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WENDY CARLOS INTERVIEW

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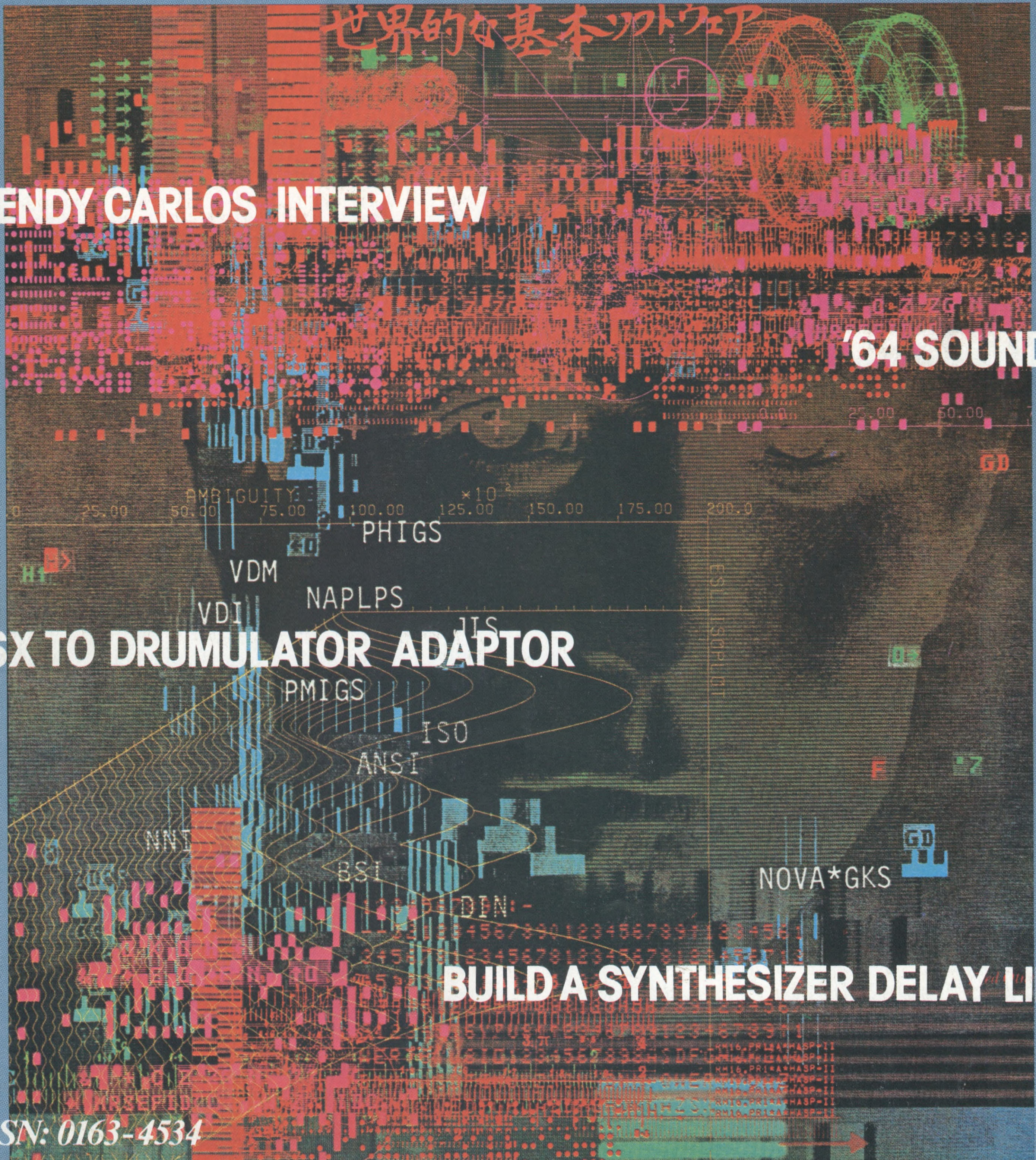

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Polyphony

Volume 9, Number 4

JUNE, 1984

FEATURES

COLUMNS

REGULARS

On The Cover: Computer Graphics generated by the NOVA GKS Details in Editor's Notes - Page 6.

Letters

CORRECTIONS & UPDATES

In the October 1983 issue, page 24, under "Click Track Formulas": Film is projected at 24 frames per second, not 245 frames per second as given in the article.

In Robert Carlberg's February review, there were a couple of missed numbers. Mark Isham's **Vapor Drawings** is Windham Hill #1027, and K. Leimer's **Imposed Order** stock number is POL .17. Also, in the review of Brian Eno's **Working Backwards**, the word "unrealised" should read "unreleased".

Several errors crept into the artwork for Ron Oberholtzer's DIY Compander project in the February 1984 issue. Sorry! Page 11, Fig. 2: The input to the gain cell from R7,8 pin numbers should be 5,11 and not 7,9. Page 12, Fig. 3: The op amps listed as 1/2 IC3 should be 1/4 IC3. The unlabelled resistor going to pins 3,13 or 1/2 IC4 should be R17,18. Page 12, Fig. 4: The polarity of diodes D5 and D6 are incorrect and should be reversed.

Finally, due to parts kit production delays the second part of Kirk Austin's article on a remote MIDI controller has been postponed until next issue.

SCAM?

This letter serves a dual purpose. First, I would like to express my thanks for Polyphony magazine. Its concepts, contents, and all-round presentation make it a superb magazine.

Second, I wrote to a company called SYNTEX (702 117th Pl. SW, Everest, WA 98204) that had offered plans for a variety of synthesizers after seeing an ad in an electronics magazine. I sent off a check for the plans plus postage over a year ago, and have not heard anything since. Could you please look into this for me?

Tim Corfield
Wiley Park, N.S.W. Australia

Tim -- I checked into SYNTEX and would recommend that readers avoid this company until we see some evidence that they are indeed "real". I'd also like to emphasize that the vast majority of companies are honest; however, every now and then a scam pops up. If any readers have problems with particular companies, be sure to write and we'll see what we can do for you.

GUITAR SYNTHESIS

Are there are articles giving an overview of polyphonic guitar synthesizers? And, do you know of any particular artist/musician who creates their music wholly or mostly with a guitar synthesizer, to whom I might listen? After reading your reference in the December issue of Polyphony to GR-300 modifications that you had made, I'm starting to feel that I might consider concentrating on a guitar rather than keyboard synthesizer system. Can you offer any comments?

Glynn Black
Waynesboro, GA

Glynn -- Guitar synthesizers have had a somewhat unfortunate history since ARP's Avatar promised the moon and delivered considerably less. Many models didn't work right, were overpriced, or sacrificed those qualities which make guitarists want to play guitar in the first place. The only remaining major guitar synthesizer company is Roland, who are doing quite well with their GR-300 guitar synthesizer system and have just introduced the GR-700 system (see Current Events in this issue). Write them for literature on what they offer.

To hear guitar synthesizers in action, check out Chuck Hammer's Guitarchitecture. I haven't heard the record myself, but understand that it is mostly guitar synthesizer. Also, Robert Fripp (of King Crimson) and Andy Summers (of the Police) use the GR-300 a lot in their music.

Comments? To me, a guitar is a very different instrument from a keyboard, and what I want out of a guitar synthesizer is different from what I want out of a keyboard synthesizer. Although I play both guitar and keyboard, I generally use keyboard synthesizer more than guitar synthesizer; nonetheless, there are times when it seems that only the GR-300, and not a keyboard synthesizer, can do the job. (And the hex fuzz capability of the guitar synth is a joy.)

MIDI

First, congratulations on a terrific magazine, one I look forward to constantly. My interests lie in the areas of computer interfaces with both digital and analog synthesizer. One project I would like to see is a MIDI to CV/GATE converter for use with older monophonic synthesizers.

I built the Snare+ a couple of months ago and use it constantly. I am also looking forward to building the Hip Bass Drum. One modification for these two projects I have thought of, but need someone else to design, is voltage control of the drum pitch. This would allow the playing of tuned drums from a keyboard, computer, or MIDI.

Dwight Hawes
North Vancouver, Canada

Dwight -- The Snare+ and Hip Bass Drum are not easily modified for voltage control; they are designed primarily with cost-effectiveness in mind, which precludes precision voltage control.

Re a MIDI to CV converter, try J. L. Cooper Electronics (2800 S. Washington Blvd., Marina Del Rey, CA 90292). They make a variety of interface units that adapt MIDI to non-MIDI products.

DATABANK CORRECTION

At least you can't bounce a check at the Databank, but there was a pretty glaring error on Page 23 of the April '84 issue: The 741 pinout bears no relationship to reality as we know it. For the correct pinout, refer to the LF351 pinout on the same page, which is pin-for-pin compatible with the 741. Our apologies for any inconvenience or confusion this may have caused.

Editor's Notes

This month's cover comes to us courtesy of the people at Nova Graphics International Corporation. Assistant Editor Vanessa Else spotted some of their work in IEEE COMPUTER magazine, and thought it would make a great cover for a technically-oriented magazine such as Polyphony. They were kind enough to let us use their graphics for the cover; all they asked in return was that we talk a bit about their latest development, NOVA*GKS. That seemed fair to us, and since many of you are into computers and graphics, here's the scoop.

Nova produces the NOVA*GKS Operating Systems Software and Applications Software, which is a full implementation of the Graphical Kernel System (GKS) international programming standard. It provides programmers with an international, common graphics language; and since it describes graphics generically, applications can be both computer and device independent. The software itself uses a unique distributed architecture which prevents having to tie the graphics application software to specific operating environments and individual graphics devices. As a result, computer graphics hardware manufacturers can construct new products which adhere to the GKS standard and be assured that all features of both current and future products will be operable in any GKS-compatible environment. NOVA*GKS also streamlines the programming process for applications developers, and systems integrators who use NOVA*GKS can create systems with different brands of graphics input and output devices since NOVA*GKS is device independent.

Perhaps of even greater interest to Polyphony readers is that a microcomputer version of NOVA*GKS is scheduled for release in July. Written in "C", this new version of NOVA*GKS is compatible with Nova's mainframe/host version, and runs on virtually any 16-bit microcomputer as well as the new 32-bit supermicros. Like the original NOVA*GKS, the microcomputer version features multilayer, distributed architecture that is both hardware and application tuneable. To find out more about what Nova Graphics is up to, write them at 1015 Bee Cave Woods, Austin, TX 78746.

* * * * *

As you've probably noticed, there are more and more advertisers in Polyphony these days. And why not? We've got a great, technically sophisticated readership, and because the magazine is so specialized, our rates are very inexpensive compared to larger, more general circulation magazines.

But what's most important from my standpoint is that these are quality advertisers. They advertise musically useful products, don't try to interfere with editorial policy and are very supportive of the industry as a whole. And it looks like we'll be getting more advertisers soon, so we're definitely moving up in the world.

What can you do to show your appreciation for their support? Simple: If you see something in the magazine (either an ad or Current Events mention) that catches your eye, when you write away for information be sure to say that you saw it in Polyphony. That lets the advertisers know their message is getting through, and also gives them some feedback on what ads are most effective (as well as which products draw the most attention). If they don't know you're reading their ads, they won't continue to advertise...so make sure you mention Polyphony when you ask for information. It helps them, it helps us, and it helps you by letting us put out a better magazine.

* * * * *

Omission time: In our hurry to get the April issue out, we forgot to include photo credits on the NAMM show pictures. So, a belated thank you to Vanessa Else for coming through with the photos that really helped to dress up the article.

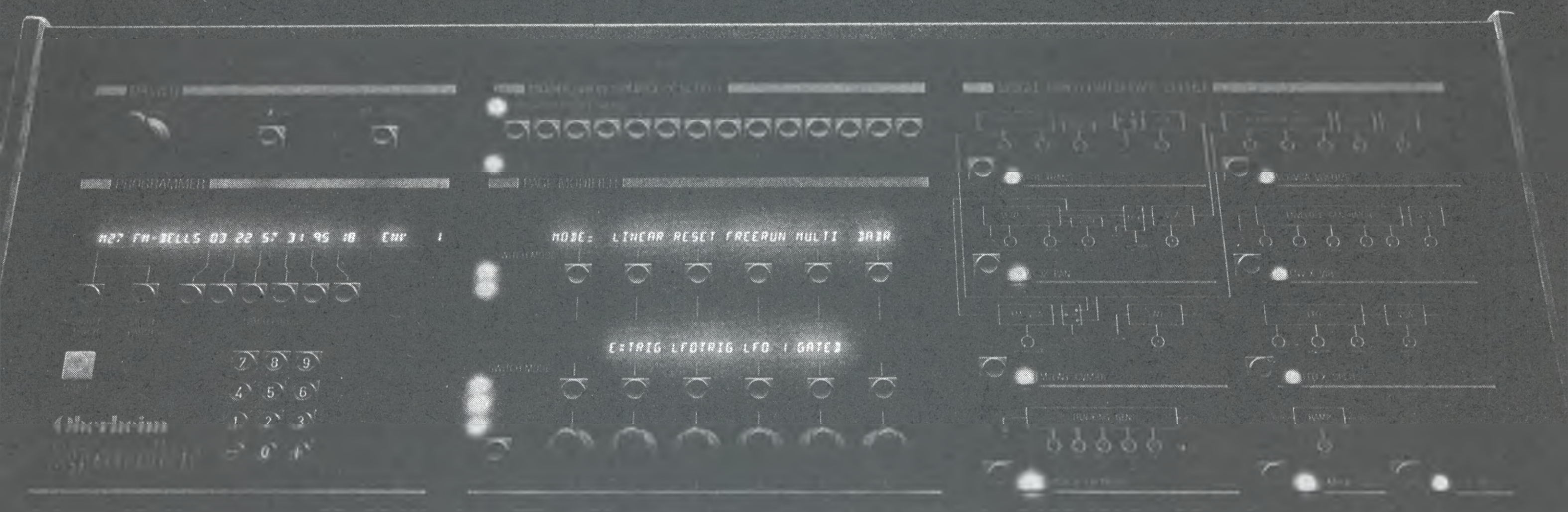
Another omission: Because we're working hard to get the magazine on schedule, there was only a few weeks between the time we finished the April and June issues -- which left Robert Carlberg without enough time to gather the records necessary for another installment of Re:View. Our apologies to Robert and his many fans; Re:View will be back next issue. And by the way, thanks to Robert for providing us with photos of the albums being reviewed.

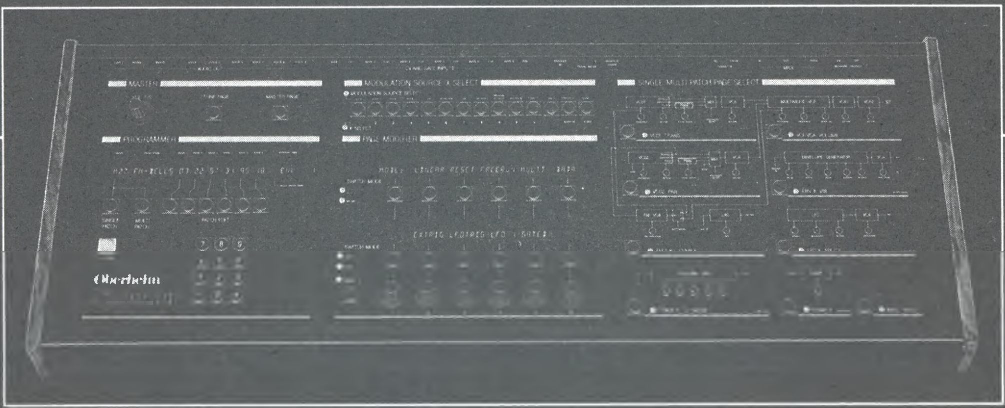
* * * * *

We just got back from the Summer 84 NAMM show, so expect a full report next issue. As usual, there were quite a few exciting new products, people to talk to, and things to see. Also, video producer David Karr and I did a one-hour NAMM show video report which will be offered through this and other magazines in VHS format. The NAMM videotape is the next best thing to being there, so watch for it.

Craig Anderton

A FANTASY REALIZED





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Applied Synthesis:

Overdubbing & Licking the Envelope Problem

By Bill Rhodes

When we speak of studio multi-track recording we automatically involve ourselves with the science of overdubbing (O/D). Overdubbing can be a pain or a joy; this depends upon the musician's ability to deal with this important process through the use of precise playing and proper envelope selection. We can approach synthesis overdubbing in three basic ways: a) chordal or polyphonic O/D, b) linear O/D (melodic or counterpuntal) and, c) effect O/D.

Chordal O/D. Chordal or polyphonic O/D requires a less tedious technique than multi-tracking many single lines. Basic tracks involve the use of chords (of two, three, four, or more individual notes) and can be thought of as the sonic foundation from which the other types of O/D are added. Thanks to polyphonic synthesizers, it is much easier to layer multiple parts today than it was a few years ago, when mono synthesizers were in vogue and every line had to be recorded individually. Remember "Switched on Bach" by Wendy Carlos or "Realization For Rock Orchestra" by Larry Fast? Every single line that was recorded (except the mellotron in the case of Fast's album) had to be recorded one-at-a-time. Even adjusting separate oscillators to different intervals didn't help much.

Polyphonic O/D is not only a basic track technique, but can be used for brass or string ensemble dubs, chord clusters, or background ambience effects. However, unless you have multiple triggering on your synth, the envelopes will be non-independent when your fingers hit the keyboard. Multiple triggering allows each finger to execute an individual envelope characteristic, which is more controllable than single triggering. Single triggering responds to groups of notes only if they are played simultaneously, and does not allow individual finger control of envelope. Therefore, polyphonic O/D is easiest because many notes are played together and there is no need to worry about individual oscillator tunings (as required to make a chord with mono synths) or separate envelope hassles.

Linear O/D. Careful technique and approach is of the utmost importance when layering single-note-at-a-time structures. If we want to play a classical piece, for example, and use <u>one</u> track for each <u>non</u>-polyphonic (monophonic) instrument, we must consider the following factors.

A. Tempo vs. envelope: If the piece is rather difficult to play due to a fast tempo, any O/D must be played precisely in time with existing parts. All the tracks that are of a linear (melodic) quality must be played articulately and sound clean.

Remember too that if we are playing, for example, string, brass, french horn, flute, and oboe lines together their combined envelopes must sound correct. We know that every instrument has its own envelope characteristic so we must be aware of that when recording each instrument. (Refer to past <u>Applied Synthesis</u> columns for examples.) The reason why a great deal of synthesized music sounds fake or too mechanical is because all the tracks of all the instruments have one basic envelope. If all (or many) of the tracks have the same fast melody sequence and therefore must be played simultaneously, make sure the final mix of all those lines does not have a single envelope or organ "feel." Each instrument must sound and "feel" independent of each other or it will be a monophonic clutter of sound. Proper stereo placement and processing during mixdown is also important to give each voice its own identity.

