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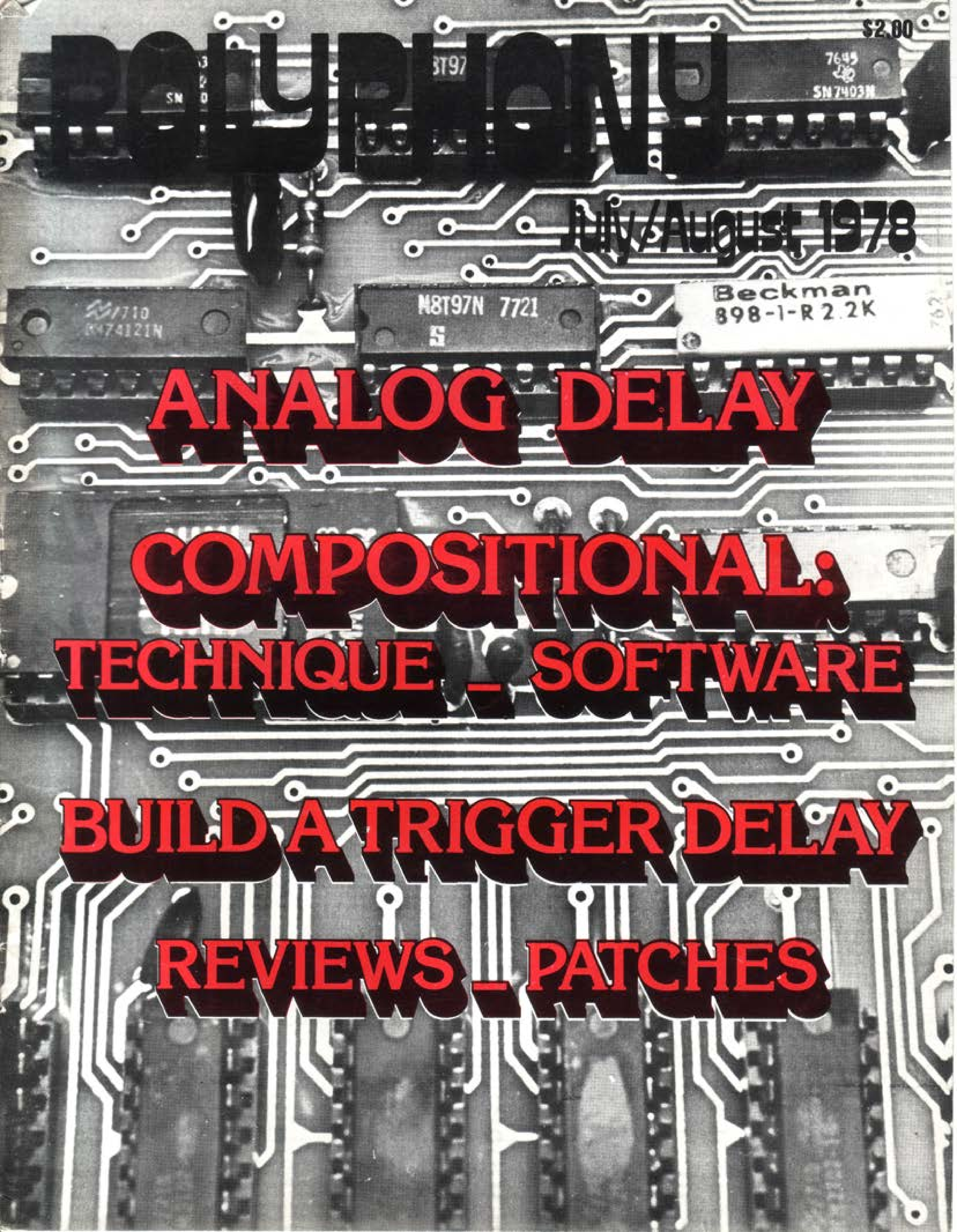
POLYPHONY

July/August 1978



ANALOG DELAY

COMPOSITIONAL: TECHNIQUE _ SOFTWARE

BUILD A TRIGGER DELAY

REVIEWS _ PATCHES

STAFF

EDITOR:
Marvin Jones

ASSISTANT EDITOR:
Linda Kay Brumfield

CONTRIBUTING EDITOR:
John S. Simonton, Jr.

PRODUCTION:
David Longmire
Donald Cooper
Marcina Howard

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VOLUME 4, NO. 1, 1978

ON THE COVER:

Digital Circuitry photographed by Marvin Jones.

Editorial

HELP US GROW!

When POLYPHONY started in 1975, the primary goal was to provide an intercommunication and users group for people with PAIA synthesizers. Initial response was good; POLYPHONY got off to a good start. Readers sent plenty of information for publication and, as PAIA employees had time to assemble articles based on past experiences with customers, the magazine began to grow.

At the end of 1976, we had a magazine which was literally too thick for us to staple. We had to make a change. Readers had been asking for more frequent publication and larger issues. We didn't have the time for more frequent publication, but we expanded to the larger 8X11 format so we could get more information in a given number of pages. Readers were again receptive and began submitting more material for us to publish. An added help during this period was an increase in public interest in synthesis, and a growing number of people looking for a publication such as POLYPHONY. During this period we had a barage of requests from owners of other types of equipment to start writing about their gear, how to modify it, and get more out of it. Unfortunately, we had to concentrate on the PAIA users group format because POLYPHONY was still a part-time venture for those involved, and PAIA was subsidizing part of our growth.

It is time to grow again, and POLYPHONY is going through some very important changes. POLYPHONY staff is devoting a larger part of their time to ensure smooth productive growth. POLYPHONY has been getting thicker with each issue, and will continue to do so. Since the beginning of 1978 we have been publishing roughly on a bi-monthly schedule. With the formal publishing schedule we have now implemented, we should be finishing the last issue of Volume IV around the end of 1978. This should resolve the "volume versus year" problem. At that point, we should be in a position to increase to six issues per volume for bi-monthly publication. We will be running more pages per issue. More photographs. Perhaps color. All this growth takes money which must come from two places. Advertisers and reader subscriptions.

We have been looking for firms who may find POLYPHONY a beneficial medium for advertising, and have found a number of people who are very excited about helping us get off the ground and, of course, reaching YOU with their ads. Please support our advertisers whenever you can. Tell them you saw their ad in POLYPHONY and that you like what they are doing. We stand behind all of our advertisers as reputable firms with products and services which are very closely related to what we want and need as synthesists. You won't see any poster companies advertising in POLYPHONY. You won't see any home correspondance school reply cards stuck between the pages. Support our advertisers. TELL 'EM YOU SAW IT IN POLYPHONY!

The biggest part of POLYPHONY's growth comes from subscriptions. At this point, the majority of you are PAIA customers, although there is a good percentage of you who use other gear and enjoy POLYPHONY's general interest articles. We want more readers, so we are making a change in the editorial policy of POLYPHONY. We are finally changing from a users group for one brand of equipment to a general purpose applications magazine for all synthesists, regardless of who's equipment they are using. With all the mail we were getting about how to work with other gear, we realized that very few manufacturers provide advanced applications information after the sale. Some provide users manuals and patch books with the instrument. Some don't even do that! When the user begins advancing and wants to interface several instruments, modify for added or improved features, or even repair the instrument ....."Sorry, you're on your own!". POLYPHONY is now here to serve YOU as well as the beginner or experimenter. The Mini-Moog modification in the last issue should indicate our intentions. Modifications for other brands are in the works. TELL YOUR SYNTHESIST FRIENDS about the change in our editorial policy. Users of other types of gear who may have previously had only passing interest in POLYPHONY should now find our contents much more applicable to their systems as well. We will continue to have homebrew projects, modifications, patches, and so on. But hopefully, we will also be adding regular columns by familiar POLYPHONY writers as well as some new faces, test reports and evaluations of equipment by various manufacturers, more product announcements, industry news, perhaps some interviews, and a lot more. Really --- we have some HEAVY DUTY material we are working on for this fall. Spread the word.

If you have friends who have successfully modified or expanded commercial equipment, or who have some unique applications, have them write it up as an article for us. Remember -- we pay! Also be sure to note our NEW ADDRESS if you wish to send anything to us. (P. O. Box 20305, Oklahoma City, OK 73156) If you know of manufacturers who may like to reach the POLYPHONY audience, tell them to contact us, or let us know and we'll contact them.

Help us grow. Get a friend to subscribe. Support our advertisers. You will be helping POLYPHONY grow into a whole new role, continuing to provide the best information on experimental electronic music for everyone. THANKS.

Marvin

LETTERS! LETTERS! LETTERS!

TAPE EXCHANGE PROPOSED

Hello!

I'm wondering if you might consider setting up some kind of service for those of us who are getting into home or semi-professional recording and would like to try private distribution of our products. What I envision is some kind of review column for those who send their product in. Then for those reviewed, advertising space giving details about purchasing the tapes reviewed. I think that this would be a wonderful alternative to the corporate controlled system of music marketing that exists today. Such a system would be particularly helpful to all of us electronic music enthusiasts who want to contact other enthusiasts and hear their work.

Thank you very much for your time.
Chuck Larrieu
Corte Madera, CA

POLYPHONY would indeed be interested in helping musicians distribute their their home recordings. There are several ways such a project could be handled, and I would like to solicit opinions from our readership on how they think it would best be implemented.

First, the entire operation could be completely outside of Polyphony with one or two people solely responsible for maintaining a library of tapes, cataloging recordings, sending lists to interested persons and selling tapes to other readers. Polyphony could aid by publishing reviews of the new recordings as they are received and periodically listing tapes that are available. Another alternative could be to operate an individualized exchange program. As a reader finishes a recorded work, or compilation of works, a copy could be sent to Polyphony for review. Interested readers could then send the musician a recording of some of their own work, and in exchange he would make a copy of his reviewed composition and send it to the reader. Possibly this could be handled as an extension of Local Happenings. Readers could ask to be listed in a TAPE section, along with the format they have available to work with - reel, cassette, or 8-track (or even video, I guess!). Interested parties could then contact others directly to work out a trade. Of course, those artists who have published their work on album or tape can send a copy to Polyphony for a regular review. With reviews we inform the readers of cost and ordering procedure. I'd like to hear your suggestions as to how we can make this system work. Send your ideas to us; we'll print them for everyone's consideration, and hopefully we can get a system such as this into operation a couple of issues from now.

-Marvin-

COMPUTER AIDED MUSIC

Marvin-

I was just reading over my old issues of POLYPHONY and found some things I want to comment on.

Concerning Charles Bodeen's letter on automated music typography (Nov.'77 issue -- you know, the one with the great front cover), I'd like to offer an alternate approach to a "universally accepted" computer music language. The problem I see with a high-level language of this nature is that it can't take into account many aspects of computer generated or computer controlled music. Obviously the technique used will affect the subroutines that make up a language -- that is, the actual commands of a language will not be affected, but the way they are executed will differ greatly from a computer-controlled analog synthesizer and a wholly digital synthesis scheme. What that means, then, is that a language carried beyond the format and logical structure stage into actual working implementation on a particular system, is bound for obsolescence. How useful will the MUS1 PROM be on a totally digital PAIA 14700/S? Again, the commands for the language won't change, but the implementation will and probably new commands will be added for "extended" versions of the language (which often tends to confuse the language). It would be nice to create a language which could, indeed, cover all possible aspects of music, I just don't think it is possible, or at least, it will be a big, clumsy, expensive system. More importantly, it will take a long time to develop and would require establishing totally new (and totally universally acceptable) conventions for certain aspects of the music. One of these new conventions would be music

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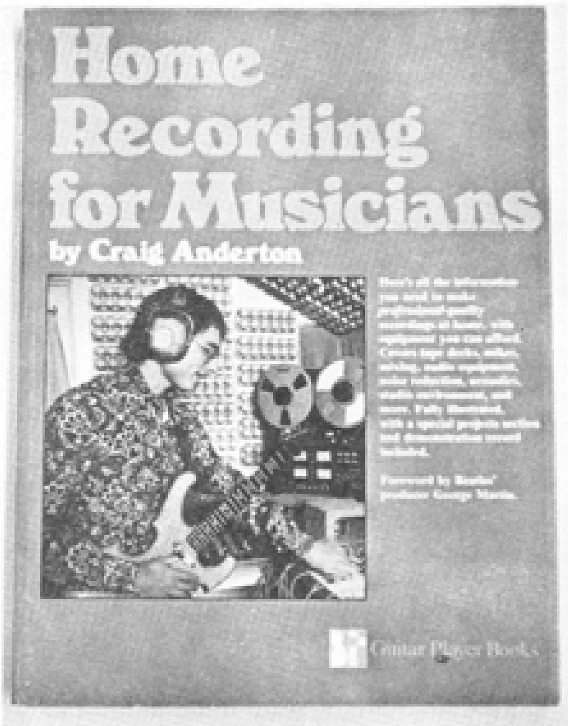
POLYPHONY REVIEWS:

By: Marvin Jones



HOME RECORDING FOR MUSICIANS
By: Craig Anderton
Guitar Player Books, Box 615,
Saratoga, CA 95070 - $9.95

There's no two ways about it -- the 4 channel multi-track recorder has changed the paths of most any amateur or semi-pro musician who has been exposed to its power. But it's not surprising that many musicians get somewhat confused with all the new terminology and technical details involved with their new venture. If there is any one place the aspiring musician can turn to find help, it's "Home Recording for Musicians". As is typical of Craig's writing, the reader can tell that the book was written from Craig's everyday struggles and experiences with his equipment. One tends to build a very close friendship with Craig through his writing, because he implies a true concern and desire to help you.

When you don't know where to start, most people agree that the beginning is best. In chapter one, just enough technical background is given to allow the musician to understand what is being said in later sections of the book. Tips are included on buying recorders and planning a home studio, so this book should become part of your home studio long before the equipment does. Later chapters discuss features and operation characteristics of consoles, microphones, and accessories. The "Recording Techniques" section provides all those handy little tips that normally take you forever to learn, allowing a much higher initial concentration on the creative aspects of home recording. Important, but often neglected, information on tape editing, assembling, handling, recorder maintenance, and even the psychology of tape recording are also given a good share of the book.

The final section of the book covers a number of general audio processing elements as construction projects. Described are mixers, submixers, reverb sections, pre-amps, noise gates, tone controls, and so on. Each could be used separately for special effects processing, or Craig describes how to tie them all together into a small mixing console for a 4 track studio.

With tape recording being such an essential prerequisite for realization of synthesized music, this book should easily rank among the top "handbooks" to be kept in your studio/lab reference library. It will teach you the basics; it will save you time; it will improve your recordings; get it.

"The Death of the World of Now", music by: Ron Di Iulio
Available for $6.25 postpaid from:
Trifid Publications
904 Keller St.
Benbrook, TX 76125

I'm really glad to see so much "home-brew" music being released non-commercially. "The Death of the World of Now" is a concept album about the destruction of the world, the resulting havoc, and final restoration into a better world than before. Like "Astral Warrior" and the "Craig Anderton Music Tape", which we have reviewed in previous issues, this album was recorded completely in Ron's home studio using primarily PAIA equipment and, perhaps, some Arp gear too. The music is mainly a "jazz trio" (piano, bass and drums) with synthesizer lead lines and effects. Unfortunately, the music is generally mixed too far back to keep track of the actual musical progressions. The music serves primarilly as a backdrop and catalyst for the lengthy poetry from which the entire album concept was adopted. Ron, and the poet/narrator involved with this project, are also adapting this album for use as a soundtrack in a regional laser/planetarium presentation. All in all, quite an extensive undertaking. I would recommend this album for those interested in seeing how others are releasing their music, and to hear other productions stemming from a home studio type environment. If you are more interested in just hearing synthesized music, you may want to wait for future solo releases by Ron where you will be able to really hear his music and synthesis highlighted.

Hope by Klaatu
Capitol ST 11633

Some of you may remember Klaatu as the band which was supposedly the reunited Beatles when their first album was released a couple of years back. I never heard that album, but from the second album I can surely see how someone could confuse them for the Beatles. More specifically, Klaatu seems to have been influenced by the Beatles arranger and producer, George Martin. The songs themselves lack a bit of flair and originality, and after several listenings were obviously not written by Beatles. The slick commercial nature of this album implies that members of Klaatu have been around for a while, perhaps as studio musicians or staff writers; they know their job. The overall theme of Hope concerns itself with a lonely lighthouse (laserhouse?) keeper on the planet of Politzania who sends out continuous pleas for rescue and help for the decaying planet. The album starts off with "We're Off You Know" which is a lighthearted, old-timey piece similar in many respects to the Beatle's "Your Mother Should Know" or "Fool on the Hill" The instrumentation and scoring are pulled off well, as is the mixing of the wild array of instruments. The following "Madman" is filled with chorused acoustic guitars and organ which show off Klaatu's mastery of studio processing equipment. The processed bass on this song is a good test for most speaker systems, and is well executed. "Around the Universe in 80 Days" features flanged fuzz bass which explodes with power. The backwards cymbals, flanged drums, and synthesized "computer" sounds are all standard effects, but work well in this piece. Perhaps the high point of this song is the voice processing using backwards tape echo, flanging, and pitch shifting. The result is a good alien voice simulation. "Long Live Politzania" which closes the first side is a poor excuse for an anthem. It's no wonder the empire fell. The flanged concert band doesn't seem to be an appropriate application of technology, and the William Tell Overture rip-off is cute but doesn't seem to work with the song at all. "The Lonliest of Creatures" which opens the second side features some extensive multi-tracked vocal chorus much like Queen has done, including the use of multi-speed recording during the track building procedure. The result -- a most effective mass chorus. In "Prelude", the deeply flanged organ in the middle of the song is an excellent effect, but seems out of context with the rest of the song. There are tape effects galore, and a good processed piano part. The strong part of "So Said the Lighthouse Keeper" seems to be the miking of the drums and a convincing voice processing effect. The closing "Hope" is a slow, commerical ballad which presents a moral for this album's drama, saved only by the vocal harmonies and weeping guitar a la Abbey Road.

I still haven't made up my mind whether I like this album or not, yet I keep listening to it. Effects are good, production is good; engineering is good; but the material is a real distraction to what artistry is presented here.

Pyramid by the Alan Parsons Project
Arista AB 4180

One begins to suspect that Alan Parsons was, perhaps, a sequencer in a previous life. This album is an excellent recapitulation of his magical "hit record" formula which has appeared on his last two "solo" albums. Persistent useless drum beats, droning chords, vocal choirs, and a touch of "movie music" orchestration have become synonymous with Parsons. To this list he has now added .... overuse of time delay processing. Boy, I've never heard so much flanging, echo, chorusing, stereo imaging and motion in one place before. It is evident that Parsons likes his delay unit, but enough is enough. "Voyager" starts off with some free form echo effects but soon reverts to the mechanical drumbeats and chorused guitar which is the signature for this album. "What goes up...", and "Eagle Will Rise Again" are saved primarily by Colin Blunstone's vocals, which have been absent from pop music far too long. The autoharp in "Eagle..." is a cute effect that helps add a bit of Zip (sic) to the piece. "One More River" is an average straight rock ditty which is disrupted by interjections of non-tonal echoed oscillator sweeps. It sounds like someone trying to decide whether they want to be a synthesist or not. Another of Parson's cliches pops up here - the brass (especially French Horns) ensemble. "Can't Take it With You" closes the first side, and is familiar to many as it is getting a good deal of airplay right now. The synthesized whistling is well executed, and seems to have a bit more feeling than Tomita's whistling synthesis. The tricky change to a 15 time signature brings the first side to a decent conclusion. Side two tends to be a bit more electronic oriented, and starts with "In the Lap of the Gods" which sports an excellent smooth flanged electric bass part. "Pyramania" is short and cute, and has a good synthesized tuba solo (watch the speakers). "Hyper-Gamma-Spaces" is the big instrumental for this album. As in the past, the main ingredients are droning choirs and orchestras, endless drum and bass rhythms, and so on. The bit of sequenced bass background seems to fit into Parsons' structure well; not nearly as mechanical as it would be in other works. This side closes with the slow ballad "Shadow of a Lonely Man" which seems to draw a lot from "Fools Overture" by Supertramp.

I'm not sure where Parsons is headed. He is definitely a good producer. Any one of his recent albums is very good, but they are all basically the same. Perhaps he should go back to producing other bands so there would be more creative voices involved in the album's decisions. Perhaps he should find new composers to collaborate with instead of always using Woolfson. Perhaps his next will be better.

