POLYPHONY

Nov./Dec. 1981
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ELECTRONIC MUSIC & HOME RECORDING

SYNTHESIZING STRINGS

PRO-ONE REVIEW

CIRCUITS: VCO DELUXE

Jay Lee Interviews: E-Mu Founder DAVE ROSSUM

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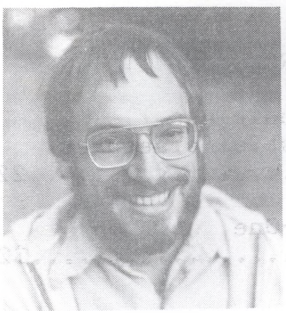

I just got back from the 70th AES show in New York. No, I'm not going to tell you about all the fabulous new toys; what impressed me most about the show was the many musicians I talked to. If they're representative of where musical electronics is going, we're in good shape. In fact, we may even be on the threshold of something very exciting.

The biggest change I felt was an openness, a "we're all in this together" attitude that contrasts sharply with the "I have a bigger synthesizer than you do" attitude so often evidenced in the past. Perhaps as musicians, we allowed ourselves to become seduced by technology; people now seem much less interested in what kind of instrument you play, preferring instead to analyze the music.

Maybe this emphasis on music is due to our being comfortable with the technology. I'm sure when you heard your first synthesizer record, your reaction - like mine - was "how do they do that?!?" But after you've heard your 100th synthesizer record, you know how they do that and now the concern becomes "is this good music?" That's a healthy sign. In essence, the novelty has worn off, and now we can see musical electronics for what it is: An additional set of tools for musical expression.

Perhaps some of the co-operative attitude was also spawned by the lack of public acceptance of electronic music. While some people still believe that the public is dense and doesn't "appreciate" what we're doing, the fact is that various pieces of electronic music have reached people emotionally over the years (through movie soundtracks as well as records). So, we know that public acceptance is possible. The bottleneck appears to be distribution - matching the right records with the right audience. Most record companies simply don't know how to market music that doesn't fit into easily definable categories (rock/jazz/folk/classical), which indeed presents a marketing problem.

Once musicians came to grips with this problem and realized they weren't going to be signed by the "biggies" (possibly a blessing in disguise, but that's another story), they dealt with the situation in a variety of ways. They pressed their own albums, and did their own marketing. They made the required "trillions of calls", as Bill Rhodes said a couple of issues ago, to record stores and radio stations.

I talked to several people who had set up local independent distribution, specifically designed to get the right kind of people and the right kind of music together. I was surprised when one independent distributor mentioned that he specialized in "electronic music, free jazz, and blues" records. What do these forms of music have in common? Arguably, not very much. But the people making these records have a lot in common. These are the self-reliant folks, the ones who don't roll over and play dead when the major labels don't return their calls (or their cassettes). They recognize that driving a limo doesn't make for better music, that record company parties where everybody gets smashed doesn't make for better music, that full page ads in the music trades don't make for better music. In fact, a $20,000 synthesizer isn't going to make better music than a $1,000 synthesizer...it's the musician, and the musician alone, who determines whether the music will be good or not.

The beneficial effect of marketing your own music is incalculable. For one thing, you're generally freed of the legal hassles, producer hassles, and will-it-get-airplay-and-does-it-last-less-than-3-minutes hassles that can sap your artistic vitality. Sure, it's hard work to pound the pavement and bug the local DJs, but it's work that needs to be done and no one is going to do it for you. Interestingly, I have yet to meet a musician who has put out a self-produced record that didn't eventually break even...which is more than the big labels can say. I've seen far too many musicians lose their soul as they saw control over their music slip away from them. The do-it-yourselfer doesn't have that problem.

Another beneficial effect is the feeling of co-operation this whole situation creates. When I mentioned that I would be releasing an album soon, people willingly offered me advice, turned me on to radio stations that play independent music, and the like. I think people realize that the success of ANY self-made music helps us all. Some independents have said to me that they feel like they've been beating their heads against a brick wall in trying to get their music marketed. Well, I think so many of us have beaten our heads against the wall that some real cracks are showing, and several musicians have even punched a hole in the wall and come out the other side.

The future? Simple. Major record labels have proven that they are incapable of identifying, analyzing, and capitalizing on trends. Boston's first album sold 8 million copies, and yet it almost didn't get released; the most successful Peter Gabriel album so far was the one judged as having the least commercial potential by his label; and there are plenty of other examples. I can easily picture a time when it's the do-it-yourselfers who are generating the talent, with the major labels principally serving as distributors, which seems to be what they do best. I also see the independents virtually taking over the market on albums selling 100,000 copies or less, a market which the majors have more or less written off.

We may be on the threshold of something very exciting. Keep your eyes on these pages in the months ahead.

TODAY'S TECHNOLOGY TODAY

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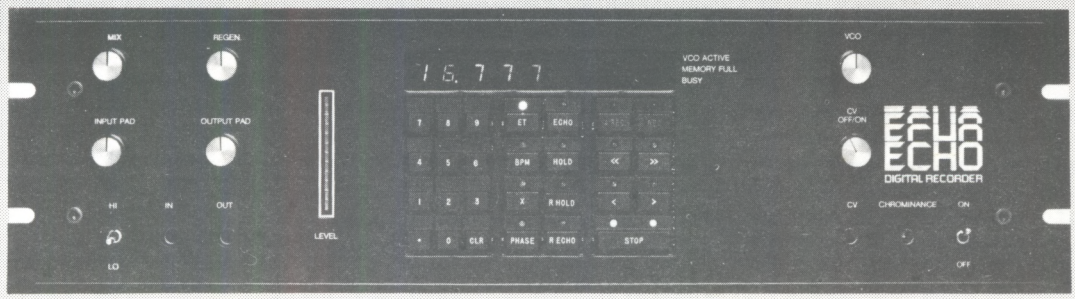

7 OCTAVE (82 NOTE) AUTOMATIC CHROMATIC TUNER FOR LESS THAN $200

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LETTERS

TAPE TOY TIP

I've come across an extremely helpful device - one of those that makes you wonder how you ever got along without it. It's called the VOXCOM photo recorder, and its advertised purpose is to record sound on photographs via self-sticking magnetic tape (or magnetic tape tabs recorded when the picture is taken).

