POLYPHONY

Nov./Dec.
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....just makin' waves...

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FORMAT FEEDBACK

In keeping with the "feedback" spirit of POLYPHONY, here is a short letter of praise- criticism- info. First of all, I am glad to see that you guys have caught up with your issues. The July/August publication was one of your best yet (love that cover!). The whole format is far more polished as of late, and I am glad to see more and more advertisers with each issue. I think this is a good move on their part as I feel that POLYPHONY tends to reach those of us in electronic music who are really concerned about the maturation of the medium. As I stated in my "Jot" article (by the way, that was another great cover) the free exchange of ideas is of the highest import to any art. I feel that your magazine will become the "paper telephone" of the electronic music world - it is doing so already.

There is one thing you have yet to clean up - your justification! The spacing between your words is sometimes galactic! If there is anything to fault POLYPHONY on, that is it. 'Nuff said.

John Mitchell
San Luis Obispo, CA

REVIEW TACTICS

I would like to comment on a couple of reviews that have appeared in the last two issues. One is the review of Pat Gleeson's unreleased album. The other is the Roger Powell review, again of an unreleased album.

I realize that this is done more for political purposes than any others. That is, write a good review, and when enough of these appear the record company will be pressured into speedy release and distribution of the album.

But this also points out the fallicy of continuing in the corporate structures as they exist. Might it not be better for Gleeson and Powell to release albums privately and distribute them privately. When the big artists start going this route, that will signal the end of the corporate domination of the arts. As long as these people go through the whole record company routine, the corporations will continue to suffocate the arts.

I realize that I get somewhat overenthused about this possibility. But such is the way of the revolutionary. And you, as a publication dealing with computer applications, must understand the possibilities that the computer can realize. That is, you can see how the computer can totally transform society. When the home computer becomes as common as the telephone, the changes will be swift and all inclusive. So it is in the arts.

But it is going to take the efforts of a lot of people to bring this to full potential. That means people who are "names" as well as those who are not yet stars. As long as people continue to kow tou to the corporate interests, this cannot occur.

So I say to Pat Gleeson and Roger Powell, screw those record companies, pay them back in kind for what they are doing to you. If the record companies were really interested in you as an artist then there would be no problems with releases. But since they are only interested in sales, and not in minority markets such as synthesizer music, you owe them nothing and should act in your own interests.

Well, deep breath as I calm down. I feel very strongly about this. I get tired of hearing nothing but disco on the AM stations, nothing but hard rock on the so-called "progressive" FM stations, and so on. And I get tired of a few monopolies controlling what I listen to, what I buy in the stores, and what gets recorded in the first place. Guys like Mike Danna, Mark Petersen, Robert Banks aka Greenberg, Mike Gilbert, and I don't know how many others are already in the vanguard of the revolution. Gleeson and Powell (and Eno and Fripp and Schulze and everyone else) are welcome to take part.

Chuck Larrieu
Corte Madera, CA

industry reports

vocoder plus

Roland presents a totally new concept in polyphonic keyboards with the introduction of the Vocoder Plus, an instrument that combines vocoder circuitry with two other tone-generating sections (string, human voice) to achieve a dramatic and usable effect. Each of the three sections may be independently assigned to cover the whole keyboard, or either the upper or lower half. In addition, each half of the keyboard feeds into its own output so that the Vocoder Plus can be run in stereo. The Vocoder Plus contains a balance control between all sections as well as vibrato controls that allow selection of rate, depth, and delayed vibrato.

The string section produces orchestral string sounds with independent control of tone and attack time. The release time is shared with the human voice section. In the human voice mode, a lifelike chorus of human voices is produced with one female and one male chorus on the upper half of the keyboard, and two male choruses on the lower half. The vocoder section processes the spoken or sung human voice, and uses this information (or program) to modify another musical signal (known as the carrier). The vocoder section uses the 'human voice' signal as its carrier, but will also process an external signal if desired. The microphone input will accept either phone plug or XLR connector.

In a live performance, the Vocoder Plus can be used to strengthen a band's vocal capabilities by literally adding a chorus of voices singing the same part. The String and Human Voice sections offer additional enhancement. The list price of the Vocoder Plus is $2695. For more information, contact: RolandCorp US, 2401 Saybrook Ave., Los Angeles, CA 90040.

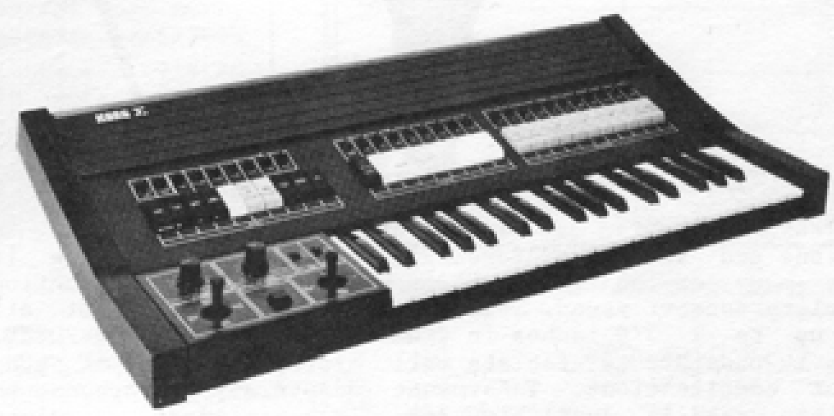
price reduction

Eventide Clockworks is pleased to announce a reduction in the list price of the model H910 Harmonizer. A unit complete with digital pitch ratio readout and second output (delay only) will now cost only $1500. The previous price was $1865. This price decrease is made possible partly by a reduction in material costs, and partly by improved production methods allowed by product standardization. Eventide expects to be able to deliver the H910 Harmonizer with readout and second output 'off the shelf'. For more information, contact: Eventide Clockworks, Inc., 265 West 54th Street, New York, NY 10019.

synthesizer seminar

Polyphony columnist Craig Anderton, nationally known author on the subject of musical electronics and all around good guy, will conduct a seminar on "Synthesizer Basics" in Concord, California on January 16, 1980. Topics will cover basic synthesizer concepts, keyboard synthesizers, and mating synthesizers to conventional instruments such as guitar and woodwinds. For more information and free tickets, call 415-676-3151 or write Mau's Music, 1450 Monument Blvd., Concord, CA 94520.

performance synth



Korg announces a new concept in performance oriented keyboard instruments: the new KP-30 "Sigma" monophonic synthesizer. The Sigma features nineteen mixable voices which can be used separately or in stereophonic unison for a sound previously never heard from a performance instrument. The KP-30 utilizes two separately tunable VCOs with six sub-octaves, and a separately programmed synthesizer module (waveshaper, VCF, VCA, and EG) for each voice! Moreover, each voice is user variable in its most important parameter (eg. Fc, Attack, etc.).

Voices are divided into two groups with separate outputs: Instrument (conventional acoustic instruments) and Synthe (synthesizer voices). A mono output is also available. The Synthe section features voltage controlled low pass and high pass filters. Twin joysticks control pitch bend, modulation by LFO, modulation by noise, and filter cutoff frequencies. Other exciting features include programmable touch sensitive keyboard, single/ multiple triggering, sample and hold, ring modulation, and full interface patching.

Whether you want the sound of a flute, tuba or electric bass, the sound of four seperately programmed synthesizers -- or all of these at the same time -- they are yours instantly with Korg's new Sigma synthesizer. Suggested list price is $1400. For more information, contact: Unicord, 89 Frost Street, Westbury, NY 11590.

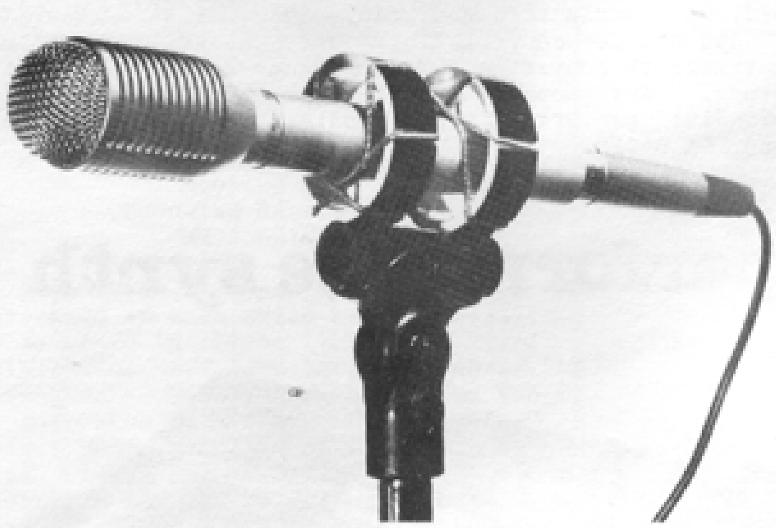
portastudio & catalog

The Tascam division of Teac has announced their M-144 Portastudio which is a combination four-in two-out mixer and multi-track (4 track) cassette recorder that weighs less than 20 pounds. The Portastudio is a musical instrument on which up to 10 musical instruments or vocals can be recorded using Teac simul-sync "ping-pong" recording with only one-time dubbing for each instrument. Dolby and double speed recording are used to obtain better than average quality from the cassette recording format. It is a versatile creative tool that can be used by musicians, composers, audio-visual technicians, educators, and recording artists.

Also available is a new product brochure for the Tascam series Professional Products by Teac. Recorders such as the 35-2, 40-4, 80-8 and consoles like the model 5 and model 3, as well as many other products, are explained in detail with panel close-ups and color layout throughout. The book is being sent to dealers nationwide as well as consumers who inquire about Tascam products.

For more information, or a catalog, contact: Teac Corporation of America, 7733 Telegraph Rd., Montebello, CA 90640.

mic isolation



Tensimount is a device of particular interest to musicians and sound reinforcement specialists, because it solves many on-location problems which plague the production of quality concert sound. Tensimount allows isolation of all mics up to 1 3/8 inches in diameter at greater than 20 dB, making it possible to isolate all microphones in a setup without complications. Tensimount adapts all microphones to fit into readily available 3/4 inch clamps; complete standardization of microphone clamps for your sound system is the result. Isolation of vocal mics which must be unclipped and hand-held by the performer is possible with Tensimount, as is emergency mounting of mics where space or number of stands is limited.

Microphone isolation, long appreciated by studio workers, has been neglected in live music where monitors, speakers, flimsy stages, drum sets, and instrument amps make shock mounts a much greater necessity. In general, all mics used in live music applications should be isolated. Tensimount, a simple, sturdy, unobtrusive device now makes this possible - at an affordable price! Suggested price is $9.95 with storage box and instructions. For orders or more information, contact: Brewer Instruments, PO Box 163, Newton Highlands, MA 02161.

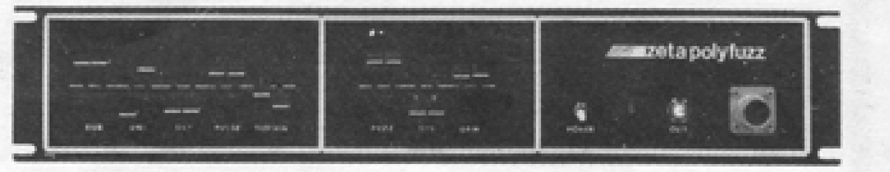
module giveaway

Aries Music announces their first contest, the subject being discovery of interesting uses for their AR-334 Sequencer and AR-335 Switches modules. The AR-334 is a potentiometer-memory, 8 step by 2 row sequencer with position gate outputs along with reset and run, enable & step inputs. The AR-335 is a unique set of 4 bidirectional switches: 2 SPDT (pulse controlled), 1 SP4T (pulse controlled), and 1 SP4T (voltage controlled).

Contestants will be asked to submit a block diagram for a patch which takes advantage of as many features of these modules as possible, while producing a musically interesting and useful result. The twenty best patches and their descriptions will be published and sent to all the contest entrants. The five finalists will have their patches and descriptions published in Polyphony, whose readers will be invited to vote for the best patch. The winner will be able to select $600 worth of Aries Music modules. (Which, coincidently enough, is the value of one each, Sequencer and Switches, assembled, or three modules in kit form.) The finalists will be selected by Bob Snowdale, president of Aries Music, Ron Rivera, designer of the modules, and Mark Styles & Ken Perrin, noted Boston area composers of electronic music.

For more information, contact: Aries Music, PO Box 3065, Salem, MA 01970, (617) 744-2400.

guitar processor

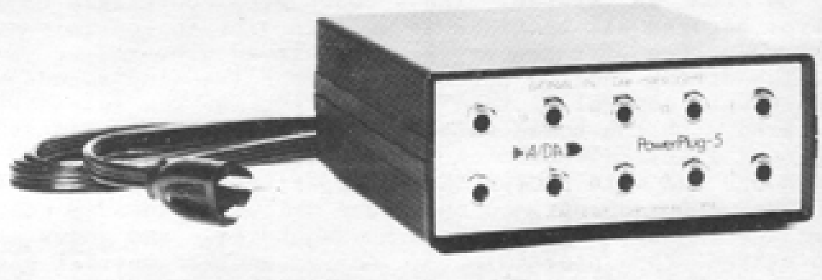


H.E.A.R., Inc. brings the fuzztone to the '80s with the Zeta PolyFuzz. This hexaphonic fuzz unit is a six channel modifier for the electronic guitarist who is ready for hex processing. The Zeta PolyFuzz's array of sounds offers the guitarist a world of timbral possibilities at an affordable price. Operating with most hex pickups, the PolyFuzz generates five different kinds of fuzz for each string on the guitar, allowing the guitarist to play rich, distinct chords and harmonies. The PolyFuzz's incredibly wide range of pitches and overtones complement and enhance other guitar effects such as flangers and phasers.

The effects offered are Sub-Octave Sawtooth, Unison Sawtooth, Skysaw (an octave up effect that changes with your picking attack), Sub Octave Pulse (a modulated pulse wave), plus traditional overdrive, and the clean pickup sound. Separate mixing controls for high (E, B, G) and low (D, A, E) strings are side by side for ease and flexibility of control. On all fuzzes except the traditional, a dual sustain control adjusts the amount of sustain or dynamic following. Multiple access jacks on the back, including dual envelope follower out and stereo out allow for flexibility in patching. Fast stage accessability and solid construction assures the guitarists of superb live performance capability. The unit uses 3.5 inches of rack space.

For more information, contact: H.E.A.R. (Holt Electro-Acoustic Research), Inc., 1122 University Ave., Berkeley, CA 94702.

universal power supply

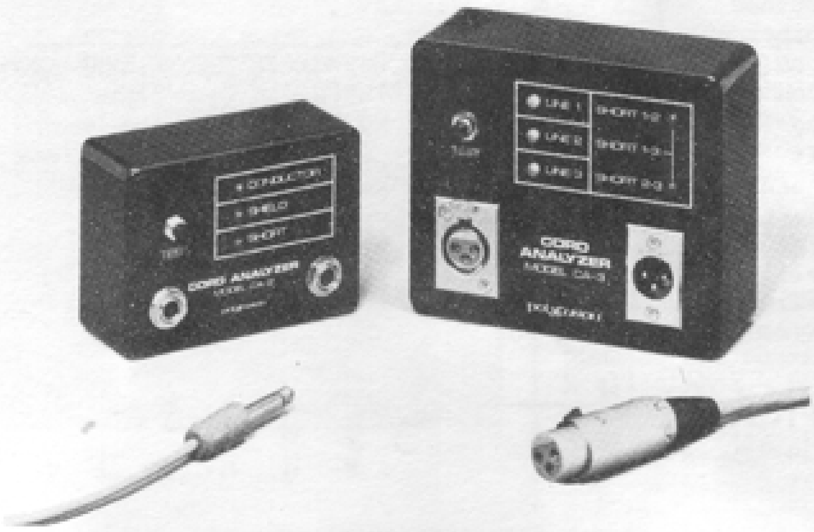


The A/DA PowerPlug-5 is the first universal battery eliminator designed for the musician who uses 9 volt battery powered sound modifiers and pre-amps. Capable of powering up to five devices simultaneously, the A/DA PowerPlug-5 overcomes the problems of using batteries and one-accessory battery eliminators, providing a single, compact, light weight, cost-effective unit. Also provided on the rear panel are two grounded AC outlets to provide power for devices with built-in supplies.

The A/DA PowerPlug-5 is compatible with devices having external power jacks as well as devices that have no means for running from external power, as the power is brought into the sound modifier via its input jack. No special modifications to any device which may void its warranty, such as installing power jacks or transformers, are necessary. Also, there is no longer any need for having different different battery eliminators with their various power ratings and connectors that are only suited for the specific device that they were designed for.

A fused power supply for protection against AC line transients, a grounded AC cord for safety, short circuit proof input and output terminals ensure the musician that A/DA reliability is designed into the PowerPlug-5. Housed in a rugged attractive ABS enclosure, the PowerPlug-5 also includes an LED power indicator. The PowerPlug-5 comes complete with five 18" stereo cords and four 18" mono cords, a user's manual, and a one year parts and labor warranty. Suggested retail price is $149.98. For more information, contact: Analog/ Digital Associates, 2316 Fourth St., Berkeley, CA 94710.

cord analyzers



The CA-2 and CA-3 Cord Analyzers have been designed to provide a fast, accurate means of testing the two most commonly used audio cables. The CA-2 checks guitar cords (or any cord with ¼" phone plugs on each end). Three LEDs give a visual indication of the cords condition as it checks for continuity and shorts. Housed in a compact bakelite enclosure and powered by a single 9 volt battery, the CA-2 lists for only $19.95. The CA-3 checks microphone cords with three conductor XLR connectors on both ends (one male and one female). Five LEDs are provided to give instantaneous readout of the cords condition. Each of the three lines is checked for continuity and shorts to any other line. The CA-3 operates on two 9 volt batteries, is housed in a black bakelite enclosure, and lists for $29.95. For more information, contact: Polyfusion, 160 Sugg Road, Buffalo, NY 14225.

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Home Studio Techniques

by Brian Folkes

MAXIMUM MUSIC from MINIMUM EQUIPMENT...

One of the main reasons musicians/composers purchase synthesizers is to create and realize their own compositions. But unless you own a multi-track studio or a computerized unit with multi-tracking capabilties, you may still have a problem. Hopefully this article will help alleviate some of the problems. The single biggest problem facing a potential electronic music composition is noise. Equipment noise...tape noise...noisy noise. Unless you have an inheritance or invested in Xerox during their first year, you probably are limited to the quality of equipment you can get your hands on. I'm going to show how some of the noise can be eliminated. This, coupled with recording technique, a little music theory, and some good maintenance habits can go a long way in helping you make cleaner sounding recordings.

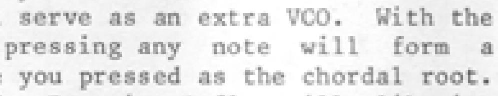
Composing

Before we get into multi-tracking itself, let's cover a few basics. In addition to tape recording techniques, much of the clarity of your recorded work will come from your compositional technique. For a recording to sound full you must put as much on each track as possible unless, of course, the piece or section is intended to sound sparse and uncluttered. Uncluttered. Now there's a key word. On one hand I suggest putting as much into the recording as possible and then I contradict myself by saying to keep all of the musical lines and ideas clear, distinct, and uncluttered. All of this means that, as in symphonic writing, a melody, counterpoint, rhythmic line along with harmonic support and a bass line are generally all that is being heard musically at one time. However, several instruments may be playing the same parts; this is known as compositional 'voice doubling'. The counterpoint line, for example, may be scored for clarinet, oboe, bassoon and cello. By applying this technique to electronic music, you can have the same line or idea occuring in multiple VCOs to add to the fullness of the line in much the same way a composer will score a melody and employ several instruments to play it. Melody, counterpoint lines and bass lines are easy. Some oscillators have simultaneous waveform outputs to add to the fullness; additional VCOs can be offset by an interval. However, harmonic background presents a problem for which two solutions come to mind. The first is a compositional answer; write independent lines that imply harmonic progression much the same way Bach wrote 2 and 3 part inventions. The other is with equipment; with this method you are only limited by the type of synthesizer and other musical instruments you own, plus keyboard ability if you will use more than one keyboard. Since most of us still don't have polyphonic synthesizers, it is a problem using VCOs to lay down 3 or 4 part harmony. It takes too many tracks. If you have an electronic piano, organ or guitar, you can interface it to your synthesizer for producing full chords. Better yet, many of the newer organ/synthesizer hybrids include gate outputs specifically for external processing. If all you own is a synthesizer with 2 or 3 VCOs and a multi-modal VCF or two, you are on your way to establishing a harmony background for your work.