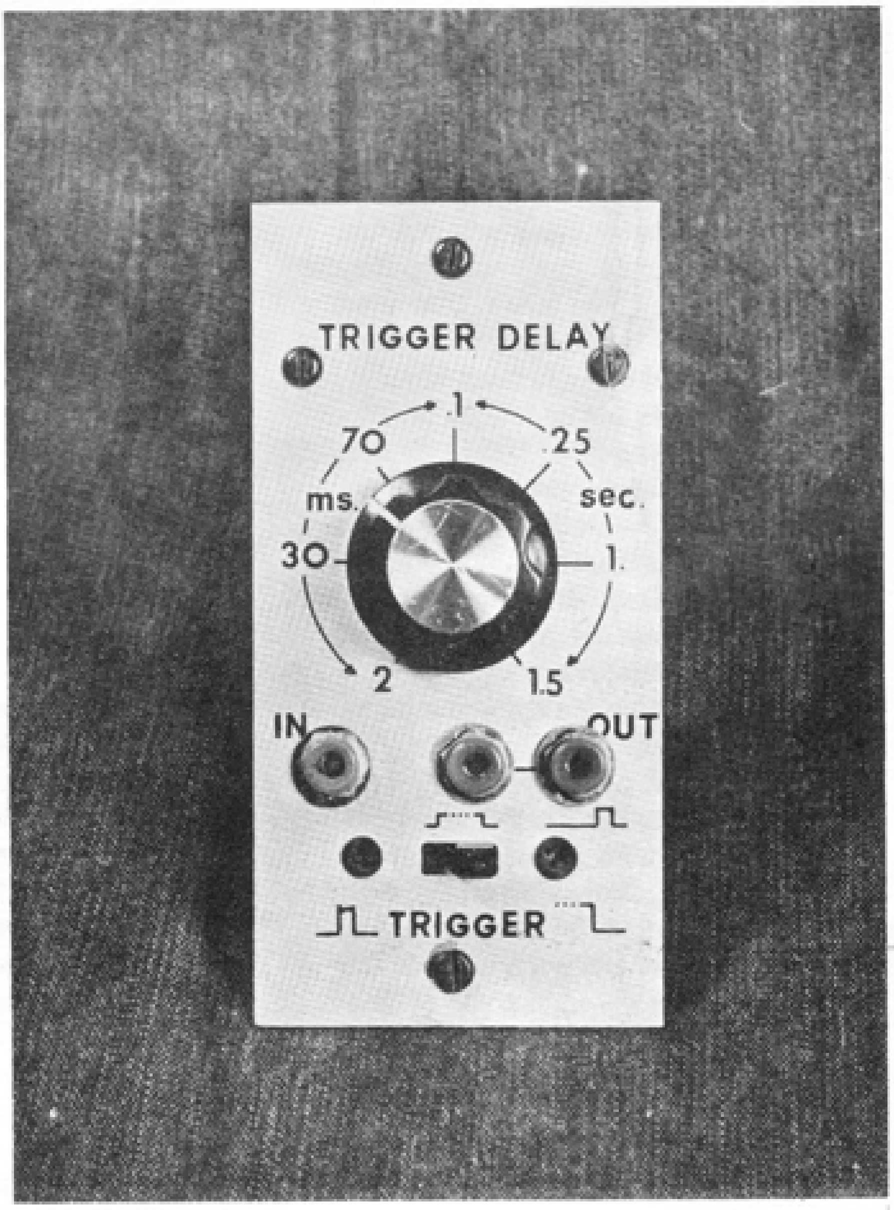
UK by UK
Polydor PD-1-6146

This is one of the strongest debut albums I've heard in quite some time. Before even listening to the album, you can imagine the power from the impressive lineup of former members of Yes, King Crimson, and other innovative bands. Keyboardist/violinist Eddie Jobson plays a big part in the electronic attitude of the band, and drummer Bill Bruford has developed a keen sense of complex time structures which definitely sets the group apart from other progressive bands. The overall feel is similar to Kansas, but with an increased tendency towards jazz and electronic influences. The opening suite, "In the Dead of Night" displays most of these qualities. Part one features a complex

——— continued on page 33 ———

CONSTRUCTING A TRIGGER DELAY

BY Larry Pryor



Here is a handy circuit designed to use in conjunction with AR and ADSRs. It is a step and trigger delay. It can delay the start of a step or trigger pulse from the keyboard. The delay is from 2 ms. to about 2 seconds. When a key is pressed it will output a step or a trigger pulse after a preset time determined by a front panel control. If a key is struck again before the timing cycle has completed, it will re-set and start a new cycle on the last key down. Figure 1 shows the complete schematic.

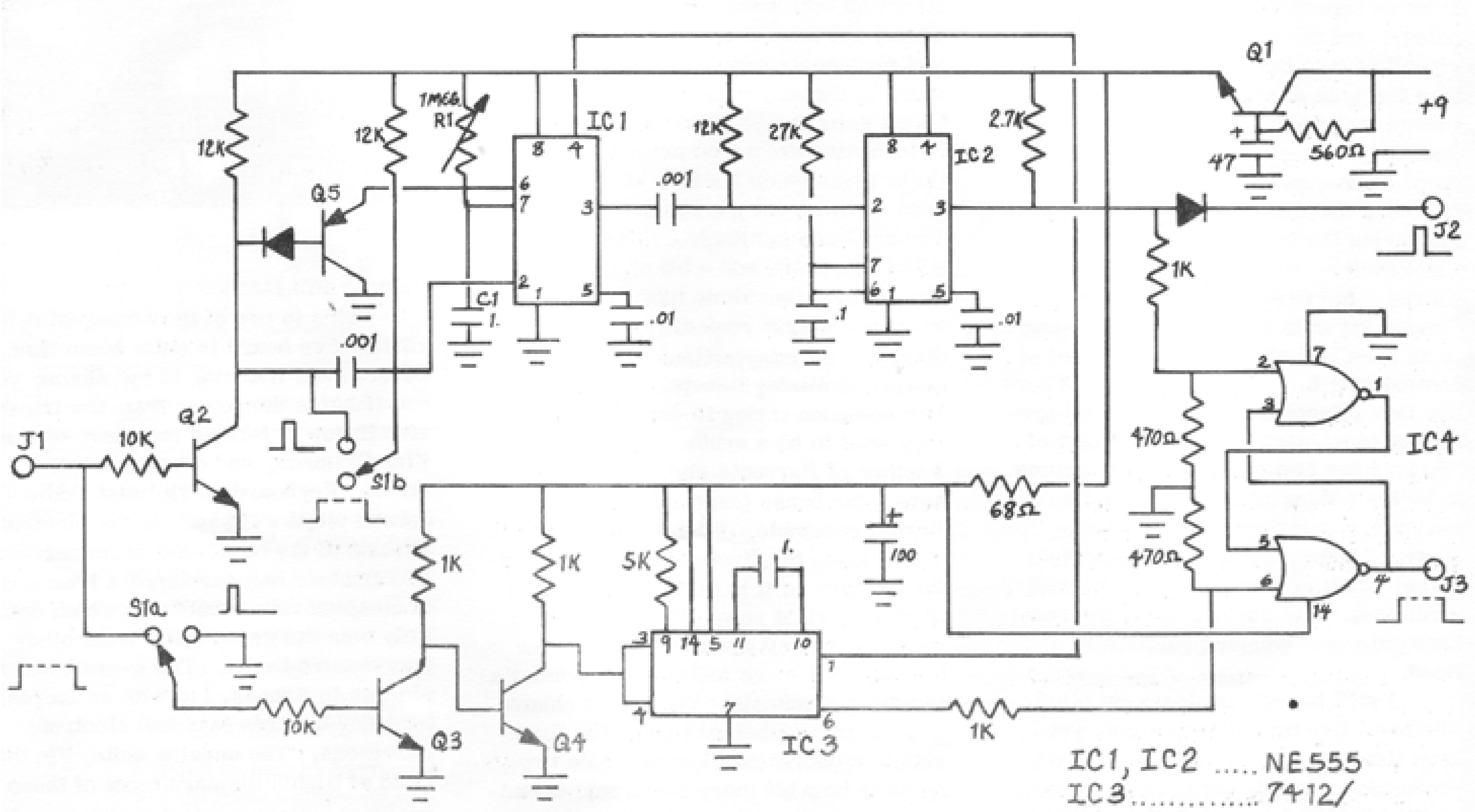


Figure 1

A step or trigger pulse is inputted into J1 and inverted by Q2, its collector is high when there is no input. The first 555 timer, IC1, is a variable one-shot. R1 and C1 set its "on" time. R1 is the front panel control. With the 555 hooked up in this configuration, it will "one-shot" on a negative going pulse. So when Q2's collector goes low the cycle begins and pin 3 of the 555 will go from low to high and back to low again when the cycle is complete.

S1 on the panel is labeled _⎍_ and _⌐¨¨¬_ or trigger and step. When S1 is in the _⎍_ or trigger mode, and a trigger pulse from the keyboard is supplied, Q2's collector will be the inversion of this and will fire the "one shot", IC1. When a step pulse is used the delay begins when the key is let up because the capacitor C1 is being held down by Q5. When the key is released, Q5 no longer shorts C1 and C1 begins to charge, completing the cycle.

Now, when you input a trigger pulse and hit the key before the timing cycle is complete, the cycle will start on the last key down. Why? Because of Q5. If you look at the location of Q5, you will find it placed right across the timing capacitor shorting it out and beginning a new cycle whenever another pulse is entered before the timing cycle is complete.

Now place S1 in the _⌐¨¨¬_ or step mode, and input a step trigger from the keyboard. This now places a capacitor between Q2's collector and pin 2 of IC1. The timing cycle begins on key down because Q5's output is integrated and begins the cycle. This also places Q3 and Q4 and IC3 in the picture. Q3 and Q4 are buffers for IC3. When you press a key down and hold it, you will get a pulse output from IC1. If you press a key, release it and press it again you get a pulse out. But, if you press a key, release it and don't press a second key, no output appears. You have just cancelled the instruction to delay with the aid of IC3, a TTL 74121, one shot. IC3 is wired in such a way that it will fire on the falling edge of the step. When it sees the falling edge, it fires. This output pulse goes to pin 4 of the 555's. Pin 4 of these ICs are labeled re-set. Normally this pin is held high for "normal use", but taking it low for a time re-sets the one shot. Pin one of IC 3 is normally high for "normal use" of the trigger delay except when cancelling the output pulse altogether.

Now we get to IC2 and IC4. IC2 is a fixed one shot that is used for the new trigger pulse output. It fires when IC 1

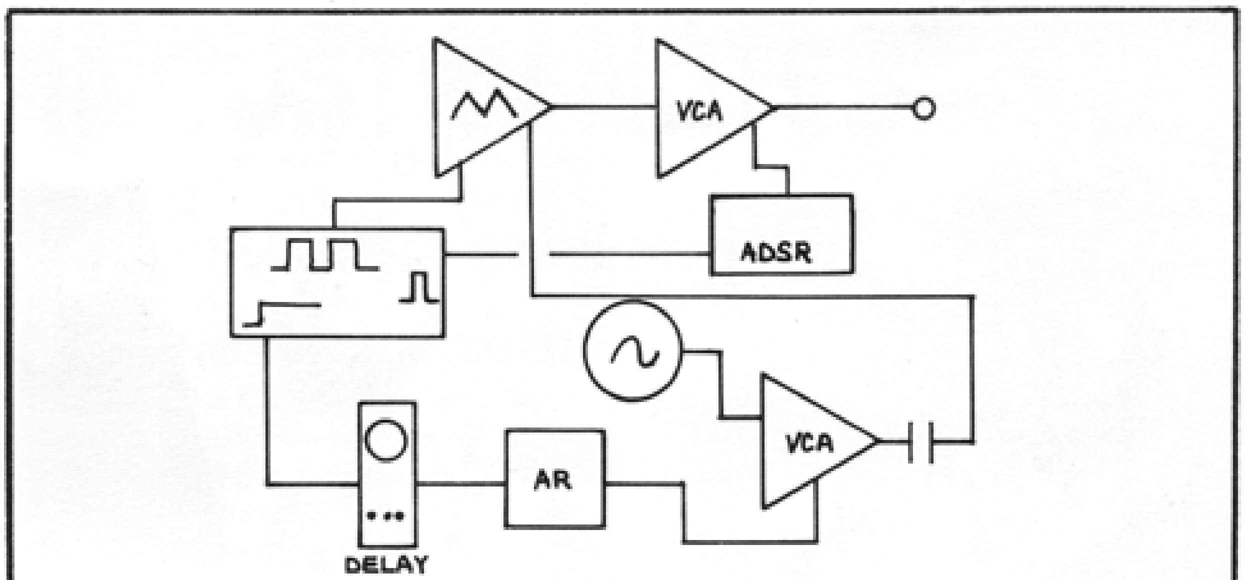
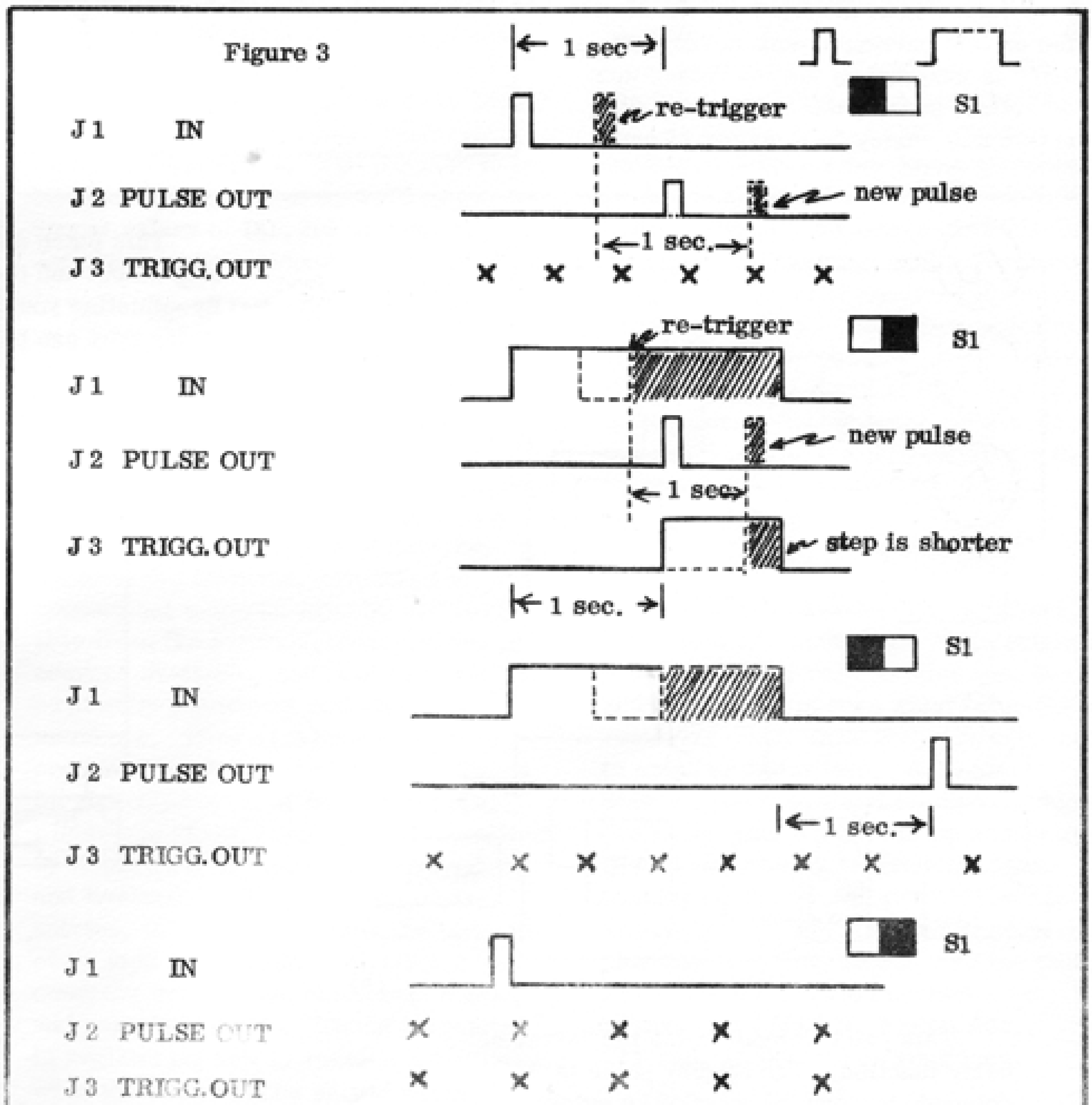


Figure 2

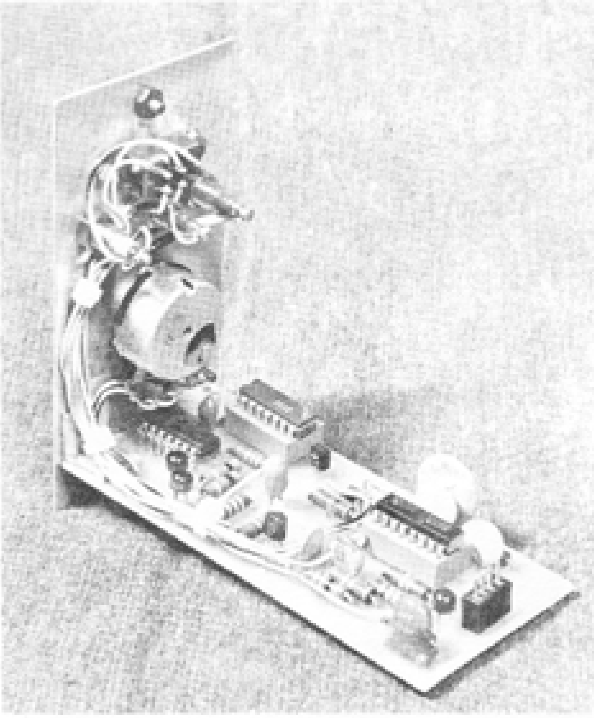
goes from high to low and then fires a fixed output pulse of about 2 ms. to J2. This is the delayed trigger. IC4 is a Quad NOR gate and is wired as a set-re-set type circuit. Pin 3 of IC2, the delayed trigger, is used to set the circuit, and latches pin 4 high. When the step pulse is released, pin 6 of IC3 fires and re-sets the latch and pin 4 goes low. What we have here is a delayed step which outputs at J3. This mode works only when S1 is in the step mode, and a step pulse from the keyboard is used. (NOTE: This trigger delay works fine with the PAIA D/A converter, use the "D6" output for the step input, this will give step and trigger delay.) This is very handy if you want delayed vibrato, see figure 2.

If all of this has you thoroughly confused, we'll add to the confusion with the timing chart shown in figure 3.

Figure 3



R1 IS SET FOR ONE SECOND DELAY



All four ICs and associated parts can be assembled on a 2" X 4" printed circuit board or perf board. With the values of R1 and C1 as shown in figure 1, 2 ms. to about 1.5 seconds can be expected. These values can be changed to expand the time. Use good quality capacitors for C1.

The photograph, figure 4, shows the orientation of the front panel. Notice that on this particular unit the circuit board is mounted on the top rather than the bottom of the panel. This was done on this unit purely as a matter of convenience in fitting it into a custom system configuration. The circuit board can be mounted in any convenient location.

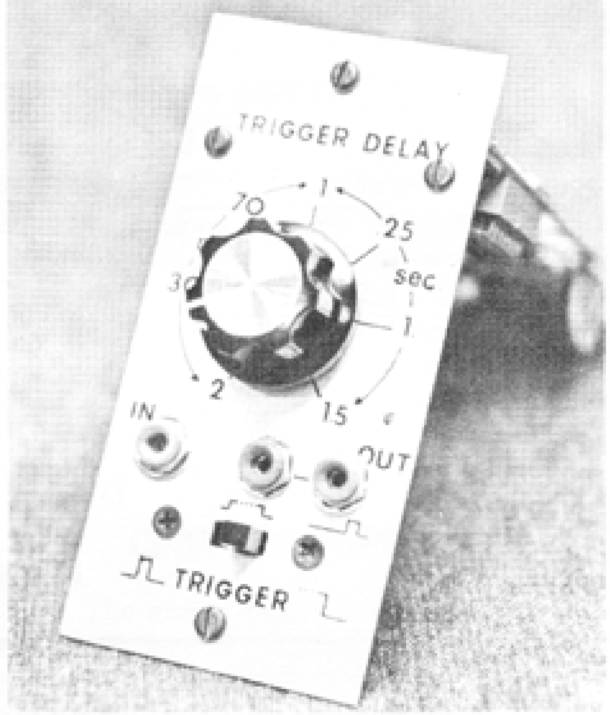


Figure 4
photographs by: Larry Pryor

Calibration of R1 was done with a triggered sweep scope, and the delay time read directly off the scope. See figure 5. Example: The time base is 500 ms./division and you press a key down, this starts the sweep. When the delay is over a pulse will appear. If the pulse appears after the scan has passed three divisions, then the delay time is 3 X 500 ms. or 1,500 ms. or 1.5 seconds. Or, you can guess at the time delay like you guess at the time of the AR, ADSR, Filters etc. The one thing lacking is calibration.

Figure 6 provides some sample patches to aid in familiarization and initial experimentation with this module. Like all new toys, you will be busy for hours.... have fun.