I've found this very useful when filing patches. For some time now I've been filing patches for my P-4700/J, and all its numerous modifications, on index cards. I had the cards printed with a graphic representation of the front of my five road cases. When I find a useful patch, I draw in the patch placement and control knob settings. The problem has always been describing the sound well enough in words so that three months later I would know what I was talking about. Recording the sounds on tape caused a time problem in indexing and retrieving, which made that alternative impractical.

Well, you've probably guessed what I did. With a piece of magnetic tape on the index card, each patch has a 10 second (longer if larger cards are used) sample of the patch. When I am looking for a particular sound or effect, I find the category in my index file and run a few cards through the photo recorder until I find what I need.

Another application is with computer software. Cassette interface has always been a time consuming problem to me - not so much the actual loading time, but the cueing time for multiple programs on one cassette - and putting each program on a separate cassette creates a cost problem. Recording the program on index cards using the magnetic strip is much faster, and only costs a couple of pennies per copy.

The device might seem overpriced but I think it's worth it. It's available for $69.96 from:

VOXCOM
PO Box 2520
Peachtree City, GA 30269
404-487-7575

The modification for computer input is $17.80 and rolled tape costs $12.50 for three rolls (180 inches per tape roll), or $2.95 per roll from Edmund Scientific (Edmund also sells the recorder, but for $89.95).

Mike Bailey
Creative Rediffusions
Panama City Beach, FL

Mike - Thanks for the tip. For yet another use, how about storage of special effects, sound effects, and the like for theater or film projects?

REVIEWER REVIEWED

In the July/August issue review section, my name was mentioned in reference to a recording by Randall McClellan. The context in which it was used was entirely the opinion of the reviewer, and definitely not mine. To critique a composer's very early works ten years later, in the manner of comparison to a (then) future student, is of questionable benefit.

In general, I enjoy Polyphony very much, including the reviews. Thanks for your continued support of independent musicians.

Michael William Gilbert
Amherst, MA

Michael - Your point is well taken. However, I think that Robert was trying to be constructive in his criticism, and simply pointed to your works as an example of what he meant. He certainly did not mean to imply that you shared his views. In any event, thanks for writing and giving us an opportunity to clear up any possible misunderstandings.

MORE REMCO COMMENTS

I was happy to see so many small electronic instruments reviewed in the March/April issue as I need one. While reading the description of REMCO's FX machine, though, I noticed an apparent similarity (control names, functions, internal patching, etc.) between the FX and the SN76477 (see Thomas Henry's recent columns for more information about this chip - Ed.) It seems more than remotely possible that the 76477 is the heart of the FX. I can't be 100% sure of this since I have never used or taken apart an FX, but if someone out there with an FX could check this out, that might open up many possible modifications to make the unit even better.

Dave Wilson
The Cheerleaders Corp.
Fairfield, CT

PATCHES?

One comment I have about Polyphony in general is that I've been noticing the "Patches" section getting smaller and smaller over the months. Is the problem simply a lack of patch contributions from the readers or is it something else? I enjoy this section very much and would like to see it grow. I have loads of patches that I could submit, but if the magazine doesn't want this sort of thing anymore, I won't send them in. Let me know, OK?

Charles J. Lauria
Towaco, NJ

Charles - Actually, there are probably as many patches as ever, just that the packaging is a little different. For example, Jim Aikin's article in the May/June and July/August issues had a wealth of patching information, as did your own article in the September/October issue. Rather than giving specific turn-these-knobs-this-way patches for specific synthesizers, these articles have presented a more "background" approach that will (hopefully) allow musicians to patch creatively. It's like the old saying, "give a man a fish and he'll be fed for a day, but teach a man to fish and he'll never be hungry again".

If you have loads of patches, perhaps you could wrap an article around them (as you did with the "Psycho-Acoustic Experiments" article) so that whether the reader has a minimoog, Pro-1, 2600, or PAIA 4700 they'll be able to put your information to good use. How about it, readers? Is this a good approach, or would you rather have patches oriented towards specific machines? Let us know your feelings and we'll do the best we can to keep everybody happy.

Ca

CORRECTIONS:

march/april

In the March/April issue (0605) the name of Rare But Fair Tape Service was inadvertently omitted, and the description of their service was listed under Passport Imports. RBF is at 7514 W. Indianola, Phoenix, AZ 85033 - tel 602-247-1778 and they are the tape distributor. Passport is another record distributor.

july/august

In the July/August issue (0701), page 24, figure 3, the unlabelled resistor at pin 7 is the one that's supposed to be removed according to instructions given in the text.

Also in the July/August issue, page 12, "Tube Tip", after "...paid is some crossover distortion" add the following: "First, check if there is any connection using pin 1 of the output tube sockets as a tie point. If any, lift. Then bridge pin 1 and pin 8 with a short piece of wire, or better yet, with a 1 Ohm, half Watt resistor."

Please make these corrections in your issues.

