POLYPHONY

ELECTRONIC MUSIC & HOME RECORDING

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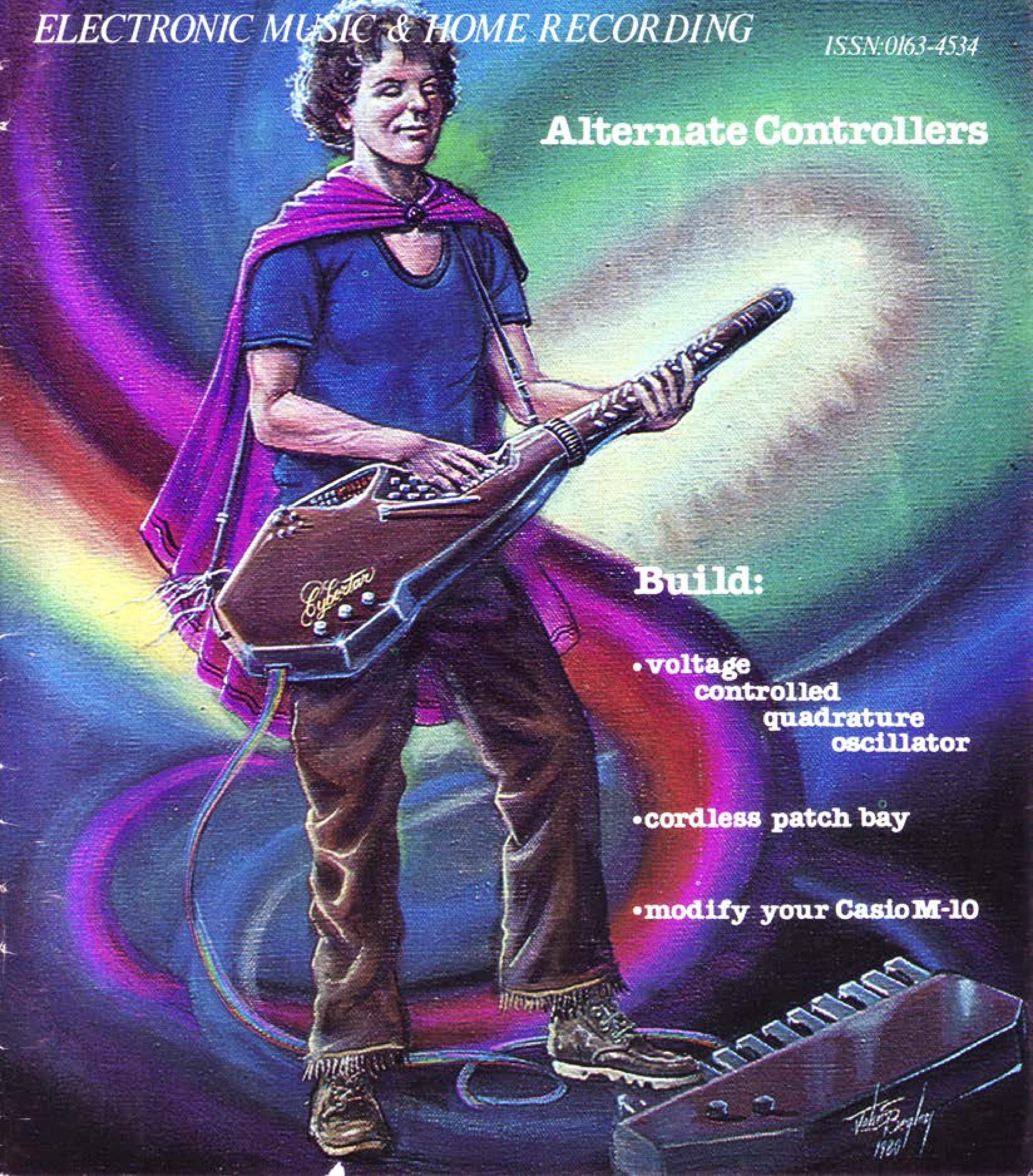

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November/December 1980

Volume 6, Number 3

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CONSTRUCTION AND MODIFICATION

COLUMNS AND REGULARS

"Polyphony is a source of useful information for those of us who are into the technical side of electronic music. I look forward to receiving future copies, and to being kept in touch with the ideas and activities of electronic musicians everywhere."

Bob Moog

NATIONAL ASSOCIATION OF MUSIC MERCHANTS
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LETTERS

Easy does it

I recently purchased my first synthesizer and I am just beginning to learn the basics of electronic music. Since I realize there is much more to playing a synthesizer than turning knobs and flipping switches, I subscribed to Polyphony in the hopes that I might learn something about synthesizers and electronic music. What a mistake!! I can't decode any of the strange scribblings and I can't decipher any of the bizarre hieroglyphics. Since I am a beginner, I understand very little of your magazine. I could learn as much about electronic music by reading the phone book! Now don't get me wrong. I think your magazine has great potential, if only you would consider the beginner. Possibly you could have one regular article about Basics. I think this would make your magazine 100% more enjoyable to me and other beginning synthesists. Thank you at least for an intriguing magazine.

Dave Scott
Cincinnati, OH

I think you will find a wide variety of simpler projects in this issue. Due to the nature of the material in many of our feature articles, it is sometimes hard to get the point, or all the details, across without turning the article into a 'monster'. Also, it just happened that we received a number of in-depth articles recently. I'm sure the material slated for upcoming issues will provide plenty of smaller construction, modification, and tutorial features.

Marvin

OOPS!

Imagine my disappointment upon receiving the first issue of my new subscription to Polyphony and finding incorrect and misleading information given prominent display on page 9. I am referring to the section entitled "Prophet Update", and particularly to the paragraph describing the variable scale tuning system. This is not a "totally unique feature" as your article exclaims. Nor is it the "first time, in a standard commercially available instrument, the musician can use other scales." This feature has been available in the P-series synthesizers from Korg for some time.

I also question the amount of flexibility provided by ±½ semi-tone. I would hope that before a new product would be so highly praised, some comparisons to other available products would be made. Since that has apparently not been the case, perhaps a review of the Korg instruments is now in line.

Eric Rothenbuhler
Worthington, OH

All material featured in our Industry Report column comes directly from the businesses involved, in the form of press releases. Naturally, the manufacturers are going to point out the better qualities and features of their products, and the enthusiasm about their products is also many times quite obvious. Nevertheless, these press releases offer first-hand (and in many cases, preproduction) announcements of new products and events which might otherwise take several months to be advertised or picked up by the media in other ways. We run as many of the announcements we receive as possible in order to give our readers an overview of what's happening and what's available in the industry. We hope that the emergency of the material presented in Industry Notes will outweigh the occasional "gloss" and "promo" which might be provided by the manufacturers and organizations.

Marvin

Poly-critique

First I would like to compliment your efforts in Polyphony, however there is a noticable weakness in technical editing. I will not accuse the writers as they appear to be little more than hobbyists or experimenters reveling in their latest successes. I can appreciated being limited to contributed works and finite resources. I think your magazine has a flavor that experimenters can identify with, because of the source material. However you could tighten up the technical content without sacrificing the style or image.

I will take a few examples from your May/ June issue.

1. In Mr. Perrin's answer to a poorly worded question he makes an assumption that qualifies his answer but will confuse the readers. Indeed Lissajou patterns can be used to compare phase of complex waveforms, while two microphones will generate non-coherent signals. Even in the given case it will be instructive to look at the phase relationship caused by the location and directionality of the microphones.

2. In Mr. Lewis' article on micros in audio, his choice of words could be confusing to anyone approaching the subject for the first time. The transversal filter (comb) he describes is commonly known as a flanger. The phase

shift block is actually a time delay. The transfer function he plotted is almost correct. Assuming he is adding the delayed signal to the direct in phase, the first notch will occur at ½ the frequency whose wavelength is equal to the delay time with notches spaced regularly at increments of that same frequency. Were one of the signals inverted before summing, the notches would still be spaced the basic frequency apart but the first notch would occur at the basic frequency itself. A computer program to create a phaser type phase shift would be quite complex.

3. Mr. Anderton's discussion of pitch shifting was good, but to my taste a bit harsh on the analog approach (which, by the way, is patented). The basic problem for time base compression/ expansion, be it digital or analog, is how to fit the stretched waveform into the original space or, in expansion, what to do with the space left over after you shrink the sample down. While I will concede that digital techniques offer easier splicing routines, the basic problem exists for both. While the best sounding unit I've heard is digital (a $5000 French unit), I've heard decent performance from an analog unit as far back as '71 (in a lab, at least).

4. In the digital audio delay line construction article (Neves/ Kolupaev), the qualification that the design "has slightly less dynamic range" is approaching criminal understatement. The few hardy souls who build that turkey will get an instant education in why pro audio uses 16 bit converters! I can see why the authors recommend a -20 dB nominal setting (a 48 dB dynamic range will make a cassette deck sound hi-fi by comparison). Just for the record, home delay systems run nominal levels of -3 dB to -9 dB with initial delays in the 30 - 50 mS range.

I apologize if some of my examples are nit-picking, but that is exactly what I set out to do. You have an excellent magazine that can (should) be made much better. It is very important that experimenters and beginning hobbyists don't get sent in the wrong direction and discouraged from a truly rewarding field.

John Roberts
Phoenix Systems
Monroe, CT

more LETTERS

Thank you for your observations. You are obviously one of our readers for whom precise, highly technical articles are the order of the day. But as a previous letter attests, many Polyphony readers are not so technically oriented, indeed are sideline recording or performing musicians, "weekendwarriors," tinkerers. Polyphony will strive to the depth of its resources to be interesting and accurate in all articles, regardless of depth and scope. You'll soon see fun little construction "cameo's" and modifications sandwiched between feature in-depth articles. Only in this way can we walk the wire. The DDL project by Neeves has recently been updated to include a compander for 11 bit plus sign resolution, and better than 70db dynamic range. Not bad for some $80 worth of major parts!

Jim Riter

EDITORIAL

Left to right back row: John Simonton, Cynthia Rueb, Peggy Walker, Dinky Cooper. Front row left to right: Linda Kay Brumfield, Ramona French, Cathi Diehl, Cindy Edwards, Jim Riter.

Dear Readers;

Polyphony is currently undergoing a period of rapid change and readjustment. Marvin Jones, our Editor of over 6 years is leaving us to pursue an independent business venture. Although we wish him the very best in his new endeavor, his shoes are hard to fill. During our reorganization we've slipped back the cover date. (Just in case you had not yet noticed the gap in receiving your subscription copies.) We chose to do this rather than put out a less than quality product on deadline. Your subscription is still good for the same number of issues, the volume and issue number are in sequence, it's just that the post office gets awfully jumpy about delivering mail with a stale cover date. They think they'll be accused of delaying the mail.

We've got some really exciting things in store beginning with the expanded January/ February, 1981 issue. We're growing, re-shaping and moulding our publication to meet the needs and interests of all of our readers, from the novice through the experienced electronic music enthusiast.

Meanwhile, we thought you might be interested in actually seeing the people who are involved in bringing you POLYPHONY. Some are familiar faces and some are new but they all have in common the goal of producing the best possible product, from editorial content, through Polymart management, to the actual physical production and mailing of POLYPHONY.

Hope you'll bear with us during this transition period.

See you next month!
Linda Kay Brumfield
Managing Editor

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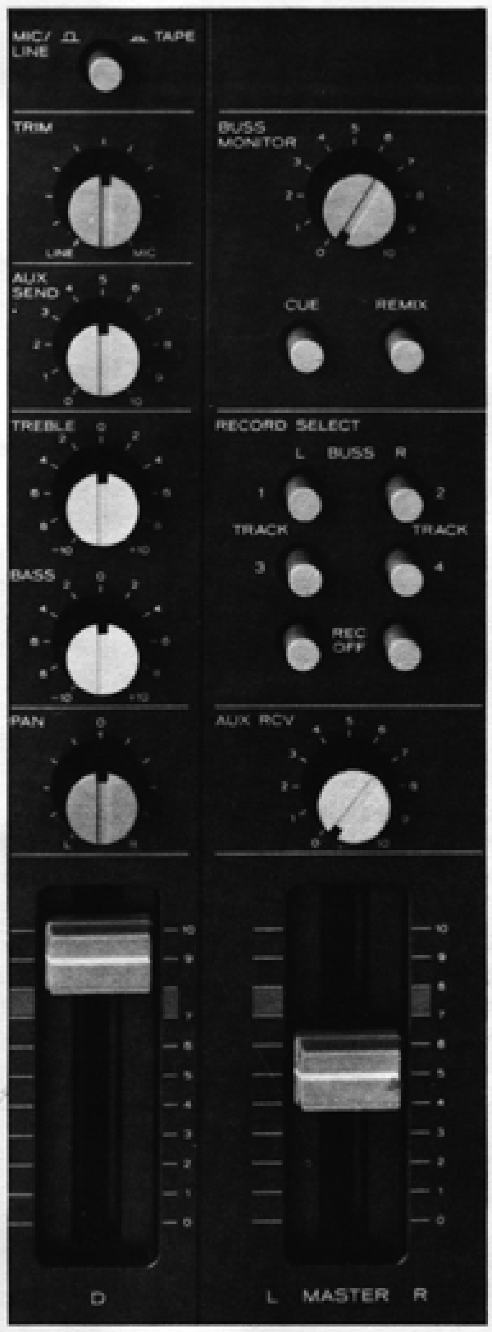

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Alternate Controllers

BY JIM RITER

For quite a while now, the keyboard synthesizer controller has seemed to reign supreme, and indeed, when most people think of a synthesizer, they visualize an organ type keyboard. But is a keyboard necessarily the logical choice for controlling a synthesizer? Might there not be other controllers which, though perhaps new and unfamiliar now, may supercede the keyboard?

Consider the growing popularity of the Moog Liberation or the Performance Systems Syntar which exploit the use of light, mobile controllers. The fact that the Dream Module contest in the last issue was won (over many impressive contributions) by the Cybortar concept of alternate controller may be a portent for the '80's.

Though this is not intended as a construction article we will include schematics for proven circuits that can be developed into full-optioned controllers. In most cases a controller consists of a basic circuit which is repeated a number of times. (see figure 2) So while a non-standard controller may be conceptually easy to understand, it may be the case that the actual construction is rather tedious.

We'll start with an axe conceptually not far removed from the AGO keyboard. Bass pedals (especially those made by Hammond for spinets, such as the M-3) lend themselves admirably to contoller applications. But before considering some basic applications we need to tackle the problem of where to get the basic pedal board unit. Mine came from an old M-3 that I stripped down before going on the road. Perhaps you could find one through a dealer or at a warehouse sale, from a little old lady, or maybe even from a keyboard player with an M-3 who has no taste for what the thing can do.

The unit I have is exceptional in that one and only one note at a time "opens" mechanically when a pedal is depressed. In effect, this is a mechanical "sample and hold", and so drift is not a problem. All that is needed to complete the project is a batch of some simple inverters to obtain the positive logic for the D/A and computer interface. By the way, in addition to the switches mentioned above, there is a contact switch available that closes anytime ANY note is depressed. That's provision for your gate signal right there.