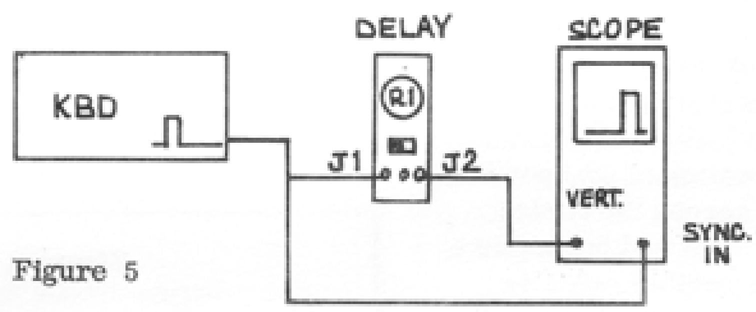
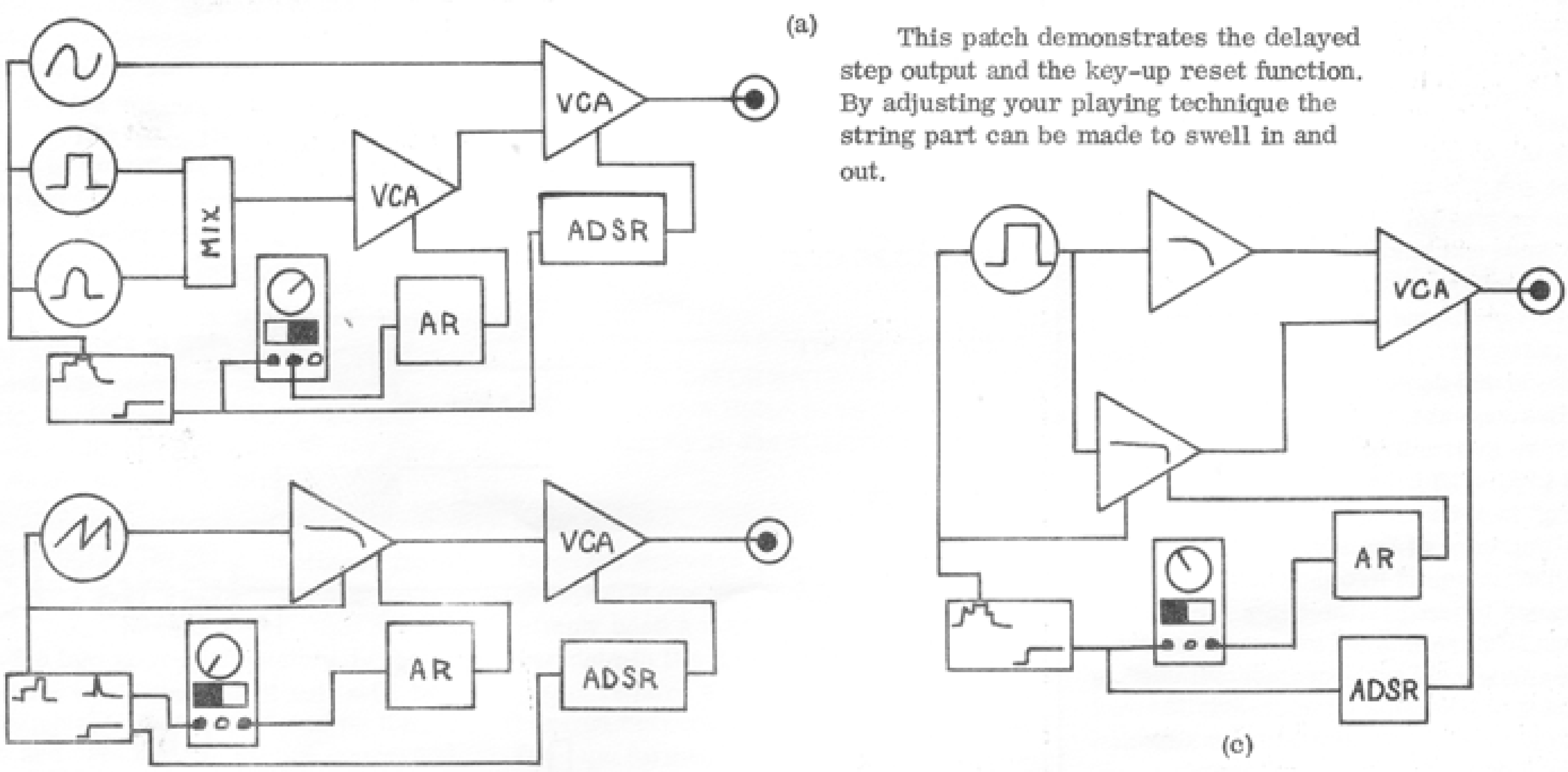


Figure 5



(a) This patch demonstrates the delayed step output and the key-up reset function. By adjusting your playing technique the string part can be made to swell in and out.

(b) This patch demonstrates the "straight" delay function. The trigger pulse is delayed, but may be stopped by quick, successive key depressions.

(c) This patch demonstrates the delay period starting on the release of a key and may be cancelled by depressing another key.

Figure 6

A COMPOSITIONAL METHOD FOR ELECTRONIC COMPOSERS

BY: DAVID ERNST

All forms of creative expression embody structural principles regardless of how avant-garde or 'far out' the end result appears. Musicians, along with other artists, view the concept of structure from two perspectives: practical versus theoretical. Composers generally tend toward the former, whereas the latter view is often more important to musicologists and theorists. Needless to say, no musician should restrict his or her knowledge to either practical or theoretical considerations; balance between the two is essential because they are truly inseperable.

One difference between practical and theoretical approaches is temporal -- composers (practical) are concerned with the present (now I want this type of sound....), whereas musicologists and theorists look at music that has been written to encompass both the recent and remote past. As a result, it is often said that composers break the rules established by theorists, but this statement is not a true indication of the evolution of compositional aesthetics.

With respect to structure, composers learn much from studying music of the past; having acquired this knowledge, composers gain the ability to develop individual concepts that stem from the work of their predecessors. For a contemporary composer the study of earlier music often centers around compositional techniques so that fluency of expression in a particular style or medium is attained.

Throughout past centuries numerous treatises and books have been written to teach the techniques of musical composition, and many of these texts have been, and still are, quite useful for both students and composers. Although true creative expression cannot be taught, those techniques necessary for its development and articulation may be practiced and mastered. In such treatises compositional rules are often formulated as an aid to composers -- not as an obstacle to composition. The following essay is patterned on this concept. Its purpose is to illustrate one possible method for the realization of an electronic composition, and it is certainly not intended to be totally inclusive with respect to compositional techniques. Guidelines are presented to assist in the formulation of a compositional method so that specific techniques serve only as the means to an end. Hopefully the following material will suffice as a point of departure for those interested in electronic composition.

MICROCOSMIC STRUCTURE

SOUND SOURCES

Since the sounds and their transformations common to electronic music are frequently only remotely associated with those of conventional music we will attempt to develop a compositional method suitable for all types of sound. Because there are no restrictions as to the nature of the sounds that we may choose, we must define the sonorous characteristics of each composition. In other words, it is up to the composer to define the limits within which he will work. As soon as the sonorous nature of a composition is determined (probably a vague or general notion at this point), the ingredients for a structural framework are present.

We will deal purposely with a simple sonorous structure so that our compositional method remains visible. The choice of one basic sound will thereby enable us to concentrate upon structural relations more readily than if we choose a variety of sounds. At first this self-imposed restriction may appear to be too severe, but we shall soon become aware of its many advantages.

Thus far we have spoken of structure and its importance in an electronic composition. What is structure? It is nothing more than a self-imposed restriction that generates sonorous COHERENCE. With respect to our compositional method, the fewer the basic sounds the greater is the possibility of sonorous coherence. However, this alone does not guarantee a 'good' piece; our discussion only includes the technical components of structural coherence so that all other elements (imagination, emotion, etc.) must be accounted for by the individual.

If we choose only a single sound as the source of a composition it is obvious that we run the risk of sonorous boredom. In order to prevent this situation, yet to retain structural coherence, we must be able to generate 'new' sounds from this source. This is possible via electro-mechanical transformations of the original sound as filtering, modulation, frequency shifting, reverberation, envelope shaping, phasing, panning, mixing, tape transposition, tape reversal, tape loops, tape delay, splicing, and all combinations of these (and other) techniques. After all of these possibilities have been explored the composer may choose those sounds which he prefers in order to proceed to the next compositional stage. The following steps summarize this initial process.

1. Choose a single sound as the primary source for a composition.
2. Subject this sound to a variety of electro-mechanical transformations.
3. Select only those sounds which conform to the desired sonorous nature of the composition.

At this point, a library of interrelated sonorous materials has been assembled. Although the individual sounds may appear to be quite diverse they are structurally related because they originated from a single source. Thus far our compositional method has enabled us to assemble a library of sounds that includes both a high degree of sonorous coherence along with necessary diversity.

David Ernst holds a Ph.D. in Music Theory & Composition from Rutgers University and is the author of The Evolution of Electronic Music and numerous articles on electronic music theory and composition. He is currently the Director of the Electronic Music Studio at York College, City University of New York.

SUBSTRUCTURE

Now that we have decided upon the type of sounds to be used in our composition we must begin to arrange selected sonorities as they will appear in the final version. Again, simplification of this process will enable the composer to exercise maximum control over the end result. One of the first questions to arise concerns the beginning of the piece, and we may choose any sound from the previously assembled library (step #3). The next step is perhaps the most difficult for there are no extra-musical guidelines to assist the composer; the second sound (or possibly silence) must now be chosen.

An easy way to accomplish this task is to imagine what type of sonority should follow the initial sound, and then try to find this sound (or an acceptable substitute) from among those of the library. The structural value of the sound library becomes increasingly important, for sonorities chosen from the library are at least physically related to each other; this acoustic interrelation helps to establish a coherent sonorous structure so that a more or less logical succession of sounds results.

After the first two sounds have been chosen they should be listened to carefully, and the next sonority may be obtained from the sound library in the same manner as the previous sound was acquired. Repeat this process until a complete sonorous idea evolves. Do not worry about the overall duration of the sequence of events at this point, for there are no standards upon which to base your work.

Since our compositional method has provided a physical structure for the piece, at least some degree of logic is maintained among the sounds. The more sensitive and imaginative the composer, the greater will be the 'musical' value of the composition. Such value judgements, however, are too subjective for the present discussion, and they are best discussed in person with an experienced composer. For now it will be sufficient if you listen carefully to all that you do, always striving for complete satisfaction with the resultant succession of sounds. The following steps are the continuation of steps 1-3, and they summarize the second stage of our compositional method.

4. Choose a sonority from the sound library as the initial event of the composition.
5. Choose the second sound from the library.
6. Repeat this process until a complete sonorous idea is developed.

A relatively small part, or subsection, of the composition is now completed. Listen to it repeatedly so that its sonorous characteristics become familiar.

MACROCOSMIC STRUCTURE

Up to this point we have been concerned with minute details in order to generate a foundation upon which an entire composition may be realized; the logic supporting our course of action stems from the development of a compositional method. Further development of this method will promote the completion of an entire electronic work.

Now that the first subsection is finished, the nature of the next section must be determined. This may be accomplished by forming new subsections from the sound library, or by electro-mechanical transformation of previous sections. In either case, several critical decisions must be made, including: the number of subsections (to determine the duration of the composition), and sonorous interrelations among subsections (to establish varying degrees or homogeneity and/or contrast). All of these considerations are associated with the overall structural design of the piece, and they may be incorporated within our compositional method.

The chief value of the compositional method has been to generate a variety of diverse sonorities from a single sound source, from which a library was assembled and a subsection eventually derived. As previously stated, dealing with subsections elevates structural concepts to a higher level. We may now apply those same electro-mechanical procedures (used to form the library) to the first subsection (step #6) in order to develop enough material for an entire composition. In this manner any number of interrelated subsections can be produced. Figure 1 graphically depicts this entire process.

Another application of the compositional method for the derivation of subsections is shown in figure 2, whereby many different sounds are extracted from the library to produce interrelated subsections of a possibly more diverse sonorous nature. Since both methods are of equal importance we will discuss their particular advantages with respect to the overall structure of the composition.

The degree of homogeneity and/or contrast among subsections will ultimately determine the duration of a piece. As before, these choices remain the responsibility of the composer, but a wise application of our compositional method may make this task easier. Method A (figure 1), for instance, will most likely yield a shorter composition than method B (figure 2) because of the restricted introduction of 'new' sounds from the library in the former process. Continual development of previous material characterizes the nature of method A, whereby a rather homogeneous sonorous texture results.

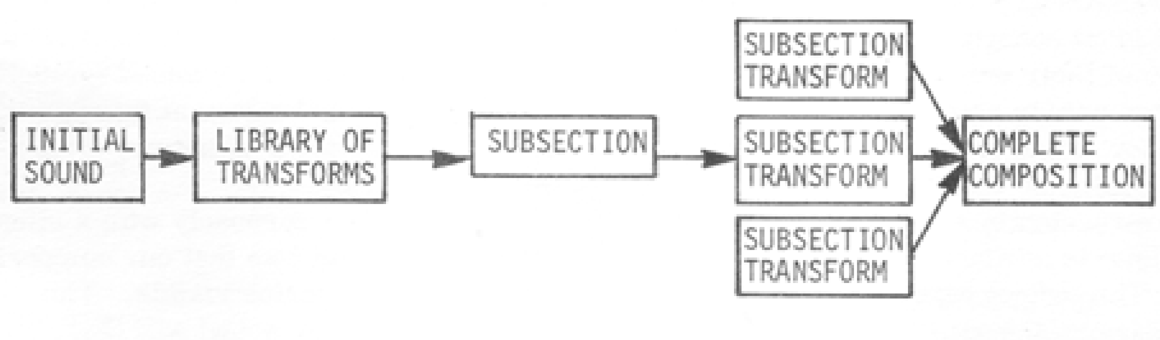


Figure 1 OUTLINE OF A COMPOSITIONAL METHOD (A).

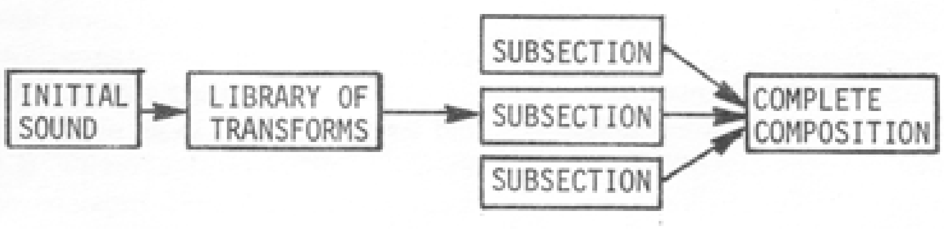


Figure 2 OUTLINE OF A COMPOSITIONAL METHOD (B).

On the other hand, method B provides for a greater number of sounds from the library so that individual subsections may be more remotely associated to one another. The greater possibility of sonorous contrast afforded by method B therefore more readily suggests a composition of longer duration.

Once these basic structural principles are understood it is an easy matter to combine both methods, as well as to formulate new ones. Regardless of the manner in which a piece is conceived and developed it is imperative to remember that compositional techniques, methods etc. are only the means to an end -- the actual composition. The composer's satisfaction with the final result is essential, and it must be remembered that the aforementioned procedures have been suggested only as an aid for the attainment of this goal.

The following discography and bibliography include materials relevant to the topics discussed in this essay. They are not intended to be exhaustive, but they do serve as a point of departure for serious electronic composers.

Selected Discography

Berio, Luciano. "Thema" (Omaggio a Joyce), Turnabout 34177.

Henry, Pierre. "Variations on a Door and a Sigh," Philips 836-898 DSY.

Luening, Otto. "Fantasy In Space," Folkways FX-6160 and Desto 6466.

Reich, Steve. "Come Out," Odyssey 32160160.

Schaeffer, Pierre. "Etude au piano II," Ducreter-Thompson DUC-8.

Stockhausen, Karlheinz. "Gesang der Junglinge," Deutsche Grammophon Gesellschaft DG 138811 SLPM.

Ussachevsky, Vladimir. "Sonic Contours," Desto 6466.

Selected Bibliography

Cope, David. New Music Composition. New York: Schirmer, 1977.

Ernst, David. The Evolution of Electronic Music. New York: Schirmer, 1977.

Howe, Hubert. Electronic Music Synthesis. New York: W. W. Norton, 1975.

Stockhausen, Karlheinz. " Music and Speech." Die Reihe 6 (1960): 40-64.

Ussachevsky, Vladimir and Otto Luening. Electronic Tape Music. New York: Highgate Press (Galaxy Music), 1977.

EXPERIMENTING WITH ANALOG DELAY

by : Marvin Jones

I guess it's not unusual that analog delay lines and time domain processing have found such a comfortable home in today's electronic influenced music. After all, music is an artform which is 100% dependent on one single dimension for its realization - - time. Musical works are performed in time. Even the tones from instruments are smaller recurring time periods, and musical harmony or intervals are ratios of time periods (frequencies). Looking at the situation from this standpoint, one begins to wonder how we lived without delay lines for so long. Up to now the delay lines have been used primarily to process audio rather than generate audio, so we haven't even begun to realize the potential of these devices. Perhaps one reason for the heretofore obscurity of these devices in the home-brew artist's repertoire is the relative high cost of the ICs themselves, probably due to (or at least compounded by) the lack of availability on the hobbyist or consumer components market. PAIA's recently announced release of the EK-5 Delay Line experimenter's kit should help provide the basic tools and inspiration required to get these nifty devices into the hands of all you mad wire freaks, so let's discuss a bit about how the Reticon SAD-1024 works and some basic applications to get you started.

ABOUT THE SAD-1024

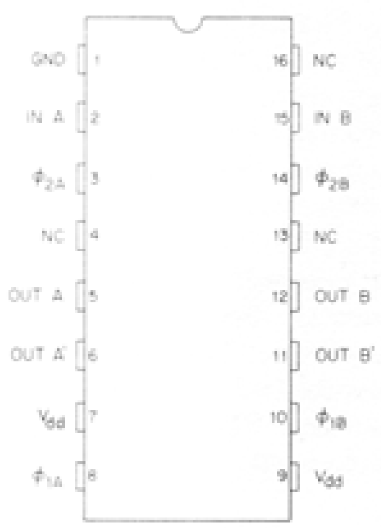
The SAD-1024 represents a third generation of bucket brigade analog delay line technology. As a result of the design and the use of N-channel silicon gate fabrication technology, the SAD-1024 provides operational specs and simplicity far surpassing most of the comparable delay lines. As a direct result of the use of N-channel technology, these delay lines feature: single supply operation, higher speed, better frequency response, better linearity, greater dynamic range, lower harmonic distortion, etc. With proper termination, the SAD-1024 can achieve unity gain, eliminating the need for gain restoration stages (and thus, added noise) after each delay line. Power requirements are greatly simplified since the SAD-1024 requires only a single supply. Many of the P-channel delay lines require three or four supply and bias lines. While specified to operate at 15 volts, the SAD-1024 can operate on supply voltages as low as 3.5 volts, allowing easy application in battery powered equipment, or direct control from 5 volt logic supplies and clock sources.

The SAD-1024 contains two independent 512 stage analog delay lines. The pinouts of the IC are shown in figure 1. Figure 2 shows an equivalent circuit schematic for one of the two delay lines in the IC. Figure 3 shows the schematic of one half of the PAIA EK-5 circuit configuration. Armed with these visual aids, let's see how the chip works.

Most of you, as synthesists, will be familiar with the concept of "sample and hold". This is used quite extensively in synthesizer keyboards, and basically operates as follows. An applied input voltage is always moving, or can be varied. At some point in time, the sample-hold circuit can be triggered

Pin configuration. Note: Unused outputs should be connected to V_{dd}; all other unused pins should be connected to GND, Pin 1, including those marked N.C.

Figure 1



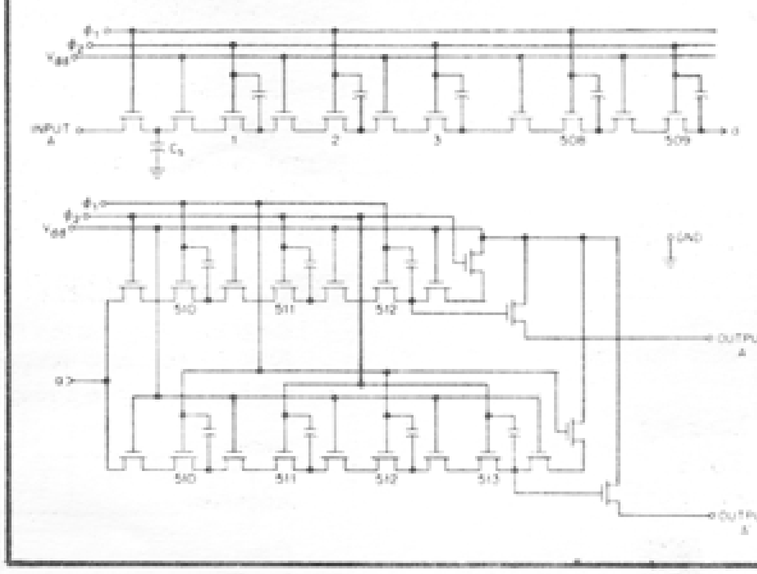


Figure 2

Equivalent circuit diagram of either 512-stage section of the SAD-1024.

to sense what the voltage is, and this voltage is stored on a capacitor until the next trigger pulse causes the stored voltage to change to a new level. For the most part, this is how an analog delay line works also. The primary difference is that there are 512 sample and holds all in a row, and on each "trigger" the stored voltage is passed from one holding capacitor to the next until the voltage once again appears at the output. The analogy most often used to explain the analog delay lines is a bucket brigade. Most people have heard of or seen firefighters using a bucket brigade to pass water from person to person to get it to the fire. Well, think of the line of firemen as the analog delay line, and think of the buckets of water as the charges stored on the capacitors in the delay line. This is where these devices picked up the nickname of Bucket Brigade Devices (BBD).