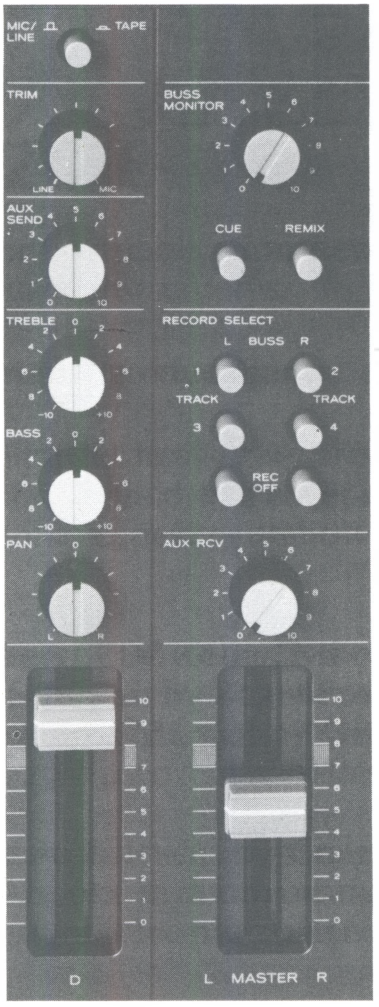
PORTASTUDIO.

LEARN ON IT.
REHEARSE ON IT.
CREATE ON IT.

Portastudio gives you all the essential functions of multitrack recording in one compact, self-contained unit.

You can work anywhere you have an electrical outlet. Connect electronic instruments directly. And try out new musical ideas. Will a keyboard work well in the break? Do you want a chorus behind the vocal? With Portastudio, you can find out before you put out hard cash for studio time.

Whether you're recording basic tracks, overdubbing, punching-in or mixing down to another cassette deck, Portastudio helps you get all the signals to the right places.



Portastudio's versatile 4 x 2 mixer section gives you mic/line/tape switching, trim control, high and low EQ, fader, pan and Aux Send for each input. The failsafe group switching matrix lets you record on up to two tracks at the same time. And the master fader gives you overall level control during recording and mixdown.

The full-logic cue system in Portastudio lets you hear everything you're doing all the time. Input and tape cueing, monitoring for recording or mixdown are all available. And every signal can be metered. Coming or going.

TEAC engineers created a totally unique format for Portastudio. Four tracks in sync on cassette tape at 3-3/4 ips. It's fast, simple, reliable and economical.

Portastudio's drive system is built specifically for the rugged needs of multitrack recording. Transport controls are all solenoid-operated for faster, easier switching. And you get a built-in variable speed control that lets you add special effects, fix a flat note or solve timing and cueing problems.

You can work with Portastudio using nothing more than headphones and a microphone. Or send the output through your home audio system. You'll also find the patch points and controls that let you use Portastudio with other equipment like echo units, equalizers and additional mixers.

Nothing else in the world hands you so much multitrack function in such a sensible way. So see your dealer today for a demonstration of the very affordable Portastudio.

TASCAM CREATIVE SERIES
TEAC Professional Products

re-view

by Robert Carlberg

Jean-Michel Jarre **Magnetic Fields** (Polydor 1-6325). Like Jarre's two previous solo albums, Magnetic Fields consists of two long suites combining several shorter pieces. Like his previous albums he builds up lush, hummable melodies using string-synthesizer like voices. Unlike his other discs, though, Magnetic Fields builds these tunes on a base of electronic percussion.

Probably this is the influence of the new wave/electrobeat artists such as John Foxx, Gary Numan, Joy Division, Orchestral Manoevres, Duran Duran, Spandau Ballet, etc. But despite this new element lending a somewhat harder edge to his music, Jarre still retains much of his lyrical, Romantic quality.

There is another change in Magnetic Fields, too. Equinox, Jarre's previous album, opened side two with a clap of thunder and featured a thunderstorm on the last track. Magnetic Fields puts much more emphasis on this interest in Musique Concrete, with subway trains, voices, lots of mechanical and water noises, and vocoder all used to effect. These occur not only in the segues between tracks, but also sometimes in the middle of a track, when the instruments drop the melody, "get weird", and the effects float in. Jarre, who has always been the synthesist for non-electronic music fans, takes a bold step - the AM radio housewives will probably like this album less (indeed, if it gets airplay), but diehard synthesists are sure to like it more than his previous discs. To top it off, with all the hard sounds this is the best recorded Jarre disc to date.

I find Magnetic Fields particularly gratifying because Jarre has carefully integrated new elements into his already successful formula. This proves not only his attention to what is going on around him, but also his willingness as an artist to keep moving. I admire that.

Hans-Joachim Roedelius **Lustwandel** (Sky 058). Having finally completed the housecleaning represented by his Selfportrait series, Roedelius once again gets down to some new composing. Happily, the interlude seems to have been used to advantage.

As on Cluster's Grosses Wasser, Roedelius once again seems to be exploring the textural possibilities of standard instrument voices. Close-miked piano and percussion are common enough, but today direct-into-the-board Mellotron, Clavinet, and string ensemble sound almost revolutionary. This technique encourages the listener to hear these instruments as self-entities, as instruments in their own rights and not necessarily imitative of anything else. In fact, the whole electronic music field seems to be growing out of the imitative stage - but that's another discussion for another time.

Lustwandel is sparse, with just these few unprocessed instruments (plus synthesizer) playing against each other. Where percussion occurs it is occasional, not regular. Roedelius' composing, as it often does, consists mostly of one instrument gently improvising over a repeated phrase on another instrument, with a third adding "color". In fact, with piano being the dominant voice, it is easy to view Lustwandel as a suite of small chamber pieces for keyboards, of which synthesizer just happens to be one.

The sparseness and naked tonalities leave Roedelius' composing as the only focal point, and luckily the material holds up. Grosses Wasser, which was in many ways similar, included too few really memorable musical moments for me, but Lustwandel has a number of very nice moments. It's as if Roedelius is telling us "Look, elaborate sound is secondary, the music is the reason why we're here".

