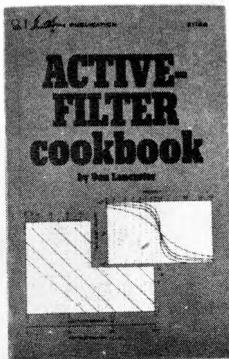
John Mitchell, from San Diego, CA, recently took time to write us about the interesting things he has been doing with his PAIA equipment. John got interested in electronic music in 1968 when he built his first Theremin. Since that time, he has built quite a few instruments for himself, as well as for other people. He also has been collecting electronic music albums, and has over 55 of them. John's equipment setup is based on an early PAIA 2720 system with the "button controller". He also has several tape recorders, miscellaneous audio processing products and a LOT of homemade equipment.

John not only uses his setup for personal enjoyment, but also synthesizes and records music and sound effects for the San Diego Junior Theater. The theater is run by professionals, and is for youths who take acting seriously. Since each production is highly polished and seen by thousands, the sounds he synthesizes must be very convincing. He said they are almost always a challenge for him.

The projects John has taken on are diverse and sound as if they were large scale productions. He has composed and produced a "dream ballet" for Tom Sawyer, synthesized an avalanche, a dragon's roar, and "magic sounds" for The Hobbit. He also provided live electronic music for I Sincerely Doubt... and Dracula. For one production John created an ocean sound complete with seagulls. John's current project is composing and realizing two electronic backgrounds for the San Diego State University Readers Theater Troupe on PAIA, Buchla, and homemade equipment. The compositions will premier in England this year.

ACTIVE FILTER COOKBOOK

by Don Lancaster



#21168 - \$14.95 - Howard Sams & Co., 4300 W. 62nd, Indianapolis, IN 46206

When you meet Don you come away with the feeling that here is a person that thinks he knows everything about everything.

You don't have to read too many of his books and articles before you begin to agree with him.

It would be difficult to do a better review of the book than Don does himself in the preface where he says:

"This book is about active filters. It is user-oriented. It tells you everything you need to know to build active filters, and does so with an absolute minimum of math or obscure theory.

If you know nothing at all about active filters and simply need a frequency-selective circuit, this book will serve you as a catalog of "ripoff" circuits that are ready for immediate use - with math ranging from none at all to one or two simple multiplications.

If you are interested in the how and why of active filters, there is more-detailed information here that lets you do more-involved design work, optimizing things to your particular needs and perhaps using a simple handheld calculator for the actual final design effort.

Finally, if you are an active-filter specialist, you will find in this text a unified and detailed base that includes both analysis and synthesis techniques that can be easily expanded on by using a computer or programmable calculator. This book should be extremely useful as a college-level active-filter course book or supplemental text."

I agree but wish that he had broadened his definition of "active" beyond just op-amps to include the serial analog delay devices that will ultimately play an important role in the design of many useful filter sections.

But other than that minor point I consider "Active Filter Cookbook" to be a definitive reference work on the current state of the art in filter design - how they work, why they work and how to make them work. If you're serious about electronic music (or just electronics generally) you should add this book to your library.

John S. Simonton, Jr.

ELECTRONOTES

"So you want technical articles, huh?"

Response to the questionnaire last issue showed a great interest in technical articles and construction projects. If you are an advanced hobbyist with a real passion for knowledge, Electronotes will fill your needs plus some. Their articles are primarily technical dissertations on results of current research being done, schematics of circuits which may be useful in synthesizers, and educational articles which teach the novice what he needs to know to be able to dig into the deeper material-circuit design, microprogramming, application and realization of theoretical material. Electronotes is becoming heavily involved in utilizing microprocessors and digital technology in electronic music. Current articles which are representative of their format include: "D/A Conversion for Equally Tempered Scales". "Design and Applications of Sample-and-Hold Modules with Slew Limiting and Cascading Features", a continuing series entitled "Back to Basics" which discusses fundamental electronic concepts, and a series entitled "Microprocessors in Electronic Music" which deals with how processors work, how to program them and what all that microprocessor lingo means. There is an average of about six schematics and construction articles per issue, and a "buy, sell, trade" section in the rear. All issues are loose sheets punched for 3-ring notebooks, and back issues since 1972 are available. Current subscription rate is around \$16, and if you write to them, I think they may send you a sample copy with subscription information.

Electronotes, B. A. Hutchins, 203 Snyder Hill Rd., Ithaca, NY 14850

M. J.

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.

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Springdale, OH 45246
(513) 851-2155

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Jim Johnson
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Seth Zirin
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Ronald Greene
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(203) 929-7077

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(919) 851-2759

Edward Zimmerman
619 W. 18th
Wellington, KS 67152

Jay Truesdale
1400 Hudson Rd.
Ferguson, MO 63135
(314) 521-0567

If YOU would like other PAIA user's to contact you, drop us a line giving permission to publish your name and address. Should anything come of these meetings such as jam sessions, lecture/demonstrations, rap sessions, etc., be sure to fill us in on the details. A little publicity couldn't hurt, could it?

Random Noise

A sharing of miscellaneous hints which might prove helpful in an electronic music studio but are not directly concerned with the Synthesis equipment.

More Patch Cord Racks

Greg Leslie, from Ponca City, OK, is still in search of the Utopian patch cord rack. (See Polyphony Vol. 1, No. 1 for Greg's first two ideas.) The first idea is a round, rotating rack mounted on a music or microphone stand. This unit can be set in the middle of a room, next to your synthesizer, and is easily portable. Cut a circle of plywood with a diameter of 10 inches or more (depending on how many cords you have), and use a protractor to lay out a number of radii (again depending on how many cords you have). Saw the lines from the outer edge of the circle to within 2 inches of the center of the circle. Mount a dowel or something similar in the center of the circle to allow the rack to be mounted on the stand. If, in the future, you need more room for cords, you can (1) cut more slots, (2) make a bigger disk, or (3) build another rack and put one on each side of the synthesizer. Drawbacks: You still have cords in front of cords, and the one you need is always behind umpteen others. Also, people tend to come up to it, scream "Wheeeeeeeee", and spin the disk as hard as they can. Bad news.

Idea #2 is the Velcro rack. It is pretty simple, but takes a while to make up. Go to the sewing department of your local department store and get several feet of black Velcro (loop and pile fastening material). Round up all your patch cords, adaptors, screwdrivers and anything else you normally keep with your synthesizer. Using the furry half of the Velcro, wrap a layer around the body of the cord plugs and cement in place with epoxy, etc. Be sure to prepare both ends of the patch cords. Fasten a length of the hook type Velcro material to wherever you want to hang your patch cords (the wall, an easel, or the synthesizer case). Just touch a cord to the rack and it will stick in place. To remove it, pull it away from the rack. This idea may be handy for transporting cords, also. If space permits, fasten two Velcro strips to the inside of the 4762 Road Keyboard Case lid. Fasten both plugs of a cord to the Velcro strips and they're always there when you need them and they won't get tangled.

Wire Markers Help Organize Connecting Cables

If you play in a band, or have a small recording studio, you probably spend a good amount of time manually tracing cables when a problem occurs. Even in a small 4-track studio, you may have a dozen cables from mikes or instruments to the mixing console, four cables feeding the recorder, four cables returning to the console from the recorder, four cables feeding the monitor amps, and speaker cables for each amp. If you have extras such as EQ, noise reduction, bi- or tri-amped monitor systems, you can see that tracing all these cables by hand would be "hassle city". Brady wire markers could save a lot of time. Identical markers should be applied to each end of the cable, and identical terminal markers can be applied next to the jack which receives the similarly numbered cable. Wire markers are available with several different labeling systems: numbers 0 through 45, letters of the alphabet, the 10 standard NEMA color codes, and special markers such as

continued on page 36...

A Time Trip-READY: _

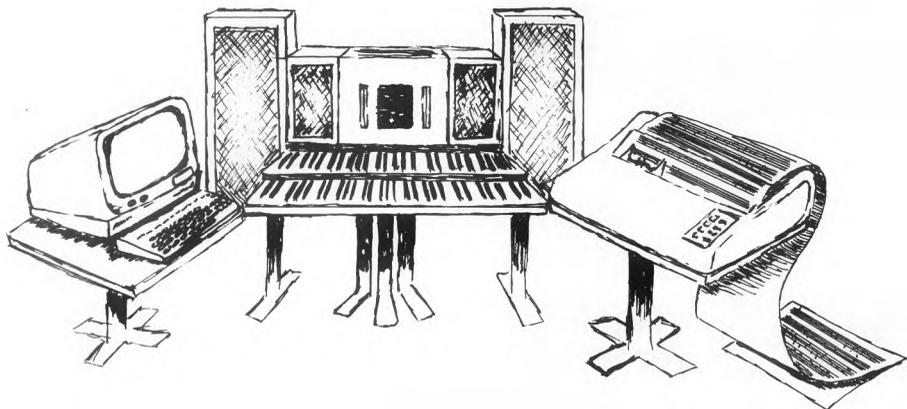
by: John S. Simonton, Jr.

I'm walking down a hall of oval cross section. Ahead and behind me the walls are colored by flowing patterns of blue, green, purple. As I pass, the sections of the walls closest to me burst into rapidly changing patterns of yellows and red. Softly, softly I hear a melodically changing pattern of notes and chords. I notice these things no more than you noticed the color of your walls when you woke up this morning.

A sign inlaid in the wall announces:

UNDERGRADUATE MUSIC LAB

and I stop. In my mind I form a picture of a section of the wall sliding up and it s. I pass through the portal and picture the door closing.



ore me is a group of cabinets that obviously is a musical instrument. It's dominant feature is two stacked AGO keyboards. Above the keyboards, a panel with two central vertical doors about 1 by 6 inches flanked on either side by sections of grille cloth. To the right of the central console is a high speed printer and to the left a second keyboard, a typewriter keyboard, and above it a video terminal. In the upper left hand corner of the screen is written this word:

READY: _

I open one of the narrow doors in the central console and insert a square cardboard jacket removed from a pocket in my assignment binder.

Sitting down, I type:

:LOAD "MUSE"

There is a barely audible click from the central console and after an instant's delay a colon appears on the screen. I type again:

```
:PLAY "TUBA", K1, OUTL, OCT1
```

```
:RUN
```

and as I touch the lower AGO keyboard fat, juicy tuba notes come plopping out to the left speaker. Nice.

I type:

```
:PLAY "TUBA", K1, OUTL, OCT1
```

```
:PLAY "STRINGS", K2, OUTR, OCT5
```

```
:RUN
```

and now I get the tuba from the lower keyboard and left speaker while the right speaks phasy strings in response to my touch on the upper keyboard. Say, this is alright.