On a more technical level, let's look at the internal schematic as shown in figure 2. For the sake of argument let's say we are feeding the clocking inputs with a 100 kHz. square wave. The two clock lines, ø1 and ø2 require complementary signals (square waves 180° out of phase) with minimal ringing or undershoot. That is why you will so often see a CD4013 flip flop driving a delay line. The CMOS flip flop is fast, creating minimal undershoot, and automatically generates complementary outputs. To initially get a signal into the delay line, ø1 must be at a high logic level, near +v. This turns on the first input transistor (N-channel MOS) to connect the A input to the first storage capacitor, Cs. Concurrently, the LOW logic level on the ø2 clock line keeps the transistor switch of the first stage (immediately above the "1" designation in figure 2) shut off. At the next clock transition, the instantaneous voltage at the input terminal is "frozen" on Cs as ø1 drops low (near ground) and shuts off the input transistor switch. However, note that ø2 is now high, thus enabling the #1 transistor and allowing a path for the charge on Cs to discharge into the first capacitor, in stage 1. During this half period of clocking, there is readjustment of charge from Cs to cell 1. During the next clock half cycle, ø1 is again enabled to take another sample from the input, and this time also allows the charge in cell 1 to pass to cell 2. In this way, each sample is passed along until it appears at output A with a delay of 512 clock half periods. With the 100 kHz. clock frequency we are using as an example, the output will appear at output A 2.56 milliseconds after it was input. Note that only one sample is taken at the input per clock cycle, and the output appears at "A" only while ø1 is high. When the ø1 falls, the "A" output will fall to its normal low output. An identical output signal will now appear at output A'. At the 510th cell, the signal is split and sent to two outputs, one line having an extra cell to provide an extra half clock cycle of delay. Thus, through summation of the two outputs, a continuous stair-step approximation of the original input is obtained. Without the second output the output would have a great deal of high frequency clock content, as the analog signal information would be presented on only the top pedestals of every other clock half cycle as shown in figure 4. As seen from the timing graph of figure 4, the summed outputs produce a more constant audio signal with only minor, short duration clock switching glitches which are easily removed even with crude passive filtering.

An important concept involved in the use of analog delay lines is sampling theory. Specifically, how many samples are required to accurately reproduce an audio signal? Of course, the more samples taken of a waveform, the more refined the "stair-step" approximation at the output of the delay line will be. However, in an attempt to get longer

NOTE: Pin numbers in parenthesis indicate pinouts for second half of ICs. Component designators identical for each half of circuitry.

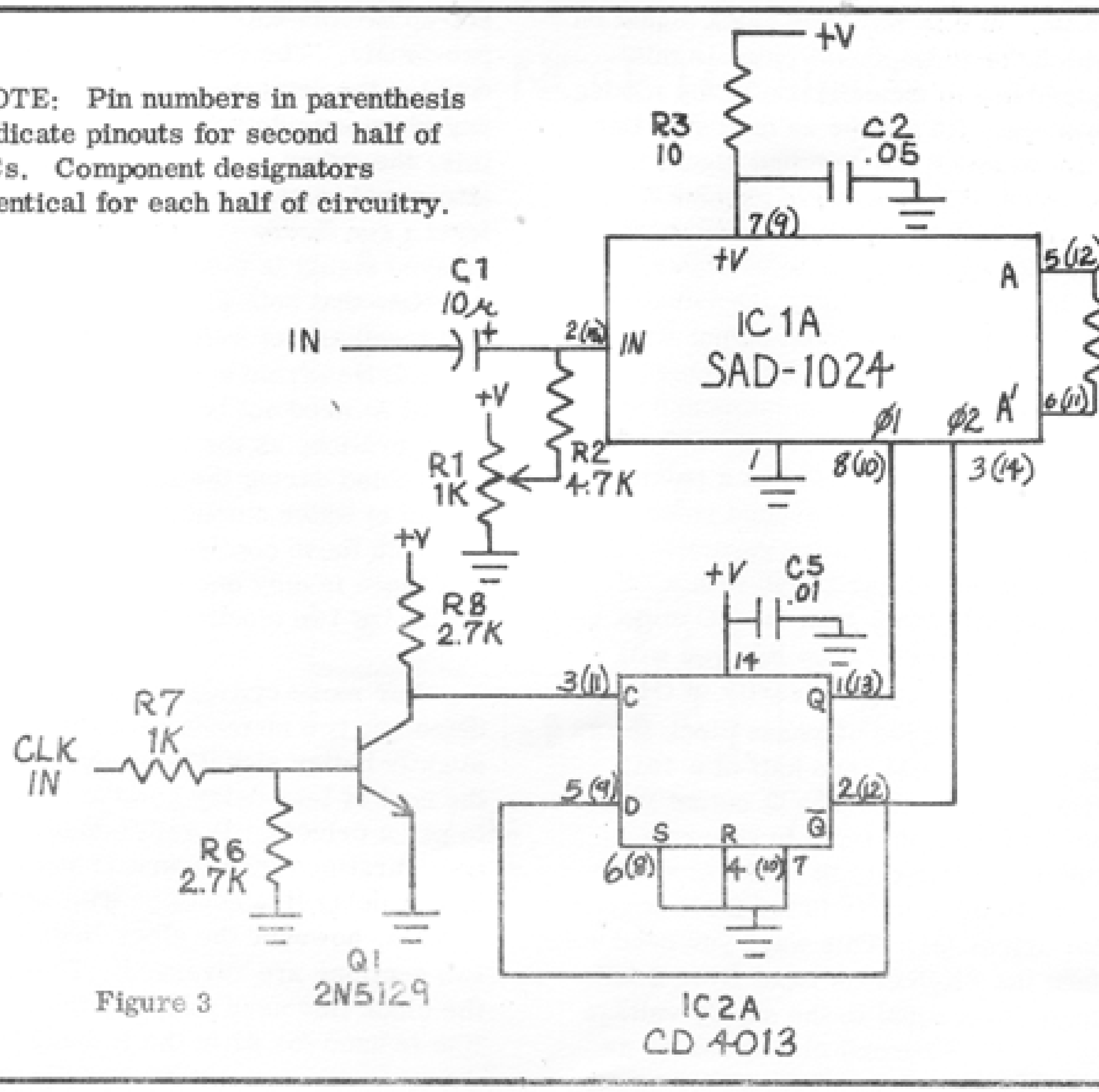


Figure 3

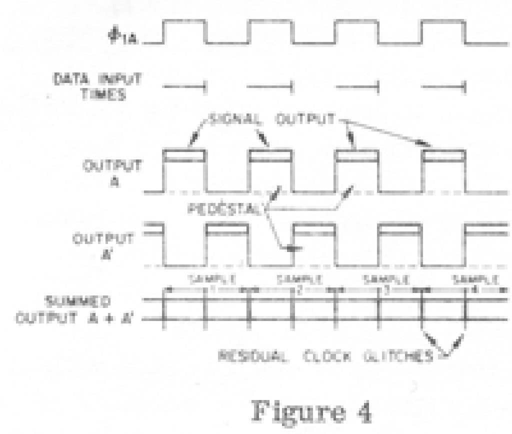


Figure 4

delay times out of the analog delay line, the designer runs head-on into sampling problems. Good reconstruction of waveform occurs with 10 samples per cycle or more, but as the clocking frequency falls towards 2 or 3 times the highest frequency to be sampled, the delay system approaches what is called "critical sampling". Theoretically, a sine wave of specified frequency can be reproduced with only two samples per cycle. This, of course, is a theoretical limit, so we should try to maintain a clock frequency which is at least 3 or 4 times the highest frequency to be processed. In an audio system requiring bandpass to 15 kHz., this would then imply a minimum clock frequency at Ø1 and Ø2 of around 50 kHz. Recalling that most delay systems will use a flip-flop to provide bi-phase clock signals, and remembering that this flip-flop circuit configuration generates a frequency division of 2, this means that the system clock should have a minimum of around 100 kHz. In a mathematical sense, each component of the input is convolved with the Fourier transform of the sampling frequency, fc, to give a sin x/x amplitude response modifier at the output. This, of course, assumes an ideal output smoothing filter. The first "zero" of the sin x/x amplitude modifier occurs at the clock frequency. Thus, at the proposed maximum useful input frequency of fc/2, the output amplitude would be down to .637, or a 4 dB loss. Any realistic viewpoint should also consider the less than ideal roll-off characteristics of output filters, and the possibility of charge dispersion inside the delay line itself.

In a bucket brigade device, each shift of the charge packets leaves behind residual charges, picks up charges from switching signals induced into the audio path, and (at higher frequencies) can radiate energy from the delay line path. All these problems cause inefficiencies which are accumulated over the length of the delay. These inefficiencies account for most of the "Noise" or distortion typically associated with analog delay lines. However, the Reticon SAD-1024's utilization of N-channel technology provides better noise and dynamic range specs than their P-channel counterparts. The typical 70 dB dynamic and noise ratio of the SAD-1024 is perfectly adequate for most experimental applications, and noise gates or companding noise reduction is easily implemented with the delay lines in finished versions of circuits requiring higher specifications.

ABOUT THE EK-5

The small amount of circuitry involved in the EK-5 is the minimal configuration required to get the chip into operation. Naturally, more complicated or refined applications will require additional circuitry. Positive power is applied to pins 7 and 9 through the decoupling network consisting of R3 and C2. This helps stabilize the supply and keep noise from getting in or out. Trimmer R1 and current limiter R2 are used to apply bias at the input, pins 2 and 15. The voltage at the input pins must be set to 40% of the supply voltage in order for the IC to pass a signal. An input signal of up to 500 mV peak-to-peak can be applied to the input pin via coupling capacitor C1. Note that the polarity of this capacitor assumes a ground referenced input signal. If you are using circuitry which places the audio signal on a DC level higher than the SAD-1024 bias voltage, you will need to reverse the polarity of C1. The primary output, providing 512 clock half cycles of delay, and the secondary output, providing an extra half period of delay, are selected from pins 5 and 3 (12 and 14) respectively. R4 provides capabilities for varying the gain of each output in an inverse ratio. This allows matching the two output stages so the DC levels of each output are the same. In this way, the clock signal on which the audio signal rides is minimized due to cancellation in the mixing process. R5 serves as the delay line load on which the summed signal is dropped. C3 is a crude passive filter to help further eliminate some of the clocking glitches and noise above 15 kHz. or so. Despite the inclusion of this filter, additional output filtering with a sharper cutoff should also be used to reduce clock noise modulation of the audio signal. C4 serves as the output capacitor which has a polarity to drive an assumed ground referenced load. As with the input capacitor, if you will be driving a load with a DC component higher than the DC component of the signal on R5, you will need to reverse the polarity of C4.

To drive the bi-phase clock inputs of the SAD-1024, one half of a 4013 flip-flop is used. The $\overline{Q}$ output is tied back to the Data input to generate a divide-by-two circuit. The clocking input to the divider is fed from transistor driver Q1. This way, you need not feed the EK-5 clock input from a full logic level equal to the supply voltage of EK-5. External clocks can be as simple as a low level (down to about 1 volt peak) differentiated spike from a simple UJT oscillator, or a full logic level clock generated by optional on-board components. The rest of the circuitry on the EK-5 board is an exact duplicate of the first half, except using the other halves of the ICs. Also provided in the EK-5 layout are two sets of uncommitted component mounting holes, each of which can accommodate a 16 pin IC, or smaller 14 pin IC, or even two 8 pin ICs plus locations for required resistors and capacitors.

The edge of the EK-5 board is configured to fit into a 22-point edge connector with .156 inch spacing between connections. This style connector is readily available as a 44-point double-sided edge connector from your local Radio Shack stores or many mail order parts houses. Alternatively, you can use the copper fingers as solder points for hardwired applications. Four corner holes allow mounting the circuit board with #4 size hardware.

APPLICATIONS

There are several ways that the two independent delay sections can be used in conjunction with each other. The normal single section configuration involves using only half of the SAD-1024 as shown in figure 5. This is the basic set-up used in EK-5 and described previously. The serial configuration doubles the permissible delay time for any given sample rate. To accomplish this, the output from A is slightly attenuated to restore the original input level (see figure 6). This initially delayed signal is then applied to section B. Note that both Ø1 clock lines are tied together, as well as the Ø2 lines being driven from a common source. Output A' need not be used in this configuration, as the input to section B is disabled during the same time period in which output A' is inactive. For both these configurations, note that there is only one sample per clock cycle, but two clock "glitches" in the output.

For more critical applications, there are two methods which provide slightly better signal reproduction, at the cost of less delay (you always have to pay a price!). Parallel-Multiplex configurations (see figure 7) supply two analog delay line sections with identical signals, however the clock lines to the two sections are reversed. That is, the clock line used for Ø1 in the A delay line is used for Ø2 in the B delay line, and so on. In this way, samples are alternately taken by the two delay line sections thus providing twice as many samples for a given clock frequency. This circuit uses one output from each delay line to be summed as usual for reconstruction of the input signal and nulling of the clock glitches. This circuit configuration was used in the PAIA Phlanger. Note that in any of these configurations where one or more of the output pins are not used, the unused pins should be tied to the positive supply to disable the internal source follower FETs.

In the Differential configuration, it is easier to obtain cancellation of clocking glitches and even-harmonic distortion. As shown in figure 8, each half of a differential signal is applied to each half of a delay line. At the outputs, the normal (A and B) outputs are summed with the extended (A' and B') outputs as usual. This delayed differential signal is then applied to a differential amp where the "common" clock glitches are cancelled.

Any of the above methods can be combined for multiple chip operation. For example, you may wish to use the differential mode for low clock noise feedthrough, plus the techniques of serial configuration for longer time delay. Possibilities are numerous, and each has benefits for particular applications.

With all the above information out of the way, you should have a fairly good idea how the SAD-1024 works, and how to use it in its own audio delay subcircuitry. Now you are ready to fit the EK-5 delay circuitry into whatever application you have in mind. Some effects are obvious. For example, an echo unit would be configured by using a chain of delay lines (to get a long enough delay to be able to actually hear the "echoed" signal) and a mixing amp to provide an output signal containing the original input and the delayed signal. For fancier versions, you could provide front panel control of the clock speed to vary the time of delay. And, how about a control to tap off part of the output of the delay line and feed it back to the input for repeating echos. All these features would be desirable in an echo unit. All are straightforward and easily implemented, so we won't go into any detail here. A few more advanced applications, and guidelines for their implementation, are presented below.

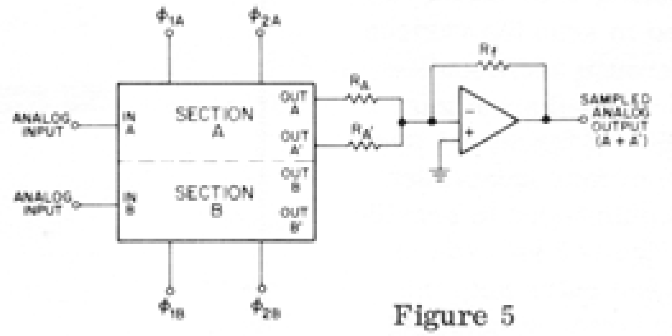


Figure 5

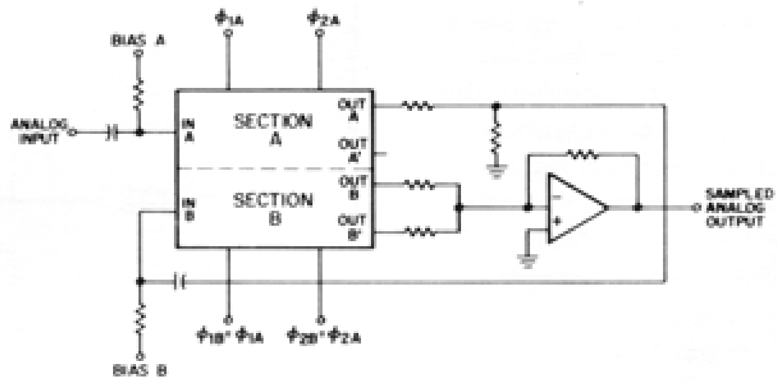


Figure 6

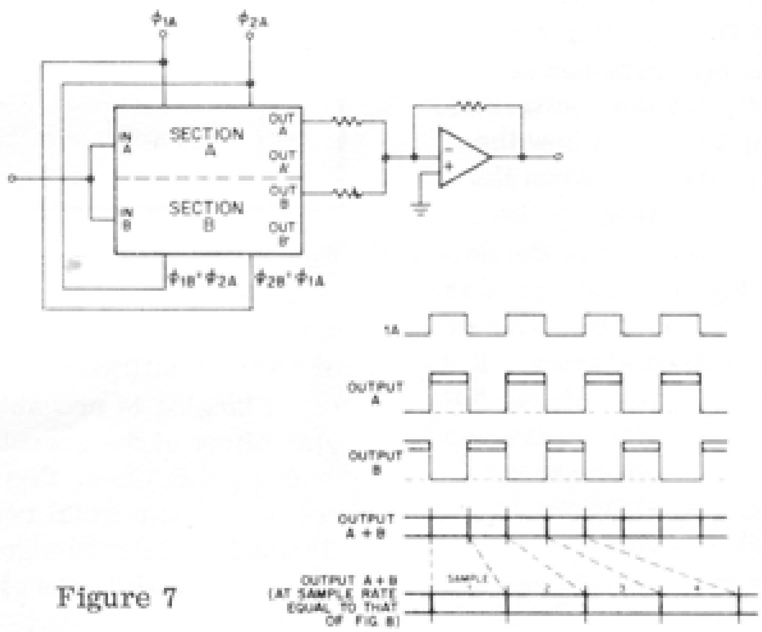


Figure 7

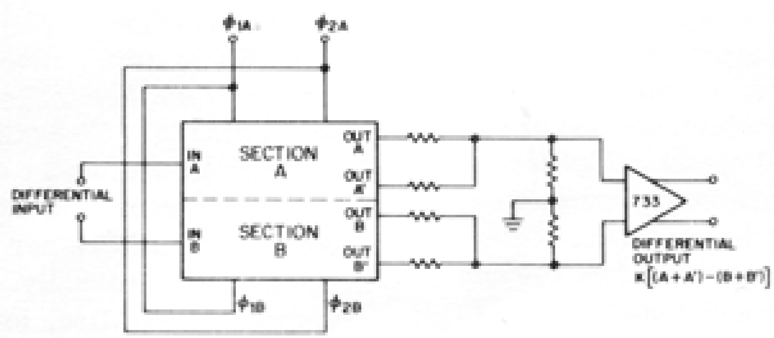


Figure 8

Reverberation, although very similar and often times confused with echo, is a complicated and delicate effect to synthesize. Multiple delay paths, each with different delay times, provides the basis for reverb, as it simulates sound waves bouncing off various walls and structures in an enclosed area. To provide the effect of multiple reflections and persistence of the reflected sound, a feedback path should be provided to send the various delayed signals through additional (different) time delay paths. Figure 9 shows a block diagram for such a system. A minimum of four independent delay paths is recommended to provide the randomness required for reverb. Of course, additional paths help the system do an even better simulation.

Vibrato can be obtained by slightly modulating the high frequency clocking signal for the delay line. The resulting pitch shift is the derivative of the waveform being used to modulate the clock. Thus, a sine wave used to modulate the clock will provide a cosine variation of pitch, or the most common smooth vibrato effect. Using specialty waveforms can produce special effects. Modulating the clock with a triangle wave can produce a square wave vibrato. Using a ramp will provide a constant shift of the pitch - shifting up if the ramp has a positive slope, down if it has a negative slope. Unfortunately, the solid pitch shift is marred by the flyback period of the modulating ramp wave. Additional circuitry can solve this pitch "glitch", but the configuration is far from simple. Here's how the vibrato works (basically). When the signal has been passed through the delay line, it is clocked out of the delay line at a slightly higher clock speed than when it went into the line, the pitch of the signal will be raised slightly. If it is clocked out slower than normal, the pitch will be lowered. The continuous cyclic variations in the clock speed then translates to the variations in pitch of the signal -- vibrato.

Chorusing is very similar to vibrato in that you wish to make small variations in the pitch of the signal. However, to best simulate the effect of two or more independent signal sources, a random fluctuation of clock speed more closely approximates the random technique differences between human-type musicians. Summing the original with one delayed signal can give the effect of two musicians playing in unison. Adding either another delay path with its own independent random clock signal, or using another delay stage to double the first doubler output (as shown in figure 10a) should be sufficient to confuse the ear into thinking it is hearing not three or four instruments, but a rather large group playing in unison. Initial delay times as short as 1 millisecond can be used for the chorusing stages as shown in

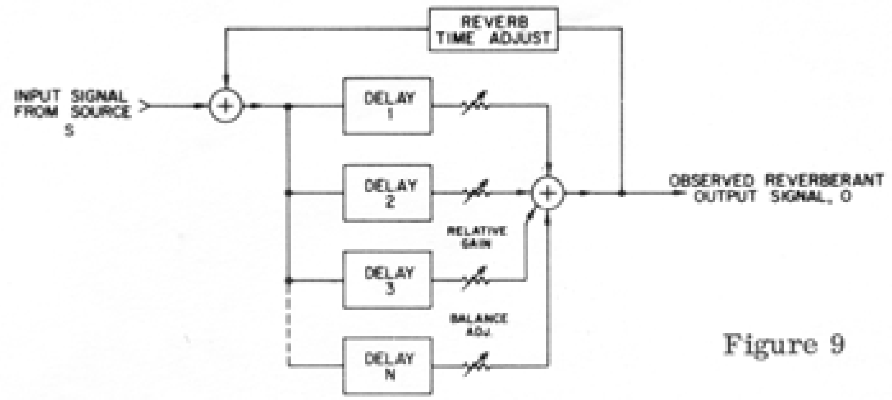


Figure 9

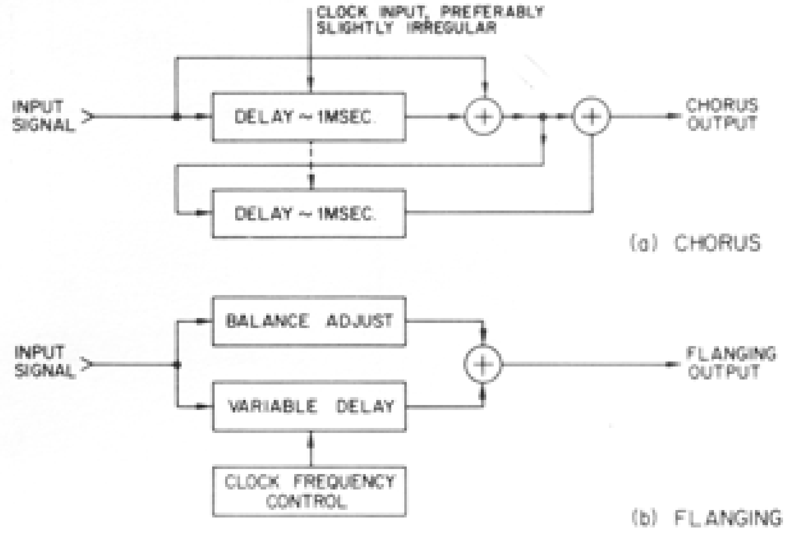


Figure 10

figure 10a. However, the effect seems to be more pronounced and realistic with initial delays in the neighborhood of 10 to 20 milliseconds.

