POLYPHONY

T.M.

ELECTRONIC MUSIC & HOME RECORDING

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1980

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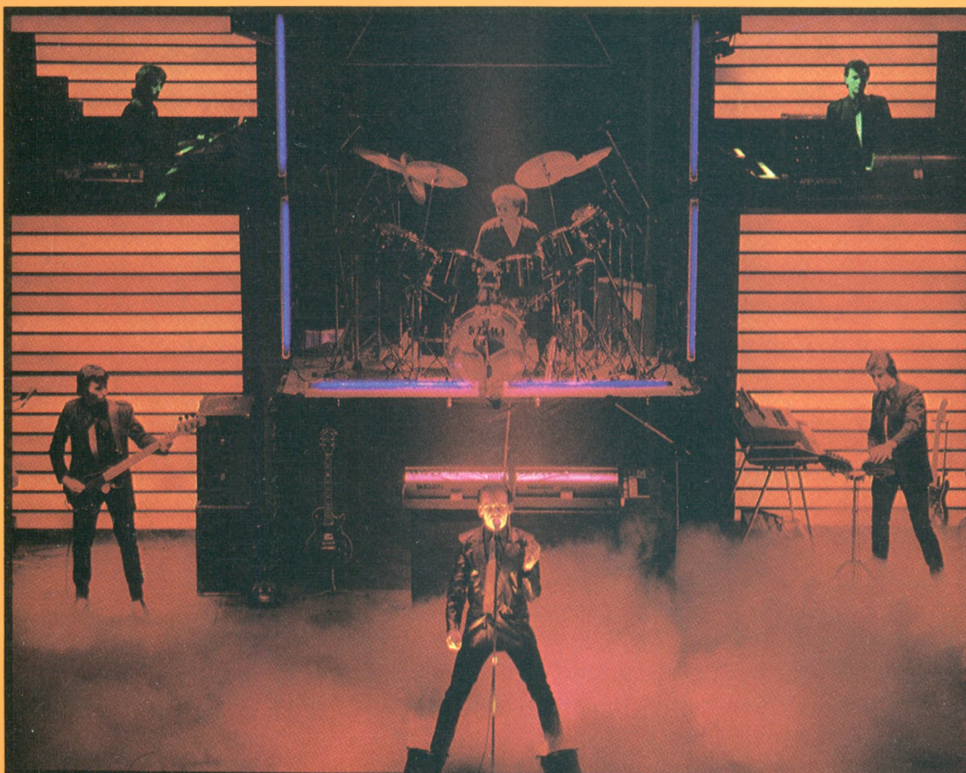
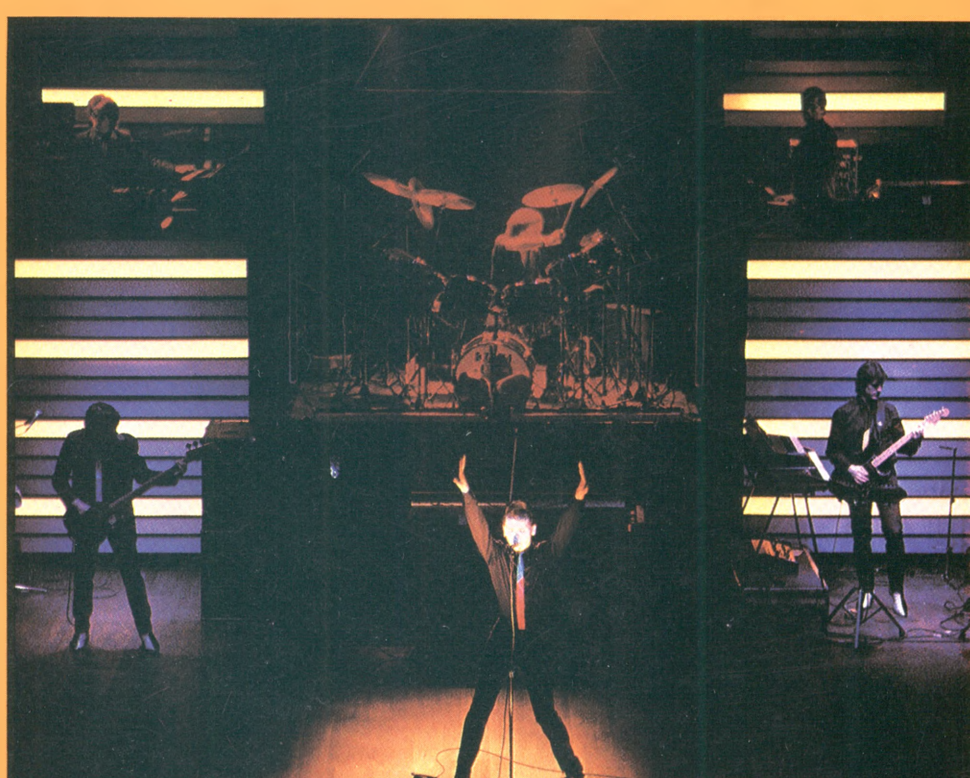


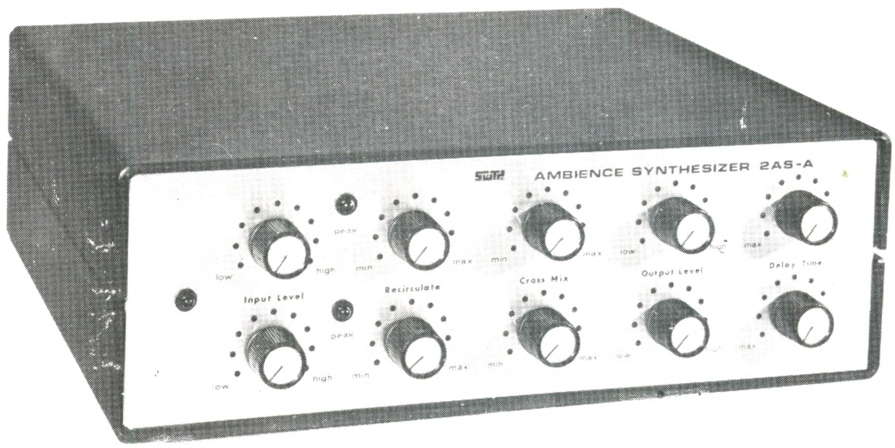
GARY NUMAN

BUILD Digital Delay



Writing Documentation

SWTPC ANALOG DELAY



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May/June 1980

Volume 6, Number 1

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FEATURES

COLUMNS & REGULARS

LAB NOTES is taking a short vacation (the first time in four years) while author John Simonton finishes a few of his many other projects.

ON THE COVER: Gary Numan in San Francisco.
Photos Copyright 1980 by Ken Friedman.

The last cover photo was "Violin and Chips"
Copyright 1980 by Marvin Jones.

"I use not only the brains I have, but all I can borrow."
-Woodrow Wilson

LETTERS

PMS Questions

(Addressed to author Ken Perrin) In your extremely informative article (Using Sophisticated Modifications in Creative Synthesis, September/ October 1979) you mention the Lissajou patterns. (1) In figure 13 (page 30), what is the effect if the two sine waves (90 degrees out of phase) are of different frequencies? (2) What is the effect if the two waves (90 degrees out of phase) are of different forms, i.e. a square versus a sawtooth wave, etc.? (3) Are the circles in example 14 actually rotating, are the straight lines stationary? (4) Could the horizontal and vertical designs and their interactions also be the result of two orchestrated segments in and out of phase? My sincerest thanks for any help you can offer.

Maury Deutsch
Brooklyn, NY

First off, let me respond to two of the questions simultaneously by answering neither of them directly, but by talking instead about an oscilloscope.

Internal to the scope is a sawtooth oscillator which pushes the beam along the X or horizontal axis. If you look at a scope which has no signal at its input, you will see a single horizontal line. This is the result of a single dot being pushed along the X axis as the sawtooth goes high and snapped back to the beginning as the sawtooth goes low. Of course, if this dot moves fast enough, we see it as a solid line. When a signal is displayed on the scope it is a voltage which pushes the beam along the Y or vertical axis. Assume, for the time being, that it's a sine wave and if the frequency of the internal sawtooth were identical to the displayed sine wave, then we would see one cycle of the sine displayed on the screen. If the sine wave were twice the frequency of the sawtooth we would see two cycles of the sine.

But let's assume the sawtooth and sine are of the same frequency but have a phase difference of 90 degrees; the sawtooth starts 90 degrees after the sine. In this case we still have one cycle of the sine displayed but it apparently starts at its highest point, swings low, and then swings back to its highest point. In spite of the fact that it looks like the curved part of the Greek letter PSI, it is still a sine wave.

Now, take a different case, or maybe a different scope. This scope has an input which allows the user to use any waveform in lieu of the internal sawtooth. I'd assumed in the article (without actually stating it) that we were using this kind of scope to do the displays. Let's start simply.

Obviously, if we use a sine wave to deflect the beam horizontally and a sawtooth of the same frequency to deflect it vertically, we have the same display as in the first example three paragraphs ago except this time it's turned on its side. The sine wave begins at the bottom of the screen and then goes upward.

Next, consider the case of two sines of the same frequency and phase. In other words, a case where X = Y. As the X sine wave pushes the beam horizontally, the Y sine wave pushes it vertically. We end up with a 45 degree line, just as you'd expect when X = Y. If we slow down both sines to about 1 Hz we could watch the dot move upwards into the +X+Y quadrant, back to 0, then down to the -X-Y quadrant. When the sines are 180 degrees out of phase we have X = -Y and so we get a line perpendicular to the first line. When the sines are 90 degrees out of phase, the beam circumscribes a circle by moving clockwise; it moves counter- clockwise if the sines have a 270 degree phase difference.

So, above are the answers to the second and third questions. The answer to the first question is easy. It doesn't make sense to talk about the phase difference of waveforms having different frequencies. The fourth question is the tricky one because it encourages me to trot out my ignorance and show it in print. What is an orchestrated segment? I'm going to assume that it means some part of a piece which has been arranged for orchestra. Deciding that it is a musical term is a gamble because the phrase has such a trigonometric ring to it, nonetheless -- you only go around once. Besides, it makes the answer easier.

The answer is that it doesn't work that way. If you had a microphone hooked up to the X axis and another microphone to the Y axis and somehow got the signals to difference by 90 degrees, the display would be nothing but jumbled lines. You can see why if you consider the characteristics of the waveforms generated by such sources.

Jean Paul Fourier (contemporary of Robespierre, Jefferson, Beethoven, and that crowd and also the deviser of the mathematical description of wave motion) says that a waveform has no beginning or no end. He wasn't being cosmic, but rather mathematically correct. Another waveform characteristic is that it repeats its cycles without variation. Neither of these are characteristic of a musical tone. Musical instruments produce boring scope displays and interesting scope displays are the sonic pits.

Ken Perrin
Boston, MA

Chameleon Problems

I saw Jon Balleras' Chameleon 0.25 program in your Nov/Dec 1979 issue which I thought would be really nice for playing live, but I have encountered a few problems and was wondering if you could help me out.

The first time I tried to run the program I couldn't make any sense at all. Then I noticed that in the listing, starting after address $024E the listing jumped to $027C and continued out to what should have been the end of the program at address $02A2. But it doesn't end there; instead it continues on with the part of the listing that was skipped. In other words, the part of the program from address $027C to $02A2 should be interchanged with the part of the program from address $024F to $027B. If you examine the program listing starting at address $024E it's fairly easy to notice. This presented no real problem as I found the mistake soon after I started looking for the trouble.

My biggest problem was that after I had corrected the first problem, the program appeared to be running fine. The transpose was working; the glide was working; also the clear and tune functions were working fine. But when I hit any of the pads designating the number of voices to be used, the program shut itself down and changed the value of $07 at address $0220 to $03.

I was just wondering if there is another misprint in the listing that I can't find, or what? Any help would be appreciated.

Charles Brown
Montgomery, AL

How about it readers? Have others experienced similar problems? Let us know so we can publish the corrections.

Marvin

CA3280 Updates

I will not let Polyphony err twice! The 24 dB/octave filter presented in "Using the CA3280" by James Patchell (Jan/Feb '80 issue) is guaranteed not to work. As stated, the circuit is similar to Gary Bannister's in the May/June '79 issue, and both circuits share the same fate: oscillation. This is inevitable because, to paraphrase Routh and Hurwitz (remember them?), it is a necessary condition for stability that each coefficient of a complex polynomial be non-zero; in other words, each integrator output must drive some type of feedback loop. Obviously, the first, second, and third integrators in Mr. Patchell's design do not drive feedback loops. (And yes, I have constructed the circuit to verify its instability.)

Readers would be better off building a state-variable circuit like that described in the article when using the CA3080 or CA3280 because the Gm from one OTA to another can vary as much as 100%. (A 24 dB filter requires some way to match the OTAs.) If 24 dB is necessary, I would suggest using the CEM or SSM filter IC; the price/ performance combination is hard to beat.

Howard Cano
Arvada, CO

There were a few boo-boo's in my article, Using the New CA3280. In figure 1, the terminal V2 should have a negative, not a positive, sign. In figure 3a, the TL074 pin #1 should be pin #3; pin #3 should be pin #1. This was my error; sorry. Figure 10 shows the 75K R18 incorrectly connected to -15 volts; it should tie to +15 volts.

Also, some of my friends were unclear on what I meant by Vin Peak in figure 5. 5 volts peak = 10 volts peak-to-peak or 3.5 volts RMS AC. I used peak because the circuits can respond to DC.

Speaking of figure 5, there seems to have been a slight error when the plot was redrawn. It would be OK with me to send readers who are interested a photocopy of the original if they will send me a stamped addressed envelope. The error shows up mostly at 6 volts. In my article I say that the diode current is .35 mA where the chart says .34 mA. A small error, but I am unfortunately a perfectionist.

James Patchell
533 San Blas Pl
Santa Barbara
CA 93111

Garson Fan

I really enjoyed the article on Mort Garson in the Jan/Feb issue. I've been following Garson's work ever since I heard "Electronic Hair Pieces" (A&M SP 4209, not 4029 as stated in the discography) ten years ago. The music was so far advanced beyond the other commercial Moog stuff being foisted on the public that I couldn't understand why Garson was being hidden from the public eye. I began to notice his touch in many TV commercials and game show themes; but there wasn't much under his name in the Schwann catalog. I finally did gather together just about all the albums you mentioned in the article, and I was utterly fascinated to hear about what really went on in the production of the Elektra "Zodiac" album!

I must take issue with Ms. Bryant on one significant point; she states that "it was with the 1967 release of his album 'Zodiac' that the public was first introduced to the phenomenon of synthesized sound." I can't let that go by without a challenge -- perhaps Ms. Bryant is too young to have any perspective in this regard. Leaving aside all the avant-garde composers who had been laboring away in EM for decades before 1967, the real introduction of 'the public' to the phenomenon of synthesized sound had to be the 1955 album "The Sounds and Music of the RCA Electronic Music Synthesizer" (RCA LM-1922), which just may have been the first use of the word Synthesizer! RCA's room-sized extravaganza was certainly the first significant total system for producing sounds from scratch, and the album itself is an interesting mini-course in synthesis.

Another introduction of the public to synthetic sound was the way-ahead-of-its-time totally electronic score to the 1956 MGM film "Forbidden Planet" (available on Planet Records PR-001) by Louis and Bebe Barron. The Barrons even used some computer-like circuits to generate sound patterns they could build upon.

Taken as a whole, though, the Garson article was a really fine job, and a bit of overdue recognition to one of the most talented systhesists - and musicians - around.

Steve Klein
San Bernadino, CA

Reading Schematics

I subscribed to Polyphony in August of '79. I have been very pleased with the articles but I feel I am missing a lot. The reason behind that is ... I don't have any knowledge about reading schematics. Are there any books you couls refer me to so I could put my feet on the right path?

Thank you for your time and a great magazine.

Jeff Bundy
Fayetteville, NC

My first recommendation would be Craig Anderton's Electronic Projects for Musicians, which is an excellent introduction to all aspects of electronic music projects which musicians and 'non-tech' types might not be familiar with. Also, look for How To Read Schematic Diagrams by Donald Herrington (Howard Sams #21127, $5.50). These should get you going.

Marvin

Drum Mods Wanted

While I find the overall design and usability of the Paia Programmable Drum synthesizer better than most drum machines, I think a couple of additions and changes could make it even more useful.

1. Three Discrete Outputs - When you put the drum output through reverb, it really muddles up the bass before it has much effect on the snare or other higher components. A provision for three separate outputs, one for snare, one for bass and accent bass, and one for the other oscillators, would be very helpful. You could put the bass through straight, put the snare through a VCF and put reverb on the rest.