FIGURE 1

Before we get into the actual mechanics behind this let's cover a little bit of music theory. Take a look at the C major scale and chords in figure 1. Three chords are major sounding: C, F, and G (these chords are composed of a minor 3rd over an interval of a major 3rd). Three of the remaining chords are minor: Dm, Em, and Am (a major 3rd over a minor 3rd). The last chord is diminished: B (minor 3rd over a minor 3rd). In actual chordal functions there are 3 primary chords, the others are variations. Refering back to figure 1, the iii (Em) and vi (Am) chords are similar in function to tonic I (C). The iv (F) is similar to ii (Dm) and vii (B dim.) is related to V (G). By therefor restricting your use to only 3 chords- I, ii and V- you have a very basic harmonic structure to work with. If you are using tracking VCOs, however, you will have to use, for example, all major chords so it is no problem substituting IV for ii. (Ah yes...the old I, IV, V progression.) Now these chords tend to get pretty bland and colorless after a while. But since most popular music is based on these 3 chords they will serve a useful purpose. Now by using a sequencer programmed for root progressions and tied to 3 VCOs tuned to a major chord, we will have a recurring harmony pattern. By setting some stages to minimum, we can create rests and the run will be syncopated. If you think that I-IV-V is boring, wait till you hear them over and over and over and over...there is another solution. This time use a keyboard tied to 3 VCOs still tuned to a major chord. If you only have 2 oscillators it's no problem. The most easily deleted note of any chord is the fifth above the root. In harmony, due to the overtone series, the interval of the fifth is always implied whether it's there or not. Obviously it's better to have it there. (Back to fullness again.) However it can be left out. Alternatively you could use your multi-modal filter to accentuate that harmonic in the VCO that is tuned to the root. With your filter set to oscillate and track the keyboard, this will serve as an extra VCO. With the VCOs tuned to a major chord, pressing any note will form a major chord with the note you pressed as the chordal root. Hit C and out comes a C chord. Pressing A flat will likewise result in an A flat chord. Fantastic!!! As long as you are writing only with major chords you've got it made; however, most composers don't write that way. Again, a little music theory helps. If you've ever arranged music for 4 voices or have had to write Bach-type chorales, then you are aware of voice doubling, or adding notes that are not specifically in the basic chord structure. A tonic C chord being scored for 4 voices could have the C doubled, or you could add a B to make the chord a C major 7, or add an A to make a C6 chord. With a minor chord you can always add the natural 7th, so a Dm could be scored with an added C to make the entire chord Dm7. The upper 3 notes (f, a & c) spell out an F major chord as depicted in figure 2. Therefore, since a minor chord can

FIGURE 2



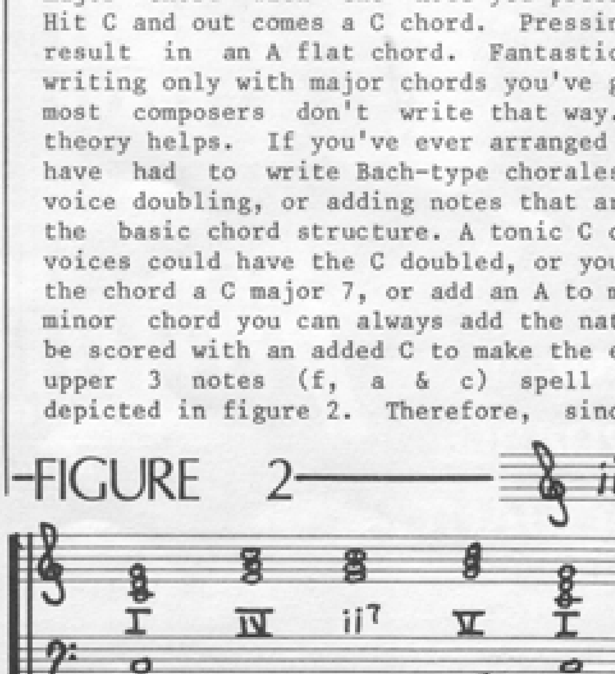


FIGURE 3

always have a 7th added, and have the last 3 notes spell out a major chord, then one track for the harmony part will consist of major chords and the second track will have the bass line. This bass line will then determine whether the harmony on the first track is major or minor. If you have two keyboards, one will have the bass patch while the other will have a harmony part. If the progression is I-IV-ii-V-I, play C-F-F-G-C on the chord programmed keyboard. The bass line will then be C-F-D-G-C as written in figure 3.

Let me backtrack a bit. The method used for major and minor chords works great. But...not all chords are just major or minor. There are diminished 7ths, major and minor 7ths, augmented, half diminished, 9ths, 11ths, major 7ths with sharp 9ths and flat 11ths, etc. The only way to get these chords would be to have more VCOs tracking at the interval that you want. When the more complex chords come up open the mixer input they're in and lower the note(s) you don't need.

Recording

Now that we have a way to establish harmonies and bass lines in just 2 tracks, let's try to put something together. There are 2 methods for overdubbing. The most preferable is a tape recorder with at least 4 tracks that can be synchronized with each other. Such a unit would be the TEAC A3440. However, after you've spent your allowance on synthesizers you might not have the $1600 for the TEAC. Well don't give up. The other method for multi-tracking is available on most stereo machines. That is called sound on sound (S.O.S.). The trouble with S.O.S. is that it's noisy. The Sound on Sound technique is based on bouncing your recorded track back and forth between the left and right channels adding a new musical line with each channel transfer. Practically speaking, the first voice is recorded on the left channel. When satisfied with your performance and volume level, the tape is rewound to the starting point. The recorded part is played back and recorded along with a new part onto the right channel. Voice one is now on both channels but voice 2 is only on the right channel. Also, the parts on the right channel have been time delayed due to the distance between the record and playback heads during the transfer process. Thus, the tracks can never be played back as stereo; the last channel recorded will give you your composite mono mix. Now voices 1 and 2 are played back and recorded along with another voice onto the left channel. The procedure is repeated until you finish the piece or can't stand the noise, which ever comes first. Since every voice is re-recorded several times, the noise level and quality loss due to every track transfer builds quite rapidly. How many tracks can be built up is dependent upon how clean your machine is, the quality of tape used, plus other factors to be covered later under tape maintenance. Most 3 head stereo tape recorders come equipped with a provision for S.O.S., generally a switch marked 1-4 & 2-3. (If your recorder doesn't have this provision, both SONY and TEAC make an adapter for most recorders so they can do S.O.S.) On my tape recorder- AKAI 4000DB-the output from the synthesizer mixer connects to my left line input on the back of the recorder. The monitor switch is set to source and S.O.S. is on. The volume level is set now on the V.U. meters. Very Important!!!!!! Volume levels are the key to good clean recordings. Since the sound will degenerate every time you bounce the tracks, record the least important parts first; save the most important parts for last. If the first track peaks on the V.U. meters at 0db, then with each subsequent voice set back the volume control somewhat. This way a good blend will occur plus the volume won't end up distorting your later voices that are added. With good levels, a clean machine and using a brand of tape recommended for your recorder, compositions of 5 to 8 tracks are possible without having the noise override your composition. Once your piece is recorded and edited (covered in detail later) you should master your tape onto another reel to reel or quality cassette deck. Since you'll be adding another generation, it'll help if your master machine has a dolby noise reduction unit. Another helpful hint especially for S.O.S. recordings- always record at your highest tape speed. This uses more tape, but now is not the time to be frugal with tape. Using the higher speed (most stereo recorders will have 7½ i.p.s. as their top speed, some even have 15 i.p.s.) will increase your signal to noise ratio and will give you a wider frequency respose and less wow and flutter.

The ultimate semi-pro recorder to use is a four track synchronized reel to reel. A machine such as the TEAC A3440 coupled with a 6 to 8 input 4 output mixing board will result in excellent quality tape recordings. To have a quad deck is not enough. The recorder must be able to play back any of the 4 tracks with the record head while other material is being recorded at the same time. This function is marketed under several trade names such as Simul-Sync, Sel-Sync, and Multi-Sync. In these modes, the record head is used temporarily as a playback head to allow "time-aligned" monitoring of previously recorded tracks. The record head has a limited playback frequency response so the highs are lost, but when the sync mode is disengaged the normal playback head with a full frequency response is engaged. As with S.O.S. the first voice is recorded alone on track 1. The tape is rewound and channel 1 is then placed in sync and channel 2 is ready to record the second voice. This procedure will result in 4 perfectly synchronized tracks of music, all first generation. With at least a 4 input mixer, each track has its own volume control during mixdown. The beauty of this recorder is that you don't have to stop with just 4 tracks. Ten tracks of material can be recorded with 4 tracks being first generation and the other tracks only second generation. Remember, 10 tracks using S.O.S. will make the first track 10 generations away, the second track 9 generations away, and so on. To achieve 10 tracks on your four track machine, record your first three voices on the first three channels. With all three tracks still in sync, set the output assignment channels on the mixer so all three tracks will mix down to channel 4. Set up your 4th voice to go into channel 4 also. Channel 4 will now consist of voices 1, 2 and 3 (all second generation) plus voice 4 (first generation). Now put channel 4 into sync and take the first 3 channels out. Record voice 5 onto channel 1, and voice 6 on channel 2. Set the output channel assignments for tracks 1 and 2 to mix down to channel 3 along with your new input for voice 7. Now channel 3 will have voices 5 and 6 just second generation and voice 7 will be first. By repeating this procedure you'll end up with channel one consisting of the 10th track of material at first generation and channel 2 will have voice 8 at second and voice 9 at first generations. As with S.O.S., record the less critical parts on the tracks that will become second generations. Also it is still important to set proper levels. Since you now have up to four different voices on one volume control and the voices will be of different generations, it is imperitive that the levels of the individual parts be correct. Once mixed to another channel with more material, only the overall volume of that channel can be changed. If 10 tracks isn't enough for you, further submixes can yield 20 voices, 4 of which will be first, 6 will be second and 10 will be third generations. Again, remember you'll have even less control over the individual volume than before. With 20 tracks, 10 of these will be on one channel and volume control alone!

Arranging

Now that we have some tools and a way of multi-tracking, lets put it all together. Figures 4 and 5 are excerpts from a score to the "Overture From Tommy" to be realized on the synthesizer. It is scored in these two examples as it would

FIGURE 4

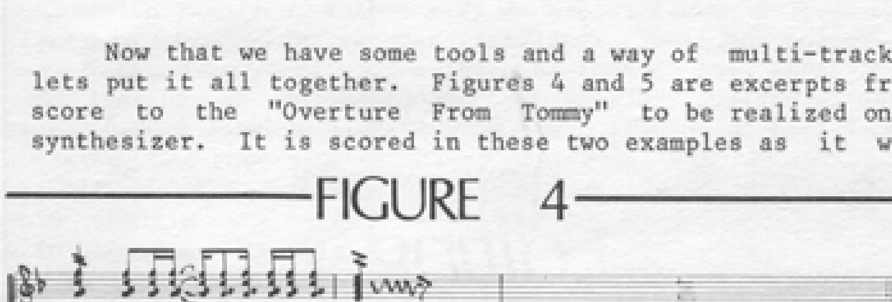
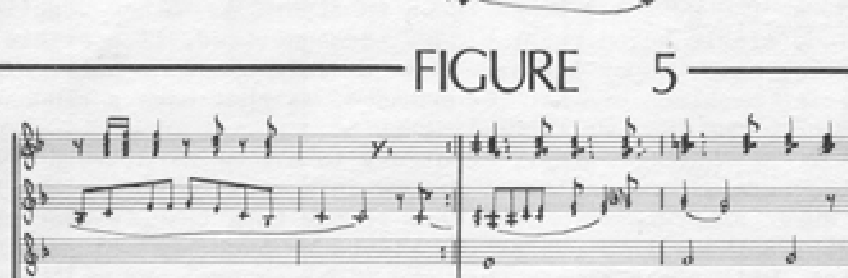


FIGURE 5



be scored for conventional instruments. Figures 6 and 7 are rescored and broken into components necessary to multi-track with. Compare figure 4 to 6. The only moving voice in the rhythmic chord pattern is the resolution of the suspended fourth (F to E in the C chord). Two VCOs are used. In the first track the VCOs are tuned to a fourth (C and G); in the second track the VCOs are tuned to octaves. The other two tracks consist of only one VCO. Which voice should be recorded first? If using a 4 channel deck, it doesn't matter since only four tracks are being laid down. If a S.O.S. type recorder is being used, it does matter. Since the syncopated pattern is established before the other voices are added, the top two parts are recorded first. System one will be recorded first since the C note is used in two other parts plus the G note is less important. Before starting to record I have found that working with a metronome to establish tempo works best. When the tape starts to roll I play four notes in time to the metronome before I play the downbeat. This serves as a reference of tempo and downbeat for recording future tracks. The four beats are later cut out when the piece is assembled and edited.

FIGURE 6

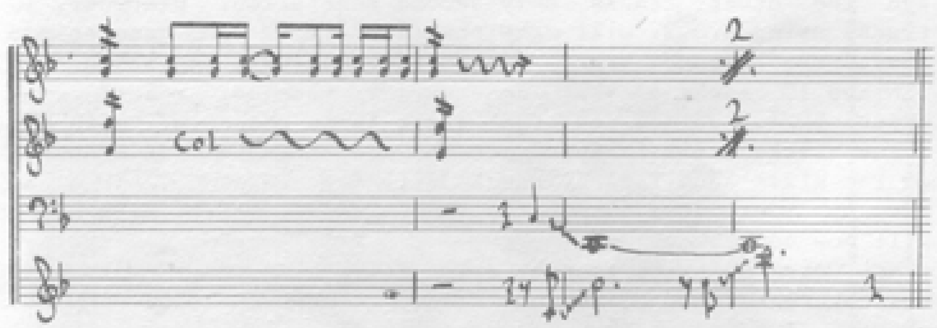


FIGURE 7

The two examples of figure 5 and 7 offer a different situation. First, notice the difference in the way the background harmonies are voiced. Figure 5 is more like the way chords would be structured for conventional instruments. Figure 7 is the way it has to be structured by the method outlined earlier. Notice that the harmonies are all major. Only the bass line will determine whether the resultant harmony is major or minor. Also, figure 7 is scored for three tracks with capabilities for adding three more optional parts to be added first since they are less critical. If S.O.S. is being used I would record in this order: optional harmony, counterpoint and lead. Then bass, harmony and lead. If a four channel reel to reel is used, record bass and harmony on channel 1, optional harmony and counterpoint on channel 2 optional lead on channel 3 and main lead on channel 4.

Editing

The final topic to cover is putting everything together through editing. To edit you have to have a splicing block such as the Edit-All block shown in figure 8. Other supplies include single edged razor blades (demagnetized, if possible), splicing tape, paper leader, and a soft lead pencil. The splicing machines are not recommended, as they have a tendency to mangle the tape at the splice point.

I have found that, contrary to all opinion, 90 degree splice cuts as opposed to 45 degree cuts work best for stereo or quad recordings. Consider: as the tape travels over the heads and the splice passes the playback head, a 45 degree splice will first appear over the right channel and will move toward the left channel. A slight panning effect is heard which is most noticable at slower speeds. This sound, while not that objectionable, nevertheless will call attention to the fact that a splice has been made. If the cut is at 90 degrees, the right and left channels on the tape will reach the playback head at the same time. I have never experienced any popping or clicking noises since editing this way.

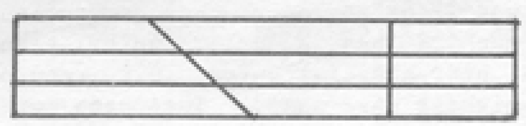
Where to splice is the next problem. The answer is entirely dependent on what it is you're editing. Listen first to the tape. When you get to the part that is to be cut, put the recorder in the "pause" or "edit" mode. This will leave the playback heads contacting the tape so you can exactly locate the cut by rocking the reels back and fourth until you hear the beginning of the sound to be edited. Rock the tape back to just where the sound will begin. Carefully mark the tape with your pencil; don't mark the heads or use enough pressure to deform the tape or misalign the head. Repeat this procedure for the end of the edited section. Remove the tape from the machine and place the center of the mark in the center of the blade track you have selected to use on the splicing block. Make a good clean cut; repeat the process for the second cut. Remove the scrap tape, butt together the remaining ends, and apply a 3/8" length of ¼" splicing tape to the cut line. Remove air bubbles with your fingernail. When all looks secure, grab the tape by both ends and snap the tape out of the block. This won't hurt the tape and if the splice survives this shock it'll survive just about anything. Replay the spliced tape to make sure it was done right. Always save the cut out tape in case a mistake has been made.

Some things to watch out for when editing: Unless there is to be a change in volume from splice to splice, watch your VU meter to make sure the overall volume doesn't change during the splice. Also, always use the same type of recording tape throughout the piece. Tapes with different biases and quality will result in noticable splices. If you want silence between passages in your piece, don't use leader tape. Since leader tape has no magnetic qualities, tape noise will drop out during the leader section, calling attention to the splice. Rather, use blank tape for silence. This will maintain a constant noise floor (which most people will psycho-acoustically ignore) throughout the composition. With these things in mind, splices will be less obvious and will sound

Maintenance

These methods will help reduce the number of tracks to be used with the useful side effects of fewer generations, less signal loss, and less noise. However, improperly maintaining your tape equipment will dramatically increase your noise and distortion level. Two things that must be done regularly are cleaning and demagnetizing your heads. The instruction manual for your recorder will suggest how often to clean the heads and rollers. I find it best to clean them every time you finish a major session. Make sure you use denatured alcohol or a solution specifically for use on recording heads. Isopropyl or colored solutions leave deposits and film on the heads. Moisten a cotton swab with the alcohol and rub the entire surface of all the heads. Also clean all parts that the tape comes in contact with, including the capstan and pinch roller. Since the magnetic particles on the tape can magnetize the heads, it's also a good idea to get a head demagnetizer. Turn off the recorder and, starting with the erase head, move the demagnetizer in a tight circular motion past all the tape heads and metal parts in the tape path. Pull the demagnetizer slowly away from the heads, widening the circular motion until you are several feet from the recorder.

FIGURE 8



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A CONSTRUCTION PROJECT

PERCUSSIVE NOISE VOICE

BY: JOHN BLACET

Noise is something that engineers spend a lot of time trying to eliminate in audio circuits. At the same time noise is a very natural part of some musical sounds and nearly every synthesizer has some sort of simple noise module. It turns out that there are a lot of different types of noise, classified according to their spectral energy distribution. Most synthesists are familiar with white, pink, and 1/F noise. These are listed in order of decreasing higher frequency content. Further on down, we encounter brown noise.

Another way of differentiating between types of noise is by their degree of autocorrelation. This merely means how "together" they are, how the noise fluctuations at a given point relate to the previous fluctuations, as opposed to their degree of randomness. White noise is completely random and brown is highly correlated with the others in between. As we go from white noise downward, large changes tend to occur less often than small changes. Very Zen.

All these types of noises occur in nature. White and pink sounds are obvious; surf, rain, wind, etc. 1/F noise is interesting because its variations are typical of both human and natural events. Music based on 1/F noise tends to be the most interesting of "random" music. Both events and music can be characterized as having more small changes related to each other than large unexpected changes. Although this area does not seem to be well understood, it may be that 1/F variations will turn out to be important in the timbral qualities of electronic instruments.

circuit analysis

This particular noise voice came out of working with Texas Instruments sound chip, the SN76477. Although this is not a precision device and has some features implemented in a non-standard way, it is very inexpensive and turns out to be just fine for some applications. There is after all, no other chip with so much sound generating capacity for three dollars. A look at Fig. 1 will give you an idea of the chip functions.

All of the chip functions are used, resulting in a combination noise and tone source. Controls include: noise clock frequency, noise filter cutoff, super low freq. oscillator speed, VCO frequency and envelope decay rate. The envelope can be triggered by a rising edge or a pushbutton to a positive supply.

There are a number of interesting circuit "tricks" that should be pointed out. An external noise clock is used instead of the limited range internal clock (pin 3). The noise gererator itself is the pseudo-random type constructed with shift registers. The filtered noise output is tapped off pin 6 and applied to the VCO voltage control pin 16. This results in a different sound than the usual route through the mixer, which is a digital AND gate rather than an analog mixer.

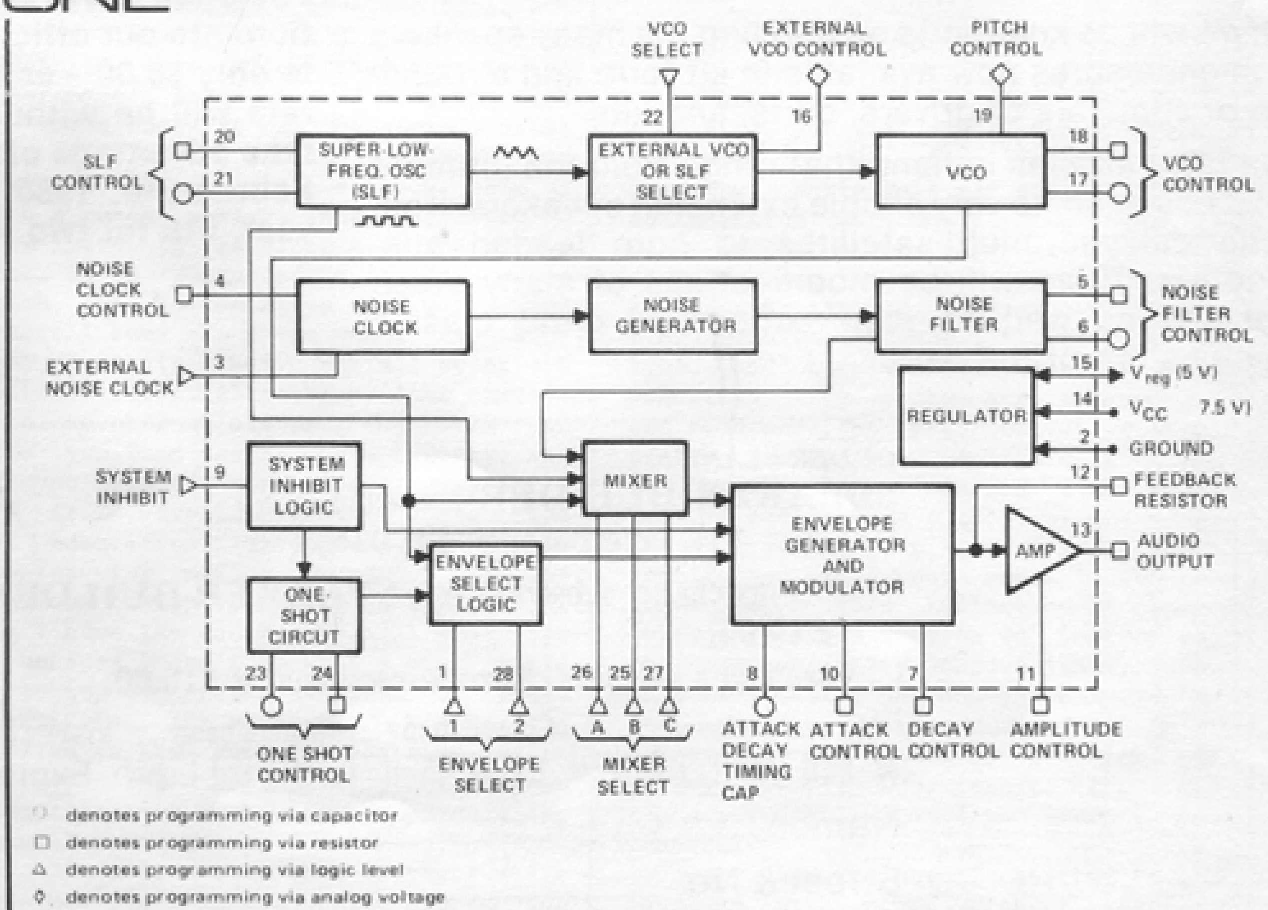
In a like manner, the SLF oscillator triangle wave output is picked up at pin 21 and applied to the VCO pitch control (pin 19). This pin controls the duty cycle of the VCO pulse wave output. Since the noise output is also present here, a nice phaseshift type sound results. A diode at pin 21 is connected to a one-shot that is triggered each time the voice is triggered. This discharges the SLF capacitor and provides a synchronized sweep. This in turn means the sound will be the same each time.

The same one-shot provides a brief pulse that is capacitively coupled to VCO input pin 16. This causes a brief frequency offset that is a chief psychoacoustic clue common to percussive devices.

The triggering circuitry uses a one-shot and an inverter to produce a falling edge which is what the chip circuitry responds to. For the VCO frequency pot, a cermet type should be used for maximum temperature stability.