Borbetomagus **Borbetomagus** (Agaric 1980). First off, I don't like this record. I can try to be as fair as I want, but chances are you'll know I don't like it anyway. There's just no way to review it without spilling the beans - but maybe that's a reason to review it.

Borbetomagus is a 4-piece band: two reedmen, electric guitar, and "electronics". The electronics sound like a Buchla, but it really doesn't matter because there isn't much room to hear the electronics. You see, everyone is feeding back and overdriven at 100% distortion, so you have to listen very carefully to hear what is electronic and what is simply distorted beyond recognition.

I don't know what they did to the saxophones - either played them very close into a cheap microphone or processed them through the electronics - but they don't sound very comfortable. Raspy, raucous, raped and very rude, but not "like a dry martini". The guitar mostly sounds like Hendrix burning his Strat at Monterey, and the electronics...well, it's hard to tell.

Taking a step outwards from the instruments being used, let me try to describe the music. All tracks appear to be made up on the spot - there is no discernable organization to the tracks, and no discernible interrelation between the instruments...except possibly processing (see above). There are limited notes used, mostly two (rapidly alternating), except when the reed players accidentally lapse into octaves, giving them-

Primary Colors

Synthesizer

Strings

Organ

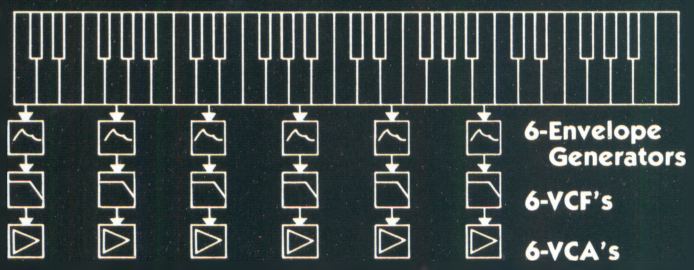
TRILOGY

Crumar's Trilogy combines polyphonic synthesizer, orchestral strings, and cathedral organ in one instrument, then lets you combine them in any proportion for limitless creative potentials. Just as the primary colors are combined by the artist to produce all the textures we see, the Trilogy allows you to combine the primary sounds to produce all the textures you'll want to hear and perform.

THE TRILOGY LETS YOU MIX SOUND LIKE COLOR TO CREATE A MUSICAL MASTERPIECE.

POLYPHONIC SYNTHESIZER WITH 8 PRESETS

The **Trilogy** combines the power of a fully polyphonic synthesizer with the performance features of the world's best lead line synthesizers, the result of its Unique Circuitry. Six separate, 24 dB lowpass filters, six independent, 4 parameter envelope generators, and six high performance VCA's span the keyboard, controlling the two independent oscillators and convincing you at least six lead line synthesizers are all playing at the same time.



Unlike many polyphonic synthesizers that limit you to 4 or 5 keys, you're not limited to even the six we just mentioned. Lay 10 fingers across the keyboard and all 10 keys will sound!

Oscillator/Pitch Control Network

The oscillators can produce sawtooth, square and mixed waveforms, be tuned to a unison or perfect fifth, or be detuned to anything in the middle. You can even see the degree of detuning with the LED tuning indicator, so you can tune by sight, on stage, before you even play a note.

Preset Selection

The **Trilogy's** Presets utilize one variable to let you explore your own sounds, and seven of what we consider the most popular synthesizer sounds, from acoustic piano to polyphonic brass to synthesizer variations. After familiarizing yourself with the **Trilogy's** variable mode, you may want to create your own presets. See your Crumar dealer for details on re-programming.

Special Keyboard Control Feature

For years, synthesists have pointed out the advantages of single and multiple triggering. The **Trilogy** is the only synthesizer that lets you choose several triggering modes at the same time, each assigned to a different musical effect. For example, multi-triggering can be used so that each key resets the delayed entry of vibratos and filter sweeps from the LFO. It's like having a delayed vibrato on each key. Very musical!

ORCHESTRAL STRINGS

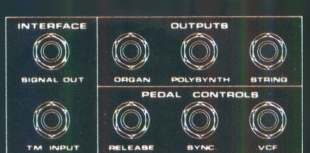
The general consensus throughout the music industry is that the Crumar Performer and Orchestrator produce the finest string sounds on the market. This same technology is used in the **Trilogy.** There are two selectable footages, 8' and 16', and separate volume, timbre, attack and release controls.

CATHEDRAL ORGAN

Anyone who's kept current on the music scene has obviously noticed the resurgence of the organ as a prime component. That's why we incorporated a traditional pipe organ sound into the **Trilogy** as a separate feature. There are four different footages, and a volume control for the section. The four footages can actually be combined to generate different waveforms. Each footage is a square wave. When these squarewaves are mixed in the right proportion, new waveforms are produced. This gives you an array of possible sonic foundations upon which you can add strings and polyphonic synthesizer.

OUTPUTS: AN EXPANDED REAR PANEL

On the back of the **Trilogy,** you'll find separate outputs for polyphonic synthesizer, organ, strings, and a special one called Signal Out, designed for connection to any effects or processing devices. Collectively, these outputs offer the musician maximum flexibility for setup (Stereo or Quad systems), with the added ability to have individual equalization and signal processing per output.



See your local Crumar dealer or write to: Music Technology, Inc., 105 Fifth Ave., Garden City Park, N.Y. 11040

re-view

selves away.

Inside the record cover was (as if in explanation) a favorable review from Cadence magazine, but I confess I couldn't have assembled this disc from that recipe. I have successfully "gotten into" some free jazz before, but there's just nothing here on which to hang my hat. I will admit that this disc afforded me several minutes of entertainment in trying to figure out exactly what was being played for this review, but I was also nagged with the question of WHY they did it. Perhaps I missed the point of the music, or perhaps there isn't any, but my gut reaction is that this emperor's dishabille.