By far the most common application of this bass pedal board to Paia or other digitally controlled synthesizers is the tying into the scanning encoder at a point "lower" (so far as binary words are concerned) than is occupied by the AGO keyboard. The easiest way to do this is to use two 4049 hex inverters (for 12 note total) to provide positive logic from the one and only note depressed, and "encode" the pedals by using diodes to build the binary word required for a D/A. This is the most cost effective

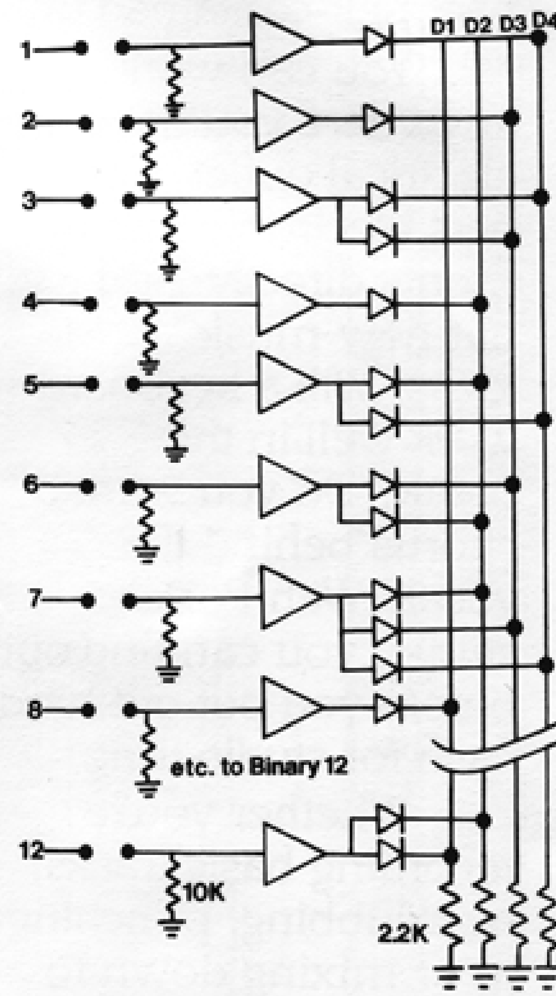
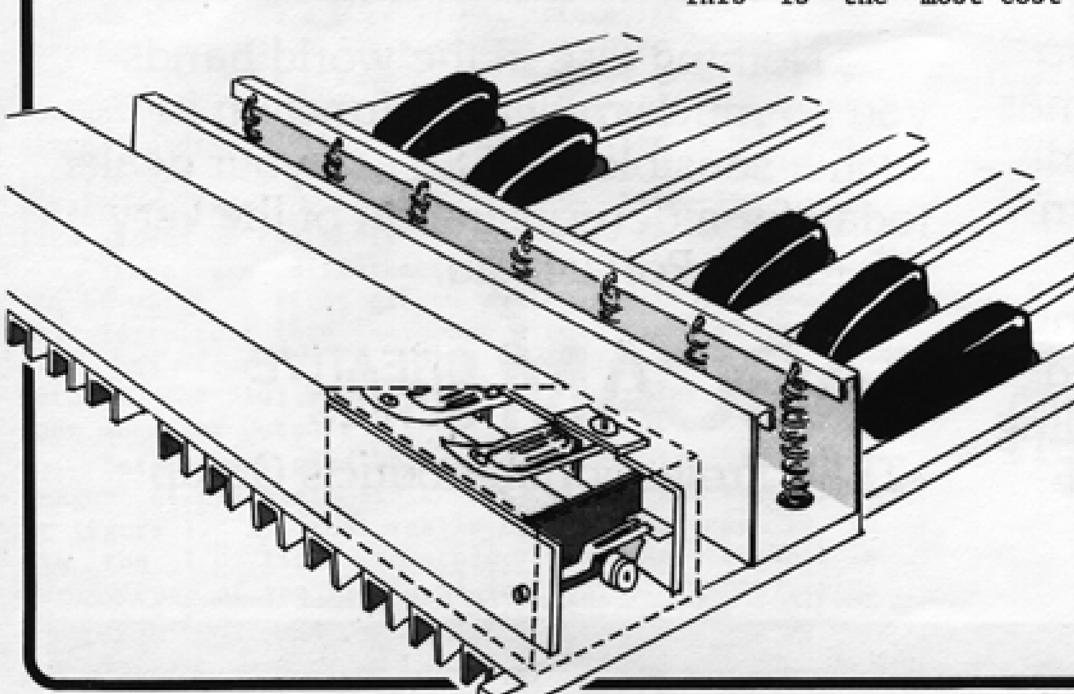


Figure 1: Switch opens when depressed and latches

method if using the pedals as a seperate controller for an individual exponential or linear DAC. (see figure 1)

Once inverters are installed giving positive logic, analog switches can be tied to each output, to switch in any scheme of resistors or pots thus obtaining selectable analog controllers. (see figure 2) No traditional "sample and hold" circuit is needed since each note of the pedal contains a latch and will stay fixed until a new one is pressed. The momentary contact common switch, to the right of all



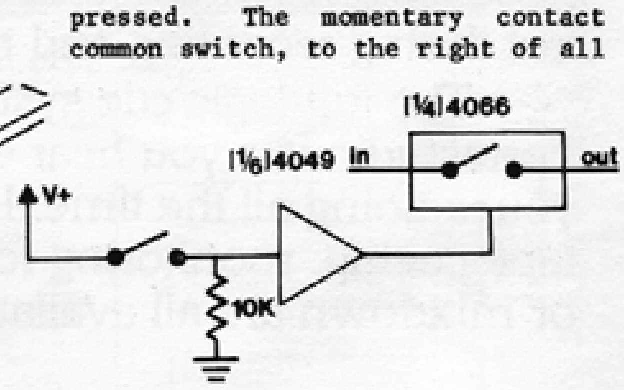
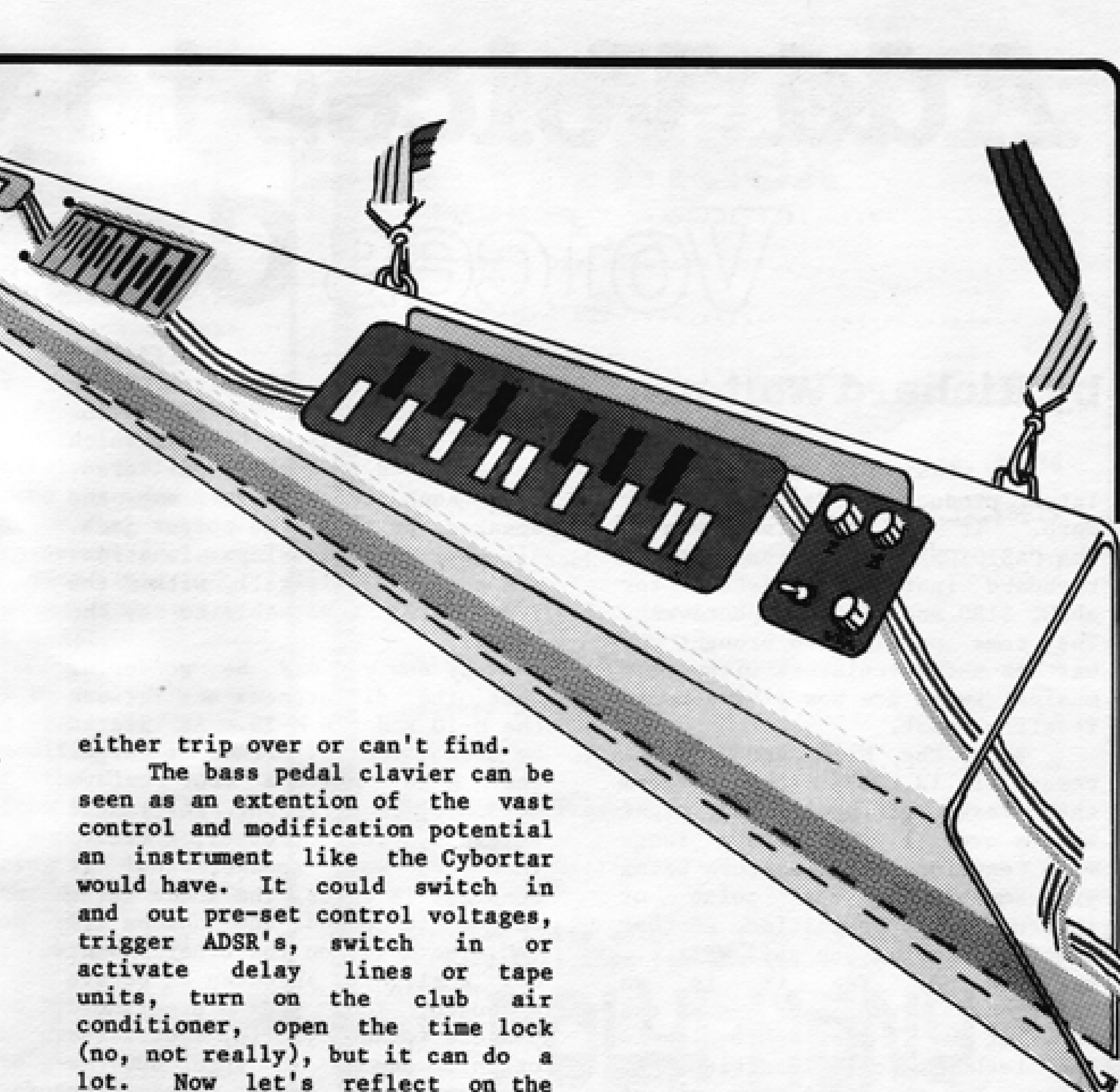
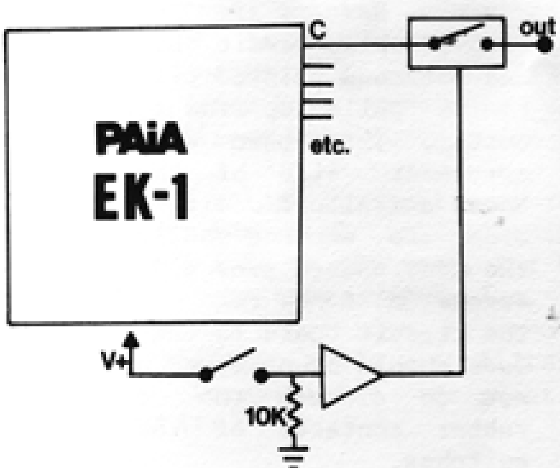


Figure 2: Analog switching for bass pedals



the switching banks, can be easily used for a note down "gate". Precision 100 ohm resistors in a string configuration, taped at each node and driven by a voltage regulator chip on top, will work for exponential oscillators. Those of you with left over -8 Paia divider boards can put part of them to use in this application.

Figure 3: Top octave bass synthesizer

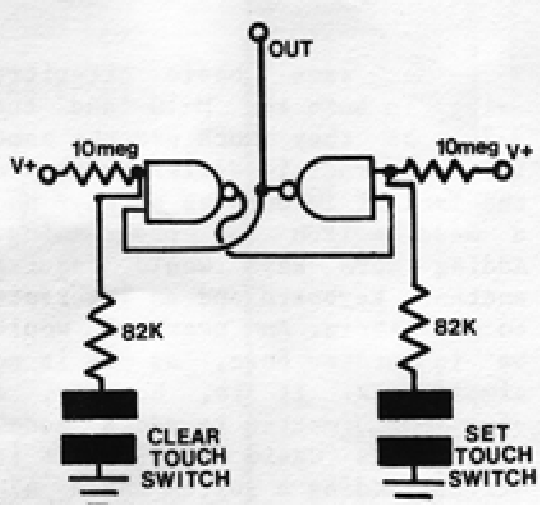


Another way to use the pedal arrangement with the 4066 analog switches, is to rig a switching matrix for the Paia EK-1 top octave experimenters's kit. (see figure 3) For bass pedal application you will want the master clock frequency to be quite low. For more information refer to POLYPHONY 2/76 or the off-prints that serve as instructions for the EK-1.

My last suggestion ties in with Kurt Shultz's Cybortar concept and offers some suggestions along those lines. If you use DC switching lines for your effects, (such as the Mutron Bi-Phase, or any effect switching electronically as explained in Craig Anderton's book "Electronic Projects for Musicians", project #15) then pedal switching mentioned above is a natural extension. Use a circuit like that in figure 2 to cut the effect in and out. With this scheme each note controls a different effect, eliminating the traditional clutter of floor switches that you either trip over or can't find.

The bass pedal clavier can be seen as an extention of the vast control and modification potential an instrument like the Cybortar would have. It could switch in and out pre-set control voltages, trigger ADSR's, switch in or activate delay lines or tape units, turn on the club air conditioner, open the time lock (no, not really), but it can do a lot. Now let's reflect on the special controller that goes hand in hand with the bass pedals.

Inherent in my concept of a Cybortar (as well as the originator's concept, as far as I can tell) are touch switches, specifically, set/re-set latches such as the one shown below. These simple touch controlled switches have the property of either momentary (touched) action or on/off alternate switch action. This lends itself to some new, powerful playing techniques. Each tone from a top octave chip is switched through its associated section of 4066 Quad Bi-lateral Switches on command from its respective touch switch latch.



The touch pads are laid out in association with the equally tempered keyboard protocol, and spaced accordingly. Each pad on the otherwise stripped circuit board has its copper "tinned" with a blob of solder for better tactile feedback. The master re-set bar on the back of the instrument acts as a common "off" for all switches. Here's where the novel playing style comes in. If you so select, you first touch a note to play it, then must hit that note again (or the master re-set bar) to end it. Hit four notes and they will all sustain, hands in the air, until you slap the re-set bar. How's that for impressive on-stage stuff!

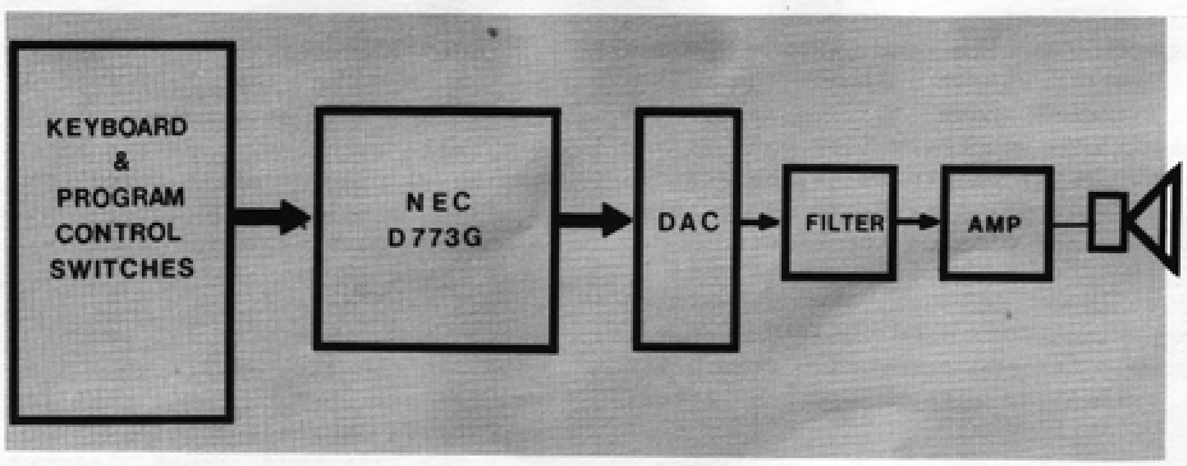
The left hand touch latches select octaves so that you could do a full keyboard run by simply re-playing the same notes and "jumping" octaves at the appropriate time. The left hand master bar is the final processing trigger. Since DC switching is used thoughout no power or audio cables need be brought into the instrument body. Extra waveshaping circuitry, top-octave chips, processors, etc., are all the individuals custom affair, and suitable circuits can be found in Application Notes and Cookbooks (see Polymart).Σ

Add Voices to your Voices Casio M-10

by Richard Wolton

The CASIOTONE M-10 is the latest product of Casio Computer Corp. It is the infant twin to the CASIOTONE M-20. These portable keyboard instruments retail for about $180 and $600, respectively. The same people who brought you watches and calculators with that musical twist are now giving music itself a twist.

Both the M-10 and M-20 are remarkable in that they perform the task of generating eight voices over a six octave range with envelope and waveform being programmable to the point of instrument approximation. Another electronic organ you say? Well.... yes and no. Yes it is an electronic organ, and a good one too. But no, it does not follow in the technological tradition of others. For the first time, all of the above mentioned features are contained on one IC, and the approach is somewhat different. The block diagram outlines the basic system:



As outlined in the block diagram, we see that the process is quite simple and almost entirely digital. The keyboard and program controls are all entered into the 773 by way of a scanned matrix, not unlike the scanned matrix commonly found in computer typewriters. This matrix operates in two modes - 'play' and 'program'. The 'play' mode generates up to eight voices as a function of the depressed keys. In the 'program' mode, the last key depressed determines the instrument to be synthesized. The AGO keyboard is used for preset selection! This, currently, permits up to 29 different instruments from which to chose and program into four presets. The output is straight binary, which is converted to analog, filtered, and sent to a power amp and speaker, or to a line output jack. All wave and envelope functions are performed digitally within the 773. Quite remarkable to say the least.

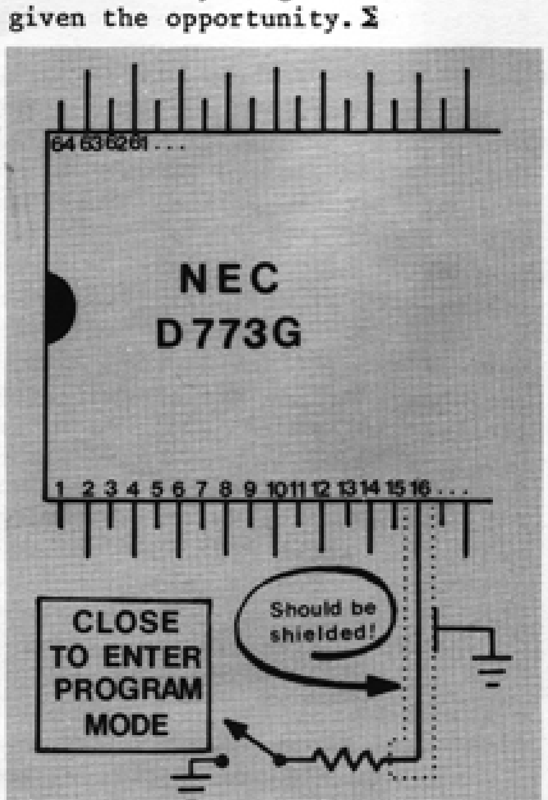
By now you may be wondering what the differences are between the M-10 and the M-20. As stated by Casio, the difference is large: the M-20 has a four octave keyboard, 29 different instrument voicings, four presets, a twelve inch two watt speaker and it consumes 18 watts. The whole thing weighs 15 pounds and runs on 117 VAC. The M-10, on the other hand, has a slightly shrunken 2½ octave keyboard, four pre- programmed presets (piano, violin, flute, and organ), and a 2½ inch 300 mW speaker. Total power consumption is 2.8 watts from five "C" cells or an external power supply. The purchase price is explained by these differences. But don't let them fool you.