Flanging is probably the most popular effect of the seventies. Without analog delay lines, this effect would not be a commercial reality. For flanging, a delayed signal is mixed with the original while the clock signal for the delay line is slowly varied across a wide range. Delay time span for a typical flanger is from about .5 millisecond to about 10 milliseconds. Figure 10b shows a block representation of such a system. Also, see the flanging article in the April/May issue of POLYPHONY. To easily accomplish flanging (or vibrato or chorusing) you will probably want to use a voltage controlled clock rather than varying clock speeds manually. A typical manually controlled, low-cost clock could be generated as shown in figure 11. Alternatively, extra money could be invested in using a 566 type VCO which is a much more versatile clock. It is also much smaller than a "gate-type" clock, so TWO of these 566s could be put on the EK-5 board where you could fit ONE of the 4001 or 4011 types. The basic connections to the 566 are shown in figure 12a. NOTE that a bipolar supply is required, but the square wave clock output switches between +V and ground which is perfect for driving our 4013 clock shaper. For using the 566 as a high frequency clock, the timing capacitor (at pin 7) should be small. The values between 15 pf and 100 pf generally work well, depending on application you are using the delay line in. Pin 5 is a frequency modulation input, but only operates over a narrow voltage range and only delivers a frequency deviation (predictably) of 2:1, or up and down one octave. For a wide range clock sweep it is best to tie pin 5 to a fixed reference and play with the timing resistor input at pin 6. A fixed resistor or pot can be used as shown in figure 12b and c. Best yet, a voltage controlled current source similar to figure 12c can provide clock sweeps over several decades for flangers and wide range echo systems. For additional information on using the 566 as a voltage controlled oscillator, see Lab Notes in the November, 1977 issue of POLYPHONY.

As mentioned earlier, don't forget to use output filters to roll off the

4001 OR 4011 GATES

CAN BE USED.

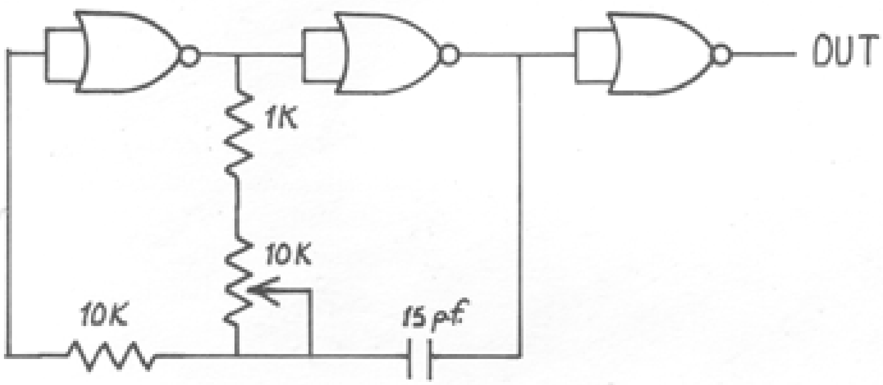


Figure 11

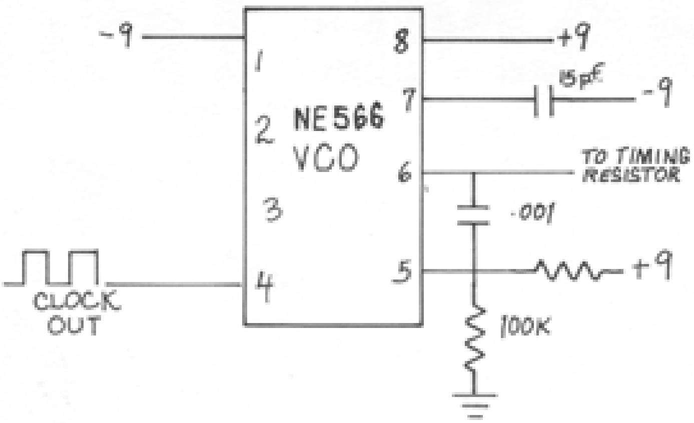


Figure 12 (a)

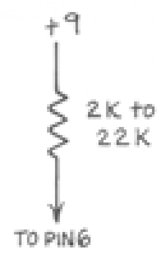


Figure 12 (b)

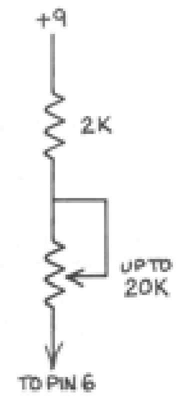


Figure 12 (c)

response of the delay line at a point at least 1/2 or 1/3 of the clocking frequency. This is important for minimizing clock frequency feedthrough and intermodulation of clock frequencies with audio signals. A typical output stage utilizing a two-pole active filter with a cutoff frequency of 20 kHz. is shown in figure 13. Your filter need not be this complex for initial experimentation, but don't discount its importance if you experience noisy output.

I wish to thank the people at RETICON for allowing us to use many of the drawings and ideas from the manuals and data sheets they have published. Their well-planned publications have been instrumental in letting many a designer get down to the business of applications rather than first spending a lot of time just learning how the chip works.

Hopefully, this will get you interested in playing with one of the neatest ICs to come around for some while. The EK-5 experimenter's kit is available from PAIA Electronics, 1020 W. Wilshire Blvd., Oklahoma City, OK 73116 for $24.95 postpaid.

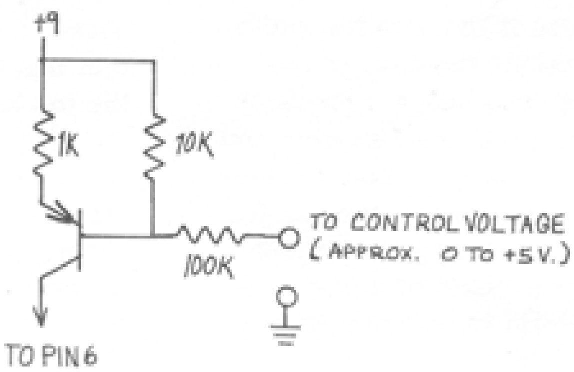


Figure 12 (d)

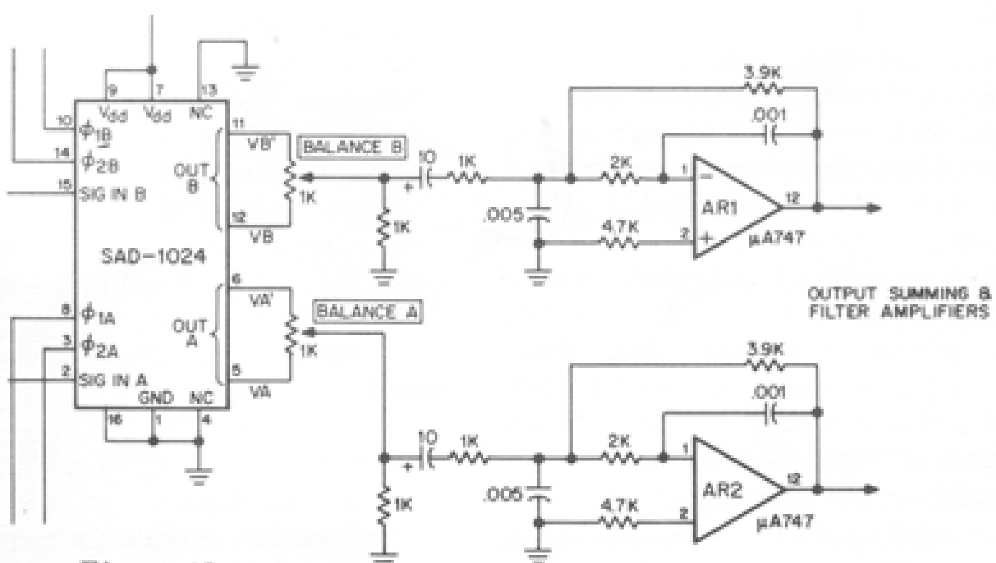
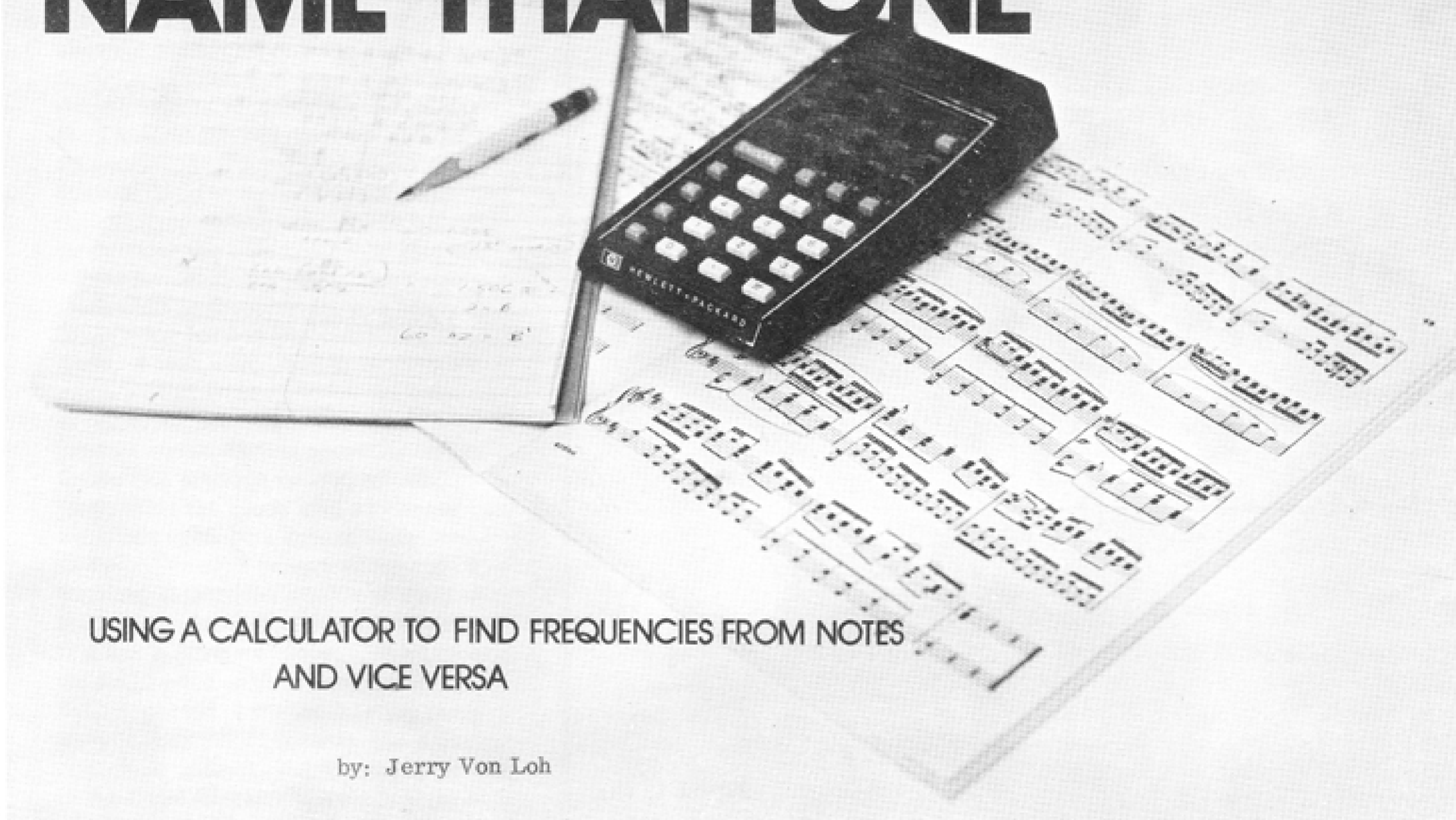


Figure 13

NAME THAT TONE



USING A CALCULATOR TO FIND FREQUENCIES FROM NOTES AND VICE VERSA

by: Jerry Von Loh

Have you ever heard a note and wondered about where it fits into the audio spectrum? Consider the case of the sound technician who hears a feedback tone in a professional sound system and wonders what band on his equalizer will filter it out. Or the case of the music theory student who wants to know just what the frequency range of a piano, or flute, or bass violin is and only knows the tonal range. In this article, I've worked out several formulae that solve these problems, while at the same time, pointing out some of the mathematics of audio in general. These come in handy in the synthesis field, where tonal ranges must be converted to frequency spans. Quickly, what would be the range needed on an oscillator to cover a span from C^2 to A^6? Well, the sequence below will show how it is done.

First we need a way to convert the alphabetical part of the note to a number that can be used in our formulae. The equivalency line shown in figure 1 works out fine and it is this system that we will use since the most common tones are in the equally tempered scale.

The octave numbers will remain the same, assuming A^4=440 Hz. For reference sake this means that on a typical 88 key piano, the lowest key is A^0 with each key above it also having the same octave number until the next A comes up, that one being A^1, etc. all the way up to the top key that turns out to be C^7.

The first formula is the note to frequency one, and it is fairly basic to anyone who has studied music theory. First, some constants must be established. We will be using 27.5 as a base frequency since it is equivalent to A^0 or 0^0. This was derived from A-440 in the fourth octave being divided by two over and over four times bringing it down to the 0th octave(any frequency divided by two is equivalent to the same note one octave lower.) The next constant is the infamous $\sqrt[12]{2}$ here being equal to 1.05946309435. This is the basis of the equally tempered scale, and its derivation is found in the PAIA synthesizers user's manual. Any frequency multiplied by this constant is equal to the next note in the scale.

Now, knowing these facts, I've come to this equation:

$$F=27.5 \times 2^{(O)} \times (\sqrt[12]{2})^{(N)}$$

which works out with all frequency standards. (F freq., O octave, N note)

Now for a few examples:

What is the frequency of B^{b^6} ?

Using the equivalency line, $B^b(A^\#)$=1 so:

$$F=27.5 \times 2^6 \times (\sqrt[12]{2})^1$$

Frequency is 1864.655 to three places.

What is the frequency of Middle C (C^3)? C=3 so:

$$F=27.5 \times 2^3 (\sqrt[12]{2})^3$$ Freq. is 261.626

This is a nice formula, but not half as nice as when it is used in reverse. In this form you can put in any number, a readout from a frequency counter, the year you were born, or a preset clock frequency and find the nearest note. It's a bit more complicated, and has two steps to it, but is quite useful, especially to electronic designers.

Here it is: first find the octave:

$$O = \frac{\text{Log}\left(\frac{F}{R}\right)}{\text{Log}(2)}$$

Where F = Selected frequency
and R = Ref. Freq. = 27.5 Hz.

Note: Truncate octave to Integer value. Do not round!

And then the note:

$$N = \frac{\mathrm{Log}\left(\frac{F}{R \cdot 2^{(O)}}\right)}{\left(\sqrt[12]{2}\right)}$$ Rounded to the nearest integer.

What is the note value for line frequency? (60 Hz.)

$$O = \frac{\mathrm{Log}\,\frac{60}{27.5}}{\mathrm{Log}\quad 2} = 1.125 \text{ or } \frac{1}{} \text{ to the first integer}$$

$$N = \frac{\mathrm{Log}\left(\frac{60}{27.5 \times 2^{1}}\right)}{\mathrm{Log}\ \sqrt[12]{2}} = 1.506 \text{ or } 2$$ when rounded. and 2 = B on line in figure 1

from first equation

so 60 Hz. $= B^{1}$

What would be the note associated with a 5 kHz. oscillator tone?

$$Oct = \frac{\mathrm{Log}\left(\frac{5000}{27.5}\right)}{\mathrm{Log}\ (\ 2\ )} = 7.506,$$ or 7 when cut to the integer

$$N = \frac{\mathrm{Log}\left(\frac{5000}{27.5 \times 2^{7}}\right)}{\mathrm{Log}\left(\sqrt[12]{2}\right)} = 6.07 \text{ or } 6 = D^{\#}$$ (from fig. 1)

from previous formula

There are a few odd ball cases where this formula doesn't seem to work, and these are when any frequency is below the reference of 27.5, because the answers are negative. But they do work, it's just a matter of reversing the note/number equivalency line, as below. Whenever an answer turns out negative, as before the octave is chopped to the nearest integer, with any value less than -1 being equal to 0. The note value is still rounded, though, and this value is used in the new tone line.

Example: What is the note of a 20 Hz. tone? (the theoretical limit of hearing)

$$O = \frac{\mathrm{Log}\left(\frac{20}{27.5}\right)}{\mathrm{Log}\quad (2)} = -.45 \text{ or } 0$$ by dropping the decimals.

$$N = \frac{\mathrm{Log}\left(\frac{20}{27.5 \times 2^{\cancel{0}}}\right)}{\mathrm{Log}\ \sqrt[12]{2}} = -5.513$$ rounded to -6

In fig. 2 below, 6 = $D^{\#}$ so 20 Hz. $= D^{\#\,-1}$*

* Note: When counting backwards, this automatically lowers octave by one.

To check this, the frequency of 40 Hz. should be one octave higher, or

$$O = \frac{\mathrm{Log}\left(\frac{40}{27.5}\right)}{\mathrm{Log}\quad (2)} = .5405 \text{ or } 0$$ to integer

$$N = \frac{\mathrm{Log}\left(\frac{40}{27.5 \times 2^{0}}\right)}{\mathrm{Log}\ \sqrt[12]{2}} = 6.48 \text{ or } 6$$ when rounded.

Using figure 1, 6 = $D^{\#}$, so 40 Hz. $= D^{\#\,0}$

It checks!

I hope these formulae will come in handy for you someday, either in the synthesis field, electronic design, or just for a little math fun. Now if someone says, "Name that tone!" you'll know how to go about doing it.

NOTE EQUIVALENCY CHARTS

A	A#	B	C	C#	D	D#	E	F	F#	G	G#
0	1	2	3	4	5	6	7	8	9	10	11

Figure 1

A#	B	C	C#	D	D#	E	F	F#	G	G#	A
-11	-10	-9	-8	-7	-6	-5	-4	-3	-2	-1	0

Figure 2

LAB NOTES: PINK TUNES

By: John S. Simonton, Jr.

As we begin this month's journey into the bizarre I should warn you that I'm operating in a somewhat altered state of consciousness.

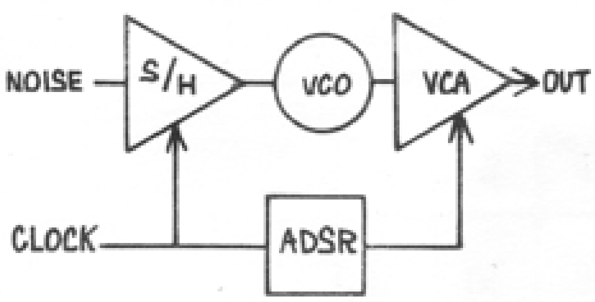
Oh, not from chemicals or nature's own, none of that. There's just some very nice color graphics going on the Apple II and the background music is a slightly oriental feeling 4 part harmony being composed by a P4700/J. It's really a most unique environment.

Wait. Composed by the synthesizer? Surely not, surely just something pre-recorded and played back.

Well, I suppose that I wouldn't attack someone who asserted that I composed the piece. I'd be flattered, but it wouldn't be entirely correct. I knew before the tune started what sort of texture (for lack of a better term) it would have. But I have no idea what exactly is coming next.

And that, in case you hadn't already guessed, is what we want to talk about this time. Computer programs that compose music.

Let's start at a very elementary level. Probably you've seen or connected synthesizer patches that look something like this:



It's a relatively common configuration in which, at regular intervals, the instantaneous value of the noise source is captured by the S/H and the resulting voltage used to set the pitch of the VCO. The ADSR and VCA give us some knobs to twiddle and control dynamics, but otherwise are just window dressing.

If you've done one of these, you know that the results are interesting, but certainly not a musical composition in the traditional sense. As a composing device, it's hard to know what the biggest fault is here, but certainly it must be the fact that there are no guarantees that the series of pitches produced are going to be equally tempered intervals (or any known tempering for that matter). In fact, you can almost guarantee that they won't be; the control voltages applied to the oscillator are completely random.

And therein lies the tale.