The power supply may be a nine volt battery, a regulated 5 volt or a 15 volt supply with a dropping resistor. Note that 10 volts is the maximum safe input

ONE



functional block diagram

Reprinted by Permission from TI Bulletin # DL-S12612 (pin 14). If a 5 volt supply is used, connect to pin 15. This pin becomes a 5 volt source when pin 14 is used as a power input. All external voltages should be sourced from pin 15.

The full operation of this chip is covered in a lengthly bulletin from TI. Radio Shack also sells the chip and has a similar application note. One thing to be aware of here is a difference in the minimum resistance value at a number of inputs. TI lists 7.5K minimum, while Radio Shack shows 2.7K on one diagram. This could result in excessive current flow and reduced chip reliability. In our application 10K is used, as this is a common value.

applications

What we end up with is a rather interesting matrix of noise and tone based sounds that sound a lot more complicated than the tiny PC board or cost would indicate. In fact, using conventional modules, I found that I needed a noise source, a VCO, a VCF, and a VCA just to duplicate some of the sounds--lots of tied up panel space.

What kind of sounds are available? All the standard sorts of wind, surf, rain, explosions, earthquakes are there of course; plus all the things in between. When you start to add tone to it all, the complexity increases rapidly, resulting in cymbal and gong-like sounds, spacey 1/F modulated tones, and so on. It becomes difficult to describe these sounds in terms of familiar instruments. The comparisons are valid and helpful as a kind of map, but we are not out to just duplicate existing sounds but to use the potential of electronics to create new ones.

The original intent of this module was as an addition to a drum set. If you go this route, it's easy and cheap to go multiple units. Calculator keys and other low bounce push-buttons are easy to convert into drumstick activated triggers. Use trimpots on the boards for even greater economy. One incarnation has 20 modules and is a strange sort of instrument all by itself. Very Harry Partch!

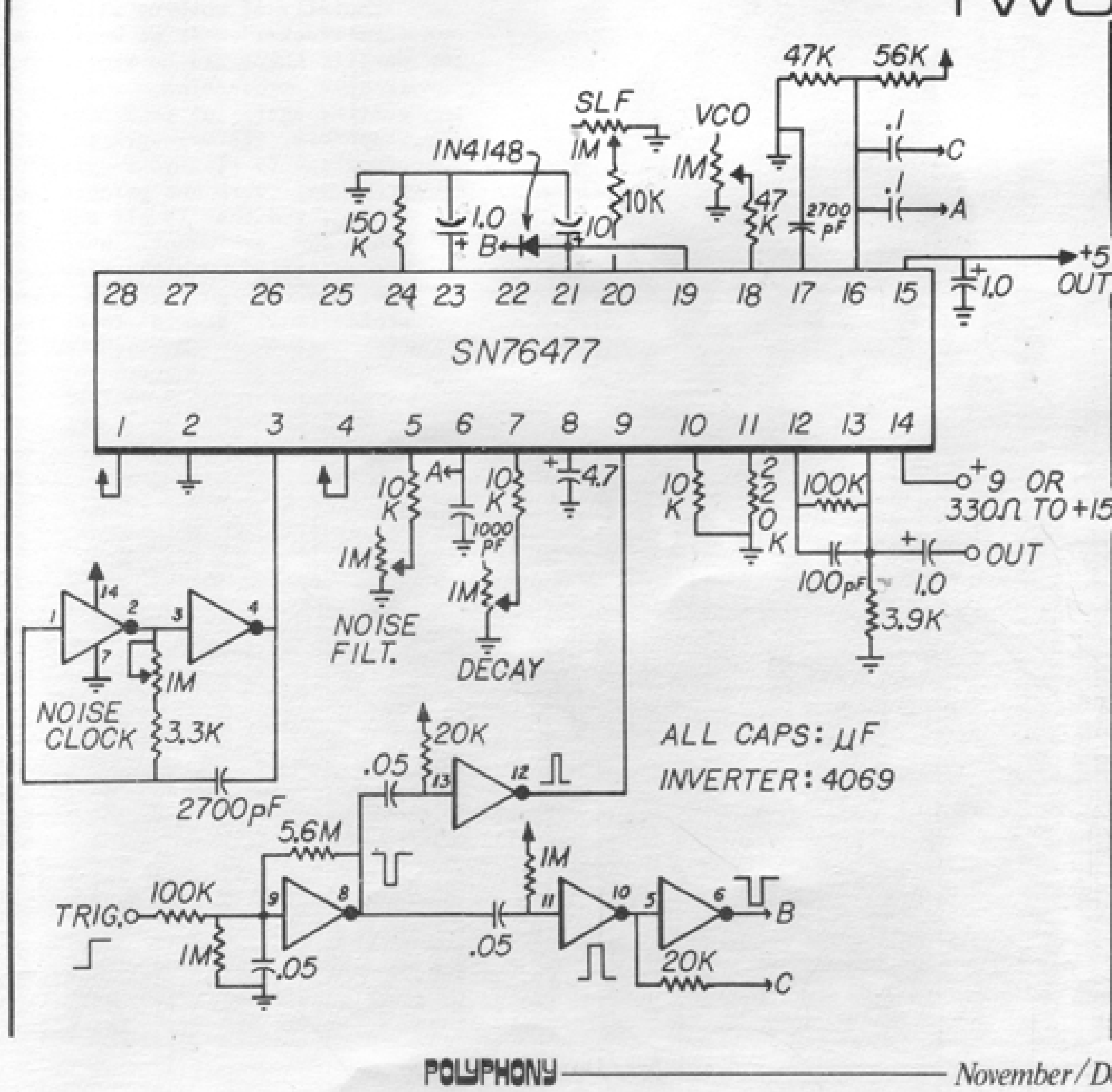
As a regular synthesizer module, you may want to put an attenuator on the output so it can be used as a triggered modulation source for voltage controlled blocks. Try a little with a VCO and trigger via the keyboard. This adds a little controlled distortion back into the waveform for more life. In this case, the character of the distortion can be varied considerably.

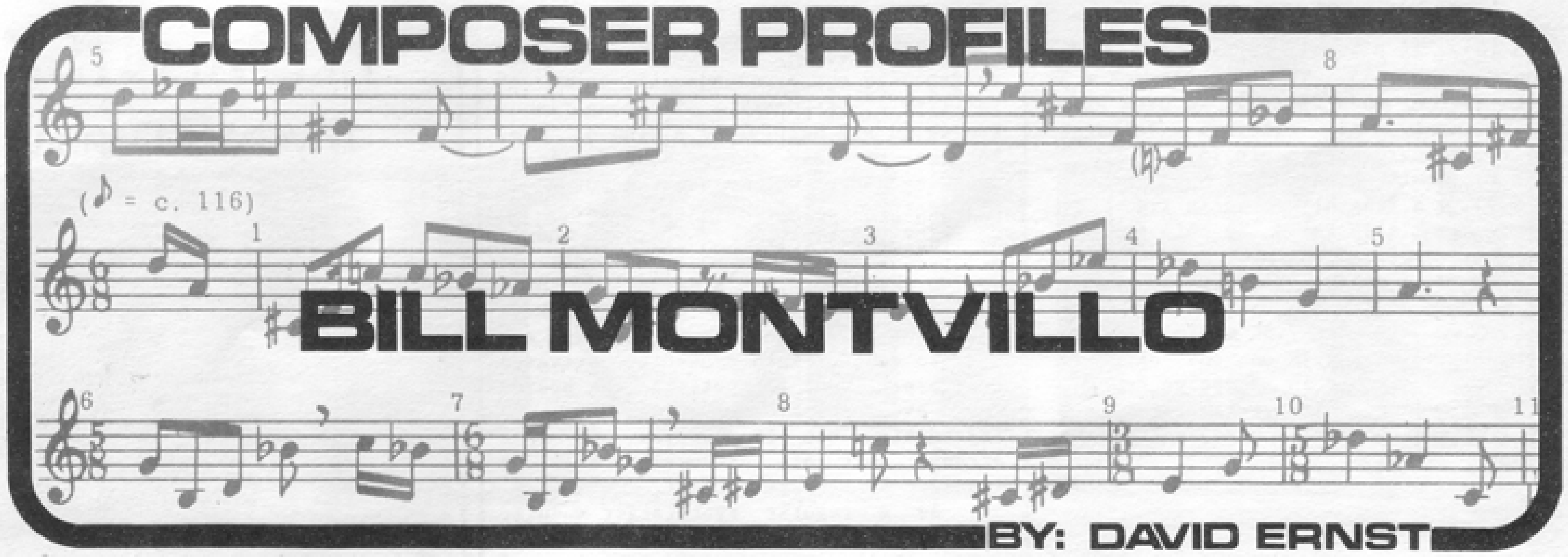
A third alternative is to put the board in a small box, power via a nine volt battery and use as a hand held percussion device tambourine, etc.

In the future, we will look at more interesting uses for the SN76477. TI also has some sound chips designed for computer interface and applications for those will be discussed.

NOTE: A complete kit for the noise voice is available from Blacet Music. Your choice of on board trimmers with easy to adjust shaft, or regular pots: $19.00 ppd (USA, Canada, Mexico).

TWO





COMPOSER PROFILES

BILL MONTVILLO

BY: DAVID ERNST

In this issue we are going to learn about a very lucrative profession -- production of retail music, or commercial jingles. Although this field is highly competitive it is well-suited for the home, i.e. small scale synthesist; local radio and/or TV spots are particularly promising. We will obtain first-hand information from Bill Montvillo (b. 1953), a talented New York composer, arranger, organist, synthesist, and conductor. And in our effort to present a true picture of the retail music industry, this article is divided into two sections: prerequisites and procedures for jingle production. As in all fields of endeavor there are no short-cuts, no substitutes for hard work and study. If you feel that this is the profession for you, then this article should help you to get started.

prerequisites for jingle production

"Buy a synthesizer and become a musician instantly?" "This is nonsense," says Bill. There are three major satisfied before one can really expect to produce successful jingles. First is the question of a composer's musical background. Bill has been around long enough to observe that "since competition in this field is fantastic I have never met a musician who didn't have his act together." This means that you must be able to read and to write music, and that you must possess at least an elementary knowledge of basic music theory and orchestration -- even if dealing with synthesizers. Performance ability (especially on keyboards) is also desirable, as well as familiarity with diverse musical styles, e.g. Baroque, Classical, Romantic, Jazz, Folk etc. Listen to all types of music and try to imitate stylistic mannerisms with respect to melody, harmony, rhythm, and orchestration. Analyze the music of other composers to discover their techniques and special skills, and then use this information as a point of departure to develop your own ideas. Many books are available to assist you, and remember that a solid musical background will help you to satisfy the demands of your clients---more about this later.

PHOTO BY LORI SEID

The second requirement involves a composer's technical background including topics as microphone placement, recording studio techniques, synthesizer operation, and basic acoustics. You must develop good recording and mixing procedures to maintain high sound quality with minimal noise. Again, you are serving the needs of a client who expects professional results---both musically and technically. Take the time to listen carefully to the fidelity, as well as the musical content, of radio and television commercials. You will soon agree with Bill when he says that "Now-a-days, simple beeps, whistles, and bells are not sufficient for network distribution." The necessity of technical perfection is manifest most clearly in the small, home-studio situation, where you will be in competition (there's that word again!) with professional 16- and 24-track studios.

This leads us to the final, although not the least important, prerequisite for jingle production---equipment. Bill, along with most professional musicians, encourage the purchase of "the best equipment that you can afford; and then learn how to use it." A minimal do-it-yourself home studio should contain a polyphonic synthesizer, 4-track tape recorder, microphone(s), preamplifiers, at least two graphic equalizers, decent playback system, mixer, cassette recorder, and tape editing materials. Other useful equipment includes a ½-track tape recorder, string synthesizer, electric piano and/or organ, effects devices (phase-shifter, echoplex etc.), headphones, stop-watch, pitch-pipe, metronome, and electronic percussion unit. If you intend to be a one-man-show you will probably need the extra keyboards and electronic percussion, and Bill advises you to "play as much as possible in real-time to avoid dumping, especially if working with only four-tracks." As we mentioned earlier there are no short-cuts in this profession, and good quality equipment is expensive. Therefore, it may work to your advantage to plan your jingles at home, work out patches and timing, and test it all out on your own equipment. When you are satisfied with the flow and feel you can go into a more professional studio to do the finished work. If you have an above average home studio, you should be able to even lay down some background sub-mixes or cue tracks which can then be transferred to the larger multitrack when you go to the big studio. Regardless of the method that you choose, the next portion of this article will prepare you for the actual production of jingles.

procedures for jingle production

Bill has given us a detailed outline of how to go about producing retail music, and you may be surprised at the amount of research required. We are now speaking of local spots where, according to Bill, "the composer is under pressure to produce quickly and efficiently. All clients want to have heard the jingle yesterday." This is why good equipment, along with a solid musical and technical background, are necessary. You must always remember that you are writing music for a client who has a product to sell, so Bill's advise is to "find out as much as possible about the client and his product, in addition to previous jingles and advertisements.

Personal contact with your client will help to keep you on the right track so that you will be able to keep the musical style consistant with the product and the image that the client wants to project." You must not only have a clear idea of what is expected of you, but it is also helpful if you know the product's advertisement history.

After the background research is completed you are ready to begin laying out the overall format of the jingle. First, you must know the length of the spot---usually 30 or 60 seconds, then you will be able to map out voice and music sections. It should look similar to the example in Figure 1. Later in the article Bill will explain how the use of click tracks facilitate the solving of timing problems. The next step is to know how many tracks will be available for recording, e.g. 4, 8, 16 etc.; keep one of these channels open for the click track. Careful planning enables you to work quickly, efficiently, and economically.

Now we come to the musical style. As Bill mentioned earlier, the musical style should be consistent with the product, and this is where a vast musical background is invaluable. If a march is called for, listen to records and look at musical scores of marches. Observe how the melodies are constructed, the nature of the harmonies, the rhythmic organization, and the types of instruments used. This applies to all musical styles, and you will frequently get ideas from listening to 'classical' music. Also, know the role of your music---background, foreground, or alternating between the two. After this is accomplished, write the music for piano; it will be orchestrated later. When the music is finished get a precise timing by using a metronome and stop-watch. By referring to the format sketch (see Figure 1) you will know the exact amount of music needed. Be sure to clear the text with your client if you had to write your own, for he will probably make suggestions. Bill also suggests that you make a cassette of this piano/vocal arrangement and play it for your client (personal contact). It is best to make changes prior to the final recording, and perhaps your client will give you some ideas for future reference.

Bill also suggests that, since we are thinking of local stations, you should consider the type of radio station that will air the end product. The musical style of your jingle should be compatible with the type of music played on that station. For instance, if your jingle is in the Baroque idiom and it is played on a rock station, it will probably not be very successful.

If all goes well you are now ready to orchestrate the jingle. The first question concerns the instruments---synthesizer or acoustic. If you adopt the one-man-show concept you will probably go for the former; it is certainly less expensive than paying studio musicians. But don't rule out the possibility of having acoustic instruments, even one or two, just to add an extra 'something' to the jingle. This choice is up to you and your client. Many composers, including Bill, have found that it is not said; this decision is up to you.

Regardless of the instrumentation that you choose there are a few items that Bill has mentioned that will be of great assistance with respect to the overall texture of the end product. Because of the relatively high energy of oscillators as compared to that of acoustic instruments, it is often wise to keep synthesizer lines as widely spaced as possible. When recording two synthesizer lines on a single track Bill prefers to keep them in different frequency ranges, e.g. soprano and bass. This technique is useful if you need to conserve tracks, yet it permits discrete equalization of both registers. Synthesized orchestrations generally require thinner textures than acoustic arrangements, so be sure to allow yourself the flexibility to make last-minute changes.

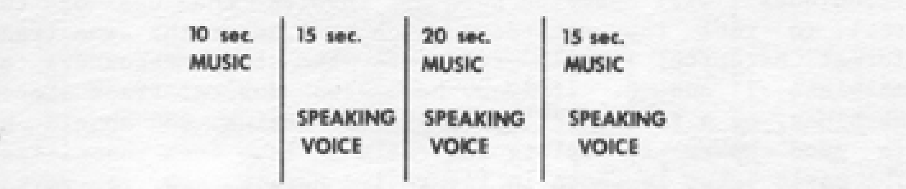
Octave doublings are also useful in achieving a particular acoustic effect---'dry' or 'wet'. In current rock and disco music the drum is often boosted at about 1k Hz. to add clarity. If you are using synthesized drums it is esential to obtain the 'right' sound, which usually involves playing around with the equalization. A 'wet' percussion sound can be obtained via slight reverberation or echo, but keep the modifications minimal unless you want a specific effect.

Bill has emphasized the need for documentation of all aspects of the orchestration and recording stages. Keep records of patches, equalization and recording levels, and track content. Try to get the sound that you want before equalization, so that the equalizer will be used primarily to accentuate timbral subtleties. When finished you should have a complete musical score. You will be able to play around later with spare tracks to add 'sweetening'. The importance of all this is revealed in Bill's comment: "Treat the studio as an instrument, and write down everything that you do. You can't learn from your mistakes unless you remember them."

Now we come to the actual recording of your jingle. It might have seemed that we would never reach this stage, but all of the preliminary work will pay-off---especially if you will have to go into a professional studio and pay for recording time and possibly musicians. Also, if your client is going to pay for recording time you will want to be able to give a close estimate of expenses. Bill's advise for the recording stage is: "Start simple. Give your client exactly what he wants. Then, if you have time, prepare one or two alternate versions that incorporate more original ideas."

Decide on the recording sequences---one track at a time or everything at once. The one-man-show approach will demand the former, but if you are using only studio musicians the latter may be preferable. We will assume, however, that each track will be recorded separately. Record the click track first and, to give yourself some flexibility, make two or three click tracks for future use---revisions, alternate versions, etc. Since the click track functions as a metronome it is easily timed: MM ♩ = 60 = 1 second; MM ♩ = 90 = 2/3 second; MM ♩ = 120 = ½ second, etc. At least in the beginning try to keep your click track a simple multiple of standard metronome settings to avoid tedious arithmetic conversions. Rhythm tracks (percussion, bass, guitar, piano) are generally recorded next, and it is wise to add these to the extra click tracks recorded previously. Try to put each instrument on a separate track to make future revisions easy, but this may not be possible if you are using only a 4-track recorder. In this event you will want to play two parts simultaneously (e.g. percussion and bass) and mix them on to a single channel. On the other hand, if you do have eight or more tracks available don't feel that you must use all tracks. Remember Bill's advice: "simple is best."

1



2

After finishing the rhythm tracks, record background fills, followed by vocal and lead instrumental parts. To help keep your place while layering parts, Bill suggests liberal use of scratch tracks---piano or vocal cues recorded on empty tracks and eventually erased. Scratch tracks often save time while recording intricate passages or instrumental entrances preceded by long rests. Another hint offered by Bill involves ritards and accelerandi. These are precisely timed and work with the click track as illustrated in Figure 2. As shown, the ritard is written out as specific note durations so the exact duration is known, and the duration of the ritard (or accelerando) can be modified by changing individual note values. Working in this manner you will not have to worry about ritards and accelerandi being too long or too short; you have complete control over the duration of the jingle.

Bill's final advice is: "Be honest. Don't fool yourself. Don't record what you aren't satisfied with." If you take the time to obtain the sounds you were after in your preliminary work there should be no problem now. If you documented all your work properly, it will be easy to reproduce any part. Listen very carefully to the completed jingle. If you have a free track, and if you feel the music could be enhanced by special effects (echo, phasing etc.), you are free to experiment. You may even decide to make an alternate version, replacing one or more of the original tracks with new or modified material. This is perhaps the most creative stage of retail music production. Treat the studio as an instrument; take advantage of its numerous resources.

continued on page 20...

TAPE LOOP TECHNIQUES FOR MUSIC SYNTHESIS by Tim Fluharty TAPE LOOP TECHNIQUES FOR MUSIC SYNTHESIS by Tim Fluharty TAPE LOOP TECHNIQUES FOR MUSIC SYNTHESIS by Tim Fluharty TAPE LOOP TECHNIQUES FOR MUSIC SYNTHESIS

by Tim Fluharty

The use of tape loops is an often overlooked technique in music synthesis. This is understandable- there is only so much that can be done musically with a piece of magnetic tape with its ends spliced together in a physical loop configuration. Also, the splicing and editing processes required to fully utilize the discrete loop approach can be tricky and laborious (though anyone willing to take the time can master the technique).

Fortunately, there are other kinds of tape loops. The techniques I will describe here all involve the use of two reel to reel tape recorders which must be of the same track format. Hereafter I will refer to the tape recorders as machines 1 and 2. If you have two quarter track stereo machines, or a four track and a stereo machine, you should be in good shape to explore the realm of tape loop sonorities. The basic setup is shown in figure 1. Here's how it works: (for the sake of simplicity, assume that we're using only track one of each machine - temporarily disregard the other tracks) The input signal (synthesizer, guitar, microphone, etc.) enters the mic input of machine 1 and is recorded on the tape. After a variable delay (up to 10 seconds or more) the signal is played back by machine 2 and is fed back to the line input of machine 1 where it is mixed with whatever else is coming into machine 1. The process repeats continually -- much like tape echo, only at a much slower rate.

Setting Up

Position your tape recorders about 36 inches apart, horizontally, on a clean floor or large table. Make sure they are level and, if the machines are not the same height, put books under the shorter one to bring it up to the same horizontal plane as the other. Put a fresh reel of tape on machine 1 and string the tape, between the two machines, over to the takeup reel of machine 2. Plug your musical instrument into mic input 1 of machine 1. Patch the line output of machine 2 into the line input of machine 1 - if your machine has mic/ line mixing, this is ideal. Alternately, you may use a mixer connected to machine 1's line input; put your musical instrument signal and the line output from machine 2 (the feedback path) through the mixer directly into machine 1's line input (in this case, forget about the mic inputs). Adjust machine 1's recording level to not quite 0db for the maximum level of your musical input. If you are using a mixer, adjust it for a 50/50 mix of input and feedback signals; for mic/ line mixing, set the levels the same (50/50). Turn the output level control of machine 2 to minimum (completely off). Set machine 1 to record and machine 2 to playback (pause buttons are handy here), and turn both machines on simultaneously (make sure both machines are set to the same tape speed - 3 3/4 ips or 7½ ips - 15 ips is too fast to get a long delay time). With both machines running , start playing (anything) into machine 1 -- wait a few seconds, and then <u>slowly</u> bring up the level of machine 2's output. Keep playing - you should begin to hear your input signal repeating slowly and gradually fading out. To optimize the overall levels of the system, you will need to find the setting of machine 2's output level that is too high. At this point, the overall level will be getting louder with each successive repetition, and possibly distorting instead of fading out (decreasing in amplitude). Incrementally decrease machine 2's output level until the point is reached where the repetitions gradually decay and the distortion ceases. You have now found the optimum level for your tape and machines. Rewind your tape and thread it up again - you are now ready to go to work.