The same basic circuitry exists in both the M-10 and the M-20, as they both use the same IC. The crucial differences are the lack of 1½ octaves of keys and a mode switch for programming. Adding more keys would require another keyboard and an interface to the matrix. An overhaul would be in order here, as it is no simple task. It is, however, a very simple matter to add a "mode" switch, as Casio simply left it off. By adding a switch and a 47K ohm resistor, the M-10 takes on the feature of 20 different instruments with four presets - only nine less than the M-20!

Where did these nine go? They would be the voices corresponding to nine of the missing 1½ octaves of keys. Even with just twenty instruments, this becomes a fascinating instrument.

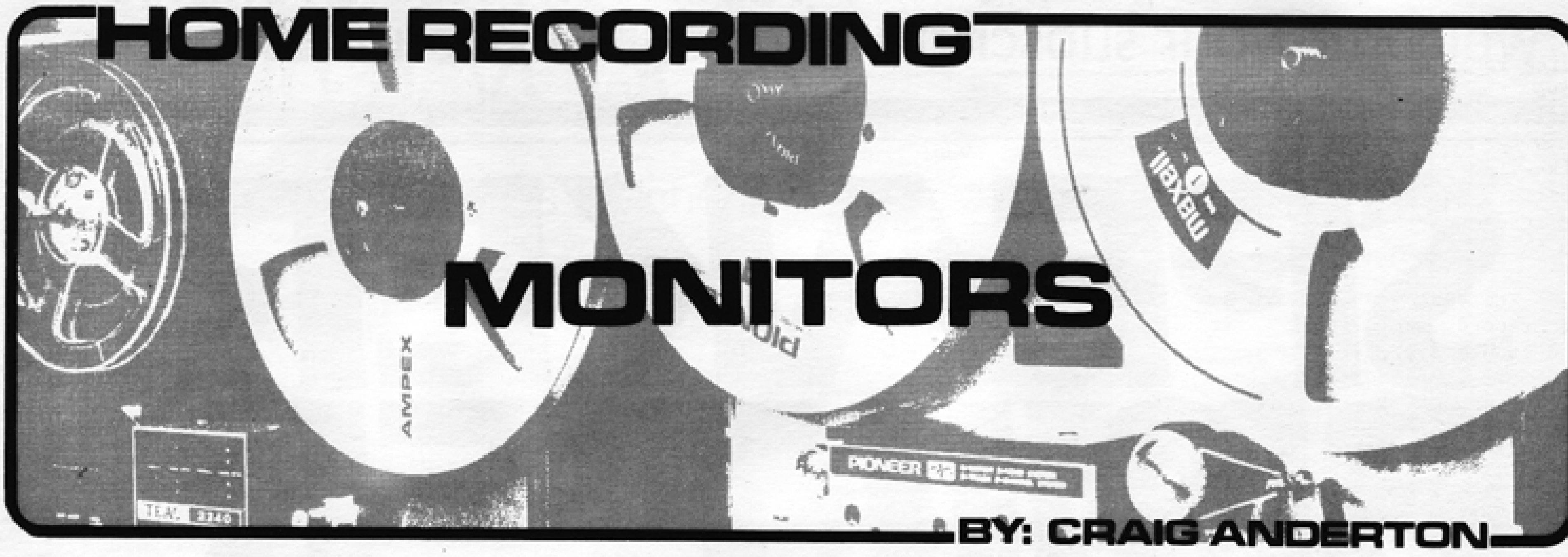
It is, however, a modification which requires some care. Within the circuitry are some static sensitive devices which should be protected from static discharge. In order to open the M-10, the five screws on the rear and the three control knobs are removed. Pressure is then applied along the encircling seam. This removes the top plastic cover. Next, six screws are removed which permit the keyboard and attached printed circuit board to be pulled up from the plastic bottom. This then exposes the component side of the circuit board and allows modification. In order to examine the traces (and the many unused pins of the 773), screws must be removed that hold the circuit board to the keyboard. Care should be observed here so as not to disturb the conductive rubber contacts of the keyboard switches.

The true joy of the M-10 is it's portability. Never before has it been so easy to race down the highway while playing a polyphonic keyboard as the world flashes by. A 'moving' experience! And although cows can't applaud, they make a very eager audience when given the opportunity. Σ



HOME RECORDING

MONITORS



BY: CRAIG ANDERTON

No matter how you look at it, transducers have nowhere near the fidelity - or the sophistication - of the electronic equipment to which they mate. Consider that all this time after the invention of the phonograph record, we're still dragging a piece of rock (albeit diamond) through thousands of feet of plastic... pretty primitive! In addition, all of the tonearm's strange mechanical resonances, coupled with the average cartridge's unusual impedance characteristics, make for pretty unpredictable response characteristics.

Tape heads aren't all that great either. Since they only react to changes in energy, square waves are virtually impossible to accurately reproduce. Besides, the levels a tape head puts out are so small you have to have special, low noise preamps just to get the signal up to a usable level.

Then, there are the human ears listening to the equipment, which are far from ideal. You'd be surprised how few people can hear the horizontal oscillator of a TV set, which produces an audible tone in the 15 KHz range. People who have been subjected to high distortion, high intensity sounds (such as those found in an industrial environment) may not be able to hear much past 5 KHz.

This brings us to the transducer that we'll be examining in this column, the studio monitor. We may as well start by saying that speakers are not very good transducers either. They are highly inefficient, have more peaks and dips than the Rocky Mountains, and often use crude crossover devices to drive the various speaker elements assigned to different frequency ranges. While bi-amping and tri-amping techniques help to create a better speaker sound, they also help create a lower standard of living for those who wish to invest in such a system.

Unfortunately, our studios are virtually dependent on monitors; they are the gateway through which all our subjective evaluations concerning the fidelity of a tape, or the sound of an instrument, must pass. As a result, big-time studios spend vast quantities of bucks on monitor speakers that try to achieve (and often come very close to) the ideal of a flat, and accurate, frequency response.

For those of us with budget studios, after spending $1000 or so on a tape deck it seems like adding insult to injury to have to lay out hundreds of extra dollars for monitors. If you have a decent stereo system, one alternative is to do "time-sharing" with your hi-fi speakers; however, after about the 15th time that you lug your speakers back and forth between hi-fi and studio you might question the wisdom of this approach. Since this column is mainly about how to get a good sound for the least bucks, I figured that it might be time to talk about my personal feelings towards studio monitors.

BASIC CONCESSIONS

First, we're assuming that you do not have lots of money to spend on monitors. If you do, you might as well skip to the next feature... this article is for those who have about $100 left over after scrimping and saving to get the recorder paid for and the mixer built.

Second, we're going to have to agree among ourselves that all speakers are imperfect. Therefore, whether you spend $100, $300, or even $500 on a speaker, you're still going to get something that isn't all that efficient and has peaks and valleys in the response. One speaker may sound a little sweeter than another, one may sound a little fuller; one may sound a little more harsh. But, you're assured that none of them will have a consistency of response that's even close to that of your power amp.

Remember, though, that the purpose of a studio monitor is to be flat; and, we've just agreed that any speaker we can afford is not going to be flat. Talk about a no-win situation! However, there is an answer... provided that we look at monitors in a little different manner.

THE SOLUTION

The following rule sums up a solution to our problem: You can use any speaker you want to for a monitor, regardless of cost, provided that you learn all about it's sound qualities. There! That takes the burden of buying a big monitor speaker off our shoulders. But, we've added a new burden, which is to learn about the speaker.

LEARNING THE SOUND OF A MONITOR

Now the hard part begins. You've chosen a speaker, either because you can afford it or because it sounds better than any of the other types in it's price range. Now you have to identify exactly in what ways the speaker deviates from flat response. Is it deficient in the treble region? Bass region? Is there a wicked midrange peak that makes your voice sound like it's coming through a megaphone? For a cheapo monitor to do the job for you, it must be quantified as accurately as possible... let's talk about how to do this.

The best way to learn a speaker is to take a tape or record and listen to it over a variety of stereo and studio systems. Make sure that the various tone controls are consistently set for the same settings (flat response may not sound best, but remember we're comparing here). Ideally, you should listen to the tape or record over a really superb system (such as an expensive studio system, or maybe the local audiophile's "golden ear" system). Don't listen to the music; listen to the relative levels of the instruments, how hard the cymbals crash through on the high end, how even the bass response is, and so forth. If at all possible, listen to the same program material over a variety of top-class systems. You may be surprised at the variations that you encounter as you do these listening tests, even among supposedly "flat" response systems. While much of this variation is due to differences between speakers, room resonances also enter into the equation. We'll touch on this subject a little more later on in the column.

If you can't listen over some really fine systems, then listen over as many systems as possible and take an "average" to figure out what the record was supposed to sound like when it came out of the studio.

Now that you know what the program material should sound like, play it over the speakers you've chosen as monitors. Sounds pretty bad, doesn't it? But, you can still hear the music and while the tonal quality may not be exceptional, at least you will hear the various instruments with some semblance of accuracy with regards to balance, stereo imaging, etc. Now, note as many differences as you can in the response from a subjective standpoint. Where does the high end poop out? The low end? If the vocals and guitars sound more prominent on your setup, you know that either the midrange is being boosted or the bass and treble response is so bad that it makes the midrange appear boosted by comparison.

After learning the peculiarities of your speaker, take into account that the sound you're hearing is still being shaped by the room in which the monitor is located. If you have a graphic equalizer with enough bands, you might be able to smooth out some of the speaker's response anomalies and some of the room resonances as well.

THE REWARDS OF KNOWING YOUR SPEAKER

Now that you know your speaker, you can feel much more confident about your mixes. For example, let's suppose that your speaker has poor bass response. If you play a drum sound through these speakers, you'll probably turn the kick drum way up in order for it to sound good. But, if you know your speakers are deficient in bass response, you can then compensate by trimming back the kick drum a little. While it may sound bass-shy on your speakers, when you play the same material over higher quality systems the kick drum will fall right into place if you've compensated by the correct amount.

This may seem like an awkward way to mix; however, it has been my experience that you really have to learn the characteristics of <u>any</u> speakers, even the good ones. If you can't afford a good speaker, don't worry about it... learn about the one you've got and you'll still be able to put out a pretty good mix.

MY MONITORING SYSTEM

Since I fall under the category of "not being able to afford the monitors after putting the studio together", I settled on a pair of Auratone speakers, which set me back $70 (that's for both speakers, mind you). Auratones are <u>not</u> the best little speakers you can buy; I've heard some tiny hi-fi speakers that gave amazing response. Rather, the Auratones can take a lot of power and still not blow up (unless you're really crazed for volume), and best of all, just about all the big studios use them as reference speakers in order to get an idea of what something will sound like when it hits the real world of tinny transistor radios, mono car radios, and $50 mass-market stereos. In my particular case, having worked at a number of studios that used Auratones, I got to hear material over both excellent, high-ticket speakers and the Auratones. Over the years, I've managed to learn Auratones so well that they've become my reference speaker. So when I needed to get speakers for the studio, that's the kind I purchased.

An additional advantage of mixing on inferior speakers is if you can make things sound good on them, just wait until you hear the sound over really good equipment. (continued .... page 20)

INDUSTRY REPORT

'Tell Them You Saw It In Polyphony'

New Hybrid

E-mu Systems has announced the availability of the AUDITY computer-controlled polyphonic synthesizer system. The AUDITY system consists of a central computer with dual floppy disks, a remote programming console, a 16 channel polyphonic keyboard/sequencer with its own disk for sequence storage, and up to 16 voice cards, each containing a complete analog synthesizer with associated circuitry to allow total computer control of all voice-definition parameters.

Each AUDITY voice card contains two VCOs with PWM and hard sync, a 24 dB/octave lowpass filter, a 24 dB/octave highpass filter, a multimode resonant filter, a VCA with linear and exponential response, four ADSR transient generators with delay, a variable spectrum noise source, an LFO with reset, and four independent modulation busses, including one capable of delayed modulation.

Unlike other programmable synthesizers, the AUDITY's central computer has independent control of each voice card, allowing the assignment of different sounds to different synthesizer channels simultaneously. In addition, entire 16 voice orchestrations can be stored in memory as "system presets". Both sounds and orchestrations are stored on floppy disk, allowing the user to create a library of essentially unlimited size.

Although the AUDITY can be controlled by virtually any 1 volt/octave controller, it was designed especially for use with the E-mu 4060 polyphonic keyboard. With the 4060's built-in 16 channel memory sequencer the composer or arranger can create multitrack compositions and then experiment with orchestrations in real time--creating and modifying timbres while his or her piece is actually playing.

A complete 16 voice AUDITY system costs $69,200.00. For more information contact E-mu Systems, Inc., 417 Broadway, Santa Cruz, CA 95060

Prophet-Link

External control of the PROPHET 5 synthesizer is now available in a control voltage and gate interface device from Ampersand.

The LINK is installed as a modification into the PROPHET, and terminates at a remote box having phone jacks for each of five voices' CV and gate inputs and outputs, and five independent voice select switches are provided. A high quality connector scheme provides reliable quick-connect setup; PROPHET circuitry is isolated and buffered for protection.

Typical controllers used with the LINK are the E-mu Systems Microprocessor Keyboard, the Roland Microprocessor, and guitar syntheziers having 1V/octave outputs. Price: $990 installed. Write to: Ampersand, 9548 E. Zayante Rd., Felton, CA 95018

POLY·PICK·UP

The HEAR P2V is a hexaphonic (six channel) pitch to voltage converter enabling the guitarist to control their choice of polyphonic synthesizer(s) from their own guitar. The tiggering speed, tracking accuracy and dynamic response of the P2V marks the end of the wait for the ultimate guitar to polyphonic synthesizer interface.

The front of this system is the patented HEAR hexaphonic pick-up/bridge assembly which easily replaces the tune-a-matic bridge on any guitar. The P2V provides the basic pitch and gate voltages hexaphonically as well as 23 additional outputs (per string).

The other outputs available for each string are: five extracted guitar voices, six extracted fuzz voices, two types of triggers, three (other) types of gates, four types of envelope followers, a speed follower, a timbre voltage and a period voltage.

Available factory direct only-$2500.00. Write or call HEAR, Inc., 1122 University Ave., Berkeley, CA 94702 or 415-848-6262

improved Drummer

PAIA has announced a low power redesign of its popular Programmable Drum Set which extends the time that rhythm patterns may be stored to over one year.

Unchanged is the PDS's simple programming system which allows even first time users to structure the unit's bass, tom, snare, wood-block and clave sounds into any rhythm in any time signature.

The improved design also retains the original's versatile memory organization which provides simultaneous storage of two separate rhythm patterns each with its own bridge rhythm. Bridges are activated either from a control panel touch plate or optional foot switch and are automatically synchronized to the main rhythm.

Redesigned memory circuitry allows a "save" mode of operation that draws only slightly more power from the unit's "AA" size batteries than they loose normally with no load at all. Battery life during normal operation is also extended to several hundred hours.

The Programmable Drum Set (Cat. #3750) is available directly from PAIA Electronics, Inc., 1020 W. Wilshire Blvd., Oklahoma City, OK 73116 at a price of $99.95 in kit form and $159.95 assembled.

Space Available

CREATIVE SPACE, a composers pre-production workshop will open in August, 1980. The facility offers seven self-operated recording suites and a real-time copy room designed specifically for the musician/composer who is developing material for future record production.

Each suite is complete with all the necessary equipment for the production of a song demo, including the 4 track TEAC 144 PORTASTUDIO with Dolby, Yamaha piano, programmable rhythm machine, reverb unit, Aurotone speaker monitors, stereo amplifier, tuning device, studio quality 2 track cassette recorder, microphones and headphones, SPECIAL FEATURES include; professional acoustic design and treatment, independent climate control and electrically filtered air in every suite.

All rooms are designed for acoustic instruments and vocals. Any electric instruments can be directly connected to the input of the Portastudio, thus eliminating the need to transport heavy speakers from the gig or home.

Industry accounts are welcome for established composers, labels, publishers and producers who want to evaluate songs before purchasing major studio time. A Creative Space suite is available for $12.50 an hour. For the struggling musician/composer we offer a 20% discount making suites available for a low $10.00 an hour.

Room and equipment demonstrations will be held twice weekly or by appointment. For further information contact Janis Thompson (213) 384-3704.