2. Lower Drum Sounds - Lower and deeper outputs from the tom and conga would be far more useful. As they are now, they are fine for a salsa band or if you are anticipating the 1983 mambo and bossa-nova revival, but for rock or disco they leave much to be desired. With a deeper drum sound you would lose that sound which is characteristic of all 'drum machines'.

3. Cassette Load and Dump - This would make programming a whole lot easier (but is least practical of the three).

continued on page 29....

About The Cover...

GARY NUMAN

by kirk austin

The house lights dimmed as the curtain rose on a dark stage. Slowly the dry-ice fog began to creep onto the stage, and then out into the audience. Audience and stage became one as the fog engulfed the first twenty rows, and the solitary synthesizer line that is the beginning of "Tracks" echoed through the auditorium. The drummer, playing sixteenth notes on his hi-hat, joined the synthesizer as the fog and darkness added an other-worldly atmosphere to the piece. When the bass finally joined in, the stage became alive with light --- huge columns of alternating white and purple light.

This is how the show began March 8, at the Fox-Warfield theatre in San Francisco. Gary Numan had not even appeared on the stage yet, but the audience was already in awe of the spectacle. The instrumental from the Pleasure Principle album was a perfect opening number, allowing the audience to concentrate its attention on the visual aspects of the stage set-up. When Gary Numan made his appearance after the first tune he was just as hypnotic --- moving slowly and deliberately in a stylized pantomime. For the most part he did not play the instruments, but stood up front and talked/sang the lyrics. The five other musicians carried the music well, though, and were able to duplicate the recordings accurately.

Needless to say, the 21 synthesizers that Numan carries with him on tour provided the bulk of the sound. Minimoogs, Polymoogs, ARPs, Rolands, and racks of outboard gear (digital delays, flangers, etc.) brought the multi-tracked records alive in vivid fashion. A couple of Snyare percussion synthesizers were also used in order to duplicate the multiple hand clap tracks on the records.

When I talked to Gary Numan I found him to be quite different than what I would have expected. For starters, the door to the hotel room was answered by his mother. His father, Tony Webb, gave me the once over before introducing me to Gary. I guess that this is not too bizarre since Gary had just turned 22 that day, but I was rather surprised.

Numan is a pretty soft-spoken person, and he didn't talk about cybernetics, or the future, or any of the other things that the record company promo says he talks about. He mostly talked about manipulating the media, making his fortune, and getting out of the business as soon as possible (within a year or two). He is very deliberate and calculating, and his success has been carefully planned by himself and his father. Gary is aware of what it takes to sell an image, and what the market is prepared to accept. His real interest, though, is air planes, and he hopes to make enough money from the music business to finance a charter airline for himself. He already has all of his hours completed, and just needs to take some tests in order to get his pilots license.

Obviously, Numan has definite ideas about music, too. Although he originally started out as a guitar player, he was fascinated by a synthesizer that was set up in a studio he was using to record some punk demos (after he got a recording contract he dropped the punk act). Gary found the synthesizer to be an extremely powerful sounding instrument, and he couldn't understand why it was not being used by more bands. Even the bands that did use synthesizers (Ultravox, Roxy Music) always used them in supporting roles, still relying on the guitar for the basic rhythmic structure. Numan thought that the synthesizer should be used in place of the rhythm guitar, building the rest of the song around it. This idea led to his first album, Tubeway Army, which was done with his old friend Paul Gardiner on bass, and his uncle, Jess Lidyard, playing drums. The album was not well received, though, and Numan went back into the studio to record Replicas with the same band. Replicas was released six months after Tubeway Army, and went to number one on the English charts. The single,"Are Friends Electric?", also made number one, and Gary was on his way.

He still didn't have a band that could perform the songs, due to the extensive overdubbing that was used on the records, so he went back into the studio and made the Pleasure Principle album. This album was released six months after Replicas, and also went to the number one spot on the English charts. Gary Numan had three albums out within one year, and they were all selling like mad (the Tubeway Army LP found its way into the top twenty after the success of the other albums).

Gary does not consider himself to be a very good musician, and, in fact, plays "one-finger" synthesizer. He has realized the potential of the instrument, though, probably better than his musical contemporaries. He manages to make relatively simple patches musically effective. Out of all of his equipment he still prefers the Minimoog and its three oscillator sound over the other synthesizers at his disposal. It's strange how the success of the Minimoog has not inspired other synthesizer manufactures to make three oscillators the minimum number required to create a voice. According to Gary: "The three oscillators make it sound much fatter. I can always get a good sound from the Mini".

Numan also likes non-tonal sounds like the 'wash' on "Me I Disconnect From You", and the sample-and-hold on "Praying To The Aliens". One of the few times that he did play the synthesizer during the performance was when he held down an arbitrary key for the solo on "Praying To The Aliens".

Another favorite of Gary's is the MXR digital delay. Gary said, "I love MXR, and I buy anything that they come out with. I bought all of the available digital delay units before I left England." The flanging and automatic double tracking that was so prominent on the records was reproduced with the MXR gear.

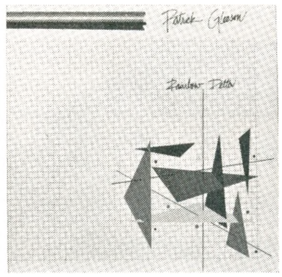
Expect one more album from Gary Numan before he hangs up his rock and roll shoes. His career in music will be short, but he has taken the synthesizer out of the background and given it a place next to the glorified guitar power chord. Electronic music will never be the same.

Editor's Note: For those of you wanting additional information about Gary Numan, Kirk will have an extended interview with Gary Numan in the July/August issue of M.I. Magazine (Box 6395, Albany Station, Berkeley, CA 94706).

INDUSTRY REPORT

'Tell Them You Saw It In Polyphony'

POLYPHONY PUB.CO. VOLUME 6 / ISSUE 1



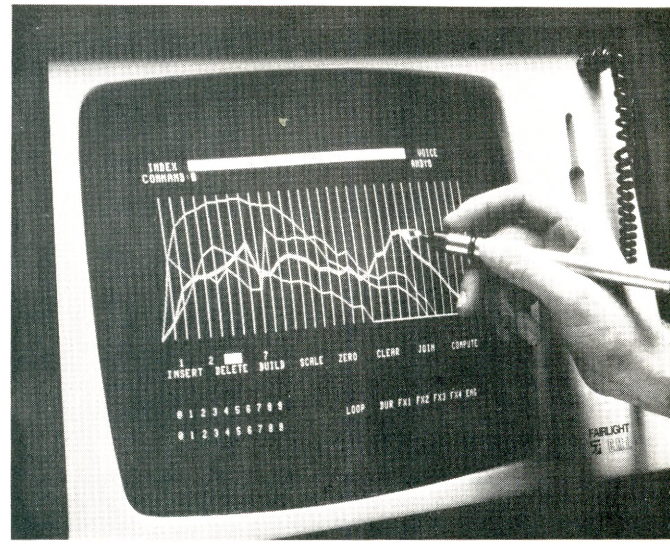
NEW ALBUMS

Those readers interested in Roger Powell's Air Pocket album and Pat Gleeson's I Just Got Here Myself album previously reviewed in Polyphony will be happy to hear that both have finally been released. Pat Gleeson's album has been renamed Rainbow Delta and has been released by PVC Records (#PVC 7914), but is otherwise as reviewed in the Jan/Feb 1979 Polyphony. Rainbow Delta is available through Polyphony's PolyMart mail order service. Roger Powell has released Air Pocket on the Bearsville label with a rearranged playing sequence, but otherwise as reviewed in the May/June 1979 Polyphony. Air Pocket is not currently stocked by PolyMart; if you have trouble finding it, let us know and we will get it for you.

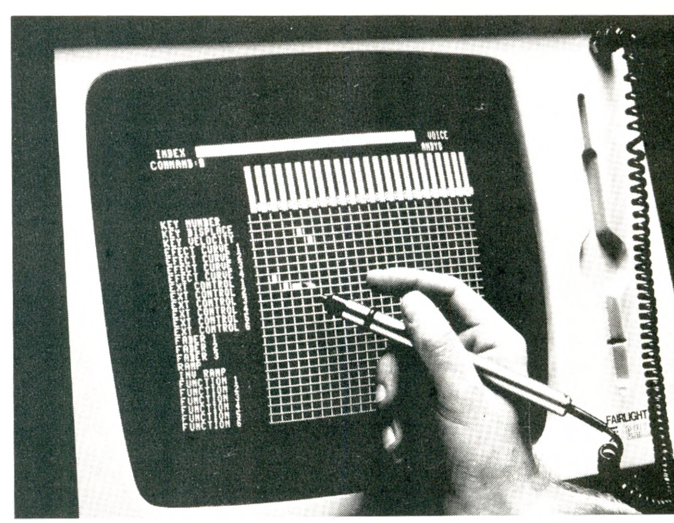
STATE OF THE ART

A new development has been made in electronic musical instruments. The concept of the instrument is that ANY sound whatever may be "played" in a controllable and musical fashion, on a musical keyboard. It can also be "programmed" by the musician to perform itself.

The unit is called the "Fairlight C.M.I." (Computer Musical Instrument), and uses digital sound generation techniques in conjunction with microcomputers to allow the most subtle variations in the characteristics of "sound" to be either "recreated" or "created" (synthesized).



In the case of "recreated" sounds, a sample of any sound may be inputted via a microphone or tape-recorder, stored, and then "played" musically. The characterisitics of the sound may be studied on a video graphics display and manipulated using a "light-pen".



In the case of a "created" sound, a number of methods are used whereby sounds can be synthesized. One of these methods is by "drawing" a series of harmonic profiles (envelopes) each with its own character of attack and decay.

In another mode, sounds can be created by "drawing" the complete sound (waveform) from beginning to end using the light-pen. By examining and manipulating the waveforms of sampled or synthesised sounds an insight may be gained as to the natural laws governing the creation of sounds.

The "Fairlight C.M.I." can be used for "live performance" as well as educational and studio use. Eight separate "voice generating" channels are provided allowing up to 8 notes or 8 different sounds to be played simultaneously.

Although the system is to some extent capable of "playing" the sounds of existing classical instruments (by nature of its ability to reproduce a "sampled" sound), this is not the intention of it's design. More importantly, it is expected that the instrument will free the musician and composer to use not only sounds of existing instruments, but

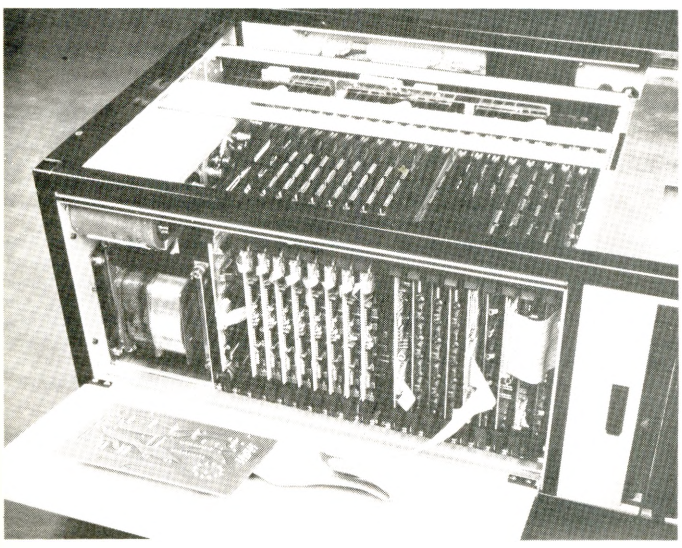
also the infinite variety of sounds occuring "naturally" or created by the musician.

The capabilities of the instrument are largely determined by the programs (software) which are loaded into it via two double-sided floppy diskettes. This allows the system to be enhanced as new programs are made available, either from other users or from Fairlight Instruments.

Fairlight Instruments was formed in 1974 by Kim Ryrie and Peter Vogel for the purposes of designing the C.M.I. System. They joined forces with Mr. Tony Furse of "Creative Stratagies" who had originally conceived some of the basic principals of digital sound generation in 1965 and had taken out copyrights and provisional patents relating to some of these.

The very first digital waveform generator prototype, built in 1966, used hundreds of diodes, resistors and transistors and was controlled by a "Prime" minicomputer. After nearly eight years of research and development, a system called the QASAR M8 (Multimode 8) was designed using TTL and Dual Microprocessor Technology and made for the School of Music in Canberra, Australia. This was the first of their all digital synthesizers and was installed in 1976.

The new FAIRLIGHT C.M.I. incorporates all of the capabilities of the QASAR M8 system but uses MOS Large Scale Integration technology extensively and incorporates many powerful features that were not possible on the original M8 system. It is also much smaller and less expensive than the original system.



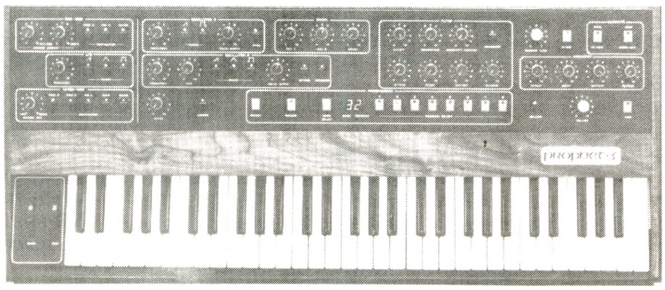
The complete C.M.I. System is the result of fourteen years of development and in the last two years of development has occupied nearly twelve people full time on both software and hardware development.

The C.M.I. is made in Australia by Fairlight Instruments Pty. Limited and licence for the manufacture of some items is granted by Creative Stratagies Pty. Ltd. American distribution and representation is handled by Bruce Jackson, PO Box 401, Lititz, PA 17543. Polyphony highly recommends that you send for their brochure and ask to be put on the mailing list for their newsletter.

ELECTRONIC MUSIC FESTIVAL

The Tenth Annual Electronic Music Plus Festival will take place at the Creative Arts Center of West Virginia University, Morgantown, WV 26506, on October 17-18, 1980. Candidate scores and tapes must be submitted by August 15, 1980. Compositions must be electronic plus live performer(s) and/or multi-media. Live electronic music (with or without tape) of composers/performers will be especially welcome. All submissions will be returned unless the composer indicates that the score and tape is a donation to the West Virginia University Music Library.

For more information, or to submit scores and tapes, contact: Dr. John Beall, Electronic Music Plus, Creative Arts Center, West Virginia University, Morgantown, WV 26506



PROPHET UPDATE

For 2½ years, the Prophet-5 has produced incredible sounds for the world's top musicians. Now, the industry's first completely programmable polyphonic synthesizer from Sequential Circuits has been extensively redesigned and improved. Major technological advances have made it possible to add the following features, among others, to the Prophet-5.

*A built-in CASSETTE INTERFACE enables the transfer and storage of complete sets of 40 programs onto regular cassette tape. The programs can then be reloaded or transferred to any other Prophet. It is possible to store an unlimited number of programs for use at a later date. Virtually any cassette recorder may be utilized, expensive "component" models as well as less expensive portable units.

*A totally unique feature of the Prophet is the ability to use different tuning scales. A special "variable scale" mode allows you to actually tune each of the 12 notes in an octave to different frequencies. The range for each is about $\pm$ ½ semitone from its normal equal tempered value. These tunings can then be programmed into memory, which enables instantaneous switching from one scale or key to the next. Since they reside in memory, they can be stored on cassettes with the other programs for future use. For the first time, in a standard commercially available instrument, the musician can use other scales; such as just intonation, mean tone, Pythagorean, etc., as well as their variations (and different keys). This tuning feature (along with all the other features, of course), makes the new Prophet-5 ideal in educational environments, as well an excellent tool for experimentation and performance by all musicians. Imagine --- a keyboard where the thirds don't beat!

*The EDIT feature has been improved to provide the utmost simplicity in program modification. Turning a knob or hitting the desired switch will immediately modify any program in memory. No special "edit" mode switches need to be touched. The modified program may then be permanently saved, or the old program may be recalled.

*A VOICE DEFEAT system enables you to easily disable a defective voice in an emergency situaiton, even while playing.