Using this process, it is possible for a single person with only a monophonic synthesizer (or guitar) to produce gigantic revolving sound textures - and with a good familiarity of the system and appropriate juggling of voicings and pitch ranges, these textures can often approach symphonic proportions. With a polyphonic input, textural possibilities obviously increase - but I would not advise attempting it until you get a good feeling for the possibilities using a monophonic input.

Now that we are familiar with the basics of the process, let's look at some of the many specialized and unique possibilities for sound processing and manipulation.

Feedback Processing

Note the box shown in the feedback signal path in figure 1 labeled "optional signal processing". You can easily insert any type of signal processing in the feedback path - phasing, flanging, filtering, balanced modulation, equalization, delay lines, etc. The effect of doing this in the feedback path is that the signal will be reprocessed with each successive repetition - it will continually re-phase, re-flange, etc. and build up unbelievable harmonic textures which constantly permutate and form new sonorities. I should also mention that natural chorusing will result from the same notes overlapping as you build textures with the repetitions, which in itself will create a rich fabric of sound. You will have to be careful about the system levels, since harmonically processing the feedback signal can produce a very "peaky" response, and some frequencies may accumulate too much energy and possibly distort. No problems should result if you are careful about your levels. If you have a compressor or limiter, you can insert it <u>after</u> the processing to even out the level a bit. Along the <u>same line</u>, if you put a graphic equalizer in the path, it's best to only <u>cut</u> frequencies, not boost, for if you boost you will almost certainly have frequency build up problems.

Up to now, we have been talking about a very flexible single channel system. It gets better.

Stereo & Quad Motion Generation

By setting up the connections shown in figure 2, we can easily also generate an automatic stereo panning effect - each successive repetition will alternate between the two channels. This effect was used with the guitar synthesizer in Terra Incognita from Synergy's Cords, and also on Delta 4 from the new Synergy Games album. Note that you still have the option of processing the two feedback paths to produce incredibly complex effects. If you had two four track recorders, you could use the same approach to make your signal sequentially appear in each of the four channels in a continuous circle.

Alternate Music Systems

What we're really talking about, when considering tape loops as a musical structural element, is an alternate music system - very different from other types of music. This system has it's own logic and flow, and it also has its own unique limitations. Obviously, when attempting to integrate slow repeats of melodies and harmonies into a musical form, the tempo and technique of playing will relate to the delay time/repeat frequency. You can't just play your favorite song into a loop setup. Well, you can, but it might sound a trifle chaotic. It's best to play with the setup for a while and get a feel for the possibilities. With delay times of ten seconds or more, melodies or melodic fragments can turn into complete compositions by building up layers of melodic variations and harmonic complements.

I think specialized music production systems are a valid alternative to other more "traditional" approaches. Very often composers are dissatisfied with existing systems and new processes become inevitable. At the very least, playing with a system like this can open up your awareness to how we perceive such subjective phenomena as melodic and harmonic progressions. I think this kind of research is an essential involvement for any serious synthesist.

Suggested Listening

No Pussyfooting by Robert Fripp and Brian Eno, 1973, Island Records HELP 16.

Evening Star by Robert Fripp and Brian Eno, 1975 Island Records HELP 22.

Discreet Music by Brian Eno, 1975 Obscure Records Obsc. No. 3, Obscure Records distributed by Island Records.

Peter Gabriel second solo album, 1978, Atlantic Records 19181, looped guitar is used as background on "Exposure".

Cords by Synergy, 1978, Passport Records PB 6000, stereo looping is used on "Terra Incognita".

FIG. ONE: BASIC TAPE LOOP

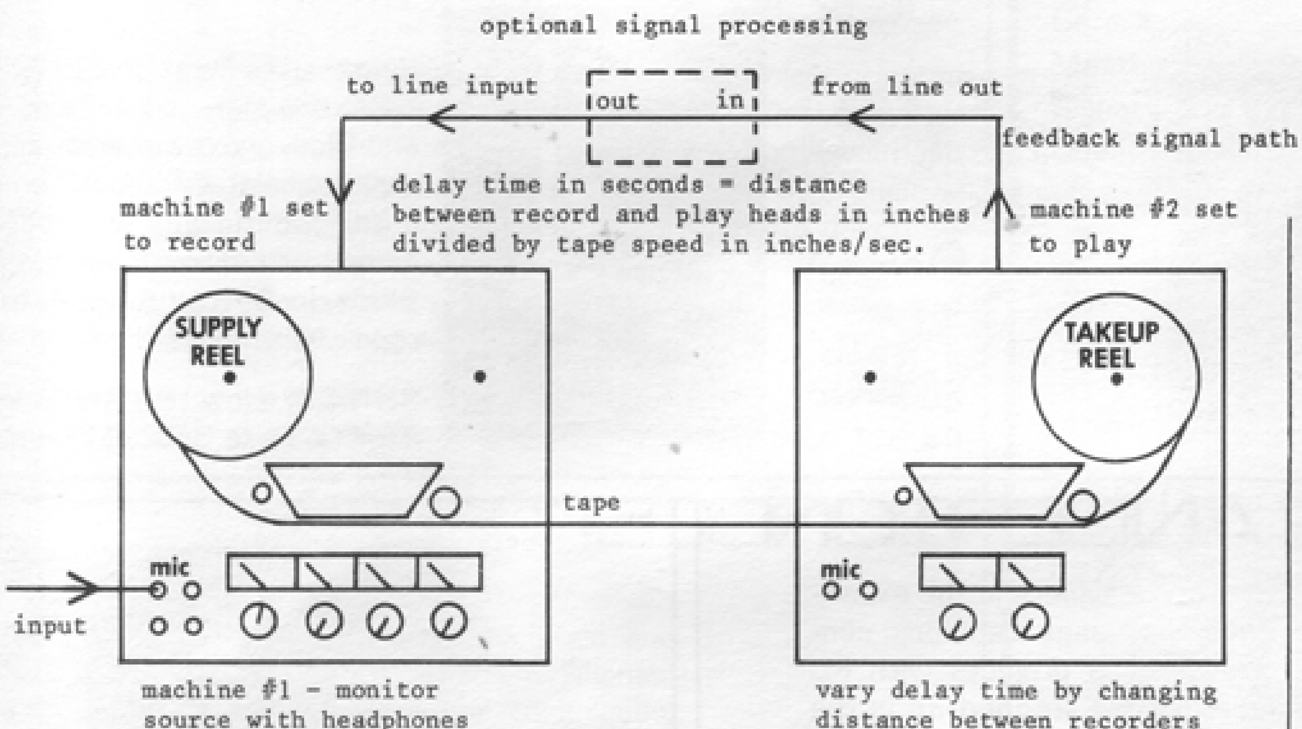
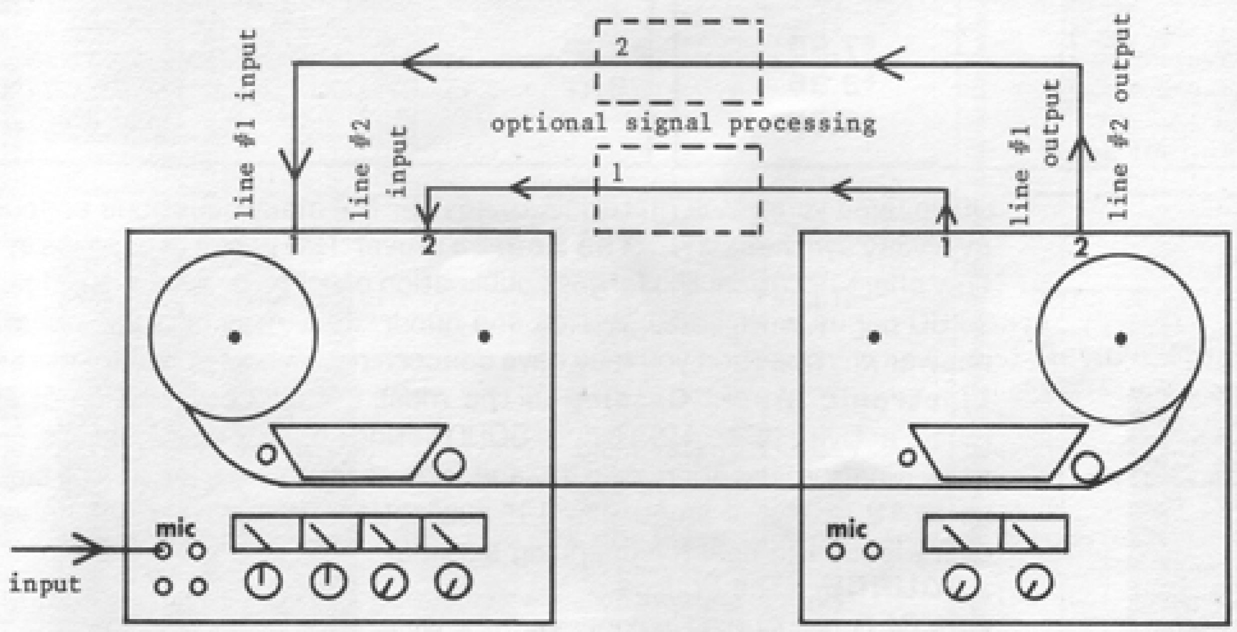


FIG. TWO: AUTOMATIC PANNING



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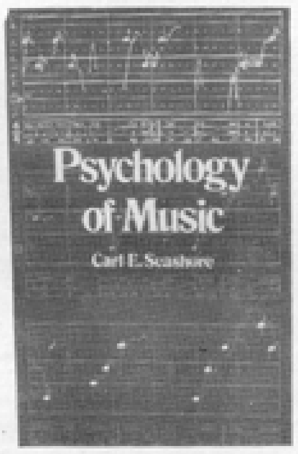

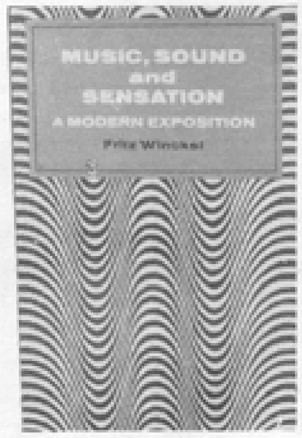

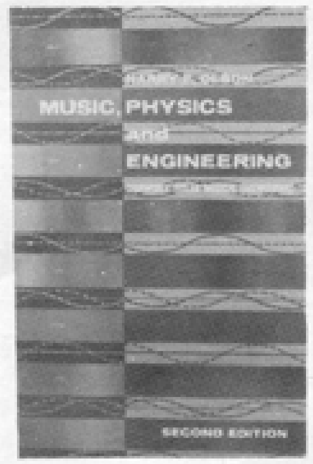

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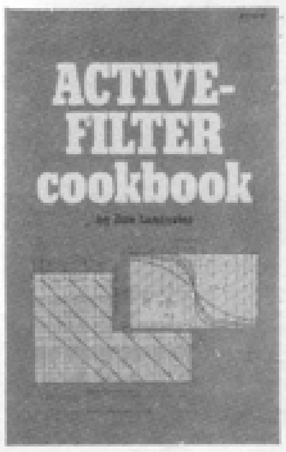

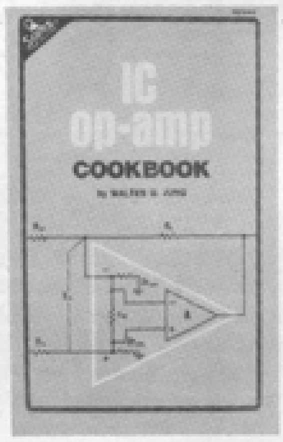

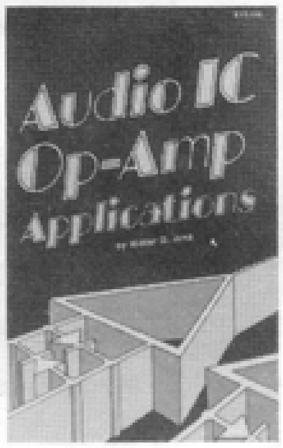

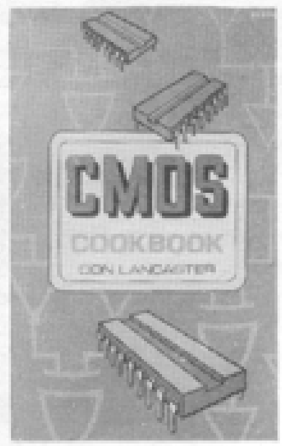

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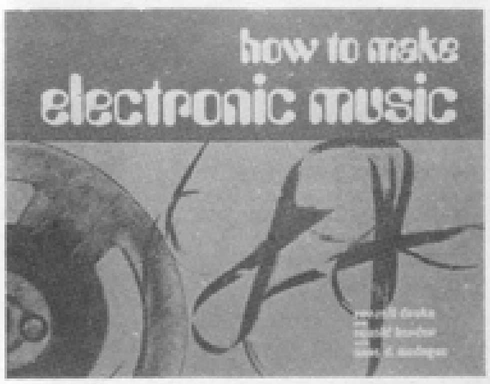

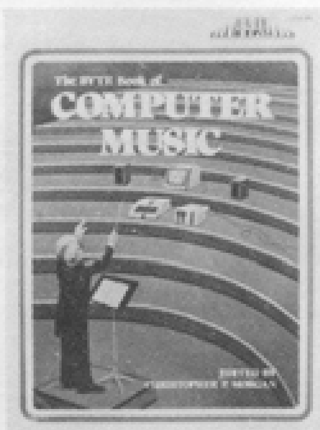

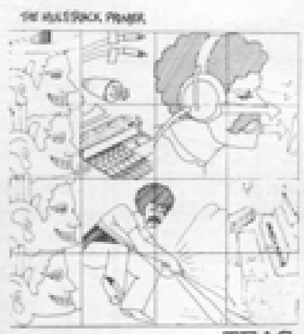

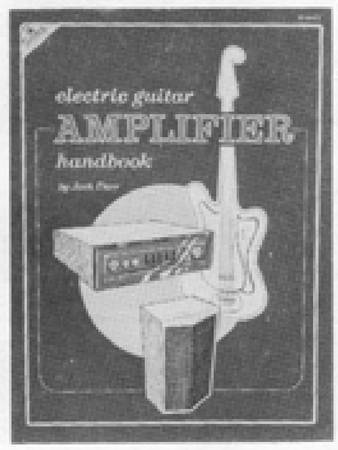

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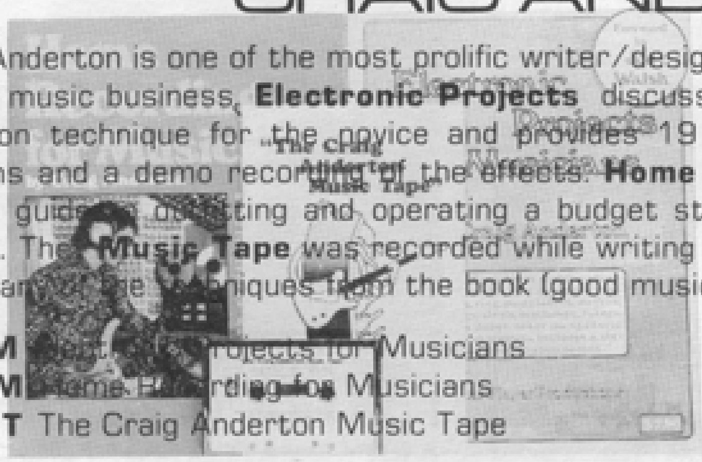

Craig Anderton is one of the most prolific writer/designers in the experimental music business. **Electronic Projects** discusses electronic construction technique for the novice and provides 19 projects with PC patterns and a demo recording of the effects. **Home Recording** is the original guide to outfitting and operating a budget studio for maximum results. The **Music Tape** was recorded while writing HRFM, so you can hear many of the techniques from the book (good music too!).

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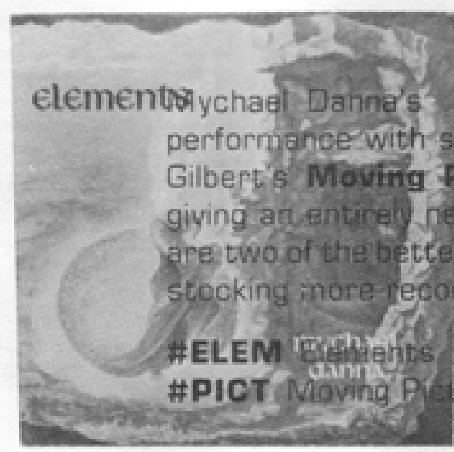
#HRFM	Home Recording for Musicians	$9.95
#CAMT	The Craig Anderton Music Tape	$5.95



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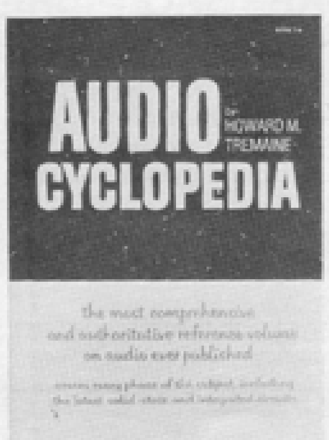

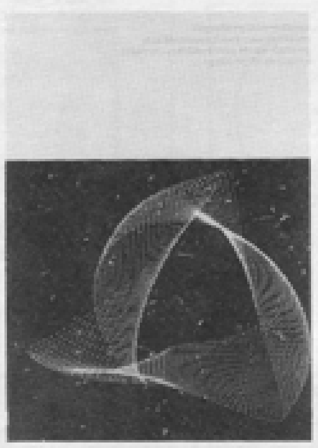

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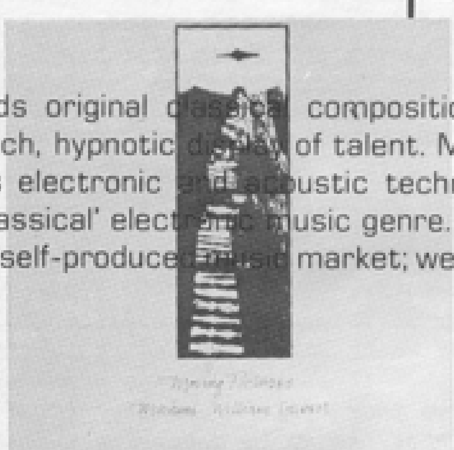

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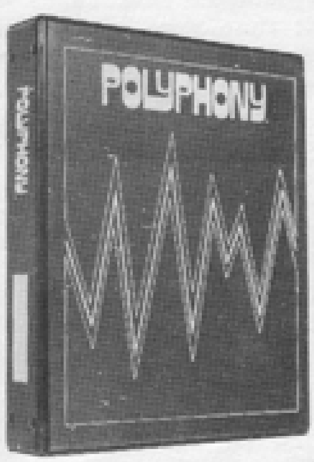

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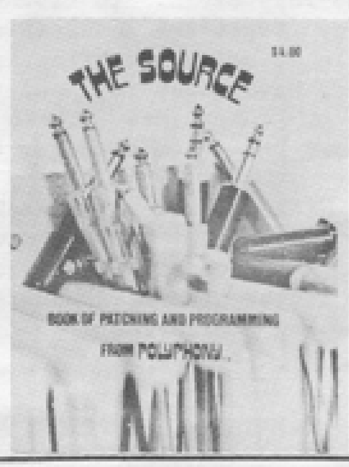

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☐ #0402: September/October 78: electronic music notation, notes on the recording of "Cords" by Larry Fast, sequencer software- part one, rhythmic control of analog sequencers, touch switch projects, modular vocoder techniques, PET as a music controller,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

☐ #0403: November/December 78: multi-purpose keyboard software, Sohler keyboard and notation system, voice frequency to voltage converter project, proposals for tape exchange, VCA project, sequencer software- part two, frequency balancing in recording, Barton and Priscilla McLean.

☐ #0404: January-March 79: add-ons for vocal F to V converter, shorthand patch notation, more on note to frequency conversions, graphic monitor project, George Russell, super VCA circuit, echo software, Vol. 4 index.

☐ #0501 May/June 79: using click tracks, PET music software, clockable sample/hold and noise source project, voice processing patches, VCF circuits, profile of John Cage, linear DAC.

☐ #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: composing on synthesizer, phase modulated sync, budget EQ, LFO project, Jan Hammer, Charlie Morrow, Poly-Split software, patches.

BILL MONTVILLO

continued from page 15...

You are now almost finished with your jingle. All that remains is the final mix, but there are special considerations for retail music. First, ask your client in which format he wants the final tape. The standard format is ½-track stereo recorded at 7½ ips, but you should be prepared to deliver mono and cassette copies. Regarding final mixes, Bill emphasizes finding the optimal mix for small speakers as used in portable radios and automobiles. Most likely, you have monitored your work on a good speaker system so the low and high frequencies were undistorted. But your jingle will be heard under quite a different set of circumstances. You don't want your great bass line to muddle the vocal part, and your client is concerned primarily with his product---not your music. To eliminate such hassles, many pro studios are set-up to monitor the mix over a car radio before the tape is delivered to their client. EQ levels may then be adjusted for the poorer frequency response of small speakers. This process can be simulated by playing a cassette mix on an inexpensive cassette player; you will have a pretty good idea of the acoustic effect of your jingle.

"Be prepared to make changes," says Bill. Do not expect your client to accept the first mix, although this should happen if you played preliminary cassettes and maintained personal contact. Nevertheless, clients change their minds. If all instruments are on separate tracks there will be no problem in preparing another version. This is why it is important to know your client and his product; if you can anticipate your client's musical preferences you could make two or three versions initially. The odds are in your favor if you deliver a choice of a few jingles.

As you can see, it takes a lot of time, knowledge, and energy to produce successful retail music. Should you decide to enter this profession Bill suggests you practice doing jingles before you look for clients. Try to create different moods and effects, build up a library of synthesizer patches, learn to work with click tracks to obtain precise timings, and get into the habit of documenting everything you do. After a while you will be able to work quickly, efficiently and economically, and be able to enter this competitive profession with confidence. Bill's final words of advice are: "SIMPLE IS BEST."

Home Studio

continued from page 10...