NEW·MINI·AMP

Polyfusion announces MS-8 "MICRO-STACK" a miniature speaker/amplifier which can be used with guitar, bass, synthesizer, microphone or any other electric instrument. MS-8 boasts big sound and features found only in much larger amps. For example; Distortion circuit, Tone Control with active circuitry, gives 12dB treble boost or cut, also, high quality 3" speaker, 500 milliwatts output. Unit operates on two 9 volt batteries. Suggested price: $89.95. Distributed exclusively by BKL International Distributing, Inc., P.O.Box 248, Neptune, NJ 07753

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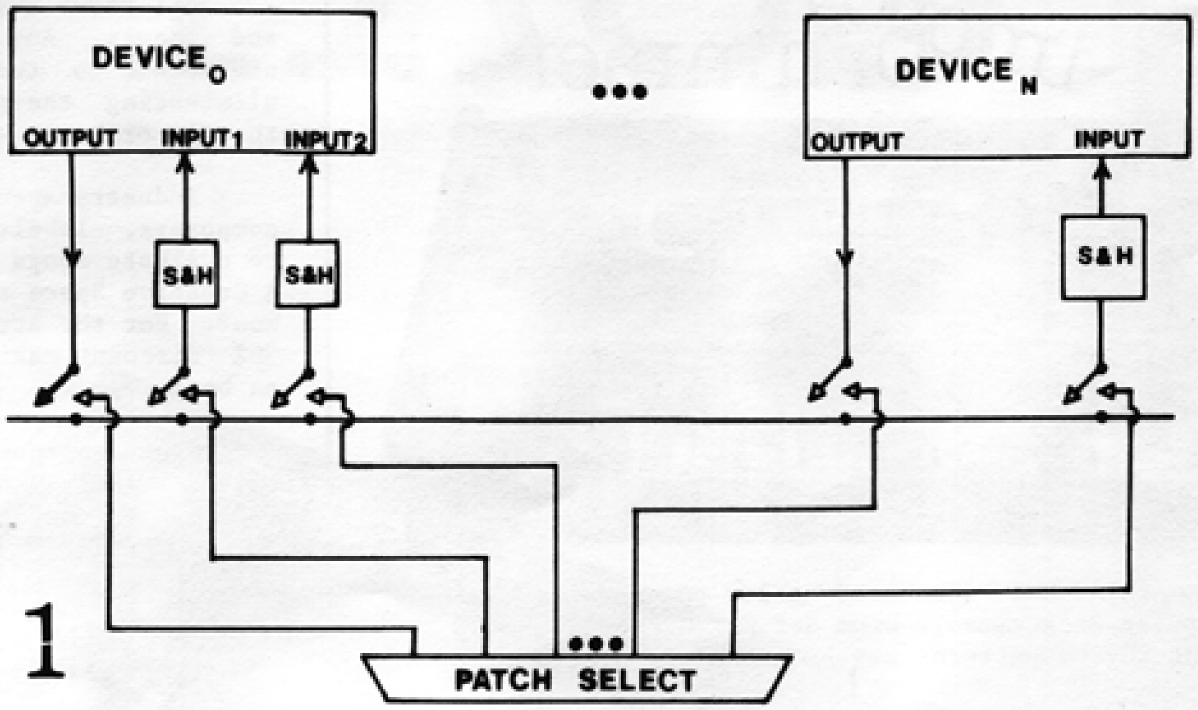
A Cordless Programmable Patch-Bay

by Colin Johnson

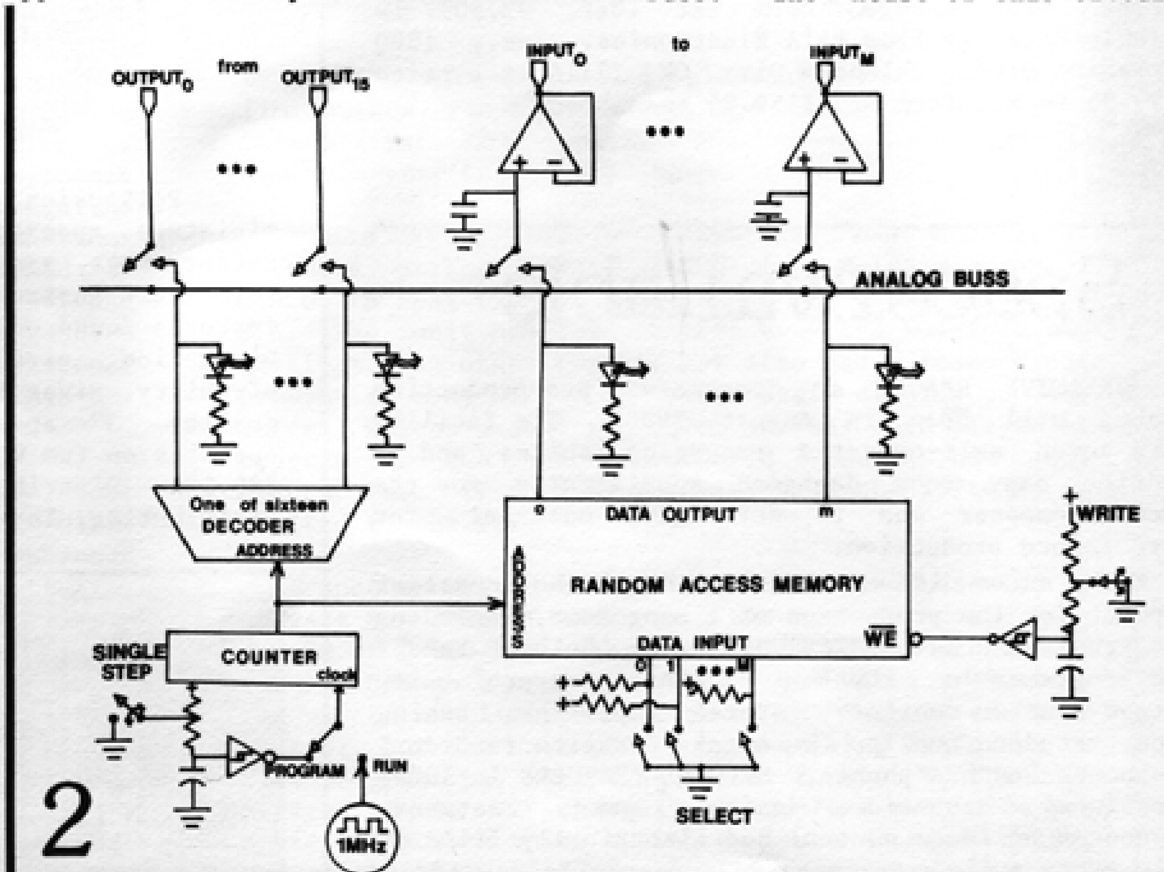
One way to minimize hardware interconnections in an analog synthesizer is to time-share a common buss. Each input and output is attached to this buss via an analog switch. In this way, any output may be connected to any input by simultaneously engaging their respective analog switches (see figure 1). The buss is then time-shared so that more than one such connection can be made. Each desired individual patch is given a time slot on the buss. During this time, the analog switches for an output and an input are engaged, and the sample-capacitor on the input is charged. Then when the switches are dis-engaged, the capacitor stores the voltage level until its next turn to use the buss. Typically, a time-slot is one micro-second out of every thirty two. This would allow thirty two patches to be maintained at full audio bandwidth (e.g. 0 - 16kHz). If frequency bandwidth is restricted to the range of control voltage (e.g. 0 - 1000 Hz), then five hundred patches can be maintained. Clearly this is enough for most musical purposes.



1

There are many methods by which the patch-bay may be programmed. In figure 2, we see a configuration that allots each output an equal time-slot on the buss, one micro-second every sixteen. A random access memory is used to remember which inputs and outputs are to be connected. LED's indicate when an analog switch is engaged. When running, the LED's flash so fast as to appear continuously lit. But when switched to the program mode, you may SINGLE STEP through the outputs while WRITing the inputs to be engaged into memory using the SELECT switches and LED indicators. The start-up procedure is : 1) switch to PROGRAM 2) SINGLE STEP until the desired LED is lit 3) set the SELECT switches to the desired input and observe its LED light up 4) toggle WRITE to enter the patch into memory 5) repeat for all outputs.

A more flexible system is shown in figure 3. Here more economical use is made of the buss. The heart of this circuit



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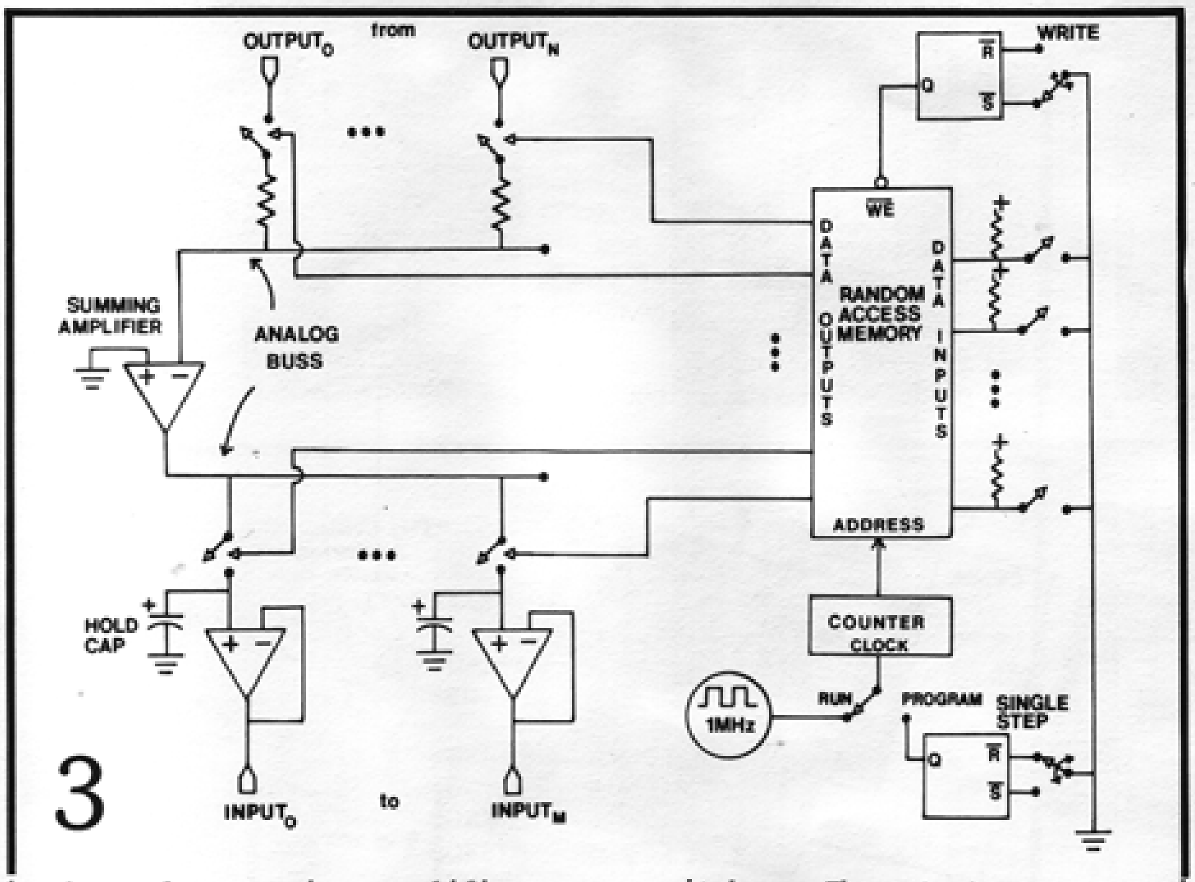

is the analog summing amplifier. It allows any sum of the outputs to be fed to any number of inputs. The memory is expanded to make room for a rememberance of the output sums. The system clock and address generators sequentially access memory. When the clock advances the memory engages some combination of the analog switches. The outputs are summed at the summing amplifier and fed to any number of inputs. The beast is programmed by SINGLE STEPPING the system clock and WRITing the various output-sum-to-inputs-combinations into memory via the SELECT switches.Σ

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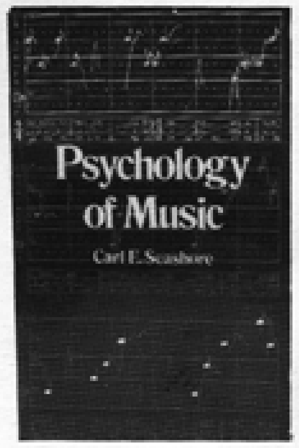

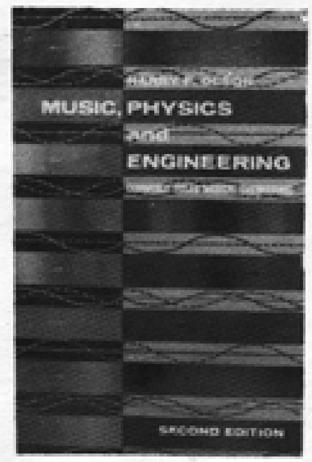

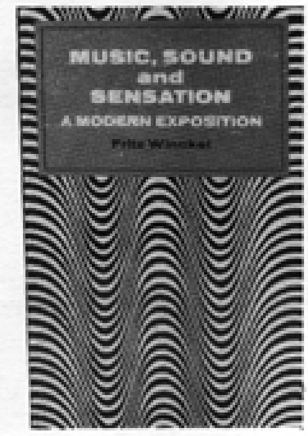

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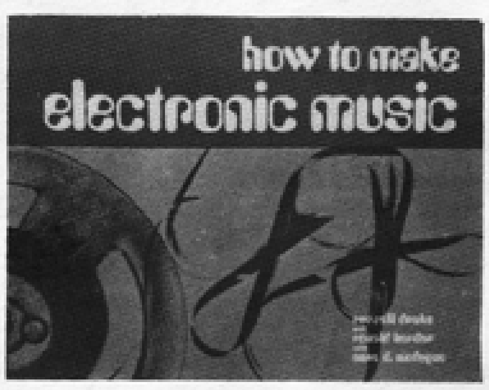

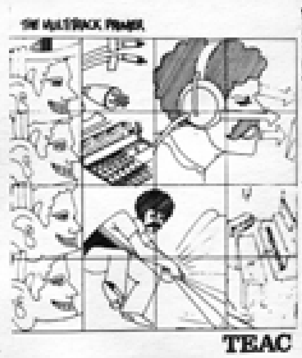

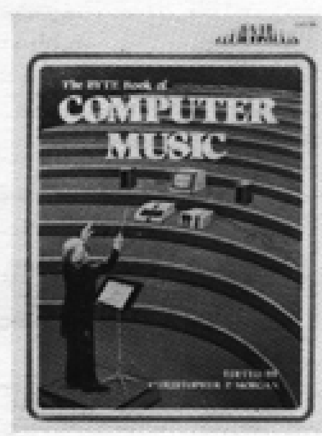

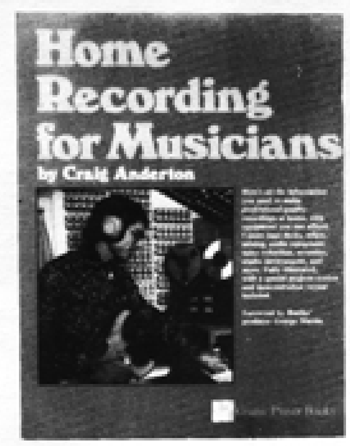

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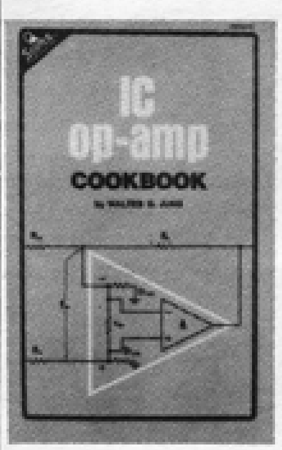

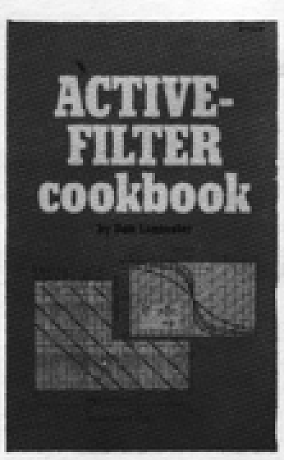

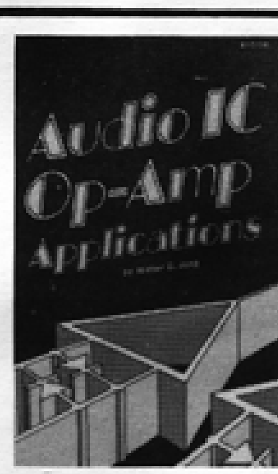

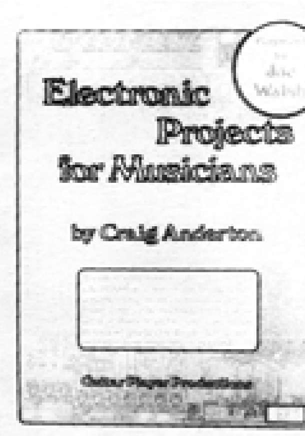

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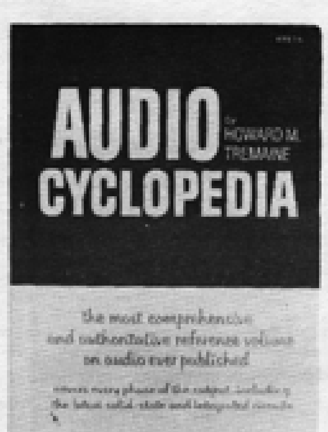

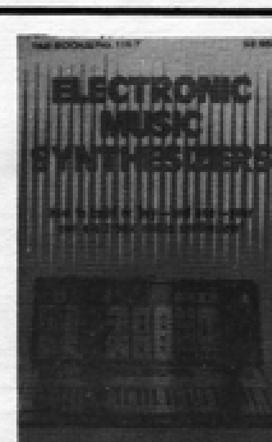

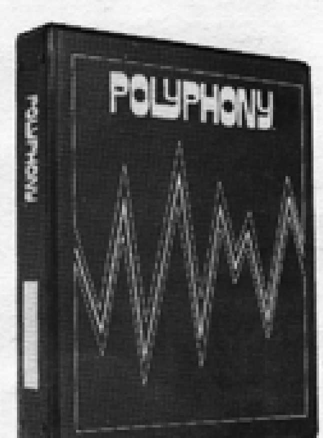

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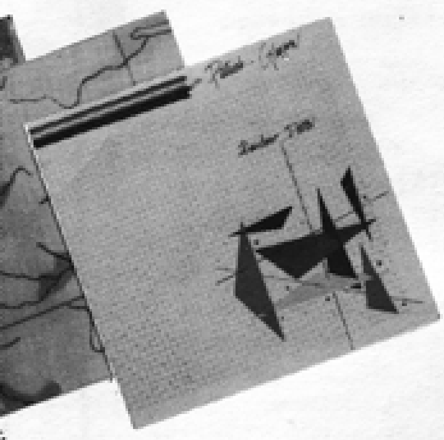

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#0401: July/August 78: analog delay lines (theory and projects), composing for electronic music, note to frequency (and vice versa) conversions, build a trigger delay, software for computer composition, low cost VCO circuit, patches.