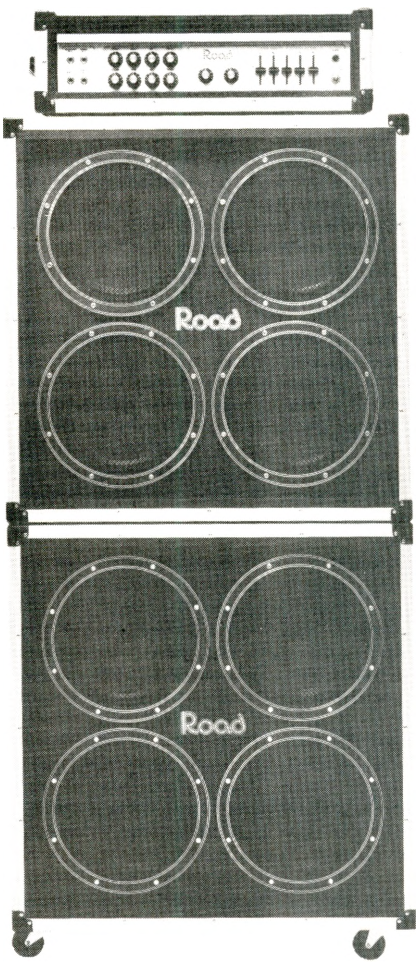
Other advances allow for fewer electronic parts in the Prophet-5. A radically new oscillator design, coupled with a unique new computer-correction scheme, practically eliminates tuning problems. The new circuitry cuts service problems and dramatically improves the realiability and roadworthiness of the instrument. As further support, Sequential Circuits has established a service center network in eight major cities nationwide. Presently, there are centers in New York, Boston, Chicago, Cleveland, Toronto, Washington D.C., Atlanta and Los Angeles, with more to be opened in the next few months. All other features on the Prophet, including the incredible Prophet sound, have not been altered.

The new Prophet lists for $4595. For further information, contact Bob Styles at (408) 946-5240. Sequential Circuits, Inc., 3051 North First Street, San Jose, California 95134.

POLYPHONIC COMPUTER

Dataton AB (Box 257, S-581 02 Linkoping, Sweden) announces a new extended capacity version of their original 3301 Polyphonic Computer, the 3301-E. Update kits are available for those who purchased the earlier version.

The 3301 system allows one to record control voltages in real time, overlay control voltages in real time, hand load control voltage sequences, manipulate control voltages, and build up full programs of note assignments, durations, and playback instructions for automatic replay. The 100 page 3301 manual, which is available separately, covers with great detail the programming instructions and operations which are available on the 3301. Also provided is external interfacing requirements, schematics, and introductory programming techniques. For more information on the Dataton product line and 3301 manual, contact the manufacturer at the above address.



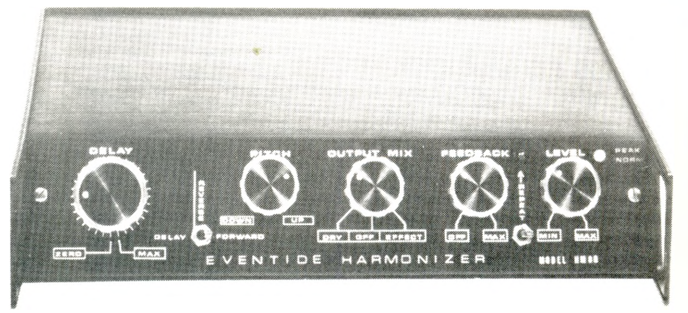
SPEAKER MOUNTS

Bob Griffin, president of Road Electronics, has announced the availability of speaker protection kits for the musician.

These kits consist of a heavy die cast aluminum speaker retaining ring, heavy guage steel mesh grid, speaker mounting gaskets and all necessary hardware. They can be used on existing sound reinforcement or musicial instrument speaker cabinets, and should prove ideal for the do-it-yourself cabinet builder.

According to Mr. Griffin, these kits will fit most speaker brands and are the same as used on the reknowned Road cabinetry, offering the ultimate in speaker protection, easy access and striking appearance as shown in the photo above.

For more information, contact: Road Electronics, 2107 E. 7th St., Los Angeles, CA 90021, Ph. (213) 623-2890.



BABY HARMONIZER

Since the introduction of the first Eventide Harmonizer in 1975, it has been the favorite special effects device of top bands and studios throughout the world. Now, Eventide introduces the HM80 Harmonizer - a compact unit with a full range of features. PITCH CHANGING from one octave up to one octave down, DELAY from 0 to 270 ms, FEEDBACK control, MIX of effect + dry signal, REPEAT and REVERSE (a new capability which gives a unique effect).

The HM80 has a frequency response of 10 kHz, a dynamic range of 80 dB, accepts line or guitar level input, weighs less than 3 lbs, and costs $775 ($800 for the 230 volt version).

While not intended to replace the H949 and H910 Harmonizers, which are designed for recording studio use, the HM80 is ideally suited to live performance. The REPEAT and PITCH CHANGE functions have been made remote controllable with this in mind, and a MIX control is included so that a console is not required.

The recommended list price of $775 puts the HM80 Harmonizer within the reach of most working musicians and small studios. Its compact size and rugged construction make it easily transportable - on stage, it can be placed on the floor or on top of a keyboard instrument, where its sloping front panel make the controls easily visible. It also has die-cut 'handles' to permit wearing it with a shoulder strap.

For more information, contact: Eventide Clockworks, 265 West 54th Street, New York, NY 10019, Ph. (212) 581-9290.

NEW STUDIO

Bruno Spoerri Recordings (Zurich, Switzerland) have just completed updating the studio with a new MCI 536 computerized console, an MCI JH-110A 2 track with AutoLocator, and the AMS digital delay and pitch shifter and the Lexicon 224 digital reverb. The studio has the widest variety of synthesizers in Switzerland, including the EMS 100, Prophet-5, different Lyricons, Arp and Moog synthesizers, etc.

Now in planning is a new large studio with board and lodging facilities. It will be built into a farm house which is situated in the Swiss countryside near Zurich. Bruno Spoerri and Irmin Schmidt (CAN) have recently been working on a new album of electronic music, and almost completed is the new Infra Steffs' RED DEVIL BAND LP. both productions are being engineered by Ron Kurz.

For more information, contact: Bruno Spoerri, Schneckenmannstr. 27, CH-8044 Zurich, Switzerland.

ACOUSTIC COMPOSER

Audionics Of Oregon has informed us of their Space & Image Composer, which is a very high performance SQ quadraphonic decoder... but it has a number of benefits and features beyond just the decoding of SQ encoded records. There is a Stereo Enhance position that allows the user to create a 360 degree sound field from any true stereo program source, whether

continued on page 29....

Part 2: SOFTWARE

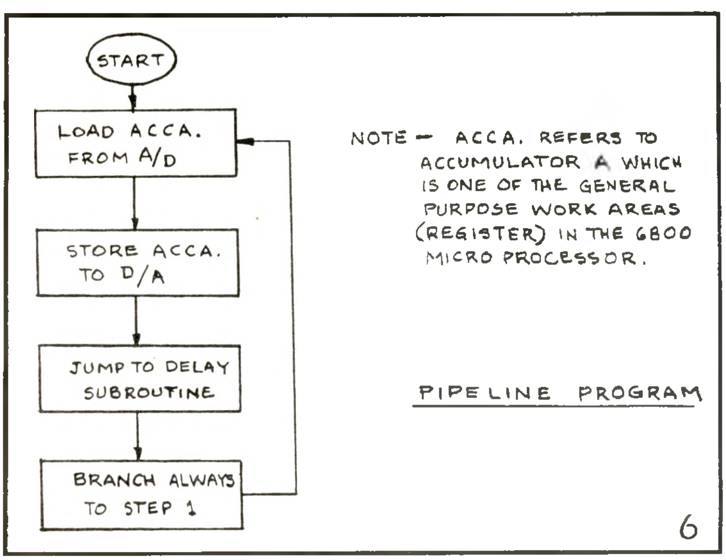
MICROCOMPUTERS IN REAL TIME AUDIO

by Tony Lewis

(In part one of this article, the author discussed the hardware and software manipulation techniques involved in handling audio information by a computer.)

Disregarding errors (such as aliasing and quantization error previously mentioned), the only changes occurring in the wave form going through the system should be due to the computer's operation on the data stream. From this it becomes obvious that almost anything could be done to the data stream by the computer, limited only by processor time. I have already discussed the grave limitations that processor time imposses on program complexity. For this reason the programs I have so far developed are not as lengthy as one would tend to suspect. All programs are written in machine language as opposed to a high level language like Basic to achieve faster execution. In discussing how the programs operate I will confine the discussion to the flow chart level avoiding source listings of the actual program, for I could not expect all readers to be familiar with 6800 machine code.

The first program that I wrote was the "pipeline" program mentioned in some examples in the previous issue. This program was useful for testing the system. Also, by slowing down it's execution by use of a delay subroutine, I could hear first hand the bandwidth limiting that I could expect more complex (therefore longer) programs to cause. The flow chart of the "pipeline" program is shown in figure 6.



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The "pipeline" flow chart is self-explanatory with the exception of the delay subroutine. A subroutine is a service routine called by another program. Upon completion of the service routine, program control is transferred back to the next sequential location in the main program after the subroutine call. This particular subroutine is part of the computer's monitor firm- ware. The subroutine loads a processor register with a delay constant and then repetitively subtracts one (decrements) from the register until it equals zero. At this point it returns control to the calling program.

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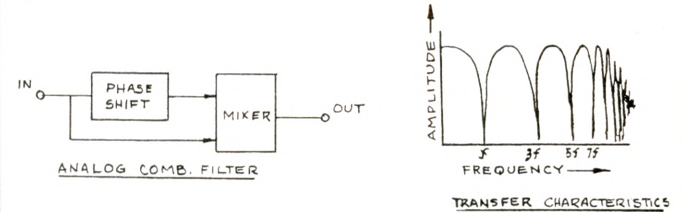
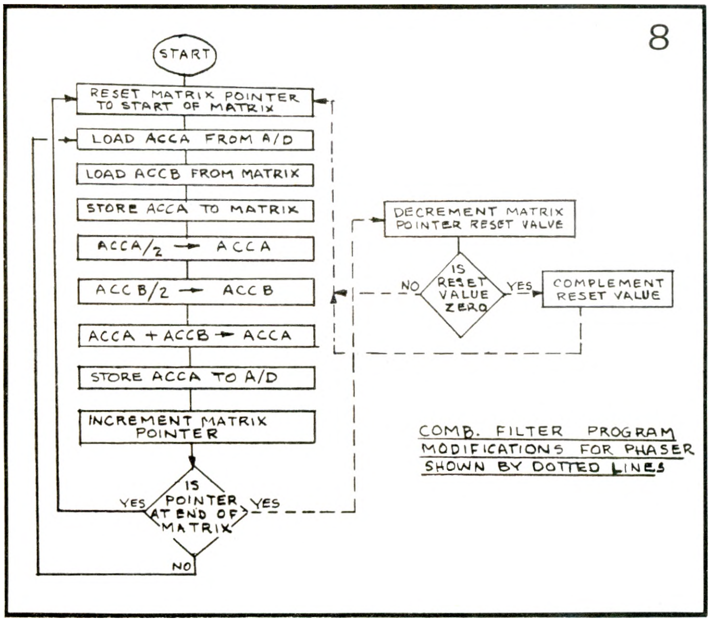


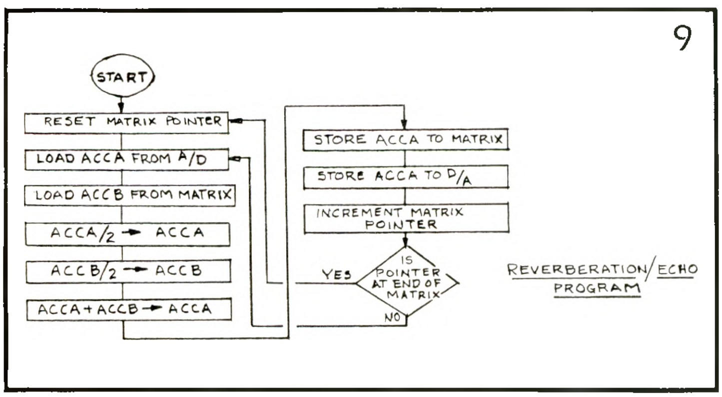
a b

The first program that actually modified the data stream was to have the computer simulate the analog device called a comb filter. Figure 7a shows the block diagram of an analog comb filter and figure 7b shows the transfer characteristics of a comb filter. When the phase delayed and original signals are mixed by the comb filter some frequencies fall out of phase and cancel while others fall in phase and add (see Kilobaud magazine, "Digital Audio" p. 75, issue #7, July, 1977). The comb filter program (flow chart in Figure 8) operates much like its matrix in memory. After setting a pointer to the beginning of the data matrix the computer receives data from the A/D and from the matrix. Then the computer replaces the data in the matrix with new data from the A/D and from the matrix. Then the computer replaces the data in the matrix with new data from the A/D and adds the old data from the matrix with the new data. Before the old and new data is added together both operands are divided by two to prevent overflow upon addition. Once data is stored in the matrix it is not retrieved until the computer has cycled through the matrix and back to where the data was stored. This process forms a digital delay line which acts as a phase delayer while the addition steps act as the mixer.

By using matrix lengths from 1 to 256 I could achieve delays from 46 microseconds to 117.8 milliseconds. An interesting effect (called phasing) is achieved when a comb filter is swept across the audio spectrum. The program just described can easily be modified for this effect. By changing the length of the matrix each time the computer cycles through it, phasing is realized. This is illustrated in figure 8 by the dotted line choice for yes at "is pointer at end of matrix".

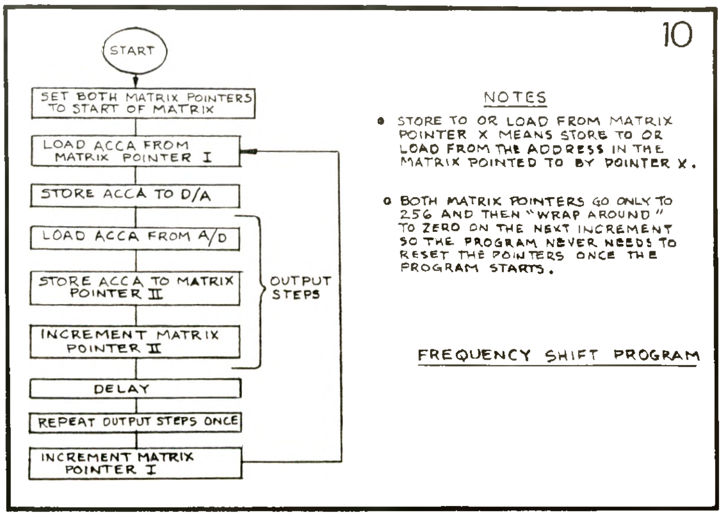


Another effect that can be realized by the matrix method is reverberation, or if the matrix is long enough, an echo. The basic change from the comb filter program is that the matrix is loaded with the result of the addition instead of the direct data from the A/D. When this is coupled with a longer matrix a good reverb or echo can be achieved. See figure 9 for the flow chart. When a signal first enters the computer it is divided by two and added to an old signal from the matrix - the result is then output to the D/A and stored in the matrix. Therefore, the signal is still present in the matrix (at ½ original amplitude due to the division) for the next time the computer comes to that point in the matrix. The next time around, the computer retrieves the matrix data, again adds it with new data and sends the result to the D/A and matrix. This time our original signal is still present in the matrix but at ¼ original amplitude. This process continues with the amplitude of the signal decaying by a factor of two each time through the matrix.



For reverberation I used matrices from 256 bytes to 1024 bytes in length and from 1024 to 8192 bytes for echo. As the length of the matrix increases, the time between echoes increases. Echoes can usually be heard five to nine times before they drop below the level of audibility (listening with headphones). Both the reverberation and echo are very "dry" sounding compared to their acoustical counterparts for each time a sound is repeated it has exactly the same harmonic content (unlike the echo in a large hall where the objects in the hall act as passive filters) and the sounds are repeated at a definite interval of time. In "real life" there are usually several different echo times due to reflection from objects at different distances. This could be simulated in the program by randomly changing the matrix length; however, I have yet to try this.

The only other type of waveform modification I have experimented with to date is pitch shifting of a waveform. It is extremely difficult to alter the frequency of a waveform by analog means in real time without changing the shape of the waveform. However, with a digital audio processor one can raise or lower a waveform in octave increments. One method of frequency shifting using analog means is to record the material on tape and then play it back at a different speed; however, this is not usable in real time. The method by which the computer achieves frequency shift in real time is closely related to the tape recorder. Here again the computer sets up a matrix (this one is of fixed length) into which it reads data at one rate of speed while outputting data from the matrix at a different speed. See the flow chart in figure 10.