I don't believe that anyone is able at this point in human development to concisely explain what makes music "musical", but most folks that have thought about it seem to feel that "good" music (ugh, all the subjective terms) combines both order and disorder. Establish a pattern in the listener's mind . . . then surprise him; pleasantly, preferably.

Like the "noise music" example above, any compositional program that we come up with today is in some way going to rely on a STOCHASTIC (big word for random) process. If it didn't, we wouldn't be writing the program that wrote the music; we'd be writing the music.

Our task then, is to bring order from disorder (in a very real sense, nothing less than reversing entropy) - but not completely. It isn't easy, but in an elementary form not as difficult as it may sound either because we now have at our disposal that wonder of wonders (which many right-thinking people say is Maxwell's Demon personified):

THE COMPUTER

By simply programming the computer to randomly select only pitches that are part of the equally tempered sequence, we've made a start, but in all honesty not much of one; still there is too much disorder. Low pitches followed as likely as not by very high ones, no identifiable key signature. It's still "noise music".

The quickest way to begin bringing the kind of order that we're looking for is to write a program that uses a random number not as the note, but as a "pointer" which is used to select one of a number of acceptable "candidate" notes from a previously entered table. We're using our intellect to select ahead of time only those notes which we know will harmonize with the rest of the notes which the computer is allowed to select. I've written a few of these kinds of programs. They're a little better than purely random notes, but not much. Still too much disorder.

There are a lot of tricks to bring rigorous order, like making random substitutions of candidate notes into previously entered melody lines. This kind of thing produces terriffic results, but it's not the computer doing most of the composition - you are.

Now comes the April issue of Scientific American and there, in Martin Gardner's consistently enlightening Mathematical Games column, is a piece on computer music. Well, not just computer music - as is usual, Mr. Gardner's mind ranges far and the column covers visual art and computer generated "landscapes" and fractal curves and the place of pink noise in "the meaning of it all". Very heavy. And buried in amongst it all is an algorithm conceived by Richard Voss (of IBM) for turning "white" random numbers "pink".

Don't let this "white" and "pink" business throw you. You're used to white noise and the pink noise that results when you filter it. We can think of the Voss algorithm as a filter for random numbers.

The realization of the Voss algorithm which is used in PINK TUNES (the program listing at the end of this column) can be likened to rolling a set of 5, four sided dice whose faces bear the numbers 0 - 3. We get the random number that we'll use as the pointer to the list of candidate notes by adding together the numbers on the exposed face of each die (I know, a 4 sided die won't have an upper face, that's not the point).

If we consistently rolled all 5 dice, we would still produce too random a number; even through, as any crapshooter can tell you , the probability is that the total of the faces will be somewhere in the middle of the range of possible numbers - just as a pair of six sided die "like" to come up 7.

The trick is not to roll all 5 dice every time, but rather to come up with a scheme that most frequently rolls one or two and infrequently rolls all 5. Since the random number that is produced is always a total of the 5 dice, this produces a series of numbers that most frequently vary only slightly from one another while still permitting periodic large changes.

Voss's scheme (and ours) is to maintain a 5 bit "pinking counter" (our term) which is incremented each time we get ready to generate a new pink number. The new value of the pinking counter is compared to the old and only those "die" which correspond to bits in the counter which have changed are rolled.

The rest of the program is "overhead". As I mentioned in the beginning, PINK TUNES actually generates a 4 part harmony (provided that we supply it with harmonizing notes in the candidate list) and the program must keep track of how long each of the notes in the 4 parts is to play and allow for the updating of the candidate list and recognize a limited number of commands from the computer's keyboard.

The fully documented listing is the best place to go to see how it all works (it's in your best interests to understand it as fully as possible) and specific details and asides are covered in the boxes.

After entering the program and its data base (note part of the program is on page zero, part is on page one and the data base and working registers are on page zero), first save a copy on tape. If something goes crazy, you don't want to have to enter it all again.

Set up the synthesizer and start running the program starting at the hard start location of $0003. The data that you loaded is for the pentatonic scale composition that I mentioned in the opening paragraph and you should immediately hear the synthesizer producing the composition. It should go without saying that you will undoubtedly have to call the tuning function (control key #1) and tune the oscillators before it makes music.

You have the ability to change the candidate note list while the program is running simply by pressing keys on the keyboard, but bear in mind that the candidate list is 16 notes deep. As you enter a new note, the one that was entered "16 notes ago" disappears from the list. If any of the 16 notes are inharmonious, the program will periodicly produce discordant sequences.

With PINK TUNES running, three of the computer's control keys have meaning:

Key 0 "scrambles" the random number generator to produce a new tune. This is really only useful if you are in the cyclic mode (see box).

Holding key 1 provides a tuning function by causing all 4 outputs to produce a triggered middle C.

Touching key #2 initiates a muted shut-down of the synthesizer and branch back to the monitor, allowing changes in the memory locations described in the boxes.

After making changes using the monitor, always start the program running again from the soft start location $000B.

The program runs very nicely, but is experimental and not intended as a finished product. Skillful polishing should reduce its length by at least 15 - 20% and it would be nice to make changes in timing, etc. "on the fly" without having to shut down the synthesizer.

At the same time that the program is primarily "just for fun", don't dismiss it as trivial. It definitely produces 4 part harmonies and even those that are not directly useable in a composition can serve as inspirational lubrication to the gears of creativity. If you're involved in producing commercial jingles, this is a terriffic tool.

As you play with the program you will begin to get a feel for how various probabilities affect the composition and you're sure to learn some things about composition that you never knew before.

Finally, a very special thanks to Bob Yannes who sent me a listing of a similar program (PINK FREUD) which generates 4 part canons on a P4700/J. I haven't reviewed this program thoroughly yet, but knowing Bob it's sure to be neat. I'm sure that he wouldn't mind my sharing copies of the listing with anyone who sends a SASE.

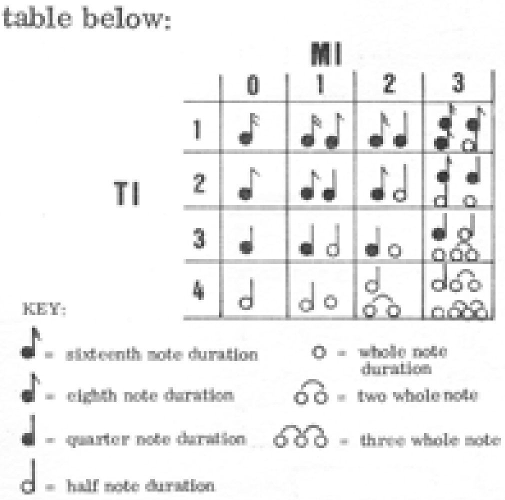
'Til next time, my best to all.

NOTE DURATIONS

Each of the 4 output channels has associated with it its own duration timer and two variables in the computer's memory which determine what characteristics the time values of the notes produced by that channel will have. In the interest of convenience, we'll name these two variables MASK and TIME; or, simply M and T.

We need to think of each of these variables as being composed of a high half-byte (hhb) and a low half-byte (lhb). The hex number $F3 (an arbitrary example) has an hhb of $F and an lhb of $3. This is necessary because the half-bytes determine two separate parameters.

The lower half-bytes of MASK and TIME (Ml and Tl respectively) interact to determine what time values are possible from a given channel. A channel can be restricted so that it produces only 1/16 notes or 1/16 and 1/8 notes or a wide variety of other possibilities as summarized in the table below:



Note that this is a partial table intended only to demonstrate the pattern.

Other combinations of Ml and Tl produce other possible time values. Some combinations not listed will produce undesirable results.

The high-half-bytes of MASK and TIME (Mh and Th) interact to determine the probability that the note being produced by that channel will be dotted (its duration extended by half of its actual value).

In actual practice, it is most convenient to set Mh to $F and regulate the probability using only Th. The influence of Th on the probabilities of a dotted note is illustrated below:

Th	Probability of dotted note
$8	one in two
$4	one in four
$2	one in eight
$1	one in sixteen
$0	zero

EXAMPLE: A channel which has MASK and TIME values of $F3 and $11 respectively will be capable of producing 1/16, 1/8, 1/4 and 1/2 notes with a one in sixteen probability of the note being dotted. A channel with M and T of $F0 and $01 will produce nothing but 1/16 notes, none of which will be dotted.

The page zero addresses of the MASK and TIME parameters for the four output channels are given below:

CHANNEL

	A	B	C	D
MASK	$8F	$8E	$8D	$8C
TIME	$8B	$8A	$89	$88

TEMPO

By using the MUS-1 subroutine LOOK to gather data from the AGO keyboard, PINK TUNES follows our standard protocol of using the keyboard encoder clock rate as the system master clock. Analog control of tempo may be provided by varying this clock rate as has been mentioned in previous columns.

PINK also has a variable at zero page location \$A9 which gives gross digital control of tempo. The recommended range of values for this variable are from \$FF (far too fast) to \$F0 (insanely slow).

GLIDE AND TRANSPOSE

PINK uses the MUS-1 QuASH drivers (NOTE) and therefore allows for both independent pitch transpositions of any and/or all 4 channels as well as providing a means of enabling or disabling glides.

Though not strictly true, it is most convenient to think of these variables as being divided into high half-byte and low half-byte with the hhb controlling glide (\$8 turns the glide on, \$0 turns it off) and the lhb determining transposition. For example, a channel which has this transposing variable set to \$8C will have its glide turned on and be playing notes an octave higher than the actual note selected by PINK.

Here are the transposing variable addresses:

	CHANNEL A	B	C	D
TRANSPOSE	\$CF	\$CE	\$CD	\$CB

CYCLE CONTROL

The variable at zero page location \$D3 controls the number of notes which will be played before the cycle repeats. Changing the contents of this location to \$20 (for instance) will cause 4 bars of eighth notes to be played before the tune repeats. \$40 would produce 8 bars of eighth notes.

Setting the contents of the location to \$00 amounts to enabling a "free run" mode in which the patterns do not repeat (in practical terms).

If you want to get really fancy, you can change program location \$188 from its current value of 85 (STA to the zero page) to EA (a NOP) and the result will be that on successive cycles the time values of notes will not change but the actual notes played will, producing a strong rhythmic tie from cycle to cycle. It also doesn't always work, sometimes a repeating loop will be entered anyway. Other times the duration of a tune will be 2 or more times as long as the actual cycle time.

To change a cyclic tune, touch control key 0.

DE-PINKING

To get some feel for the effect that the Voss pink-ing algorithm has on the composition, you may want to change it slightly. There are a couple of easy ways that this can be done. By changing the current instruction at program location \$11C from \$45 (Exclusive-OR on the zero page) to \$EA (a NOP), you slightly de-pink the note selector, making it somewhat more random. You may have to listen a while before you notice the difference, but there is one.

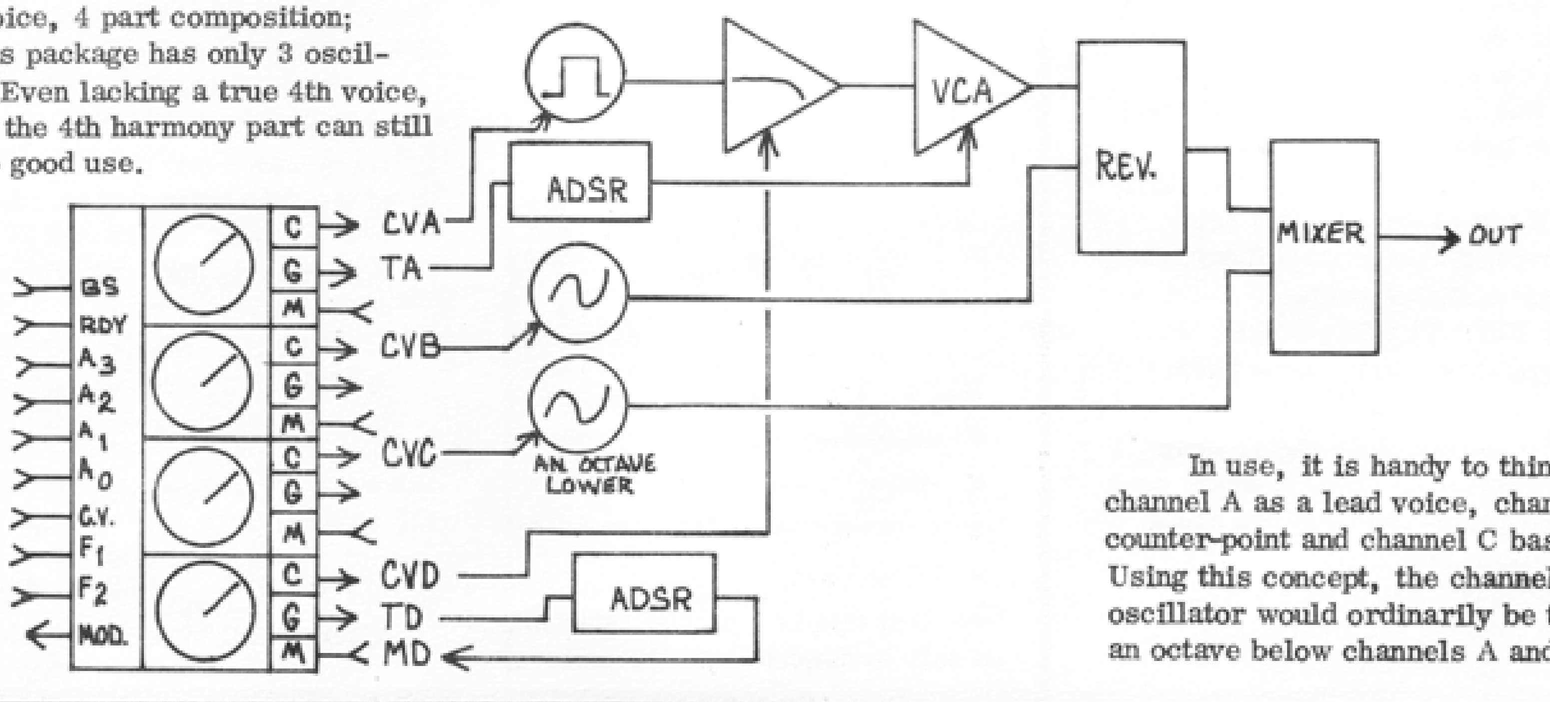
To completely eliminate the Voss algorithm make these substitutions beginning at location \$118; A9 FF EA EA EA EA. This change is equivalent to rolling all 5 of our alleged 4 sided dice each time a new note is selected and will produce changes that even a tone deaf aborigine would recognize.

THE SYNTHESIZER

The module complement of a P4700/J is not large enough to do a true 4 voice, 4 part composition; since this package has only 3 oscillators. Even lacking a true 4th voice, however the 4th harmony part can still be put to good use.

Here is the most universal of the patches used during the development of PINK TUNES:

Note that the 4th harmony part (from channel D of the QuASH) is used to set the center frequency of the VCF.



In use, it is handy to think of channel A as a lead voice, channel B counter-point and channel C bass line. Using this concept, the channel C oscillator would ordinarily be tuned an octave below channels A and B.

THE CANDIDATE NOTES

Selection of the candidate notes that you give PINK TUNES and the order in which they're entered play a big part in the feel of the final composition. As an obvious example, the pseudo-pentatonic scale resulting from entering only accidentals (sharps and flats) tends to produce oriental sounding compositions.

The selection of notes is "pinked" on a compositional (rather than a per-channel) basis, which means that the 4 notes being played at any one time tend to cluster around a relatively short series of entries in the candidate table. The significance of this is that it allows statistical control of changes in key signature. For example, entering the candidate sequence C1, E1, G1, C2, E2, G2, C3, G2, A2, F2, D2, A1, F1, D1, F1, A1 will produce a composition that periodicly changes from the key of C to D minor.

It is important to remember that the candidate table will always contain 16 notes and in order to produce consistant harmonies, all 16 notes must be harmonious. Also remember that notes at the ends of the table (oldest and newest entries) have a lower probability of being played than the notes in the middle.

LOADING THE PROGRAM

NOTE THAT PINK TUNES CONSISTS OF THREE MAJOR SECTIONS: THE MAIN PROGRAM ON PAGE 0 OF MEMORY, SUBROUTINES ON PAGE 1, AND DATA BASE ON PAGE 0.

BEFORE ENTERING ANY PROGRAMMING, MAKE SURE THAT THE MONITOR STACK AND USER'S STACK ARE BOTH SET TO $FF (SO THAT THE STACK DOES NOT OVER-WRITE PROGRAMMING ON PAGE 1) AND THAT THE STATUS REGISTER IS SET TO $00 (TO INSURE THAT THE CPU IS WORKING IN THE HEXADECIMAL MODE) USING THESE ENTRY SEQUENCES:

```
0ED-DISP-FF-ENT (SETS MONITOR STACK)
0FE-DISP-FF-ENT-00-ENT (USER STACK
             AND STATUS REGISTER)
```

ALL OF THE FOLLOWING PROGRAMMING, DATA BASE, AND INITIALIZATION OF MUS-1 NEED BE DONE ONLY ONCE. THEY WILL SUBSEQUENTLY LOAD TO THE COMPUTER'S MEMORY FROM THE MASTER TAPE THAT YOU WILL GENERATE AT THE END OF THE LOADING PROCESS.

INITIALIZE THE MUS-1 VARIABLES CTRL ($0E8) AND ODLY ($0E9):

```
0E8-DISP-00-ENT-20-ENT
```

ENTER THE DATA BASE LISTED BELOW BEGINNING AT LOCATION $088 USING THIS ENTRY SEQUENCE:

```
088-DISP-02-ENT-04-ENT-01-ENT (ETC)

****************************************

              DATA BASE

088:02 04 01 01 F2 F0 F3 F3
    5A 5D 5F 62 64 62 5F 5D
    5A 58 56 53 51 53 56 58
    00 00 00 00 00 00 00 00
    FA

****************************************
```

NEXT LOAD THE MAIN PROGRAM:

```
000-DISP-4C-ENT-C0-ENT-FF-ENT (ETC)
```

AND THE SUBROUTINES:

```
100-DISP-8A-ENT-48-ENT-A5-ENT (ETC)
```

BEFORE TRYING TO RUN THE PROGRAM SAVE IT ON TAPE FROM LOCATION $0 TO $1A0:

```
0-0-0-0-0-1-A-0-0-1-D-D-TAPE
```

BEGIN RUNNING THE PROGRAM FROM THE 'HARD START' LOCATION $3:

```
003-RUN
```

AFTER A SHORT (3 SECONDS OR SO) DELAY, THE PROGRAM WILL BEGIN PRODUCING THE COMPOSITION.