B. Use of headphones: No matter how much some people complain about using headphones when recording, headphones are necessary. If you are listening to monitor speakers but are located at some distance from them, two phenomena occur. First, there is a minute but noticeable time delay between what you play and what you hear, thereby distorting the concept of envelope in your playing. Since envelope is a simple function of time versus volume, if the O/D is complex you might not sync that O/D correctly with the previous tracks because of this delay. Secondly, objects in the studio tend to rattle and give off other sounds that distract you while recording. How many times have you heard the snares ring, or the windows rattle, or strings vibrate when you are recording without headphones? Once again, a distraction can cause a time delay element when overdubbing. Most headphones (good ones) reproduce your sound with considerable quality and put the sounds directly in your ear -- not in the air!

C. Crescendo and decrescendo: The rising and falling action of sound once again involves the envelope. If you have many tracks that are crescendoing simultaneously, make sure the settings for each instrument and track have the correct attack characteristic. You may want to have a staggered or cascaded envelope of attack in some instances; adjust your envelope accordingly. The same goes for decrescendo (fading away) of volume.

You can approach the crescendo problem in another way by mechanically riding each fader in the final mix. This is difficult because if you have multiple tracks, you are going to need that many fingers just to control each individual track in the mix. However, you could submix those instruments having the same attack characteristics into two (L + R) stereo faders. The same comments apply to decrescendo.

D. Recording instruments in sequence. Always record the basic track (whether it be a polyphonic synthesizer foundation or rhythmic structure) as cleanly and precisely as possible. You will kill yourself trying to overdub to a track that is out-of-time, played sloppily, or poorly recorded.

I have found the following order of recording to be quite efficient when using an eight track to record keyboards and voices.

1. Basic tracks (if premix, left track)
2. Basic tracks (if premix, right track)
3. Bass
4. Rhythm structures for drive or continuity, or harmonic polyphonic brass or keyboard textures for rhythm.
5. Lead lines
6. Contrapuntal or counter melodies
7. Special effects, such as filter whistles, explosions, bells, cymbals, etc.
8. Voices

Tracks 1 and 2 are premixed basic tracks that could contain sequencers, digital or analog drums, and a polyphonic foundation mixed down to a stereo 2 track format from a multi-track preliminary recording. You may want to have more voices, so you should probably double up on certain tracks where you can. If you have 16 - 24 tracks free for recording you have less to worry about -- except for the bigger recording bill!

Effect overdubbing. Usually the effects are put on next to last or last because they are the "icing" on the recording. Envelope would be important in certain crescendoing filter sweeps, or filter modulated trills and whistle. For example, percussive sounds such as synthesized gongs, drums, bombs, explosions, +7 dB spikes(!), cymbals, etc. would be easier to perform because they are not extremely precise musical or melodic devices at all times. The envelope is important for proper attack decay and release, but the passages need not be thought of as individual notes that must be correct in pitch as well as timing with respect to previously recorded parts. We are not avoiding the envelope problem altogether, but these sounds require less overall precision.

As you can see, envelope is yet another parameter to consider when recording synthesizers. A good way to practice overdubbing and envelope technique is: a) Rent time at your local studio and bring your American Express Card or, b) practice the music you want to record at home using different instrument sounds and envelopes. If a piece requires a synthesized flute to play along with cello and trumpets, practice the passages on each individual "instrument" at home, with the correct envelope for each sound, before you spend your money in the studio...the only envelopes a studio knows about are the ones that contain bills for you!

An Interview With WENDY CARLOS

By: John K. Diliberto

In every art form there are dividing lines, points of demarcation that illuminate the shift from one style to the next. Charlie Parker neatly divided Be-Bop from Swing. Jack Kerouac clearly set off the Beat generation and Delacroix etched the cut from Romanticism to Impressionism. In the brief history of electronic music, Wendy Carlos is such a divider. Her importance approaches that of Robert Moog, with whom she is inextricably intwined.

Carlos's 1968 hit record, Switched-On Bach, propelled electronic music into popular consciousness. While many electronic purists cringe at the thought of imitative synthesis and the now cliched "Switched-On" approach, there can be little doubt that electronic synthesis would not be where it is now without it. Her Switched-On albums were masterworks of electronic craft, forcing the synthesizer into the rigorous terrain of Baroque and Classical masters (especially Bach). Breaking the rules is an important statement, but doing it after you've shown that you can play by them is profound. After proving that the synthesizer could be musical, Carlos wrenched the classical standards into the electronic age with her horrifying score to Stanley Kubrick's equally horrifying movie, A Clockwork Orange. The vocoded voices of Beethoven's 9th and the metallic decay of Purcell's Music for the Funeral of Queen Mary (The Clockwork Orange Theme) signalled Carlos's and electronic music's shift from the past into the future, whether Tomita wanted to come along or not. Her own composition, Timesteps, proved that no further questions needed to be asked about the validity of electronic music.

Unfortunately, few of Wendy Carlos's own compositions have been committed to vinyl, so the general public is still unaware of how far-reaching her conception of making music electronically stretches. A hint was gleamed from Kubrick's The Shining, but he only used two pieces from what was to be an entire film score.

Then there was the soundtrack to Tron, Disney's science-fiction computer epic. Here, Carlos alloyed electronic and acoustic sounds in a merger as complete as chromium. Using the digital synthesis of the GDS (General Development System) synthesizer, Carlos had the orchestra and synthesizer reinforcing each other. To the casual listener, it sounded orchestral; and in fact, many Carlos fans were disappointed that it wasn't an overtly electronic score. To be sure, the Tron soundtrack was not as striking musically as other works by Carlos, but it did open up the possibilities of electronic sound for her. She could make it sound electronic, acoustic, or like sounds you've never heard and she could do it with a full orchestra or the most advanced computer synthesis. As she showed on her Sonic Seasonings LP, the possibilities of music are as wide as nature permits.

Carlos employs the GDS synthesizer not only to make music but to create the basic sounds, the actual "clay" from which she forms her compositions. She eschews the pre-programmed approach of most synthesists, instead seeking out new sounds and forms that offer her the timbral richness and texture that she loves in acoustic instruments. Her modular Moog system, custom designed and modified by Robert Moog, still occupies a prominent spot in her multi-track recording studio; yet one suspects that it will occupy less space in her future work.

John Diliberto: When did you first start playing electronic instruments?
Wendy Carlos: I think a fair date would be '56, '57. By '57 I was already playing around with my own home-brewed stereo tape machine, and by '61 my own home-brewed quad machine. Usually the things that I would put on those machines were some form of musique concrete with a few oscillators added in. So it's the real American way of combining both the German and the French classic styles that eventually led to the traditional synthesizer, which seems to be more of an American invention in its own way.
JD: What inspired you to go in that direction?
WC: Records! The Brown University music department had collections of quite a few records. There were also silly myths going around that Brown had an audio-research laboratory, which now they do have, but at that time it was a hoax. I was naive enough to believe that they did have something like that, and that I was going to do something which would convince them that maybe they'd want me as a student when the time came for me to look for a college. The impetus was that funny turn-on to find ways of making sounds that would be exciting, using the new technology of stereo taping.

"...I suppose in one sense my background is almost stuffily academic, yet my attitude is anti-academic, maybe as a reaction."

JD: So you were just in high school and doing this on your own with no academic background.

WC: Nothing in this field, although I'd been studying piano since first grade and went through organ lessons as well. I had read a lot of books on counterpoint and harmony, so I suppose in one sense my background is almost stuffily academic, yet my attitude is anti-academic, maybe as a reaction.

JD: In 1958, all of your friends must have been listening to Elvis Presley and Bill Haley.

WC: Oh yes indeed! Were they ever! Funny thing is that music in the popular side of things during that particular period couldn't have bored me more. I just loathed 1-4-5 harmonies, even in other forms of music, or the simple chugga-chugga-chugga of a 4/4 beat. I was just simply intolerant as you are best able to do when you're that age. You can never quite pull it off as well later on.

"(Electronic music) was my way of making something that was more timely than Beethoven or Mozart, and yet was not a part of this boring staff that all my friends listened to."

Then when pop music shifted and became filled with all kinds of meaty things I suddenly was fascinated and wanted to make a hit in that area. Then when it reverted back in the mid-seventies to that same chugga-chugga, once again I said "Oh that's right! I used to hate that stuff then and I hate it now." So this (electronic music) was my way of making something that was more timely than Beethoven or Mozart, and yet was not a part of this boring stuff that all my friends listened to.

JD: When did you get involved in an academic sense?

WC: When I was going for my Bachelors Degree in music composition, I talked the faculty advisors at Brown University into allowing me to do a special research seminar as one of my music courses. I put together a couple of more or less serious works, limited very greatly by the available technology. If you want to talk about composition in a more serious manner you'd have to talk about when I got involved with the Columbia-Princeton Electronic Music Center. I came here (New York) in 1962 and did my first serious composition with the technology they had there. It involved an awful lot of splicing, but it allowed you to compose in a sense that was composition and not improvisation.

JD: As you look back on that whole era from the late 50s and the 60s with your music, Otto Luening's and Vladimir Ussachevsky's, do you see it as people experimenting and trying to find something or do you see these works as fully realized compositions and performances?

WC: I'd say the latter. There was experimentation, for sure. Even those who had been working at it for years, and had their techniques more or less perfected, still had some elements of experimentation going on. The process itself was unremoveable from that, in the same way that a jazz musician is always experimenting. But more than in traditional composition, there was a great deal of experimentation that went on in all of these works. They color the way you look at it now. You see them as something that arose from that period of history that cannot be removed, as well from that period as music that came both before and after, at least in other styles.

JD: When I listen to your early works like "Music For Flute and Magnetic Tape" and "Dialogue For Piano and Two Loudspeakers", there's a clear distinction between the electronic sound and the live musician. But with the Tron soundtrack, the acoustic and electronic sounds are much more intermeshed.

WC: Yeah, they came out of different things. Are you aware that at that time it was a deadly thing to play a tape at a concert? We all had our four-track tapes and we thought we'd just dim the house lights and leave a little light on the stage, announce the piece, step back and hit the start button and play the tape. It was horrible. No audience was ever able to get through more than a couple of minutes with feeling uncomfortable and you could feel it for them if you sat out there. We had to find something to do to focus the attention, which you get automatically with a live performance. So we just put the two together.

With Tron, it didn't matter one wit how you obtained any particular sound. It was the final product that counted. I was looking for the gradation of color that you can get by going from the most synthesized sounding electronic effects to the most orchestral and everything in between. In those earlier pieces you'd play up the contrast between the two -- the mechanistic portions of the rhythm that you would get from the tape, versus the live performer who could be more rhapsodic feeling and improvisatory. I simply wrote a score with two lines for the left loudspeaker, two lines for the right, two lines for the solo instrument or in a four-channel piece, a few more lines as well. And then I'd tediously realize that portion on tape until I had all the notes and timbres and everything was specified on the score. The performer would practice and synchronize with the tape and invent techniques to allow some freedom between where the synchronization points were. It was experimental in the execution, but the music was less experimental than the pure electronic music.

JD: How have multi-tracking techniques affected your music?

WC: The kind of control you get with multi-tracking is that of a live performer. You can actually spend the time worrying about each individual note, as a performer does on the cello. You're turning knobs and moving and shaping sound as you play it. You actually perform in your best performance of each line, line-by-line, note-by-note. With some of today's devices where you're hitting chords you're more like an organist. There's less of this (individual sound-shaping) and in my opinion the music suffers. It doesn't have quite the human side of it that I was excited about hearing in multi-track music, be it electronic, rock and roll of the time in the 60s which pushed the music from two, to four to sixteen tracks and upwards, adding on tracks ad infinitum.

JD: I notice that you have a modified quadraphonic set-up in your studio, and that a lot of your music is also concerned with the placement of sound in space.

WC: Yes, antiphony. It's an important parameter and the truth is there hasn't been that much done, although they fooled around with it years ago. The most recent example is Bartok's "Music For Strings, Percussion and Celeste." Antiphony is another axis of freedom. The idea is that part

"(Antiphony) is an important parameter and the truth is there hasn't been that much done... so much of classical music doesn't explore this other color on the palette."

of the excitement of the electronic medium is that it gives you possibilities that you couldn't have before. Distributing the sounds all around you is a nice way of making it clear what's happening in a piece of music. You could repeat a sound exactly except place it in a different location, which would create a different reaction in you than it would if it were all collapsed down to a couple of musicians sitting on stage a couple of hundred feet away from you. So much of classical music doesn't explore this other color on the palette.

JD: Do you think this is because your music is of a recorded medium instead of live performance?

"If I did a live performance I'd do what a lot of avant-garde composers do — I would request that certain instrmentalists sit in certain locations."

WC: No, I don't think so. If I did a live performance I'd do what a lot of avant-garde composers do -- I would request that certain instrumentalists sit in certain locations. In fact, in the orchestra for Tron, we had a reseating to pull off an antiphonal effect, which got ruined because of an ignorant engineer who collapsed it all down to mono. But it's nice, as long as you have two ears and a good acoustic environment, to be able to pull things apart that might otherwise blur together.

JD: Do you think that electronic instrument sounds are not as rich as acoustic ones?

"To me, synthesizers are like eating bland food like most Americans eat."

WC: Yeah! If you look on a graph they look like a computer plot. Live instruments look like someone tied a pen to a kitty's tail and it scrawled this plot while playing with a ball or something. Acoustic instruments have a wealth of detail, some of which is unimportant to the ear, but a lot of it is subtle stuff that is important to the ear and produces a little friction in the ear that spices up what you're hearing. To me, synthesizers are like eating bland food like most Americans eat. Once you get turned on to ethnic food it's like acoustic instruments. You start blending all kinds of exotic spices and herbs, and you miss that when suddenly you don't have it. It is important and it does make a difference.

"The GDS is the only instrument built that allows you to do things with the subtlety, precision and nuance — and the complexity — that exists in acoustic instruments."

Until the GDS came along there was nothing on the market to allow you to cook with exotic herbs and spices. The GDS is the only instrument built that allows you to do things with the subtlety, precision and nuance -- and the complexity -- that exists in acoustic instruments. But no one has been able to fully realize that yet, including me. It's a ton and ton of work and specification to get the instrument to be able to do that.

"...for every parameter you can control, you must control. A lot of people aren't willing to get into that amount of work."

Again, this recalls my old rule that for every parameter you can control, you must control. A lot of people aren't willing to get into that amount of work.

JD: With Tron, there's a merger of electronic and acoustic sounds.

WC: Well don't you think that's kind of the reason for continuing in this direction? If you're going to make instrumental colors on a digital synthesizer that sound more and more acoustic and if you keep your finger in the pie of all the contemporary things one can do with orchestras that sound more and more electronic, I think the two overlap and start blurring very violently. You're ending up with something that, as a composer, you need to face honestly and say, "OK, I've got two regions now that used to sit apart, but now they're two circles that are overlapping." I have to face that I'm working in this double range of possibilities in which some of my colors can be done with either the orchestra or an electronic device and no one will tell the difference, even me. Other colors could only be done by the orchestra and yet others by only the electronic device. It's like a multi-media event. You've got performers on stage, motion picture film, and you can do things by causing the two to interact. Tron was the first score to my knowledge that blurs the distinction with such a vengeance. I wanted a score where you couldn't tell one from the other and Tron was a perfect vehicle, being a film that explores the ambiguity between the real world and the electronic world.

JD: The music released from The Shining was mostly electronic, wasn't it?

WC: No, much of that was musique concrete. That's one of the things I miss about not working with Rachel Elkind. The other thing that can be added into the melting pot is musique concrete; it hasn't died. Because it seemed that electronic music grew up and forgot about it doesn't mean that we still can't go in with microphones, and take a bow with rosin and rub it against a trumpet mouthpiece or something and get a squeak and record it, process it and do something with it. You might come up with sounds that we can't presently make.

JD: When you use the GDS, is it like you're creating a whole new instrument or do you create a line note-by-note?

WC: You're creating a note at the moment. You try and make it so it's broad enough to be several notes at least, so you try to create an octave out of it if you can. Then in a compositional sense you don't want to think about that. So you put those down on a floppy disk and call up the sounds that you've worked on hours, days, weeks, or maybe years ago, and start playing with them the way a composer plays with flute, oboe, sine-wave oscillator, it doesn't matter. So there are two hats you have to wear. It's hard doing a composing step and a digital synthesis step at the same time. When you're making sounds on the GDS, very often it's both intuitive and logical. You're using the full brain pretty completely -- just as when you're composing you're involved with the drama, the tempo, the moment, the loudness and the line.

JD: Do you think, that even on the GDS, you can get the freedom of spontaneous expression you would obtain with a conventional instrument?

"I've come down hard on the artificial intelligensia. I just don't believe that we can describe what we embody as a performer."

WC: At the moment that you're performing, you're likely to do things that you can't specify. It's a spontaneous natural reaction. In fact, as a performer you train yourself to depend on that seat-of-the-pants instinctive way of working; if you had to think about every muscle you were moving, you'd never get anywhere. It's like driving an automobile. That kind of thing has got to be instinctive. Could you write a program to drive the car the way you do, or walk? It's why I've come down hard on the artificial intelligensia. I just don't believe that we can describe what we embody as a performer. I embody a human being that took years of practice and training and feedback with other human beings as part of something that has been passed from generations over centuries. But you don't really know what that "thing" underneath me on which I'm standing really consists of. So how can you specify it with the thoroughness and the rigor that these devices, like the digital computers of today, require. You can do it for an accounting job because you can teach somebody how to calculate an extrapolated regression, but it's hard to teach a task that you can't describe very well, namely, the tasks that you do more from muscular, physical things, or audible feedback tactile things ing on the touch sensitive keyboard like the GDS or the Moog keyboard that Bob (Moog) custom built for me. These allow you to stop thinking about the expressivity a little bit and start just feeling it. I think the problem with most electronic music is that the instruments don't allow you to express: they require that you concoct the expressiveness in your head where it doesn't belong, in the left hemisphere. The verbal, descriptive side of your brain is now doing a right hemisphere task, and it isn't capable of doing that very well. It will try its best, and some people come out with things that sound okay. It's sort of like the dancing dog; it's a miracle that it happens at all, not how well it happens.

"I think the problem with most electronic music is that the instruments don't allow you to express: they require that you concoct the expressieveness in your head where it doesn't belong..."

"...right now the field is depending more on craft and less on art, let's be honest."

Your question suggests that I should be able to craft together all of the nuance and performance value I might want with these instruments. And I'm saying to you that I can't do that. I challenge anyone to do that. You can't program the art. But you can program the craft and right now the field is depending more on craft and less on art, let's be honest.

JD: Do you think that the technology will date the music?