This program maintains two matrix pointers which cycle through the matrix. Notice from the flow chart that matrix pointer II (output data pointer) moves through two matrix locations in one program loop while matrix pointer I (input data pointer) only moves through one location per loop. The net result is a doubling of frequency of the input waveform. The purpose of the delay routine is to insure that the same amount of time elapses between outputs. If this delay were not present, the output instructions would be right next to each other in the program loop. Without the delay the computer would produce two outputs (one right after the other) and then take time for the rest of the loop which would increase distortion of the output waveform. If the delay loop takes as much time as the rest of the program loop (excluding the output instructions) the outputs will be evenly spaced.

While the delay routine eliminates one cause of waveform distortion, another cause is inherent in the way in which the program functions. Whenever the output pointer crosses the location of the input pointer a glitch (rapid jump in voltage that is not part of the normal waveform) occurs. This is because before the output pointer crosses the input pointer's location, the computer is outputting "old" data but as soon as it crosses the input pointer's location it is in "new" data. If the new data happens to be at the same voltage level as the old data no glitch would occur, but this situation can only be achieved in one way. If the frequency of input is such that one cycle of the waveform fits the matrix length exactly, it will seem like a standing wave and the glitch problem will disappear. It is for this reason that I selected a matrix length that would fit one cycle of the estimated average frequency of the input waveform.

Since this program outputs data at twice the rate it inputs data the output frequency is doubled which corresponds to a pitch shift of one octave up. By modifying the program to output data at one half the input rate a downward pitch shift of one octave can be achieved.

Since this program, when shifting a pitch upward, must in essence manufacture half of the output data and since there is not enough time within the program to cure the glitch problem it is by far the most "noisy" (distortion introducing) program of those I have so far written. A similar problem exists when shifting pitch downward except that the program must ignore one half of the data. This concludes my discussion of the functioning of the real time audio processor system. By now you should have at least a basic idea of how the system hardware functions and how the software achieves the various functions of which the present system is capable.

I will now outline my results to date, and plans for the future of the real time system. The initial test of the hardware was performed without the computer in the system by

continued on page 18....

HOME RECORDING

APPLYING TRANSPOSING DEVICES

BY: CRAIG ANDERTON

In the past few years, an entirely new family of effects devices has made its appearance; these are called "harmonizing" or "pitch transposing" devices. Examples include the Eventide Harmonizer, MXR Pitch Transposer, A/DA Harmony Synthesizer (now discontinued), and many others. Most of these rely on digital technology, and as the cost of digital circuitry has decreased, so has the cost of these tools.

Unfortunately, the price still remains rather high. The A/DA unit, based on analog techniques, sold for $500; however, the sound quality was not exactly what you'd call hi-fi. The basic MXR unit costs $800, but if you want to add a preset selector footswitch and display, the cost goes up to $1150. Above $1000, you have several options, including a revamped version of the original Eventide Harmonizer. At these prices, few musicians - especially us homebrew/experimenter types - have had the opportunity to play with these devices and become familiar with their applications. However, since prices are coming down, you're likely to run into more studios, PA companies, and even bands that are using pitch transposing devices...so I think it would be a good idea to discuss the possibilities that these units offer for the home recording enthusiast. No doubt in the years to come, the price of pitch transposers will fall --just as the prices of other digital products have fallen in the past.

WHAT ARE PITCH TRANSPOSERS? Basically, you feed an input signal into the unit, and you get out a signal that is typically anywhere from an octave above, to one or two octaves below, the original signal. You can set the interval of the output signal with a continuously variable (or sometimes stepped) control knob. This allows you to get harmonies that are a fifth above, an octave above, a third above, and octave below, and so on. Unlike your everyday octave dividers and multipliers, pitch transposing devices can work with a polytonic signal. Guitar or piano chords, for example, work very effectively with harmony synthesizing devices.

Additional effects are obtainable by recirculating the pitch-shifted signal back to the input, creating a regenerating effect that can add considerable versatility to pitch-transposed sounds. We'll talk some more about what you can do with regeneration later on in the column.

HOW THEY WORK. If you have a device such as a voltage controlled flanger, you can experiment with very crude pitch transposing by sweeping the clock frequency of the flanger with a positive or negative going sawtooth wave. For best results, this should be at a sub-audio frequency -- but not too slow, or the effect disappears. What you are doing is basically storing signals at one rate, and reading them out at another rate by varying the clock speed. If you increase the clock frequency as you input a signal, the output will be <u>higher</u> in pitch than the original signal. Decreasing the clock frequency produces an output that is <u>lower</u> in pitch, since you are "<u>slowing down</u>" the signal. In digital units, which work much better and sound more natural, the input signal is usually read into some kind of memory, but read out at a different rate to create the pitch transposition effects. Unfortunately, during this write/read process, you have to either cut out cycles or add additional cycles in order to create a continuous audio tone. This process is called "splicing" and the object of splicing is to smooth out the output signal so that it sounds as natural as possible. As you can imagine, the more you pay for a given harmonizing device, the better the sound quality. At least with digital devices, the splicing glitches can be minimized; it is much harder to do so with analog units.

APPLICATIONS OF HARMONY SYNTHESIZING DEVICES. There are 3 main categories of applications; let's go over each one in turn.

1. <u>Creation of harmony lines.</u> While this is the most common use of pitch transposers, there is some question as to how musically valid a fixed harmony can be. Octaves work out all right and fifths sound OK most of the time...but try playing a harmonized line with major 3rds only, and you'll run into problems. An analogous situation occurs with multiple synthesizer VCOs; most synthesists I know tune them to octaves of fifths, or to near-unison for "flanging" and doubling effects.

As a result, some units include footswitching or preset options to allow for the selection of different intervals "on the fly". The A/DA unit, for example, includes a momentary footswitch which, when depressed, adds a different harmony line. In a situation like this, one preset could be for major thirds, and another for minor thirds. However, in studio applications, this does not seem to be too useful; after all, multi-tracking is all about the ability to add things like harmony lines. Another problem is with vocals. While the idea of using harmonizing devices for voice is appealing, most of these units change the timbre along with the pitch. Therefore, at an interval of something like an octave above, your voice sounds like it's on helium; with lower intervals, you get that "recorded at 15 ips but played back at 7½ ips" effect. So, again I think you're better off just double-tracking instead of using a pitch transposed line. Nonetheless, for "live" use and for applications involving chords and other complex sounds, the notion of a fixed harmony has a certain amount of validity. You can also achieve some very nice sounds with fuzz guitar by either taking the combined output of the harmony synthesizer and original signal and fuzzing them, or placing the fuzztone before the device.

2. <u>Creation of ambience.</u> This is perhaps one of the best uses for harmonizing devices that I have found. By setting the pitch interval very slightly lower that the original signal, or very slightly higher, you achieve a doubled, thickening sound. Adding regeneration adds a doppler kind of effect that is most interesting. The reason why this happens is because when you take a pitch that has been transposed by, say, one-half tone upwards, and then recirculate that tone back to the input, after passing through the pitch transposing circuitry the second time it will come out one full tone higher. Recirculate that again, and the next time it appears at the output it will be one and one-half tones higher. So, you can hit just one note on your instrument, and by properly adjusting the pitch transposer create an upwards or downwards cascade of notes. Figure 1 graphically shows this glissando type of effect.

Figure 2 shows what happens when you delay the recirculation signal. the time between interval changes increases, thereby creating more of an arpeggiation sound. With a 1 second delay, for example, a transposed note will occur every second. The sound is related to echo, in the sense that something happens at a regular rate; however, instead of having constantly decreasing amplitude (as is the case with echo units), you have constantly increasing or decreasing pitch. Subjectively, the ear is fooled into interpreting these pitch changes as some kind of ambience...at least that's the way it sounded to me. I'll leave any further analysis to the psychoacoustics experts in the crowd.

When using harmonizing devices as ambience enhancers, it's a good idea to keep the transposed signal at a much lower level than the original signal, as is usually done in the case of reverberation. This makes for a more subtle sound that also appears more acceptable to the ear's sense of reality.

Octave above sounds, especially with a little regeneration, can add string synthesizer-like effects to a variety of instruments. Also, true polyphonic octave division is a viable technique with pitch transposers. When mixed in at a low level with the original sound, the octave above adds a feeling of brightness, while the octave below adds a kind of rumble that can add more power to things like rhythm guitar and bass (when being played up high on the neck).

3. Creation of special effects. This is where you can really go to town, especially when you have a regeneration option. Varying the pitch while feeding constant tones into the device, varying the amount of regeneration, processing acoustical sounds to an extreme -- all of these techniques can produce sounds that range from the beautiful to the bizarre. Being able to pedal control the harmonized interval, thereby leaving your hands free, allows the instrumentalist additional latitude. I've gotten some great sounds on guitar by holding an E-Bow over one of the strings, and slowly varying the interval pitch while adding lots of regeneration. This produces "bell-tree" like effects that sound like a time warp being turned back around on itself...I realize that's a sketchy description at best but to do any better you'd just have to hear it.

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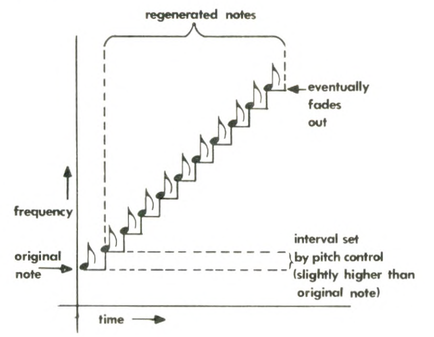

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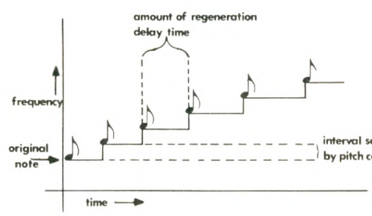
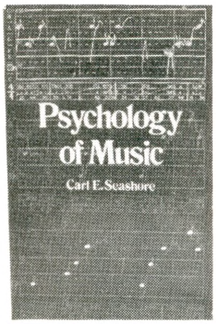
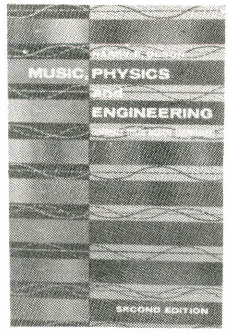
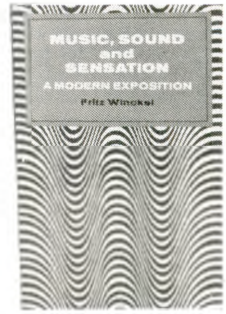


ADDITIONAL COMMENTS. There are several questions people ask me concerning pitch transposers. One of the favorites is "Will it make my 6 string sound like a 12 string?" Well, personally I don't think so. Subjectively, a flanger/doubler type of unit seems to do a more convincing job. Getting a transposed sound an octave above the original is difficult to do, and the quality of sound suffers as a result. Another question relates to vocals, but as I've said before, the timbre change really creates a problem. At this point, the questioner usually goes away disappointed...not realizing that harmonizing devices are perhaps better suited to applications other than creating musical harmonies. So, next time you get to play with a pitch transposing device, free yourself from any preconceived notions and check out the ambience and special effects potential of the unit. You may find that you've got a whole new tool to add to your arsenal of sounds.

POLYMART







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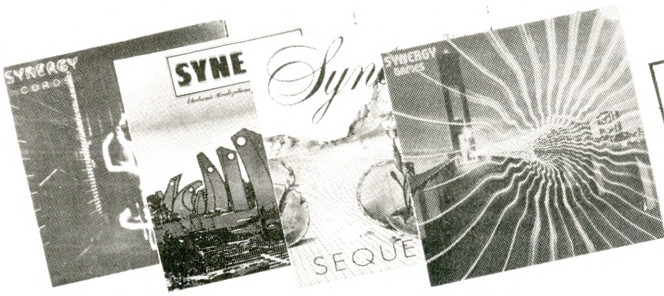

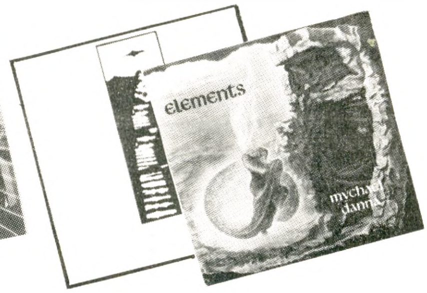

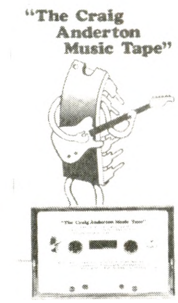

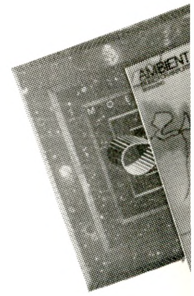

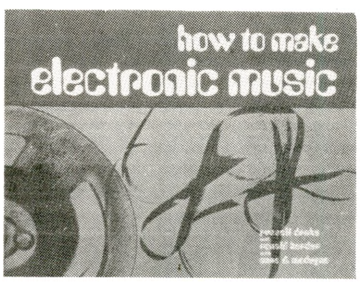

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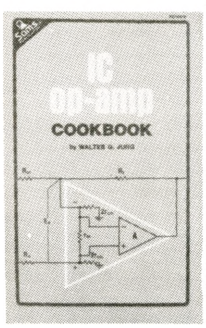

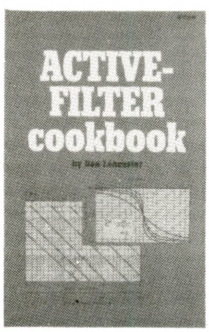

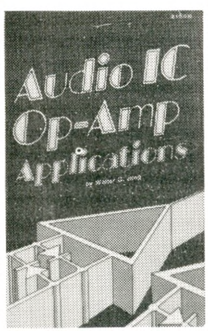

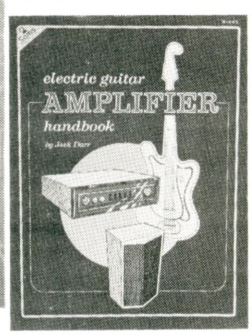

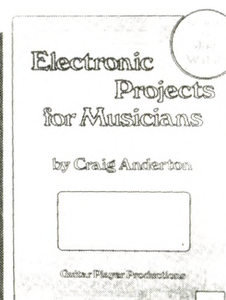

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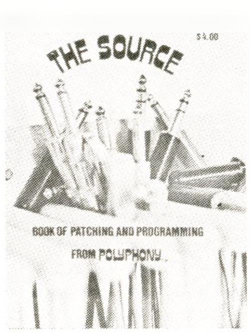

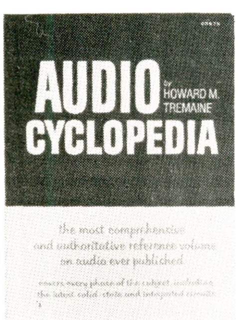

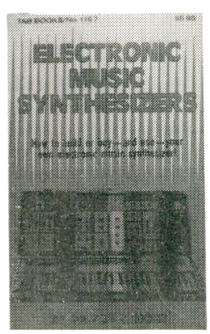

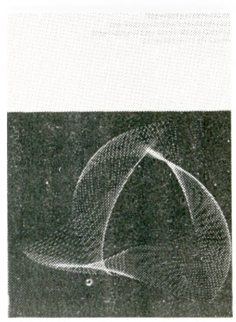

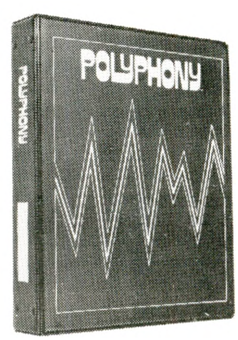

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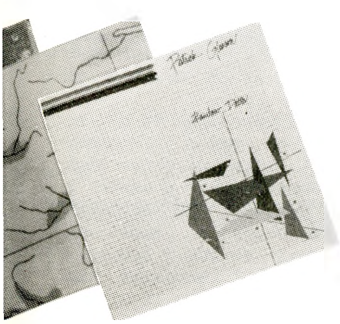

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MICROCOMPUTERS IN REALTIME AUDIO

continued from page 13....

connecting the A/D directly to the D/A. In this setup the system hardware functioned flawlessly on the first try with a bandwidth in excess of 18 KHz. Once the computer was placed in the system running the "pipeline" program, signals still passed through the system but with narrower bandwidth and the addition of some noise. Using the delay in the "pipeline" program to simulate the running of a longer program, the bandwidth was dropped to approximately 8 KHz. At this point, I adjusted the conversion speed of the A/D to more closely match the computers speed by dividing the A/D's clock input by two. The aforementioned noise was, after considerable experimentation and frustration, found to be a mixture of noises from the following sources:

1. Short term (approx. ½ second) squeals from 80 Hz. to 3 kHz - This was found to be caused by the radio frequency signals inside the computer. Two or more of these high frequency signals would zero beat (mix together to produce a frequency equal to the difference between the original frequencies) to produce the squeals. The squeal's amplitude was measured at a maximum of ± 0.2 volts on the ± 2.0 volt output signal. I felt 10% was intolerable so I increased the shielding on all cables and reduced the squeals to 2% of the output signal.