#0402: September/October 78: electronic music notation, notes on the recording of "Cords" by Larry Fast, sequencer software- part one, rhythmic control of analog sequencers, touch switch projects, modular vocoder techniques, PET as a music controller, patches.

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#0502 July/August 79: hex VCA/mixer project, electronic music schools and studios, modify the Oberheim Expander Module, profile of Ernest Garthwaite, budget microphones, digitizer projects and software, bar graph ICs.

#0503 September/October 79: SOLD OUT

#0504 November/December 79: SOLD OUT

#0505 January/February 80: Joseph Byrd, Mort Garson, Larry Fast on 'Games', composing for 'live plus tape', using the CA3280, recording vocals, ADSR circuits.

#0506 March/April 80: Computers in Music: real time audio processing- hardware, Powell sequencer system, Max Mathews, advanced STG software, PortaStudio, phase modulation, Volume 5 index.

#0601 May/June 80: Gary Numan, harmonizing devices, real time audio processing- software, digital delay project, writing documentation, Richard Hayman.

MONITORS

continued from page 13

THE VALUE OF HEADPHONES

Most people say that you should never mix on headphones; I agree. But that doesn't mean that headphones can't be part of the mixing process.

I've always felt that headphones are more cost effective than speakers in the sense that $100 headphones invariably had better subjective fidelity than $100 speakers. So, after getting the Auratones I treated myself to a pretty accurate pair of headphones (the LV-10s that Radio Shack sells - don't worry, they're manufactured by Koss). Since I do a lot of listening to records on headphones, as I listen to my studio tapes on the headphones I can compare the overall balance with that found on a variety of albums. After a while, you start mixing for an "average" balance that will hopefully sound right on a variety of systems, even when listened to by a variety of sets of ears under widely varying conditions.

CONCLUSION

Sure, it would be nice to be able to spend thousands of dollars on the best monitors made... but even if you can't, by taking the time to really learn a set of speakers you will at least be able to produce tapes that are relatively accurate. They may not sound fantastic over your own system, but they will at least sound pretty good on all systems, and excellent on some. In this world of imperfect ears and transducers, you really can't ask for more.

* * * *

I regret to say that this is my last Home Recording column; my time is getting crunched to the point where I've had to let some projects go. I'd like to leave you with a thought: If you enjoy your music, that's all that matters. Accept criticism graciously, but don't let anyone tell you what you "should" or "should not" play... you are unique, and have your own contribution to make to this planet.Σ

PINK DRUMS

Scott Lee
PAIA Tech Services

Pink Drums can be an interesting alternative to your standard rythym box, or for that matter , a drummer and his drum set. You can obtain a wide variety of percussive sounds with this set-up that range from a true "trap set" type of drums to very "abstact" type sythesized sounds, or both.

Basically, "Pink Drums" is a patch between a P4700J and a P5700 drum synthesizer. They are "pink" in the sense that they are almost random, but like pink noise, not entirely so, having some (in this case controllable)correlation. The computer determines which channel will be triggered and how often, all with respect to a basic four-part harmony. Those of you already familiar with John Simonton's program "Pink Tunes" (see July/August '78 Polyphony) probably already have a good idea of what is going on. For those who don't, read on!

"Pink Tunes" is a program written for the 8700 computer/ controller that composes four part harmonies by randomly picking musically "valid" candidate notes from a table. Since this is a "polyphonic" type approach, a D/A and Quash are needed for the basic 4 part configuration. If you wish to take a minimum investment approach you can get by with just the 8700/80/81 configuration (approx. $220) mated to a P5700 "Drum" package. In this case you would enter your candidate note values in hex from the computer keypad (see above mentioned article.)

The actual drum sound generation is accomplished with the P5700 . This package is a cost effective professional drum synthesizer with a convenient array of user controls and interface jacks. The trigger outputs of the Quash are first patched into the trigger inputs on each drum channel. This will require some special patch cords that terminate in pin jacks for the Quash outs and 1/4 in. phone plugs on the Drum trigger inputs. The same type patch cords are used from the Quash output (control voltages) to the c.v. inputs of the Drum. Then adjust the drum controls on the panels to achieve the desired sounds.

The sounds I tuned the P5700 to consisted of a bass drum snare, a synth-type drum with a short upward glide, a percussive bell sound using a higher pitched sine wave, and a stunning envelope modulated filter sound on the last channel. The Drum's capability to accurately reproduce the traditional drum sounds of a working professionals drum kit, or even symphonic kettle drums make this system much more than a plaything.

To load "Pink Tunes" into the 8700 start the program running at 0003 and select some notes on the AGO keyboard (if you have the P4700J) or enter your candidate notes in hex on the computer. The system should now start generating the "pink" rhythms.

Further information can be found in the Friendly Stories about Computer/Synthesizers manual. The manual for the 5700 Drum gives an explanation of the controls and operation of that unit. Σ

1980 PACS computer music festival

by John Lazzaro

On August 23, 1980, the Philadelphia Area Computer Society (PACS) held their annual Computer Music Festival at the Civic Center in Philadelphia, PA in conjunction with the Personal Computer and Small Business Computer Show. PACS is a non-profit organization, and all the planning and preparation of the festival was done by volunteers. A high quality sound system and two large screen televisions were available for the presentations, and for certain speakers 200 people filled the hall. An electronic music concert featuring many of the speakers and synthesizers of the festival was held Saturday night.

The presentations could be broken down into three categories: software to enable microcomputer owners, with the help of a D/A, to program and play back musical compositions in a sequencer-type fashion; self-contained computer/ keyboard systems that enabled the user to build accurate waveform compilations of acoustic instruments, assign them to a polyphonic keyboard, and multi-track up to 8 voices through the keyboard out onto disks; systems that fit into neither of these two categories. The first category, aimed at the computer hobbyist rather than the musician, had prices ranging from over $1000 to only several hundred dollars; the second category, aimed at educational markets and large studios, carried prices into the $30,000 range.

The hobbyist-oriented D/A instruments were some of the least expensive items to be shown. Cliff Ashcraft and Frank Covitz presented their enhanced musical software for a 6502 processor. The software, designed to run on a KIM-1 or AIM 65 and a D/A, played programmed pieces of up to four voices in real time, and featured the abilities of building up waveforms with Fourier analysis. Like every system on display, the tones were produced completely digitally, with the exception of the analog conversion of the D/A, obviously. The notes were programmed in through the terminal keyboard, thus aiming for the hobbyist market rather than the musician's market. The voices produced by the system sounded pleasant, but had their limitations. The piano voice, for example, was more "piano-like" than piano. Then again, a non-musician would be very happy with "piano-like".

Mike Reily, of AB Computers, had a similar software package that played four voice programmed pieces in real time on a Commodore PET using an 8 bit D/A. The unit had less control over the waveform and envelope than other models, but also had a feature not present on many of the others: a staff display on the screen which showed the notes as they were played or programmed. The voices sounded okay, but not spectacular. Because of the graphics, the software is limited to the PET.

Another system along these lines was presented by Stewart Newfield of Newtech. His software for the TRS-80 or S-100 bus enables the playback of programmed tunes through a D/A, with 2 or 4 voices in real time. The voices are limited because of the lack of envelope control, but the unit offers many editing features to ease programming, including repeat functions. The unit offers up to 14 changes of waveform per voice per piece, with each voice having a seven octave range, and allows the use of non-equally tempered scales.

Mountain Music Systems presented a system of programming and playing back music. George Willibanks and Bill English presented this system, which runs on a 48K Apple, and can have up to 16 voices. A musical keyboard can be added, although the system is aimed primarily at the home/ educational market. The unit offers stereo outputs with a bandwidth of 13 Hz to 13 KHz. The software features an easy scoring medium, with on-screen staff and cursor.

The Cashead Musical System, presented by John Bondy, is a system which is designed for the S-100 bus. It plays back programmed sequences for 32, 16, or 10 channels, each of which can play 4 (optionally 16) stored waveforms per composition, with 256 amplitudes available for each voice. As the number of voices goes up, the frequency response goes down, but when John demonstrated the system with 32 channels, the frequency response sounded adequate. This unit uses a real time clock, and offers such features as attack/ decay control, waveform computation, echo, chorus, and vibrato. The number of voices available results in rich textures when voices are doubled and tripled. The highlight of the demonstration was when John, playing bass, teamed up with Dave Gastfriend, playing violin, with the computer to play an original piece, CINEMAJIG, by Dave Gastfriend. The synthesizer lines blended well with the violin to create an interesting sound.

Many of the systems mentioned previously share some similarity with the pioneering work in real time synthesis done by Hal Chamberlain. In fact, the system shown by Cliff Ashcraft and Frank Covitz began with Hal's foundation. However, Hal Chamberlain's talk centered not on real-time synthesis, but non-real time synthesis, or "computed" music, as he refers to it. As he explained in one of the best attended talks of the day, real time synthesis is limited in many ways due to machine time constraint. Basically, real time difficulties come from the non-linear interpolation of waves due to limited time (sampling theory and rate), and resulting in noise and distortion. In computed, non-real time synthesis, which involves the storing of the entire piece first, usually on disk, and playback after creation, the time constraint is eliminated. As a result, the "stop clicks" that occur in real time synthesis are gone and, in addition, computed systems offer higher quality sound, better sampling rate, better frequency response, and less noise. The only drawback is the amount of memory needed: large.

Hal's NOTRAN system, for example, has a 20 KHz sampling rate, 12 bit DAC resolution, rhythmic control that produces any rhythm representable by a fraction (Hal used 1/147th notes in a piece!), unlimited number of voices, unlimited piece length. The memory capability: 20 seconds of music per disk.

Hal played Fugue in D Minor by Bach, and the organ simulation was truely impressive. The unlimited number of voices allowed for an all-stops organ sound on the long sustain chords, and the lack of noise and clicks was remarkable. Hal was kept very busy replacing disks every 20 seconds!

That finished the "program and playback" portion of the talks. Each system has its own merits and ideal purpose.

In a change of pace, RCA representative Rebecca Mercuri presented a talk on different types of on-screen representation of musical composition. She reviewed the different types of conventional music writing (4-voice vocal harmonies, piano score, full chord stems, chord + melody scoring), and contrasted

them with different ways of computer graphic scoring. Her basic message was to customize the system to its purpose, instead of devising a cure-all system. Additional concepts discussed were the necessity of eliminating accidental erasure possibilities during scoring, and the preference of having turnable "pages" of staves rather than the moving staves often used.

The next category of instruments displayed were the large synthesizer systems, represented by Fairlight and Crumar. Although each systems has its own unique personality, they have several things in common: the ability to create waveforms highly accurately and to compile the waveforms as harmonics of an instrument (thus creating very accurate synthesis of conventional instruments) the ability to assign these voices to a polyphonic musical keyboard; the ability to multi-track up to 8 voices in a digital sequencer; and the ability to store and call up presets to individual voices of the keyboard.

The Fairlight CMI operates 100% in real time, and constructs its waveforms in two ways: by drawing the harmonics of a waveform with a light pen on a video screen, or by playing any sound into a microphone connected to the unit. The unit then analyzes the sound, displays its frequency graph on the screen (for possible light pen alteration), and the sound can then be assigned to any or all of the 8 voices on the polyphonic keyboard. Preset sounds can be stored and called up to the keyboard at any time using the numerical keypad near the keyboard. As an example, a violinist played a tone into the Fairlight during the demonstration; after assignment, the violin sound was accurately reproduced by playing any note on the keyboard. The sound was assigned to all 8 voices, and fairly realistic polyphonic strings were available. I say 'fairly' only because later on in the demonstration, even more accurate simulation was called up from the presets. In addition, accurate simulation of clarinet, sax, and drums were auditioned. Their sound is what you would expect for $30,000. Options include using 8 keyboards for 8 monophonic synthesizers. A tape was played demonstrating real time playback of a piece constructed using the 8 channel sequencer ... impressive. All in all, a very sophisticated instrument.

The Crumar General Development System is equally as sophisticated. As Kevin Duren of Crumar explained, the system lacks the microphone input/ analyzer that the Fairlight has, but Crumar has concentrated instead on constructing a waveform through other means, and on a greater emphasis on user performance features. 32 digital oscillators are available for waveform construction, and pressure sensitive keyboard allows for increased musical expression. The string presets on the machine are outstanding, and their sequencer demo tape was just as convincing as the Fairlight's.

Both these units, while out of the range of the normal home or even professional synthesist, express the state of the art of computer based music synthesis.

The final system presented, the Alpha Syntauri, was the only system presented that aimed its target market at the average musician. It is a polyphonic keyboard interfaced to a 16 voice computer system. It works with the Apple II system in real time, and offers programmable presets, ADSR control on each

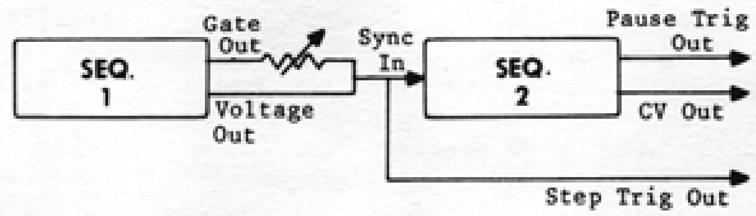
(continued .... page 29)

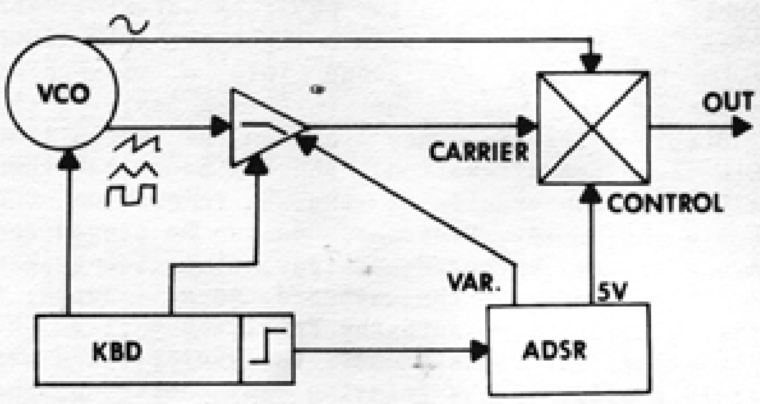
PATCHES:

Creating Spaces In Sequences

This is one of those "trust me" patches. Sequencer 1 will selectively step sequencer 2. First, feed just the gate of #1 out through an attenuator and cut back on its level until the second sequencer doesn't step. Then adjust the stages on sequencer 1 as follows: set the voltage low for notes you want a step on; set high for where you want pauses. The attenuator acts as a sensitivity control. As an interesting side note, sequencer #2 outputs a gate right before the step it pauses on; this could be used for additional triggering of events.

Chris Meyer
Hamilton, OH





Round Sound

This patch takes advantage of the VCA and frequency doubling capabilities of the balanced modulator. Using one VCO, the sawtooth, triangle, and/or pulse can be mixed for a complex waveform. The sine adds a very pleasing "roundness" to the signal with some extra harmonics. It's a very thick sounding patch for those of us with only one VCO.