WC: Music is always that way. Beethoven would not have written quite the same music had he been born a hundred years later. His music is stylistically dated. Mozart is particularly so. He wrote very much in the accepted style of the time and he did it brilliantly. But the style is still there.

These instruments are definitely dating when they were done and so is the music that was done in those years, because it was done at a particular time in history. You're probably getting yourself into big trouble if you think you can get away from the dateability and try to write in a more cosmological sense. I doubt that that can be done since we're calling upon resources within our humanness and our humanness is very much more temporal, very non-cosmological.

JD: Where is it all going?

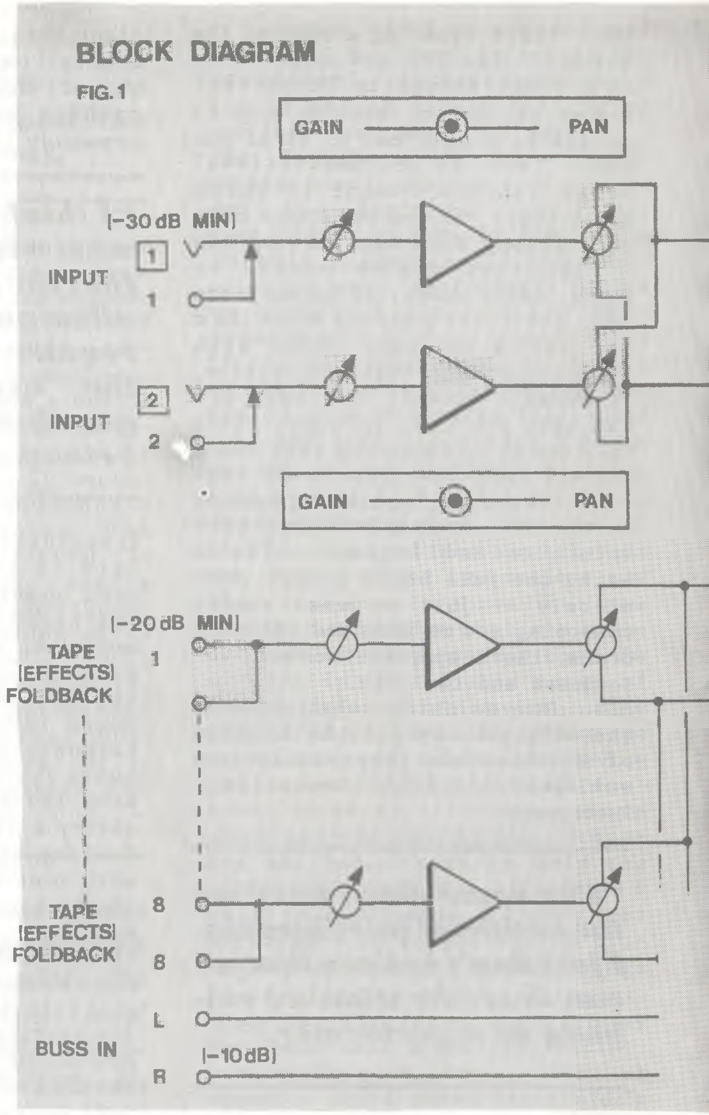
WC: It's obvious that the technology is moving in directions due to a lot of things like NASA -- specifically, more and more microcircuits that are allowing us to do things cheaply that were difficult to do earlier. So, instruments like the GDS will be on one chip and you'll be able to have eight GDSs on one little inexpensive machine. If I had my way, there will be people who will do it with wonderful keyboard-type controllers for people like me or guitar-like controllers for guitarists, or wind-type controllers in order to allow musicians who have spend many years learning their skills and art form to run this fancy new way of making sounds with nuance and subtlety.

* * *

This interview with Wendy Carlos is taken from TOTALLY WIRED: ARTISTS IN ELECTRONIC SOUND, a 26-part radio documentary examining the artistic development of electronic music through interviews and music of the artists. The series is currently running in most major markets on public radio stations through the fall and winter of '83 - '84. TOTALLY WIRED was produced by John Diliberto and Kimberly Haas. It was funded by Sequential Circuits, Inc., Yamaha Corp., the Pennsylvania Humanities Council, and the Pennsylvania Council for the Arts. For more information about TOTALLY WIRED, or to obtain cassettes of the programs, write to: TOTALLY WIRED, Box 5426, Philadelphia, PA 19143.

Fostex 2050 Review

by: Craig O'Donnell



When we installed our 16-track machine some months ago at Acme Recording (a 16/8/4-track professional recording studio in Chicago), we needed another 16x2 stereo mix available at the patch-bay. Our Neotek 18x8 console could only cope with 8 monitored tracks, and replacing it was not an option. What to do? Simple. We spent a minimum of cash (less than $335) on a pair of Fostex 2050 Line Mixers. These small mixers sit daintily atop the Neotek meter bridge. Now, as far as our main mixer knows, the two-channel monitor mix is a stereo tape deck; actually, of course, it is the 16-track machine. Line inputs enter the Fostex mixers and connect via Foldback jacks (parallel built-in "Y" connectors) to our console line inputs.

Foldback? FOSTEX? Sixteen track? What...!

No, we don't use a Fostex 1/2" 16-track machine -- we use the industry-standard Tascam 1" 85-16B with Autolocator. Still, the folks that made 1/4" 8-track and feather-weight four-track cassette recorders have come up with a versatile little gem. The 2050 Line Mixer typically sells for under $190.00 "on the street". It provides eight channels of line level mixing at the -10 dB semipro standard, and two Auxiliary inputs at -30 dB for synthesizer or pre-amped guitar. Each channel has a volume knob with concentric pan control. Looking at the 2050 from the front, there are two 1/4" inputs for the Aux channels, the two Aux volume/pan controls, eight channel volume/pan controls, a large knob called "REMIX", a phone jack with volume pot, and the phone-amp input select button.

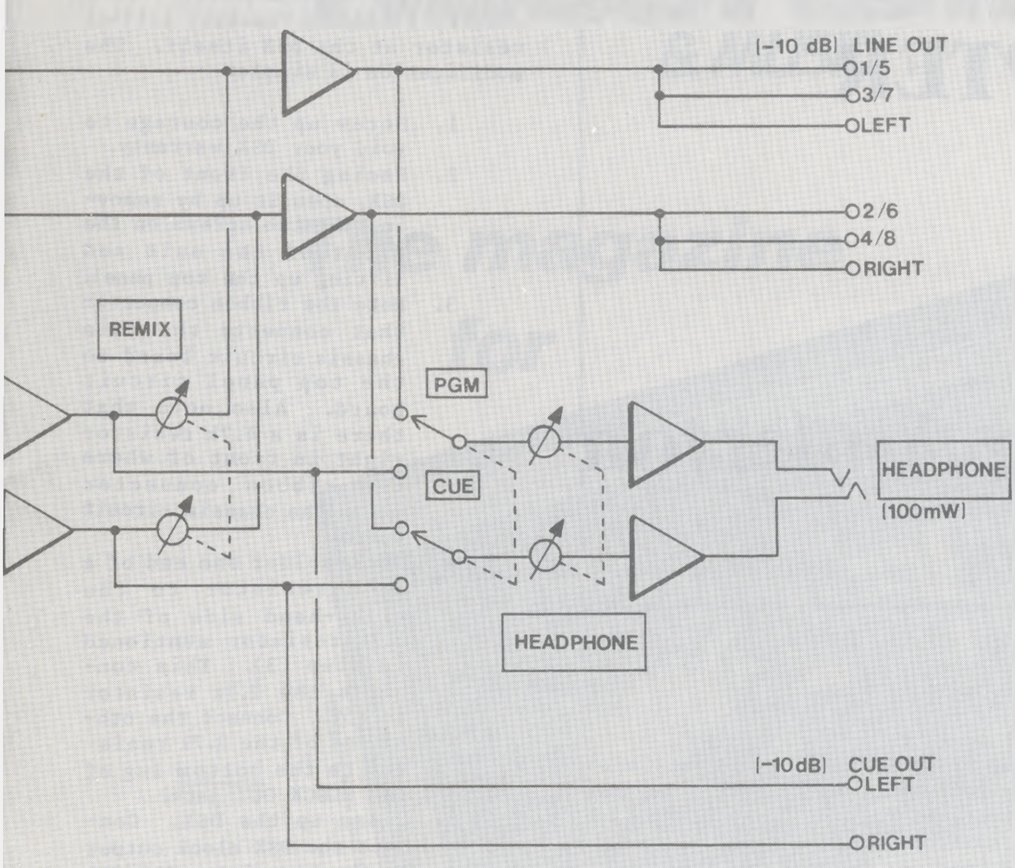
Flip it around and you'll see a phalanx of RCA connectors aligned in vertical pairs. At the left side, we have the Output section with Cue, Line "1/5", "2/6", "3/7", "4/8" and Buss In. If you're stacking for more than 8 channels, connect the Line Out on mixer "A" to the Buss In on mixer "B".

Next come eight vertical pairs of Line Inputs (the lower one is called "Foldback" and is really a parallel). At the extreme right, Aux In 1 lies above Aux In 2. When used, the front-panel phone jacks disconnect anything plugged into these rear inputs.

Fostex is wisely not competing with the Peaveys, Biamps, and Tascams that do nightly gigwork or session duties as small-studio mixers. There is no Gain set, EQ, LED display, or channel patching; but this unique box is four mixers in one. Use it, as Acme did, to supply another stereo submix; to submix effects or drumboxes and syndrums; or with PA to give an extra stereo effects send, recording buss, or monitor mixer. Plus it's a small-studio workhorse.

For overdubbing in the home studio, the Aux inputs can be fed directly into a tape deck while

✳ Features and/or specifications may change without notice.



inputs 1-8 are used for a cue mix. Or, the 2050 can be a track-bouncing mixer to take six tracks down to two on an eight-track machine. Obviously, it will bounce three tracks to one on a four-track machine. If you're on a tight budget, the 2050 can be your entire mixing console: The eight line inputs mix to stereo, with the REMIX pot acting as master fader. The foldback outputs can be your effects sends and the two Aux Ins your effects returns. Not bad for under two bills!

I've included the company's block diagram of the mixer (see **Figure 1**), which explains in no time how this little box pulls off all these various feats. You'll see that the Remix pot gradually adds the eight line inputs into the L/R outputs as it's advanced. If REMIX is shut down, the eight line ins are isolated from the two Aux Ins used to route material to whichever tracks you might have in record mode. On the phones, "PGM" gives all ten inputs while "CUE" accesses only the eight line inputs.

Fostex means for the "L/R" outputs to feed a mixdown deck; "Cue" out should hit your monitor amplifier for overdub mixes; and the "1/5", etc., outputs are for your multitrack tape recorder. Doubling outputs saves space and dollars.

Acme pressed a third 2050 into service as a reverb submixer to take ten effects inputs (a stereo chorus, three DDLS, and five reverb returns) into a stereo effects return on the Neotek console. A simple mod (see **Figure 2**) allowed us to add Mute toggles.

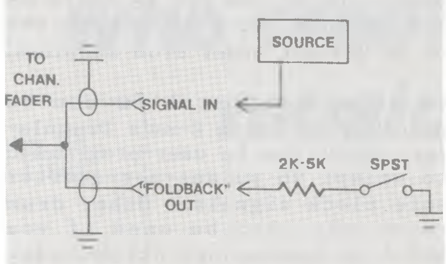
The 2050s easily match the semipro -10 dB level standard. Since there are no mic inputs, parts are at a minimum, and so is noise. Test-by-ear discerned no important difference in the sound of a tape track (snare) monitored through the 2050 and also through the Neotek. Of course there is slightly more noise, but not much. The outputs produce up to about +18 dBm without clipping -- more than enough for the applications this mixer sees.

Don't overlook the 2050 for live submixing into a bigger console. The soundman for our band, the "Scientific Americans", had two tape echoes, two DDLs, a flanger/doubler, a reverb, and a synthesizer jammed into every extra input on our Biamp 1621. A 2050 would have eased things a lot! In fact, a tinkerer can come up with a couple mic preamps for those situations where most recording is done direct at home -- or get an MXR Micro-Amp with a cannon-to-1/4" adapter to inject a mic signal into the Aux inputs.

The 2050 is only 14" wide, 6-1/2" deep, and one rack space (1-3/4") high. Compact. One quibble, though: why no rack ears? They're $20 extra...for that price, buy some panels from Polymart and have a metal-worker friend cobble a custom panel to suit. You can put a patchbay or other module in the remaining 5" of panel space.

The phone output cranks reasonably loud and distortion-free and drives a pair of phones without too much fuss (I use Yamahas or Koss Sound Partners). Two pair sound fine if levels are within reason. Conclusion? Highly recommended for the home studio, beginners, or those on a tight budget. In the case of the rugged and compact Fostex 2050, cheap cost does not mean cheap performance.

FIG. 2 "MUTE" MODIFICATION
Typical Channel - Fostex 2050



Switch grounds Input through 2K - 5K resistor, muting. Note : Chassis and PCB design makes it extremely difficult to install mutes any other way.

DSX-to-DRUMULATOR ADAPTER

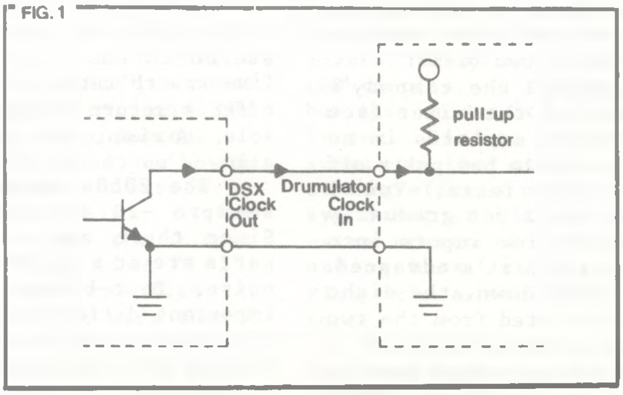
by: Craig Anderton

The Oberheim "System", one of the first (if not the first) integrated synthesizer/sequencer/drum machine setups, has been extremely popular since its inception. However, for those who want to use drum machines other than the Oberheim DMX or DX with the DSX, there are some problems which must be overcome. First, the "System" uses a high-resolution clock that runs at 96 pulses-per-quarter note (which is generally not compatible with the majority of other drum units), and second, the output structure of the DSX is designed specifically for the Oberheim DMX or DX and not other drum machines.

The first problem can be overcome by using E-mu's Drumulator, which can be user-programmed to accept 96 pulses-per-quarter note clock signals. Other drum units can also be used if you build an appropriate divider circuit (divide-by-two for drums which accept 48 pulses-per-quarter note clock signals, or a divide-by-four circuit for drums which accept 24 pulses-per-quarter note clock signals). The second problem can also be overcome, but requires a simple mod to the DSX.

The DSX clock output is basically an open-collector transistor (see **Figure** 1), with a pull-up resistor inside the DMX serving as a load for this transistor. This kind of structure is standard practice in the computer industry for optimizing a transmission line, and since the DSX and DMX/DX were specifically designed to work together, it makes sense to take this approach. However, if you try to use the DSX clock to drive some other device, unless the other unit includes a pull-up resistor to provide a load for the DSX clock output transistor you will not be able to interface the two.

FIG. 1



One solution is to add a pull-up resistor at the unit being driven, however, since I use the DSX to drive a variety of devices this would have meant installing several pull-up resistors in several pieces of equipment. As a result, I added a "master" pull-up resistor at the DSX itself. The modification is simple:

1. Screw up the courage to void your DSX warranty.
2. Facing the front of the DSX, open it up by removing the two screws on the front of the unit and lifting up the top panel.
3. Note the ribbon connector that connects the main chassis circuit board to the top panel circuit board. Also note that there is a 4.7k resistor right in front of where the ribbon connector meets the chassis circuit board.
4. Tack-solder one end of a 2.7k resistor to the right-hand side of the 4.7k resistor mentioned in step (3). This connects the 2.7k resistor to +5V. Connect the other end of the 2.7k resistor to the <u>bottom</u> lug of the CLOCK OUT jack.
5. Close up the DSX. Connect the DSX clock output to the Drumulator CLK/CAS IN, set the Drumulator external clock for divide by 4 (as described in the manual), and you should be ready to go.

There you have it: A simple way to get the DSX and Drumulator to "talk" to each other. And, I think you're going to like the way they converse.

MODERN RECORDING & MUSIC

The magazine for

Rindy and Marv Ross of Quarterflash. (Photo by Denny Anderson.)

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64 SOUNDS, Part 1

by: James A. Lisowski

Editor's note: James' last article, "Meet SID Sound Interface Device" (April 1983 Polyphony), was extremely popular. His latest, "64 Sounds" describes a number of useful programs you can enter into the Commodore-64 that take good advantage of the SID chip's capabilities. Part 1, presented this issue, covers introductory material which is a prerequisite for subsequent installments. Part 2, scheduled for the August issue, covers a Sound Test "software breadboard" (along with some sound effects) that will help you explore the SID chip while Part 3 (scheduled for October) presents a number of useful mini-programs (metronome, guitar tuner, hand clap/percussion unit, external audio filter, etc.). Have fun!

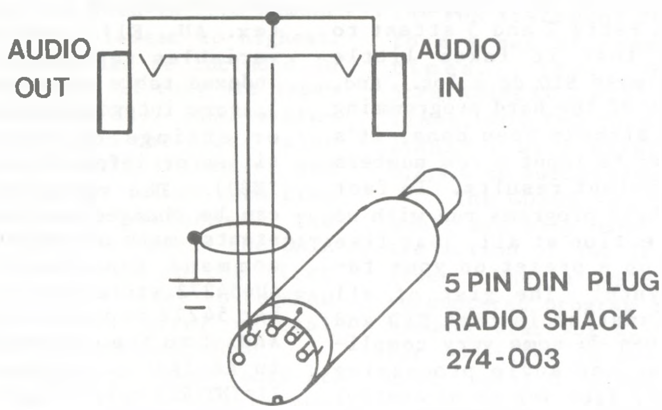
***

I mentioned in "Meet SID" that purchasing the Commodore 64 microcomputer (C-64 for short) was one of the easiest ways to obtain and control the SID (Sound Interface Device). The C-64 now sells for less the $200 in mass market outlets in America as well as Europe. "Sells" is too light a word -- because of its outstanding value in memory, 6502 microprocessor compatibility, multiple interface ports, breakthrough sound quality and graphics features, the C-64 has been deservedly topping the charts. Its snowballing popularity has also created some of the lowest cost microcomputer software and hardware yet, much of it from Commodore itself. Needless to say, all this has had a significant impact on electronic experimenters and musicians. SID is an amazing digitally controlled audio chip whose sound producing and processing capabilities make many less flexible synthesizers and music effect boxes blush in envy. The C-64's other ingredients make it an attractive low cost candidate for numerous audio profession uses: MIDI controllers (as implemented by Sequential Circuits with their C-64 compatible sequencer cartridge and by Waveform, the makers of MusiCalc), timebase generators, digital oscilloscopes, music tutorial systems, videodisk controllers and general business uses such as databases, accounting, word processing and electronic communication. AND don't forget about computer literacy! Owning and programming a C-64 helps develop or maintain an understanding of computer skills that have become survival tools of the Later 20th Century -- as well as providing a means for the music experimenter to do innovative and ground-breaking research without major funding -- all with the same investment! So, fire up your amp, flip the computer's power switch to ON, and start punching the keys as we explore and define the expanding frontiers of computer assisted sound.