2. Continuous white noise hiss - This was traced to an impedance mismatch between the preamp and input filter. The use of op amps in the input filter made it easy to redesign the filter for a higher input impedance. Before redesign the hiss was at a ± 0.1 volt level; redesigning the circuit lowered it to ± 10 millivolts.

3. Short term (sub one hundred milliseconds) glitches (not related to glitches caused by pitch shift programs discussed earlier) - These glitches occurred at random intervals often being absent totally for several minutes and were at a voltage level of ± .5 at their maximum amplitude. This was traced to glitches on the house power lines (caused by appliance cycling) that got through the power supply to the op amps. Additional bypass capacitors reduced the amplitude of the glitch on the output to ± 0.1 which is still rather high but considering the infrequent occurrence of the glitches the one tenth volt level was not found objectionable.

4. The presence of certain frequencies in the output wave form that were not in the input waveform - When applying a relatively complex wave form to the input I would sometimes find a waveform at the output that should not have been the result of the computer's modification of the input wave form. After two weeks of total frustration I discovered that the problem was due to aliasing. When a frequency above the bandwidth of the system is applied to the input, a frequency within the bandwidth of the system would result at the output. Some of the complex wave forms I was using contained harmonics above the systems bandwidth that could still get past the input filter. A simple readjustment of the input filter's cutoff frequency totally cured this problem.

After bringing the aforementioned problems under control, I found that I could modify an audio signal in real time to produce a comb filter effect, a phaser, an echo, a reverberation, and an octave interval pitch shift using a digital microcomputer and the system discussed in this paper. While the bandwidth was still rather limited I found the setup a very useful addition to my audio laboratory. By virtue of this fact I feel that I have proved that an eight bit microcomputer can be valid for real time audio work with usable results.

In the future I have plans to develop a microcomputer which is ten times faster than my present one. The new computer is still going to be an 8 bit version for that particular word size has been found to be both practical and economical. The new computer will be strictly for real time audio and will most likely be based on the RCA CDP1802 microprocessor. The new computer should cure any bandwidth problems and free the main lab computer for its other duties. A small control computer (similar to a KIM or 8700) is also in use running several other digital synthesizer modules.

CONSTRUCTION!

DIGITAL AUDIO DELAY UNIT

CONSTRUCTION!

By-John Neves & Steve Kolupaev

Many musicians will tell you that the typical American living room is one of the most "dead" places possible, acoustically speaking. That's why musicians prefer concert halls for performances. If a musician had to perform in your living room, he would probably bring along an electronic reverberation unit to add a slight echo to your living room. The effect of a barely perceptible echo added to a sound system is truly amazing, and must be heard to be believed.

This article describes a simple but effective audio delay unit which you can use to simulate the acoustical setting of a concert hall in your living room. It uses 22 ICs, most of which are available on the surplus market by mail order. The approximate cost of the ICs, as purchased from the suppliers we used, is about $68.

In designing this project, we scaled down or eliminated many of the features found in professional units, leaving just enough performance to give a credible feeling of depth for the average living room. The simpler design presented here lacks the varied controls and switches of a professional unit; it has slightly less dynamic range, and has only one channel instead of two. But its cost is less than a tenth that of the professional units. We hope that with these inducements you will build one, and share the experience of throwing a switch which seems to move the walls of your living room back 100 feet in every direction.

Our delay unit is intended to be used with one or two auxiliary power amplifiers and loudspeakers. Ideally, there should be two delay units, one per channel. The sound should be delayed about 20 milliseconds in each delay unit, and delivered to your listening room at a level about 20 dB below that of the undelayed sound. The layout of this setup is the 'standard' four channel setup where the front channel preamp outputs feed the rear channel delay units, power amps, and speakers in addition to the front channel amps and speakers. The delayed echoes in a real auditorium suffer frequency-dependent absorption and phase delay, so in your living room the delayed sound doesn't need to be as accurate as that from the main speakers. Less expensive speakers and amps for the delayed sound channels can be used with no problems.

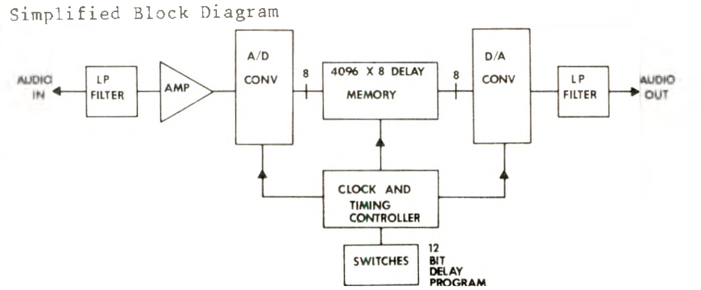
If you cannot add 2 low power amplifiers and speakers for the delayed sound channels, you could use a single amp, speaker, and delay unit. Audio from the two main channels should then be mixed together and fed into the delay unit as a mono signal. If an extra speaker and amplifier are not available, one can still achieve a depth effect by mixing the delayed signal back with the straight signal prior to being fed to the front channel power amps. Most amplifiers have a jumper plug from the preamp to the power amp which makes this option very easy to implement.

The amount of delay produced by the digital delay is adjusted in small steps, so you can tailor it to the needs of your particular listening room. About 20 milliseconds is a good starting point. Provisions have also been made for feeding back some portion of the delayed signal to the input amp for recirculation. This can generate a depth effect as well as echo, but if the signal level in the feedback loop is too high, oscillation will occur.

How Audio Travels Through The Delay

Refer to the simplified block diagram in figure 1 and the schematic in figure 2. Audio first enters the delay unit through the lowpass active filter U1a. This filter attenuates frequencies at and above 10 KHz, which is half the sampling frequency of the delay unit when the clock has been set to 1

1 Simplified Block Diagram



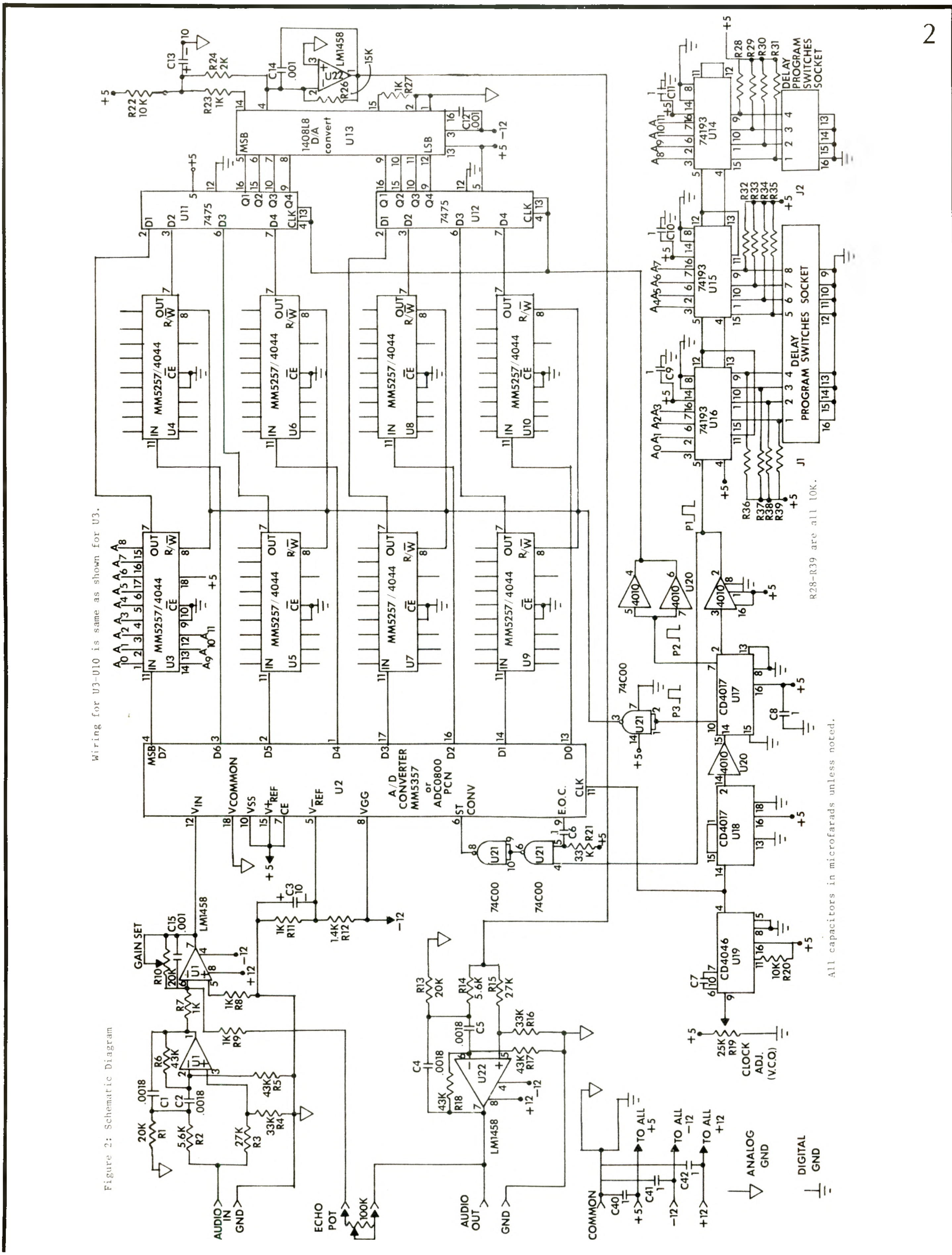


Figure 2: Schematic Diagram

Wiring for U3–U10 is same as shown for U3.

R28–R39 are all 10K.

All capacitors in microfarads unless noted.

MHz. The audio is then amplified by U1b such that the maximum input level to U2 is 10 volts peak-to-peak.

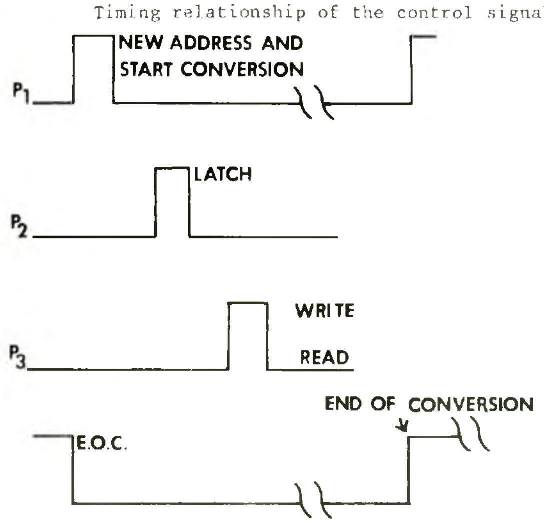
Next the audio is presented to the analog-to-digital (A/D) converter U2. U2 takes a reading of the input voltage and converts it to an 8 bit binary code to be stored in the memory. U21 provides the logic to prevent spurious conversions by U2, and the RC network is needed to 'power on reset' the A/D converter. The 8 bit code from the A/D conversion is then stored at one of the addresses in the 4096-by-8 bit memory, U3 - U10. The code is read out of the memory after a programmable storage period, which is variable up to about 200 milliseconds. The 8 bit code read out of the memory is loaded into an 8 bit latch consisting of U11 and U12, which holds the data steady while it is being converted back to an analog audio voltage by the digital-to-analog converter (D/A), U13.

This voltage is almost identical to the original voltage presented to the A/D converter (U2) sometime earlier, multiplied by a scaling factor. It may differ from the scaled original by a few millivolts, due to what is called "quantization noise". The reconstructed audio from the D/A converter (U13) is amplified by op amp U22a, bringing its level to about one volt peak-to-peak. It then passes through an active lowpass filter, U22b, which also begins attenuating at 10 KHz. The purpose of this filter is to attenuate the sampling frequency components in the recovered signal.

A 4046 is used as a clock generator in position U19. Although the 4046 is a complete CMOS phase lock loop, we use only the voltage controlled oscillator section. This VCO is the master time base for the whole digital audio delay unit. Because it has a voltage controlled time base, this digital audio delay unit can be used in conjunction with a larger voltage controlled studio or synthesizer to experiment with vibrato, flanging or frequency modulation if desired. However, in this design we simply use a front panel control to provide a variable voltage source to adjust the clock frequency. For use in music systems, the rate control could be replaced by a summing amp designed to accept external control sources.

The clock signal is fed to the A/D converter U2 and to the divider chain U17 and U18. Decade outputs of these counters provide the proper timing sequence for selecting new addresses, latching old data into U11 and U12 and then allowing new data to be written into memory. A buffer, U20, is used to convert the low current CMOS outputs into signals capable of driving the TTL chips used for latches. The manufacturer of the A/D converter specifies that the clock rate has to be at least 44 times larger than the sampling rate. Two CMOS 4017 ring counters divide the clock frequency by 50 and provide 3 time staggered pulses P1, P2 and P3 at the same rate, which is our sampling frequency. See figure 3. P1

3 Timing relationship of the control signals.



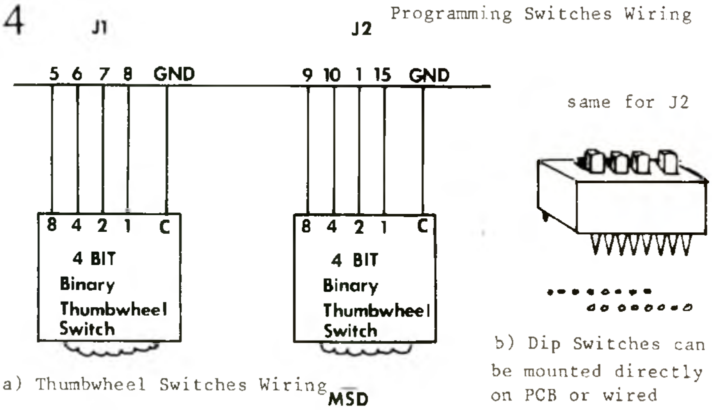
commands the start of an A/D conversion in U2 and bumps the memory address counter up by 1. The resulting memory address is the address where the oldest existing sample is stored. U2, which was started on a new conversion by P1, is now outputting the result of the previous conversion via an on-chip buffer. It will hold this output until the new conversion completes. The new conversion will complete 10 clock pulses before the next P1 pulse. The memory is normally in the 'read' mode. This means the memory constantly displays at its output the 8-bit code stored at the address pointed by the memory address counter. P2 is applied to the memory output latch consisting of U11 and U12, causing it to load the result of the READ operation and hold it for D/A conversion by U13. U13 does a conversion in less that a microsecond and will maintain its output until the newest conversion is completed.

The last staggered pulse, P3, causes a 'write' operation in the memory, which overwrites the old sample with the 8-bit code currently being displayed by the output buffers of U2.

The memory address counter consists of U14 through U16. It normally cycles through the 4096 or less different memory addresses. In any given step of the cycle, an old audio sample is read out by the P2 pulse, and a new sample overwrites the old one by P3. The delay produced by the digital audio delay unit can be figured by multiplying the number of different memory addresses being used by the sampling time interval. If the sampling frequency is adjusted for 25 KHz, the sample interval equals 1/(25 KHz) or 40 microseconds. If all the addresses are being scanned, the total delay will be 4096 X 40 microseconds or 163.84 milliseconds. If the clock is adjusted for a sampling frequency of 20 KHz, the total delay will be 4096 X 50 microseconds, or 204.8 milliseconds.

A set of thumbwheel or dip switches connected to J1 and J3 as shown in figure 4 should be used to change the modulus

4 Programming Switches Wiring



a) Thumbwheel Switches Wiring

b) Dip Switches can be mounted directly on PCB or wired accordingly

of the memory address counter, allowing delay memory length to be set anywhere from 1 to 4096 bytes. This permits the audio delay to be varied while holding the sampling rate constant. Although the use of switches provide the most convenient way to adjust the delay to the needs of the room where it is being used, the delay can also be programmed by fixed jumpers, or interfaced with external TTL signals such as a microprocessor.