Asmus Tietchens **Biotop** (Sky 057). Then there are the records that you ultimately like, but for reasons which are difficult to pin down. Biotop is not a groundbreaking album, not even an unusually intelligent one. But there's something about the spirit behind it.

Biotop consists of 16 approximately two-and-a-half minute instrumentals. Helping out Tietchens (on "elektronik") are three other synthesists, a rhythm box operator, a flanger operator and a ring modulator operator. Perhaps consequently, all of the sounds on the album - and there are a great number - are unique. Some are positively brilliant, especially a lot of the electronic percussion which sounds like no acoustic percussion ever invented.

But the way in which these sounds are organized - the music - is the real selling point. The tracks themselves are not really earth-shattering - just little 5 note patterns over electronic percussion and 3 or 4 backing synthesizer tracks. But each is pretty vastly different from the one before it, and each is carefully crafted to blend a variety of very different sounds. Together they make up a manifesto for diversity in metallic electronic new wave.

Except this isn't really new wave, either. In spite of the metal beat sound, the tracks are mostly contemplative and unhurried, sounding closer to the Eno/-Cluster collaborations, though shorter.

Like I said, the album does not lean in any one direction to make my job easier. It tends to define its own territory, which I could name "dreamy metal beat" or "new wave academic". Whatever name you give it, it would be an interesting effort if only for the fact that it's not easily classified. The fascinating sounds (and eye-popping hot pink and lime-green cover) are an unexpected bonus.

Ixna **Mi Ne Parolas/Ixna Portal Exo** (Dumb PJ101). This is a single sung in Esperanto (the "universal language" that never quite caught on) on side one and a made-up language on side two. It proves once again that lyrics are rarely heard for their meaning, and do not suffer at all from being essentially meaningless (most are anyway).

The imagination displayed in the lyrics carries over into the music, too. "Mi Ne Parolas" is given sort of a surf-song treatment, with atypically ponderous drum line supporting a bass, guitar, and synthesizer (or is it double-speed guitar?). At the center of the song, where Jan & Dean or the brothers Wilson would insert a guitar solo, Ixna have collaged multiple tracks of the immediately-identifiable guitar opening of the surfer's anthem, "Surfing USA". Far from being a parody or venomous putdown, "Mi Ne Parolas" is an affectionate tribute, capturing all of the fun-loving innocence of surf-music in a modern package. With lyrics to match.

The flip side, "Ixna Portal Exo", is supposed to be in a "counterfeit language", but with lines like "Cookie frei bin chocolate/shonuf two" you get the feeling that it really does mean something. It reminds me of the Russian/English Nadsat language in "A Clockwork Orange" or the future English in Russell Hoban's "Ridley Walker". But pondering the lyrics is only half the fun.

This side is constructed of a S/H synthesizer bassline, a couple warning/air raid sirens, a bell tree and, in the last half, a bass guitar. It sounds, as you might guess, slightly apocalyptic, but the lyrics and their varied delivery (whispered, sung, recited) makes it fascinating.

Ixna is made up of Marina LaPalma (singing and composition) and Jay Cloidt, and unless I miss my bet we'll be hearing more from them in the future. Dumb Records may not be widely distributed; their address is 625 Post Street Suite 129, San Francisco, CA 94109. e

Series-Para))el Details: ((Sum-Difference

by Dennis Bohn

Most musicians know that effects boxes sound different when you hook them up in series or in parallel; fewer musicians are aware that parallel effects sound different depending on whether you add or subtract their outputs. Two phase shifters, for example, produce dramatically different effects depending on whether they're in series or parallel, or summed/differenced in the parallel mode. The problem, of course, is figuring out some universal electronic device that lets you experiment with these different combinations.

Former DEVICE readers may recall one solution, written up in an article by Craig entitled "Generalized Series/Parallel Switching" (V1, #7, page 7), where he showed how to wire a DPDT switch, in conjunction with an inverting summing amplifier, to create a series/parallel switching box. For convenience, and as a jumping off point for this article, Figure 1 shows his original schematic. The circuit is pretty straightforward and quite useful. In the parallel position, the outputs of both effect units are summed together by the op amp mixer, and in the series position the output of effect 1 feeds effect 2 which is then passed, with unity gain, by the mixer.

Two and one-half things bothered me about this circuit. First, there is a 6 dB level difference between the parallel and series positions. Second, the output is inverted in either switch position. The inversion is easy to see, since the output summer is an inverting op amp. The 6 dB difference can best be seen by assuming a +1V input level and also assuming that each box has unity gain; therefore, there is +1V at each of the two 10k Ohm resistors. So, the output of the inverting summing amplifier is -2V - twice as much as went in, or an increase of 6 dB. Now, in the series position, the +1V input comes out of effect 1 as +1V and back into effect 2, where it again emerges as +1V. This is inverted by the mixer to produce a final output of -1V; hence, a level difference between the two modes of 6 dB. The half-thing that bothered me is that the figure 1 circuit cannot be operated remotely by a DC voltage; i.e., it is not electronically switched.

Okay, so I got to thinking about how to create a switching box that doesn't have these objections, and the following is the result.