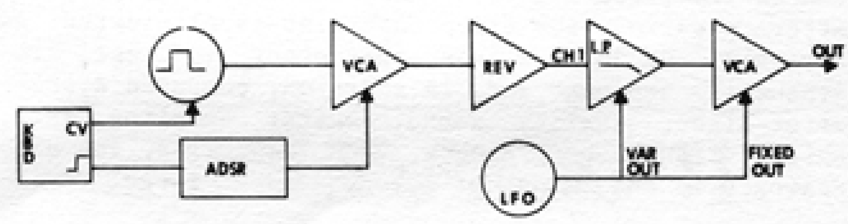
Chris Meyer
Terre Haute, IN

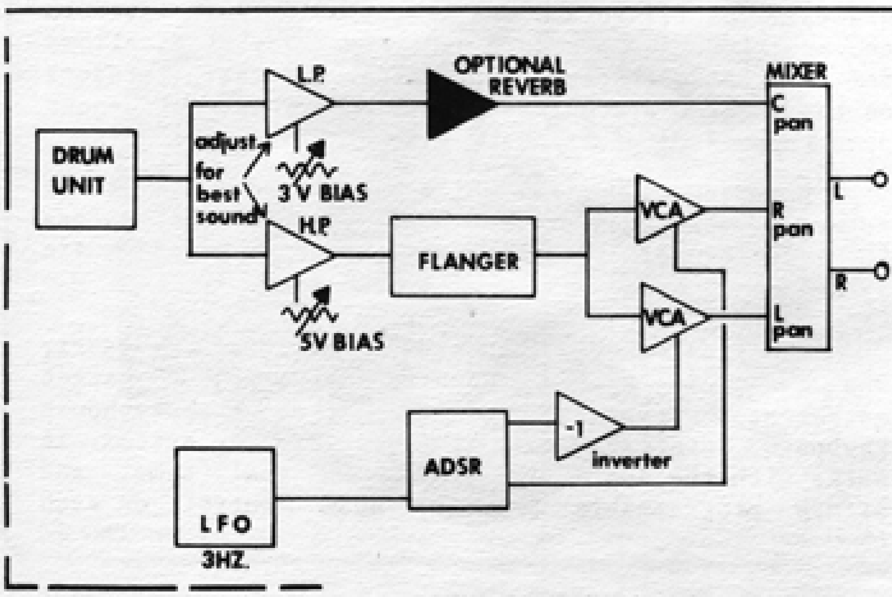
Harmonizing Echo

VCO: square wave output, low freq's best
KBD: glide off
ADSR: A - 25%, D - 25%, S - 60%, R - 40%
Output level - 50% (much more causes distortion)
REV: 100% reverb signal, no dry signal
VCF: Hi range, initial freq - 75%, Q - 60%
Lowpass is best, bandpass will work also.
LFO: approx 6 Hz., 40% output level

Play in time with the LFO. Broken chords (CEGCGEC or CFCGFCGFC for example) work best because the last note played can still be heard from the reverb unit. If any note and the one or two notes before it form a chord, the result will be a mixture of all 3 notes thus a sort of delayed harmony is created. As usual, experiment with the settings.

Jim McConkey
Rockville, MD





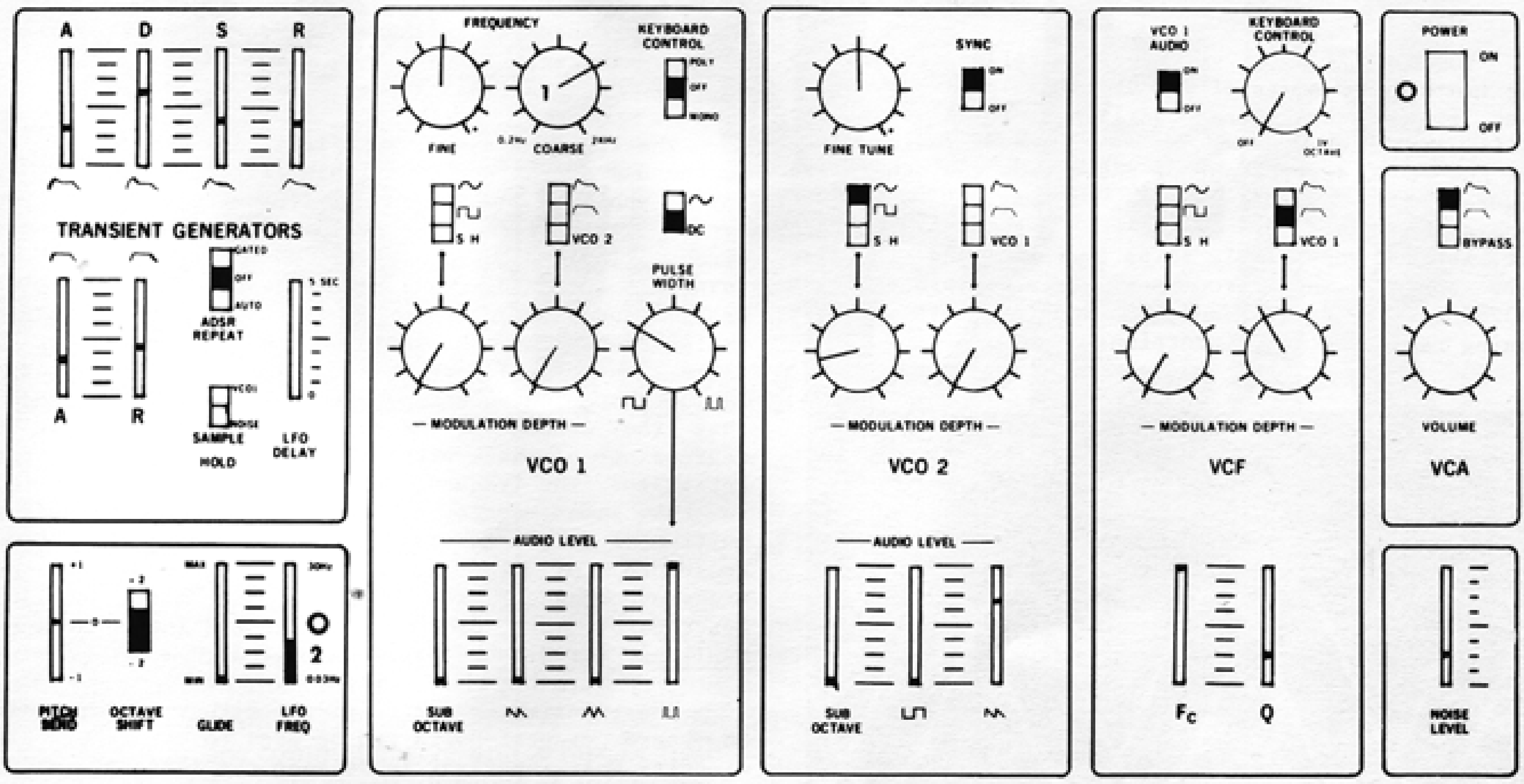
3D Drum Mix - Simulated Stereo With Rotating Cymbals

Optional Reverb should be at low setting
LFO: 3 Hz.
ADSR: A - 100%, D - 100%, S - 100%, R - 25%

Auto rhythm units are great when you don't have a live drummer to do the job, but their sound can get very boring if used on every track. Well, here's a patch to add a little sparkle to your mix. Warning: listening through headphones may cause brain damage!

Gordon McAlister
Ridgewood, NJ

the CAT srm synthesizer by OCTAVE Electronics Inc.



SOUND Pseudostring

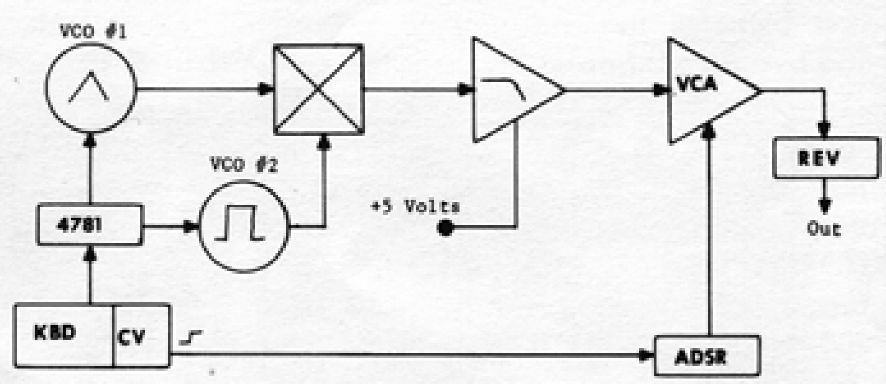
Notes: 1) Tune VCO 1 between B & C in indicated range.
2) LFO frequency control is phrasing control. Vary as indicated.

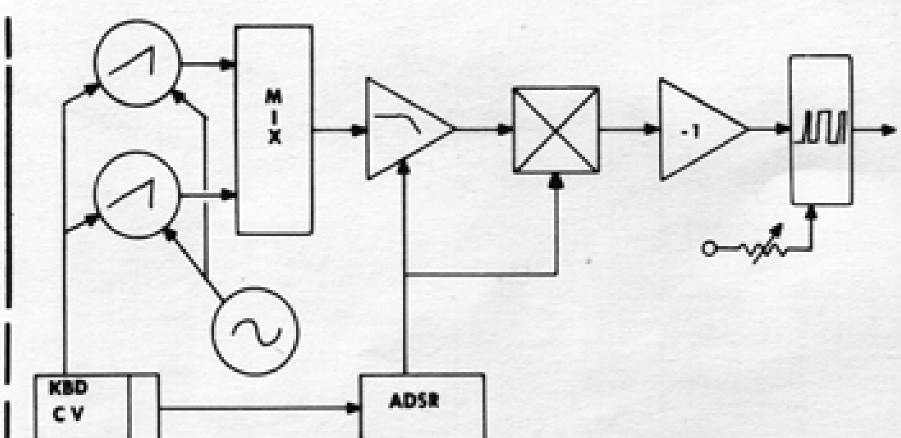
Jim Johnson
Terre Haute, IN

Metallic Voices

Keyboard: midrange
VCO #1: 20% pulse width (approx)
VCO #2: triangle out, transposed some degree to give metallic sound
VCF: low pass output, Q = 60% - 75%, vary Init Freq quickly from 50% - 100% to give the patch voicing imitations.

Dave Smith
APO San Francisco, CA





Heavy Metal Lead Guitar

LFO: 5 Hz.; vary level with note duration
VCF: high range; Q approx. 80%
Bal Mod: set mod input for VCA use
Inverter: 20 dB gain
Bias: 3 - 5 volts

Initially very slight LFO into VCOs. Increase level as note decays. Has very compressed dynamics full of gritty distortion. A pedal into the filter input allows very expressive playing. The last module is a Paia Triangle to Sine/PWM Converter.

Steve Meehleder
Grand Rapids, MI

Build a VCQO*

by James Patchell

VCQO? That's a voltage controlled quadrature oscillator. For those of you who may not know what that is, a quadrature oscillator is a sine wave oscillator which has two outputs. Each is a sine wave, but one is 90 degrees out of phase from the other (see figure 1). To put it another way, this module generates a sine and cosine wave output. As it turns out, it is a very simple thing to build and is also very useful. The circuit in this article uses the CA3280 (see Polyphony Jan/ Feb '80) for both a wide tuning range and low distortion.

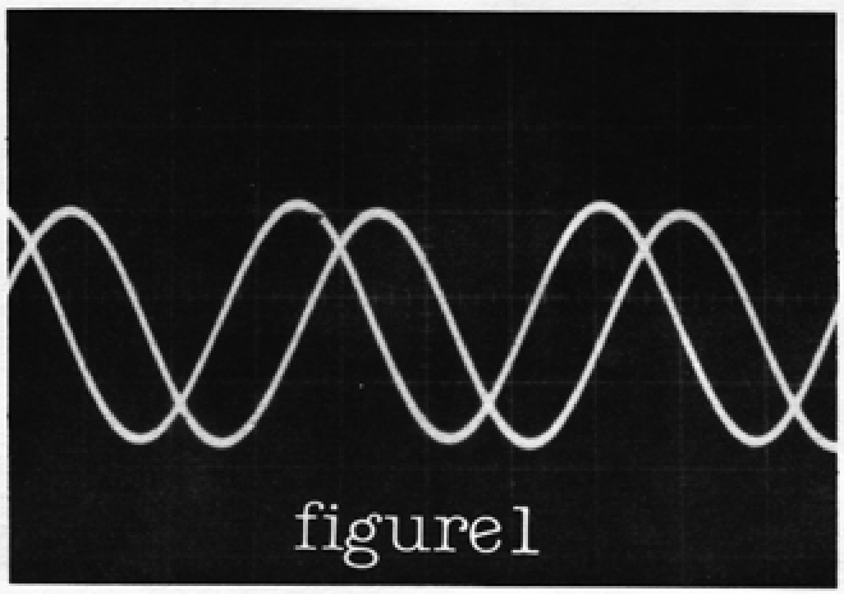


figure1

Looking at the schematic you can see that the circuit looks familiar. The reason for this will be evident in a moment. The circuit can be broken down into three sections (see figure 2):

1. exponential voltage to current converter
2. limiter
3. state variable filter.

The familiarity of the circuit is most likely due to the use of the state variable filter configuration. Anyone who owns a multi-mode filter module for their synthesizer will see that this is practically the same circuit. In this case, however, there is no input. And instead of using degeneration (to prevent oscillation), regeneration is used to sustain oscillation. The reason for the two outputs being 90 degrees out of phase is that the phase shift through an integrator network is always 90 degrees. This means that the output of the second integrator must be 90 degrees out of phase with the output of the first integrator. The frequency of oscillation is controlled in the same manner as the center frequency of a multi-mode filter is controlled, changing the time constants of the integrators by changing the transconductance of the OTAs. For a more detailed description, one can be found in the Analog Devices Nonlinear Circuits Handbook on page 79.

Construction is not critical. Either a home made PC board or perf-board can be used to construct this circuit. Most important is to keep the leads as short as possible, especially between the outputs of the CA3280 and the inputs of the TL074. High quality capacitors such as polystyrene (.001 mfd.) should be used in the integrators. Also, if accuracy is not a critical factor for your specific application, use a standard resistor for the Q81 temperature compensation

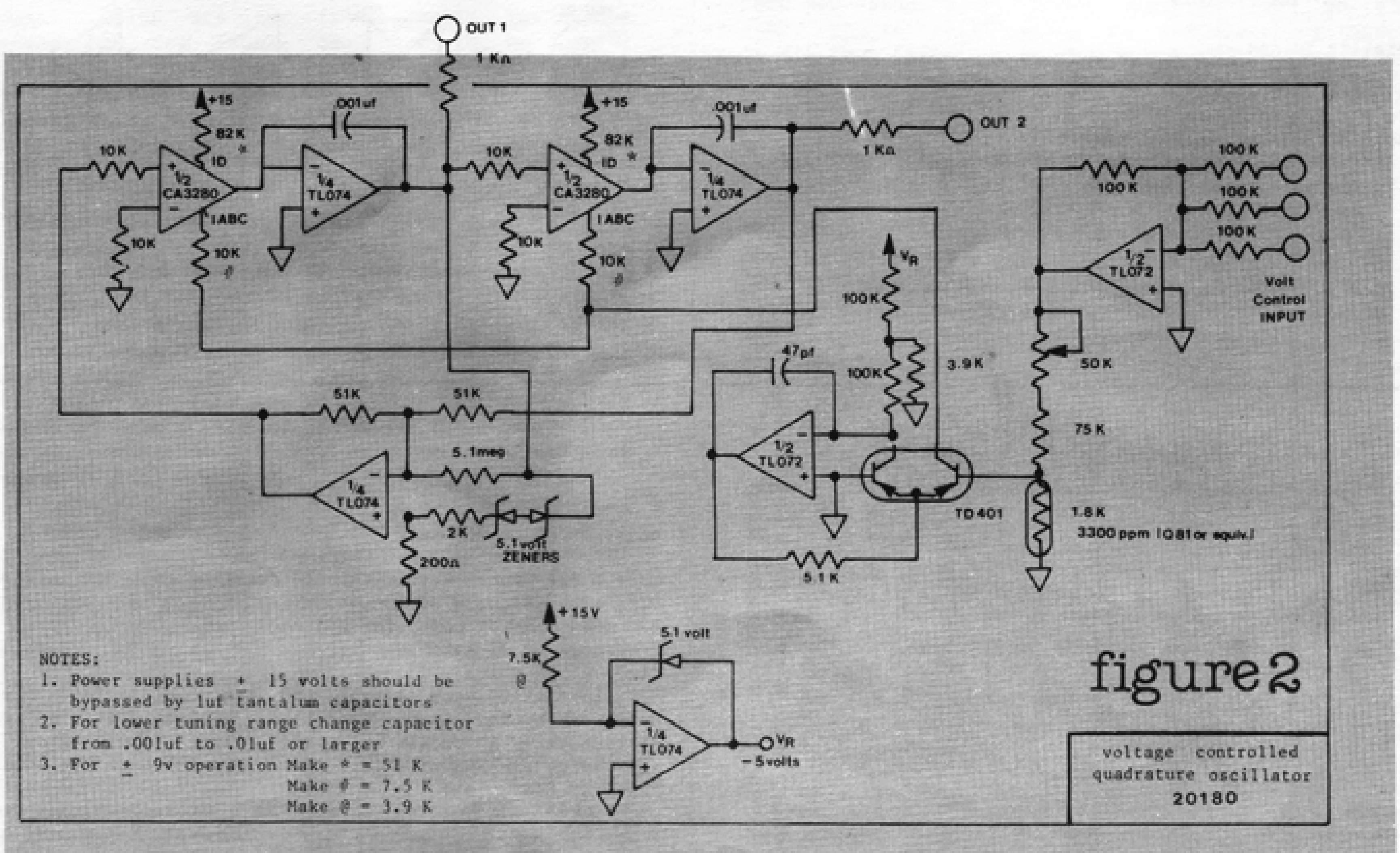


figure2

type resistor. This will save you about $2. Use caution in handling the ICs as they can be damaged by static electricity.