Construction

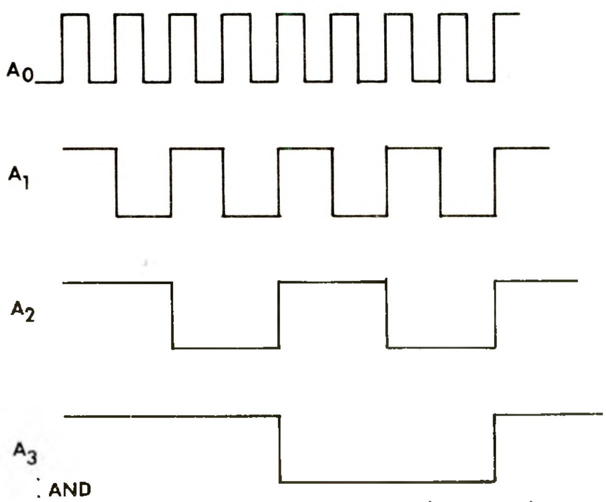
Given the multiplicity of connections, this circuit would best be implemented either by wire wrapping or using a printed circuit board. Care should be exercised in the ground connections. In order to avoid noise generated by ground loops, the analog and digital grounds should be totally separated and merged together only at the power supply input connection. The analog grounds are those connected to the operational amplifiers in the audio input and output circuitry. Note the different grounds on the schematic. The address counter outputs feeding the memory section should be kept as short and far away from the audio section as possible. Needless to say, this device is complicated enough that a single miswiring will prevent the unit from operating properly. The use of IC sockets is recommended for all packages, and especially for the expensive parts or when parts of questionable quality are used. Observe polarity of the capacitors. Wire the programming switches according to figure 5. If you are using thumbwheel switches, two digits will be more than enough for practical purposes. Dip switches fit directly into the circuit board. Thumbwheel switches will need cables with molded dip connectors. All soldering should be done very carefully in order to avoid solder bridges and cold solder joints. Orientation of IC's should be observed carefully. The digital audio delay unit needs +5 volts @ 450 mA, and +12 and -12 volts, both @ 20 mA each. The 5 volt power requirement could be reduced by using the 74L counterparts for U11, U12, and U14 through U16.

Adjustment

Very little tweaking is needed to get the unit to work. The clock pot adjusts the sampling speed, and thus controls both the frequency response and time delay of the unit. If the sampling frequency is set too low, it will limit the high frequency response. These high frequencies will cause distortion products known as aliasing products. Theoretically the frequency bandwidth is one half the sampling frequency. The audio gain pot adjustment will match different input audio levels. It should be adjusted for a 10 volt peak-to-peak audio input at the A/D converter when the maximum audio amplitude is fed to the audio input connection. Most home stereo systems will output 1 volt peak-to-peak of audio at the tape output jack, making it necessary to adjust the pot for a gain of 10 in the op-amp; i.e. the trim pot will be set near mid position.

The power supplied to the circuit should be well filtered and regulated. Most of the three terminal IC regulators readily available will do the job. An oscilloscope would be useful to check out the unit although this is not essential. In case of trouble, the waveforms should be checked as they would indicate any improper operation. If the clock generator has been adjusted for, say, 800 KHz, the output of the first clock divider (U18) will be 160 KHz. Tracking down the clock signal, the output of the second clock divider (U17) will be 16 KHz. The waveforms shown in figure 5 illustrate the most important waveforms in case there are problems.

5



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ETC

Memory address counter is a straight binary counter. A-11 will be 4096 times slower than A-0.

The memory address counter built around U14, U15 and U16 is checked by setting the delay program switches for maximum delay. In this condition each IC will count down the pulses by 16. The scope is then used to see that A0 (address line #0 - the least significant or "fastest" counting line) through A11 (address line #11) have outputs of decreasing frequency.

Likewise the audio signal can be followed through the input amplifier and filter and to the A/D converter. The digital output of U2 will vary in sympathy with the audio signal being fed to the digital audio delay unit. The important detail in this test is to verify that all bits are changing, which indicates proper operation. If a DC level is fed to the input through a voltage divider pot, the changes of the bits could be observed as the DC level is varied. The outputs of U2 are complementary. 0 volts input will produce all "ones" and 1 volt will produce all "zeros", with 254 binary numbers in between. If the unit is to be used with sampling rates lower than 20 KHz, the value of the filter capacitors in the input and output filters should be changed by an amount which is inversely proportional to the amount of sampling rate change in order to achieve the optimum signal to noise ratio. As an example, halving the sample rate would require doubling the filter capacitor values.

PARTS LIST

RESISTORS: (all are 1/4 watt, 5%)	
R1, R13	20 K ohms
R2, R14	5.6 K ohms
R3, R15	27 K ohms
R4, R16, R21	33 K ohms
R5-6, R17-18	43 K ohms
R7-9, R11, R23, R27	1 K ohms
R10	20 K ohms pc mount pot
R12	1.4 K ohms
R19	25 K ohms pc mount pot
R20, R22, R28-39	10 K ohms
R24	2 K ohms
R26	15 K ohms
CAPACITORS:	
C1, C2, C4, C5	.0018 ufd mylar
C3, C13	10 ufd electrolytic or tantalum
C6	.1 ufd ceramic
C7	10 pf ceramic
C8, C9, C10, C11, C40, C41, C42	1 ufd mylar, tantalum or 'lytic
C12, C14	.001 ufd mylar or ceramic
INTEGRATED CIRCUITS:	
U1, U22	LM1458 dual op-amp
U2	MM5357CN or ADC0800PN Analog to Digital Converter. National Semiconductor. Available from Elmar Electronics, Hamilton Avnet or any National Rep. Cost approx. $9.50. Also available from Digi-Key (See magazine ads).
U3 - U10	MM5257 4K X 1 Static Ram. Available from any National Semiconductor Rep. or EXATRON (3555 Ryder, Santa Clara, CA 95051). Set of eight $50. Replacements: TI 4044, EMM 4044, IM 7141, MCM 2141, 4104, 2613, HM 4315, HM 4847. Also available from Advanced Computer Products.
U11, U12	7475 or 74L75 four bit latch.
U13	Motorola 1408 Digital to Analog Converter. Available from James Electronics (See magazine ads).
U14-U16	74193 Up-down synchronous 4 bit counter.
U17-U18	4017 CMOS decade counter.
U19	4046 CMOS phase lock loop
U20	4010 CMOS / TTL buffer
U21	74C00 CMOS quad two input NAND gate.

MISCELLANEOUS:
- Printed Circuit Board - Available from Neves, P.O. Box 10327, Stanford, CA 94305.
- IC sockets
- 2 digit thumbwheel switch, or two 16 pin dip switches
- 100 K ohms potentiometer for echo control
- Phono connectors for input/output
- Case (preferably metal)
- Power supply (see text)

Writing Documentation: Bridging The Two Cultures

by Ken Perrin

Documenting a project should be the most natural thing in the world, but it's often treated as a boring, distasteful job. In the spectrum of things, it's typically placed somewhere between eating liver and visiting the dentist. It's viewed, at best, as a necessary job which "I'll get around to sometime in the summer." At worst, it's "something I wouldn't do even if I got paid for it." Since I do get paid for doing it, I can offer some suggestions to make it easier and to help make your documentation effort improve the quality of your finished project.

Documentation is history. Your project's documentation is a written description of how you envisioned the project at the time when you wrote down your ideas. Of course, until the final version of your project is completed, working, and you're about to do something else, your project is in a state of continual change. Consequently, you will have to periodically (or aperiodically) up-date your documentation. Take a look at the three kinds of documentation your project requires.

The first of these kinds is a FUNCTIONAL SPECIFICATION. The functional specification should be written after you've given the project a good deal of thought but before you start deciding how you will build the device. The functional specification is a detailed description of what the device will do. It should be organized in a sensible and methodical manner and each section of the document should include sufficient detail so that someone else could build the device from you functional specification.

The second kind of documentation that you'll need is a TECHNICAL SPECIFICATION. This document is the "nuts and bolts" document and in it is the information specifying which engineering choices you'll make in order to implement the functional specification. The functional specification outlines the problem (building the device) and the technical specification outlines the solution. The technical specification closely follows the functional specification and it may even be organized along the same lines. It may help to differentiate between these documents if you consider who, in a large company, writes them. When a company begins a project, the marketing department writes the functional specification and sends it to the engineering department who then writes the technical specification.

The third kind of documentation is the final re-write of the technical specification and it is called the TECHNICAL DESCRIPTION. It is a history of all the changes and decisions which you made during the time when you were actually building the project. It will include good drawings, at least a wiring diagram and a schematic drawing. These should be much more than pencil sketches. They should be done using standard drafting techniques and tools. Although it takes time, you'll find the effort to be well-spent. Of course, all documents should be typed and finished with as much professionalism as you can muster.

There has been much written regarding writing style; from the quality of writing that I often see, much of that available literature has obviously not been read. Writing well is not difficult, at least it's no more difficult than thinking logically. Each time you make a logical decision you arrive at that decision by weighing variables and choosing the most attractive alternative. Documenting your project, is simply writing a history of these decisions and, in some cases, the factors which led you to make your choice.

It's very common to find people are able to explain something quite well orally, but can't write worth a darn. While there's no simple solution to this problem, there are a couple of techniques you can use to help make sure you're not one of these folks. First, pick a person who you know personally to use as your "target audience." As you are writing keep asking yourself if this person could understand your explanations. This gives you a sense of how much detail to include and at what level of expertise you should aim your writing. The second trick is to write straight-forward sentences. Writing cumbersome sentences is somehow associated with writing about technology. While many engineers do write poorly, there is no sane reason for emulating them. Because your document is difficult to understand is no indication that your work is significant or that you are working on the cutting edge of technology. Third, THINK ACTIVE. Many technical documents overuse the passive voice. This makes the reading dull and tends to communicate less information. Finally, read something written by William Safire and try to write like he does. (Mr Safire writes the column "On Language" in the New York Times Magazine.) In a set of rules for writers he has suggested that you...

> don't string too many prepositional phrases together unless you are walking through the valley of the shadow of death.

But rather than discussing writing entirely in the abstract, let's take a sample project and work out a format for documenting it. For the sake of an example, assume that we're going to build a VCA. Furthermore, assume that we decide at the outset that it will use the SSM VCA chip. Of course, we couldn't put an entire sample documentation package into this article, but we can include a sketch of a spec. We begin by considering the organization of the functional specification.

The functional spec will be organized into four chapters. The first chapter will include general information and specs about the module and the necessary information about the rest of the system with which we will use our module. Chapter two will discuss the inputs, Chapter Three the multiplier, and Chapter Four the outputs. The functional spec for another project, however, might be organized differently. There is no standard form that is applicable in all cases.

The accompanying example (see box) is an example of one chapter of the spec. Of course your own spec should be less sketchy and include more detail than this sample.

After you've written a functional description of the project, there are probably still some decisions which you haven't been able to make. You should write these down in a separate section of the spec, perhaps in Appendix A. This will act as a reminder and will show you which areas deserve your special attention and energy. Also it will keep these issues from being forgotten instead of resolved.

When you've finished the functional specification, mull over the issues for a few days and when you resolve each one, add pages to the appropriate section of the spec showing your final decision. At this stage of the process, never erase anything, cross anything out or discard an obsolete page. There may be a time in the future when you'll want to know what you had, at one time, decided.

After you've given it sufficient thought and all the issues are decided, your spec will be a mess of corrections and changes and will probably be barely legible in places. Before doing absolutely anything else, re-copy the spec and make it neat, presentable, and readible. If you intend to try to sell your VCAs, this step is crucial. But even if you have no intention of marketing your module, having a good spec to work from gives the project a good solid beginning.

By now you probably have a pretty good idea of what information should be included in the other chapters of this spec. Keep in mind that the functional specification is a description of the module and is neither a plan to build it nor a description of the circuitry. If you were to have an engineering consultant build this module for you, this is the document you would give the consultant. Of course the other thing you'd give him/ her is the family forture, which is why you're doing this yourself.

The next step is to write the TECHNICAL SPECIFICATION. This is the document which outlines how you intend to build the module. As a guideline to writing the technical specification you should have before you your completed

FUNCTIONAL SPECIFICATION
VCA Module based on the SSM VCA chip

CHAPTER ONE

1.0 Introduction
This document describes a voltage controlled amplifier made expressly for use with the BLANK synthesizer system. The VCA will be built using the SSM VCA chip and its design will closely follow the applications notes suggested in their literature.

1.1 System Requirements

1.1.1 Power
The system operates on voltages at +/- 15 volts and this module shall not draw more than 100 milliamps of current at either voltage. The decoupling capacitors for the supply shall be 1 microfarad tantalum capacitors.

1.1.2 Signal Levels
Signal levels at which the module will operate are -10 to +10 volts at the audio inputs and 0 to +10 volts at the control inputs.

1.1.3 Signal to Noise Ratio
The signal to noise ratio will be better than 65 dB.

1.1.4 Impedances
The input impedances shall be 100K ohms and the output impedances shall be 1K ohm.

1.2 Inputs

1.2.1 Audio
There will be four audio inputs into this module. The first two inputs will have associated attenuators and the third and fourth inputs will be un-attenuated. The second audio input will be an inverting input. Signals simultaneously present at all four inputs will be summed within the module.

1.2.2 Control
There will be four control inputs into this module. The first two inputs will have associated attenuators and the third and fourth inputs will be un-attenuated. Signals simultaneously present at all four inputs will be summed within this module.

1.3 Multiplier
The multiplier is a 2 quadrant multiplier and shall be the SSM VCA chip. The gain factor shall be unity.

1.4 Outputs

1.4.1 Audio
There will be three indentical audio outputs from the module.

1.4.2 Control
There will be one control output from the module. The signal present at this output is the inverted sum of the control voltages input into the module. This is available to make panning patches, by using a second VCA, more easily patchable.

1.5 Features

1.5.1 Operating mode: linear/exponential control
The module will be capable of being controlled by a linear and/or exponential voltage. This choice is to be made by using a 2-position toggle switch mounted on the face panel. The size of the switch is to be consistant with the rest of the hardware to achieve a unified appearance. The limits of the exponentially converted control voltage will be 0 to +10 volts.

1.5.2 Input coupling
The module will be both capacatively and directly coupled at the audio inputs. There will be a 2-position toggle switch to select the input coupling. The control inputs will be directly coupled.

1.5.3 Initial gain
The module will offer manual control of the initial gain. This control will "open" and "close" the VCA manually from a front panel knob. This voltage will be summed with other control voltages and shall be subject to the operating mode. The inversion of this voltage will appear at the control voltage output along with the other control voltages.

1.6 Face panel
The module will have a front panel consisting of a 3" by 9" by 1/16" piece of anodized aluminum. It will be mounted via two screws at the top and a single screw at the bottom. It will be predominantly black and the graphics will be silver-white. Knobs will be the WW variety so as to be consistent with the rest of the system.

1.7 Jacks
Jacks will be Switchcraft XX jacks so as to be compatible with the Switchcraft YY patch cords and ZZ male connectors.

1.8 Reliability
The module will have a MTBF (mean time between failure) of 2000 hours and a MTTR (mean time to repair) of 2 hours.

1.9 Other considerations
The module will be built using the SSM VCA chip. All other parts must be available from stock from at least two reliable sources. All parts must be deliverable within 6 weeks from the time of order.

version of the functional specification. It's natural that the technical specification be organized along similar lines as the functional specification. Keep in mind that the functional specification is an outline of various engineering problems and the technical specification is an outline of their solutions.

The technical specification contains descriptions of the circuits and, where appropriate, schematic drawings. Again, the idea of the document is to present the information clearly enough that someone else could build from your descriptions. The amount of detail needed in this document is about "one level" more detail than you currently think is necessary. Of course you need not describe how individual components work. After all, in our example the SSM VCA chip is a "black box" anyway and there's no practical reason for knowing how it works. Your decision on how to handle the linear and exponential sensitivity however is different. This is something which you should document in detail.