THE 'SOFT START' LOCATION IS $00B

```
                :*********************************
                :*                               *
                :*          PINK TUNES           *
                :*                               *
                :*     A COMPOSING PROGRAM       *
                :*    FOR FOUR PART HARMONIES    *
                :*                               *
                :*     BY JOHN S SIMONTON, JR    *
                :*(C) 1978 PAIA ELECTRONICS, INC *
                :*********************************
                :
                :
                :FIRST ATTEND TO HOUSKEEPING--
                :
000-   4C C0 FF  BEG  JMP BRAK      :BREAK VECTOR
003-   20 21 0D  STAR JSR INIT      :SET UP SYNTH
006-   AD 10 08       LDA KBD       :INITIALIZE RANDOM
009-   85 D0          STA *NTMP+01  :NUMBER GENERATOR
00B-   20 71 01  LOOP JSR SET       :INIT PINK TUNES
00E-   20 2B 0D  LP0  JSR NOTE      :PLAY NOTES READ AGO
                :
                :CHECK FOR ADDITTIONS TO CANDIDATE
                :NOTE TABLE
                :
011-   A5 E7     MAIN LDA *KTBL+08  :ANY KEYS DOWN?
013-   F0 02          BEQ OUT1      :NO-CHECK FOR TIME OUT
015-   C5 EC          CMP *TEMP     :YES-A NEW KEY?
017-   85 EC     OUT1 STA *TEMP     :SAVE FOR NEXT TIME
019-   F0 0A          BEQ OUT       :BRANCH IF SAME KEY
                :
01B-   A2 10          LDX 10        :IF NEW KEY SHIFT
01D-   B4 8F     LP3  LDY *NBUF,X   :ALL 16 CANDIDATES
01F-   95 8F          STA *NBUF,X   :DOWN BY ONE
021-   98             TYA
022-   CA             DEX
023-   D0 F8          BNE LP3       :NOT DONE-LOOP
                :
                :NOW CHECK FOR CLOCK TIME OUT
                :
025-   A5 BF     OUT  LDA *CLCK     :GET MASTER CLOCK
027-   D0 1A          BNE TEST      :AND IF TIMED OUT
029-   A5 A9          LDA *TMPO     :SET TO TEMPO VALUE
02B-   85 BF          STA *CLCK     :CALL SUB FOR NEW
02D-   20 53 01       JSR ALOC      :NOTES (IF NEEDED)
030-   A5 A8          LDA *LNTH     :GET CYCLE STATUS
032-   8D 20 08       STA DISP      :SHOW IT AND IF ZERO
```

```
035-    F0 0C       0700        BEQ TEST     :CYCLE IS COMPLETE
037-    C6 A8       0710        DEC *LNTH    :IF NOT DONE, DCRMNT
039-    D0 08       0720        BNE TEST     :IF NOT ZERO NOWLEAVE
03B-    20 71 01    0730        JSR SET      :IF ZERO, REINIT
03E-    20 53 01    0732        JSR ALOC     :GET FIRST NOTES AND
041-    F0 CB       0735        BEQ LP0      :BRANCH ALWAYS TO PLAY
043-    20 00 0F    0740   TEST JSR DECD     :GET A COMMAND
046-    D0 0C       0750        BNE TST2     :NOT ZERO,NEXT TEST
048-    A2 03       0755        LDX 03       :COMMAND 0,NEW TUNE
                    0757   :                  SET POINTER/COUNTER
04A-    20 00 01    0760   TST1 JSR RNDM     :GET RANDOM NUMBER
04D-    95 CF       0762        STA *NTMP,X  :NEW INITIAL RANDOM
04F-    CA          0764        DEX          :POINT TO NEXT
050-    D0 F8       0766        BNE TST1     :NOT DONE - LOOP
052-    F0 B7       0770        BEQ LOOP     :BRANCH ALWAYS
054-    C9 01       0780   TST2 CMP 01       :COMMAND 1, TUNEING
056-    D0 0C       0790        BNE TST4     :NOT 1, TEST NEXT
058-    A2 04       0800        LDX 04       :4 OUTPUT BUFFERS
05A-    A9 5C       0810        LDA 5C       :PUT MIDDLE C IN ALL
05C-    9D DB 00    0820   TST3 STA NT0B,X   :OUTPUT BUFFERS
05F-    CA          0830        DEX
060-    D0 FA       0840        BNE TST3     :NOT DONE-LOOP
062-    F0 AA       0850        BEQ LP0      :BRANCH ALWAYS
064-    C9 02       0860   TST4 CMP 02       :COMMAND 2, STOP
066-    D0 A6       0870        BNE LP0      :NO COMMAND - LOOP
068-    20 71 01    0880        JSR SET      :CALL TO ZERO OUT-BUFFS
06B-    20 2B 0D    0890        JSR NOTE     :THEN MUTE SYNTHESIZER
06E-    00          0900        BRK          :AND RETURN TO PIEBUG
```

SUBROUTINES

```
                    0220   :     RANDOM NUMBER GENERATOR
                    0230   :
                    0231   :ESSENTIALLY A 22 BIT LONG SHIFT
                    0232   :REGISTER WITH EX-OR TAPS AT
                    0233   :STAGES 22 AND 21 FED BACK TO
                    0234   :INPUT.
                    0235   :
100-    8A          0240   RNDM TXA          :SAVE X
101-    48          0250        PHA
102-    A5 A5       0260        LDA *NOIS+01 :LAST BYTE S/R
104-    0A          0270        ASL          :ALIGN BITS 22 &
105-    45 A5       0280        EOR *NOIS+01 :21 AND DO EX-OR
107-    0A          0290        ASL          :THEN SHIFT RE-
108-    0A          0300        ASL          :SULT TO CARRY
109-    0A          0310        ASL
10A-    A2 03       0320        LDX 03       :SET UP PNT/CNT
10C-    36 A4       0330   LP1  ROL *NOIS,X  :AND SHIFT 3 BYTE
10E-    CA          0340        DEX          :SHIFT REGISTER
10F-    D0 FB       0350        BNE LP1      :BY ONE BIT LEFT
111-    68          0360        PLA          :WHEN DONE RE-
112-    AA          0370        TAX          :STORE X REG.
113-    A5 A7       0380        LDA *NOIS+03 :AND LEAVE WITH
115-    60          0390        RTS          :WITH NO. IN ACC.
                    0400   :
                    0410   :        NEW NOTE
                    0411   :
                    0412   :TAKES CARE OF PICKING PINK NOTE
                    0413   :FROM CANDIDATE NOTE TABLE AND
                    0414   :CALCULATES AND UPDATES NOTE TIMERS
                    0415   :NOTE THAT Y POINTS TO CHANNEL FOR
                    0416   :UPDATE
                    0420   :
116-    A2 05       0430   NWNT LDX 05       :SET UP PNT/CNT
118-    A5 EA       0440        LDA *OUTS    :GET COPY PINKING
11A-    C6 EA       0450        DEC *OUTS    :COUNTER,DEC ORIGINAL
11C-    45 EA       0470        EOR *OUTS    :PATTERN OF CHANGED
11E-    85 EB       0490        STA *OUTT    :BITS - SAVE CHANGES
120-    A9 00       0500        LDA 00       :PREPARE TO SUM DICE
122-    46 EB       0510   NW1  LSR *OUTT    :CHECK FOR CHANGED
124-    90 0A       0520        BCC NW2      :BIT - IF CHANGED,
126-    48          0530        PHA          :SAVE CURRENT TOTAL
127-    20 00 01    0540        JSR RND      :GET RANDOM NUMBER
12A-    29 03       0550        AND 03       :MAKE RANGE 0 TO 3
12C-    95 9F       0560        STA *RAND,X  :SAVE VALUE FOR NEXT
12E-    68          0570        PLA          :RECOVER TOTAL
12F-    18          0580        CLC          :PREPARE ADDITTION
130-    75 9F       0590   NW2  ADC *RAND,X  :ADD VALUE OF DIE
132-    CA          0600        DEX          :POINT TO NEXT
133-    D0 ED       0610        BNE NW1      :LOOP IF NOT DONE
135-    AA          0620        TAX          :USE TOTAL AS POINTER
136-    B5 90       0630        LDA *NBUF,X  :GET CANDIDATE
138-    F0 03       0640        BEQ DURA     :ZERO, DO NOT CHANGE
13A-    99 BF 00    0650        STA NTB7,Y   :PLACE IN TEMP BUFFER
13D-    A5 A5       0660   DURA LDA *NOIS+01 :A CHEAP RANDOM NO.
13F-    18          0670        CLC          :PREPARE
140-    39 8B 00    0680        AND MASK,Y   :MASK DURATION VAL.
143-    79 87 00    0690        ADC TIME,Y   :ADD MINIMUM VAL.
146-    29 0F       0700        AND 0F       :AND MASK RESULT
148-    AA          0710        TAX          :USE AS COUNTER AND
149-    A9 01       0720        LDA 01       :DO DURATIONS AS
14B-    2A          0730   NT2  ROL          :POWERS OF 2. CARRY
14C-    CA          0740        DEX          :SET DOTS NOTE
14D-    D0 FC       0750        BNE NT2      :NOT DONE - LOOP
14F-    99 C3 00    0760        STA NTB8,Y   :PUT RESULT IN NOTES
152-    60          0770        RTS          :TIMER AND RETURN
                    0780   :
                    0790   :ALLOCATION 0151
                    0791   :
                    0792   :SEES IF NEW NOTES ARE NEED AND IF
                    0793   :SO GETS THEM.  ALSO CLEARS TRIGGER
                    0794   :OF NOTE OUTPUT ONCE IT IS PLAYED.
                    0800   :
153-    A2 04       0810   ALOC LDX 04       :DO 4 NOTE CHANNELS
155-    D6 C3       0820   LP6  DEC *NTB8,X  :DECREMENT NOTE TIMER
157-    D0 07       0830        BNE LP5      :AND IF TIME OUT
159-    8A          0840        TXA          :TRANSFER X REG. TO
15A-    A8          0850        TAY          :TO Y
15B-    20 16 01    0860        JSR NEW      :AND GET NEW NOTE
15E-    98          0870        TYA          :AND DURATION AND
15F-    AA          0880        TAX          :RESTORE X
160-    CA          0890   LP5  DEX          :DECREMENT COUNTER
161-    D0 F2       0900        BNE LP6      :IF NOT DONE - LOOP
163-    A2 04       0920        LDX 04       :AGAIN, FOUR CHANNELS
165-    B5 BF       0930   AL1  LDA *NTB7,X  :GET NOTE FROM TEMP
167-    95 DB       0940        STA *NT0B,X  :BUFFER, SAVE IN OUT
169-    29 3F       0950        AND 3F       :BUFFER, CLEAR FLAG
16B-    95 BF       0960        STA *NTB7,X  :PUT BACK IN TEMP.
16D-    CA          0970        DEX          :POINT TO NEXT
16E-    D0 F5       0980        BNE AL1      :NOT DONE - LOOP
170-    60          0990        RTS          :DONE, RETURN
                    1000   :
                    1010   :          SET
                    1011   :
                    1012   :PREPARES KNOWN STARTING POINT FOR
                    1013   :CYCLIC TUNES.
                    1014   :
                    1020   :
171-    A9 00       1030   SET  LDA 00       :TO ZERO THINGS WITH
173-    A0 01       1040        LDY 01       :PRESET FOR NOTE CNTRS
175-    A2 04       1050        LDX 04       :DO 4 CHANNELS
177-    95 DB       1060   LP10 STA *NT0B,X  :ZERO OUT-BUFFERS
179-    95 A0       1070        STA *RND0,X  :ZERO 4 DICE
17B-    94 C3       1080        STY *NTB8,X  :PRESET NOTE TIMERS
17D-    48          1090        PHA          :SAVE THE ZERO
17E-    B5 CF       1100        LDA *NTMP,X  :SET UP RNDM'S S/R
180-    95 A4       1110        STA *NOIS,X  :AND CYCLE COUNTER
182-    68          1120        PLA          :RECOVER ZERO
183-    CA          1130        DEX          :POINT TO NEXT
184-    D0 F1       1140        BNE LP10     :NOT DONE - LOOP
186-    85 A0       1150        STA *RND0    :ZERO 5TH DIE
188-    85 EA       1160        STA *OUTS    :ZERO PINKING COUNTER
18A-    60          1170        RTS          :AND RETURN
                    1180   :
                    1190   END  .EN
```

LETTERS:

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notation. In order to derive maximum utility from the language, the notation system would have to be capable of expressing all characteristics of the music, not just tempo, pitch and duration, but patch changes, different types and depths of effects -- everything that can be executed on the computer music system (presumably the ultimate music system will be capable of every parameter that can be controlled). The old system of notation really can't handle this, at least, it wouldn't be the most efficient at handling this. Another problem of this language is that, somewhere along the line, it must be able to interpret real-time playing of the material which is an entirely different set of circumstances than composing by typing in the information. The actual playing of the piece involves complex human inflections which usually aren't notated and are not precise, yet the program must interpret these accurately. It should be clear that the techniques for real-time interpretation are quite different from straight composition. All in all, it sounds like this high-level language would be one hairy beast and any attempt to limit it to a certain form would only hasten obsolescence (can you predict what we will be doing musically in 2001?).

I would prefer that you simply establish some conventions as to how a few of the more commonly notated characteristics can be coded onto a cassette. In this manner, someone who already has a computer music system can link the notation subroutine to his language, no matter what techniques he uses to produce computer music (presumably all systems can specify pitch, note durations, tempo, etc.) and with it dump a coded cassette with the proper data and proper format to be interpretted by the automatic typography machine. A good analogy would be something like the BYTE standard cassette interface for microcomputers. This interface is carried out using many different hardware approaches and many different types of control software, yet they are compatible. They even have been carried out on different processors, yet it is still possible to read someone elses BYTE cassette on your BYTE interface. You can load a program that was dumped by

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INDUSTRY NOTES

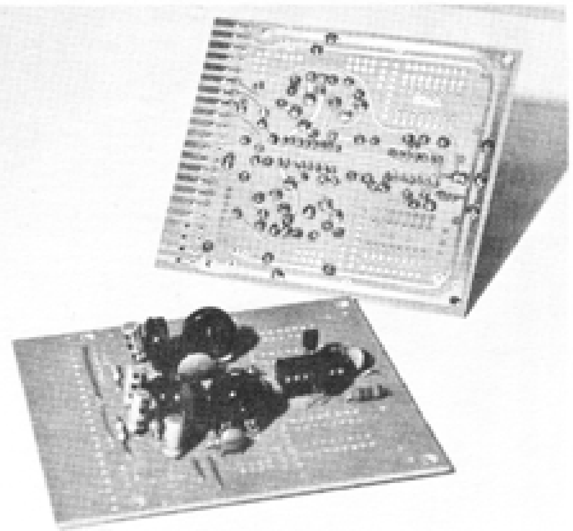
Tell them you saw it in POLYPHONY

JOB OPENING - OBERLIN COLLEGE

Oberlin College Conservatory of Music announces an opening for an Electronic Music Technician. Duties will include general responsibility for the maintenance of all equipment in the Technology in Music and the Related Arts (TIMARA) program, over-see the electronic music studios and give aid to the users, design and construct new equipment as deemed necessary by the TIMARA program committee, and supervise student assistants in the maintenance of the studio. If qualified, may be requested to teach a course in analog/digital circuit design. Qualifications include a thorough knowledge of analog and digital technology, interest and experience in electronic music synthesizing systems, digital sound generation systems, and tape recording systems. A degree in electronics such as BSEE is desirable, but not required. Interested persons should contact Dean David Boe, Oberlin College Conservatory, Oberlin, Ohio 44074. The position is to be filled by September 1, 1978 at an annual salary of $10,200 or higher, depending on experience and qualifications.

NEW LITERATURE

The Institute of Sonology (University of Utrecht, Plompetorengracht 14-16, 3512 CD Utrecht, The Netherlands) is publishing a small Newsletter twice a year to inform interested parties of events at the Institute. In previous issues have been listings of performances at the Institute, manuals and educational material used by and available from the Institute, descriptions of courses available, and published papers written by permanent and visiting staff members. They also announce that preparations are in the advanced stages for the publication of an International Electronic Music Discography, complied by Kondracki, Bergshoeff-Stankiewicz, and Weiland. With more than 2000 entries, this discography is the largest of its kind in the field of electronic and concrete music, live electronic music, and computer music. Those interested in registering now to receive information when the book is done should write directly to the Institute at the above address.



ANALOG DELAY EXPERIMENTER'S KIT

PAIA Electronics, Inc. (1020 W. Wilshire Blvd., Oklahoma City, OK 73116) announces the release of the EK-5 Analog Delay Line Experimenter's Kit. Designed as an introduction or departure point for the advanced hobbyist, the kit contains a Reticon SAD-1024 analog delay IC, a CD4013 dual flip-flop for clock conditioning, and several other passive components required to get the delay IC working. Additional parts are to be supplied by the user, and requirements vary depending on the intended application. The kit sells for $24.95, and can be purchased directly from the manufacturer.

"SILENT PLUG" AVAILABLE

The Switchcraft #172P1 "Silent Plug" is a 1/4 inch phone plug designed to eliminate the common thumps, hum, and buzz which occurs when inserting or removing plugs while they are connected to amps or recorders. An integral switch is used to short the two conductors of the plug before the tip connection has an opportunity to touch the jack bushing or the musician's body. The switch is mechanically opened when the plug has been fully mated with the jack. For more information contact : Switchcraft (5555 N. Elston Ave., Chicago, IL 60630) or your local stocking distributor.

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INDUSTRY NOTES

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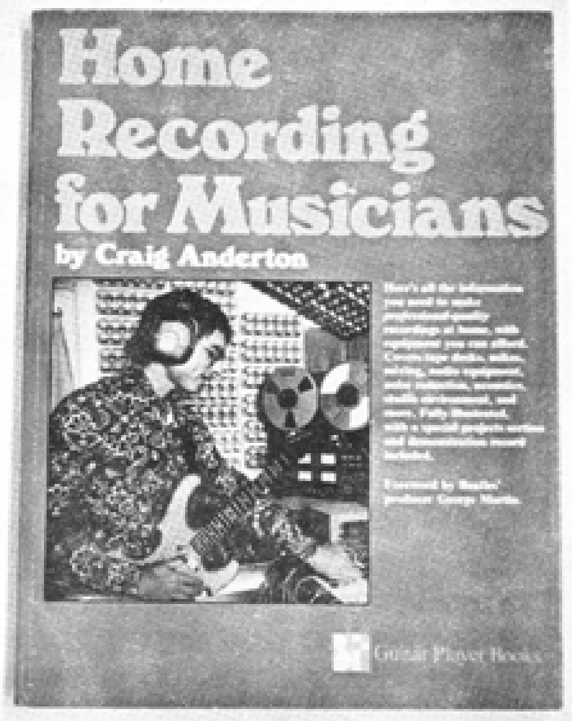

"HOW TO" BOOK ON HOME RECORDING

Guitar Player Books (box 615, Saratoga, CA 95070) announced the release of Craig Anderton's second book for them, "Home Recording for Musicians". Selling for $9.95 (mail orders include 50¢ postage/handling, California residents add 6 1/2 ¢ sales tax) the book provides a complete introduction on setting up a home demo studio. Included are several hobbyist projects which allow the reader to construct a small mixing console. A soundsheet is bound into the book, and demonstrates several of the effects and techniques discussed throughout the book.

MICROTONAL NEWSLETTER

INTERVAL / A Microtonal Newsletter is a new quarterly publication dedicated to coverage of microtonal music, instruments, theory and techniques. Subscriptions are $8 per year, and should be sent to Interval, Box 8027 San Diego, CA 92102. The premier issue contains several reviews of books, live performance, and an article on design theory and construction of a Chrysalis (microtonal instrument). George Secor and Ivor Darreg also discuss the options available for microtonal tunings, including the 17 tone octave and the Secor Nineteen-Plus-Three.

COMPUTER SHOWS

The end of summer and approaching fall mean lots of people are getting ready for this year's hobby computer fairs. Many of the fairs are featuring a bit less hobby material, and focusing more on small business systems and such. These shows are still of interest, however, and here are the major ones currently slated:

August 24 - 27
Personal Computing 78 (the biggy!)
Civic Center
Philadelphia, PA

September 15-17
2nd National Microcomputer Expo and Conference
Coliseum
New York, NY

September 29 - October 1
International Microcomputer Expo
Expocenter
Chicago, IL

November 3 - 5
3rd West Coast Computer Faire
Los Angeles, CA

FOR MORE INFORMATION on any of the products listed in this column write directly to the manufacturer.

LETTERS:

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an 8080 into your 6502, naturally the program won't run, but it will still be the same data that was stored on the 8080 and data is what we are concerned about here. As long as the system can provide data for pitch, tempo, etc. in a format that can be interpretted correctly, it doesn't matter how that data was generated. It is not necessary that everyone be running the same music language software (that universally acceptable one you mentioned) they simply need a routine to dump the proper data in the proper format. This could require a complex interpreter to convert between the user's language and the tape format language (ie the conventions of the tape-format. The conventions might require that A sharp be stored as an ASCII A# or that A sharp be stored as some other hex code, etc. These conventions must be determined, with many variables going into this determination-- i.e. should it be stored in ASCII so people can understand it? Or should it be stored as the HEX code to a DAC, or is there some easy way to express things so that interpretation between the user's language and the conventions are simplified). I would prefer that the linkage between the user's language and the typography conventions (wait, I'll use one of John's favorite words, make that typography protocols) be as simple as possible so that the interpretting software at the typographers handle the major burdens (remember, the interpreter must be capable of actually making the music, even if you never intended to have it do so -- annotating the music is another form of performing it, at least to the computer). This would put the least burden on the user by not forcing every user to buy a complex, expensive language that he/she may not want. It's much more efficient to have a single intelligent central system (the typographer's interpreter) with a lot of simple peripherals (the user's systems) in this case, as only one high priced system is needed. Of course, most software dealers would prefer to sell a lot of expensive systems, but that's not how PAIA operates, right?

An example of how a user's system might differ could be: One person has a computer with a video graphics display which allows him to actually draw notes on his screen. A second person has a high-level language which allows him to enter music information in english words (like "A#") and a third person has a solely real-time data entry system like a digital sequencer. In these cases, each must have a mini-interpreter which will convert from their systems storage format (graphics bit patterns for the first, ASCII words for the second and binary words for the third--obviously, all three have their information stored in binary code, but all three codes are different) into standardized format of the music notation system (whatever is decided to be the most efficient in terms of hardware and interpretting). Each person's cassette interface may be executed in a different manner, but each must be capable of dumping the interpreted information in the standardized format (for example, a possible format might be one block for pitches, 1 block for on duration, 1 block for off duration, 1 block for tempo, etc. Or each block may contain all the information for a single note.

'Til next time,
Bob Yannes

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VCO

By: Russell Grokett, Jr.

Here is a VCO system using one IC and a minimum number of parts.