Calibration is straightforward, and proceeds as follows. Input 0 volts at the control input. Note the frequency of oscillation. Input 10 volts and adjust the frequency scale control for an oscillation frequency that is 10 octaves higher (X 1024). You should find the circuit easily tunable from 1 Hz to 16 KHz. You should also note that the control voltages can add up negatively as well as positively. The more negative the control voltage, the slower the oscillation. You must also be aware that this oscillator is most likely not accurate enough for use as a pitch source. This is due to the fact that the exponential converter has no means for error compensation. However, it can be added easily.

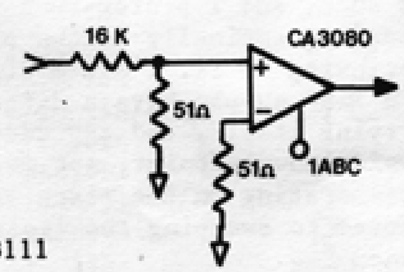
The circuit as shown was designed to run with a ± 15 volt regulated supply (such as a PAIA 4771 or Godbout supply). The circuit may be modified to use a ± 9 volt supply if the value of the diode bias resistors for the CA3280 are changed from 82K to 51K. The 10K current sharing resistors should also be changed to 7.5K. If this circuit is going to be used with linear control equipment (such as Yamaha CS series, PAIA, some Korg and EML) you may want to substitute a linear voltage to current converter.

I'm not going to spoil your fun by telling you all the things for which this circuit can be used. That is best left to the user. But to get you going, try some quad position modulation, chorus vibrato, stereo FM signal generation, or even as the basis for a frequency shifter (along with filters and balanced modulators). Capabilities are great. I'm sure you'll really enjoy it. Σ

References:
1. Analog Devices Nonlinear Circuits Handbook, Analog Devices.
2. Using the CA3280, James Patchell, Polyphony Jan/Feb 1980.
3. VCF Building Blocks, Gary Bannister, Polyphony May/June 1979.

*voltage controlled quadrature oscillator

Note: CA3280's available in small quantities from;



James Patchell
533 San Blas Pl.
Santa Barbara, CA 93111
Ditto; TD401 & 1.8K Q81

If substituting CA3080 for CA3280 use above scheme. CA3080 may require offset null pots, and may cause circuit to oscillate at lower frequencies, and will need to be attenuated more due to inferior (referenced to CA3280) distortion spec.

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POLYPHONY

Solving "Sonic" Blocks

by Chris Meyer

As opposed to trying to help you get over mental or creative blocks, I would like to present a series of steps or rules to help the person new to composing and recording (or the frustrated pro) in solving that age-old syndrome of "it just doesn't sound right!" As an example, I'll be using a piece called STEELWIND that I have been working on lately. While away at college, I had composed the piece and mailed the lead sheet off to my brother, Ronald, who produces me. When I came home for the summer and set up an impromptu studio in the basement, I started to practice the piece. It was then that all of my carefully laid plans started to fall to the wayside.

RULE 1: PERFORM THE PIECE BEFORE RECORDING IT.

A score and orchestration may seem perfect in the mind's eye, even after each separate line has been carefully written and performed. However, once the separate parts are performed together, disaster may result. I'm not referring solely to the notes themselves, but also the instruments and timbres involved. STEELWIND basically consists of a series of bass guitar solos played over a slowly evolving bass sequence, augmented by several various drones and effects. I played the sequencer line on bass, and also practiced my 'soloing'. However, when I fired up the sequencer and started playing against it instead of what was in my mind, I was appalled. Nothing seemed to fit. That was when I started invoking the following procedures.

RULE 2: PERFORM THE PIECE IN THE SAME ENVIRONMENT AS IT WILL BE HEARD.

In essence this means that if you intend the piece to be heard over a home stereo system, perform it through a home stereo. You would not believe how different my synthesizer sounds through my stereo as opposed to my old beat up practice amp. Especially with synthesizers, the exact timbre can be very important. Same goes for signal processing gear- work on the mix before you even record. You can't always "fix it when you mix it"! Get the sound you want before you enter the studio, and make sure it is indeed the actual sound.

RULE 3: PLAY THE PIECE LOUDER THAN YOU INTEND IT TO BE LISTENED TO.

Inevitably, someone who has a copy of your music is going to "crank it up." This is where a piece can sonically fall apart; where unknown glitches can appear. Example: I like to overdrive my VCA to obtain a certain warm sound. However, when I turn up the monitor, an annoying high frequency buzz appears that I did not notice at lower sound levels and that grated my ears. Other gremlins that could pop up include hum, filter popping, keyboard clicks, and flanger breathing. These can all be overcome or at least masked, but first you have to know that they are even there. You may be surprised to find that these were subconsciously bothering you.

RULE 4: IF THE PIECE SOUNDS MUDDY OR CLUSTERED, CHANGE THE TIMBRE OF AN INSTRUMENT.

This section could be subtitled "equalization can help, but not always cure". Equalization is fine for minor correction and enhancement of individual signals, but it is not a cure-all when something is drastically wrong. Since I play bass guitar and usually patch bass sequences, things tend to get muddy down in the low end. Since the timbres are so close, any note played on bass that wasn't some "pretty" interval from the sequencer created beat notes, which ruined the clarity even more. Sometimes merely turning up my high pickup helped, but in my most recent STEELWIND practice session I ended up sending my high pickup through an envelope follower and the low pickup through a mild harmonic overdrive. During a demo taping I fed the bass through a flanger - it suddenly stood apart from the sequencer, and I was able to take more liberties with my solo. Now I don't suggest that you gimmick up every sound - just the ones that need it. Sonic clarity is very important, so don't be afraid to make seemingly drastic changes. Which leads me to my next point.

RULE 5: IF YOU'VE ALWAYS DONE SOMETHING ONE WAY, TRY A DIFFERENT WAY.

This may seem like a rather obvious suggestion, but old habits are indeed very hard to break - especially when you may not be aware that you've even developed a habit. The example of the overdriven VCA comes into play again. I usually feed an extra five volts of bias to the VCA along with cranking up the gain, thus gaining some extra midrange harmonics. However, these harmonics were still fighting with my bass. After trying various filter settings and equalization, I finally killed the bias and turned the gain back down. A much cleaner sound resulted. This was quite a paradox for me, because I usually overdrove it to eliminate some of the characteristic dry, clinical sound a synthesizer can have. Other examples include trying less filter "Q", different picking methods, a little glide, etc.

RULE 6: DON"T ORCHESTRATE, FILL.

If your music sounds a little sparse, don't augment the existing lines - create a counterpoint. To use an analogy, you don't smooth out a landscape by adding dirt to the mounds, you use it to fill in the valleys. Jarre used various oscillations and flying things to great effect in his work OXYGENE; Synergy's first album employed sample and hold, soaring pitches and such, and I prefer such things as wind to add something sonically to the piece and to add some continuity to it. I'm a minimalist - I believe that each line should have a different timbre capable of carrying itself, and any additional lines should provide counterpoint, not merely augment. It makes concentrating on the piece easier and refreshing, as opposed to swamping the listener with sheer masses of sound.

RULE 7: SEEK OUTSIDE OPINIONS.

An individual composition can be an intensely personal thing, but one shouldn't lose his or her objectivity. If a composition doesn't sound right to even you, ask a friend for some feedback. I'm not advocating asking any person off the street; the advice should come from one fairly well acquainted, or at least well informed, with what you are trying to do. Don Slepian, for example, actively seeks feedback to the music he puts out. Music is usually recorded in the first place for others to hear, so there is nothing wrong with asking for some advice. It may point you in some direction you may not have previously thought of. It's plain and simple - you can't think of everything, and getting help is not a sign of weakness, but one of strength and realistic approach.

Just glancing over these rules, you may think it's nothing more than common sense. However, in the pressure of a recording situation, your thinking is not its clearest. I'm not telling you to change your style or anything - you are you, and it should stay that way. Just be ready to do things differently if the situation warrants it. It is much easier to go around a wall than through it. And anyway, this article has just been good creative catharsis for me and my related problems. Just work all of your problems out before that studio time starts running.

computer music festival

continued from page 23

voice, and a velocity sensitive keyboard. Performance oriented features include the ability to select presets from instrument files while playing, echo options, and a bar graph representation of the frequency content of a voice to facilitate waveform construction. This system was designed to create new types of voices rather than to imitate conventional voices, although it does do imitative synthesis fairly well.

The final part of the afternoon was a talk on synthesis, computers, and music by well-known New York composer and synthesist Laurie Spiegel. It was refreshing to listen, for the first time that day, to a musician talking about the expectations musicians have towards technology. Her theme seemed to be that the needs of the musical community were not yet met by the technological community.

For example, in reference to music synthesis, she felt that additive synthesis was not the best way to bring the computer to synthesis. She preferred subtractive synthesis, noting that what a computer does best is to generate square waves. She also felt that cost effectiveness could be reached with subtractive synthesis, noting the high prices for a Fairlight or Crumar. She felt that the feedback loop present in analog synthesizers (move this knob to get a different sound ... instant response) was a necessity not really present in digital synthesis. She also felt that the synthesizer was the instrument that, through a computer, could bring ways of expressing valid musical ideas to people who didn't have the ability to play acoustic instruments. All in all, she asked the designers of computer music devices to think of the needs of a musician or a composer when designing equipment.

The audience sat hushed, hanging on every word of her talk. There was a great deal of interest in hearing what the musicians of the world expect from the engineers. Hopefully, this interest will result in even more useful equipment in computer synthesis. If so, then the PAC festivals of future years will bring improved equipment every year, resulting in progressive, creative synthesis. Σ

Programmable Drum Update

NEW...LOW, Low power RAM for your PAiA Programmable Drum Set multiplies battery longevity. IT CAN SAVE YOU BUCKS!

by Steve Wood

Time can change many things, but there are few that it can alter as drastically as does technology.

Take, for example, the PAiA Programmable Drum Set (cat #3750). When this device was first introduced, it was the state of the art and then some. For the first time, you weren't stuck with what some engineer somewhere thought was a funky rhythm, you could quickly and easily make up your own (and there's no one funkier than you, right?).

While the concept of user programming is still the most advanced available, almost three years have gone by and in that time things have changed. The 2112 type memory used by the PDS is for all practical purposes obsolete (as in hard to get). Time to go back to the drawing board.

Now fortunately technology rarely changes just for the sake of change (though it may often seem that way). In most cases it improves. This is the case here because while the 2112's were quietly following their line into oblivion, other parts were getting less expensive. Like a thing called a 5101.

For those of you who don't carry parts catalogs around in your head, a 2112 and 5101 are in many ways the same thing. They are both static memory chips. They are both organized in a 256 X 4 format (256 words each 4 bits wide). They are both... Well, come to think of it, after that they start to get pretty different.

The 2112's are MOS parts and in their day were considered to be fairly low power (only about 70 ma. each). Why, that's low enough to run from batteries, as long as your uncle owns a battery factory. A new set of quality batteries in a 3750 could be expected to be completely drained within a few hours of operation.

It should come as no surprise that there are some people who consider this a disadvantage.

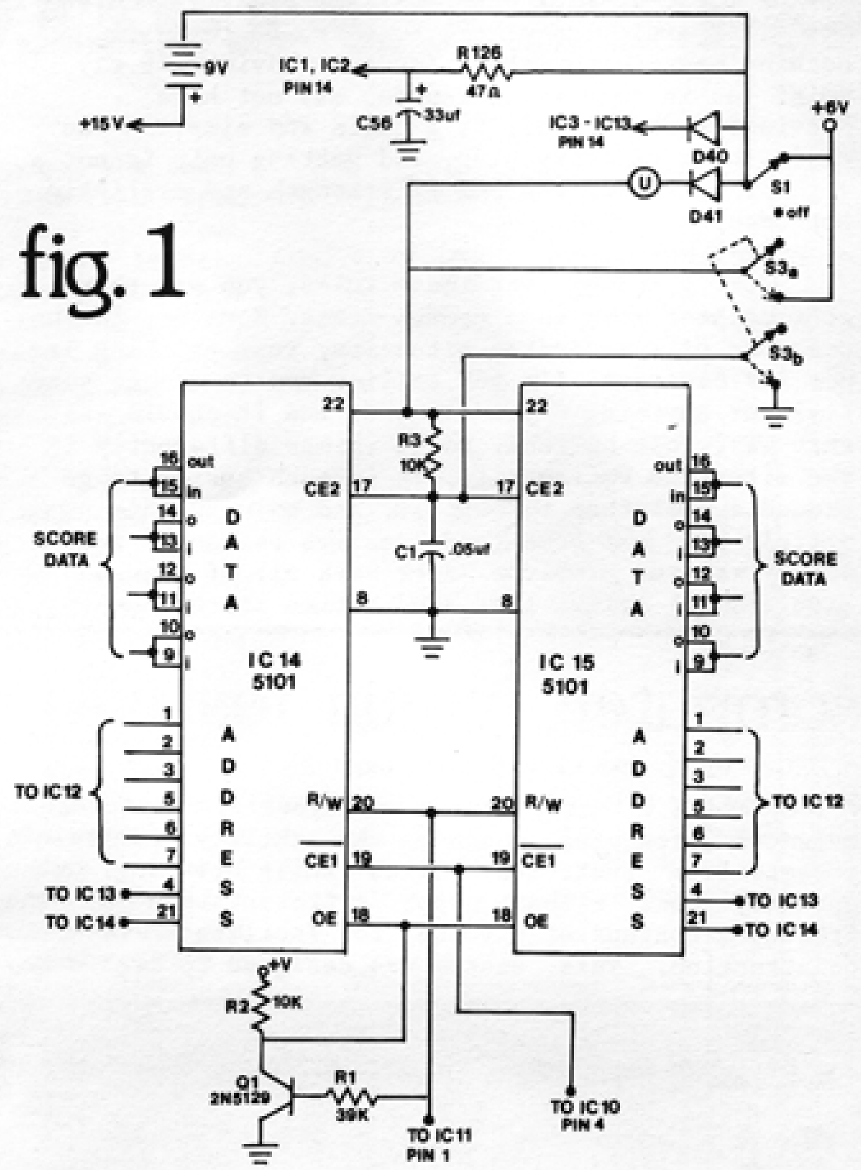
5101's, on the other hand, are CMOS parts. Not only do they draw a mere 17 ma. apiece in operation, they have provision for a super low power "power down" mode. In this "save" mode, they each draw an insignificant 15 micro (as in teeny-weeny) amperes. Since internal leakage of many batteries is more than this, a PDS with these memory chips can be expected to retain programs for as long as the batteries would last stored on the shelf not connected to anything.

There are differences between the two, at least one of which at first glance would seem to be somewhat less fortuitous than the power situation. While the 2112 has four lines which serve as both Inputs and Outputs, the 5101 has eight lines, four for Input and four more for Output.

Before your mind begins to wander too far into images of tri-state bi-lateral buffers and such, look at figure 1. All that really needs to be done is to tie the I/O lines together. The key to this is careful use of the 5101's $\overline{OE}$ (not Output Enable) pin. This pin allows us to "float" the output lines.

By making the $\overline{OE}$ line the complement of the Read/Write (R/W) line, we realize a situation where putting the chip in the Write mode (R/W low) simultaneously puts a high level on the $\overline{OE}$ so that the outputs (because they are disconnected internally) don't wind up writing back into the inputs again.

fig. 1

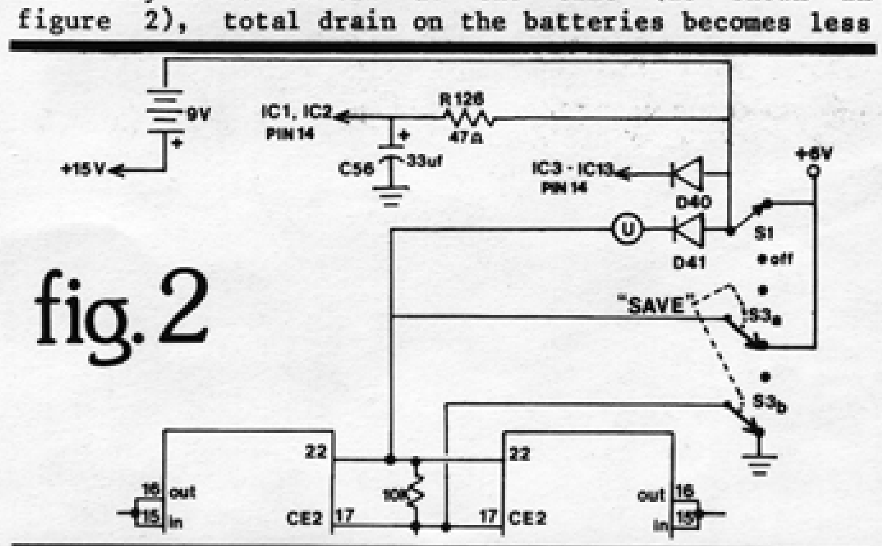


This required inversion is preformed by the transistor Q13.