This article is intended for an intermediate level of difficulty. It will help if the reader is familiar with the BASIC programming language as implemented on the C-64. Although I will try to make the program operations clear and point out details of special interest, this is not a BASIC tutorial. Even so, anyone should be able to type in these programs verbatim and get useful results. If you do not trust your typing or BASIC, fear not -- a preprogrammed disk is available, as mentioned at the end of this article. (Pricing and availability will also be mentioned at the end of Parts 2 and 3). On the other hand, if you are fluent with BASIC, don't stop here: Use these examples as groundwork for your own experiments. Either way, if you have problems or want to trade some good programs, feel free to write or call me and I'll do what I can to help.

Hardware. I'll start out with a few hardware and software pointers. Concerning additional hardware, I recommend buying or making a good shielded, 4 wire audio input/output/video output (5 pin DIN) cord so that you can use the direct audio output connector on the C-64. This allows you to route the sound directly into a high quality amplifier or tape deck (instead of getting the sound from the TV speaker), and also lets you route signals into SID's external input for processing external audio sources such as microphones, guitars and synthesizers. If a commercially available cord looks low grade or lacks the audio connection, check **Figure 1** and make your own. If you don't have a cord yet, the TV is still okay -- all of the programs listed here were prototyped on a TV set and sound great that way. (While recording, I often use both a closely mic'ed amp speaker AND a TV placed a few feet away in an open air microphone recording setup to get a good ambient stereo effect.) If you do have the cord, a generous amount of frequency equalization, reverb or digital delay can make otherwise flat or thin sounds quite a bit better. However, there are some pitfalls to obtaining higher quality sound. Computers are notorious high frequency (RF) noise emitters. Although models that meet FCC specs (like the C-64) are relatively quiet; good shields and grounds are a must so that your other audio effects do not pick up computer noise. Since the audio connector also has video and computer lines nearby, there is some low level noise present. External audio filters, companders or noise gates should be used for critical recording uses. The C-64 SID inputs and outputs are capacitor coupled, as recommended by the manufacturer, so most electronic

FIG. 1



effects can be connected to your computer audio plug without fear of damage. However, when in doubt, use optical isolators to keep voltage spikes or high level signals out of the computer. If your sound means more to you than your computer warranty, you might choose to modify the insides of your computer for lower noise, better shielding, improved frequency response, and/or direct or balanced audio connections.

The TV set itself also generates quite a bit of noise, especially in proximity to magnetic guitar pickups. Either turn off the TV during your performance or orient the guitar for the least amount of noise pickup. Note that turning the TV off won't alter your computer program, and you can program your controls to be presets, thus negating the need for visual feedback. You might also decide to design a special hardware control/indicator interface that uses LEDs or some other display technology instead of the TV. The C-64 has many I/O ports, and all it takes is the proper connector adapters and some programming to use your existing footswitches and pedals for controls. For example, if you plug paddles or a joystick into Game Port 1, the FIRE button acts just like a computer keyboard keypress -- great for a quick footswitch application.

Although I record my programs on magnetic disk, the programs presented here are short enough to be loaded from cassette tape without unduly trying your patience. For ease of transportation to a live performance, the Datasette tape storage device takes up less space than the disk drive and since it gets power from the computer, one less AC plug is needed. You might also consider placing your program into EPROM other nonvolatile (data retaining) electronic memory. A custom EPROM cartridge with on-board LED indicators, controls and jacks might be the best of all worlds -- just plug it into the 64's cartridge port and you are ready to play, with no bulky TV or disk drive to carry around. If you do take your system on the road and want the TV, I suggest using one of the miniature, 5" screen, battery operated portables and add wire mesh or transparent conductor screening to reduce EMI. Computer text and graphics are readable on those LCD based "pocket" televisions -- the ultimate in video portables. Also, you might consider using the Executive 64, Commodore's all-in-one C-64 portable that contains computer, monitor and disk drive, but costs more than a standard C-64 setup. Finally, I have one more strong suggestion: If you are going to experiment with music effects on the C-64, BUY THE GAME PADDLES! The keyboard and joystick are great for other input uses but they do not come close to the ease of turning a knob to get different input values for real-time control or experimentation. Even though paddles may not be a stock item at your local computer store, they cost less than $10 and can be ordered from the store or by mail. Most of my programs use paddles. It is not difficult to alter the programs to use other input forms but I feel that paddles give simple, fast, and flexible control operation.

SID in short. Let's take a brief look at SID to see what it will do. If you have the "Meet SID" article at hand, you might review it for a more in-depth look. In any case, for any serious work with SID, the Commodore 64 Programmer's Reference Guide is essential for the examples, charts, (including the entire C-64 computer schematic), and specs that it contains. This 400+ page book is available from your local computer store or Howard W. Sams Books.

SID is a digitally controlled single LSI sound production and modification chip that requires few external components for operation. SID has 3 independent voices, each consisting of an oscillator and an Attack/Decay/Sustain/Release (2ms to 24 second exponential ADSR) envelope generator. Each oscillator can generate Sawtooth, Triangle, Pseudo-Random Digital Noise and Pulse (Square wave with variable duty cycle) waveforms over an 8 octave (0 to 4 kHz) range. It is possible to use any two oscillators to create Hard Sync and Ring Modulation effects. Each oscillator's waveform may be reset to a known initial state for the synthesis of complex real-time waveforms. Any or all voices may be filtered by an internal variable frequency (30 Hz to 12 kHz, 12 dB/octave), variable resonance audio filter. The filter can assume high pass, low pass, band pass, or combinations (such as notch) modes. Two slow Analog to Digital (A/D) converters can read the value of external potentiometer controls (i.e. game paddles) and these values can be used by the computer to set the values of any SID control or for other program use. Oscillator 3 has some special features: The instantaneous value of its current waveform and envelope can be read and used by the computer (for example, for LFO modulation of an oscillator), and its output can be excluded from the audio output yet retained as a control waveform. One very significant feature is the presence of an External Input whereby external signals (from guitars, other SID chips, etc.) can be filtered alone or in conjunction with the other SID voices. A programmable Master Output Level control can set the combined external and internal output levels from full on to full off in real-time, computer-controlled steps. The paddle and audio inputs and the SID output (as well as RF modulated sound output for TV reception) are available at external connectors in the Commodore 64 computer.

The computer treats SID as a series of 29 memory locations (registers). Control information is written to WRITE-ONLY registers. Unlike ordinary computer RAM (read/write program and information storage Memory) or ROM (Read Only Memory, for fixed program storage), the information can not be re-read from these registers. If you want to use the value of a certain SID control in a later calculation, (for instance, increase the frequency of an oscillator from its present state), you must write the value to the SID register and also store it as a BASIC variable (or other place in RAM memory). The number of different digital information states available to operate each SID control over its entire range varies from 1 bit (Binary digit, 1 or 0, on or off) for selection of the type of waveform to 16 bits for selection of the oscillator's frequency. All the registers are 8 bits in length. Thus, to sweep an oscillator's frequency through its entire range two registers must be written to (one 8 bit coarse control, along with one 8 bit fine control, to cover the entire 16 bit range). Because of the very wide 16 bit range, the oscillators may be swept in extremely fine steps -- so fine that even though the change is in discrete steps (quantization), the change is perceived as smooth and linear by the human ear. SID registers that provide information (Oscillator 3 waveform, envelope, and paddles) are all 8 bits wide (256 different binary values) and are READ-ONLY. Writing to them will not alter their values. When they are read, the computer can save their values in RAM memory or as BASIC variables for later use, or write the value into one or more of SID's WRITE-ONLY control registers to change a control without external storage.

It may seem that all of this is very complicated and that making SID do tricks would be a major undertaking, but actually one rarely uses all of the controls and options at once. In fact, the short length of the programs presented in Parts 2 and 3 attest to the fact that it takes little effort to make SID do a lot. And, since most of the hard programming work has already been done, it's even easier to input a few numbers and get instant results. In fact some of these programs run with no input selection at all, just like punching up a preset on your favorite synth! The gist of all this explanation is that SID and the C-64 can do some very complicated computer audio processing with a very fine degree of control and repeatability, whether you're looking to create a simple metronome or a multi-voice sequencer. And you can either use programs others have written, without any programming on your part, or do some programming and customize an existing application for your own special needs -- all without buying a new instrument or picking up a soldering iron.

Software. For those who are not familiar with programming or the Commodore dialect, we'll discuss some of the essential elements of BASIC. Please consult your manual or other tutorial book for greater detail. While most of this will make sense if you are even slightly familiar with BASIC, it helps to refer to existing programs (such as INIT, published towards the end of this article) so that you can see how these statements work in context. If all this seems hopelessly complicated I suggest putting your brain "on hold" until Parts 2 and 3 are published, at which point there will be more programs to which you may refer.

BASIC is an easy to use computer programming language. BASIC itself is a program. BASIC program statements all start with line numbers (ex. 100) and contain information (data) or instructions for doing some calculation or process with the data. The line numbers act as reference points so that the program operation can jump from one line in the program to another (ex. GOTO 100 tells the program to jump to line 100 and do whatever that line instructs). Unless told otherwise, program execution starts at line zero and continues to the next higher numbered line. Multiple statements can be place on a single line if they are separated by a colon (:). Floating point program variables store numbers (including decimal point and fractional portion) and are identified by two letters, or one letter and one number, names (ex. AU, F1). Other types of variables represent items in an indexed table of values (ex. WV(3)), store integer numbers (ex. FF%), or strings of characters for titles or information sorting (ex. IN$)). The value of a variable can be changed via numerical constants, math operations or special command functions. Examples: AU=54272 stores the constant number 54272 in variable AU, AA=AA+1 adds 1 to the current value saved in AA (AA is incremented by 1), II=INT(RL) calculates the integer portion of the number saved in variable RL via the INT() function and then stores the result in variable II. DO NOT use the variable names ST and TI for your own number storage; these are special variable names used by Commodore BASIC to reflect the status of input/output operations and a running clock timer, respectively.

Most programs use just a handful of command statements to perform the bulk of the work. Here are the most used BASIC commands:

REM statements serve as comments (REMarks) for note taking or special instructions that will not be printed or executed.

PRINT statements print a message or the current value of a program variable on the TV screen (or other device).

INPUT statements print a question mark (?) on the TV screen, then wait for data to be entered from the computer keyboard (or other device). This data is then stored into a variable for later use. INPUT statements always wait for you to indicate that you have finished typing by ending your data with a press of the RETURN key. Program operation will not continue until this has been done. The last number or letter of your INPUT data can be erased (deleted) with a press of the DEL key. Optionally, INPUT statements can also act as both a PRINT and INPUT to prompt the program user with a message that describes the type of information to be entered.

cont. →

GET statements are similar to INPUT statements except that they do not print a message or question mark, and take only 1 character without waiting for RETURN.

READ statements are similar to INPUT statements except that they get the information from a fixed list of values that are found in DATA statements. DATA statements are read in line number sequence, from lowest to highest. Several values can be included in a DATA statement if they are separated by commas (,). READ statements start with the first value in the first DATA statement and select subsequent values each time another READ is performed. The RESTORE statement resets the DATA to the first item in the first DATA statement in order to re-READ the same values after several READs have already been done.

The PEEK() statement looks at a specified place (address) and returns the value it finds as an integer number. PEEK is often used to get a value from a READ-ONLY SID register.

The POKE statement takes the integer portion of a specified number and stores the result into a specified memory location. POKE is often used to write information to a SID WRITE-ONLY register.

The GOTO statement indicates that program execution should not continue with the next line but should instead GOTO the line specified in the GOTO statement.

IF and THEN statements allow the program to do something only IF a specified logical condition or equation is evaluated to be true. If the condition is not true, the program continues with the next line. If the condition is true, the statement following the THEN command is executed and the program continues with the next line. If a line number follows the THEN statement and the condition is true, the IF acts as a GOTO.

The FOR, TO, STEP and NEXT statements all work together as a counting and repeating structure. For instance, the statement FOR I=0 TO 5 will first set variable I to zero and then continue on with the rest of the following program lines, until the NEXT I statement is reached. When this happens, execution goes back to the FOR statement, the variable I is incremented by one (I would go from 0 to 1, in the example) and the result would be compared to the value 5. If I is less than or equal to 5, the statements after FOR will be executed, until NEXT is encountered and the process is repeated with I counting upward until the value of I is greater than 5. When this happens program execution continues with the statement after NEXT. If the STEP command is included on the FOR line, the number following step is used instead of 1 to increment I. The FOR statement is used to perform a set of commands several times. (Ex. FOR I=1 TO 5:PRINT:NEXT I will PRINT 5 blank lines, FOR I=0 TO 6 STEP 2:PRINT I:NEXT I will PRINT the numbers 0,2,4,6 on successive lines.)

The GOSUB and RETURN statements are also used for repeating the action of one or more program lines. When a GOSUB statement is encountered program execution continues, starting at the line number specified in the GOSUB statement, until a RETURN statement is encountered. Execution then continues with the statement following the GOSUB. This allows long, often used SUBroutines to be included in IF THEN statements or other places. All the programs presented in this series (and many other BASIC programs) use only these few statements in various combinations to achieve a spectrum of different applications.

Programming tricks. At this point I have been programming the C-64 for over a year and thus have a number of techniques and programs that can save the sound experimenter much time and trouble. The first important thing a sound program should do is set all the SID registers to a known, zero state. Since a previously run program may have (unknown to you) altered a register's value or turned on a filter, your program may be working just fine but you won't hear the results because of the way the other registers are set. The next step is to set the Master Volume Register to full ON. (Usually this means sending it a value of 15, but since this register serves several functions, the number sent may be greater than 15.) When the volume goes from 0 to 15, the TV speaker will emit a pop or click and this should serve as a check -- if you don't hear it, something is wrong. Next, send SID all the frequency and ADSR values for the sound you want **except** for the WAVEFORM values. Be careful, as many of the SID registers have multiple functions and all functions for that register have to be set at the same time. For instance, the ATTACK and DECAY values occupy different parts of the same register. If you want to set the ATTACK value you must also include the DECAY value (add the two values then set the register with the sum). If you are not familiar with this operation look at Table 1 to help select the desired values.

The next important thing to remember (and one that gave me a lot of trouble until I learned it) is that the voices have to be turned OFF in order to turn them ON. The WAVEFORM and ENVELOPE GATE (or trigger), as well as other controls of a voice, all reside in the same register and all of these parameters must be taken into account. When a voice is first triggered it goes through its AD envelope cycle, and then remains at a very low audio volume until turned OFF when the WAVEFORM REGISTER is reset to zero. Note that if you want a Sustain/Release cycle DO NOT turn the voice off with zero; instead, send the same value you used to trigger the sound minus 1 (this leaves the waveform intact but resets the GATE bit). Also be sure to have some noticeable time delay period so that the Sustain will last for a certain period of time.) If a voice works as expected the first time you play it, but then doesn't seem to work or sounds very faint on the second and subsequent times, you are probably not turning the voice OFF in-between. One way to assure that the voice will work as expected, no matter what the rest of your program does to it, is to send the WAVEFORM/GATE register a zero and then send it the desired wave value.

Words of Caution: since the POKEs used to change SID registers can also alter the RAM memory your program resides in or disrupt BASIC if misdirected, always be sure that you POKE the right locations and that you enter the example programs correctly. The best procedure is to type the program in and double check it against the LISTing. Besides checking the POKEs, make sure that the variable names have been spelled correctly and that a zero is not mistaken for the letter "O" (and vice versa), and then SAVE A COPY of the program on tape or disk BEFORE you RUN the program. That way, if the program IS accidentally altered, you just have to re-load, not retype it. And, as you experiment, remember to save occasional backup copies, as one typing mistake

cont.

could ruin hours of work in a flash. One way to avoid mistakes is to always use the same variable name to do the same thing in all of your programs. Consistent variable use will lessen POKE errors and make the program action clearer.

Program #1: INIT. I use a modular approach to my BASIC programming. The sound module INIT SOUND S200, hereafter referred to as INIT, sets a consistent, mnemonically selected group of variables to the most used sound and utility memory locations. Next, it clears all of the SID registers to a zero value and then reloads the SID voices to the values found in the INIT DATA statements. Variable AU contains the value 54272 which corresponds to the memory location of the first register of the SID AUdio chip. F1, F2 and F3 reference the HI (coarse) FREQUENCY registers of SID VOICES 1, 2, and 3 respectively. Likewise, T1, T2, and T3 point to the TRIGGER GATE/WAVEFORM registers of the three voices. P1 and P3 are the SID A/D converter locations that indicate the values of game paddles when the paddles are plugged into Game Port 1. B1 shows the status of the paddle FIRE buttons. BD is set to the location of the register in the VIC II video chip that controls the color of the border of the video screen. The next VIC II location (BD+1), controls the background video color. The rest of the variables in INIT only serve temporary uses and can be reused later in the program.