Be sure to not turn the demagnetizer on or off unless the unit is away from the recorder, your tapes, and other tools. Also, do not bring the demagnetizer near any recorded tapes while power is applied. Head cleaning and demagnetizing will solve many audio problems you may be experiencing, especially dull sounding tapes that result from loss of high frequencies. If you still have a loss of fidelity, it could be that your heads are out of alignment. Either get a book on tape recorder maintenance (not a bad idea anyway) or take your unit to a service center for head alignment work. Another thing to help you get primo recordings is to use the brand of tape recommended by the manufacturer. The internal bias controls are set to get maximum performance from a particular tape, even on the newer machines which have switches to change the bias for high energy tapes. If there's a tape brand that you prefer to use, you can have your tape recorder recalibrated for that tape. It's best to go to a service shop that is recommended and authorized by the manufacturer of your unit. By using the best (and unfortunatly most expensive) tape that your reel to reel is specially calibrated for, your tapes will sound very clean. In fact I have found that some of the high energy tapes are so clean and noise free that it matters very little if noise reduction is engaged or not. Incidently, if you do have money to spend you might consider getting a Dolby, dbx or compandor. These not only reduce noise, but also improve the dynamic range.

With the recording techniques discussed here and a good reel to reel, you are well on your way to realizing the complex electronic music scores in your mind. And that's why you bought the synthesizer in the first place! Good luck finishing your dream!

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CHAMELEON 0.25

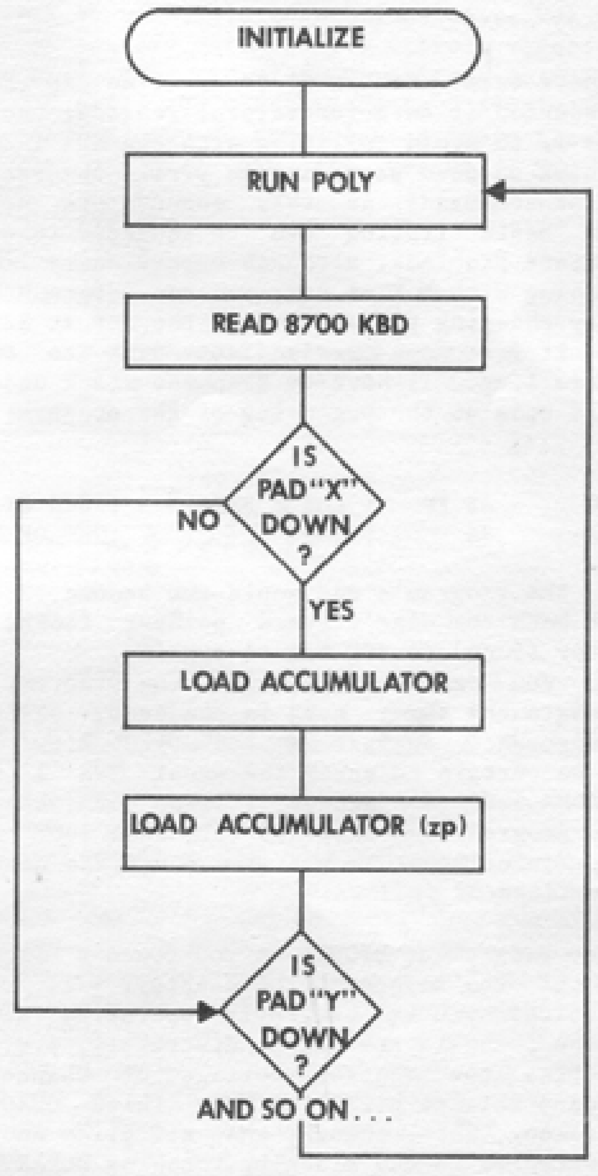
BY JON BALLERAS

There's little telling into whose hands digital music gear is falling and how it's being used. Some users running digital control type systems undoubtedly have been programming for years and are contentedly typing in their own intricate software, filled with stacks happily being pushed and pulled, indexes that know just where they're going and what to do when they get there, subroutines nesting comfortably inside subrouting, and God knows what else. Others, like me, soldered up an 8700 without knowing a bit from byte, much less being able to add $2 and $2 on the machine. Worse, users like me can easily develop a propensity toward software dependency--we become listing junkies anxiously waiting the next hit of code from Simonton or some other kindly programming wiz. Not an especially comfortable situation, especially when you realize that your high technology system doesn't do what you want it to do and, worse, Polyphony may never print a program that makes it run exactly the way you want.

All this was driven home to me a few months ago when I seriously considered playing gigs on my machine. Even though I'd done a good deal of normalizing, it became evident that I wouldn't get the same effects on a job as I did at home when leisurely laying down track by track of tape. For example, selecting glide for one channel meant shutting down the machine, remembering the transpose location for that channel, calling it up, writing in the correct code, touching ENTER, and remembering to start MUS 1 up at the correct location: eleven keystrokes in all. Switching from 1 to 2 or 4 voices meant going through an analagous process, with lots of room for error. Granted, these aren't terribly complicated procedures, but on a job there's already enough to think about: the charts, wondering if the bass player will catch the cue for the next section, what's going on in the audience. With all this happening it's nice to reserve some mental energy for just getting into the music and maybe tweaking a filter or resetting an envelope somewhere along the way.

At this point necessity plunged me into programming. My goal was to write some code that would allow on the fly control of the number of voices POLY was running and of glide and transpose values for four discrete synthesizer channels (each with hardware ADSR's). After a month of performing some simple programming exercises--reading and testing data from the command keyboard, writing a preselected value to a designated zero page location, and lots of branching -- CHAMELEON 0.25, the program at the end of this article, materialized. With one minor glitch, it does exactly what I wanted. More interestingly, it can be expanded to control other MUS 1 functions. Additionally, the body of the program looks to be a useful subrouting for programs that benefit from fast voice switching and changes in TTBL, like the "SPLITZ" portion of Bob Yannes' SHAZAM, and ECHO when run with its first preset. Experienced programmers will quickly note that CHAMELEON is far from perfect. It is extremely redundant and could easily be shortened considerably by using indexing techniques. Nevertheless, the program does run and has led me into more sophisticated coding projects. Most importantly, it has solved some vexing real time control problems that, apparently, no one else was about to take on. Programming autonomy!

The bulk of CHAMELEON consists of a long series of tests (CMP's) of the data put out by DECODE, the keyboard reading subroutine of Piebug Monitor. In the listing, this coding begins at ADDR $228. If the keypad being tested is down, the computer is instructed to write a preset value to a designated zero page location. If the keypad isn't down, the computer takes a branch (BNE) to the next CMP, and a similar routine is followed. Flowcharted, an abbreviated outline of CHAMELEON looks something like this:



some finer (yet useful) points

As the above flow chart implies, the bulk of my code is a kind of appendage to the sequence in which MUS 1 normally calls up its subroutines. As both John Simonton and Bob Yannes have pointed out in these pages, MUS 1 is not at all a monolithic chunk of programming which can only be called up at one location to do only one thing. Instead, this prom has a good number of useful entry points which you can call up when you need them with a simple JSR. If you'll look over the OPTION portion of the MUS 1 listing (beginning at ADDR $DOO) you'll find that CHAMELEON calls up almost exactly the same sequence of subroutines. But my code "opens up" MUS 1, allowing the program to continue after DECODE and perform as many KBD tests as needed. SHAZAM and ECHO are two other programs that freely enter and exit MUS 1. Studying these listings was a key to setting up CHAMELEON.

To give you a more specific idea of MUS 1's flexibility, notice that my listing contains a truncated initialization routine beginning at ADDR $205. As I experimented with each section of CHAMELEON, I found that downshifting from four voices to one, for example, didn't work. Instead of writing to QUASH channel 1 only, POLY got confused and assigned notes to all four channels. While pressing CLEAR wiped out those notes that were evidently floating around in KTBL and NTBL, TTBL was also cleared, removing any glide or transpose value the program had loaded there. Deep thought and lucky guessing led to my writing a second initilization routine, one that cleaned up those notes bouncing around in KTBL but left TTBL untouched. This rewritten version of MUS 1's analagous routine occupies ADDR's $205 to $20E. The point is that if part of MUS 1 doesn't do exactly what you need, you can always write around it. And there's nothing wrong with borrowing liberally from this prom's code, or from any other code, for that matter. If part of a program solves your problem, use it!

making it run

You'll note that CHAMELEON is written on Page 2 of memory. I started it here for several reasons: to avoid the stack on Page 1, to avoid colliding with the MUS 1 tables and other variables on zero page, and to permit the easy addition of other preset commands as they occured to me. As it happens, the basic listing can be squeezed into zero page without immediate problems, although expansion is limited. So if you're running with ½ K of RAM, you can enter CHAMELEON on zero page by changing the address of the JMP at $2A0 from OF 02 to OF 00. If for some reason you want to enter this program on Page 1, you'll have to keep the stack under control by adding this code at the beginning of the program:

```
0100    A2 FF    LDX # $FF    ;PUSH STACK
0102    9A       TXS          ;OUT OF WAY
```

The JMP at the program's end would now become 12 01. Don't forget to set both the user's stack pointer ($OFE) and the monitor pointer ($OED) to $FF before running.

Wherever you decide to locate the program, operating procedures remain the same: type in the code, saving it on tape as insurance against a bad byte blow up. Most importantly, be certain to enter the usual MUS 1 variables, starting at ADDR $OE8 ($40/$20/$01 will do the job.) When you first run the program, the displays will read $00. The 8700's KBD is now "intelligent", and the functions of all active keypads are defined as follows:

Start the program at $200. As you touch a keypad on the second rank of the keyboard, the displays will indicate the number of voices you're currently operating with. The glide/transpose channels are set up discretely, i.e., touching TRANS 2 doubles the control voltage of Channel 2 only; touching GLIDE 3 selects glide for the third QUASH channel only, and so on. Note, though, that all glide and transpose channels are cleared in one block by tcuching CLEAR. Shifting up form 1 to 2, 3, or 4 channels presents no problem, but downshifting from three or four channels can be a touch noisy if the ADSR's for these channels are still in their sustain or release cycles. As I've noted, CHAMELEON was written for a machine running with one QUASH and hardware ADSR's for each channel. If you're still running STG's, computer control of the transpose table is still possible. Just go through the listing beginning at ADDR $26E and change all STA zero page addresses designated $CE to $CD. Addresses $CD become $CB, and $CC's become $CA's. Making these changes will line up the transpose table with the CV channels of one or two STG driven QUASH. So, for absolutely quite downshifting, downshift from full polyphony only when your synthesizer is already quiet. (One deep breath after your fingers are off the AGO should do it.)

going further

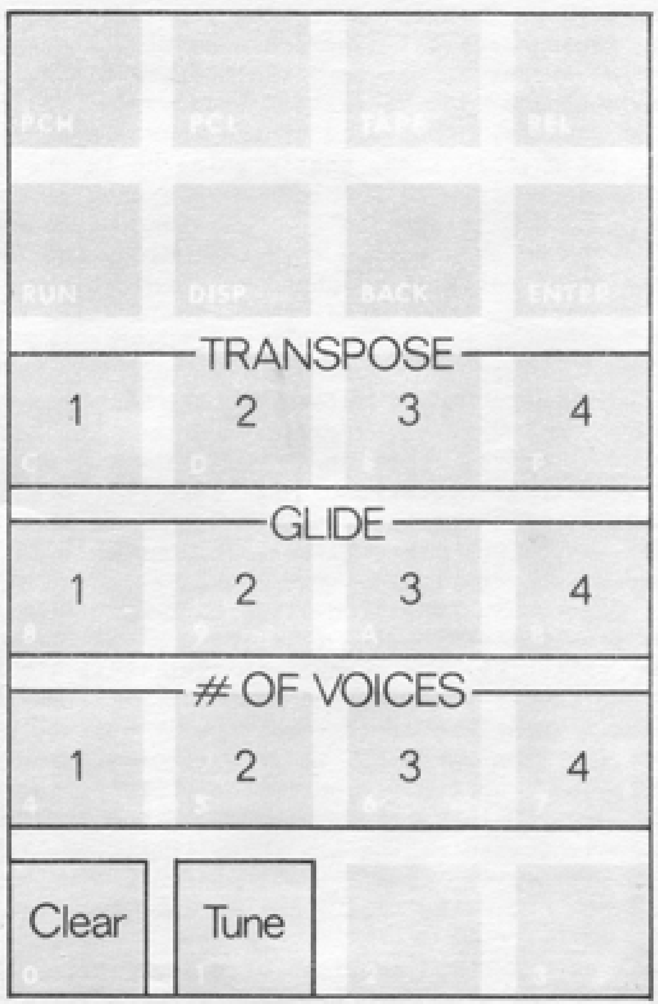
Although CHAMELEON is hardly a tight, finished program, it does have the advantage of being comfortably open ended. Once you've caught on to its basic routine of: test KBD/branch/ write to zero page, expanding the program or rededicating the 8700's control pads to different functions isn't difficult and, it seems to me, could turn into a useful set of exercises for any beginning programmer. A control pad, for instance, could be assigned to write a 1 (or 2) octave transpose to <u>all</u> channels, or to write a simultaneous glide and transpose to designated channels. Even more challenging and useful would be setting up some pads as STG envelope presets. By expanding CHAMELEON's treadmill styled routine of test/branch/write, some code to select a preset STG envelope ($10/$04/$20/$10/$3F) comes out like this:

```
02XX    C9 XX    STGT 1    CMP # $XX    ;IS PAD XX DOWN?
        DO 14              BNE STGT 2   ;NO, BRANCH TO NEXT TEST
        A9 10              LDA # $10    ;YES, PREP ATCK
        85 BA              STA $BA      ;STORE ATCK
        A9 04              LDA # $04    ;PREP DCY
        85 BB              STA $BB      ;STORE DCY
        A9 20              LDA # $20    ;PREP SUST
        85 BC              STA $BC      ;STORE SUST
        A9 01              LDA # $01    ;PREP RLS
        85 BD              STA $ BD     ;STORE RLS
        A9 3F              LDA # $3F    ;PREP PEAK
        85 BE              STA $BE      ;STORE PEAK
```

Conceivably half a dozen of your most useful STG envelopes could be stored in the program and called up when you need them.

afterwords

While I certainly hope CHAMELEON will prove helpful in solving some of your computer-assisted synthesizer control problems, I'll count this article even more of a success if it encourages those of you who are "software dependent" to start fooling around with your own code. We all known there are stacks (!) of books on programming on the market (some, like



The First Book of Kim and William Barden's How to Program Microcomputers actually talk about the 6502 in enlightening ways). After reading material like this the MOS Manual begins to make some sense. However, it's been my experience that you learn programming by doing programming. Adding $2 plus $2 is a start. As you go on, more simple exercises will occur to you, and simple exercises have a way of germinating into full fledged programs, as I hope CHAMELEON shows. The sample programs in the 8700 (or whatever computer you have) manuals are ripe for study and flowcharting -- particularly helpful activities since you'll become familiar with the actual protocols of Paia equipment (or your own system). Ultimately, each programmer has to find his or her way of coming to terms with their machine. It's not that hard, and you may well be surprised at what you learn. Happy coding!

CHAMELEON 0.25

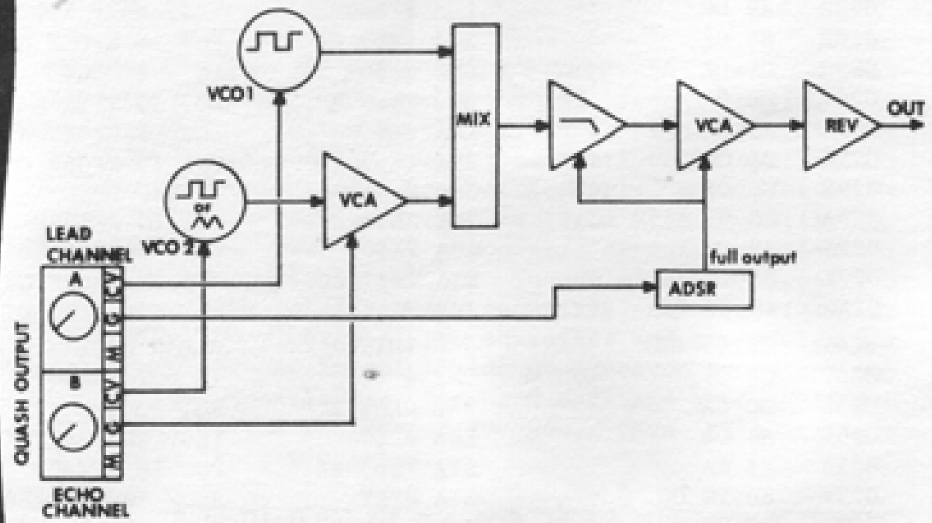
Control System For 8700 Based Synthesizers

By Jon Balleras

July 1979

ADDR	CODE	LABEL	INSTRUCTION	COMMENT
0200	20 21 0D	INIT 0	JSR INIT	CLEAR KTBL,NTBL,TTBL
0203	90 0A		BCC POLY	BRANCH ALWAYS POLY
0205	A9 00	INIT 1	LDA # $00	PREP TO ZERO
0207	A2 10		LDX # $10	SET POINTER
0209	95 CF	Z BUF	LDA KTBL, X	ZERO BUFFER
020B	CA		DEX	POINT TO NEXT
020C	DO FB		BNE Z BUF	LOOP IF NOT DONE
020E	60		RTS	BACK TO VOXTS
020F	20 71 0D	POLY	JSR POLY	ASSIGN NOTES
0212	20 C3 0D		JSR TRNGN	CALL STG'S, IF ON
0215	20 2B 0D		JSR NOTE	PLAY NOTES
0218	20 00 0F		JSR DECD	READ 8700 KBD
021B	C9 01		CMP # $01	IS TUNE DOWN?
021D	90 E1		BCC INIT 0	NO,IT'S LESS,GO CLEAR
021F	D0 07		BNE VOXT 1	GO TO VOXT 1
0221	A0 5C		LDY # $5C	PREP FOR TUNE
0223	20 52 0D		JSR FILL	PUT NOTE IN ALL VOX
0226	F0 E7		BEQ POLY	PLAY TUNING NOTE
0228	C9 04	VOXT 1	CMP # $04	IS $04 DOWN?
022A	D0 0A		BNE VOXT 2	NO, GO TO VOXT 2
022C	A9 01		LDA # $01	YES, PREP 1 VOX
022E	85 EA		STA (zp) EA	PUT IN OUTS
0230	8D 20 08		STA DSPY	SHOW VOX NUMBER
0233	20 05 02		JSR INIT 1	CLEAR NTBL
0236	C9 05	VOXT 2	CMP # $05	IS $05 DOWN?
0238	D0 0A		BNE VOXT 3	NO,TO VOXT 3
023A	A9 02		LDA # $02	YES, PREP 2 VOX
023C	85 EA		STA (zp) EA	PUT IN OUTS
023E	8D 20 08		STA DSPY	SHOW VOX NUMBER
0241	20 05 02		JSR INIT 1	CLEAR NTBL
0244	C9 06	VOXT 3	CMP # $06	IS $06 DOWN?
0246	D0 0A		BNE VOXT 4	NO, TO VOXT 4
0248	A9 03		LDA # $03	YES, PREP 3 VOX
024A	85 EA		STA (zp) EA	PUT IN OUTS
024C	8D 20 08		STA DSPY	SHOW VOX NUMBER
027C	A9 80		LDA # $80	YES, PREP GLIDE
027E	85 CC		STA (zp) CC	PUT IN XPOSE CH 4
0280	C9 0C	TTST 1	CMP # $0C	IS $0C DOWN?
0282	D0 04		BNE TTST 2	NO, TO TTST 2
0284	A9 0C		LDA # $0C	YES, PREP XPOSE
0286	85 CF		STA (zp) CF	PUT IN XPOSE CH 1
0288	C9 0D	TTST 2	CMP # $0D	IS $0D DOWN?
028A	D0 04		BNE TTST 3	NO, TO TTST 3
028C	A9 0C		LDA # $0C	YES, PREP XPOSE
028E	85 CE		STA (zp) CE	PUT IN XPOSE CH 2
0290	C9 0E	TTST 3	CMP # $0E	IS $0E DOWN?
0292	D0 04		BNE TTST 4	NO, TO TTST 4
0294	A9 0C		LDA # $0C	YES, PREP XPOSE
0296	85 CD		STA (zp) CD	PUT IN XPOSE CH 3
0298	C9 0F	TTST 4	CMP # $0F	IS $0F DOWN?
029A	D0 04		BNE RETURN	NO, TO RETURN
029C	A9 0C		LDA # $0C	YES, PREP XPOSE
029E	85 CC		STA (zp) CC	PUT IN XPOSE CH 4
02A0	4C 0F 02	RETURN	JMP POLY	DO IT AGAIN
024F	20 05 02		JSR INIT 1	CLEAR NTBL
0252	C9 07	VOXT 4	CMP # $07	IS $07 DOWN?
0254	D0 0A		BNE GLDT 1	NO, TO GLDT 1
0256	A9 04		LDA # $04	YES, PREP 4 VOX
0258	85 EA		STA (zp) EA	PUT IN OUTS
025A	8D 20 08		STA DSPY	SHOW VOX NUMBER
025D	20 05 02		JSR INIT 1	CLEAR NTBL
0260	C9 08	GLDT 1	CMP # $08	IS $08 DOWN?
0262	D0 04		BNE GLDT 2	NO, TO GLDT 2
0264	A9 80		LDA # $80	YES, PREP GLIDE
0266	85 CF		STA (zp) CF	PUT IN XPOSE CH 1
0268	C9 09	GLDT 2	CMP # $09	IS $09 DOWN?
026A	D0 04		BNE GLDT 3	NO, TO GLDT 3
026C	A9 80		LDA # $80	YES, PREP GLIDE
026E	85 CE		STA (zp) CE	PUT IN XPOSE CH 2
0270	C9 0A	GLDT 3	CMP # $0A	IS $0A DOWN?
0272	D0 04		BNE GLDT 4	NO, TO GLDT 4
0274	A9 80		LDA # $80	YES, PREP GLIDE
0276	85 CD		STA (zp) CD	PUT IN XPOSE CH 3
0278	C9 0B	GLDT 4	CMP # $0B	IS $0B DOWN?
027A	D0 04		BNE TTST 1	NO, TO TTST 1

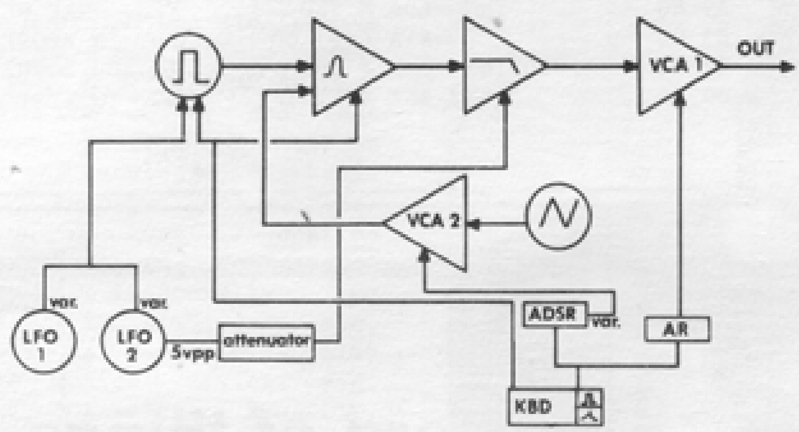
PATCHES!