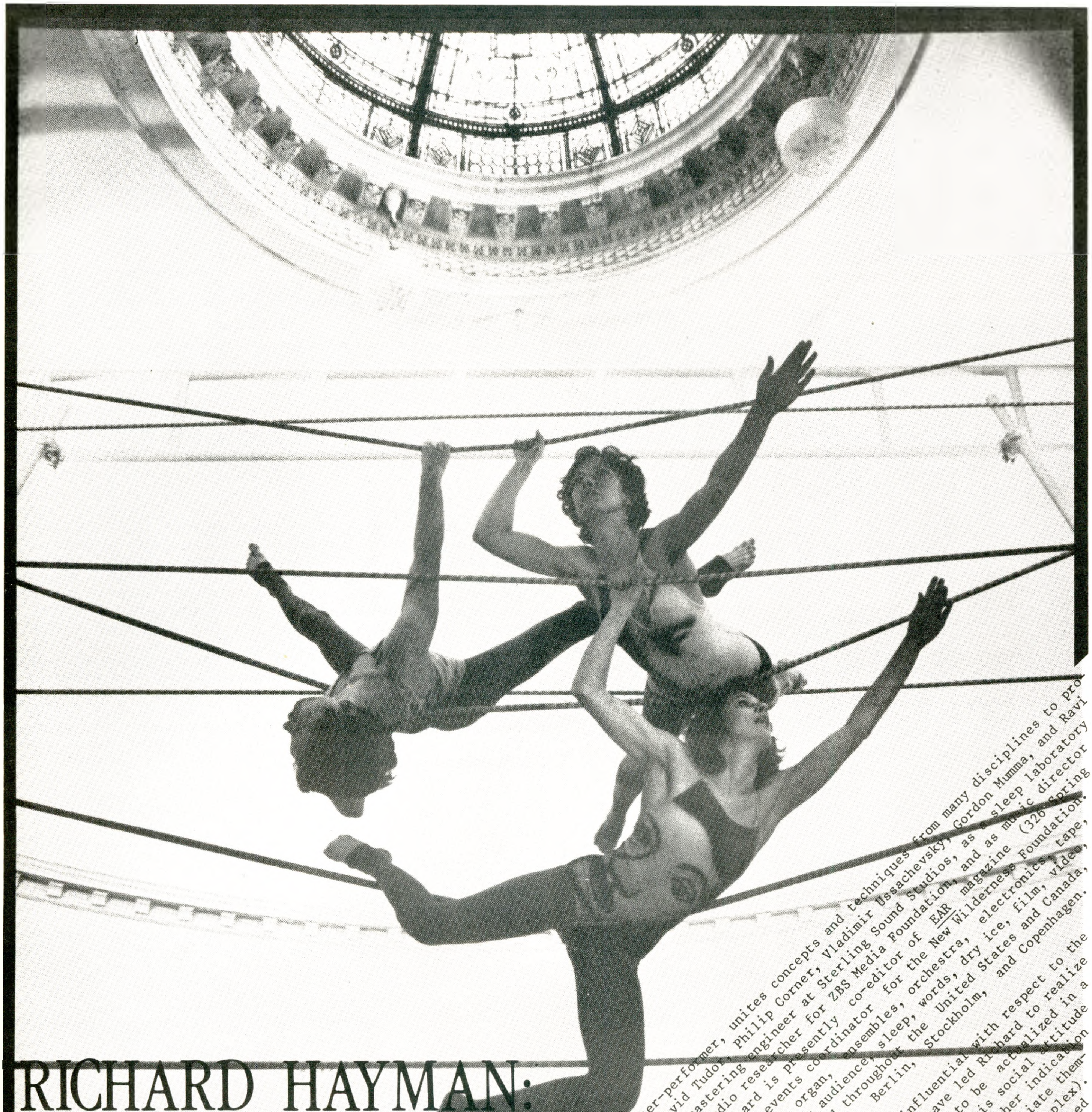
After the technical specification is complete then you actually start to build the device. When building it, have the technical spec on the bench with your work and as you make decisions about this or that capacitor value or whatever, write it into the technical spec. Do not make any technical decisions without documenting them. Things which seem obvious at the time and which you think you'll always remember, you're bound to forget sooner or later.

After you've built and de-bugged the module and are sure that it's working up to specs, you can begin the final document. This last piece of documentation is the final revision of the technical specification. Re-write the technical specification so that it includes all the decisions you made while building the project. Make a wiring diagram and a schematic drawing. We reiterate that these should not be pencil sketches made free-hand, but real professional quality drawings. Go out and get the necessary templates and drawing equipment to do a good and neat job. There will be a time when this module stops working, it's guaranteed, and making good documentation will be very helpful in finding the fault. After you've made the final revision to the technical specification, the name of the document changes. It is known at that point as the TECHNICAL DESCRIPTION and it is the last of the three necessary documents.

If you intend to market your module as a profit-making venture, you'll need a host of other documents. In addition to the obvious ones like press releases and product literature, you'll also need documents like financial proposals and business plans. These, however, are well beyond the scope of this article.

It seems like a human tendency to view one's own effort as unusually significant in the scheme of things. I can't help but see that a strong commitment to documentation is a fair indicator of one's confidence in the project and assurance of the project's success. Writing full-scale documentation for a project assumes that someone will need to have access to this information years from now. This is a demonstration of confidence in your work. Documenting a hobby-project is equally important as it shows a willingness to work thoroughly and completely. It makes the project better because you follow the proven methods and format that

continued on page 28....



RICHARD HAYMAN: Composer-Performer

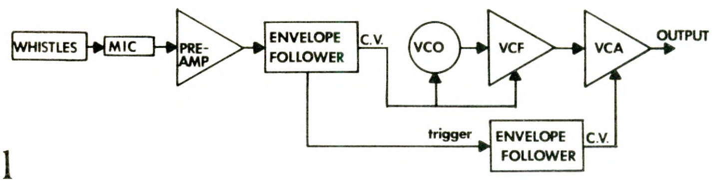
by David Ernst

Richard Hayman (b. 1951), composer-performer, unites concepts and techniques from many disciplines to produce 'new' music. He has studied with John Cage, David Tudor, Philip Corner, Vladimir Ussachevsky, Gordon Mumma, and Ravi Shankar; and he has worked in New York City as a mastering engineer at Sterling Sound Studios, as a sleep laboratory experimenter at Montefiore Medical Center, as an audio researcher for ZBS Media Foundation, and as music director of the Multigravitational Aerodance Group. Richard is presently co-editor of EAR magazine (326 Spring St., NYC, 10013), proprietor of the Ear Inn, and events coordinator for the New Wilderness Foundation. His compositions include voices, flutes, piano, organ, ensembles, orchestra, electronics, tape, bells, percussion, Indian instruments, participator audience, sleep, words, dry ice, film, video, and visual phenomena; and they have been performed throughout the United States and Canada, in addition to India, Milan, Florence, Cologne, Berlin, Stockholm, and Copenhagen.

Richard admits that John Cage and Philip Corner have been most influential with respect to the formulation of his compositional aesthetic. They have led Richard to realize that music is inherently social, and that it needs to be actualized in a social context, i.e., an audience is necessary. (This social attitude prevails throughout Richard's music, and is a further indication of the current movement among artists to disassociate themselves and their work from the 'ivory tower' complex). Richard's method of 'reaching' his audiences induced the following concert description by a Colonge reporter: " ... the border between art and non-art and the limits between artist and non-artist through spontaneity were largely eliminated." We will now attempt to discover the basis upon which Richard has developed his own compositional vocabulary.

During the early 1970's, while Richard was associated with the sleep lab at Montefiore Medical Center and the ZBS Media Foundation, he began his attempt to express the mental and physical phenomena of the dream state as sound. One of the main attractions to this realm of research was the general inability of people to recall their dreams accurately; scientists surmount this obstacle by incorporating sophiscated instruments to measure the heart beat, respiration, brain waves, etc. during the dream state. Richard undertook a 'musical' approach to this problem, from which three pieces evolved -- "Sleepwhistle" (1975), "Heartwhistle" (1975), and "Dreamsound" (1976).

The first of these, "Sleepwhistle," is described by Richard as a 'living sound sculpture'. He goes to sleep in a hammock -- before an audience -- with a few various sized whistles in his mouth; his respiration is thereby converted into the sound of gently blown whistles. Incidentally, in order to insure that he does sleep, Richard refrains from sleeping the previous night. Although these sleep-dream pieces are not directly involved with electronics, a synthesizer interface is easily developed. Figure 1 illustrates one such possibility by using an envelope follower to convert the amplitude of the whistles to a corresponding control voltage, which in turn may regulate VCO's, VCF's, and/or VCA's.



1

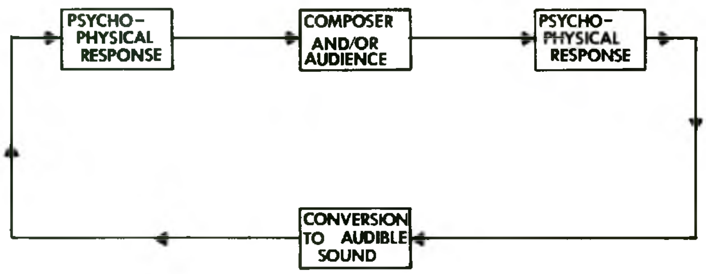
"Heartwhistle" is an audience participation piece based upon the varying pulse of the heartbeat -- that fundamental reference point (of the body) to the music we hear and all our actions. The audience is directed to find and to concentrate on its pulse, leaving one hand free to manifest the pulse by some percussive sound. After playing in time with the pulse, a continuous tone is to be whistled and held as long as possible; the tone may be changed only when a breath is taken. Richard's instructions to the audience accurately depict the resultant sonorous environment: "We will slowly build a cloud of pulses and then add our tones. We should individually fade in and out of silence, and as a group, experience a changing sound texture." Again, electronics are not present, but Richard is now developing a viable path of communication between himself (composer) and the audience. In these early works he is concerned primarily with establishing a social rapport with his audience, reflecting the aforementioned influence of John Cage and Philip Corner.

"Dreamsound," an event for sleeping audience, is an extension of the previous works, and it involves the creation and maintenance of a controlled environment. The audience (maximum of 40 people) is induced to sleep -- consequently to dream -- via a multitude of external stimuli: low light, films of a fireplace and candle flame, a silent television, serving warm milk with nutmeg and camomile tea, intermittant release of smells (e.g. concrete, including the following: background sounds (crickets, old popular songs with sleep/dream references, Bach's "Goldberg Variations," Ainu Shaman chant, Malaysian Temier dream songs, beating sine tones); "ghost cartoon" (secretly recorded conversations with visual effects); tape of Tibetan bells while recording each individual's voice message for auto-suggestive playback during sleep; tape of ocean surf until audience is asleep; intermittant sound effects (footsteps, rain, a car driving away, heartbeat, a telephone dialing, laughter, etc.); soft playback to each sleeper's ear of own voice; "Spirits" (orginal composition for piano and electronics---to be discussed later); "taps" (in slowing time on a flute); and dawn bird sounds as people arise on their own time.

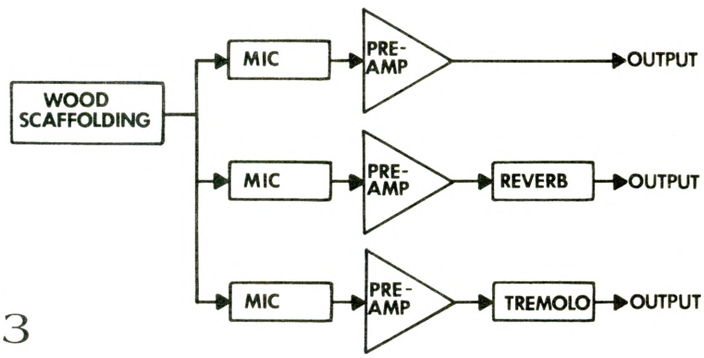
"Dreamsound" is a social event wherein sound functions as one of the foremost catalysts. Moreover, this event is directed toward the subconscious -- realized fully only in the dream state so that upon awakening the audience possesses but varied recollections of the past experience. That this piece provokes truly individual audience responses, thereby fulfilling Richard's fundamental social criteria, is evidenced by the following observation; "There were very diverse perceptions of what happened in the night. Some were reasonable misinterpretations; others were vividly imaginative. Each participant responded to different elements and no two recollections of the event were quite the same."

In the three aforementioned works Richard concentrates on people, both himself and the audience, as the pivot point of a simple feedback network (see Figure 2). A psycho-physical response (e.g. respiration, heartbeat etc.) is converted to an audible sound (e.g. whistle, tapping) which in turn elicits continued, sometimes changed, psycho-physical responses. To speak of this process quite descriptively we may say that Richard converts the inner vibrations of the human body to sound. In the next group of pieces to be discussed Richard uses contact microphones and/or transducers to pick up the vibrations of objects.

2



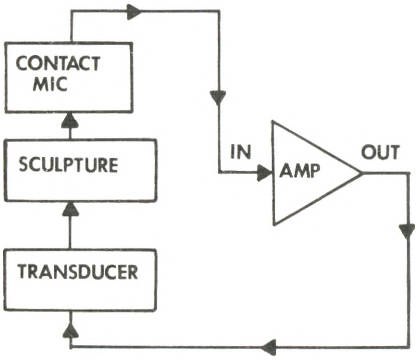
From 1974 until 1977 Richard was musical director of the Multigravitational Aerodance Group, a dance company that performed in mid-air via horizontally streched ropes (see accompanying photo). The music for this ensenble was both acoustic and electronic. The former consisted of whistles, saws etc., whereas the latter was obtained by placement of contact microphones directly on the wood scaffolding that supported the horizontal ropes. Figure 3 illustrates the electronic modification system used for the Multigravitational Aerodance Group. In this instance Richard took advantage of the extraneous sounds produced by the vibrations of the ropes and scaffolding to generate a synchronized, quasi rhythmic, accompaniment to the movement of the dancers. The resultant sonorous textures therefore comprised three main groups: acoustic, electronic, and combined acoustic-electronic.



3

Much of Richard's music is interdependent with other events, i.e. audience responses, movement of dancers etc. "Metal Sculptures" (1975) and "India Transformed" (1975), realized in collaboration with Indian sculptress Sari Dienes, continue this procedure. "Metal Sculptures" consists of three large works by Sari Dienes. They were constructed from 'found objects', and include the following: bottle garden (fish tank filled with multi-colored bottles); metal sculpture (air-conditioning ducts, mail boxes etc.); and totem pole (millinery hat forms burnt to give charred appearance). Richard then attached large transducers to each of the three sculptures to produce what he calls "resonance-sounding," for which a schematic is given in Figure 4. The transducers, wired to the amplifier output, are screwed into the sculptures so that the sculptures replace conventional loudspeakers; the vibration of the sculptures causes them to become loudspeakers, resulting in three distinct resonant (i.e. loudspeaker) timbres---glass, metal, and charred hat forms.

4



The network is completed by attachment of contact microphones to the amplifier input. The ensuing feedback loops cause each of the sculptures to generate their own sound and, due to their distinct material construction, yield discrete timbral and frequency spectra.

"India Transformed," again prepared in collaboration with Sari Dienes, utilized the aforementioned feedback technique, prerecorded tapes, and traditional Indian instruments. The tapes include flute harmonics, wooden well-wheel gears, drums, bells, stick dancers, chanting, a cremation procession, and sounds of the beach. The feedback network (see Figure 5) incorporates two large Tibetan Temple cymbals, one connected to a contact microphone, and the other attached to a transducer. The principle of operation is analogous to that of "Metal Sculptures," whereby one of the cymbals assumes the function of loudspeaker. In this instance, however, the distance separating the hand-held cymbals may be varied to produce different timbres, and the cymbals need not touch for sound to be produced. These feedback techniques reflect similar works of David Tudor and Stockhausen.