Follow the program LISTing of INIT in **Figure** 2 to see, line by line, how this entire routine INITializes the SID chip and sets the sound registers to the DATA values. LINES 200 through 212 set up the most used variables. Remember, these variables are set up to contain the ADDRESS locations of the registers, not the CONTENTS of the registers. Line 215 uses a FOR NEXT loop (repeating structure) to POKE all of the addresses starting from 54272 (the value of AU) up to and including 54300 (the last SID location) with a value of zero. These addresses are where the SID registers are located in the computer's memory and thus, SID is reset to all zero values. Also on Line 215, the SID Master Volume control register (located at AU+23, or 54272+24=54296) is POKED to full volume (a value of 15). In line 220, INIT READs a value from the following DATA statements into a temporary variable named AV. IF the value of AB is greater < than minus 1 (AV≥-1). Then that value will be POKEd into the memory location to which variable AA points. AA had been set to 54272 (the first SID register location) back in Line 200, so the first DATA value is sent to the first SID register. Continuing with the rest of the THEN statement, AA is incremented by one (AA=AA+1) so that next time around, the next SID register will get the next DATA value and finally, the GOTO 220 starts the process over again from the start of Line 220. This process continues to READ and POKE DATA values into sequential SID registers until the -1 DATA value is READ, at which point the IF condition is NOT true

TABLE 1

Table of Values for Dual Position Registers

ATTACK/DECAY	SUSTAIN/RELEASE	
0	0	Minimum
16	1	
32	2	
48	3	
64	4	
80	5	
96	6	
112	7	
128	8	
144	9	
160	10	
176	11	
192	12	
208	13	
224	14	
240	15	Maximum

Examples:

0	= Minimum ATTACK / DECAY
5	= Minimum ATTACK / Medium DECAY
208	= Long ATTACK / Minimum DECAY
85	= (80+5=85) Medium ATTACK / Medium DECAY

Table of Special Characters

Name	Code	Type this
Screen Clear	{SC}	Hold SHIFT, press CLR HOME
Gray	{G1}	Hold Commodore Logo key, press 4
Cursor Down	{CD}	Press CRSR Down Arrow key
Home Cursor	{HM}	Press CLR HOME key

```
200 AU=54272:F1=AU+1:F2=AU+8:F3=AU+15:REM INIT SOUND S200
210 T1=54276:T2=54283:T3=54290:AA=AU
212 P1=54297:P3=P1+1:B1=56321:BD=53280
215 FORII=AUTO54300:POKEII,0:NEXTII:POKEAU+24,15:REM ZERO
REGISTERS, FULL VOL
220 READAV:IFAV>-1THENPOKEAA,AV:AA=AA+1:GOTO220
229 REM  FL  FH  PL  PH  W   AD  SR
230 DATA 000,000,000,000,000,000,000
235 DATA 000,000,000,000,000,000,000
240 DATA 000,000,000,000,000,000,000,-1

READY.
```

FIG.2

```
0 REM SOUND TEST V3 (C)1983 JAL SOFTWARE
200 AU=54272:F1=AU+1:F2=AU+8:F3=AU+15
210 T1=54276:T2=54283:T3=54290:AA=AU
212 P1=54297:P3=P1+1:B1=56321:BD=53280
215 FORII=AUTO54300:POKEII,0:NEXTII:POKEAU+24,15
220 READAV:IFAV>-1THENPOKEAA,AV:AA=AA+1:GOTO220
229 REM  FL   FH   PL   PH   W    AD   SR
230 DATA 000,000,000,000,000,000,000
235 DATA 000,000,000,000,000,000,000
240 DATA 000,000,000,000,000,000,000,-1
300 POKEBD+1,15:POKEBD,15
310 DIM WV(3),WS(3)
320 PRINT"{SC}{G1}SOUND TEST V3 JAL SOFTWARE"
330 PRINT:PRINT"SET UP SOUND DATA THEN"
340 PRINT"PRESS SPACEBAR TO TRIGGER THE SOUND"
350 PRINT"OR":PRINT"PRESS RETURN TO END"
360 PRINT"THEN USE SCREEN EDITING TO MAKE NEW"
370 PRINT"CHANGES AND RE-RUN"
380 RESTORE:FORI=1TO3
390 FORII=1TO7:READV:IFII=5THENWV(I)=V
395 IFII=7THENWS(I)=WV(I)+(V>0)
400 NEXT II,I
410 SF=(WS(1)+WS(2)+WS(3)<>0)
500 GETIN$:IFIN$=""THEN500
510 IFIN$=CHR$(13)THENPRINT"{SC}{CD}{CD}{CD}{CD}{CD}{CD}
{CD}RUN":PRINT"{HM}";:LIST229-240
515 POKET1,0:POKET2,0:POKET3,0
520 POKET1,WV(1):POKET2,WV(2):POKET3,WV(3)
525 IFNOT(SF)THENFORD=1TO500:NEXTD
530 POKET1,WS(1):POKET2,WS(2):POKET3,WS(3)
540 GOTO500

READY.
```

FIG. 2 [cont.]

(AV=-1 therefore AV is equal to the comparison value -1, not greater than it, as were all of the previous zero or positive numbers) and program execution continues with the next line (without doing the statements following THEN). Line 229 is a REMark that provides SID register name labels for the data value positions on the following DATA lines: FL (oscillator Frequency L1 value), FH (oscillator Frequency Hi value), PL (Pulse width Lo value), PH (Pulse width Hi value), W (Waveform/trigger gate), AD (envelope Attach and Decay values) and SR (envelope Sustain and Release values). The DATA statements in Lines 230 to 240 have their data items ordered to coincide with the registers that control the three SID voices with VOICE 1 data on line 230, VOICE 2 data on Line 235 and VOICE 3 data (plus the end of data indicator, -1), in Line 240. Thus, for our experiments, all we have to do is change the values in the DATA lines and the SID voices will be setup accordingly when INIT is RUN. As such, INIT provides a basic "breadboard" or workshop table on which to base audio experiments. So, the first thing to do is: type in INIT, double check it and save a copy on tape or disk. Then, when it is needed, load INIT back in and add the rest of the program lines that are required to complete that particular program, and finally, check and save the finished program -- **before** you try to RUN it. One note: Normally the W (Waveform) DATA values should be left as zeroes so that the voice will initially be turned off. The SOUND TEST program presented next issue is an exception to this.

Well, we're out of space for this issue; see you next time with more information on programming the C-64. If you want to get a head start by ordering a postpaid diskette containing all 15 programs used in this series, send $20.00 in money order or cashier's check (Wisconsin residents, please add $1 for state tax) to: JAL Software, Box 128, S. Milwaukee, WI 53172. An audio demo tape of the various sounds is also available for $5.00 (Wisconsin residents add $0.25 tax) in case you want to hear the sounds by themselves.

CURRENT EVENTS

'Tell Them You Saw It In Polyphony'

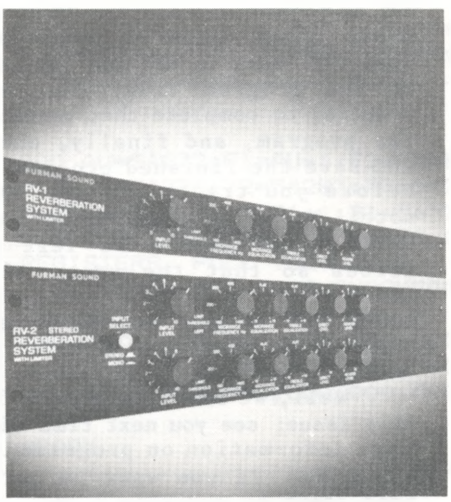
Graphic EQ on a chip. National Semiconductor (PO Box 70818, Sunnyvale, CA 94086) continues to develop interesting linear ICs. Their latest, the linear CMOS LMC835, is a digitally controlled equalizer that packs 14 frequency bands of 24 steps each on a single chip. It also delivers all the logic necessary for serial data control from a 3-wire microprocessor interface. With respect to external components, the LMC835 requires only simple op amp buffers and active inductors to make a complete, digitally controlled, stereo 7-band graphic equalizer. National claims total harmonic distortion of 0.0015% (measured at 1 kHz), and typical signal-to-noise ratio of 114 dB (controls flat). Typical step error is 0.1%. The LMC835 also contains the equivalent of 43 CD4066s to implement the microprocessor interface, and allows for a boost/cut range of either ±12 or ±6 dB.

National has also announced the LF13006 and LF13007 CMOS digital gain sets. These precision (maximum gain error of 0.5%) switching networks are designed for 3-wire microprocessor control of op amp gain. The LF13006 sets binary gains of 1, 2, 4, 8, 16, 32, 64, and 128, while the LF13007 sets gains of 1, 2, 5, 10, 20, 50, and 100. Both versions include two uncommitted matched resistors for customizing the gain and operate from +5 to ±18V DC.

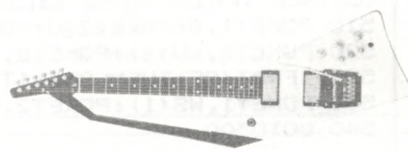
Noise reduction on a chip. Solid State Micro Technology for Music, Inc. (2076B Walsh Avenue, Santa Clara, CA 95050) has introduced two exciting new chips. The SSM2200 integrates the popular Dynafextm noise reduction system, developed by MICMIX Audio Products, on to a single chip. This single-ended system provides up to 30 dB of noise reduction and does not require any kind of encoding/-decoding process. The claimed dynamic range is 110 dB with a THD of 0.04%. The SSM2200 requires few outboard components: two op amps (for input/output conditioning), and a couple dozen other components. Target price is $6 in thousands. Although the SSM2200 is not slated to be available to hobbyists in small quantities, rumors are that RODCAR Electronic Sales (9983 Monroe Drive, Dallas, TX 75220) will be offering a reasonably priced noise reduction kit based on the SSM2200.

The SSM2014, another new SSM chip, is an operational voltage controlled element. This generalized VCA building block will substitute for any VCA circuit presently available, and may be configured as a voltage-in or current-in VCA or VCP (voltage controlled potentiometer). Class A or class AB operation may be selected at the user's option.

Finally, SSM is now a second source for the popular 5534A low-noise op amp.

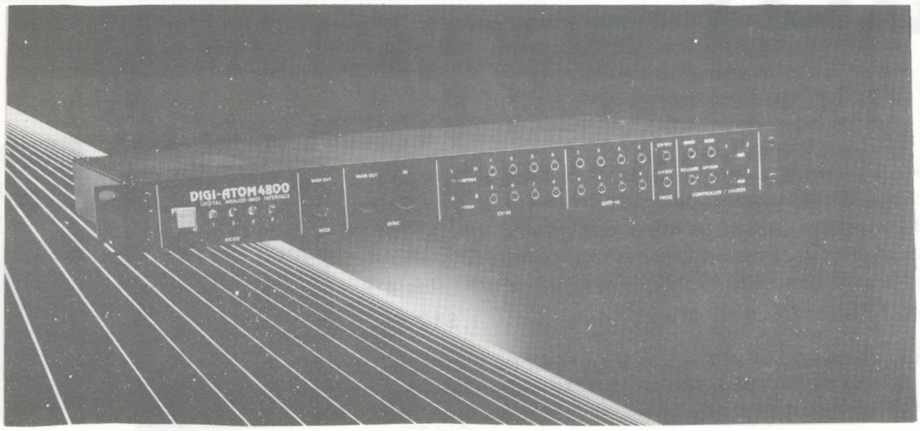


New reverb. Furman has announced the RV-2 stereo reverb, which like the RV-1 includes limiting and EQ facilities. A unique stereo-mono switch generates stereo reverb from a mono source in the "stereo" position, while in the "mono" position, the two channels can be used independently or patched in series for extremely dense mono reverb.



Roland guitar synth. The MIDI-compatible GR-700 features 12 DCOs (2 per string, thus allowing for sync), 6 VCFs, VCAs, and envelope generators, plus LFO modulation and stereo chorus. Onboard memory stores 64 patches, expandable to 128 patches; the optional PG-200 programmer is required for synthesizing sounds from scratch. Roland has also introduced the matching G-707 guitar, which features a graphite support bar. This is claimed to lower the guitar's resonance point below the frequency of its lowest note, resulting in no dead spots or unwanted resonances. The GR-700 lists for $1,995 and the G-707 for $1,150.

Low-cost stereo imager. The Radio Shack "Video Sound Processor" (stock #15-1277), while intended for enhancing the audio quality of video devices, is also excellent for stereo imaging and ambience synthesis. For a retail price of under $80, the VSP includes a stereo synthesizer, delay line with short variable delay, and an LM1894-based single-ended noise reduction system.

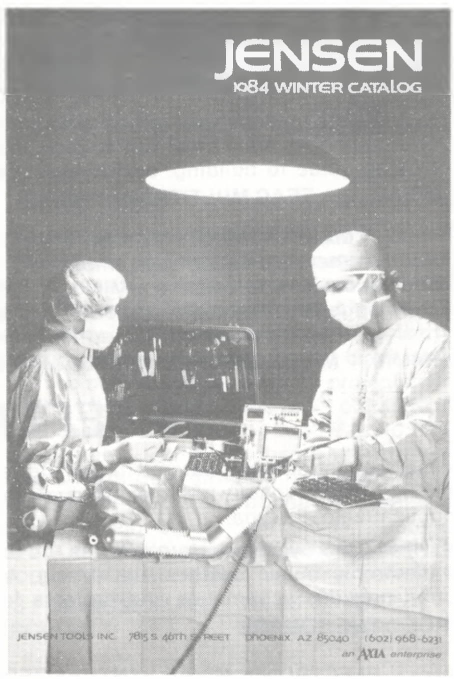


Analog to MIDI. Yes, you can talk to MIDI with an analog synth via the Digi-Atom 800 from Talk Studio (1-26-3 Zenpukuji, Suginami-ku, Tokyo, Japan 167). This device converts up to eight control voltages and gates into MIDI key data. Additional inputs include velocity control, pitch bend, modulation amount, release, and sound program select; there are also limited synchronization capabilities.

Clever bass amp. Normally Current Events doesn't find bass amps all that interesting, but this one is too hip to ignore. The AMP XB-15 is a "box within a box" (see picture) that packs an upper cabinet and lower cabinet into an easily transportable box (shown at right of picture). When set-up (at left of picture), the

upper cabinet is open at the bottom and sits on top of the ported lower cabinet, which is open at the top. The two enclosures automatically seal and the connected air space provides an internal volume of 7.7 cubic feet. Claimed frequency response is 40 Hz to 5 kHz. AMP (9829 Independence Ave., Chatsworth, CA 91311).

Precision tools. Jensen Tools (7815 S. 46th St., Phoenix, AZ 85040) has released their new, 80 page free catalog of more than 1,000 tool of interest to engineers and technicians. And the cover is excellent!



Phoenix Systems address change. Phoenix Systems, Inc. is now located at 71 Old Farm Road, Tolland, CT 06084 (tel. 203/643-4484).

Network feeder. Opamp Labs Inc. (1033 Sycamore Ave., Los Angeles, CA 90038) has introduced the Model VA-16 1-in/16-out Video/Audio Distribution System. This device is useful as a network feed for up to 16 monitors and audio amplifiers and lists for $600.

New delay. ADA (2316 Fourth St., Berkeley, CA 94710) has introduced the 2FX digital multi-effects box, which allows the flanger/chorus section to be used simultaneously with the delay section. Other features include over 1 second of delay at 17 kHz bandwidth, 10:1 flanger sweep, and optional footswitch controller for calling up presets. The 2FX lists for $599.95, the footswitch for $99.00.

POLYMART BOOKS

NEW BOOKS!

NEW

GUITAR ELECTRONICS FOR MUSICIANS by Donald Brosnac is a comprehensive guide for anyone interested in electric guitars. It clearly explains guitar electronics step-by-step with over 350 photos, drawings and schematics. Chapters include: types of pickups, design and function of hardware components, servicing electric guitar circuits, hot rodding electric guitars and more. Anyone who wants to increase his knowledge of guitar construction and function will benefit from reading this book. **# GEM..... Guitar Electronics for Musicians............$12.95**

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Synthesists must be well versed in a number of techniques and principles. "How To" and project oriented books are a great way to pick up these skills. **MULTITRACK PRIMER** by TEAC is a step-by-step guide to building, outfitting and operating your home studio.
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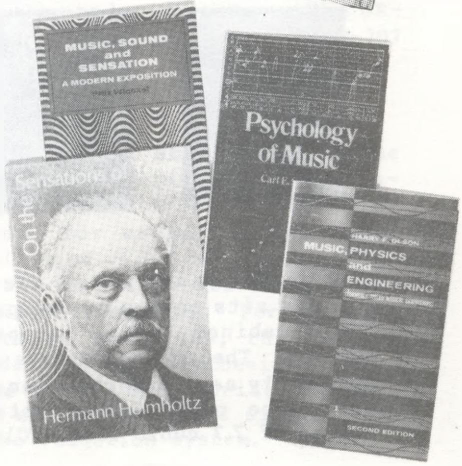

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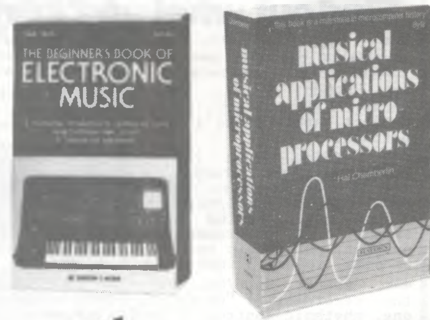

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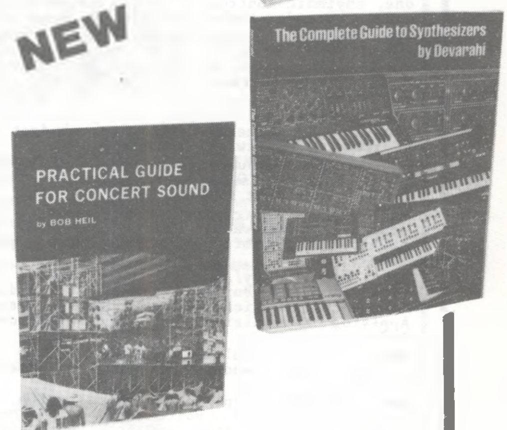

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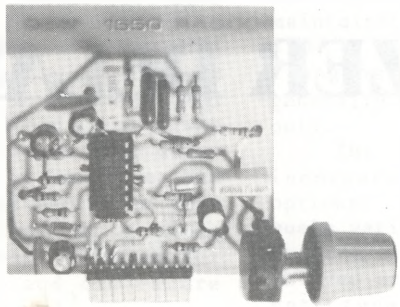

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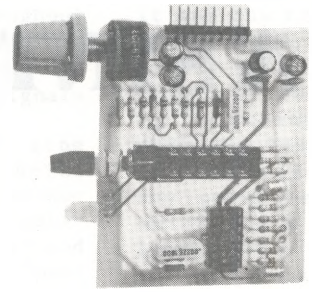

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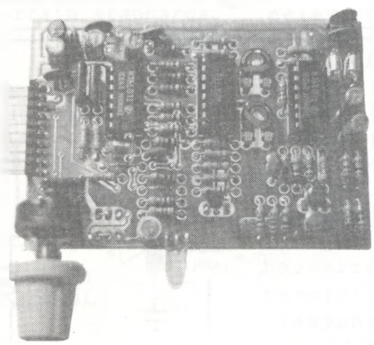

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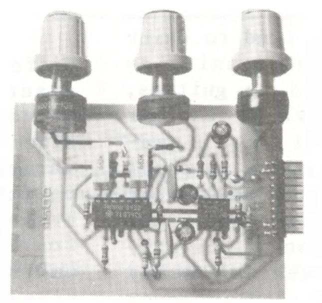

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Practical Circuitry

BUILD A SYNTHESIZER DELAY LINE

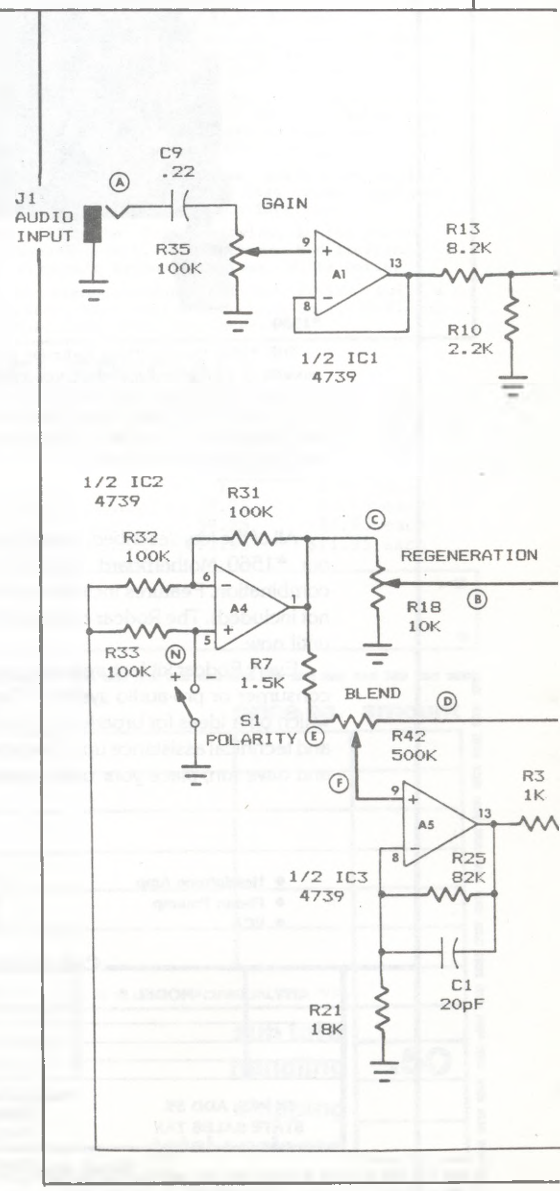
by: Thomas Henry

A lot of people tend to think of flangers and chorus units as being strictly suited to guitars, and not of any real value for synthesizers. After all, a synthesizer has many more control options available and scores of VCFs and VCAs to alter the sound; what more could a flanger possibly add to a synthesizer's already large repertoire of tonal colors? Well, lots! A delay line capable of flanging and chorusing opens up a whole new world of sonic capabilities. Considered as a "timbre modulator", a delay line can transform traditional VCO waveforms into a spectacular spectrum of harmonics that would be impossible to achieve with standard methods. I used to be skeptical about combining flangers and synthesizers, but changed my mind when I heard dozens of new sounds coming forth from this simple homebrew unit.