Acoustic/ Western Guitar

VCO 1: square wave
VCO 2: triangle wave for regular acoustic
square wave for western guitar
Tune VCOs in unison; Mix outputs equally
VCF: low pass, high range, range (init freq) = 75%, Q = 65%
ADSR: A=0%, D=40%, S=50%, R=100%
Reverb: very slight, for depth only
Keyboard/ Quash/ Echo .31 software :
* Use preset #0 (OUTS = 01, ECCO = 07, DLAY = 01, OFST = 01) for single string plucking.
* Use preset #2 (OUTS = 01, ECCO = 03, DLAY = 02, OFST = 08) for multi-string patterns; play fast, rhythmic, stacatto.

Russell Brower
Glendale, CA

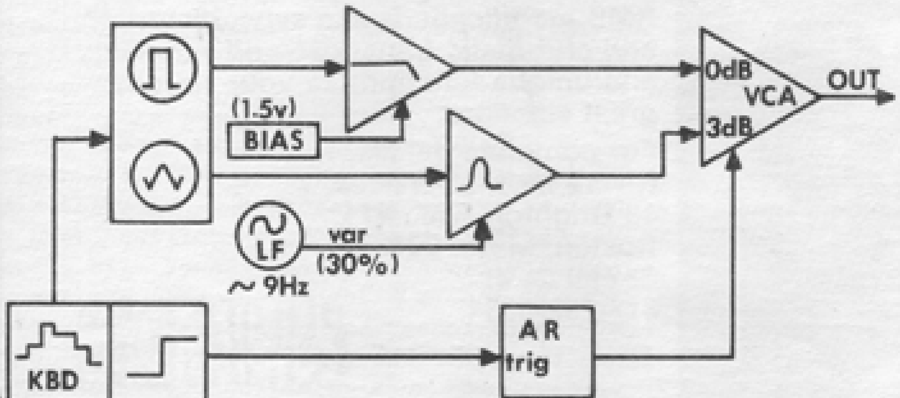


Indian Woodwind

VCO: very narrow pulse
VCF: bandpass, Q = 90% to 100%
AR: A=60%, R=80%, short timing ranges
ADSR: A=10%, D=100%, S=15%, R=15%, output level- see text
LFOs: see text

I just call this patch Indian Woodwind because, after months of trying to get the right sound, I forgot the name of the instrument! The noise is present to simulate the players breath and as such, the ADSR's variable output should be set so its presence is unobtrusive. Ideally, its peak should be easily heard, its sustain level barely audible. The two LFOs should be set near 6 and 7 Hz and the variable outputs set to provide an unsteady vibrato (to prevent it from sounding too "perfect" and therefore inhuman).

Bill Williams
Clarkston, MI



Harmonica

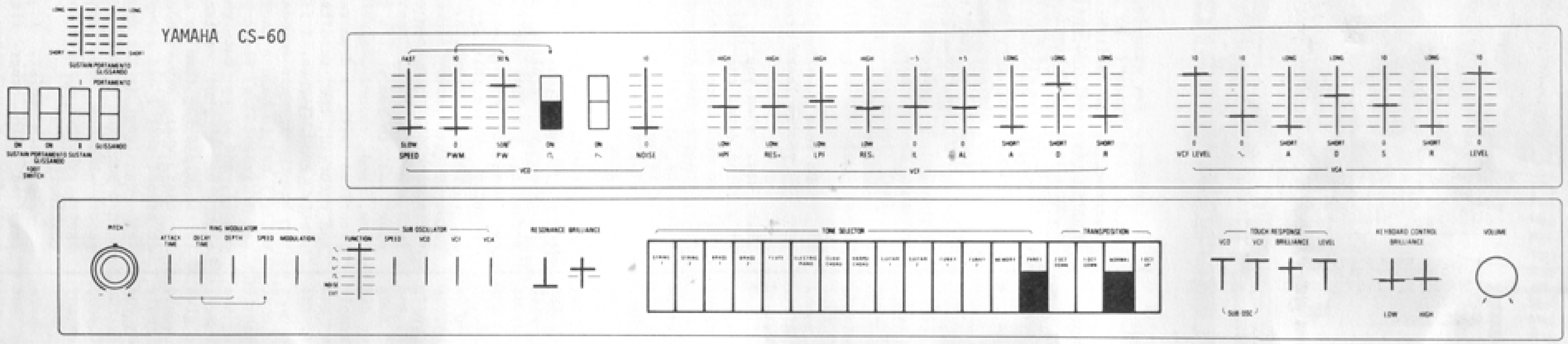
Kybd: tune to highest octave, play in top two octaves
VCO: minimum audible pulse width
AR: long timing range, A=50%, R=15%
LFO: approx 9 Hz, 30% output level
VCF: band pass, Q = 60%
VCF: low pass, slight bias to remove "edge" from pulse

Vary the amount of LFO fed into the band pass to control expression; 30% to 40% gives emotional 'fireside' harmonica

Mark Briggs
Balch Springs, TX

BILL BOYDSTUN
OKLAHOMA CITY, OK

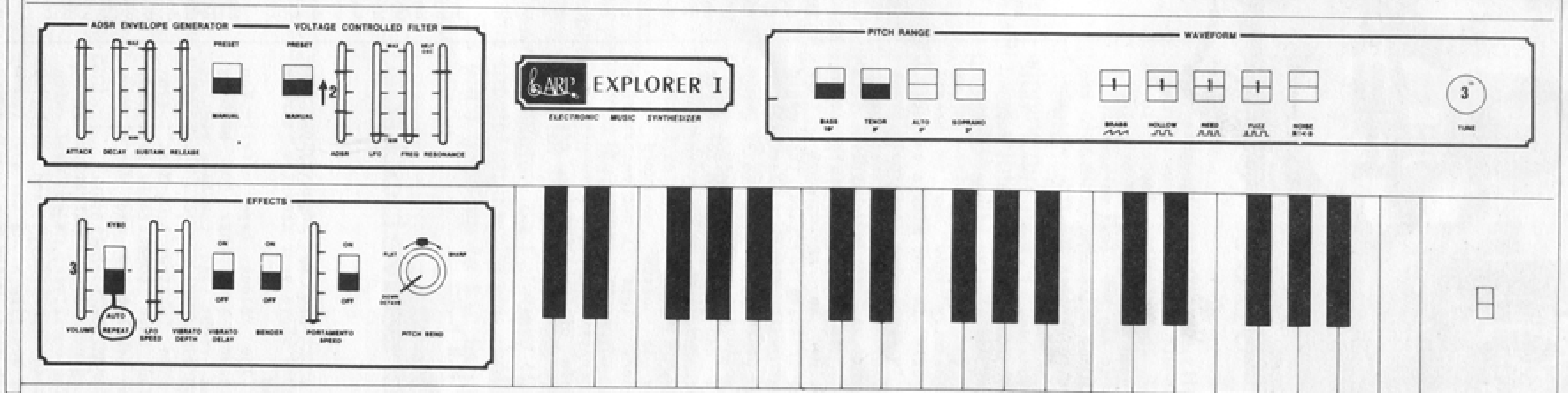
YAMAHA CS-60



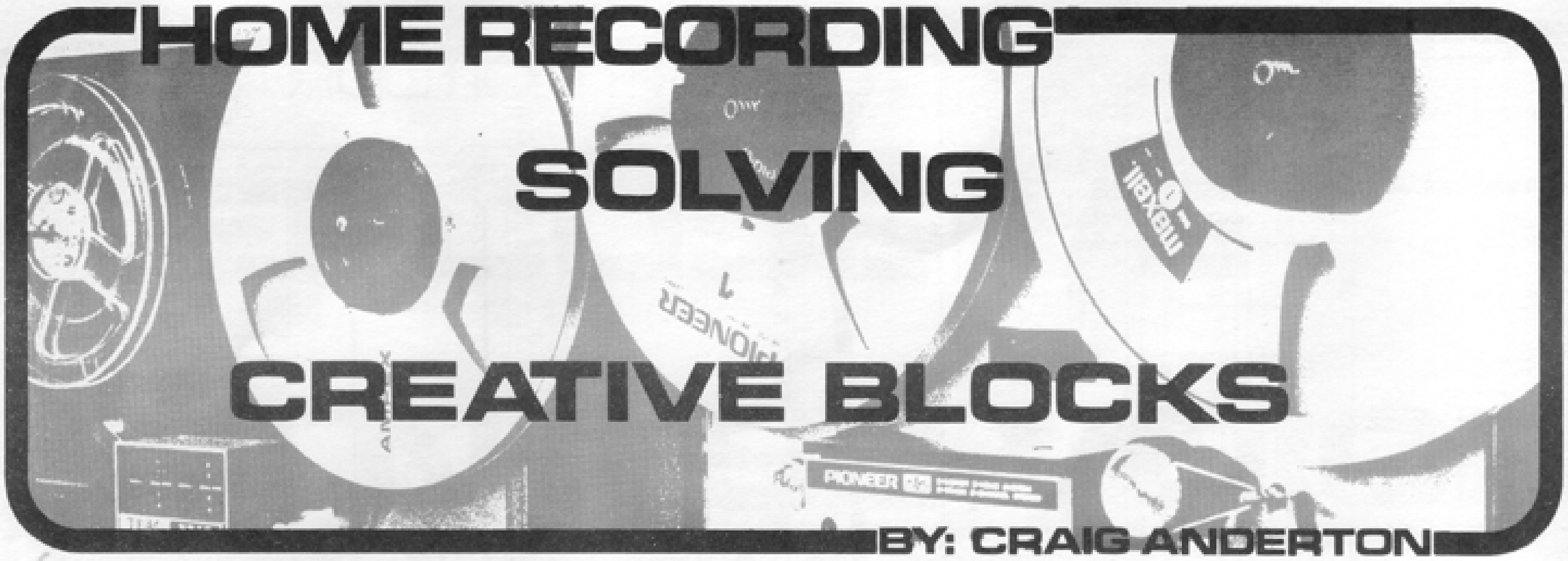
1. DIFFERENT WAVEFORMS PRODUCE DIFFERENT HARMONIC STRUCTURE SWEEPS. 2. VARY ADSR/VCF CONTROL FOR DIFFERENT "STOPPING PITCHING". 3. PITCH & VOLUME AS DESIRED.

HARMONIC SIFTER:

BILL BEHRENDT
CAMPBELL HALL, NY



HOME RECORDING
SOLVING CREATIVE BLOCKS



BY: CRAIG ANDERTON

Ideally, creative thoughts and actions should flow out of us at all times; of course, it's impossible to even approach that ideal. Nonetheless, it's fun to see how close we can get to that goal, and whether or not we can come up with devices and attitudes that stimulate creative thought.

Personally, I don't feel that creative thoughts are sudden, capricious, and elusive beings; rather, they are in us all the time, but we don't necessarily know how to access them properly in our mind's computer. From time to time, some external event or internal feeling occurs; then something clicks, and a creative idea pops out. It then follows that if we can provide an environment that encourages creative thoughts, that magic "something" will click a far greater percentage of the time, leading to more creative actions.

If I can just ramble on for a bit more here before getting into the meat of things, I think one problem is that creativity is often confused with intelligence. This is not necessarily the case. Anyone is capable of coming up with creative ideas, and people with high levels of intelligence are not necessarily the most creative.

two kinds of creative blocks

Let's begin by identifying two common cases where your creative stream is blocked.

1. The "Blank Tape" effect- You want to record something, but nothing comes out right and the tape remains blank.

2. The "What-Do-I-Do-Next" syndrome- You have some tracks recorded, but become stalemated amd don't know what your next move is.

By generating a creative environment for ourselves, we can deal with both these problems, and pick up additional benefits along the way (such as greater self-confidence from knowing that we have a certain degree of control over our creative flow).

"blank tape" effect solutions

Here is one way to arrive at a solution:

Think about the reason you're having trouble with your creative mechanism; if you can figure that out, you're extremely close to the solution.

For example, one day I just couldn't put down a drum track I was trying for; no matter what I did, it didn't sound like the sound I had in my head. After following the advice given above, and thinking about the situation, I realized that I would <u>never</u> get the sound in my head--because <u>I</u> didn't have the right equipment at that time to pull it off. Thus, my creative impulse was frustrated due to the lack of proper tools for its expression. I then made a mental note to myself that I've <u>got</u> to build a new drum set as soon as I have the time, but in the meantime, I'd better try something else. I switched over to the Programmable Drums, messed around with it while checking out the various drums with a couple of trial programs, and what do you know--I somehow came up with a catchy beat. That suggested a bass part to go along with the catchy beat, which suggested a guitar part, which led to coming up with words, and before I knew it, there was a new song that I really liked. While I never did solve the original problem, this example does show how identifying a problem can free you from its negative effects, and let you pursue a more productive path.

It is very important to note that the identification process is <u>not</u> a process for making excuses. In the previous example, saying "Well, since I don't have the equipment, I'll just give up because I can't do what I want" would be making an excuse instead of truly identifying the problem. Finding the problem in this case demands finding the corresponding solution, or your identification work will not produce any tangible results.

If there is another instance where you just can't make anything happen but still want to do something in the studio, try making a better working environment. For example, do you have a patch cord "tree"? Do you neglect the construction of switching boxes and mixers, so that you end up doing a lot of unnecessary patching? Is it awkward to edit tape? Is everything properly maintained and in working order? If not, your odds for having a creative flash will diminish because you are making demands on your time with these problems, and the more time it takes to implement a creative thought, the greater the danger of losing the original intention. We can summarize the content of this paragraph as solution #2, which goes:

Do your housekeeping when you're least creative.

That way, when you're feeling creative at a later date, the recording process will go much more smoothly. Admittedly this is a somewhat devious way to deal with the initial problem, but the end result is nonetheless beneficial.

Solution #3 is a little different, and it goes like this:

Run an Experiment.

When you really want to record something, but just can't quite figure out what to do, the idea is to do <u>anything</u>. While this may not start off as anything worthwhile, the important point is that it may serve as a bridge to something that does eventually turn into something neat. One experiment of mine was to see what a flanger sounded like, and record the results. I practiced with a drum set, flanged it, and recorded the results...it sounded pretty good. Then, I figured I'd test out how the flanger sounded with bass. Well, I had to have a chord progression, so I grabbed a piece of paper and a pencil, and wrote out what I thought might be a good collection of chords (without playing them on an instruments to hear what they sounded like), until it seemed that I had enough changes to last as long as the drum part. As it turned out, the chord progression was pretty likable too, except for a few spots...which I altered to improve the continuity.

At this point, I started to get involved with other flanger settings applied to other instruments; eventually, I ended up with the tune

"Roy Herful" on my Music Tape. In fact, for "Music Machine" on the same tape I pretty much used the same process: I wrote out a bunch of almost random chords that looked OK on paper, tried them, made a couple of arrangement changes, and ended up with a satisfying piece of music. The point is that sometimes, a cut-and-dried experiment can all of a sudden pick up a life of its own--and eventually, trigger one of those true creative moments that transforms the identity of what you're doing from an experiment to something deeper and more purposeful.

Other suggested experiments: Try making a tape with only your voice. It might not be anything you'll want to keep and play for everybody, but you'll discover sounds you never thought you could make...and probably have a good time, too. Get outrageous; that's what experiments are for. Once I tried an experiment with the ground rules being to 1) try to hold one constant note with my voice, and 2) not listen to any previous tracks while doing overdubs. It produced an interesting effect, because the voice can't hold the pitch accurately for a couple of minutes at a time, plus the breathing can't be synchronized to previous tracks since you aren't listening to previous tracks, so there is beating, wavering, and other kinds of changes that are held together by the basic premise of your experiment. Did this experiment produce a masterpiece? In a word, no. But it taught me a lot, including some techniques I want to use on future pieces. As far as I'm concerned, converting a creative block into a learning experience is just as desirable as turning that block into an actual piece that you'd want to work on and polish up.

Here's another experiment: Pretend you've been assigned to record a demonstration tape of an instrument or effects box. How would you do it? How would you show off all the things it can do? Thinking along these lines will often prod you into discovering new instrumental possibilities, sometimes after only a few minutes of playing.

Other possible experiments would be writing a piece around one instrument or sound only, playing a particular piece as sparsely as possible, finding out what particular instruments sound like together, comparing the difference of one recording technique with another, and so on.

It's important for me to re-iterate that it's entirely possible that none of these events, in themselves, will necessarily be the creative answer you're looking for. But starting something--anything--can lead you down a path where, consciously or not, you encourage a creative event to happen.

There are other possibilities, such as using chance to determine what you're going to play (try throwing dice to determine changes, where numbers correspond to chords), testing out a new invention or piece of equipment, or experimenting with unusual time signatures. All of these can start the creative process going by making creative suggestions; you can then modify these suggestions or leave them intact--whichever sounds better.

dealing with the "what-do-i-do-next" syndrome

So far, we've talked about how to initiate a creative event; now, we need to figure out how we can wind it down...after all, nobody can stay creative hour after hour. Eventually, the creative battery runs out of charge; it invariably need some rest, and recharging, before it will start putting out again.

Therefore, there is a real chance that when you don't know what to do next, the problem is not a creative block; rather, there may not be enough creative energy left. In this case, you should strive to recognize the symptoms of energy drain as soon as possible, so that you can quit while you still feel good about what you're doing, before it becomes a chore. Otherwise, you might try to drain an already depleted power source--which does nothing but cause long-term harm.

After keeping track of my performance in the studio over the past decade or so, I've found that I can seldom maintain peak efficiency for much longer than 3 hours (on particularly demanding stuff, 1½ hours is more like it). While this may be due to intendity more than fatique, the fact remains that after this point, if I try to force myself to come up with something great, it just isn't going to happen. So if you don't know what to do next, first ask yourself the following question:

Have I already done enough
for one night?

Remember that you can be creative in your mind, but your body may not have the energy to implement your ideas. This is another case where you're limited by your equipment, and you might consider getting your body more in shape if you want to extend your endurance in the studio. I never could understand session musicians who never see the sun and chain smoke...how do they do it? However, if you think that you really are in a creative frame of mind and want to continue, ponder this question...

Is the basic track in conflict
with the overdubs I want to add?

A song can change character between the time of its inception, and the time it needs overdubs; if a change has occurred, that could explain why you find it difficult to continue. At this point, you have to decide whether to start over, or modify your overdub plans. Most of the time when this happens to me, I start over; but usually finish the original anyway (no matter how crudely), so I have a rough draft of what I'm aiming for.

If you find yourself trying another instrument or effect just for the sake of trying it, or if you start feeling any kind of desperate need to finish a track, you're best off stopping. I think it's very important to leave the studio satisfied with what you've done, and trying too hard will usually give the wrong results...it's as bad as not trying hard enough.

If after approaching the problem a few more times you still don't know what to add to a track, don't erase the basic track. Come back to it a month or two later, and you might be surprised to hear it sounding much better than you remembered. This positive surprise alone can suggest what types of overdubs would be appropriate.

final comments

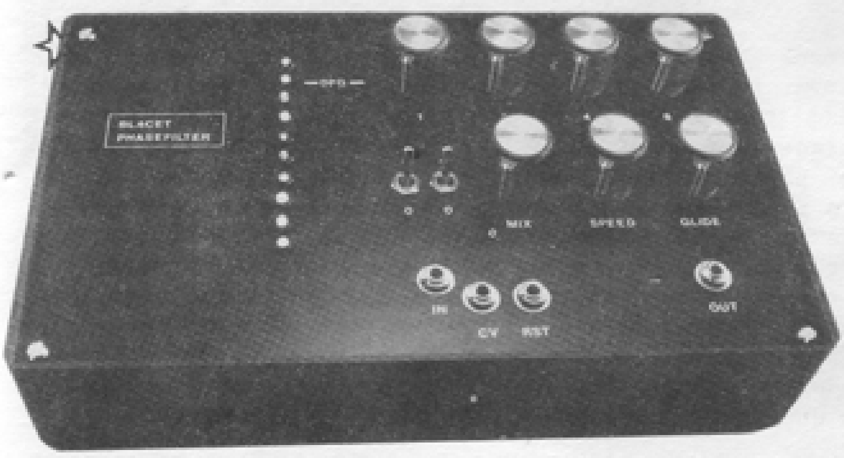
Sometimes, merely instructing yourself to act in a certain way will produce the desired results. If you orient your mind to having confidence in your creative abilities, and if you recognize that the above steps can help you unlock your natural creativity, you may "hypnotize" yourself into being creative with no further ado. In other cases, it might take a little while longer. But analyze and observe what does, and does not, lead you down creative paths. Some people have their creativity triggered by sex, some by drugs, some by jogging or exercise, some by meditation; finishing a new instrument or device and testing it out can inspire you, as can hearing any really good piece of music. By analyzing your actions, you can perform additional actions that point your life towards living in a more creative environment. That environment will, in time, inspire creative actions.

Dream Module Contest

Polyphony Magazine in Conjunction with Blacet Music Research Presents a Contest to Test Your Imagination!

1st Prize: A Custom Assembled Phasefilter.

2nd–5th Prizes: 2 Year Subscription (extension) to Polyphony.



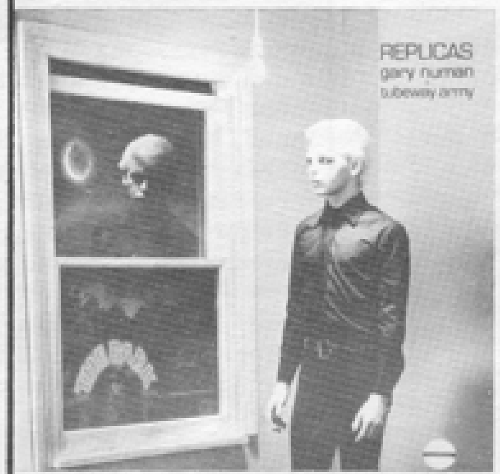
The purpose of the Dream Module Contest is to stir the imagination of Polyphony readers and to plumb their deepest fantasies. ...about synthesizers! Entries should be a description of a synthesizer module, an entire system, an effects box, or a microprocessor program. These should have some features that are not currently available. A reasonably brief description of what the module or system would do is really the key, although block diagrams or other electronic information would be welcomed for clarity.