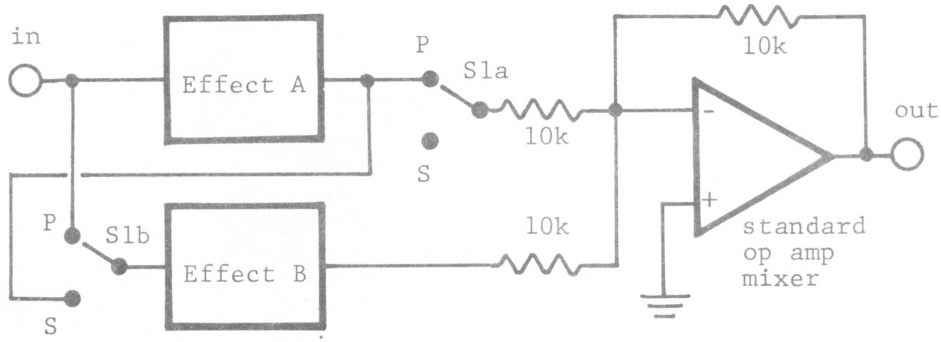


Fig. 1 Generalized series/parallel switching

Electronic switching for the millions. There is a lot to be said about electronic switching. But I am not going to say it. At least, not here. Maybe, if there is sufficient interest, I will do a DETAILS column on the various methods of electronic switching in the future. For now, only one method will be introduced, and used to generate the series/parallel box.

The most effective method of electronic switching, that achieves the best performance and specifications, is to take a field effect transistor (FET) and put it into the summing junction of an inverting op amp as shown in figure 2. As drawn, this is a junction FET, or JFET, but it could also be a MOS or CMOS device and work just as well (for this project, we'll be using CMOS). The sex of the FET makes no difference either; p- or n-channel, it doesn't matter.

The key to success in applying FETs as switches in audio circuits is to keep the drain to source voltage as small as possible. This is why the FET is located in the summing mode of an inverting amplifier. When the FET is off, it represents an arbitrarily large resistor, say, 10^{10} Ohms - in effect, an open circuit. When the FET is on, it has less than 200 Ohms across it. Let's say that the input resistor, R2, of figure 2 is 100k Ohms, and that the input signal is +1V; then, there is only 2 mV across the FET when it is on, if its on resistance is 200 Ohms. This completely satisfies out requirement of keeping the voltage across drain

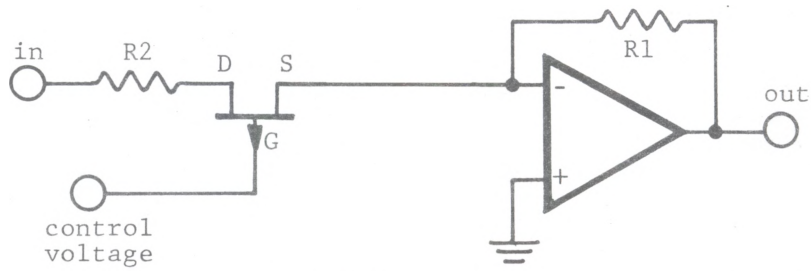


Fig. 2 FET switching in summing junction

to source at a minimum. Operation of the switch is simple. Apply a positive voltage to the gate to turn it off, and ground the gate to turn it on. For CMOS, apply a positive voltage to turn it on, and a negative voltage to turn it off.

CMOS transmission gate. Take two complementary MOS devices, one p-channel and one n-channel, glue them face-to-face, tie their tails together through an inverter and you get a transmission gate as shown in figure 3. This is the cheapest form of analog switch available today, being less than $0.25 per switch. The most popular versions are the 4016 and 4066 ICs made by just about every semiconductor house in the business. There are four switches in each package, and they are pin compatible. They differ only in that the 4066 has lower on-resistance and less resistance modulation due to input voltage variation. (If used in a summing mode, the 4066 offers no improvement over the cheaper 4016.) The idea of a transmission gate is to create a bilateral switch, namely one that will pass signals in either direction. This is why complementary devices are used and tied together. Also, this reduces the on resistance by one-half, since they are in parallel.

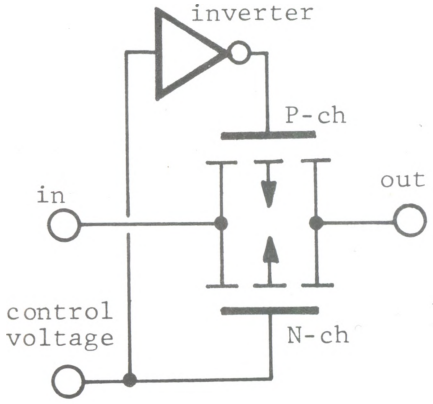


Fig. 3 CMOS transmission gate

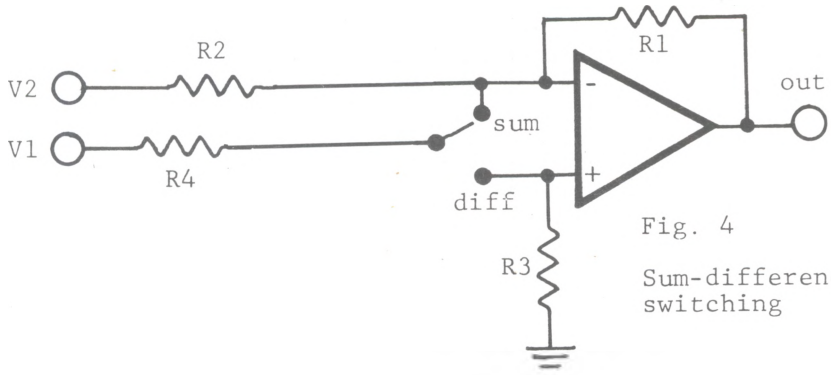


Fig. 4
Sum-difference switching

Sum-difference switch. One more diversion, then I will tie all this together into our present project. I hope.

Figure 4 shows an interesting use of an SPDT switch. In the SUM position, the output is the sum of V1 and V2, and in the DIFF position the output is the difference between V1 and V2. Thus, if V1 and V2 represent the outputs of two effect boxes, you can either add or subtract them at the throw of a switch. Notice that the switch is in the summing mode. Smell it coming?