A delay line for synthesizers. This issue's "Practical Circuitry" describes a delay line which has been especially designed to work with synthesizers. While not offering long enough delays for echo and reverberation, it is suitable for a whole range of flanging, chorusing and vibrato effects. Best of all, it's easy to build and align, and uses commonly available parts to boot. If you've been thinking about getting into analog delay projects, this might be a good place to start since the design is straightforward and relatively goof-proof.

There are basically three differences between this synthesizer delay line and the typical guitar flanger. First, a synthesizer delay line must allow for a variety of generalized control options (as opposed to the single triangle wave generator control found in most guitar-oriented units). Secondly, the assumed signal levels are considerably different (synthesizer oscillators put out far stronger signals than guitar pickups). Finally, noise considerations (all analog delay line chips suffer from some noise problems) may be relaxed somewhat. This is because with a guitar unit, the output always connects to the amplifier. Any noise is always present, so most circuits use compansion or noise gating to reduce noise. With a synthesizer, however, matters are simpler since the last module in the signal chain (before going to the amplifier) is usually a VCA. When no keys are held down, the VCA is cut off (closed) and this gates out the noise. Pressing a key down opens the VCA, but remember that the signal going through the VCA is much greater in amplitude than the background noise. Therefore, in either case background noise is simply not apparent.

Now I'm not going to claim that this delay line is the quietest unit around, but if (as described above) it's inserted into the signal chain before the final VCA in a patch, then it will perform quite admirably and noise will not be a problem. There are



always tradeoffs involved in electronic design, and the factors I tried to balance here were noise and cost. By assuming that the delay line would be followed by a VCA, I was able to trim the cost and complexity of the circuit by a large factor. The result is a delay line which is more than adequate for most synthesis and doesn't cost an arm and a leg. One of the goals I had in mind was making banks of two to four delay lines economically feasible, and the present design satisfies this goal. Using multiple delay lines opens up many possibilities for Shepard function and quadrature function generators.[1,2]

Delay line features. Let's quickly sketch out the salient features of the synthesizer delay line. As with previous projects presented in this column, the flanger follows certain standards. The audio signal is presumed to be 10V peak-to-peak, and the control signal 0V to +5V. The input impedances are 100k and the output impedance is 1k. One standard that isn't obeyed is DC coupling; the extra complexity involved in maintaining DC response throughout didn't seem worthwhile since flanging and chorus effects are generally applied to audio signals only.

The synthesizer delay line incorporates a number of useful options. For example, a continuously variable blend control tailors the mix between straight and delayed sounds. I like this "panpot" type of control much more than the individual level mixer controls found on most delay lines; a fringe benefit is that you don't need a separate vibrato mode switch (for vibrato, simply turn the blend control to the full delay position). Another option is the polarity switch. In the positive position, the delayed signal is added to the dry signal; in the negative position, it's subtracted. The two sounds are quite different, with the positive having a more metallic scrape to it and the negative generating a smooth bassy sound. Best of all, unlike many other designs that require a DPDT switch, the polarity switching requires only an ordinary SPST switch. Finally, two LED indicators keep a tab on any possible signal conditions which could overload the delay line.

All in all, while this is a simple delay line circuit, it nonetheless incorporates the kind of features that make using it a breeze. Now let's take a look at the schematic and see how it works.

How it works. Refer to

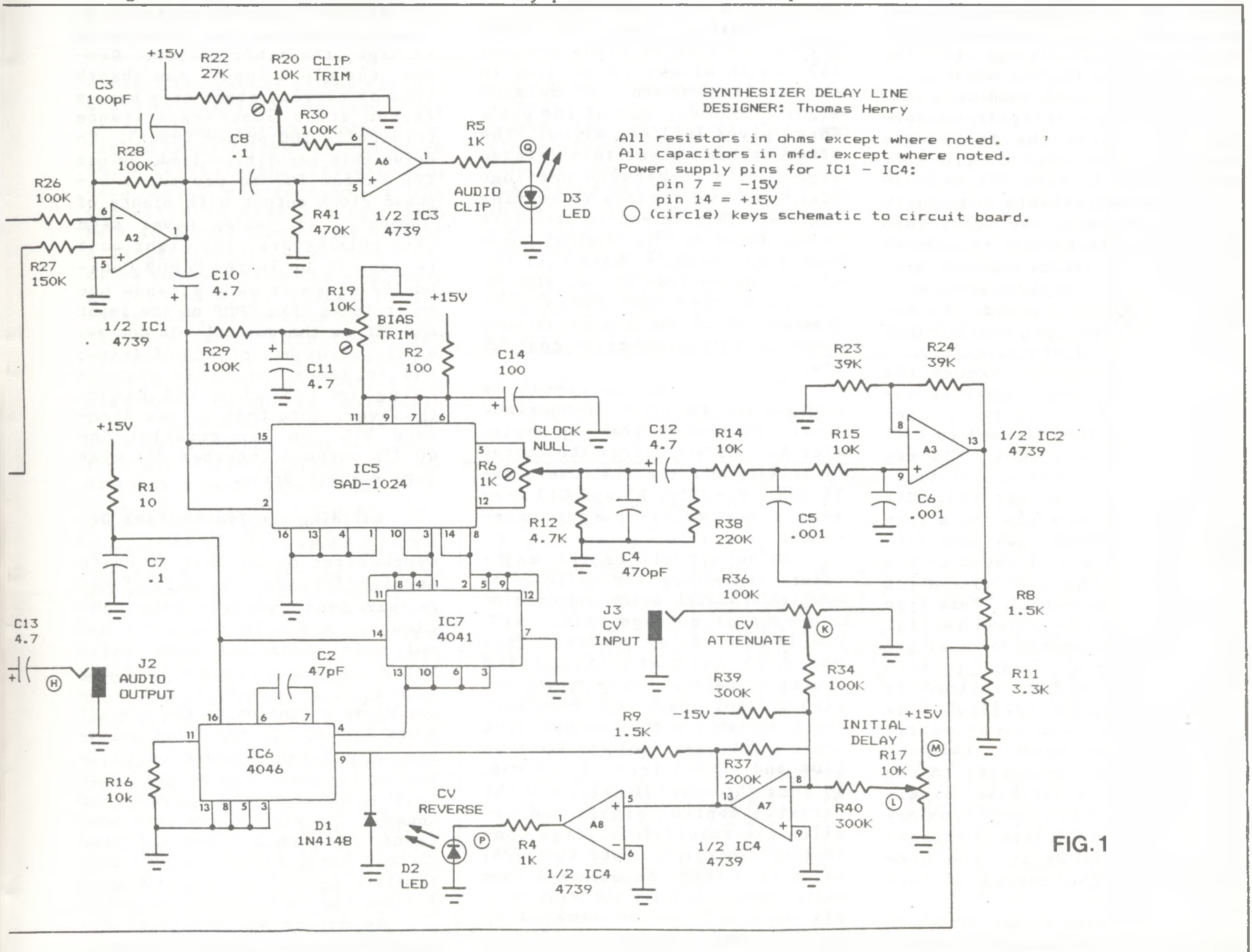


FIG. 1

Figure 1. The audio input enters the circuit via J1, then AC-couples through C9 to attenuator R35. A1 buffers the signal, which is then attenuated by a factor of ten by divider R13 - R10. (The SAD-1024 distortion increases rapidly with signal levels above 1V peak-to-peak.) The signal next goes to regeneration mixer A2, where C3 limits the high end to about 16 kHz. Restricting the response in this way keeps high frequency energy out of the SAD-1024, thereby minimizing foldover distortion.

At this point the signal splits into two paths. One path goes to the clip detector formed around comparator A6 and associated components. Adjust trimmer R20 so that the LED comes on when the SAD-1024's input signal exceeds 1V peak-to-peak. This not only lets you monitor potentially harmful conditions, but also, gives some indication of the onset of clipping. While this is a quick-and-dirty design for the clip detector, it does work!

The other path couples to the SAD-1024 via electrolytic capacitor C10. Since the delay chip requires a DC bias to properly pass a signal, R19, C11 and R29 provide an adjustable bias voltage. By the way, the delay chip is wired in the parallel-multiplex configuration, which tends to give the cleanest results when only short delays are needed. R2 and C14 serve to decouple the SAD-1024 from the rest of the circuitry.

After passing through the delay stages, the signal is applied differentially to trimmer R6, which can then be used to null out any clock glitches. The reconstructed delayed signal develops a voltage across R12; C12 then couples the signal to a second order Chebyshev low pass filter with a cutoff of about 14 kHz (the filter comprises A3 and its associated components). This type of filter has a steep initial rolloff at the expense of a slight peak in the pass band. In this application, a fast rolloff is crucial for good fidelity while any small response peaks really don't matter -- in fact, this can add some more complexity to the timbre. Bernie Hutchins' excellent manual, "Laboratory Problems and Examples in Active, Voltage-Controlled And Delay Line Networks" made the design of this filter a snap.[3]

The low pass filter serves to smooth the delayed signal from a stairstep to a more linear form, and at the same time, removes any residual clock garbage. Since the filter has a gain of about 2, resistor divider R8-R11 chops the signal back down to size, whereupon it goes to the polarity switcher (composed of A4 and related components). With S1 open, the signal is left unchanged; with S1 closed, the signal is inverted. Using this particular signal inversion circuit makes it possible to get by with a single SPST switch and at the same time keep the audio signal close to the circuit board. Note that the signal is inverted or not inverted to both the output blend control and to the regeneration loop; this is essential in order to obtain the full benefits of phase switching. (Incidentally, you may recognize this polarity changer since it appeared in the "Quadrature Function Generator" circuit.2)

The audio signal then heads off to one side of blend control R42, which allows you to dial in the desired amount of delayed signal. One extreme of the pot's throw gives full dry signal, the other extreme gives full delayed signal, and in-between settings blend the two signals to varying degrees. This versatile structure is described in "The Musical Engineer's Handbook."[4] Note that the signal coming from A4 can also be fed back into the audio path (thereby producing a more intense effect) via regeneration control R18.

The final mixed signal is brought up to full synthesizer level (10V peak-to-peak) by amplifier A5. C1 stabilizes the output by rolling off the response above 16 kHz. Finally, R3 and C13 present a nominal 1K output impedance.

We've hit all of the major points concerning the audio path; now let's switch gears and examine the control voltage path. R17, the initial delay control, lets you dial in a specific fixed amount of delay. However, you can also control the delay by injecting a variable control signal from other synthesizer modules (such as LFOs and ADSRs) into J3 and R36. A7 sums the initial delay and the variable control signal and applies the result to voltage controlled clock IC6. Note that IC6, which is a CMOS phase locked loop chip, operates from the +15V supply only and can be damaged if negative voltages are applied to its control pin. D1 shunts any reverse current to ground, and R9 limits the current flow to a safe level for D1.

In addition to safety considerations, it's useful to know if the control pin "sees" a reverse voltage since the control voltage will have no effect under these conditions. Comparator A8 turns on the "CV reverse" LED when this happens; adjust initial delay control R17, or CV attenuator R36, to get into a more appropriate region of the clock's operating range.

As mentioned earlier IC6 is a phase locked loop, but we are only using the VCO section of the chip. Pin 9 is the control voltage input, while pin 4 is the logic level output. R16 and C2 set the basic operating range. With respect to driving the SAD-1024 clock inputs, there are two major requirements: First, the SAD-1024 inputs want complementary waveforms so that when one clock pulse is high, the other is low. Second, the clock inputs can absorb lots of current since they have a rather stiff input capacitance (typically 100 pF per input). To drive this capacitive load, we use TTL buffer IC7 to provide a biphase clock output with plenty of "oomph". If you've never seen this chip before, you might want to look it up in the "CMOS Cookbook"[5] since it is a strange one -- it looks like CMOS on the input and TTL on the output! Of course, a TTL output has plenty of drive, and so keeps the clock signals going to IC5 nice and clean. (However, note that unlike standard TTL you can parallel the 4041's outputs together for even more current drive.)

Building the Synthesizer Delay Line. Building the synthesizer delay line is quite straightforward, but only if you employ good construction techniques. Since this circuit has not only audio but some hefty radio frequency signals floating around, it is essential that you pay close attention to the circuit board layout. In the final analysis, a printed circuit board is really the only way to go, since this minimizes the interaction between the various critical sections. **Figure 2** shows a tested circuit board layout for the synthesizer delay line, while **Figure 3** gives the parts placement guide.

As anyone who has ever designed an analog delay line will

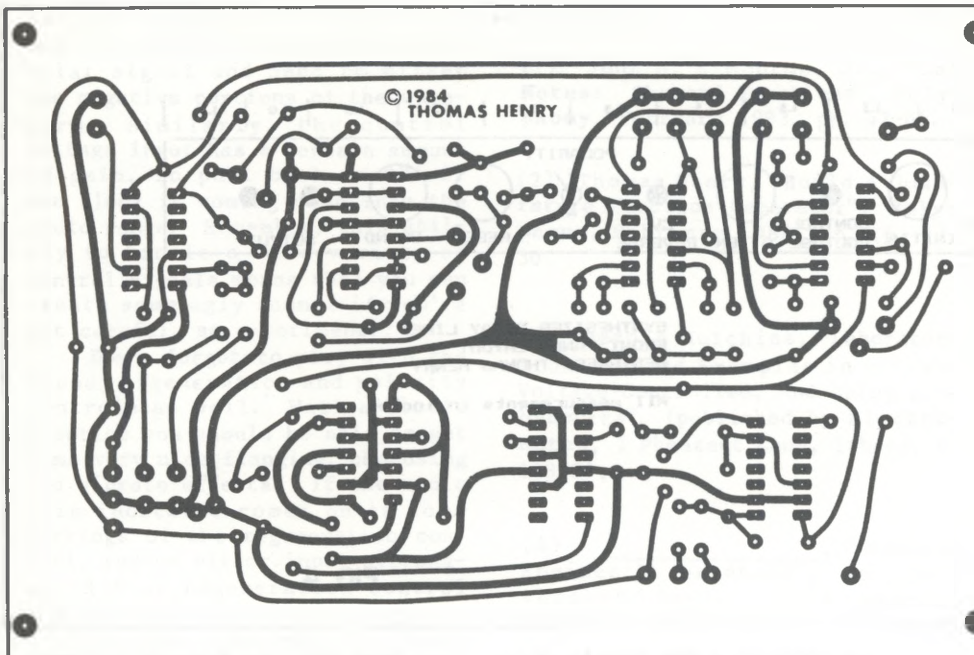


FIG. 2 PO BOARD FOIL PATTERN (not actual size)

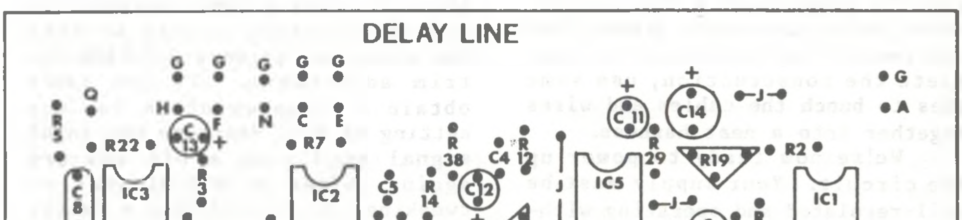
attest, the layout of the circuit board is no trivial matter. This board employs several exotic techniques to minimize hum and clock noise feedthrough. For example, each major sub-section of the circuit must have its own path back to ground (preferably through a large trace). Similarly, the clock outputs from the 4041 must go through hefty traces to the delay chip, since this minimizes the effects of inductance and resistance. Finally, attention must be paid to the actual parts placement and op amp configuration. This design uses dual opamps and situates them well away from any clock circuitry. Because of these unusual layout considerations, the circuit board may look a bit strange with its redundant ground lines and different size traces, but believe me, they're all necessary for good operation!