The entries will be judged by the staff of Polyphony magazine and Blacet Music Research. We will look for the usual things: originality, practicality, neatness, humor. All the usual contest rules apply: All entries become the property of Polyphony Publishing and Blacet Music Research, winning entries will be published in future issues of Polyphony. So, put yourself into your favorite dream state, let the ideas flow, and pick a winner!

Contest closes March 28, 1980, so get your entries sent off to:

Dream Module Contest c/o POLYPHONY
PO Box 20305 Oklahoma City, OK 73156

REVIEWS



Replicas
Gary Numan and Tubeway Army
Atco SD 38-117

Pleasure Principle
Gary Numan
Beggars Banquet BEGA 10

Since its development, the synthesizer has threatened to provide a number of styles and new directions for music. Somehow, these revolutions have failed to appear or, at best, have been short-lived. Gary Numan presents a new style of music which is heavily based on electronic music. Drawing from influences in pop music, new wave, electronic music, and art rock, Numan has forged a new sound which could easily become the most accessible branch of new wave or electronic music to date. "Me I Disconnect from You" and "Are Friends Electric" from Replicas, and "Cars" from Pleasure Principle have already made substantial showing on the charts; a dedicated group of Numan fans is growing, and consists of much more than just musicians or electronic music followers.

Numan's success is strongly based on simplistic production (although still heavily layered with underlying themes) and a sort of regression of synthesis techniques. A friend recently commented "It's all been done before". While that is true, it has been a while, and the results are somehow more comfortable now. The state of the art, and the musical development of our ears has pushed us to work harder and harder at generating more complex and more natural or imitative effects. As a result, many of us have forgotten about the inherently "new" sound of such effects as glide, pulse width modulation and detuned oscillators.

Replicas is Numan's first American release; Pleasure Principle is his third English release, just now appearing in the US. (Pleasure Principle, as well as Numan's first album entitled Tubeway Army, are both available from major record stores who stock import albums distributed by JEM Records, or by direct mail order from Import Record Service, South Plainfield, NJ 07080.) Replicas features a number of innovative compositions offering sonic as well as lyrical hooks. Themes consistent throughout the album include mans interaction with machines ("The Machman", "Are Friends Electric", "When The Machines Rock"), and extraterrestrial life ("I'm Praying to the Aliens", "I Almost Married A Human"). Replicas features two instrumentals, "...Machines Rock" and "...Married a Human", while Pleasure Principle has "Airlane". All these instrumentals are among the strongest cuts of the albums, featuring varied rhythmic structures and good melodic composition. Pleasure Principle is much less a ground-breaking album, drawing much on the stylistic research and development of Replicas, but presenting a better produced statement of the central direction of the earlier work. The lyrics on Pleasure Principle are less abstract, presenting more direct statements about everyday life as in "Cars", "Engineers", "Communication", and "Films".

Gary Numan has drawn upon a number of the best qualities from musical styles which appeal to both experimental musicians and casual listeners alike. It will be interesting to watch his future development and public response to Numan's music and upcoming American tour. This may very well be a lasting definition of the New Wave movement.

-Marvin Jones

Games
Synergy
Passport PB6003

Looking back on the development of Synergy, most listeners remember the first two albums (Electronic Realizations for Rock Orchestra, Sequencer) as electronic rock music. The music had the rhythmic and dynamic impact usually associated with rock music, and the nature of the sonic and musical composition was definitely a "reaching out" to find new territories for exploration. The music imparted a very strong "good" feeling, perhaps partially due to the "rock" nature of the music, and partly due to the composers satisfaction in realizing the sonic experiments he was trying. With the release of Cords, the Synergy sound definitely took on the compositional and performance aspects of symphonic endeavors. While an entirely different scope of music, Cords represented another form of experimentation and development of new textures, compositional techniques, and instrumental technique.

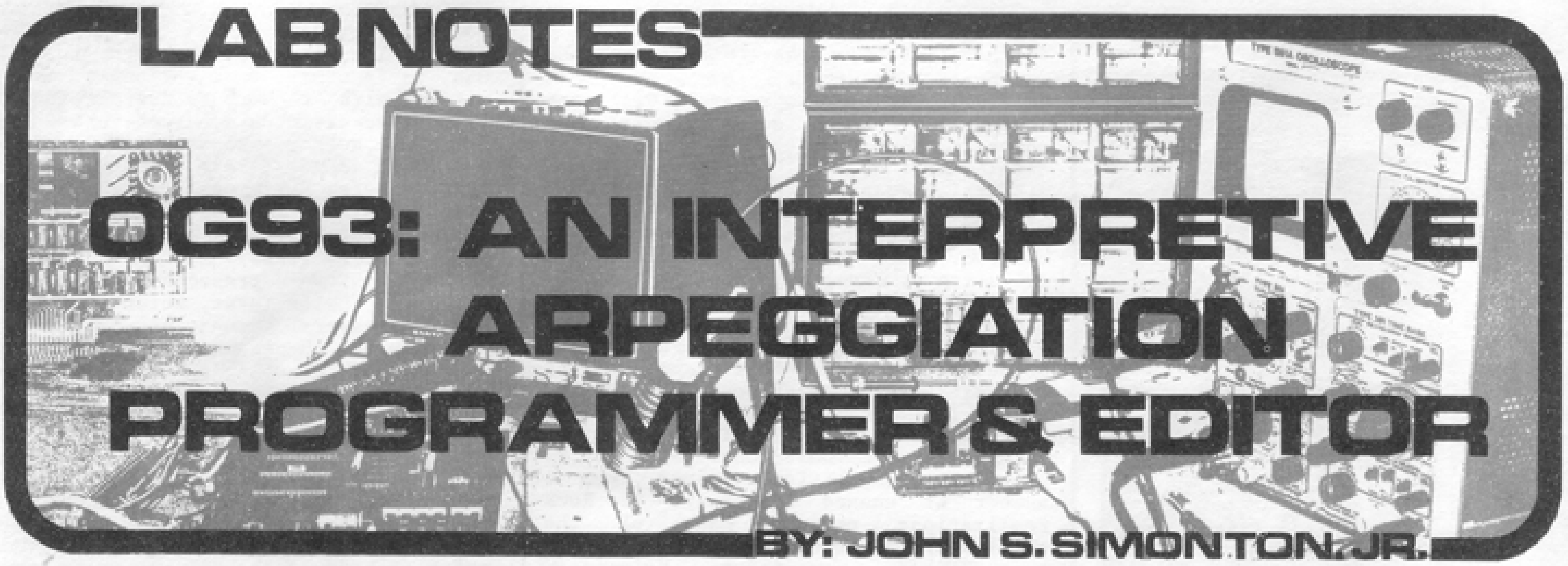
Games is much less an experimental endeavor, and shows an application and collation of the ground covered in the previous albums. Much of the livelier and more dynamic "rock" feeling of the earlier albums is present here, but melded with the symphonic textures and "classic" forms of composition explored in the last album.

Side one of Games holds Delta Two, Delta Four, and Delta One. Delta Two is noted as a composition derived from a number of unused themes from the Rock Realizations period. Many of the original signatures are heard: plucked bass lines, percussive "clavinet" voices, drums, "electronic" lead voices. The primary difference between this cut and the early albums is the notable development of string and brass patches and melodic interaction. The first song cross fades directly into Delta Four which uses an orchestral tape loop (with some excellent string synthesis) as a background for a permutation sequencer work which was done on the Bell Labs digital synthesizer. The loop work is better than that done on Cords in that the voicings and loop development (or reprocessing) are sonically more easily followed. The work done on the Bell system is interesting as far as the building and modification of the sequence, but the sequencer software used was not fully developed and lacked capabilities for significant alteration or reassignment of the voice or patch used to perform the sequence. Thus, the sequencer line gets a bit tedious, with very few timbral changes throughout the piece. The last cut on side 1, Delta One, is another rock oriented composition with a very strong rhythm and bass line structure, good brass and fuzz-tone guitar synthesis, and a rhythm solo that sounds like a drum synthesizer gone bonkers. Parts of the lead line use a non-modulated resonant delay line as a fixed formant filter- a seldom realized use for delay lines.

Side two presents the multi sectioned Delta Three, which is a suite based on several themes used as audio sound system checks on the Peter Gabriel tour- an interesting source of material for composing! There is a wide variety of everything represented in Delta Three. Symphonic composition prevails, although "electronic" patches and sound effects, FM synthesis, "rock" rhythm and bass lines, and even some convincing choir and organ synthesis appear. On listening to this side, I frequently think of many of the earlier major works by Yes. Much of the melodic, timbral, and contrapuntal organization is very similar. This cut, due to its length and diversity, is perhaps the best on the album.

If you liked any of the earlier Synergy albums, chances are good that you will find several things of interest on this album. It serves as an excellent summation of the work Larry Fast has done thus far; it projects a more relaxed feeling- as if he wanted to have fun doing this album, and use what he has learned, rather than to take off on another probe. Chances are good that the next album will be another exploratory mission and, while they are important not only to the development of Synergy, but to the state of the art as well, it will always be good to hear Larry sit down and have some fun laying down tracks.

continued on page 32... -Marvin Jones



LAB NOTES

OG93: AN INTERPRETIVE ARPEGGIATION PROGRAMMER & EDITOR

BY: JOHN S. SIMONTON, JR.

One of the major advantages that our hybrid computer/synthesizer system offers is the ability to realize a class of new tricks which for lack of a better term we'll call "keyboard effects". I have in mind new sounds which arise not so much from the timbre of each note, but from the timing and sequence in which the keys played are converted to notes and how they're allocated to available output channels.

Using this definition, I suppose that POLY-SPLIT from last time would qualify as a keyboard effect because it affects the way that keys held down are allocated to note-producing output channels. But, ECHO (January-March 1979 Polyphony, page 29) is more specificly what I feel the term should mean because with that program new effects (and at short delay settings, new timbres) arise that would be extremely difficult to accomplish without some means of juggling key activations and how they're assigned to outputs.

Another good example would be the ORGASMATRONIC GLIDE arpeggiation trick that the keyboard encoder and D/A did by themselves (remember?). Hold down a bunch of keys and the encoder, while scanning, stopped momentarily when it reached one of the down keys and played the note briefly before continuing the scan. When another key was found down, it stopped again to play that one, and so on. Altogether an alright thing that allowed arpeggiations to be played much more rapidly than they could be without electronic assistance.

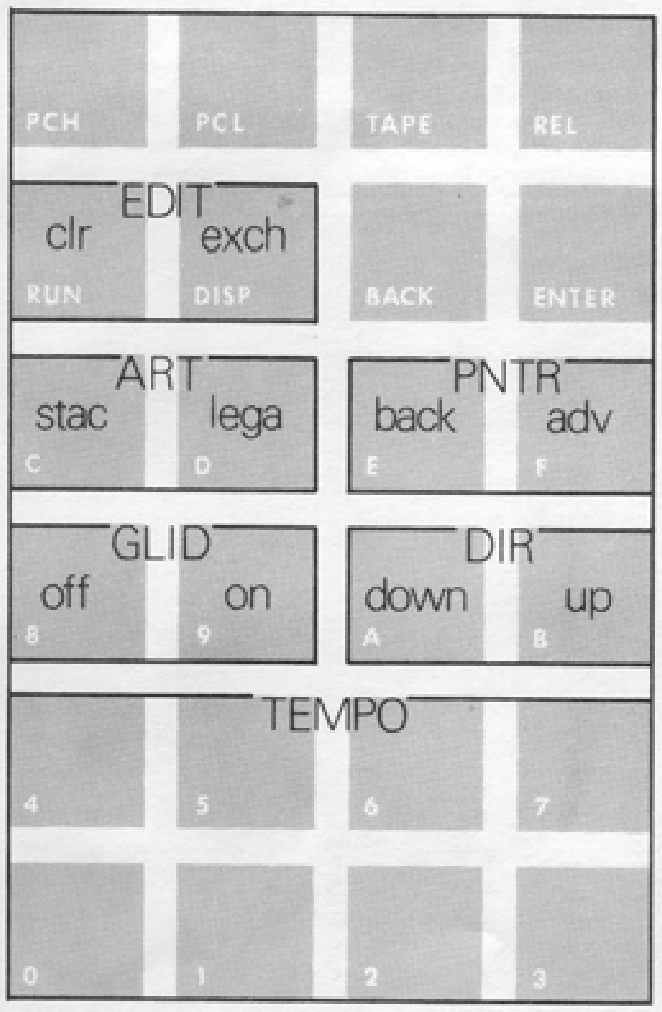
When we installed the computer in the loop, we lost Orgasmatronic Glide (OG), which maybe was not such a huge sacrifice when considering the power that was gained in the process; but still, I know several folks who mourned the loss because it was an effect that they were using to good purpose in their music.

Here's a terrific replacement. This new program does the same thing that the old OG did, hold down a bunch of keys and it plays them in sequence, but it also gives control that wasn't possible with the old "state machine" version. For instance, it can arpeggiate down-scale as well as up. And it plays staccato or legato. It also allows touch pad control of glide and similar control of the tempo of the arpeggiation.

Great. But not the greatest part, we'll get to that soon.

Enter the program as outlined at the end of the column and start it running, then press down a group of keys. If you've done everything correctly, you should hear a relatively slow down-scale arpeggiation of the notes that you're holding down. When the lowest note has played, the sequence should start again from the highest.

Now let's play with the control some. Here's what the keys mean with OG93 running:



Touching the DIR:UP pad will cause the arpeggiation to change direction from down-scale to up. GLID:ON turns the glide for the arpeggiation channel on and (you guessed it) GLID:OFF turns it off.

The LEGATO ARTICULATION pad causes the trigger signal to remain high as long as any keys are down so that there will be no re-articulation as one note finishes playing and the next begins. STACCATO ARTICULATION triggers the note the first instant that it plays then releases the trigger.

The TEMPO keys cause the rate of arpeggiation to change from slow (7) to fast (0) over a range from so slow that almost anyone could play the run manually to a rate that's so fast that the sequence begins to take on the texture of a chord (which should give you a clue to one interesting application of OG93 in a piece of music).

If you were an Orgamatronic Glide fan in the first place, we could probably stop here and you'd be completely happy - the program is a lot better than the old manual version. We'd also be stopping before we really got started, because by far the most interesting feature of OG93 is that it's an interpreter that allows us to program a series of arpeggiations and an editor that makes the entry and manipulation of those programs easier.

Each program step contains all of the information that we controlled earlier (glide on and off, up-scale or down, staccato or legato, and one of 8 tempos) and when the program is run, each step will be taken in turn and an arpeggiation of the keys held down performed using the status of the parameters specified by that step. At the end of the program it jumps back to its beginning and the sequence of arpeggiations repeats.

Each step of the program is "written" in exactly the same way that we set the parameters earlier; in fact, as you'll soon realize, you were in effect writing the first step then. The key to forming these steps into programs is the PNTR : BACK/ADV block of pads on the command keyboard. The pointer (PNTR) refers to the program step that you're writing.

One quick example should get the idea across. We'll write a program that sweeps up the keyboard at a moderate tempo, re-articulating each note, followed by a quick legato run down-scale. Program the first step by touching these keys - TEMPO:4, DIR:UP, GLID:OFF, ART:STAC. That takes care of the up part.

Now for the down part, begin by touching PNTR:ADV so the commands that we enter next are "pointed" at the second program step (which is step #1 as shown in the displays, the first step is #0) and touch TEMPO:2, DIR:DOWN, GLID:OFF, ART:STAC. Now hold down a big chord structure to hear the full effect of this dual arpeggiation.

Editing an existing program is simply a matter of pointing to the program step that you want to change and entering the changed parameter. To change the first step (#0) in the example above to a slower tempo, for example, touch PNTR:BACK so the display shows 00 and then touch TEMPO:7 (or whatever).

OG93 can handle programs up to 8 arpeggiations deep and, when you begin stacking that many steps, it's easy to get lost. The EDIT:EXCH key helps here by allowing us to remove the step pointed to from the program and replacing it with an instruction for repeat. By backspacing the pointer to step #1 and touching the EDIT:EXCH pad, we cut the program to just the first step, EDIT:EXCH again and the original program step is back in place, so that the entire program runs again. By stepping through the program and causing it to repeat after the 2nd, 3rd, etc. steps, it's fairly easy to locate where in the program a specific sound is coming from and then make changes there.

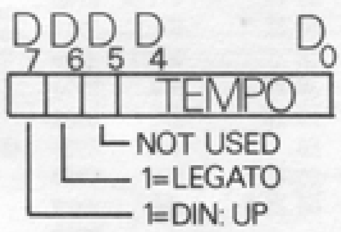
As you may surmise from the name, the EDIT:EXCH key causes the program step pointed to to be exchanged with a memory buffer location which is initialized to contain the interpreter's repeat code (00). This implies that this key can also be used to exchange two program steps by pointing first to one and touching EDIT:EXCH and then to the next and again EDIT:EXCH. In fact, this is the case; with one exception. The first step of the program may not be the repeat code 00. If it is, the interpreter will lock up as it reads the first step, finds that it's a repeat, so it reads the first step, and so on. OG93 protects against this by checking to see if you are pointing to the first program step and, if you are, checking to see if you're getting ready to make it 00. If you are, it doesn't complete the exchange. You can get around this if you want to exchange the first step with another by pointing to the other step first, EXCHanging, and then going to step #0 and EXCHanging again.

The final editing key is CLR (clear). Touching this pad clears the program, with the exception of the first step, which remains unchanged for the reasons given above.

For detailed operational information there is no substitute for the liberally commented assembler listing at the end, but let's talk in general terms about how the program works.

We use the MUS1 firmware NOTE to take care of the dynamics of maintaining analog outputs that must be refreshed and to read the AGO keyboard. The list of keys which the firmware returns as being held down is the "arpeggiation list", or the notes to be played. In simplest terms, OG93 does nothing but delay for a period of time which determines tempo and, when the time is up, pulls the next key from the list and plays it as a note. The bulk of the rest of the program checks that we're not yet to the ends of tables (keys down, program steps, etc.) and, if we are, takes car of starting from the beginning again, and controls things like re-setting the tempo timer when it expires and re-articulating between notes when playing staccato.

Most of the arpeggiation program's control information is contained in the single word of memory labeled SCTL (sequence control) in the listing. The 8 bits of that word have these uses:



When the interpreter needs the tempo, it extracts it from this word with an AND operation (statement 1440). The status of the control bits D6 and D7 are determined with BIT operations at lines 0790 and 0900.

When one run has finished and the next is due to begin, the interpreter pulls the next program step from memory (the program buffer CSEQ) and after a manipuation which immediately isolates the glide controlling bit, places the program step in the control word SCTL, where the rest of the program accesses it as outlined above. The isolated glide bit is immediately rotated into the most significant bit of the transpose buffer (TTBL) word corresponding to the ouput channel being used.


```
                        0010 :********************************
                        0020 :*                              *
                        0030 :*     ORGASMATRONIC GLIDE      *
                        0040 :*                              *
                        0050 :*  ARPEGIATION PROGRAMMER AND  *
                        0060 :*          EDITOR              *
                        0070 :*            BY                *
                        0080 :*    JOHN S. SIMONTON, JR      *
                        0090 :*                              *
                        0100 :*(C) 1979 PAIA ELECTRONICS, INC*
                        0110 :*                              *
                        0120 :********************************
                        0130 :
                        0140 :****** MONITOR SUBROUTINES ******
                        0540 :THIS IS THE MAIN PROGRAM LOOP. START BY INITIALIZING THE SYNTHESIZER
                        0550 :AND CALLING THE QUASH DRIVERS AND AGO KBD READING ROUTINES FROM MUS1
                        0560 :CHECK TO SEE IF A COMMAND KEY HAS BEEN TOUCHED; AND IF SO, JUMP TO
                        0570 :SUBROUTINE TO DETERMINE THE COMMAND AND EXECUTE IT. DETERMINE THE
                        0580 :POINTER FOR THE OUTPUT CHANNEL AND JUMP TO SUBROUTINE FOR ORG. GLIDE
*                       0590 :PROCESSING. ON RETURN, LOOP.
*1000LLL                0600 :
1000-  20 21 1D   JSR   0610      JSR INIT     :MUS1 SYNTH INIT ROUTINE
1003-  20 2B 1D   JSR   0620 LOOP JSR NOTE     :QUASH DRIVERS AND READ AGO
1006-  20 00 1F   JSR   0630      JSR DECD     :PIEBUG READ COMMAND KBD
1009-  B0 03      BCS   0640      BCS HERE     :IF NO NEW KEY TOUCHED, SKIP NEXT
100B-  20 00 11   JSR   0650      JSR CMND     :CALL COMMAND DECODER
100E-  A0 0F      LDY   0660 HERE LDY 0F       :POINTER TO ORG. GLIDE OUTPUT CHANNEL
1010-  20 16 10   JSR   0670      JSR STAR     :CALL ORG. GLIDE PROGRAM
1013-  4C 03 10   JMP   0680      JMP LOOP     :LOOP TO CONTINUE
                        0690 :
                        0700 :FIRST THE TIMER IS TESTED AND IF NOT TIME FOR THE NEXT NOTE TO BE
                        0710 :PROCESSED THE STACCATO CONTROL BIT IS CHECKED AND IF CLEAR
                        0720 :(STACCATO) BRANCH IS TAKEN TO DE-TRIGGER NOTE IN OUTPUT
                        0730 :BUFFER. IF LEGATO MODE, EXIT IS IMMEDIATE
                        0740 :
1016-  24 E7      BIT   0750 STAR BIT *KTBL+07 :ARE THERE ANY AGO KEYS DOWN?
1018-  50 1E      BVC   0760      BVC SINT     :NO KEYS, BRANCH TO RE-INIT ARP. POINTER
101A-  C6 72      DEC   0770      DEC *TIMR    :OTHERWISE, DECREMENT THE TIMER
101C-  30 05      BMI   0780      BMI ADVA     :IF EVENT TIME, BRANCH
101E-  24 74      BIT   0790      BIT *SCTL    :OTHERWISE CHECK FOR STACCATO AND IF TRUE....
1020-  50 46      BVC   0800      BVC CLRN     :BRANCH TO CLEAR TRIGGER FROM OUTPUT NOTE
1022-  60         RTS   0810      RTS          :OTHERWISE, RETURN WITHOUT CLEARING TRIGGER
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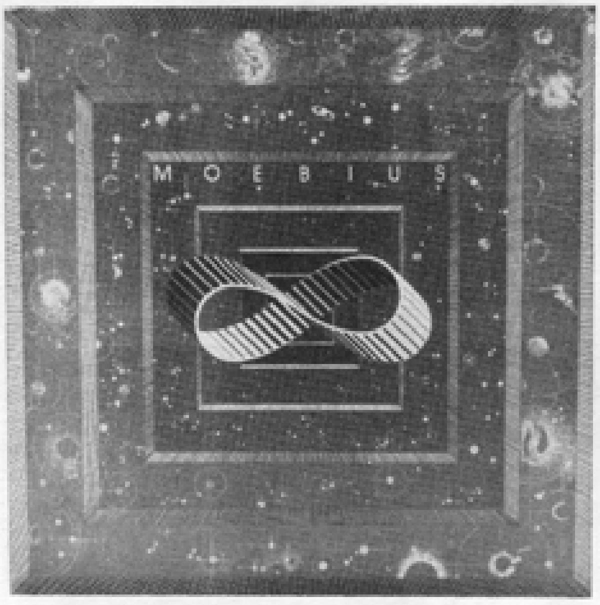
REVIEWS



Moebius
Moonwind Records
MW 33801

This album represents an original foray into an area which has been dealt with so often in cliche that its real possibilities have been largely ignored: pop synthesis. Taking up where Brian Eno and Tangerine Dream left off, "Moebius" combines some of the sonorities of these groups with the more minimalistic production values of groups like The Normal and Suicide. Add a dash of biting lyrics and a mock serious approach, and you have what amounts to the missing link between the more traditional schools of pop synthesis and the outer fringes of new wave. The result is a refreshing LP whose bold approach makes you wonder why no one has spotted this fertile turf before.

If "Moebius" straddles such broad musical boundaries, it is in no small part due to its personnel, all of whom are well grounded in synthesis. They are: Douglas Lynner, editor of Synapse and former member of LEM, Bryce Robbley, also of LEM, and Synapse correspondent Steve Roach. Their knowledge of the territory makes for an album which is not only broad in scope, but sonically inventive using synthesizers, drums and violin as well as effects such as crinkling paper, and processed voices to augment its aural images.

For my money, the best cuts are those which bear the closest ties with the new wave: tunes like "Light My Fire", where crazed vocals combine with a bassline gone bonkers to create an atmosphere which bears only a cosmic resemblence to the original; and "Clone Zone", a cut whose cryptic lyrics, naked sequencer and tongue in cheek production conjure up a time when mindless clones stalk the Earth like giant Xerox copies in search of their originals.

Of the rest of the material, the most accessible cuts are "Urth", and the title cut which uses sequencers to good rhythmic and harmonic effect. For more traditional tastes, "Prophecy" is a rolling instrumental reminiscent of Eno's dreamier moods.

There is a tendency towards overproduction which marrs some of the tunes. It's as if, in the process of rejecting the production cliches of pop music, Moebius has not yet zeroed in on what its own values are. And a couple of the tunes are simply too long. But for all its flaws, this is an interesting album whose innovative approach proves that the synthesizer has far from realized its potential in pop.

-Melodie Bryant