I type:

```
:PLAY "DYNAMUTE", K1, OUTL, OCT3
```

```
:RUN
```

but this time a message displays on the screen. The message is:

```
ERR 10
```

Very cryptic. I remove a ring-bound manual from a drawer below the keyboards. "PAIA 14700/S - Systems Manual" and I thumb through it until I find a section called "error codes". Here I find this entry:

```
ERR 10
```

```
Undefined Instrument Name.
```

Well, rats. I could have sworn that a simple thing like dynamute would have been in my instrument list. Too antique, I suppose; but fortunately it's a simple voice and I know it by heart, I type:

```
:DEFFN "DYNAMUTE":SIGNAL OSC(PULSE 10),  
FILT(BP, Q50, CC2), AMP (100): CNTRL ENVG  
(A10, D0, S0, R5000) ( FILT, AMP), KBD (OSC)  
:TRIG KBD(ENVG)  
:PLAY "DYNAMUTE", K1, OUTL, OCT3  
:RUN
```

And now as I play, the old familiar "wahp-waph"s come from the speaker. A little trite perhaps, but still musicly useful in a piece that is to have an "old classic" sound to it. And just so I won't have to enter this voice again:

```
:INSTSAVE "DYNAMUTE"
```

The central console clicks. Now, to the real work. I type:

```
:SCORE "BASS1" C2/4, E2/4, G2/4, A2/4;  
R; TF2, R; TC2, R; TG2, R; TF2, R;  
C2/4, G2/4, F2/4, A2/8, C3/8, D#3/8,  
E3/8, C3/8, A#2/8, G2/2; BRIDGE, F2/4,  
F2/8, E2/8, F2/8, F#2/8, G2/4, D2/4,  
G2/8, F2/8, E2/8, D2/8;
```

Immediately, the old familiar walking bass line "wahp"s its way into the room while I play string accompaniement on the lower keyboard. After diddling around for a while I come up with a melody line that I like OK and I type:

```
:SCORE "STR-LEAD", K1  
:PLAY "STRINGS", "STR-LEAD", OUTR, OCT5  
:PLAY "DYNAMUTE", "BASS1", OUTL, OCT3  
:RUN
```

and play the lead that I liked. Now a moment to sit back and listen again. I type:

```
:PLAY "STRINGS", "STR-LEAD", OUTR, OCT5
```

```
:PLAY DYNAMUTE", "BASS1", OUTL, OCT3  
:RUN
```

and everything that I played a moment ago is re-created. It sounds good but there's one note that's off. I type:

```
:LIST "STR-LEAD"
```

and the machine replies:

```
STR-LEAD: C4/8, G3/8, A#3/8,A#3/8, C4/16,  
D#4/16, E4/16, G4/16, A#4/16, A4/16, G4/16,  
E4/16, C4/8, G3/8, A#3/8, A3/8, C4/16, D#4/16,  
E4/16, G4/16, A#4/16, A4/16, G4/16, E4/16
```

I can see what's wrong. That third octave A sharp in the first measure should have been a third octave A natural. I type:

```
:EDIT "STR-LEAD"
```

and the score is shown again but now there is a cursor at the end of the line. Using special keys on the keyboard I move the cursor back until it's under the error and then I press a key labeled "delete". The sharp is now a natural and with a PLAY instruction I have the line repeated. Now it sounds right.

Out of habit, more than anything, I type:

```
:COMPRESS "STR-LEAD"
```

and wait while the machine scans this score and reduces the memory space required by inserting "transpose and repeat" instructions wherever possible.

Using SCORE, EDIT and PLAY instructions I lay down another six tracks and then type:

```
:DEFFN "COMP1"  
:PLAY "STRINGS", "STR-LEAD", OUTR, OCT5  
:PLAY "DYNAMUTE", "BASS1", OUTL, OCT3  
:-----etc.
```

and then:

```
:COMPSAVE "COMP1"
```

a click. And just to double check:

```
:CLEAR  
:COMPLOAD "COMP1"  
:PLAYCOMP "COMP1"  
:RUN
```

It's not bad. There are only eight parts, of course, and it did take me a little longer than the graduate students; but they have modern Cyber-net activated instruments to use too. Having to bang away at the keys takes time. And in any case, It's all my work. I didn't use the HARMONY or CREATE instructions once. Poor old Dr. Biggle will like that. Now, before I shut it down:

```
:PRINT "COMP1"
```

and the high staccato of the printer assures me that I will soon have a hard copy of the score on tablature.

I type:

```
:CLEAR
```

and the machine answers:

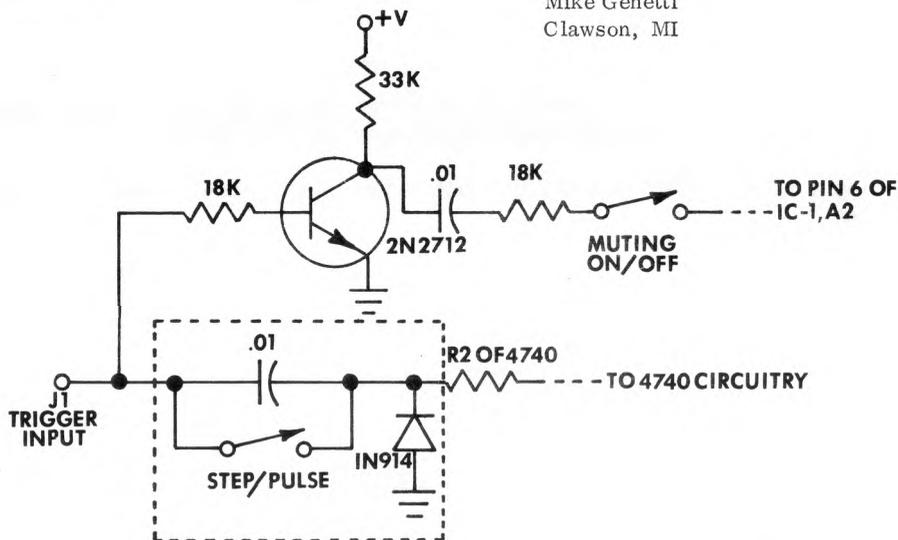
```
READY:_
```

Add Muting, Decay/Release Isolation and/or End of Cycle Retriggering to Your 4740

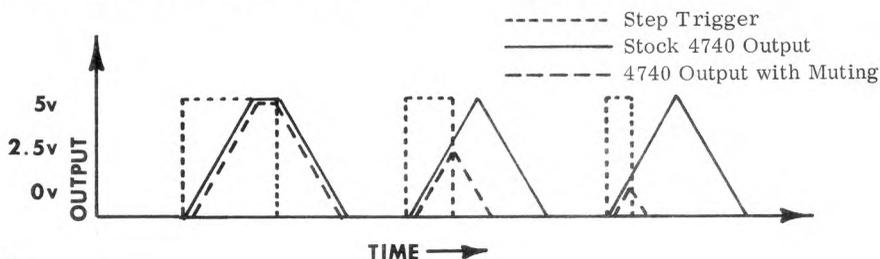
Mike Genetti and Michael Bridgers: Contributors
Marvin Jones: Editing and additional material

Here is a modification to the ADSR that will prove in making long attack settings easier and more comfortable to control. When S1 is closed, the attack cycle will terminate whenever the key is released, instead of rising all the way to a 5 volt peak before shutting off. For me, this allows a lot more expression in playing by giving a considerably wider control over dynamics. If a small toggle switch is chosen for S1, it could be installed in the manual trigger hole, or in the space between trigger button and output attenuator. The only drawback is that a step trigger must be used with the ADSR when the muting is on. If this is a problem, a pulse trigger could be derived by putting a capacitor in front of R2, but after the tap which leads to the muting modification. Also, a diode should be inserted to shunt negative trigger pulses to ground. The switch across the capacitor also selecting a step or pulse trigger to be generated within the 4740.

Mike Genetti
Clawson, MI

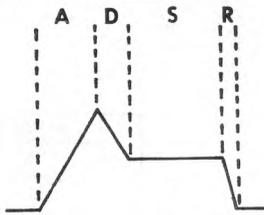


If you have trouble visualizing how muting changes envelopes, a graph is shown below which depicts trigger times, envelope outputs for a stock 4740, and for a unit which has been modified for muting.

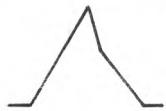


Also, an easier route to having muting on the 4740 is to lift one end of R6. This opens the feedback loop which normally keeps the 4740 triggered until the attack peak is reached. With R6 removed, the 4740 will stay in attack mode only as long as a trigger is applied to J1. A switch can be wired in series with R6 to turn the muting on and off.

The remaining modifications were sent in by Michael Bridgers, Elkhorn, WI. The first is used to make the decay portion of the envelope independent of the release setting. This circuit will only be effective when pulse triggers are used. As you may recall from the design analysis of the 4740, when a pulse trigger is applied, the unit will latch in the attack mode until the 5 volt peak is reached at the output. At this time, BOTH the decay and release discharge paths for the timing capacitor are engaged until the sustain level is reached, at which time the decay path is disabled and only the release discharge path remains active. This causes some interaction to occur between the decay and release controls, and the decay time will always be less than what is produced by the same setting when using a step trigger for sustain mode operation. Shown on the right is a representative envelope using a step trigger and sustain mode on the 4740.



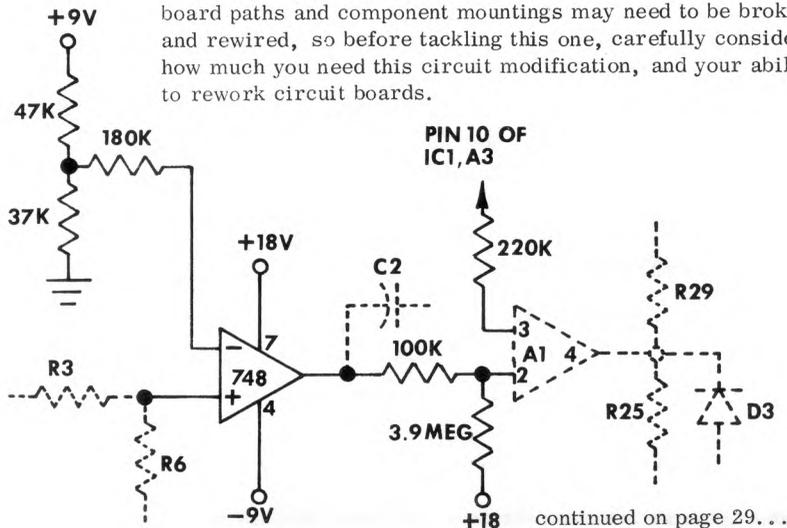
If the same 4740 settings are used, but a pulse trigger is applied, the resulting envelope would look something like:



Note that the decay section is considerably shorter than before. This is due to the timing capacitor rapidly dumping into the low impedance release circuit. Addition of Michael's isolation circuit will restore the timing of the decay cycle to its intended longer setting as:



The schematic for the isolation circuit is shown below. Note that parts labelled with R3, R6, etc., refer to parts that are already in the 4740 circuit. This circuit can be assembled on a small piece of perf-board, and suspended above the stock 4740 board. Some of the circuit board paths and component mountings may need to be broken and rewired, so before tackling this one, carefully consider how much you need this circuit modification, and your ability to rework circuit boards.