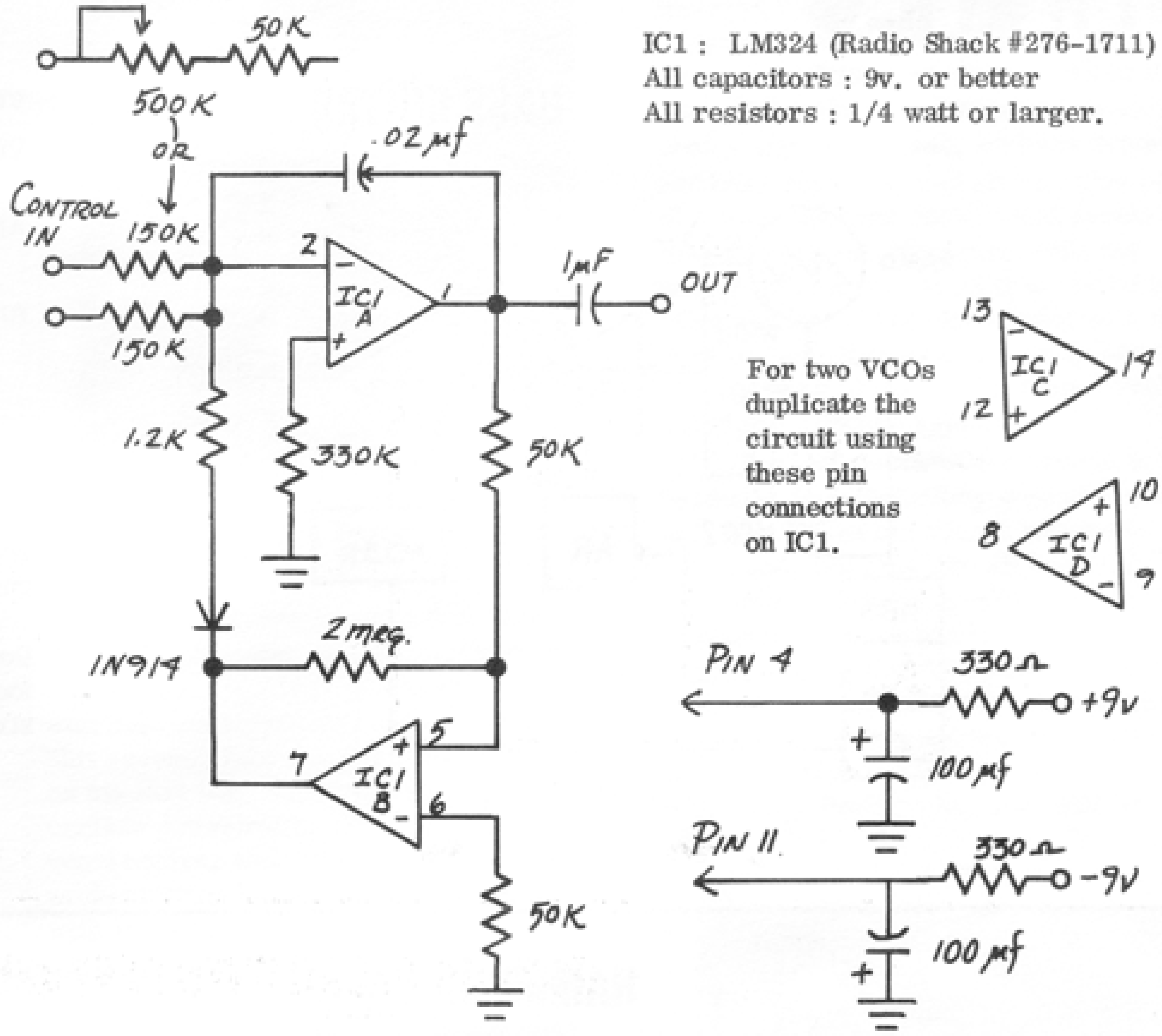
This VCO can have a frequency range of approx. 80 to 10,000 hz by varing the .02uf capacitor. Using a 0 to 5v. input and the capacitor listed, the frequency range is about 300 to 4000 hz with a ramp output of about .5v p-p.

The LM324 used in this circuit is a quad op-amp and since one VCO uses only 1/2 of the IC, each IC can be built into two VCO's.

Construction is not critical and parts can be substituted. Also: One of the input 150k resistors can be replaced with a 500k pot. and a 50k resistor to give a wider frequency range and more flexibility when using multiple VCO's.

I don't know how linear these oscillators are, but for special effects where multiple VCO's are a must, these work quite nicely.

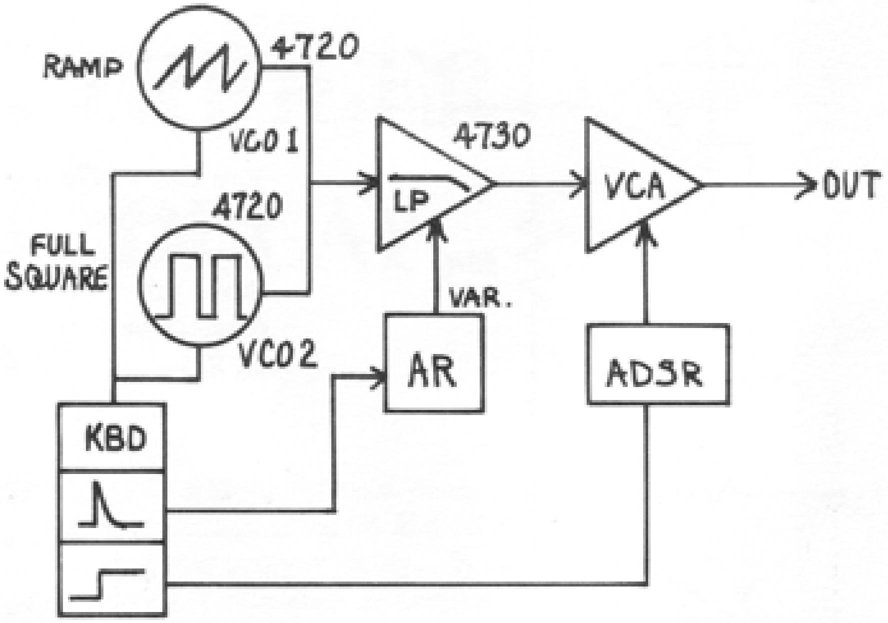
IC1 : LM324 (Radio Shack #276-1711)
All capacitors : 9v. or better
All resistors : 1/4 watt or larger.



For two VCOs duplicate the circuit using these pin connections on IC1.

PATCHES

BASS GUITAR

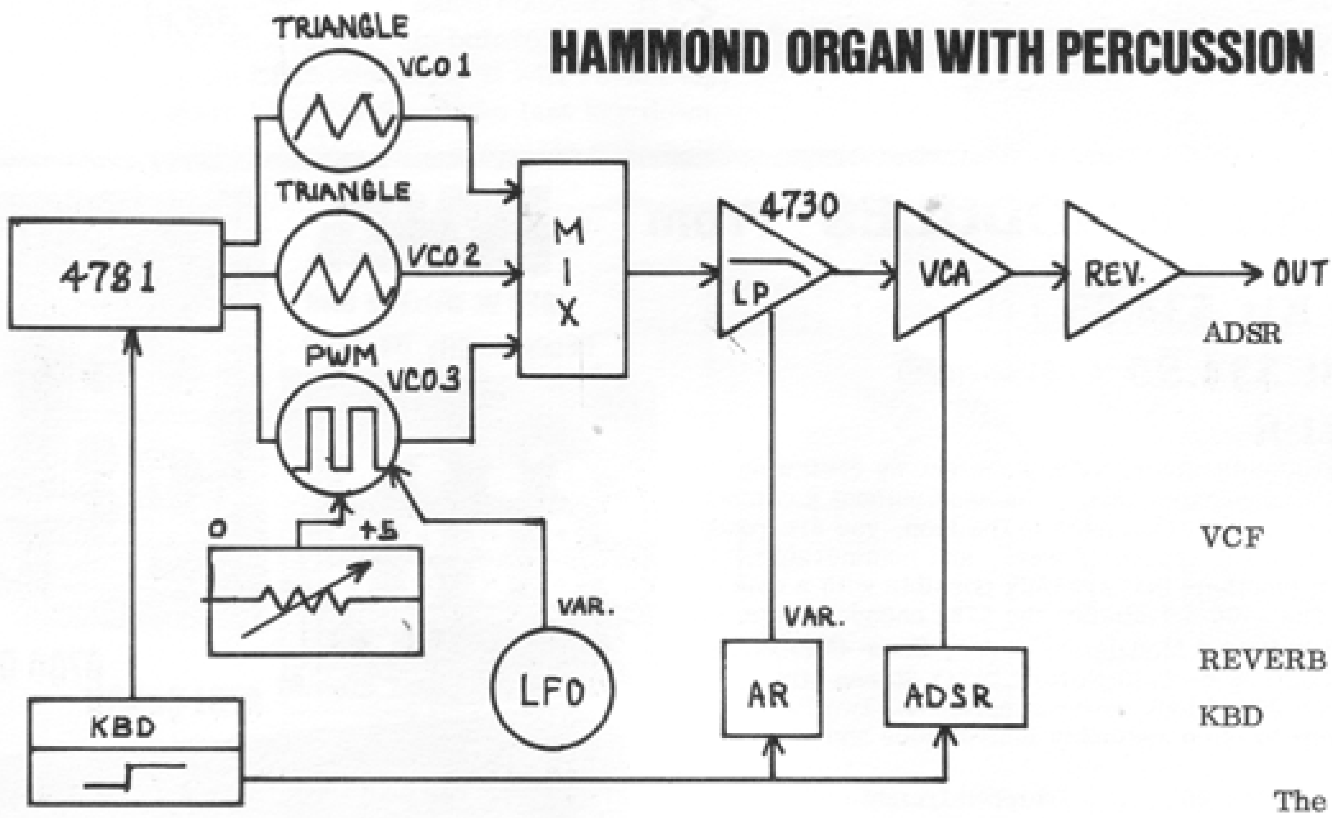


KEYBOARD:	low octave (transpose further with 4781 if needed, for proper range.
VCO 1	One octave above VCO 2.
VCF	Freq. HI; Init. Freq. MIN. Q 60%; Mode SWEEP.
AR	Expand ON; Attack 0%; Decay 40%; Variable 35%.
ADSR	Attack 10% (adj. for slight 'pluck' sound, 'thunking' attack) Decay 30%; Sustain 80% Release 45%.

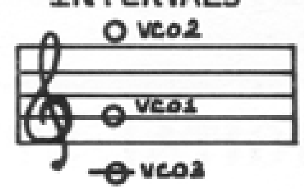
When set up properly, this patch sounds quite a bit like an electric bass guitar, especially when played as accompaniment to other instruments through a good bass amp. Be first to fool that C & W bass picker down the street.

Submitted by:
Dave Garfield
Austin, TX

HAMMOND ORGAN WITH PERCUSSION



INTERVALS



ADSR	Attack - about 5% (enough for initital 'pop' percussion sound, but not too much.) Decay - 15%; Sustain - 45% Release - 45%
VCF	Freq. - HIGH; Init. Freq. = 15% (accent VCO 1) Q - 60%; Mode - SWEEP.
REVERB	To taste.
KBD	Medium to high octave, no glide.

VCO's	2. No transposition -- KBD C.V. out. Triangle. 1. One octave lower than VCO 2, P5 higher than VCO 3. Triangle. 3. Perfect 5th down from VCO 1. Pulse Width Modulation = 25 - 40%.
BIAS	Enough to accent harmonic P 5 up from fundamental - about 1 volt.

LFO	About 5 Hz. Variable output enough for 'vibrato' - about 20% P.W. change.
MIX	VCO 1-60%; VCO 2- 80%; VCO 3-60%; Output (one channel only) - 100%.
AR	Attack 0%; Release 20%; Variable 50%; EXPAND ON.

The main point in this patch is to accent the harmonics that are a P 5 above the fundamental (fund. = VCO 3). It takes some experimenting with parameters, mixing, etc. to get the right effect, but it's worth it. By varying filter freq., and/or oscillator pitches, you can go from Keith Emerson to Deep Purple. This one is another good one for multi-channel recording.

Submitted by:
Dave Garfield
Austin, TX

more...

PATCHES

3750 3/4 DRUMPATCH

MAIN PATTERN

Prep: Reset, program

Main pattern data:

AB, R, T, R, CO, R, AB, R, S, R, R, R,
AB, R, T, R, CO, R, AB, WB, CO, R, S, S,
AB, R, T, R, CO, R, AB, R, B, R, S, R,
AB, R, T, R, CO, R, AB, CL, R, R, S, S,
Repeat, Reset.

BRIDGE PATTERN

Prep: Reset/Bridge, Program

Bridge pattern data:

AB, R, CO, R, T, R, AB, R, R, R, S, CL,
AB, R, B, R, CO, R, AB, CO, S, R, S, T,
AB, R, CL, T, CO, R, AB, R, S, CO, S, R,
R, AB, CO, CO, CO, CL, AB, CL, S, T, CO, CO,
Repeat, Reset ------Play!

EACH PATTERN UTILIZES 48 EVENTS.

3750 5/4 DRUMPATCH

MAIN PATTERN ONLY

Prep: Reset, program

Main pattern data:

AB, S, R, R, S, R, AB, R, CO, R,
AB, CL, R, R, CL, R, AB, R, S, S,
AB, S, R, R, S, R, B, R, CL, R,
AB, CL, WB, T, CO, R, AB, R, S, S,
Repeat, Reset ----Play.

TAKES UP 40 EVENTS This is a 3 + 2 style of 5/4

3750 7/4 DRUMPATCH

MAIN PATTERN ONLY

Prep: Reset, program

Main pattern data:

AB, R, S, R, B, R, S, R, AB, R, S, R, S, R,
AB, R, S, CO, B, R, S, CO, AB, R, S, T, S, R,
AB, R, S, R, AB, R, S, R, AB, R, CL, T, CO, R,
AB, R, S, CO, AB, R, S, CO, AB, R, S, S, S, R,
Repeat, Reset, Play.

Takes up 56 events
THIS IS A 4 + 3 STYLE OF 7/4.

COMMENTS

ALL DRUM PATCHES BY: CRAIG ANDERTON

The drum abbreviations are as follows:

AB - Accent Bass
B - Bass
S - Snare
T - Tom Tom

CO - Conga
CL - Clave
WB- Wood Block
R - Rest

Other programming steps are self-explanatory.

more...

LETTERS:

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A TROUBLESHOOTER'S COLUMN?

Yep, I too enjoy the new look of POLYPHONY recently. Especially the fact that you seem to be putting more into each issue.

First, I agree 100% with Dave Mays (Letters, April/May '78). A common problems and troubleshooters guide hits the nail right on the head. I guess everyone has, at one time or another, had icky little problems pop up unexpectedly. They don't seem worth the trouble to pack up and send back to the manufacturer (let alone the time involved). But sometimes we hesitate to tear into our synthesizers, expecially if it is a very subtle problem. A small tip or modification or circuit update column in each issue could be a real lifesaver.

I was somewhat disappointed that there were only six patches in the last issue. This is my favorite section!

Oh, by the way, when does my subscription expire? I figure I should be about due. I sure don't want to miss anything.

Bye-
Ron Jones
Placentia, CA

Ron -

Thanks for the comments on POLYPHONY's new look. The idea for a troubleshooters column has met with a fairly good response from our readers and we are working on making this a permanent column. With all the brands of synthesizers on the market, we could discuss various types of problems without ever running out of material! One thing we need to know from our readers, though, is: what little problems have you been having with your gear, what make is it, and what the model and serial number is. We need the latter information because many "bugs" are modified by the factory and are only applicable to units in a particular range of serial numbers. Also, be sure to check with local service agencies for your type of equipment. Many manufacturers will issue an announcement to their service centers notifying them of small circuit modifications which the factory has started adding to later models. Customers with earlier units can usually have the update installed for minimal cost at the service center. POLYPHONY occasionally runs Spotlight features on service centers around the country, and don't forget to check the Yellow Pages for service centers if your manufacturer didn't provide a list with the instrument.

I'm sorry about the decreasing number of patches in POLYPHONY, however this sections seems to get little response from most readers. If you readers are in fact interested in more patches, send us your favorites to help us build up a good selection for publication. POLYPHONY, in general, is based very heavily on suggestions, requests, and (most important) contributions from our readers. If we don't hear from readers concerning a particular column or article, we think you would rather see other types of material. Please let us know.

POLYPHONY subscriptions are valid until you receive a letter in the mail asking you to re-subscribe. If you suspect that you need to renew, please don't send a subscription until we notify you, as this confuses our computer (temperamental devils aren't they). We will notify readers in plenty of time so they won't miss an issue, don't worry! Thanks for the good words.

-Marvin-

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rhythm (in 11, I think - it's hard to keep track of the count!) as a basis for a good heavy rock concept. Part two, "By the Light of Day", is a mellower section in which Jobson displays the subtleties of his violin technique. The violin is processed through a phaser and echo, yielding a sound not unlike Jean Luc-Ponty. The middle of this piece features a good soft synthesizer solo, but quickly moves to a tremendous sequence of gliding chord structures in a low register on the Yamaha CS 80 polyphonic synthesizer. The third part of the suite is "Presto, Vivace, and Reprise". Initially free form in rhythmic structure, the piece develops into a fast stacatto discussion between synthesizer and violin similar to many of the Mothers early works. In summation, the "Dead of Night" theme is recalled. The remaining work on side one is "Thirty Years" which displays perhaps the best string synthesis ever -- on the Yamaha again. The string parts are also orchestrated VERY well, and this helps to reinforce the illusion. Far too many keyboardists are playing string synthesizers like organs, the result being an ethereal organ part. With proper orchestration and, in particular, open chord voicings which are so typical of orchestral arrangements, the effect is much better. The wide range of the string voice on the Yamaha generates cellos which are very rich, and really warps your head when the glide is turned on. The flanged vocals and wailing guitar solo add nice touches. Side two starts with "Alaska" which is an instrumental featuring the CS80. The low frequency drone throughout the piece provides a continuous suggestion of power, while various melodic occurances flit through the upper range. What appears to be a digital noise source is put through clock modulation processing and permutates into some good effects similar to HF filter modulation. The driving drums, heavy bass, and percussive organ which join in at the end of the song impart a definite ELP flavor, probably mostly due to the open harmony melodies on the organ. The instantaneous transference into "Time to Kill" via a close spliced vocal chorus provides a good start for the second cut. This otherwise basic rocker is adorned with some tasty violin feedback and harmonics during the solo, and the synthesizer does some sequencing of oscillators with FM waveshaping. "Nevermore" is probably the best demonstration on this album of the overall performance capabilities of both the CS80 and Jobson's mastery of the instrument. What starts with a smooth classical style guitar work and string accompaniment transforms into a more basic rock style, and is soon followed by a dynamic solo section. Jobson puts the Yamaha through the paces by jumping between the preprogrammed patches, all the while varying parameters on the instrument and generally providing a lot of expression with left hand technique. Mellowing out at the end, a lonely droning section with tape echo helps let you down slowly. The closing cut is reminiscent of the standard King Crimson jazz-rock feelings of their first few albums. A multi-tracked guitar solo near the end uses sustainers to give an almost organ quality to the traveling chord structures. The implied hard work, dedication, and mastery of equipment required to pull off this effect is a fitting conclusion for an album with similar implications throughout. I look forward to hearing more from these guys.

And Then There Were Three by Genesis
Atlantic SD 19173

The first thing that hits you on this album is the tremendous fullness, activity, and power for a three piece group. Of course, they weren't always three pieces. So, that makes the transition even more important. It's hard to loose a lead guitarist and vocalist, and continue with the same intricate type music. But, Genesis has managed - barely. It is obvious that Tony Banks' keyboard mastery has improved. He plays a few more instruments, processes them through a number of effects, and is a bit more creative in his use of synthesis. Musically, however, his writing and harmony realizations are much less artistic than the Foxtrot or Lamb Lies Down on Broadway periods. Perhaps he just has more to concentrate on now. Phil Collins' drum work seems a bit tighter, more powerful. His execution of complex time patterns is more definite. And on top of that, he is doing most of the vocals, which are wide in range and expression. Mike Rutherford's guitar work does a good job at filling the void left by Steve Hackett. Previously, Mike's parts had remained in the background, but careful listening would always show that his licks and fills were tasteful and exciting. Probably the largest disappointment of this album is that the combined songwriting talents of Genesis has very little scope. Perhaps partially due to the absence of Peter Gabriels intense and diversified imagination, most songs use similar instrumentation and orchestration techniques. Electronic effects are mostly time domain processing (flanging, chorusing, and echo) and the continuous richness of sound soon becomes tiresome. It's surprising that producer David Hentschel, quite well versed in electronic music techniques and previously quite tasteful, should leave this album at this level of repetition and lack of depth.

The pitch bending on the string synthesizer in "Snowbound" is a good effect overlooked by many when using strings. "Burning Rope" probably has the best musical composition, although the execution is about what you would expect by the time you get to this point of the album. "Deep in the Motherlode" uses backwards drums at the end; this is about the only excursion into tape processing. "Many too Many" contains the most convincing string synthesis of the album. "Scenes from a Night's Dream" is my favorite cut on the album due to the use of motion in the bass line, something Genesis has never used much due to the use of pedal bass instead of string bass. The alternation between triple and duple times at the end of this piece generates more interest than most alternating time signatures, and gives a different feel to the 'good ole fadeout'. The bending, expressive string/brass solo line in "The Lady Lies" is a good indication of Banks' left hand technique. It's unfortunate that he generally has to be playing more instruments, and can rarely get involved with the left hand. The current chart climber "Follow You Follow Me" closes the album but stirs little interest. Probably the best part of this song is the synthesized percussion throughout, or the recurrence of what used to be the "standard" Genesis fuzz guitar sound - bright and buzzy with lots of clean high harmonics. How strange that Rutherford should depart from that effect on this album. Well, maybe he's trying to break away. They need it.

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REVIEWS!

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Additional late releases which may be worth your perusal include the double album production piece based around "War of the Worlds". There is some good synthesis here and there, but most effects are studio techniques. Justin Hayward obviously had a lot to do with orchestrations and arrangements as many songs have a Moody Blues feel to them.

Peter Gabriel's second album is out, and displays his talents very well. A variety of synthesizer parts appear very tastefully, and Peter's songwriting is very powerful and innovative.

David Gilmore, guitarist for Pink Floyd, has a solo album out, but it tends to sound like the rockier part of Pink Floyd without the electronics they have been using lately.

Steve Hackett, formerly pickin' for Genesis, has released his second album with a tremendous offering of not only guitar performance, but also production and composition talents. Various artists help Steve with vocals, but he handles the musical performances on a variety of instruments very well indeed.

cut out with scissors

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