We're all set to stuff the circuit board now, so refer to the parts list at the end of this article. Note the special requirements on capacitors and pots; if a mylar cap is specified, then use one! If no special mention is made, then you may use whatever you have handy. Load the board, using **Figures** 1 and 3 as a guide. Notice that C9, the input capacitor, doesn't mount on the board, but is wired between J1 and R35 right on the front panel. Also, 100 uF decoupling capacitors C15 and C16 aren't shown in the schematic for clarity, but should nonetheless be installed on the circuit board. Be certain to observe the polarity of these and other electrolytic capacitors. Finally, use sockets for all the ICs, since the SAD-1024 and CMOS chips can be damaged by static electricity. Install the ICs in their sockets only when all soldering is completed and do not handle them unnecessarily.

After loading the board, prepare a suitable front panel. **Figure 4** shows the layout I used in my prototype. Everything fits comfortably behind a single width rack panel (1-3/4" by 19") and there was enough room left over to mount two paralleled output jacks. The circuit board mounts on little angle brackets behind the front panel, and #4 hardware secures everything in place.

At this point you can complete the final wiring to the panel. Again, using the parts placement guide in **Figure 3**, connect the various wires as needed. Since delay circuits are susceptible to hum pickup, shielded wire must be used for the connections to points A, B, C, D, E, F, H and N. There is a ground contact for the braid next to each of these lettered locations on the board; here's how to use it. Strip one end of a shielded cable, exposing the braid and the hot wire. Tin both of these, then solder the hot wire into hole A and the shield into the ground hole (lettered G) next to this point. Run this cable to input pot R35 and connect the hot wire to the appropriate lug on the pot. The shield wire is left floating at the front panel. Complete all of the other hookups in the same way. Again, the eight connections mentioned above must use shielded cable for hookup to the front panel. The braided shield wire connects to the circuit board only; the panel side of the shield is left floating. This prevents any ground

FIG. 3 COMPONENT LAYOUT



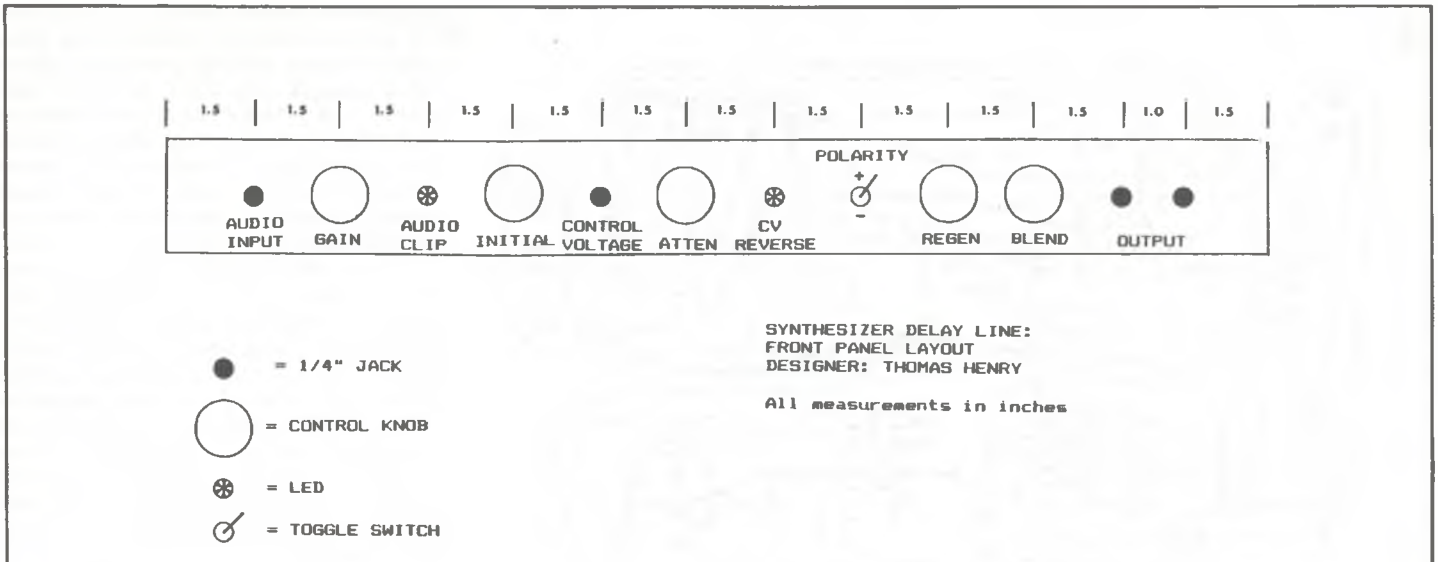


FIG. 4

loops, which can be a nasty source of hum. One exception here is the wire coming from point N; the shield hooks up to one terminal of the polarity switch while the hot wire hooks up to the other. Incidentally, Radio Shack makes a good shielded cable which is perfect for front panel work since it is barely larger than normal hookup wire and very flexible as well.

The wiring to the other controls is non-critical, so use ordinary stranded hookup wire. As a final step, run a single ground wire from the circuit board to the front panel. I ran mine to the ground lug of the output jack, as all of the ground connections to the other controls and jacks ended at this point. Thus, the front panel has a single ground connection in common with the circuit board, which minimizes ground loop and instability problems. To complete the construction, use some ties to bunch the cables and wires together into a neat harness.

We're now ready to power up the circuit. Your supply must be well-regulated and operating within its power limits (the circuit draws about -15 mA and +50 mA). It's also best if the wires running from the supply to the circuit board are short and direct. On my system, this wasn't possible; the result was hum problems, which could only be eliminated by sticking a 100 Ohm resistor in series with the +15V line just as it entered the circuit board. Short, hefty wires should prevent any problems, but if you do experience hum, try the extra resistor trick just mentioned.

Calibrating the synthesizer delay line. Tweaking the synthesizer delay line is fairly easy. Begin by double checking your work and when you are satisfied, apply power to the unit. Confirm that it doesn't belch smoke or flames! Next, turn gain control R35 up fully, regeneration control R18 down fully, blend control R42 to full delay signal and the initial delay control R17 to about a 1/3 position. Inject a 10V peak-to-peak 1 kHz sine wave into J1 and monitor J2 with an oscilloscope. Adjust the bias trim, R19, for minimum sine wave distortion. Adjusting R19 takes a little patience, since if you trim too far, the waveform will go flat on one side and not going far enough flattens it on the other side. Also, C11 adds a time constant, so turn the control slowly so that the waveform "catches up" with the trim adjustment. If you can't obtain a clean waveform for any setting of R19, decrease the input signal amplitude a bit and try again. After a few minutes of tweaking, you should have a fairly clean replica of the input. Note that this adjustment requires somewhat of a compromise as the SAD-1024's bias requirements change with clock frequency. If you wish, trim R19 to give the best possible waveform for various settings of R17.

Next clip the scope probe to the wiper of clock null trim R6, and set the timebase of the scope up fairly high (enough to see a cycle or so of the sine wave). As you vary R6 from its midpoint, you will see two waveforms. Adjust R6 so that the two signals converge.

Finally adjust the clip trim so that the audio clip indicator comes on as you slightly increase the 10V peak-to-peak signal to, say, 12V peak-to-peak. This provides visual indication that you are exceeding the nominal 10V peak-to-peak input requirements and may be getting into clipping.

If needed, you can also calibrate the unit without a scope. Instead of monitoring the output with an oscilloscope, patch in an audio amplifier. Adjust the bias trim for the least amount of distortion, as perceived by your ears. R6 isn't all that critical and may be just left at mid-rotation. And of course, you can adjust R20 simply by viewing the LED. So don't let the lack of an oscilloscope stop you from building this circuit!

After calibration, play with the delay line for a week or so to allow the components to break themselves in, then give the unit one final tweak. This should be the last adjustment you'll have to give it for quite a while.

Using the synthesizer delay line. If all has gone well, you now have a synthesizer delay line up and running. Since there are a lot of controls, be sure to really get familiar with them before assuming anything is wrong. For example, the control voltage input and the initial delay time give much more range than you'll be able to use in most situations, so be sure to tone these down if you're getting weird results. Also, the initial delay pot has a lot of "oomph" in case you're modulating the clock with a bipolar signal and need to offset the negative portions of the waveform. Similarly, the control voltage input has a certain amount of gain, so pull back on R36 if the clock is coming down into the audio range. Remember, this ability to handle a wide variety of control signals means that you can create some ugly sounds if you're not careful, so experiment.

Don't forget to play with the blend, regeneration and polarity controls as well. With a little practice you should be able to get some very nice flanging, chorusing and vibrato effects. If the audio clip indicator comes on at some settings of the regeneration control, reduce either input attenuator R35 or regeneration control R18.

Are you interested in experiencing "hypertriangular" flanging? Nothing could be simpler, providing that you have a wide-range, voltage-controlled, triangle wave generator. Modulate the delay line with a low frequency triangle wave from a VCO (like my VCO Deluxe circuit previously published in **Polyphony**), taking care to adjust R36 and R17 so that the clock is in the proper range.[6] Now feed some of the triangle wave back into the attenuated exponential control input of the VCO, which transforms the triangle wave into a "hypertriangular" waveshape. Adjust the exponential voltage attenuator on the VCO as needed to bring in the hypertriangular effect.

For other exotic sounds, put the synthesizer delay line under control of an envelope generator. This is especially neat when used with the "Snare+ Drum Voice".[7] Or how about using the Shepard function generator with multiple delay lines for some truly spacey effects?[1] Don't delay using delays any more!

Acknowledgment: Many thanks go out to Craig Anderton whose several excellent analog delay line circuits provided ideas which simplified the design of this one. Craig is truly a delay line guru, so refer to his articles on this subject for the real dope on getting these sometimes unruly chips to do remarkable things! (8), (9), (10), (11)

References

(1) John S. Simonton, Jr., "**Lab Notes:** Shepard Functions", **Polyphony**, February 1983, pp. 42-46.

(2) Thomas Henry, "Build A Quadrature Function Generator", **Polyphony**, February 1984, pp. 26, 27, 30.

(3) Bernie Hutchins, "Laboratory Problems and Examples in Active, Voltage-Controlled, and Delay Line Networks", (published by **Electronotes**, 1 Pheasant Lane, Ithaca, NY 14850).

(4) _______________, "Musical Engineer's Handbook", pp. 3a-4 through 3a-5. (see address above).

(5) Don Lancaster, "CMOS Cookbook", (Indianapolis: Howard W. Sams and Co., 1977), p. 99.

(6) Thomas Henry, "VCO Deluxe", **Polyphony**, Nov./Dec. 1981, pp. 28, 29, 30.

(7) ____________, "Snare+ Drum Voice", **Polyphony**, Sept./Oct. 1982, pp. 28-31.

(8) Craig Anderton, "Pedal Flanger", **Guitar Player**, February 1980, p. 123; March 1980, p. 124; April 1980, p. 116.

(9) ______________,"The Flanger", **DEVICE**, Volume 1, Number 9, pp. 1-6.

(10) ______________,"Building the Hyperflange + Chorus", **Modern Recording**, July 1983, pp. 36-40 and September 1983, 56-60.

(11) _______________, "Build a Chorus Delay", **Guitar Player**, January 1982, pp. 26-30, 33, 34 and February 1982, p. 116.

PARTS LIST

Resistors

R1	10 Ohms
R2	100 Ohms
R3 - R5	1k
R6	1k trimpot
R7 - R9	1.5k
R10	2.2k
R11	3.3k
R12	4.7k
R13	8.2k
R14 - R16	10k
R17, R18	10k audio pot
R19, R20	10k trimpot
R21	18k
R22	27k
R23, R24	39k
R25	82k
R26, R28-R34	100k
R27	150k
R35, R36	100k audio pot
R37	200k
R38	220k
R39, R40	300k
R41	470k
R42	500k linear pot

Capacitors

C1	20 pF
C2	47 pF polystyrene
C3	100 pF polystyrene
C4	470 pF
C5, C6	0.001 uF mylar
C7, C8	0.1 uF
C9	0.22 uF mylar
C10 - C13	4.7 uF electrolytic
C14 - C16	100 uF electrolytic

Semiconductors

D1	1N4148 (or equiv.)
D2, D3	LED
IC1 - IC4	4739 dual op amp
IC5	SAD-1024 delay chip
IC6	4046 CMOS PLL
IC7	4041 TTL buffer

Mechanical parts

J1 - J3	1/4" phone jack
S1	SPST mini-toggle
Misc.	Wire, knobs, sockets, solder, LED clips, etc.

Note: C9 mounts on the front panel between the hot side of R35 and J1. C15 and C16 decouple the +15V and -15V supply respectively, where these lines first enter the board. Although they aren't shown on the schematic, provision has been made for them on the circuit board.

MICRO PROJECT

by: Michael Bienholz

If you play piano, you have probably noticed the distressing lack of a legato (sustain) pedal on a synthesizer. Fortunately, it is very easy to build one, as shown in **Figure 1.** This circuit hooks into an existing envelope generator, as shown in **Figure 2.** The best way is probably to unsolder the wires at the decay pot and insert the footswitch circuit.

FIG. 1

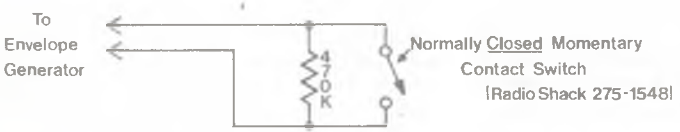
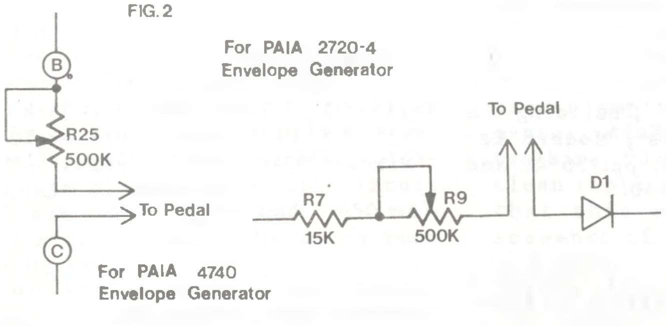
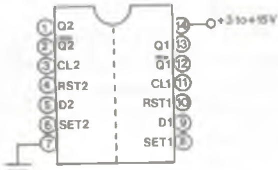


FIG. 2

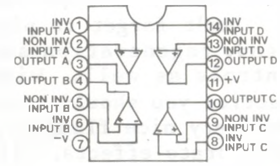


With the momentary switch closed, the envelope generator operates normally. By opening up the switch, the 470k resistor increases the sustain time to give a sustain effect. I used a closed circuit jack to connect the pedal to the envelope generator, so that it will function normally with the pedal disconnected. One could also try putting a 500K pot in place of the 470K resistor for a variable sustain time.

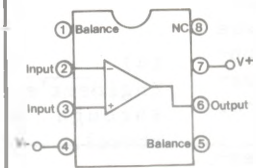
DATA BANK



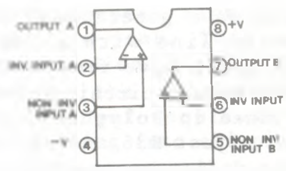
4013
DUAL D FLIP FLOP



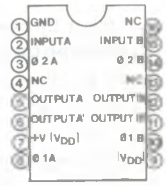
4136
QUAD OP AMP



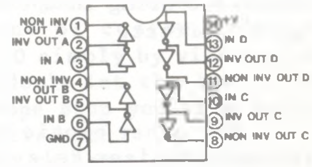
LF351
OP-AMP



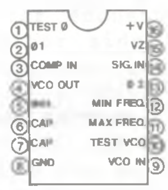
1458
DUAL OP-AMP



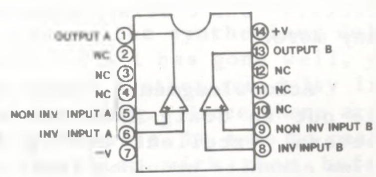
SAD1024
DUAL ANALOG DELAY LINE



4041
QUAD TTL BUFFER



4046 PLL
PHASE LOCKED LOOP



4739
DUAL LOW-NOISE OP-AMP

570/571
COMPANDER

PARTS and KITS from: PGS

LINEARS

TL061	BiFet	.72
TL062	Dual BiFet	.99
TL064	Quad BiFet	1.95
TL071	BiFet	.65
TL072	Dual BiFet	1.15
TL074	Quad BiFet	1.95
NE555	Timer	.39
NE570	Compander	3.80
NE571	Compander	2.95
NE572	Compander	4.95
UA741	Comp. OpAmp	.29
MC1456	Low Noise OpAmp	.90
RC1556	Low Noise OpAmp	1.48
CA3080	OTA	.94
CA3280	Dual OTA	1.98
RC4136	Quad OpAmp	1.10
RC4739	Dual Low Noise	1.19
NE5532	Dual High Perf	3.70
NE5534	High Performance	2.65

SPECIAL PURPOSE

SAD-1024	Analog Delay	17.50
SAD-4096	Analog Delay	37.50
MK50240	Top Octave Div.	5.95
SN76477	Sound Generator	3.45

SANYO HYBRID POWER AMPS

STK050	50 Watt Power Amp	19.40
STK070	70 Watt Power Amp	24.20

SSM- SOLID STATE MICRO-TECHNOLOGY

SSM 2010	VCA	7.50
SSM 2011	PreAmp	5.75
SSM 2012	VCA	9.50
SSM 2020	VCA	7.50
SSM 2022	VCA	7.50
SSM 2030	VCO	7.50
SSM 2033	VCO	10.00
SSM 2040	VCF	7.50
SSM 2044	VCF	7.50
SSM 2050	VCTG	7.50
SSM 2056	VCTG	5.75

THERMISTER (Temp. Sensing Resistor)

TSR-Q81	Tel Labs Q81 1k	$3.50

OPTO-ISOLATOR

CLM6000	Clairex CLM6000	$2.85

CAPACITORS (25 volt)

701-100	100 pf polystyrene	.25
701-180	180 pf polystyrene	.25
701-1000	1000 pf polystyrene	.25
701-2200	2200 pf polystyrene	.25
701-2200	3300 pf polystyrene	.25
701-3900	3900 pf polystyrene	.25
702-005	.005 uf mylar	.12
702-01	.01 uf mylar	.12
702-05	.05 uf mylar	.16
702-1	.1 uf mylar	.21
702-22	.22 uf mylar	.33
703-1.0	1.0 uf tantalum	.39
703-3.3	3.3 uf tantalum	.49
703-4.7	4.7 uf tantalum	.59
704-2.2	2.2 uf electrolytic	.21
704-4.7	4.7 uf electrolytic	.21
704-10	10 uf electrolytic	.21
704-100	100 uf electrolytic	.31
705-10	10 pf ceramic disk	.15
705-.01	.01 uf ceramic disk	.12
705-.1	.1 uf ceramic disk	.17

IC SOCKETS (soldertail)

IC-S-08	8 pin high quality socket	.27
IC-S-14	14 pin high quality socket	.30
IC-S-16	16 pin high quality socket	.34
IC-S-18	18 pin high quality socket	.40
IC-S-28	28 pin hgih quality socket	.60
IC-C-08	8 pin economy socket	.13
IC-C-14	14 pin economy socket	.15
IC-C-16	16 pin economy socket	.17
IC-C-18	18 pin economy socket	.20
IC-C-28	28 pin economy socket	.40

RESISTORS 5%, 1/4 watt

All EIA values available from 2.0 ohm to 5.1 Meg. Also availble is 10 Meg.