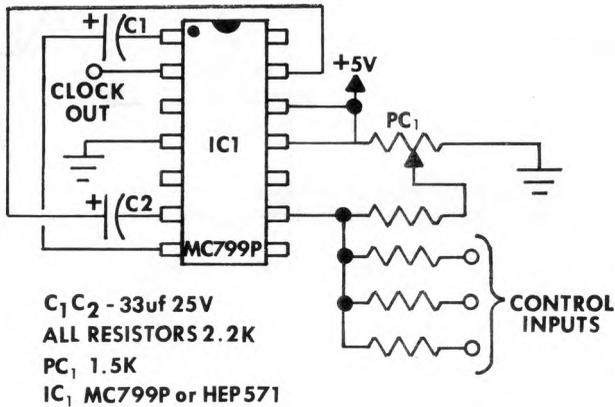


continued on page 29...

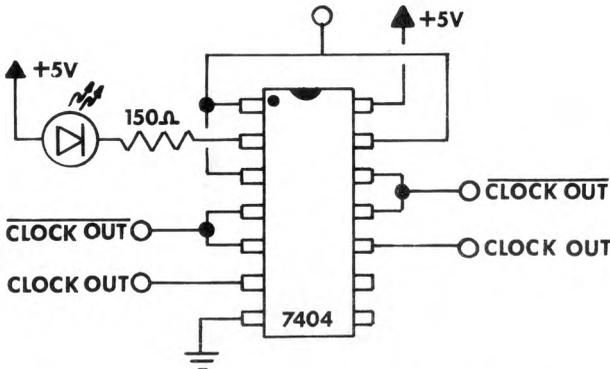
Voltage Controlled Clock

by John Mitchell

Shown are the schematic and parts list for a voltage controlled clock which I have used in my "Psychtone" and other devices for over two years now. The cost is less than \$3, and it is really pretty accurate. The frequency range is about 3 to 25 Hz. with a control voltage range of 1 to 15 volts. I can see no reason why this circuit shouldn't work as an external clock for the 4780 Sequencer by feeding it into the 4780 sync input. This circuit is so much fun to use that I'm sure other readers would enjoy experimenting with it also. Some of the applications include adding syncopation to sequencers, producing square wave vibrato which increases speed as higher frequency notes are played, and as a schmitt trigger (with proper setting of the bias control). If other readers show an interest, I can send some patch diagrams which use this circuit.



Note: PC₁, the bias control may be used to set the initial clock frequency, or set on it's low end (near ground), it adjusts the clock response.



This circuit may be added to the above clock circuit to achieve visual indication of clock rate, and inversion of the clock output pulses.

FUNDAMENTAL MUSIC NOTATION

BY M. J.

PART ONE -- PITCH

As a preface to this series, let me explain what we will try to do, and how we will do it. The majority of us are electronic hobbyists with a common interest in electronic music and audio circuitry. Some of you may have a background in music, but most are lacking in the background needed to write down the music they compose while experimenting with their synthesizers. This series is not intended to teach you how to compose, orchestrate, and notate the perfect symphony. Rather, it will be a review of the most common procedures used to represent music on paper. For those who have a basic background in music (perhaps you learned to play an instrument in grade school), this will be a good review and possibly you can learn something new. For those that have a Doctorate in music theory, you can write me a nasty letter about the things I am omitting.

The series will be broken into about four parts. The first will deal with pitch notation, including names of notes, frequencies of notes, accidentals, scales and key signatures. The second article will cover timing notation, including note durations, time signatures, rests and written tempo indications. The third article will discuss dynamics and expression, and will have some miscellaneous tidbits thrown in. The fourth installment will cover the goal of this series, notating electronic music. Discussion will include how synthesis has been notated in the past, things to keep in mind when scoring for electronics, and hopefully some reproductions of electronic music scores. Part four is the proposed end of the series, but we will continue to print new ideas about notation from our readers. Hopefully, some of you will send us your own scores to be printed!

The general format of the series will be more like a glossary or outline, rather than full text. This will allow future use as a reference source, and will cut the size of the series from a textbook to a magazine article. (Thank goodness!)

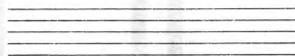
PITCH NAMES are derived from the first seven letters of the alphabet - A B C D E F G.

The STAFF is the configuration of five lines and four spaces on which the music is written.

The G CLEF is placed on the staff such that the body of the clef circles the line designated as the note G. The most common position for this clef is when the second line from the bottom of the staff is G. With the clef in this position, it is commonly referred to as a TREBLE CLEF.

The F CLEF is placed on the staff such that the line passing through the two dots is the note F. The most common position for this clef is when the fourth line is designated F. This is called a BASS CLEF.

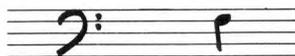
STAFF DESIGNATIONS use consecutive letters for adjacent lines and spaces. Higher locations on the



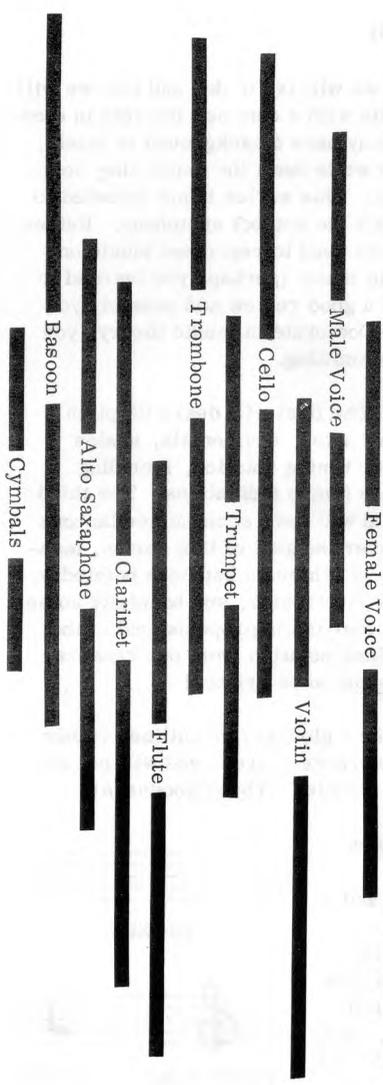
The Staff



Treble Clef G note

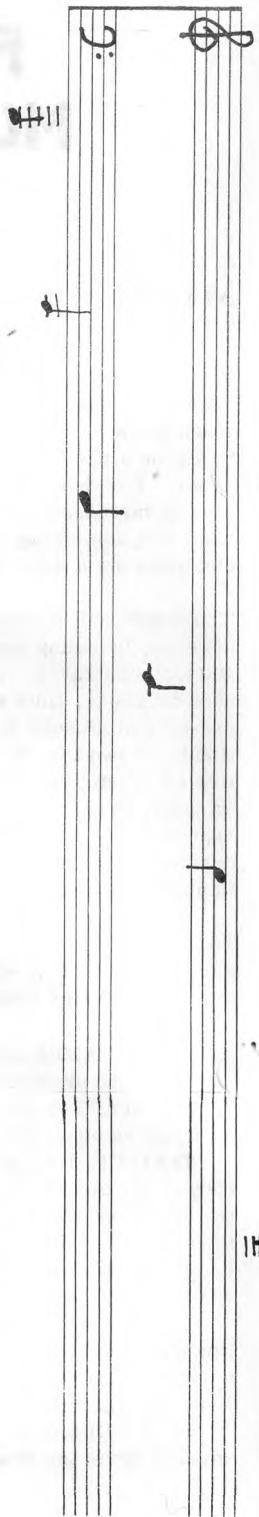


Bass Clef F note



FUNDAMENTAL
NOTATION

16.352	C ₀
32.703	C ₁
36.708	D ₁
41.203	E ₁
43.654	F ₁
48.999	G ₁
55.000	A ₁
61.735	B ₁
65.406	C ₂
73.416	D ₂
82.407	E ₂
87.307	F ₂
97.999	G ₂
110.00	A ₂
123.47	B ₂
130.81	C ₃
146.83	D ₃
164.81	E ₃
174.61	F ₃
196.00	G ₃
220.00	A ₃
246.94	B ₃
261.63	C ₄
293.66	D ₄
329.63	E ₄
349.23	F ₄
392.00	G ₄
440.00	A ₄
493.88	B ₄
523.25	C ₅
587.33	D ₅
659.26	E ₅
698.46	F ₅
783.99	G ₅
880.00	A ₅
987.77	B ₅
1046.5	C ₆
1174.7	D ₆
1318.5	E ₆
1396.9	F ₆
1568.0	G ₆
1760.0	A ₆
1975.5	B ₆
2093.0	C ₇
2349.3	D ₇
2637.0	E ₇
2793.8	F ₇
3136.0	G ₇
3520.0	A ₇
3951.1	B ₇
4186.0	C ₈



staff use later letters of the alphabet. To help students remember the proper names for lines and spaces, several phrases have been devised. The first letter of each word is the note name, working from the bottom of the staff to the top. To remember the lines of the Treble Clef, remember - Every Good Boy Does Fine. The spaces of the Treble Clef spell F-A-C-E. The lines of the Bass Clef are represented by - Good Boys Do Fine Always. The spaces of the Bass Clef are - All Cars Eat Gas.



LEDGER LINES are used to provide lines and spaces for notes outside the normal scope of the staff.

The GRAND STAFF is a joint bass and treble staff. Most instruments use one of the common staves for its music, but instruments with a wider range of pitches available generally use the grand staff. Piano, organ, harp, and condensed ensemble scores are typical applications of the grand staff.

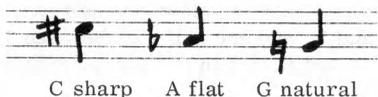
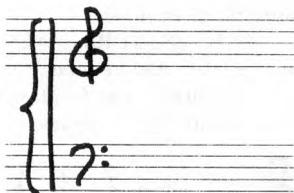
HALF STEPS occur in the musical alphabet at E/F and B/C. These notes produce adjacent musical tones. All other adjacent letter pairs produce a FULL STEP, which means a musical tone is available between the two letter notes. On the keyboard, any two consecutive keys produce a half step.

SHARPS and FLATS are used in conjunction with the letter names to achieve notation for the tones which lie between the letter notes. A sharp is used in front of a note to raise the note a half step.