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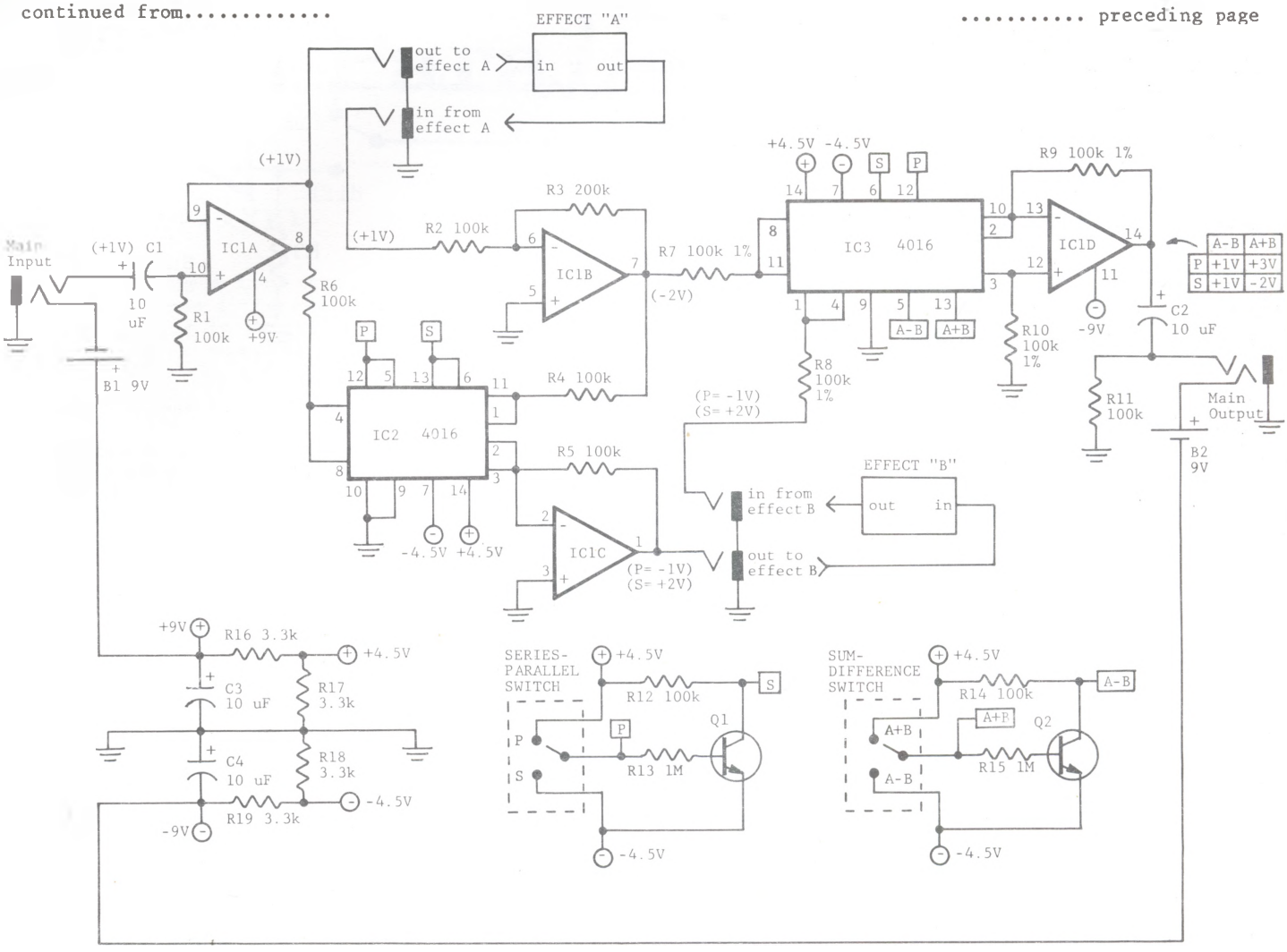


Figure 5.

Series-parallel/sum-difference box. Figure 5 shows the schematic of a series-parallel/sum-difference construction project that is both fun to build, and convenient to have around for creating new and unusual effects. It is similar to Craig's switch box of figure 1, except it brings some new tricks to the party. All switching is electronic, and it eliminates the inversion and 6 dB level difference. In addition, the user has the option in the parallel mode of either adding or subtracting the outputs. All that is required is one quad op amp and two 4016 quad switch packages. And, before you write, a printed circuit board is available (see the parts list).

The simplest way to understand this circuit is to walk through it in each of its various modes. To make this easier, I have put voltages in parenthesis at several key spots in the circuit. These are not operating voltages; they are the result of an assumed +1V sine wave input. But, before we walk, let's look around a bit.

There are two external switches (either panel mount or footswitches) which control the DC levels routed to the 4016 switches. I have labelled these DC levels, P and S for parallel and series, and A+B and A-B for sum and difference. If the voltage is positive, then the function is active; if negative, then it is not active. So, for example, P = +4.5V means that the unit is in parallel operation. Positive voltage means a switch is on; negative voltage means a switch is off. Signals go through "on" switches only.

So much for basics, let's walk. Remember as you look at the schematic that all points labelled "P", "S", "A+B", and "A-B" connect together. Ditto the +9V, -9V, +4.5V, and -4.5V connections.