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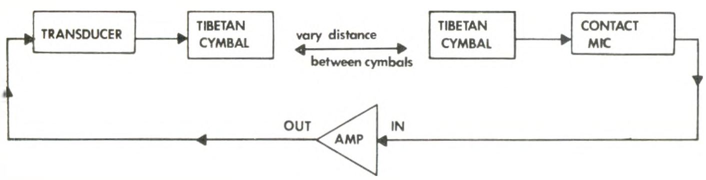
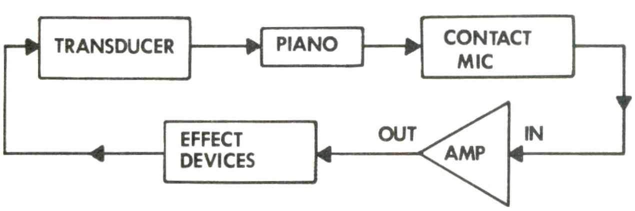


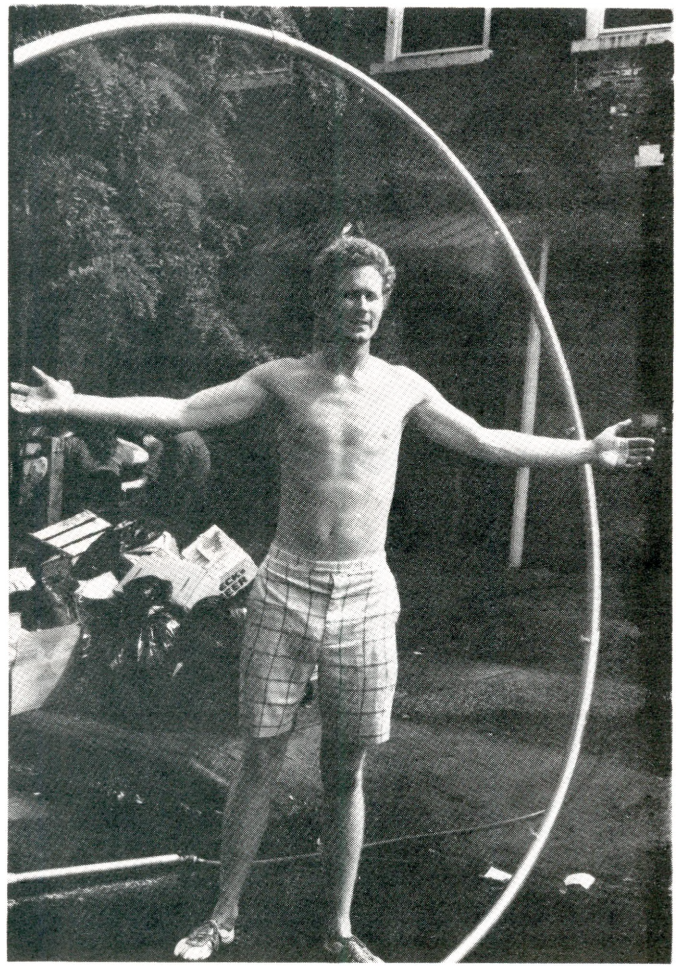
Richard applied the resonance-sounding feedback system in "Spirits" (1976), for piano and live electronics. Like the provious works, a 'found object' (in this instance piano) is played, i.e. resonated, through the piano's soundboard. Contact microphones, placed on various regions of the piano, generate different resonances; and the piano may be played in the traditional manner, along with the more recent techniques of playing on the strings, etc. Additional timbral diversity was obtained by including simple effect-devices (filter, tremolo, etc.) between the amplifier output and the transducer, shown in Figure 6.

6



"Late Nite TV" (1977) is another event oriented piece for TV movie and dry-ice. Hot objects, e.g. heated coins, soldering irons, etc. are placed on the dry-ice to produce a wide range of 'hisses', and the overall effect is reminiscent of the sonorous textures frequently encountered in the works of John Cage, David Tudor, et al. Yet another work of this nature, "Chair for John Cage" (1974), demonstrates Richard's ability to synthesize seemingly diverse elements. A push-button switch, wired to a phonograph turntable, is mounted on the seat of a chair; and an off-center recording of James Dean's hit song "Big Bad John" is placed on the turntable. As members of the audience sit on the chair they activate, via the push-button switch, the James Dean recording. Due to the off-center nature of the record, however, the stylus jumps, skips, and slides randomly across its surface, giving the impression of a warped record.

Throughout the past four years Richard has been in the process of developing a new disc mastering technique, whereby a standard LP record is pressed whose grooves criss-cross. The stylus wanders across the surface of the record during playback in much the same way as occurred in "Chair for John Cage," so that compositions recorded in this manner are different in each playing. This compositional aesthetic is consistent with Richard's other works, and it also reveals his



deep understanding of the music of John Cage and related Oriental philosophic ideas. Although many young composers have more or less drifted in this direction because it was currently in vogue, this is not true of Richard. His knowledge in these areas was not gained through second-hand methods, but was assimilated by studying and working with musicians John Cage and David Tudor, by living in India, and by working with artists as Sari Dienes, Nam June Paik, and Dali.

Perhaps the most refreshing aspect of Richard's music is its appropriateness for individual events. Regardless of the electronic complexities all of the sounds appear in context with one another, and often the listener is not even aware of the physical nature of the ensuing sounds. Such sonorous homogeneity is indicative of the balance that exists in all of Richard's work. While talking to Richard I asked him if he wished to make a statement for this column. He responded: "Persistance is the only way to get anything done, and not to get anything else done."

continued from page 24....
the largest corporations use in developing their products. Finally, there's always the chance that you will be able to sell your ideas. Having them well-documented not only makes the sale easier, it also gives you added experience in working in this professional manner.

Possibilities of uses for your documentation after completion of the project need not be limited to use as a "service manual" for the module, or to selling the module itself. There are a number of technical and hobbyist publications which are in continuous (and in some cases, desperate) need of articles for publication. With full documentation prepared, it is a simple matter to spend a few evenings rewriting your experience into prose for submission to such magazines. Black and white photos of the completed project, as well as photos during critical parts of assembly (you may want to keep this in mind during construction if you even suspect that you might want to get the project published), will round out your manuscript to present a package which most magazines would love to see.

Finally, after you have finished the project's documentation, make a copy of it. In fact, punching both the original and the copy and putting them in two seperate three ring binders is a good idea. The world's best documentation is not useful if it's packed away in some forgotten box.

Next, store the binders in two different places. The one binder you'll probably want close at hand so you can use it as a reference. The other should be kept across town somewhere. This way, in case one copy is damaged you'll not have lost all your documentation. You will have spent many hours in preparing this documentation and if it is lost you will have lost much of that information. Futhermore, your insurance company will not reimburse you for your writing time. It seems foolish not to take this extra precaution.

After all my proselytizing, I imagine you would expect me to suggest that you keep a pen and paper near your synthesizer so you can document the unusual sounds that you stumble upon. Not so. If you understand your instrument, it'll never suprise you. The suprises always come from your own ears and there's only one way and only one reason to document that, and that's as a piece of music.

record or otherwise. This capability should be of particular interest to those working with electronic music.

For example, if a pure left signal is placed into the Composer input, it will appear on the left rear channel output. Likewise, a pure right signal will be directed to the right rear output. The intermediate signals between center front and pure right or left signals will fill in the sound field. Out of phase information will be placed in the back channel quadrants. Some of the commercial recordings by Tomita are excellent examples of the capabilities of the Composer.

For copies of their informative brochure and product reviews, contact: Audionics of Oregon, Suite 160, 10950 SW 5 Ave., Beaverton, OR 97005, Ph. (503) 641-5225.

NEW DRUMMER

Boss Products introduces a major development in the field of musical rhythm devices with the introduction of the Boss Dr. Rhythm. The Dr. Rhythm contains a microcomputer that lets the musician compose, arrange, and program his own rhythms that can be recalled instantly at the touch of a button.

The Dr. Rhythm allows the programming of six sixteen step rhythms and two twelve step rhythms. Each of these rhythm patterns can contain up to three independently programmed percussion sounds: Snare Drum, Rim Shot, and Bass Drum. To this can be added programmed Accents (to liven up each rhythm), and a Hi Hat can be added in either of two patterns.

Programming of the rhythms is easily accomplished by means of two controls; one produces a beat and the other produces a rest as the rhythm is moved through its sixteen or twelve step cycle. Programming in this manner provides completely accurate rhythms that can easily be programmed by anyone with a fundamental knowledge of music. Also, each program can contain an A or B version of the rhythm, doubling the effective memory capacity and also allowing rhythms that extend over two measures.

An Accent knob lets you control the strength of the accents added to the program. Other controls include: Volume, Tone and Tempo. The Start/ Stop function can be activated by means of a remote footswitch, and a jack is provided for synchronizing the Dr. Rhythm with a digital sequencer.

With low power consumption for long battery life, the programmed rhythms remain intact in memory when the unit is turned on and off, and even stay intact when the batteries are being changed.

The list price of the Dr. Rhythm is $199.50. For more information, contact: RolandCorp US, 2401 Saybrook Ave., Los Angeles, CA 90040.

Advertiser Index

When you write to or order from our advertisers, be sure to help out the Polyphony system by mentioning that you saw their ads and products in

POLYPHONY

LETTERS

continued from page 6....

Your help in circuit changes for these would be much appreciated.

Steve Latham
New York City

How about it readers? If you have worked on any modifications such as these, write them up and send them in for publication. Equivalent modifications on similar drum machines (such as the new Roland and Korg machines) would also be welcome. Drum machines have a very definite place in the current music scene and, by exchanging our ideas and modifications, can become very useful additions to an electronic music studio, relieving larger gear for more complex or subtle sound generation.

Marvin

Live Wires

Your magazine is very well put together. I would like to see you put together a live performance article dealing with equipment setup, sound reinforcement, visual effects, and how a performer/ artist should write, select, and arrange the music to best suit live performance.

Dale Dyer

This is an excellent topic for not only a feature article, but a continuing column as well. We have looked for a person to do such a column in the past, but to no avail. If you are actively involved in live performance, or know of people who are, we would be more than interested to talk with them about doing an article or column.

Marvin

EQUIPMENT EXCHANGE

List your equipment or services for sale, trade, or wanted, plus job openings, positions wanted, and so on. Keep listings brief; enclose $1.00 for each 10 words. Prices, zip, and phone numbers count as 1 word each. Display classifieds are available for commercial ads; rates are $12 per column inch (minimum, camera ready). 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.

Services

WE'VE GOT BBDs! MN3005- 4096 stage BBD, MN3010- dual 512 stage BBD, SAD-1024, CMOS (CD40106 Hex Schmitt Trigger and many more), LM723 voltage regulator, CA3080E OTA. Great prices! Catalog and ap sheets due out soon. For more info: E Systems, Box 5305, Berkeley, CA 94705.

PET-MUSE: PET-to-Paia music software for Commodore Pet computer. $1.50 per program. Send SASE for 3 page list. R. Grokett, 401 Monument Road, #123, Jacksonville, FL 32211.

CUSTOM ELECTRONIC MUSIC equipment consulting service. Anything from just the design to complete fabrication. Also custom modification of existing equipment. Other items soon to be available. For more information or estimate, write: Jim Patchell, 533 San Blas Place, Santa Barbara, CA 93111. x

Literature

HOW TO START your own part time recording business. Our guide discusses planning, business deductions, selecting equipment, types of services, recording for demos, churches, schools, etc.; advertising and many more suggestions to make a profit using your own equipment. Send $3.95 to HSC, PO Box 4870-P, Austin, TX 78765. x

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PROGRESSIVE ROCK is discussed monthly in ANY COLOUR YOU LIKE. Contributors and members are wanted for this open forum. Write for details: John Loffink, 428 Citrus Rd., Melbourne, FL 32935. x

Wanted

EML 200 and POLYBOX wanted. Reasonable trade. See ad under Equipment For Sale. Gary Baker, 10786 Owens Pl., Tujunga, CA 91042, Ph. (213) 352-5616.

Music equipment

PAIA 4700S systems (2) $250 each or best offer. Reasonable trade. See ad under Equipment Wanted. Gary Baker, 10786 Owens Pl., Tujunga, CA 91042, Ph. (213) 352-5616.

PROPHET 5 for sale. Expertly maintained, with extras. $2500. Optional 5 voice external CV and Gate interface, $800. (408) 335-5107 / 429-9147.

BLACET SYN-BOW, fully assembled, sounds great. Sacrifice at $85 ppd. Gerard Pardeilhan, 1533 Berkeley Way, #4, Berkeley, CA 94703, Ph. (415) 540-0221 evenings.

4700 SYNTHESIZER for sale: 8780, (2)4720, 4740, 4750, 4712, 4770, 2720-1, 2720-3B, 2720-3L, (2)4761 cabinets, digital keyboard, home built regulated power supply, patch cords, documentation. Calibrated, good condition. $475 or offer. Mark Carroll, 1550 Beverly Pl., Berkeley, CA 94706.

KIM-1 COMPUTER with power supply, case, all manuals, and Microchess on tape - $125. Two SWTPC Tigersaurus 250 watt amp kits - one partially assembled, one in unopened box - $125 each. Will trade for synthesizer modules or ???. Jim Einolf, PO Box 162, Franktown, CO 80116, Ph. (303) 688-0294.

E-MU SYSTEMS synthesizer with complete microprocessor control, consisting of: 4060 uP Kbd with 16K RAM (i.e. 2,000 key depressions of memory - expandable to 6,000 and more with DOS), cassette load and dump, and many programs available through Emu; main frame of 24 state of the art modules, power supply, walnut cabinet, patch cords, firm wires, connecting cables, documents, etc. For the serious synthesists only! Burt Alcantara, Ph. (212) 691-7805.

PAIA MODULES: 8780, (5)4740, (3)4720, (2)4730, (2)4710, 2720-1, 2720-2A, 2720-5. All scope adjusted and ¼" audio jacks. $250 or separate. Dave Biddle, 4516 Navarre Rd. #26, Canton, OH 44706.

ORCHESTRON (Laser Optics Disc) includes Hammond B-3, violin, flute, and choir discs. Needs some work. Cost new: $2500. Best offer. Steven Halpern, 620 Taylor Way, #14, Belmont, CA 94002, Ph. (415) 592-3300. x

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E-MU SYSTEM for sale: mono voice & 2 sequencer voices plus lots of peripheral modules in an E-mu main cabinet. Perfect for college use or serious studio work. Expertly maintained, with warranty. $6850. Contact Ed Rudnick at (408) 335-5107 or 429-9147.

4700 MODULES: (3)4720, 4730, 4712, 4710, (3)4770, 2720 modules: -5, -11, -12. $275. Gnome $50. All in good shape with documentation. Mickey Le Blanc, 128 E. Summit, San Antonio, TX 78212, Ph. (512) 735-2149.

MISC PARTS, UNUSED: Proto board #104, $50; 2 Emu demo boards, $500; Emu hi gain amp submod, $55; Emu KB submod, $250; 79 C&K flat handle mixed pole/throw switches, $138; 102 pots, $90; 135 LEDs, $15; SSM 5101, $7; 2 SSM 2050, $13; 4 SSM 2040, $35; Belden 8221 RG59U patch cord wire in 3 lengths totaling 120', $17; 172 5" reels and boxes, $67; 2 pitch wheels 100K, $20; 3 Q81, $9; MXR Auto Flanger, Phaser, and power supply, $300 used; 15 electronics books, $25 used. Burt Alcantara, Ph. (212) 691-7805.

COMPUTER SYSTEM: 8700 w/ page 2&3 RAM, Mus-1 PROM, CS87 Cassette Interface, connectors, in custom Hamilton case. Built & tested by Paia, PS-87, 8780, 8781. $450. MODULES: 2720-7, -11, 4710, -20, -40, (2)4770. $135. 2720-8 keyboard and case w/EK3/KBM-1 digital encoder, Rnd/Arp/Variable Delay options in custom case, 4-way joystick. $150. GNOME w/vco, vcf, vca c.v. inputs, "VCA Repeat", schematics, Phlanger w/ "Sweep Defeat" & low noise modifications. $100. Everything in excellent condition & working to specs or better. Each includes all manuals. Sold separately, packages total $835. Will sell all four, including wide assortment of patch cords, for $750 or best offer. Jeff Wurstner, 8 Hickory Place, Seneca Falls, NY 13148.

DATEL SYSTEM 256 Computer I/O Analog Data Distribution System. 1) 16 lab quality (suitable for audio/ video rate outputs) output channels with 16 D/A converters (2 usec), ±10 V. outputs. 2) Address/ control/ interface logic. 3) Power supply; 19" rack mount cabinet with front panel (local) control and display; cables. 4) S-100 interface card/ cables (standard interfaces for PDP-11s, others available from Datel). 5) Perfect for audio/ video production, machine or analog process control or home automation. 6) Can be expanded to 32 D/A converters and 256 A/D channels (available from Datel). List $3350; best offer. Ph. (415) 456-0170.

SWTPC 2AS-A Stereo Analog Delay, factory assembled, unused, users manual. $200. Contact: ETC (Entertainment Technology Consultants), 2600 NW 30th, Oklahoma City, OK 73112. x

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ARP QUARTET- brass, strings, piano and organ voices. 4 octave keyboard. Brand new. Paid $800, asking $600. Gerard Pardeilhan, 1533 Berkeley Way, #4, Berkeley, CA 94703, Ph. (415) 540-0221 evenings.

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360 SYSTEMS Spectre OEM circuit board (interface section only). Fully trimmed and tested, includes power supply, PVC, string select circuitry, hex fuzz, octave and transpose selects, hex pickup and documentation. $625. Kevin Monahan, 2360 7th Ave., Santa Cruz, CA 95062, (408) 476-5630. x

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HEATHKIT IT-3121 Curve Tracer, documentation, cables, $75 ppd. Realistic sound level meter, padded case, documentation, $30 ppd. ETC (Entertainment Technology Consultants), 2600 NW 30th, Oklahoma City, OK 73112. x

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HEXAPHONIC SYNTHESIZER- expanded Paia P4700J includes computer, 12 channels S/H, 6 VCOs, 4 VCFs, 8 VCAs, 2 LFOs, 2 reverbs, 8 channel mixer, and 3 custom modules. $900. (303) 722-0508, 6 PM MT. x

Events


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