A flat is used in front of a note to lower the note a half step. Double sharps and double flats also exist, and are used to raise or lower a note by a whole step. The "double signs" are not used often. A natural sign is placed before a note when you wish to cancel a previously given sharp or flat.

ENHARMONIC NOTES are notes which are written differently but sound the same pitch. Examples of enharmonic notes are: G sharp/ A flat, E sharp/ F, C flat/ B, F double sharp/ A double flat/ G, etc.

RELATIONSHIP OF PITCH NAMES, KEYBOARD, OCTAVES, FREQUENCIES, AND INSTRUMENT RANGES are shown in the accompanying chart. The use of subscripts to denote various octaves is the system recommended by the Acoustic Society of America, and accepted by the U.S.A. Standards Association. Other systems are also in use, and descriptions can be found in most music textbooks. All frequencies given are based on the A-440 tuning standard and the equally tempered scale formula. Instrument ranges are approximate, as upper and lower extremes depend heavily on the skill of the performer.



CHROMATIC SCALES are all the half steps available within an octave (twelve), and the form is identical for all keys. The only difference is the starting note.



C Chromatic Scale

NAMES OF NOTE POSITIONS WITHIN A SCALE are useful in naming the chords built on the different notes, and naming the relationship of a particular note to the tonic or key.

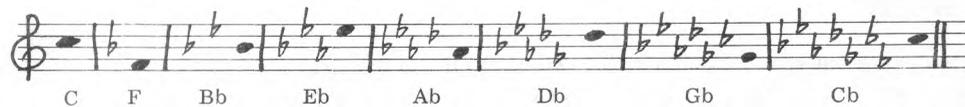
Note of Scale:

Position Name:

1	Tonic
2	Supertonic
3	Mediant
4	Subdominant
5	Dominant
6	Submediant
#6 (raised sixth)	Raised Submediant (minor only)
7	Leading Tone (major only)
	Subtonic (minor only)
#7 (raised seventh)	Leading Tone (minor only)

KEY SIGNATURES are a group of sharps or flats that are written at the beginning of a song or section of a song. These accidentals are the ones required to produce a proper major or minor scale pattern, but when written as a key signature, they need not be rewritten every time an altered note appears in the piece. Sharps and flats will still be seen in the music, but these will be accidental notes as required to fit into the chords or melody being used. Also, a natural sign can be used prior to a note which is altered by the key signature. The natural sign will cancel that particular note's sharp or flat. Accidentals, including natural signs, are good only for the duration of the measure (measures will be covered in the next issue) at which time the key signature goes back into effect as the only designated sharps or flats. The letter name of a key is the same as the tonic note of the scale which the key signature represents. Thus, a key signature with no sharps or flats would be C major, the same as a scale with no sharps or flats. Also, this key signature could represent A minor which doesn't have any accidentals. A chart of various keys and key signatures is shown below.

Major Key Signatures



Bass Clef Key Signature Positions



RELATIVE KEYS use the same key signature, but are actually two different keys. C major and A minor are relative, as are G major and E minor.

CHORDS are groups of notes which are sounded simultaneously or one after another at high speed (arpeggio). The most basic chord is constructed using 1-3-5 of the scale of the key you wish. For example a C major chord would be C-E-G, a C minor chord would be C-E flat-G. When a chord is written with the tonic in the lowest position, and the other notes represented as intervals of thirds above the tonic, the chord is called a TRIAD. The chord based on a major scale is called (amazingly enough) a major triad. Likewise for a minor triad. A minor triad with a flatted fifth is called a DIMINISHED TRIAD. A major triad with a sharped fifth is an AUGMENTED TRIAD. All four of these basic chord structures are used extensively, and each has a different texture, or sound, to it. Basic triads can be expanded into more complex chords by adding more notes. Common complex chords include seventh, ninth, augmented eleventh, and thirteenth flatted ninth type chords. Chords with a tonic note other than the tonic note of the key signature will have a specific relationship to the tonic key as outlined in the "Names of Note Positions" heading. For example, a G chord in a piece that is written in C major would be a dominant chord with respect to the C major tonic chord. If the music should modulate to the key of A minor, and a G major chord appears again, it will now be a subtonic chord.

C Major Triad F Minor Triad G Augmented Triad D Diminished Triad

E Flat Sixth Chord D Seventh Flatted Fifth Chord

This should give you the basics required to get from the notes on your keyboard to a melody on paper. Next time we will discuss how to notate different lengths of notes and timing of the notes relative to each other. In the interim, a bibliography is given for more detailed study or reference.

Harvard Dictionary of Music, by Willi Apel, The Belknap Press of Harvard University Press, Cambridge, MA.

The Acoustic Foundations of Music, by John Backus, WW Norton and Company, New York.

Introduction to the Theory of Music, by Howard Boatwright, WW Norton and Company, New York.

Horns, Strings, and Harmony, by Arthur H. Benade, Doubleday and Company.

Scales, Intervals, Keys, and Triads, by John Clough, WW Norton and Company, New York.

Also check out your local libraries, especially if you live near a university that has a good music department. Another good source of information for beginning musicians is in grade one piano instruction books. Check at local music stores for this type of book.

Adding Fine Tuning to Standard Controls

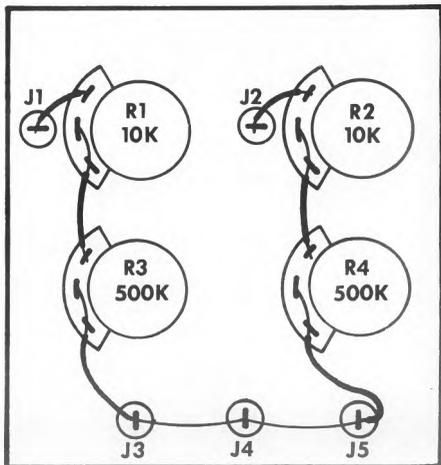
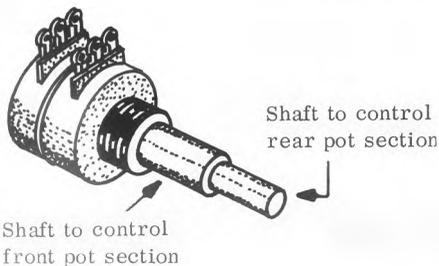
by Gary Miller

If you have more than one VCO and have a transposer to offset them, you have probably experienced some difficulty in getting them perfectly tuned. The broad tuning range of the standard 500K pots in the 4781 can make fine tuning very sensitive and difficult to do rapidly. If you have such problems, there is an inexpensive and simple way to eliminate them. Whether or not you presently have a 4781 transposer, you can easily convert it or build from scratch a Fine Tuning Transposer.

The transposer consists of two 500K pots (there are four of these on a stock 4781), and two 10K pots, wired in series, with the front panel patch controls similar to a normal transposer, except that there are two less outputs. The upper fine tuning controls can be set to approximate midrange. Then the lower pots can be used to fine tune the interval desired, but don't be concerned with getting the VCO's perfectly tuned. Go back to the upper controls and fine tune the VCO's for perfect zero beat tuning. Eventually you will be able to tune the lower pots so the upper pots will be in the center of their ranges for zero beat tuning, thereby enabling minor corrections either up or down in pitch while playing with the other hand.

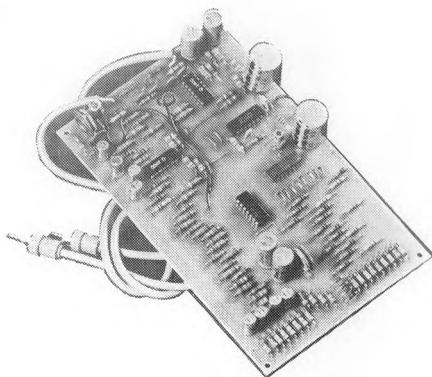
Editor's Addenda: Gary has a good idea here, but keep in mind that fine tuning can be used in other modules, too. For example, how about fine tuning on the keyboard pitch control? Also, check with your local electronic parts distributor to see if he stocks a system of "custom built" replacement pots. Rather than stock thousands of potentiometers, many dealers stock an assortment of potentiometer "bodies" with various resistances, various shaft lengths and styles, and rear mount add-on switches. If your dealer uses this system, have him make up a dual section potentiometer with concentric shafts. The front section of the pot (the section which will be directly behind the panel) should be a value equivalent to the pot which you are replacing for fine tuning (in the case of the 4781, this would be 500K). The rear section of the pot should be a value of approximately 10% of the original pot (this will be the fine adjustment). The use of concentric shafts will allow separate control of each pot section via two front panel knobs which appear to be stacked on top of each other. The rear or outer knob will control the front section of the pot, which in our case will be rough tuning. The front or inner knob of the control will adjust the rear pot section, which is our fine tuning. Use of this type of control

will allow adding fine tuning to any control you wish, and you won't need to drill a hole for another pot. This keeps wasted space to a minimum. If dual section pots are used to add fine tuning to a 4781, you can still have four outputs on one transposer, and be able to fine-tune a 5 note chord.



NEW NEW NEW

STEREO CHORD EGG*



THE ULTIMATE MEDITATION ENVIRONMENT MACHINE ?

The EGG plugs into any stereo amplifier to create soothing, flowing environmental sound textures - and through headphones it's incredible. The chords and notes EXIST in your mind, they pan and swell and phase their way through your psyche in unpredictable patterns, never repeating.

The combination of LSI organ technology with synthesizer-type processing and digital randomizing/control elements make the EGG an altogether intriguing package from either technological or metaphysical viewpoints.

Kit includes: Circuit board, all parts and step-by-step instructions.

#3790 STEREO CHORD EGG KIT \$24.95 plus \$1.00 postage
(no case available - 8 "AA" penlight batteries required)

* EGG - Encephalo-Gratification Generator

NEW NEW NEW



PHLANGER

Phil Spector's original "Big Hurt" sound, which can only be approximated by using frequency dependent phase shifters, is EXACTLY duplicated using PAIA's new constant time delay PHLANGER. This unit features a 1024 stage Charge Coupled Device analog shift register that exactly duplicates the effect of two tape recorders running slightly out of synchronization. Multiple user controls allow effects ranging from chorusing, voice doubling and reverb to full "Jet Plane" effects. Optional foot pedal provides hands free control of internal oscillator sweep speed or manual spectrum sweeping.

#1500 PHLANGER Kit \$59.95 shipping wt. 4 lbs.

SOUND SYSTEMS FOR SYNTHESIZERS

by Marvin Jones

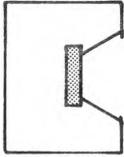
Most people who spend any time at all working with synthesizers quickly become aware of the necessity of a good sound system for the instrument. Unlike other electronic instruments, a synthesizer is capable of creating signals far beyond the extremes of human hearing. Putting together a system to accurately reproduce these signals need not be as costly or cumbersome as first suspected.