Other obvious differences between the two chips are physical size and pin count. The 5101 is larger. But fortunately not so much larger that it would not fit into the space allowed for the older memories on the original circuit board. All current Programmable Drum Sets are being shipped from PAiA with these lower power memories.

Taking advantage of the 5101's low power "save" mode is simply a matter of taking its second select line (CE2) to ground. If the power to the drum sound circuitry is turned off at the time (as shown in figure 2), total drain on the batteries becomes less

fig. 2



than 40 micro-amperes. Most batteries can survive that kind of drain for many, many months.

For those of you who are interested in retro-fitting these newer semi-conductors to your old 3750's, PAiA has a special retro-fit kit(3751-$19.95 postpaid) which includes 5101's, the transistor, and associated parts with a board to mount them all on which plugs into the existing memory sockets, as well as detailed instructions for assembly and installation. Altogether, a change worth making.

The Q9 Story

Reworking the 3750 P.C. board to accommodate the 5101 ICs gave an opportunity to correct a sort of "birth defect" that the unit had. The flaw was that the transistor Q9 was in the circuit backwards; its collector and emitter interchanged. Ordinarily a flaw like that would be amply obvious since it would produce apparent malfunctions. But in the case of the 3750, any problems caused by the backwards Q9 were so very subtle that they often went unnoticed. That's how the kit managed to get into production with such a flaw in the first place: all prototypes seemed to work great. No doubt the first two or three proto models had a correctly installed Q9. But somewhere along the long and demanding road to project completion and kit production, one tiny solder pad in the maze of foil patterns was misplaced, causing Q9 to fit on the board in an electrically backward situation. The reason this flaw went unnoticed is that in this particulat set of circumstances, the PNP transistor Q9 would appear to be doing its thing, even though it was in the circuit backwards.

Looking at the schematic drawing figure 3, we see that Q9 provides a switchable current path between the emitters of Q2 through Q8, and ground. A negative pulse from the output of IC10d is applied to Q9's base, turning the transistor on hard for the duration of the pulse. That brings the emitter of Q9, and the emitters of Q2 through Q8, to a virtual ground via Q9's collector. That's the way its supposed to happen. But with Q9 in the circuit backward (its emitter grounded and its collector connected to the emitters of Q2 through Q8), why does the circuit still work? Well, it doesn't really work right, but the difference is usually so slight that its hard to detect.

fig. 4

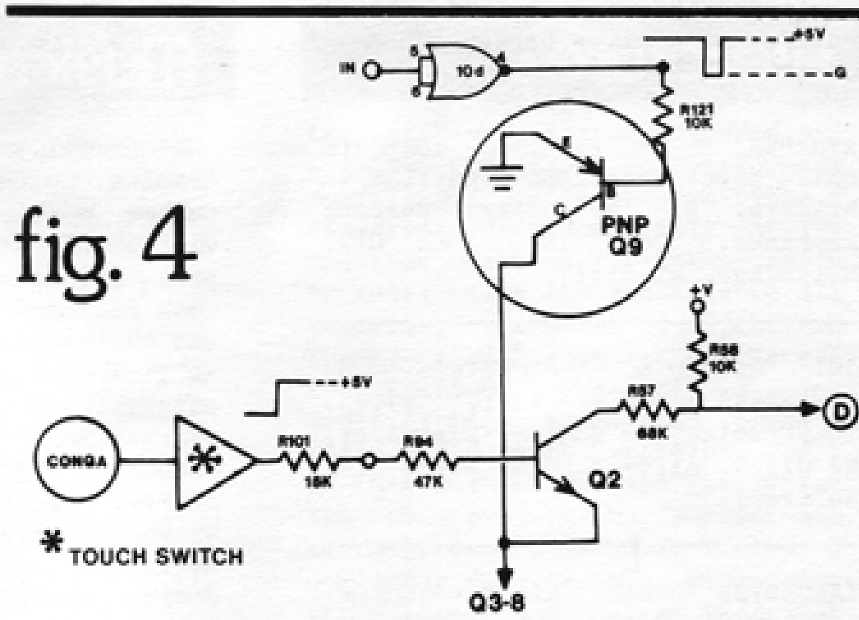
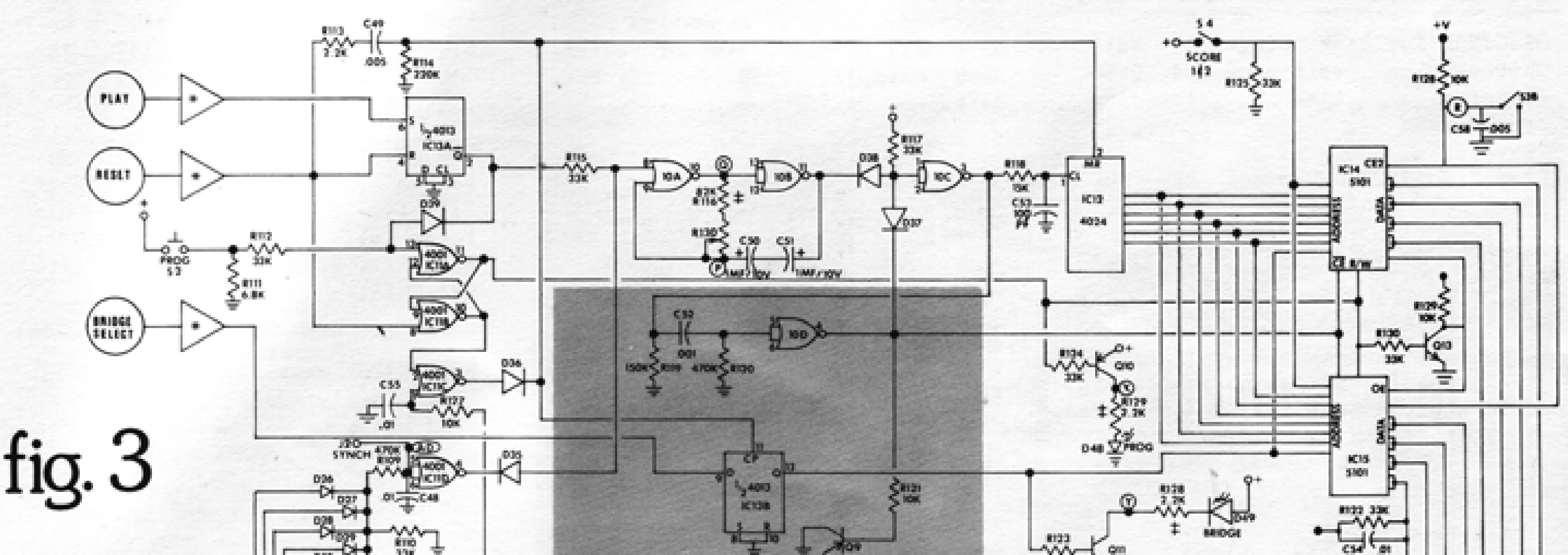


Figure 4 shows how Q9 is really connected in the original 3750 circuitry. When the negative going pulse happens at the output of IC10d and any one of the emitters of Q2-8 is at some positive voltage (possibly due to the high state of a touch switch

(continued .... page 34)

fig. 3



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equipment

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Drum Update...

continued from page 31

output) then Q9's base/collector junction becomes forward biased. This provides a limited current path from the emitters of Q2-8 via resistor R121 to the output of IC10d (which is at ground during this time). The abnormal operating conditions of Q9, together with the slight variations in value of R121, cause subtle variations from unit to unit in the amount emitter current allowed to flow from Q2-8. That caused a variation in the amplitude of the excitation pulses applied to the oscillators, so that noticable symptoms tend to manifest in the audio output. The most common symptom is a weak or inconsistent output of the noise burst portion of the snare drum sound. If you have the old 3750 and wish to correct Q9, simply remove it from the board, turn it 180 degrees and re-install it so that the emitter and collector are interchanged. It may also be desirable to change resistor R47 to about 68K, and R53 to 39K. You will need to recalibrate the unit after making these changes.

Some of you have inquired about the feasibility of programming more than one drum sound into a single event. For instance, suppose you want to enter a score that has a lick using a snare AND a bass drum on the same beat. The 3750 is designed so that when any one of the drum sound programming pads is touched, the tempo clock in the unit is enabled (allowed to run), so that after the DATA for that event is loaded into memory, the address counter can advance one count. That selects the next memory location in the sequence, so that the memory will be ready to accept the DATA pertaining to the next event. In fact, if you hold your finger on the pad, the tempo clock will continue to run, causing the address counter to keep advancing. If the unit is in the PROGRAM mode, the same DATA (drum sounds) will be loaded into each successive memory location, at a rate set by the "TEMPO" control, until the touch pad is released. So, in order to program both snare AND bass into a single event you would have to see to it that both fingers contacted their respective pads at precisely the same moment. And if you want 3 or 4 drum sounds on the same beat, well.... But, alas, contrary to the aura of impossibility I seem to attatch to the task whenever I think, talk, or write about it, it's embarassingly simple.

The installation of one SPST switch (to serve as "tempo clock disable button"), between ground and the synch input jack J2 makes it a snap! (figure 5)

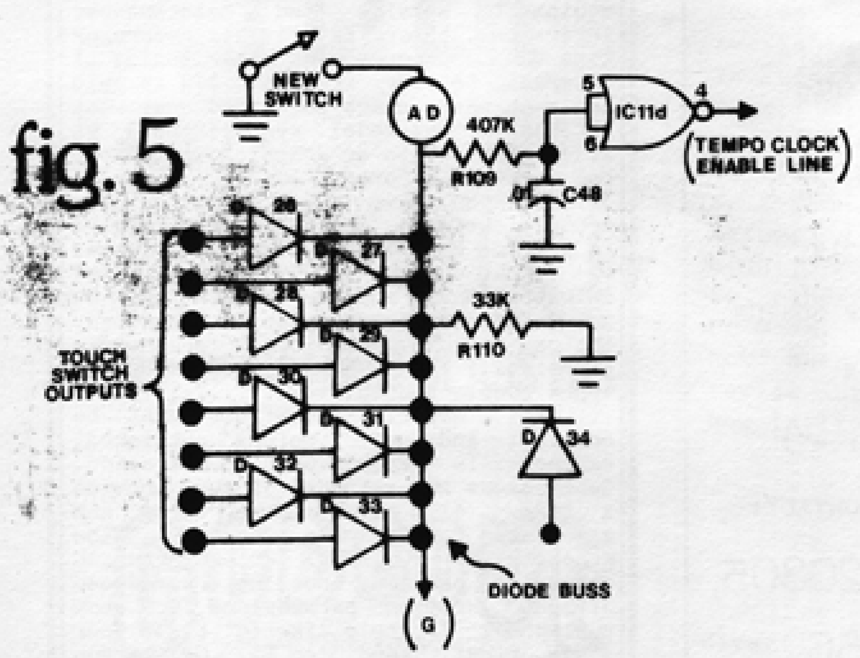

fig. 5

For instance, if you were to remove J2, you could then install a small normally open pushbutton switch (similar to that used as S2, the PROGRAM button on the 3750 control panel) in the J2 hole. The wire that was formerly connected to J2 (originating at point AD) can then be connected to one lug of the new switch, while the other lug is connected to ground. Closing the switch will ground the inputs of IC11d, so that it cannot switch when a "high" (logical one) appears on the diode buss as a result of an ON touch switch.

So to use the new switch, when you come to a place in the score you are entering where you want more than one drum, simply press and hold the button. While holding the switch on, touch the pads of the drum sounds you wish to enter on the next beat, and while holding your fingers on these pads release the button. All the drum sounds for which you are holding touch switches ON will be entered into that one event or memory location. The time between beats (as set by the tempo clock) is important when doing this because, as mentioned earlier, when any pad is touched (and the new switch is not held ON) the tempo clock is free to run at a rate set by the tempo control. So when you release the clock disable button, DATA representing the status of the touch switches is immediately loaded into memory, and you have just so long to remove your fingers from the pads before the same information will be loaded into the next memory location. You'll have to play with the setting of the tempo control and your technique to get things just right.Σ

JOHN SAYS

There's a problem with entering more than one drum sound on a single event: Don't be surprised if you don't like it. You may think you are going to hear something that sounds the way a drummer does when he hits two drums at once. But you won't, cause he doesn't. No way even the best percussionist that ever lived hit two drums at exactly the same time. Because they are not exactly synchonized you get a somewhat "fuller" sound.

The PDS is going to hit at precisely the same instant. Maybe you won't like that.

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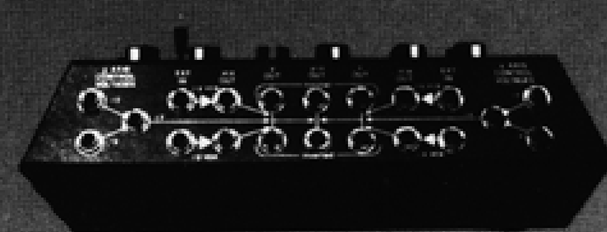

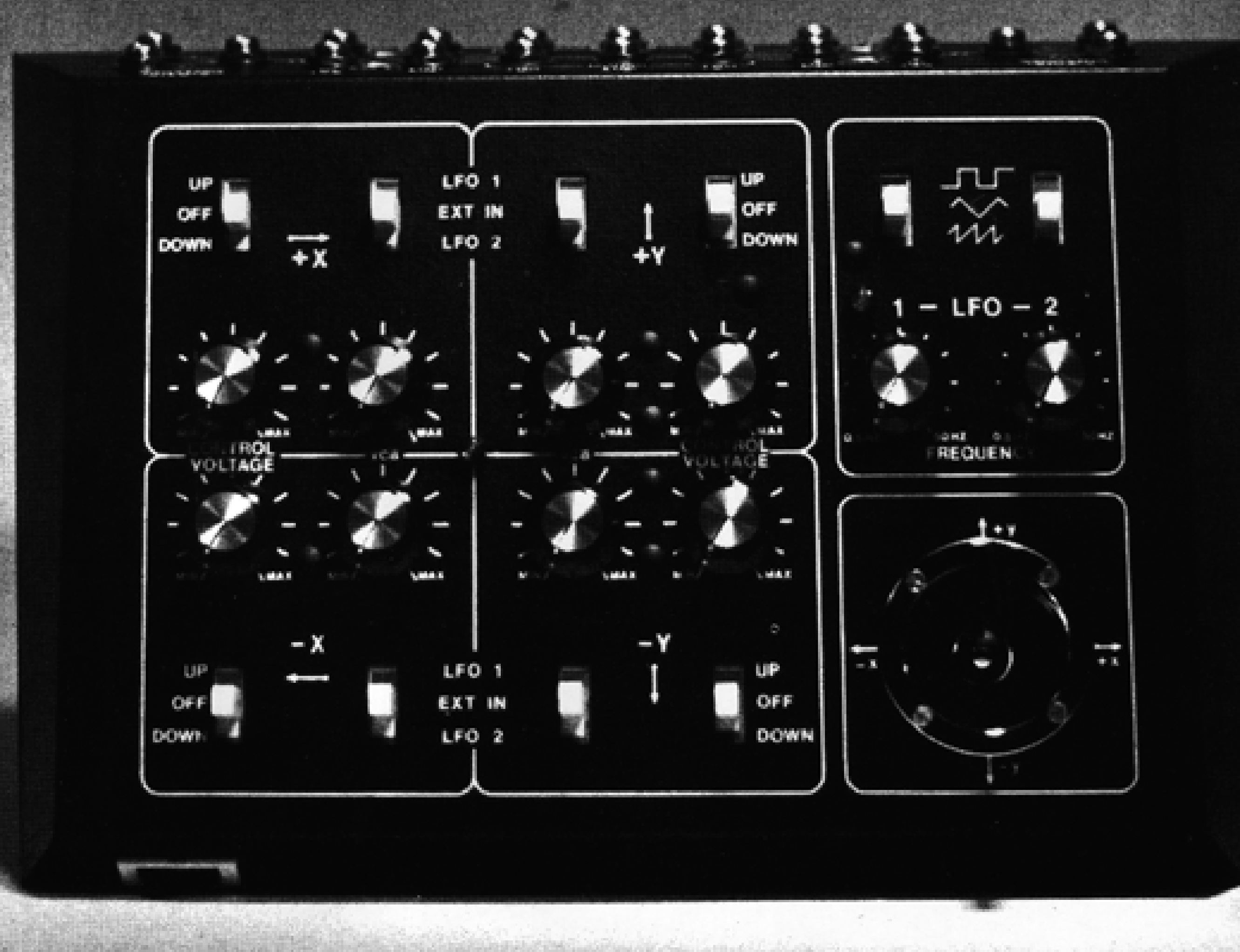

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