100 each of same value	$1.50
50 each of same value	.98
25 each of same value	.75
10 each of same value	.40
5 each of same value	.25

ASSORTMENTS

10 each of 10 values (100)	3.00
25 each of 10 values (250)	6.50
50 each of 20 values (1000)	16.00

CHORUS/DELAY KIT

This chorus/delay unit, designed by Craig Anderton and featured in Guitar Player magazine, provides flanging, slapback echo, and automatic double tracking effects. The delay range is from 2 ms to 80 ms. Due to the use of compression and expansion techniques, the unit has dead-quiet operation up to about 50 ms and only minimal noise out the full 80 ms. This project kit consists of all electronics, pots, jacks, etc. Also included are the two circuit boards (etched, drilled, and legended) needed for the project. Not included is wire, solder, case, knobs, etc. The Chorus/Delay unit also needs a well regulated bi-polar 15 volt power supply (not included). (A punched and legended rack mount panel will soon be available for this project.)

Order KT-CD777 .... $78.00

"SNARE +" DRUM VOICE KIT

This percussion synthesizer was designed by Thomas Henry and appeared in POLYPHONY magazine. Here's what Craig Anderton had to say about the "SNARE+". "At last - an inexpensive drum voice that has a punchy, full sound. .....All in all, the Snare + delivers a lot of drum sounds, and I would unhesitatingly recommend it to anybody who's tired of the thin sound found in most electronic drum units."

We offer the kit with or without a panel. Kit 3770 contains all electronic parts, switches, jacks, pots, etc, as well well as etched, drilled, and legended circuit board. Kit 3772 includes all this plus a punched and legended rack mount panel (standard 1 3/4 by 19 inches) available in black or blue (both with white legends).

Not included with either kit is wire, solder, mounting hardware, etc. The SNARE + also needs a bi-polar 15 volt power supply (not supplied).

KIT 3770 Basic SNARE + kit	$33.95
KIT 3772 SNARE + with rack panel	$44.94

THE "CLARIFIER" GUITAR EQ/PREAMP

The "CLARIFIER" is an onboard preamp/EQ module for guitar. This design, by Craig Anderton, was first seen in the pages of GUITAR PLAYER magazine. Here's what the CLARIFIER will do: Replace the guitar's standard passive tone control with a two control, active circuit which provides over 12 db of bass and treble boost and up to 6 db cut.... Buffer your pickups from external loading, giving additional output and improve high freq response.... Add a nominal 6 db of gain to give your signal a bit more punch, as well as improve the signal/noise ratio in multiple effects systems... make your guitar immune to the high freq loss caused by long cable runs.

The CLARIFIER kit is available in two options, both of which include a high quality drilled, legended, and masked circuit board, as well as complete step by step instructions. Kit 2450 contains everything needed for a complete unit.. Kit 2455 contains everything execpt the pots (for those who prefer a particluar brand of potentiometer). Batteries are not included with either kit.

KIT 2450....Complete CLARIFIER kit	$18.95
KIT 2455.....CLARIFIER less controls	$14.95

TERMS: (Check, Money Order, Cashiers Check - Add .75 if under $10.00)— ($10.00 minimum on C.O.D. (UPS only) add $1.50)— (Mastercard and Visa: $10.00 minimum. You must supply exp. date.)— (Indiana residents add sales tax.)

SHIPPING AND HANDLING: $1.00 plus 5% of purchase. We will credit any amount over our standard rate.

SATISFACTION GUARANTEED!

SIGNAL DIODE

601-60...1N914 (1N4148) signal diode	5/.35

TRANSISTORS

2N3904.....2N3904 NPN Transistor	.25
2N3906.....2N2906 PNP Transistor	.25

POTENTIOMETERS

(3/8 long shaft, 5/16 mounting hole)

854-401	10K Linear taper	1.09
854-501	100K Linear taper	1.09
854-505	500K Linear taper	1.09
855-401	10K Audio taper	1.09
855-501	100K Audio taper	1.09
855-505	500K Audio taper	1.09
856-401	10K Audio taper with on/off switch	1.25

TRIM POTS (vertical mount)

802-251	250 ohm trimmer	.40
802-103	10K trimmer	.40

MINI TOGGLE SWITCHES

403-20	SPDT (on/on) sub-mini (3A)	1.20
403-40	DPDT (on/on) sub-mini (3A)	1.50
405-10	SPST (on/off) bat handle (6A)	1.85

LED's

Please note that the typical DC forward current (I-fwd) of these LED's is less than those offered elsewhere making these LED's ideal for battery circuits or others where current consumption is a factor.

305-201	Red T-1¾ jumbo diffused (20 ma.)	.30
305-202	Green T-1¾ jumbo diffused (30 ma)	.40
305-203	Dual T-1¾ jumbo diffused (50 ma)	.90
305-204	Tri T-1¾ jumbo diffused (20 ma)	1.50

Note: 305-204 is a three lead, tri-color (green, red, yellow) device. It is essentially two separate LED's in one package. (The yellow is obtained by turning on both green and yellow.)

JACKS and PLUGS

1/4 In. PHONE JACKS

901-101	Mono standard phone jack	.45
901-103	Mono with n/closed contact	.52
901-105	Mono encl. jack (open back)	.55
902-211	Stereo standard phone jack	.70
902-213	Stereo encl. jack (open back)	.77

1/8 In. MINI JACKS

903-351	Mono with n/closed contact	.32
903-353	Mono encl. (open back)	.26
903-355	Mono enclosed with contact	.35

RCA JACKS

921-100	RCA jack, chassis mount	.34
921-200	RCA jack on phenolic mount	.25
921-300	Dual RCA on phenolic mount	.43

1/4 In. PHONE PLUGS

911-201	Mono, black phone plug	.48
911-203	Mono, red phone plug	.48
911-205	Mono, chrome (metal) plug	1.20
911-211	Stereo, black phone plug	.65

1/8 In. MINI PLUGS

913-251	Mono, black mini plug	.38
913-253	Mono, red mini plug	.38
913-255	Mono, chrome (metal) plug	.56

SWITCHING JACKS

These are stereo phone jacks that contain an independent switching sywtem that is controlled by the insertion of the plug. Jack 905-301 contains the equivalent of a DPST normally on switch. Jack 905-302 contains the equivent of a DPDT on/on switch making it ideal for switching bi-polar power supplies on and off in effects boxes, etc.

905-301	Stereo jack with SPST switch	.90
905-302	Stereo jack with DPDT sw.	1.00

PGS ELECTRONICS
Route 25 - Box 304
Terre Haute, IN 47802

EQUIPMENT EXCHANGE

PUT POLYPHONY TO WORK FOR YOU. List equipmment for sale or trade, job openings, positions wanted, etc. Equipment exchange classified rates for individuals offering goods or services for sale or trade: 25c per word, 20 word ($5.00) minimum charge; Commercial establishments: 50c per word. Prices, zip, phone numbers count as one word each. DISPLAY CLASSIFIED: $15 per column inch, one inch minimum, camera ready art to be supplied by advertiser. All classified advertising must be prepaid. Advertisers using a Post Office Box number for responses must furnish Polyphony Publishing Co. with a complete street address and phone number. Readers should respond directly to advertiser. Polyphony is not responsible for claims made in ads, or for the results of any transactions. Polyphony reserves the right to edit or refuse any ads submitted.

Music equipment

SYNERGY keyboard. Digitally generated sounds, built in 4 track sequencer. Excellent condition, never left home. $2700. Call Joe at (609) 927-4100.

Large PAIA Synthesizer for sale. Many modules have been modified for improved performance. 4 wing cabinets include 3 VCO, 4 VCF, D/A, Seq., 2 VCA, RM, RVRB, 2 Env. Gen., 2 Mix, Etc. cost $1200 will sacrifice for $400 O.B.O. Brian Folkes, 25712 #52 Le Parc Blvd., El Toro, CA 92630 (713) 380-9447.

FOR SALE: 1. Alpha Syntauri System. Includes Apple II+, all cards, software drive, monitor etc. $2900; 2. Syntauri recased in PAIA 5-octave series-V controller with all cards, software, drive monitor and lone apple: $2450; Both selections include free business, games, word processors, utilities software. 3. Vantage cherry sunburst Les-paul copy with case, strap: $95; 4. Assembled and tested PGS Chorus/Delay unit with labelled rack mount panel: $75. Tarik Qureshi, 2530 McKay Ave., Windsor, Ontario, Canada, N9E-2P5 (519) 966-2673.

KORG, Mono/Poly synth Korg SQ-10 Analog Sequencer both in excellent condition. Used only at home. Best offer. Nick Toth, 78 Milldale Ave. Plantsville, CT 06479.

TWO CHANNEL ANALOG DELAY UNIT. Made by Southwest Technical Products. Ambience Synthesis and Delay effects. $100 or trade for Casio keyboard. Gerard (415) 527-4026.

FOR SALE: PAIA Synth - (4) 4780 sequencers, (5) 4720 VCO,s (4) 4730 VCFs, (4) 4740 ADSRs, (4) VCAs, (2) 4710 Balanced Modulators, (2) 4711 Mixers, (2) 4712 Reverb, 8780 D/AC, (2) 8781 Quash, VC phaser, 3 octave keyboard w/glide, and Stringz 'n' Thingz in custom cabinet - $700. PAIA Phlanger - $30. Aries 15 unit case, power supply, AR-318 S&H/clock/Noise, AR-324 Dual LFO/Lag/Inverter, AR-328 Stereo Reverb & output - $300. Sequential Circuits Pro-One - $300. ADC 12 band stereo graphic equalizer - $100. TEAC A-3340S 4 track tape deck - $500. Each with all cables and documentation. Mark harper, 4761 Kings River Ct., San Jose, CA 95136, (408) 265-8043 evenings.

OLD TECHNOLOGY STILL GOOD. Micromoog $300, ARP String Ensemble $400, Soundcraftsman Preamp/Equalizer $200, TEAC model 2 mixer $200. P.O. Box 16211, Seattle, WA 98116. I'll pay shipping.

COMPUTER/SYNTHESIZERS FOR SALE. AIM-65 6502 based computer; cassette storage and tapes; excellent 4 output power supply, Custom 5 oct polytonic synthesizer. Very large PAIA/custom modular synthesizer. Synthesizers are less than perfect but operational. Computer is prime. make offer. Call Steve at 616-399-3775 or 616-392-1491 Ext. 290.

Parts

SID CHIPS $20.00
D. Paul, P.O. Box 391, Center Valley, PA 18034

Misc.

ACME REAL-TIME CASSETTES
Copies affordably priced up to C-92. TDK SA/AD Equivalent Tape, Dolby, Audiophile specs. Better sound than many LPs! Rates, Free Sample: Acme, 3821 N. southport, Chicago, IL 60613 (312) 477-7333.

Recordings

Rough Order of Magnitude: The release of a new experience! Synthetic music on cassette from SYNTHETIC FOUR MUSIC. $6.95 to: SYNTHETIC FOUR MUSIC, 6049 Butternut Dr., West Olive, MI 49460.

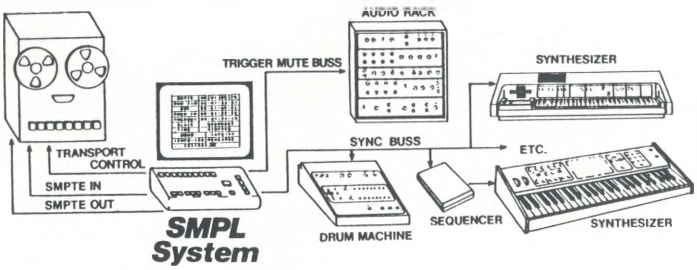
AD INDEX

AT LAST! COMPUTER AUTOMATION FOR THE SMALL STUDIO

THE SMPL SYSTEM BREAKS THE PRICE BARRIER FOR SMPTE TIME CODE

Synchronous Technologies' SMPL System is the only time code device specifically designed to solve the problems of the smaller recording studio. In one integrated package it provides functions and features which can't be duplicated with existing time code equipment even at many times the system's low price. Functions include:

- SMPTE Time Code generator
- SMPTE Time Code reader
- Automatic Punch In/Out
- Drum and Synth Sychronizer
- Programmable 8 event sequencer
- Autolocator
- Time Code Metronome
- Recorder Remote Control



IT'S THE ENGINEER YOU ALWAYS WANTED

With the SMPL System, insert editing no longer requires the combined skills of engineer, musician and juggler. During review, Punch In and Punch Out points are set on the fly and saved in the computer's memory. Separate Rehearse and Take modes allow you to rehearse and preview the edit points as many times as necessary before committing to tape.

Eight programmable event outputs are useful for triggering effects, changing instrument presets, fractional measure channel muting and much more.

The eight autolocator points let you get from section to section with a minimum of hassle and wasted time. And a separately programmable CUE point controls the recorder for a looping function at the end of rehearsals and takes. You concentrate on the art, the system attends to details.

SYNC-LOCK THE NEW GENERATION OF INSTRUMENT/RECORDERS

Through the SMPL System's MIDI standard 24 tick/beat synchronizing buss, an ever increasing number of Polyphonic Synthesizer Sequencers and Electronic Drum Sets can be precisely synchronized to material on tape. Many pre-MIDI instruments also conform to this standard and other non-standard sync formats can be handled with modest additional equipment.

Unlike tone or click-track type synchronizers, the SMPL System can be started at any arbitrary point in the work and the computer intantly calculates the correct phase of both metronome beat and synchronizing signal. You save time and aggravation by not having to play through the entire work to do an edit at the end.

MORE, HIGHER QUALITY "TRACKS"

Since much of today's commercial music involves digital drums and sequencer controlled polyphonic synthesizers, the SMPTE track can replace numerous tracks which might otherwise be recorded as audio. Not only does this effectively increase the number of tracks available, it allows these tracks to be mixed first generation to the master tape. No more loss of quality from ping-ponging and dubbing.

AN OFF LINE TERMINAL FOR THE ENTERTAINMENT INDUSTRY'S SYNCHRONIZING NETWORK

The benefits of using industry standard non-drop format SMPTE Time Code can't be overstated. With the SMPL System, tapes produced in the small studio will transport to larger studios and be compatible with automatic mix-down and chase-locking equipment.

Even if you never need to sync audio to video, this compatibility has compelling economic advantages. Tapes produced on machines with limited tracks can be "pyramided" to 24 and 40 track studio machines, allowing you to create in your own environment at your own pace and still have easy access to expensive studio facilities on an as-needed basis. In many cases, your savings in billed studio time will quickly pay for the SMPL System.

A VERY HUMAN INTERFACE

Either a Color or B/W Monitor or TV set can be used as the display device for the SMPL System. The easily readable display provides all current information on the operation of the system including operating mode, metronome tempo, current time, In/Out points, CUE point, recorder status and more. And the SMPL System doesn't require an advanced engineering degree to operate, all functions are straight forward and obvious.

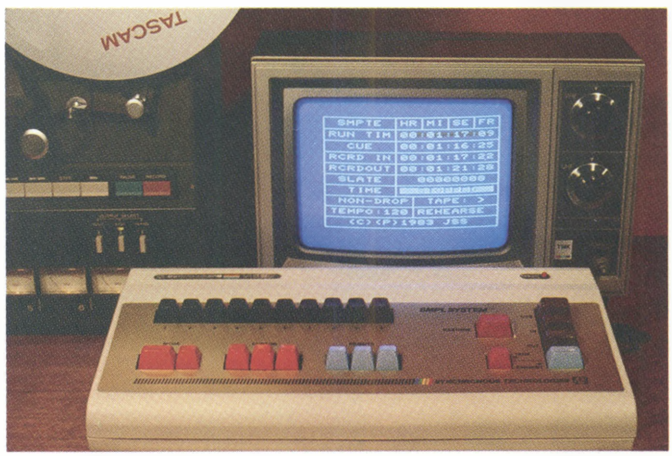
IT'S A COMPLETE, LOW COST SYSTEM

Not only is the SMPL System itself low in price, it's designed to be used with lower cost multi-channel cassette or open reel recorders by simply plugging into their normal remote control jacks. Neither tachometer output nor speed control input are required. Even recorders without remote control jacks can usually be modified for use with the system.

The complete SMPL System consists of: Personal Computer with keyboard modified for SMPL functions, SMPL System Software/Interface cartridge, VHF channel 3/4 modulator, power supply and Using and Installation manual.

SMPL System................................$995.00 (12 lbs)

CALL OR WRITE FOR THE NAME OF YOUR NEAREST DEALER.



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SYNCHRONOUS TECHNOLOGIES

P.O. Box 14467 • 1020 West Wilshire Boulevard • Oklahoma City, Oklahoma 73113 • (405) 842-0680

Psycho-acoustic Satisfaction

A speaker design with your ears in mind

When we designed the "SD" Speaker Series, our goal wasn't to produce impressive specs — we wanted to produce great sound.

Sure — extended frequency response was important. So was a smooth crossover between lows and highs. But most important was psycho-acoustic satisfaction.

Acoustics is the science of sound. But psycho-acoustics goes deeper. It's the science of how the brain perceives sound. We wanted a speaker that pleases the brain, so we focused on the characteristics that affect your ears, not our speakers.

Characteristics like tonality . . . transparency . . . and faithfulness to the original sound.

So we designed a three-way system, not two-way, for an all-around sound. For a punchy, tight bottom end, we incorporated acoustic filtering on the front panel. And we put in a moving coil tweeter for a clean high end with lots of definition. In total: great sound.

And great sound means great specs.

Specs like 360 watts continuous power handling. . .a very sensitive 102 dB efficiency rating. . .and smooth response over a wide 50 Hz to 20 kHz bandwidth.*

The Thiele-Small aligned bass reflex cabinets are a real plus. So is the high-frequency attenuation that lets you tailor the SD's output to your performing requirements and the room acoustics.

*Model 38SD

Then there's ergonomics:

- ☐ a protective steel mesh
- ☐ recessed handles
- ☐ interlocking corners for easy stacking
- ☐ a built-in adaptor for stand-mounting

Small. Lightweight. Built to take the abuse of the road. Sold with a two-year warranty. What else is there?

Besides psycho-acoustics, that is.

For complete technical data call or write:

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