Sound systems can be broken down into three classifications of equipment: amplification, speakers and special effects or audio processing equipment. Amplification encompasses the raw power amps to get the little signal voltages converted to BIG voltages which are capable of moving the speakers. Speakers are the device used to change those big dancing electrons into dancing air waves that a human ear can hear. The processing equipment is everything that's left over: preamps, tone controls, equalizers, electronic crossovers, compressor/expanders, mixers and - the list could go on for pages. Hopefully, the following pointers will help you sort all this and put together the type of system you need at minimum expense of time or money.

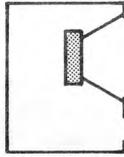
Speakers are the most important stage of any sound system, because they are what actually connect the sound to our brains (via air and our ears). Unfortunately, they are also the weakest link of any sound system, primarily because they are semi-mechanical devices. They are generally inefficient (much amplifier power is lost in the conversion to motion), and they have a high degree of distortion (the mass of the speaker cone tends to resist musical transients, and the cone may flex unevenly or create standing waves on the surface of the cone, etc.). But, it's the best device we have so far, and acoustics people have found ways to get around most deficiencies, so we use them and hope for a better day. The most important device for making a speaker usable is the speaker enclosure.

When a speaker is set in motion, air waves will be created in front and behind the speaker. If the speaker is hanging in midair, the waves from behind the speaker will bend around and mix with the waves in front of the speaker. Due to the time lag involved, much of the sound will be cancelled because the rear waves will end up being 180 degrees out of phase, or exactly opposite in polarity, to the front waves. The speaker enclosure eliminates this problem by doing one of two things: stopping the rear waves, or routing the rear waves through a mathematically derived passage-way and releasing them from the front of the cabinet to reinforce the speaker's front waves. Many sizes and shapes of speaker cabinets have been developed, but they all boil down to a few basic designs. The infinite baffle is a sealed box that only allows the front of the speaker to radiate waves to the listener. These enclosures tend to be inefficient because the pressure of the air in the enclosure works against the speaker cone movement. Also, an infinite baffle must be rather large to have good low frequency response. The ported or bass reflex enclosure is much more efficient because the air in the enclosure is allowed to leave the cabinet to reinforce the front speaker waves. A ported enclosure will produce roughly twice the acoustical output of an infinite baffle with a given electrical input. Ported enclosures also allow more accurate "tuning" of the speaker enclosure which gives a wider, smoother response in the low frequencies. The third basic enclosure style is

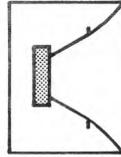
the horn. These can come in front loaded horns, rear loaded horns, folded horns, reflex horns and more. A horn acts acoustically like a transformer acts in an electric circuit. With a horn system, the speaker is gradually "matched" to the air pressure in the room, so the speaker cone vibrations can produce as broad an area of air motion as possible. Unfortunately, an ideal horn enclosure which would have good low frequency response would need to be very long and have an opening larger than most peoples living rooms. Most commercially available speaker enclosures are a combination of two or more of the above styles. This allows one to have the different advantages of several systems rolled into one. The popular Altec Voice-of-the-Theater system has the low frequency speaker mounted in a horn enclosure which is also a reflex system. The high frequencies are generated separately by a smaller horn system.



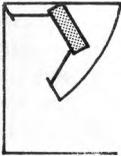
**INFINITE
BAFFLE**



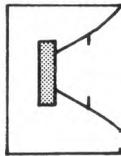
**PORTED OR
REFLEX CABINET**



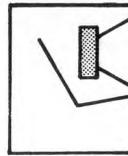
**FRONT LOADED
HORN**



FOLD HORN



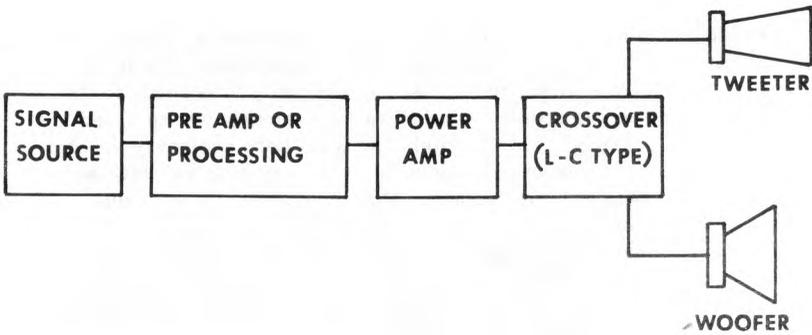
**REFLEX HORN
OR PORTED
FRONT LOADED**



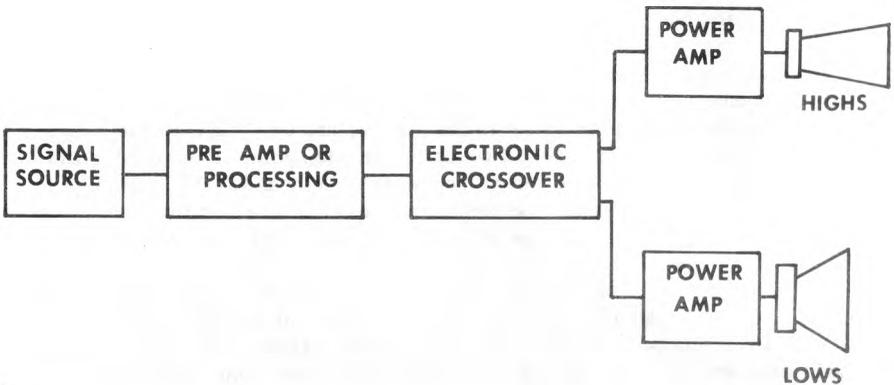
**REAR LOADED
HORN**

This brings up the point of multiple speaker systems. If a horn or reflex type enclosure is designed to have good low end response, the midrange and high frequencies will not be reproduced because the system will be too large and massive to be able to vibrate at these frequencies. With two speakers, the low frequency unit is called a woofer, and the high frequency unit is called a tweeter. More complex multi-way speakers can also be set up so each speaker can concentrate on an even narrower band of frequencies. It is not uncommon to see six-way speaker systems for use at large high-volume concerts.

Crossover networks are required when more than one type of speaker is driven from one power amp. A crossover is a simple circuit using capacitors and coils to allow only the low frequencies to get to the woofer, and only high frequencies to get to the tweeter. The frequency at which sounds stop going to the woofer and start going to the tweeter is called the crossover frequency and is designated f_c . The main problem with L-C (coil-capacitor) type crossovers is that the components must be able to handle the high voltage and current being supplied to it by the power amp. These components are expensive and large in size. Also, when using L-C crossovers, the power amp still needs to work on the complete audio range. This can be a lot of work for the amp and may limit the amount of available power to handle musical transients. The end result could be distortion in the amp, or overheating of output devices. All this depends on the size of the amp, what kind of signals you are amplifying and so on.



Electronic crossovers are devices which have recently become very popular, and are being used more and more in commercial sound systems as well as home stereo systems. And for good reason: The benefits obtained when using these devices are many. First, an electronic crossover is a low power device which is used between the preamp or mixer and the power amp. The circuit is very similar to the filters found in your synthesizer. A full range signal is applied to the input, and the electronic crossover divides the sound into as many frequency ranges as you have speakers for. Being a low power circuit, parts are easy to find, and are cheap. It is also easy to have switch selectable crossover frequencies to add more versatility to the system. After the sound has been divided into various bands, and reinforced with some small amp stages, the outputs are applied to multiple power amps. Each power amp directly drives the appropriate speaker section. Now you may think it's dumb to save some money on a crossover and have to spend a bunch more on power amps - right? Wrong! When electronic crossovers are used, not only is each speaker section working on a separate range, but the power amps are now limited to working on a smaller range of sounds. Thus you can use much lower power amps (which are cheaper) and end up generating more sound than you could with one large amp. If you used two 10 watt amplifiers in a bi-amped system, you could get as much sound output as using one 40 watt amp in a system with an L-C type crossover. That's pretty good - huh? Well, it get's better. If you have a four way speaker system, and a four way electronic crossover, and you use four 10 watt amps, you can get the same amount of sound as using a single 160 watt amp!! Talk about a bargain. And when you consider the fact that you will probably be using amps more in the 50 to 100 watt range, you're talking about a lot of breeze from those speaker systems.



Power amps are a necessity in any sound system, and can be somewhat ambiguous in capability when you first start shopping for them. The main specifications you will see associated with power amps are distortion and power ratings. Distortion you want to be as low as possible, although when you start talking about fractions of a percent (.2% and lower) the ear has a hard time detecting much improvement. So don't overbuy lack of distortion. Power ratings can also be tricky in the following sense. Doubling the amp power will not double the amount of volume you can get from a system. This fact is commonly misunderstood. Perception of sound intensity by the human ear varies logarithmically. Loudness perception also varies with changes in frequency and harmonic content. This is evident when you listen to a triangle wave and a ramp wave from your synthesizer. Although both signals are of the same amplitude, the ramp sounds louder due to the complex harmonic content. Also, when you mix two identical waveforms together, you can hear an increase in volume, but by no means is the resultant sound twice as loud. Yet the signal amplitude IS producing twice as much power from the amp. So now you are asking why everyone is so concerned about high power amps. Two reasons primary-

If an amp is driven at a lower output power, the amp isn't working as hard, it will produce less distortion in addition to not putting a strain on the components in the amp. Also, if you have a high power amp and have it operating at a lower power level, when a musical transient comes along (cymbal crash, attacks of brass or plucked string instruments) the amp will have enough reserve power to reproduce the transient without being pushed into an overload condition and causing distortion.