```
                          :
                          :IF IT'S TIME FOR A NOTE TO BE PROCESSED, THE POINTER TO THE INPUT
                          :BUFFER IS ADVANCED (EITHER FORWARD OR BACKWARD) AND IF THERE IS NO
                          :MORE BUFFER LEFT WE DROP THROUGH TO ADVANCE THE POINTER TO THE SEQUENCE
                          :BUFFER TO GET THE NEXT SET OF GLIDE PARAMETERS.  IF WE ARE NOT YET
                          :TO THE END OF THE IN BUFFER, WE BRANCH OUT TO RESET THE TIMER, ETC.
                          :
1023-  A6 73     LDX      ADVA LDX *PNTR    :GET POINTER TO INPUT BUFFER
1025-  24 74     BIT           BIT *SCTL    :CURRENTLY ARPEGGIATING UP?
1027-  10 05     BPL           BPL DOWN     :NO, BRANCH TO DO DOWN
1029-  CA        DEX           DEX          :TO GO UP-SCALE, DECREMENT POINTER
102A-  30 07     BMI           BMI SADV     :IF POINTER NOW <0, BRANCH
102C-  10 26     BPL           BPL STIM     :STILL IN RANGE, BRANCH ALWAYS
102E-  E8        INX      DOWN INX          :DOWN-SCALE, INCREMENT POINTER
102F-  E0 08     CPX           CPX 08       :OUT OF RANGE?
1031-  D0 21     BNE           BNE STIM     :STILL IN RANGE, BRANCH
                          :
                          :IF WE GET HERE (SADV) IT MEANS THAT WE HAVE PLAYED ALL OF THE KEYS
                          :THAT WERE DOWN AND HAVE REACHED THE END OF THE INPUT BUFFER
                          :NOW IT'S TIME TO GET THE NEXT ENTRY FROM THE CONTROL SEQUENCE.
                          :WE TEST TO SEE IF WE ARE AT THE END OF THE SEQUENCE AND IF SO THE
                          :POINTER IS RE-INITIALIZED. OTHERWISE, THE COMMAND IS FETCHED AND IF
                          :ZERO IT MEANS THAT IT IS THE END OF THE SEQUENCE AND THE POINTER
                          :IS ALSO REINITIALIZED
                          :
1033-  A6 76     LDX      SADV LDX *SPNT    :GET CONTROL SEQUENCE POINTER
1035-  CA        DEX           DEX          :POINT TO NEXT SEQUENCE ENTRY
1036-  10 02     BPL           BPL GSEQ     :IF NOT TO END, BRANCH
1038-  A2 07     LDX      SINT LDX 07       :RE-INIT SEQUENCE POINTER
103A-  86 76     STX      GSEQ STX *SPNT    :SAVE SEQUENCE POINTER
103C-  B5 77     LDA           LDA *CSEQ,X  :GET COMMAND FROM CONTROL SEQ
103E-  F0 F8     BEQ           BEQ SINT     :ZERO ENDS THE SEQUENCE, BRANCH
                          :
                          :A NEW COMMAND FROM THE SEQUENCE. FIRST USE IT TO SET OR CLEAR THE
                          :THE GLIDE CONTROL BIT FROM THE TRANSPOSE BUFFER.  IN THE PROCESS,
                          :THE NEW COMMAND IS SHIFTED ONE BIT TO THE LEFT; WHICH MULTIPLIES
                          :THE TEMPO VARIABLE BY 2 AND SHIFTS THE UP/DOWN AND LEGA/STACC BITS
                          :INTO MORE EASILY TESTED POSITIONS.
                          :
1040-  85 74     STA      GLID STA *SCTL    :SAVE SEQUENCE ENTRY IN CONTROL BUFFER
1042-  B9 C0 00  LDA           LDA TTBL,Y   :GET THE CURRENT TRANSOSE BUFFER ENTRY
1045-  2A        ROL           ROL          :ROTATE GLIDE BIT TO CARRY
1046-  06 74     ASL           ASL *SCTL    :ROTATE CONTROL WORD GLIDE TO CARRY
1048-  6A        ROR           ROR          :ROTATE CARRY TO GLIDE BIT
1049-  99 C0 00  STA           STA TTBL,Y   :THEN RETURN TO TRANSPOSE BUFFER
                          :
                          :THIS LITTLE ROUTINE DETERMINES WHETHER SCAN IS UP OR DOWN AND
                          :INITIALIZES THE POINTER TO THE PROPER VALUE
                          :SKYP-SET KEY POINTER
                          :
104C-  A2 07     LDX      SKYP LDX 07       :PREPARE FOR ARP. UP INITIAL POINTER
104E-  24 74     BIT           BIT *SCTL    :CHECK COMMAND BUFFER - ARP. UP?
1050-  30 02     BMI           BMI STIM     :YES, BRANCH
1052-  A2 00     LDX           LDX 00       :NO, ARP. DOWN INTIAL POINTER
                          :
                          :NOW THE ROUTINE TO RESET THE TIMER. SINCE ALL KEY POINTER MANIPULATIONS
                          :WIND UP AT THIS POINT, THE FIRST INSTRUCTION IS TO SAVE THIS POINTER
                          :THE TIMER VALUE IS EXTRACTED FROM THE CONTROL WORD SCTL
                          :STIM-SET TIMER
                          :
1054-  86 73     STX      STIM STX *PNTR    :SAVE INPUT BUFFER POINTER
1056-  A9 1F     LDA           LDA 1F       :PREPARE MASK AND
1058-  25 74     AND           AND *SCTL    :GET THE TIMER (TEMPO) VALUE
105A-  85 72     STA           STA *TIMR    :AND SAVE IN THE TIMER VARIABLE
                          :
                          :NOW WE GET THE CURRENT NOTE OF INTEREST FROM THE INPUT BUFFER
                          :AND IF THE KEY IS NOT DOWN, A CHECK IS MADE TO SEE IF ANY KEYS
                          :ARE DOWN. IF NONE ARE, THE TIMER IS TRICKED INTO TIMING OUT THE
                          :NEXT TIME THROUGH WHICH WILL THEN RESULT IN THE WHOLE COMMAND
                          :SEQUENCE FOLLOWING SYSTEM BEING RESET
                          :
105C-  B5 E0     LDA           LDA *KTBL,X  :GET THE CURRENT KEY FROM INPUT BUFFER
105E-  D0 0D     BNE           BNE BOUT     :IF ZERO, NO KEY - BRANCH
1060-  24 E7     BIT           BIT *KTBL+07 :ARE ANY KEYS DOWN?
1062-  70 BF     BVS           BVS ADVA     :YES, BRANCH
1064-  A9 01     LDA           LDA 01       :NO, PREPARE TO MAKE TIMER RUN OUT
1066-  85 72     STA           STA *TIMR    :NEXT PASS THROUGH
1068-  B9 D0 00  LDA      CLRN LDA NTBL,Y   :GET THE CURRENT OUTPUT NOTE
106B-  29 BF     AND           AND 0BF      :CLEAR THE TRIGGER FLAG
106D-  99 D0 00  STA      BOUT STA NTBL,Y   :AND REPLACE IN OUTPUT BUFFER
1070-  60        RTS           RTS          :RETURN
                          :
```

```
                               :NOW SOME TEMPORARY LOCATIONS AND THEIR INITIAL STATES
                               :
1071  00                       TEMP .HS 00
1072  01                       TIMR .HS 01
1073  08                       PNTR .HS 08
1074  C4                       SCTL .HS C4
1075  07                       PPNT .HS 07
1076  07                       SPNT .HS 07
1077  000000                   CSEQ .HS 000000000000E404
                                    .OR 10E8
                               STUP .HS 402004
                                    .OR 1100
                               :THIS IS THE COMMAND KEY DECODING AND SEQUENCE EDITING SUBROUTINE
                               :# OF COMMAND KEY IS IN Y REGISTER
                               :
1100-  A6 75     LDX           CMND LDX *PPNT    :GET THE EDITORS POINTER TO COMMAND SEQ
1102-  B5 77     LDA                LDA *CSEQ,X  :GET THE COMMAND POINTED TO (IN ACC, DON'T FORGET)
1104-  C0 10     CPY                CPY 10       :IS KEY 10 - CLEAR COMMAND SEQUENCE
1106-  F0 0F     BEQ                BEQ CLR      :YES, BRANCH
1108-  90 1F     BCC                BCC CNXT     :NO, IT'S LESS THAN "F", BRANCH
                               :
                               :THE KEY IS 11 OR GREATER.  EXCHANGE THE COMMAND POINTED TO WITH
                               :TEMPORARY STORAGE LOCATION TEMP. NOTE THAT THIS CAN BE USED TO
                               :EXCHANGE TWO OR MORE COMMANDS IN THE SEQUENCE
                               :
110A-  A4 71     LDY                LDY *TEMP    :GET THE COMMAND IN THE TEMPORARY BUFFER
110C-  D0 04     BNE                BNE ELP0     :IS THE COMMAND FROM TEMP A 0? NO, BRANCH
110E-  E0 07     CPX                CPX 07       :POINTING TO FIRST COMMAND?
1110-  F0 2B     BEQ                BEQ RTN      :YES, BRANCH. DON'T WRITE ZERO AS FIRST COMMAND
1112-  94 77     STY           ELP0 STY *CSEQ,X  :PUT COMMAND IN THE SEQUENCE SLOT POINTED TO
1114-  85 71     STA                STA *TEMP    :AND THEN SAVE OLD COMMAND IN THE TEMP LOCATION
1116-  60        RTS                RTS          :THEN RETURN
                               :
                               :THE KEY IS "10", CLEAR THE COMMAND SEQUENCE. NOTE THAT THE FIRST
                               :ENTRY IN THE SEQUENCE IS NOT CHANGED.
                               :
1117-  A2 07     LDX           CLR  LDX 07       :SET POINT TO FIRST SEQUENCE ENTRY
1119-  86 75     STX                STX *PPNT    :AND SAVE IT
111B-  CA        DEX                DEX          :DECREMENT THE POINTER(SKIP FIRST ENTRY)
111C-  A9 00     LDA                LDA 00       :AND GET READY
111E-  8D 20 18  STA                STA DISP     :ZERO THE DISPLAYED EDITOR POINTER
1121-  85 71     STA                STA *TEMP    :AND THE EXCHANGE REGISTER
1123-  95 77     STA           CLLP STA *CSEQ,X  :ZERO THE SEQUENCE ENTRY
1125-  CA        DEX                DEX          :AND POINT TO NEXT ENTRY
1126-  10 FB     BPL                BPL CLLP     :SOME LEFT, LOOP
1128-  60        RTS                RTS          :RETURN
                               :
                               :NOW WE TEST FOR "E" OR "F", BACKSPACE OR ADVANCE THE EDITOR'S
                               :EDITOR'S POINTER TO THE COMMAND SEQUENCE. NOTE THAT INCREMENTING THE
                               :POINTER PRODUCES A BACKSPACE.
                               :
1129-  C0 0E     CPY           CNXT CPY 0E       :IS KEY "E" OR "F"?
112B-  90 18     BCC                BCC STMP     :NEITHER AND LESS THAN "E", BRANCH FOR NEXT TEST
112D-  F0 0F     BEQ                BEQ BACK     :IT'S "E", BRANCH TO BACKSPACE
112F-  CA        DEX                DEX          :IT'S "F", ADVANCE THE POINTER
1130-  30 0B     BMI                BMI RTN      :AND IF OUT OF RANGE, BRANCH TO LEAVE IMMEDIATELY
1132-  86 75     STX           COUT STX *PPNT    :SAVE NEW POINTER
                               :
                               :IN THIS SECTION THE POINTER (WHICH IS 07 FOR THE START OF THE SEQUENCE
                               :AND 00 AT THE END) IS CONVERTED TO AN INCREASING NUMBER FROM 0-7 FOR
                               :DISPLAY PURPOSES.
                               :
1134-  8A        TXA                TXA          :POINTER TO THE ACCUM. FOR A CALCULATION
1135-  38        SEC                SEC          :PREPARE FOR A SUBTRACTION
1136-  E9 08     SBC                SBC 08       :TWO'SD COMPLEMENT
1138-  49 FF     EOR                EOR 0FF      :COMPLEMENT OF THAT
113A-  8D 20 18  STA                STA DISP     :SHOW VALUE IN THE DISPLAYS
113D-  60        RTS           RTN  RTS          :RETURN
                               :
                               :BACKSPACE POINTER AND MAKE SURE IT IS STILL IN RANGE, THEN BRANCH
                               :
113E-  E8        INX           BACK INX          :BACKSPACE THE POINTER
113F-  E0 08     CPX                CPX 08       :OUT OF RANGE?
1141-  F0 FA     BEQ                BEQ RTN      :YES, BRANCH TO LEAVE IMMEDIATELY
1143-  D0 ED     BNE                BNE COUT     :NO, BRANCH ALWAYS TO SAVE POINTER, ETC.
                               :
                               :IF THE KEY IS ONE OF THE TEMPOS, ADD 1 (0 TEMPO NOT ALLOWED) AND
                               :FIT IT INTO THE CONTROL SEQUENCE ENTRY POINTED TO
                               :
1145-  C0 08     CPY           STMP CPY 08       :TEMPO KEY?
1147-  B0 0A     BCS                BCS SGLD     :NO, BRANCH
1149-  C8        INY                INY          :YES, ADD 1 TO KEY #
114A-  29 F0     AND                AND 0F0      :MASK PRESENT TEMPO IN COMMAND TO ZERO
114C-  95 77     STA                STA *CSEQ,X  :SAVE CONTROL FLAGS IN CSEQ TEMPORARILY
114E-  98        TYA                TYA          :BRING NEW TEMPO TO ACC
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114F-   15 77     ORA          ORA *CSEQ,X  :COMBINE WITH OLD CONTROL FLAGS
1151-   D0 1A     BNE          BNE SAVA     :BRANCH ALWAYS
                          :
                          :NOW A SERIES OF TESTS WHICH RESULT IN THE CARRY BIT BEING SET OR
                          :CLEAR. A SERIES OF ROTATES BRINGS THE CARRY TO THE APPROPRIATE BIT
                          :IN THE COMMAND WORD
                          :
1153-   2A        ROL     SGLD ROL          :ROTATE THE GLIDE COMMAND BIT TO CARRY
1154-   08        PHP          PHP          :AND SAVE THE CARRY ON THE STACK
1155-   C0 09     CPY          CPY 09       :IS KEY GLIDE ON OR OFF?
1157-   F0 12     BEQ          BEQ ROT1     :9-GLIDE ON, BRANCH
1159-   90 10     BCC          BCC ROT1     :8-GLIDE OFF, BRANCH
                          :
                          :THE KEY WAS NEITHER GLIDE ON NOR OFF, TEST FOR DIRECTION UP OR DOWN
                          :
115B-   28        PLP     SMOD PLP          :GET THE OLD GLIDE BIT FROM THE STACK
115C-   2A        ROL          ROL          :ROTATE DIRECTION BIT TO CARRY
115D-   08        PHP          PHP          :SAVE IT ON STACK
115E-   C0 0B     CPY          CPY 0B       :IS KEY UP OR DOWN?
1160-   F0 08     BEQ          BEQ ROT2     :B-UP, BRANCH
1162-   90 06     BCC          BCC ROT2     :A-DOWN, BRANCH
                          :
                          :THE KEY HAS TO BE C OR D (STACCATO OR LEGATO)
                          :
1164-   28        PLP     SDIR PLP          :GET THE OLD DIRECTION BIT
1165-   2A        ROL          ROL          :STAC/LEGA BIT TO CARRY
1166-   08        PHP          PHP          :SAVE IT
1167-   C0 0D     CPY          CPY 0D       :CARRY SETS IF KEY IS "D" - LEGATO
1169-   6A        ROR          ROR          :ROTATE COMMAND WORD BACK INTO PLACE
116A-   6A        ROR     ROT2 ROR
116B-   6A        ROR     ROT1 ROR
116C-   28        PLP          PLP          :WASTE A STACK SLOT TO COMPENSATE
116D-   95 77     STA     SAVA STA *CSEQ,X  :SAVE THE COMMAND WORD IN THE SEQUENCE
116F-   60        RTS          RTS          :RETURN
                          :
                          END  .EN
```


EQUIPMENT EXCHANGE

List equipment or services for sale, trade, or wanted, plus job openings, positions wanted, events and so on. Keep listings brief; enclose $1.00 for each 10 words. Prices, zip, and phone numbers count as a single word each. Display classifieds are available for commercial ads; rates are $10 per column inch (minimum, negative supplied). Respond directly to the advertiser. Please don't write to POLYPHONY. Polyphony is not responsible for claims made in ads, or the results of any transactions. We reserve the right to edit or refuse any ads submitted.

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WANT TO CORRESPOND with anyone who has used microcomputers to customize non-Paia monotonic synthesizers to polyphonic. Ron Rockwell, 25 Pleasant St., Bethel, CT 06801

PAIA MODULES: 8700 Computer with PS-87 and CS-87, D/A with Quash, (3) 4720 VCOs, 4730 VCF, (2) 4710 Balanced Modular/ VCAs, (2) Envelope Generators, 4711 Stereo Mixer, Reverb unit, 2720-5 LFO/ Noise, 8782 Keyboard, (3) Watt Blocks. All assembled, cabinets available. Guy Kilpatrick, 510 College, Apt #416, Grand Rapids, MI 49503, (616) 451-0578.

PAIA 4700/S with Envelope Follower, $500. Oz, $85. Gnome, $50. Phlanger, $60. All fully assembled and working. Dave Mason, 2222 Harriman Ln., Redondo Beach, CA 90278, (213) 379-5601.

DIGITAL GROUP Z80 computer, 10K RAM, video and cassette interface, four parallel I/Os, 12 A power supply, ASCII keyboard, documentation. $750 or best offer. Marvin Jones, Box 20305, Oklahoma City, OK 73156.

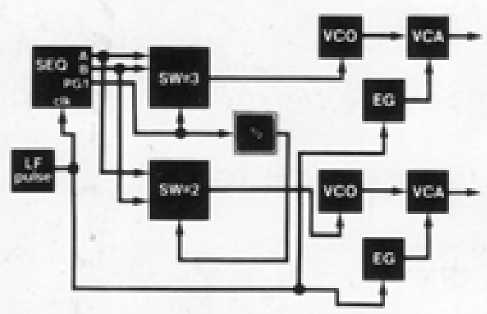
Sequencer & Switches Contest

We at Aries Music are happy to announce our first contest, the subject being discovery of interesting uses for our new AR-334 Sequencer and AR-335 Switches modules. The AR-334 is a potentiometer-memory, 8-step by 2 row sequencer with position gate outputs along with reset and run, enable & step inputs. The AR-335 is a unique set of 4 bidirectional analog switches: 2 single pole, double throw (pulse controlled); 1 single pole, four throw (pulse controlled); and 1 single throw, four throw (voltage controlled).

Contestants will be asked to submit a block diagram for a patch which takes advantage of as many features of these modules as possible, while producing a musically interesting and useful result.

The twenty best patches and their descriptions will be published and sent to all of the entrants in the contest.

The five finalists will have their patches and descriptions published in *Polyphony*, whose readers will be invited to vote for the best patch. The winner will be able to select $600 worth of Aries Music modules. (Which, coincidently enough, is the value of one each, Sequencer and Switches, assembled, or three modules in kit form.)



This patch produces a cyclical pattern of 32 notes.

The finalists will be selected by Bob Snowdale, president of Aries Music; Ron Rivera, designer of the modules; and Mark Styles and Ken Perrin, noted Boston area composers of electronic music.

So send today for the contest details, which will include the 47-page Aries Music *Owner's Manual Supplement* on the AR-334 and AR-335 modules. The contest entry deadline is January 31, 1980.

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