Accessory units are available to do about anything you have in mind, and can vary from foot pedals for volume control to a 36 channel studio type mixing console. Of course all of these units aren't necessary (thank goodness, we'd all be broke) but some of them can be a big help. A mixing system of some type is important to most keyboard players since you may have a synthesizer, electric piano, organ, maybe a string synthesizer and more. A basic mixer should have separate level controls for each input, tone controls for each input, output assignment (if the unit has more than one output channel), and some master level and tone controls. Tone controls on a multi-keyboard mixer are fairly important and shouldn't be overlooked. The standard treble/bass controls leave a lot to be desired, and will tend to limit your capabilities as much as helping. Parametric equalization is a much better choice, but of course they cost a little more. Parametric EQ is usually divided into three sections: low, mid, and high. In addition to adjusting the amount of cut or boost in these areas, you can also select the frequency at which the tone control will have it's effect. Master output levels are important, but the master tone controls can be eliminated if you have input tone controls or an equalizer on

output. Graphic Equalizers are getting more popular, and prices are going down as filter technology increases. If you are going to get an equalizer, don't bother with less than five bands. These give one control for each two octaves of audibility, and invariably the frequency you want to adjust will be right between two of the controls and you will need to alter four octaves to adjust the frequency you are working with. Nine or ten bands, on one octave centers, is about optimum and these units can be found for under \$100. Some companies make equalizers with as many as 30 bands, and while a very detailed tonal adjustment can be made, they are tedious to use and are more trouble than they are worth. Another handy device is a noise gate or a compressor/expander. A noise gate grounds the audio signal line whenever there is not a signal present that is greater than a preset threshold level. This eliminates hum and hiss from your system when you aren't playing anything. A compressor/expander can do the same type of thing plus more. When in an expansion mode, it will make your loud signals louder, and your soft passages even softer. Thus when your keyboards aren't being played, the volume will be turned down somewhat, and noise will not be as apparent. When a loud signal is played, it will come

out a little louder than usual. Transients from the synthesizer or the percussion on your Hammond will be accentuated, and your overall dynamic range can be increased by as much as 100%. In the compression mode, exactly the opposite will occur. Loud sounds will be decreased in volume, and soft sounds will be amplified. The overall sound will then be at a more consistent setting all the time. This effect is good for vocalists in bands, because once the volume is set on the PA system, the vocalist can whisper or shout and will always be heard above the band.

Well, by now you should know a little more about what to consider when developing a sound system for your synthesizer. Sadly, though, we have only scratched the surface of this subject. Following this article is a rather extensive bibliography of material covering in detail the things mentioned in this article. Some of the magazine articles have construction projects explaining how to build and use the items mentioned. Some are more technical discussion of acoustics, speaker design, and other subjects. Also listed are names of some manufacturers that have good info available on what their products will do.

Good luck putting together your personalized sound system, and may all the subtlety of your creative musical expression be accurately reproduced.

SPEAKERS

How to Build Speaker Enclosures, by Alexis Badmaieff and Don Davis, Howard W Sams & Co, Inc., Indianapolis, IN. This book gives good background on the types of enclosures and how to build them.

Loudspeaker Enclosures - Their Design and Use, \$2.00 from Altec, 1515 S. Manchester Ave., Anaheim, CA 92803. Also ask them for information on their raw speaker components.

The JBL Enclosure Construction Kit, \$5.00 from local JBL dealers or James B. Lansing Sound, Inc., Technical Information Department, 3249 Casitas Ave., Los Angeles, CA 90039. The kit includes material dealing with enclosure design, theory and construction methods. Also included are construction plans for five of JBL's enclosure designs. Also, ask them for information on their raw speakers for both monitor and instrument use.

Speakerlab, 5500 35th N.E., Seattle, WA 98105. These guys put out stereo type speakers in kit form, and sell enclosure plans and raw speakers. Although I haven't had an opportunity to do business with them (yet), their catalog implies a concern for the customer and high quality merchandise.

Audio Systems Handbook, by Norman Crowhurst, Tab Books, Blue Ridge Summit, PA 17214. A good generalized book covering all angles of audio reproduction.

Electronic Projects for Musicians, by Craig Anderton, \$6.95 from Guitar Player Productions, P.O. Box 615, Saratoga, CA 95070. This book has plans for building mixers, preamps, tone controls, compressors, etc.

Southwest Technical Products Corporation, 219 W. Rhapsody, San Antonio, TX 78216. Send for this catalog for sure! They have the absolute best buys on kits for power amps, equalizers, compressor/expanders, and a lot more.

The Audio Amateur (magazine), Box 176, Peterborough, NH 03458. This magazine is a continuing source of excellent, high quality audio circuits. They have just started a four part article on mixers you can build, and in the second issue of 1972, they had an excellent circuit for electronic crossovers complete with theory, modifications, and applications.

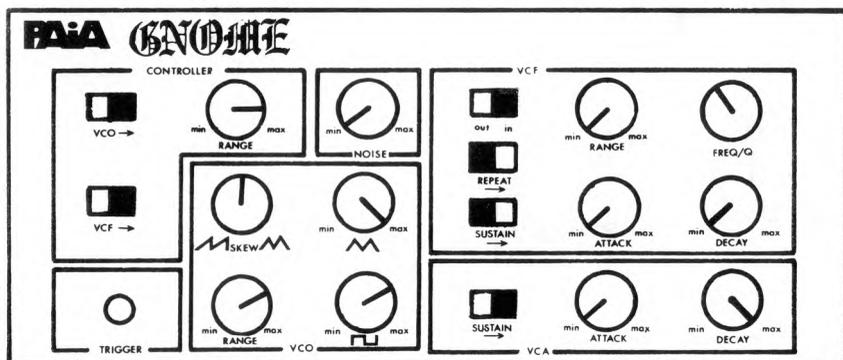
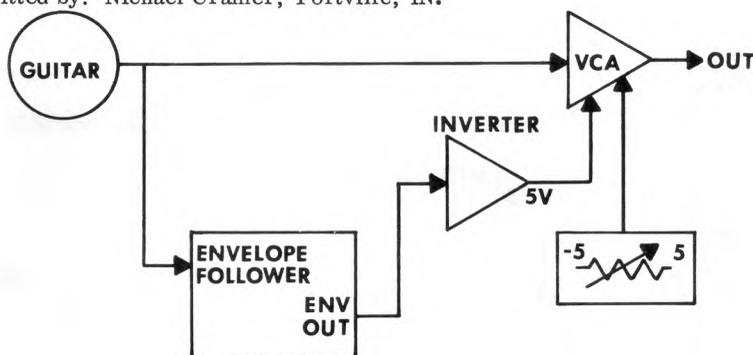
Modern Recording (magazine), 15 Columbus Circle, New York, NY 10023. In the June/July '76 issue they printed the first of a three part article entitled "P.A. Primer". This article covers important material with much detail and technical analysis. This article is recommended.

patches

POOR MAN'S COMPRESSOR-SUSTAINER

A guitar signal, initially loud, will drive the output of the envelope follower to an appropriate amount as adjusted. This is inverted, offset by 5 volts and mixed with the -5 to +5 volt bias source. As the signal fades, the VCA opens up all the way, allowing more of the original signal through. By adjusting the bias and sensitivity of the envelope follower, the signal can be maintained at a set level - thus approximating a compressor-sustainer.

Submitted by: Michael Cramer, Fortville, IN.



PROP-TYPE AIRPLANE

Controller: VCO: on
 VCF: on
 Range: 75%

VCO: Skew: 50%
 Triangle: maximum
 Square Wave: 60%

*Hit the trigger and move the skew, keeping the probe on the lower one-third of the controller strip. You can move the skew and probe simultaneously for a different effect.

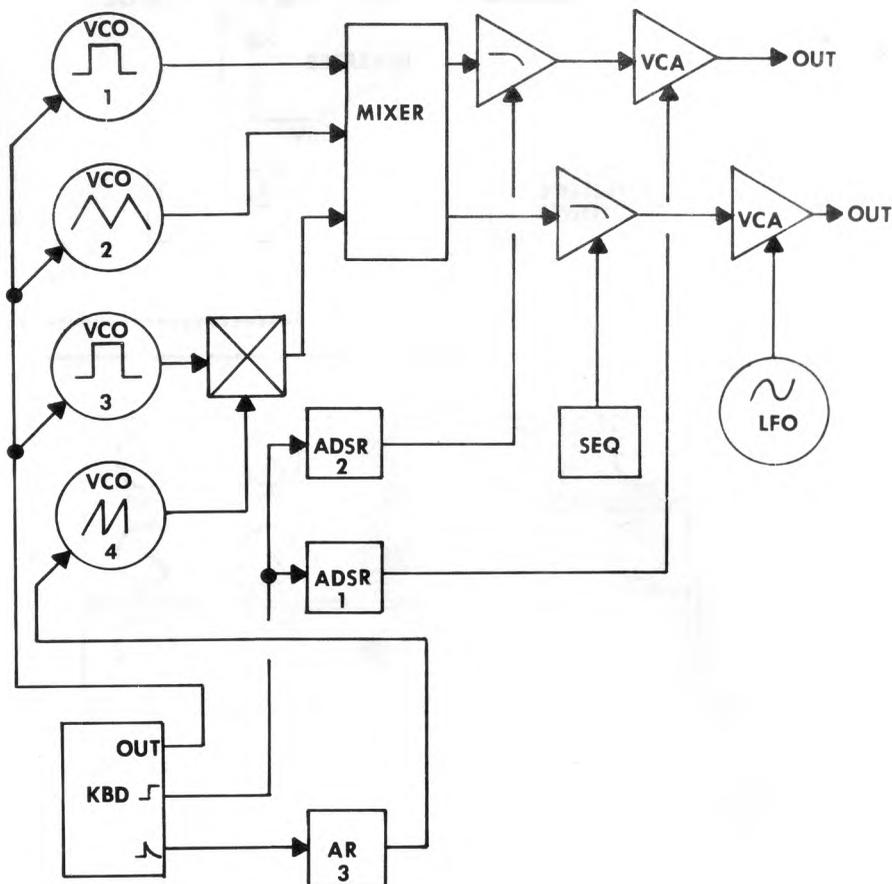
Submitted by: Jay Truesdale, Ferguson, MO.

more...

patches

Jim said this patch surrounds you with the pulsating beat of conga drums, birds flying out of the bushes, and hyenas laughing in the distance. Bringing the level of VCO 1 and 2 up, you can play a tune in the jungle. Applying a pulse to AR #3 and setting the decay for maximum gets a screaming banshee. Applying a very low control voltage to VCO #4 modulates the pulse of VCO #3 for a repeat effect (about 1 every 2 seconds) which then rings through the filter for an eerie sounding drum. Sequencer produces the background conga drums.

Submitter: Jim Riter, Austin, TX.



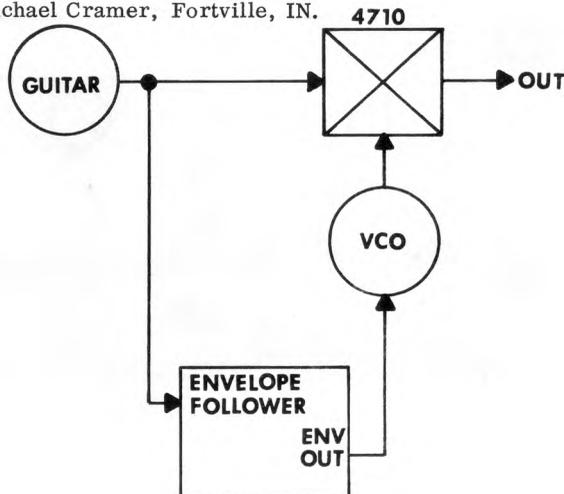
more...

patches

BANSHEE GUITAR

Other instruments with a good dynamic range, such as voice or electric piano can also be used. Besides giving an additional sound to the guitarist, a good deal of subtlety and variation is possible. The musician can control both signal pitches going into the ring modulator; the original pitch and the VCO pitch by the volume of the original signal. You can also use the gate from the envelope follower to trigger an AR generator to drive the VCO. This will give a fixed pitch to modulate against.

Submitted by: Michael Cramer, Fortville, IN.



JUNGLE PATCH

VCO's: 1 and 2 tuned to unison
3 and 4 tuned one octave lower

Keyboard: No Glide

Mixer: All panpots at 50%

Filters: Both filters set to low range
Maximum Q, Lowpass output

ADSR 1: Attack-10%, Decay-30%, Sustain-70%, Release-100%

ADSR 2: Attack-50%, Decay-70%, Sustain-100%, Release-80%

AR 3: Attack-50%, Release-100%
Use variable output

LFO: 9 Hz.

Sequencer: Rate-60%, No Glide
12 stage recirculation

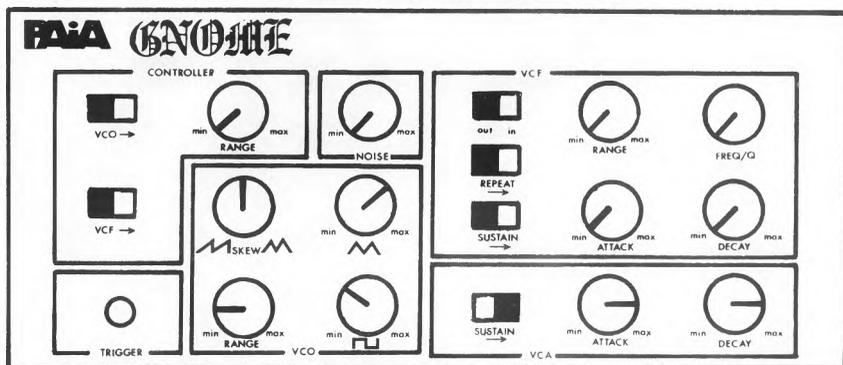
more...

patches

CAT PURR

Adjust VCO range to a frequency of about 10 Hz. or so. For best accuracy, use a real cat for comparison and adjust wave form. Hold trigger down until VCA reaches maximum gain, then release until gain falls to about 1/2 or 1/3 maximum, then press again. This imitates the cat's breathing. (I think cats breathe about 70 times a minute - adjust attack and decay rates). Note well: Turn amplifier to very soft - unless your cat is a lion.

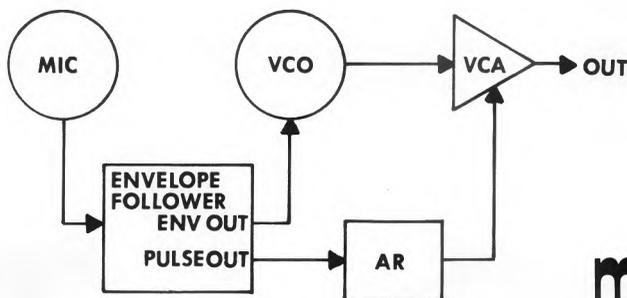
Submitted by: Mark Lutton, Schenectady, NY.



DRUM TRIGGERS

Adjust envelope follower for desired range of pitch. A small mike, such as one from a little inexpensive cassette recorder should be taped to the bottom head of a drum. A mounted tom-tom is usually pretty good. Cassette recorder mikes are generally rather insensitive and can be used on drums where they will not pick up other instruments. They often have mini-jacks which make them compatible with PAIA modules. The envelope follower can be used to trigger any patch you may wish to use.

Submitted by: Michael Cramer, Fortville, IN.



more...

patches

JETLINER LANDING

Eric Hanson's patch of a jetliner take-off (Polyphony 1/76) was not complete without a jetliner landing. This patch is so realistic that an aviation expert friend of mine swore that it was a recording of a real jet!

To do this sound, it is necessary to modify the time constants of the 4740 by increasing the timing capacitor to 25 mf. P.S.: If anyone comes up with a way to create the sound of the wheels screeching at touchdown, let me know.

VCO: #1) Maximum frequency.
#2) 10 Hz. less than VCO #1

SR: Modifications: on

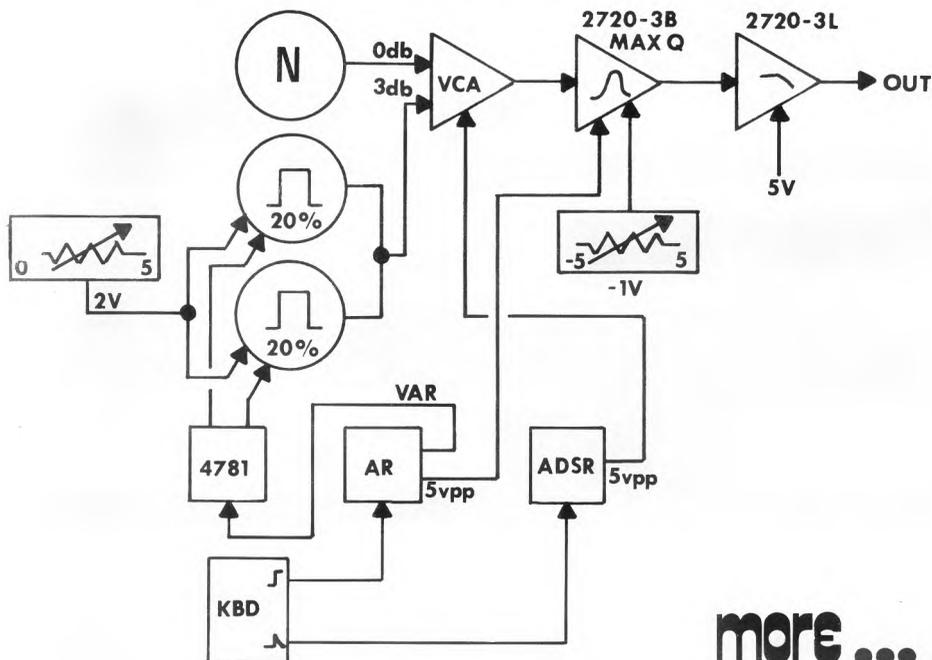
A: 50%
D: 100%
S: 100%
R: 100%

AR: Modification: on

Expand: on
A: 25%
D: 100%

Pressing any key on the keyboard starts the cycle.

Submitted by: Charles Lauria, Staten Island, NY.



more ...

patches

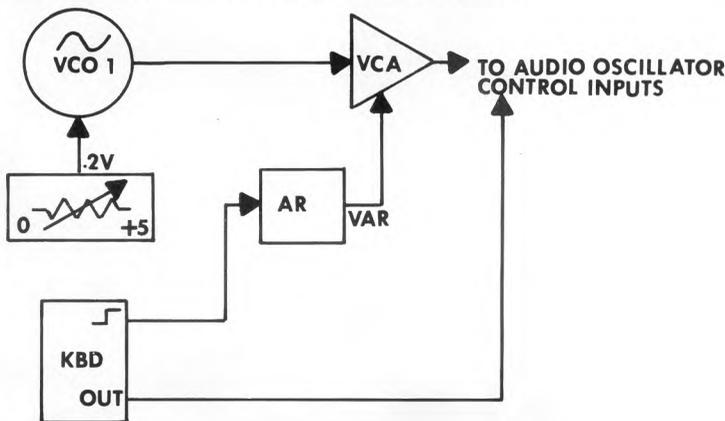
DELAY VIBRATO

I found a good patch for a "delay vibrato" effect. It's uses can be varied but I found it produces a quite interesting violin sound when applied to one VCO and played in conjunction with another VCO at the same pitch (for phasing effects).

AR: A: 100%
R: 10%

Comments: Output of AR can be varied to alter depth of vibrato. AR attack time can be varied to alter amount of delay. Bias to VCO #1 can be changed to alter speed of vibrato.

Submitted by: Michael Wilson, Ottawa, Ontario, Canada



more next issue...

Random Noise ... continued from page 8

+, -, GND and blank write-on labels. The standard wire marker is made of vinyl covered cloth, but markers are also available in self-sticking aluminum foil, sub-surface printed mylar and oil-resistant reinforced plastic. Check with your local electronics distributor or write:

W. H. Brady Company
727 W. Glendale Ave.
Milwaukee, WI 53201

Ask for catalog #100 which should contain listings of the various markers available.

LAB NOTES

by John S. Simonton, Jr.

Are you ready for a computer peripheral that's fun, instructive, low in cost and provides an impressive demonstration that even your least technically inclined friends can understand? How about this one - a computer controlled drum set.

Yes, there are a lot of automatic percussion units available and yes, they're pretty much a drag. Useful, under just the right set of circumstances, but even then their incessant BOOM-chik-chik can really turn into a BUMMER.

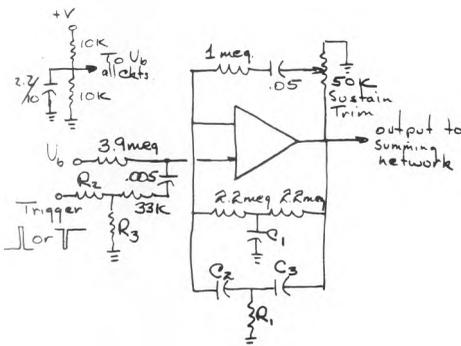
Now, think about computer interfaced drums. You're not forced into using what somebody else thinks is a cha-cha; you write your own rhythm pattern for the specific timing you're doing, even if it's in 7/16 time. And the real world niceties like bridges and intros? They're simply not possible with conventional electronic drummers. But with drums tied to a computer - no sweat.

If you're interested, we've got a lot of ground to cover; drum circuits, interfacing and programming. Let's get on with it.

DRUM OSCILLATORS

Most drum units generate their percussion sounds using a simple active filter section like that shown in figure 1. There are two ways to think of this circuit. Either it is almost an oscillator which when excited by a pulse "rings" for a short period of time; or, it is a high Q filter section which extracts from a pulse excitation function a single sine wave component. Either of these explanations is valid. Either produces an accurate description of what happens. When you hit the circuit with a pulse it responds by generating a damped sinusoid, which is the electrical analog of the mechanical action of a drum head.

Table 1 shows a realization of this circuit when using one stage of an LM3900 type quad current differencing amplifier as well as the component values required to generate a number of different drum/percussion sounds.



Drum Sound.	C1	C2,C3	R1	R2	R3
Tom-Tom	.01	.001	68K	68K	10K
Conga	.01	.001	27K	68K	10K
Wood Block	.01	.001	27K	68K	10K
Heavy Bass	.01	.005	15K	27K	10K
Light Bass	.01	.005	15K	39K	10K
Clave	500pf	500pf	68K	68K	330

Figure 1 - Typical Drum.

Table 1 - Component Values for Various Drums.

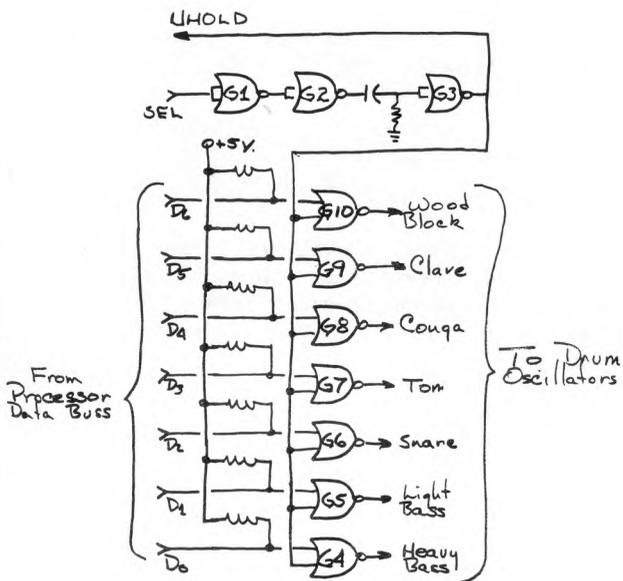


Figure 3 - Drum Interface.

Also, the output of G3 is routed back to a pin on the processor called NHOLD. This is an interesting input to the processor, because when this pin is grounded it causes the machine to STOP with the data and address lines held in the same state they were in when the hold command was issued. This has the effect of latching the address and data lines of the processor for as long as NHOLD remains low (a time determined by the R1, C1 time constant) and serves to stretch the micro-second write cycle out to the few milli-seconds required by the drums.

We need to discuss programming some, but before we do let's consider for a moment which memory location(s) we want the drum set to occupy. If you are using one of the larger systems by SWTP, IMSAI or MITS (or even some smaller systems such as the F-8), the problem is academic. These machines have provisions for output ports and you will use them - why fight the system. In these cases the SEL and NHOLD lines can be used for the handshaking "data ready" and "data accepted" lines respectively.

On smaller systems you will want to base your decision on what the cost (both financial and emotional) will be of using a specific location or group of locations. May I make a suggestions that might not occur to you otherwise? Use the same group of addresses that is occupied by whatever ROM your system has. This is not "ordinary" (you never write into the ROM locations with the usual programming because it won't do anything - they're Read Only Memory) but, there's nothing wrong with it and they have the tremendous advantage of being locations that are already decoded. On the SC/MP this is accomplished as shown in figure 4.

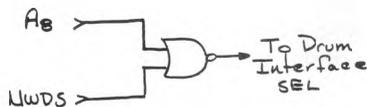


Figure 4 - SC/MP Address Decode.

PROGRAMMING

There's another big plus to using the ROM locations as an output port, this one from a programming standpoint. And we're both going to have to go slowly here because it can be confusing otherwise.

ROM, even in minimal systems, represents a chunk of memory - ordinarily at least 256 bytes. This means that in a machine like the SC/MP, which has 16 address "lines" (some are multiplexed onto the data bus, but forget that) only the most significant 8 bits are required to address the output port.

Most machines have a scheme of addressing memory which is referred to as "indexed" and they have, internally, one or more 16 bit wide "index" registers that can be used to "point" to a specific memory location. Since most of the processors used by hobbyists work on only 8 bits of information at a time it is obvious that it will take two "groups" of instructions to load the 16 bits into the index register - one "group" to load the "low order" 8 bits and a second "group" to load the "high order" 8 bits. But if we're using 256 bytes of memory occupied by ROM as a single output port, we don't have to worry about the low order 8 bits, they can be anything and we will still be addressing the output. Since they can be anything a still work, we can't have to worry about loading them and as a result we've saved at least three (and probably much more in a complete program) lines of program code.

Programming for the drums can be as simple or elaborate as you like. Regrettably, there is sufficient space here for only a few tips. The instructions for the experimenter's kit goes into much greater detail with sample program listings for a variety of machines.

Let me point out a few hardware considerations that will affect programming. Each bit of data in the accumulator at the time of the write operation will determine whether a specific drum sounds or doesn't sound. A typical coding arrangement was shown in figure 3 and with this arrangement the drums would be coded like this (remember that a 0 sounds the drum).

<u>drum sound</u>	<u>binary</u>	<u>hex</u>	<u>octal</u>
heavy bass	11111110	FE	376
light bass	11111101	FD	375
snare	11111011	FB	373
tom-tom	11110111	F7	367
conga	11101111	EF	357
clave	11011111	DF	337
wood block	10111111	BF	277

This is confusing. It would be much easier if when we were programming we could just write a 1 for the drum we want to sound. We can do that if we write a program flow that reads a byte of drum data but before writing it to the drums does an Exclusive Or Immediate with FF₁₆. As many of you will realize, this has the effect of complementing every bit of data so that the drum sounds we want can be written in memory in this more logical form:

<u>drum sound</u>	<u>binary</u>	<u>hex</u>	<u>octal</u>
heavy bass	00000001	01	001
light bass	00000010	02	002
snare	00000100	04	004
tom-tom	00001000	08	010
conga	00010000	10	020
clave	00100000	20	040
wood block	01000000	40	100

This is particularly easy if we want to sound two drums simultaneously; for example a heavy bass down beat and a snare drum at the same time would be in binary 00000101 (05₁₆) (005₈).

For very simple repeating patterns, this complementing action even has an advantage in that we can use FF₁₆ as a repeat indicator that is stored along with the drum data. Program flow would be 1) load drum data, 2) complement (XOR IMM. FF₁₆) check for zero (11111111 complemented is 00000000) and if zero start again, 4) if not zero, write accumulator to drums, 5) delay (tempo) 6) get next data and go to 2.

Various machines have an amazing variety of ways to test the accumulator (test for: zero, not zero, pos., neg., carry, overflow, etc.) but one thing they all have is a test for zero.

Notice that the 8th data bit in the arrangements that I've shown does not have a drum associated with it. There are two reasons, first, we've got a bunch of drum sounds already and secondly, we can use that 8th bit in our programming as an indicator that the data we've loaded is not to be played as drums, but to be interpreted as an instruction. A simple example would be to suppose that we have a rhythm pattern we wish to play 16 times and then stop. We can use a program based on the flow chart shown in figure 5 to accomplish this. Notice that in this case the program reads the drum data, tests to see if it is in fact an instruction, and if not plays it. When it does get to the byte that is the instruction it decrements the "count", saves the data back in the location it got it from and iterates the pattern. On the last pass through the program the decrement operation on the "instruction data" results in the last seven bits being 0, which tells the machine that it is through (or to go to the next pattern, or whatever). There are, of course, as many ways to handle this as there are people to write programs.

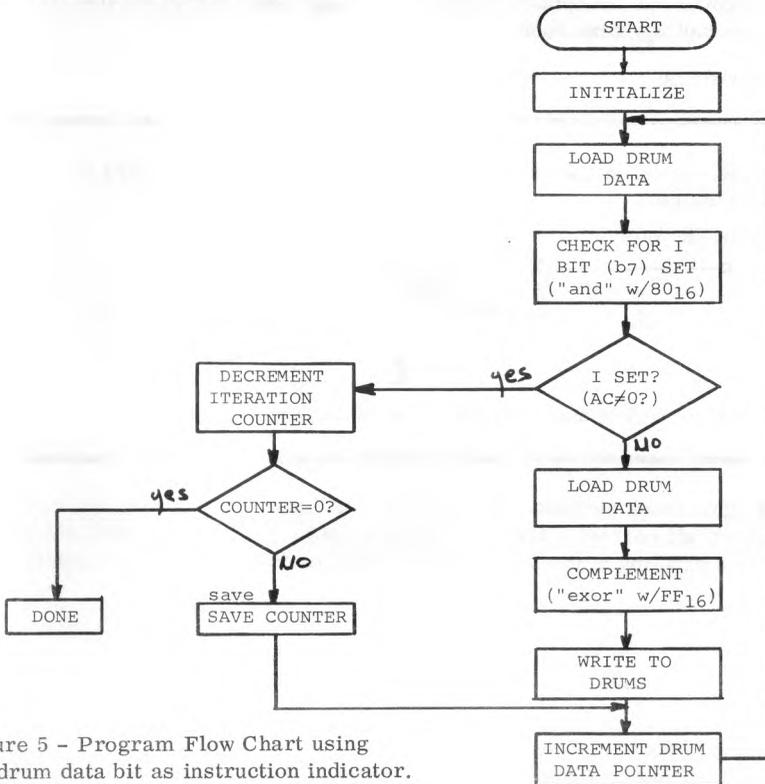


Figure 5 - Program Flow Chart using 8th drum data bit as instruction indicator.

Equipment Exchange

Equipment Exchange is a new addition to Polyphony, and 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: PAIA 2720/A plus -12, -14, -9, (2)4770, 4780, and 4710. Most modules operating properly, some need minor repair. \$220 for all, or modules available separately. Jeff Davies, 1118 Independence Center, Independence, MO 64057. Phone: (816) 795-0125.

FOR SALE: PAIA 2720/A with -9, 4710, 4720, and a PAIA 3712 Ping-Pong. All equipment operating properly, and has had little use. \$300. Bruce Fruechting, P.O. Box 1234, Las Vegas, NM 87701.

FOR SALE OR TRADE: PAIA 2720/A plus -11, -12. All built and tested, and less than a year old. Will sell or trade for amplification equipment. Mike Fennell, 712 Yorkshire Drive, Carlisle, PA 17013. Phone: (717) 243-8321.

Next Issue:

- * PART 2 OF FUNDAMENTAL MUSIC NOTATION: Timing.
- * MODIFY YOUR 4780 TO HAVE TWO INDEPENDENT TRACKING OUTPUTS.
- * BUILD A LOW COST AR ENVELOPE GENERATOR.
- * A SPECIAL "SPOTLIGHT" ARTICLE.
- * SOME "FAR OUT" NEW PRODUCTS.
- * MORE OF THE STANDARD GOODIES.
- * AND A BIG SURPRISE - Watch for it around Christmas!

We offer our apology for the delay in producing this issue of Polyphony. We were in the process of installing new printing and production equipment. The next issue is already in process and will be back on schedule, possibly in a new, larger format.

-Linda K. Brumfield-

PAVA POLYPHONY

P. O. Box 14359

Oklahoma City, OK 73114

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The mu:zines project is a non-commercial, labour of love archive of music production magazines from the 1970s-2000s.

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