Assume we have selected "parallel" and "difference" modes. The +1V input signal is buffered by IC1A and fed out to effect box "A", where it returns as +1V to the input of IC1B which is an inverting amplifier with a gain of -2. The output at pin 7 is, therefore, -2V. Since both switches of IC2 with "P" on their gates are on, and those with "S" are off, the -2V signal is blocked from IC1C and only goes to IC1D. Meanwhile, the +1V input signal goes through IC1C and comes out as -1V, since it is an inverting amplifier with a gain of -1. After going through effect box "B", it returns as a -1V input signal to IC1D via resistor R8. Since we have selected the "difference" mode, then the "A-B" switch is on and the "A+B" switch is off; this means IC1D is configured as a difference amp whose output is the difference between -1V and -2V, or +1V, since -1V - (-2V) = +1V. So, in parallel, +1V input equals +1V output.

Now, we select "series" and "difference" modes. Our +1V input still presents -2V at resistors R4 and R7, but this time the +1V at resistor R6 is blocked

and the output of IC1C becomes +2V, due to the -2V signal at R4. The +2V comes back in from effect box "B" to resistor R8. IC1D is still configured as a difference amp, except that R7 is grounded, preventing the -2V signal from going anywhere. Resistors R8 and R10 now form a voltage divider such that +1V is applied to the positive input port. IC1D simply acts as a non-inverting, unity gain buffer to produce +1V output. Therefore, in series, +1V input equals +1V output - as in parallel. So, whether in series, or parallel, the output is unity and non-inverting. Hooray!

Now, in addition to the above, when in the "parallel" mode, we can also choose "sum". With "sum" selected, the "A+B" switch is on, and the "A-B" switch is off. So the output is the inverted sum of -1V and -2V, or +3V.

(If "series" and "sum" are simultaneously selected, IC1D reverts to an inverting unity gain amplifier and the output is -2V. While it may not make conceptual sense, it does work and can be useful as an additional 6 dB of gain and inversion stage, if desired.)

The table shown near IC1D summarizes the various modes and respective outputs for quick reference.

Construction details. The printed circuit board makes things quite easy, but is not required. The circuit can be hard-wired using perf board or similar material. Layout is not overly critical as long as leads are kept short and neat. Neatness is everything in good construction jobs. Use good soldering techniques and clean the flux afterward.

I used two 9 Volt batteries to power the circuit, letting the stereo input/output jacks act as power switches (you would use these with mono plugs, of course). However, there are always pros and cons regarding battery use. They are certainly not required. The PC board is laid out to accept plus, minus, and ground wires. These could be battery wires or DC supply leads. The circuit works fine when powered from a standard ±15V supply, but **do not use more than ±18V or you will damage the ICs.** On the other extreme, do not use less than ±4V. Within this window, everything will work fine.

I used a simple folded sheet metal box as a chassis with panel mounted switches. The most useful configuration for the switches is to make them footswitches and add two more stereo jacks to accommodate them. Of course, that is the fun of DIY -it is your project, so be sure to put some of you in it.

Bye until next issue.

SPECIFICATIONS

Input resistance
- Main & input A: 100k Ohms
- Input B (sum): 100k Ohms
- Input B (diff): 200k Ohms

Output resistance
- All outputs: less than 1 Ohm

THD+N (20 Hz - 20 KHz): less than 0.02%

S/N ratio: -82 dBV re 1.0V output level (A-weighted)

PARTS LIST

B1, B2	9V battery
C1 - C4	10 uF, 16V electrolytic
IC1	TL074 quad op amp
IC2, IC3	CD4016 quad bilateral switch
Q1, Q2	2N5210 or equivalent NPN transistor

The following, unless otherwise stated, are 1/4 Watt, 5%, carbon-composition or carbon-film resistors:

R1, R2, R4-R6, R11, R12, R14	100k Ohm
R7 - R10	100k, 1% metal-film
R13, R15	1 Meg Ohm
R16 - R19	3.3k Ohm

Misc. (4) mono 1/4" phone jacks, (2) stereo 1/4" phone jacks, (2) SPDT switches, (3) 14 pin IC sockets, battery holders, suitable enclosure, wire, solder, hardware, etc.

Parts kit available: The following is available from TOLECO Systems, Box 401, Kingston, WA 98346: Complete kit of parts (less batteries and all "misc" items), plus plated, drilled glass-epoxy PC board, for $14.00 plus $1 postage and handling in U.S. and Canada. Order kit #SW-1. Also available separately is the etched, drilled, and plated glass-epoxy printed circuit board, #SW-2, for $6.00 postpaid in the U.S. and Canada. Washington state residents add 5.3% sales tax. Sorry, no COD orders. e

Applied Synthesis: Strings

by Bill Rhodes

The electronic imitation of strings has developed to the point where today, it is almost commonplace for a keyboard player to have this "ensemble" sound at his or her fingertips. Many synthesizer companies offer string units, either as constituent parts of larger units or as separate machines. Their variation in sound establishes their own particular electronic identity; some keyboards (such as the Crumar Orchestrator or Elka Rhapsody) sound more electronic than, say, the new Korg Trident or Delta synthesizers. As with the human voice, and in this case even a violinist, no two personalities are exactly alike.

The reason why one commercially available string synthesizer sounds different from the others occurs because of many factors. The "lush" string sound usually is associated with the machine's chorusing facilities. The chorus feature gives the effect of two or more oscillators playing "desinfanado" or slightly out of tune. The wider the frequency between the two tones, the greater the fullness of modulation; however, too much chorusing will make intervallic relationships between the oscillators macrotonic instead of microtonic. Some units use bucket brigade circuits to add delay upon delay to the sound, while other keyboards depend on the cascading effect of multiple chorusing oscillators to create richness.

There are many parameters we need to look at when trying to obtain convincing string sounds. If you want to record a classical string quartet sound, consider playing one track or one voice at a time and eliminating multiple oscillators. Pay careful attention to your phrasing and vibrato as well. But if you want a full, heavily chorused orchestral neo-classical rock sound (i.e. Yes, UK, etc.), play polyphonically and layer as many textures as the situation warrants. In any case, the many parameters of the synthesis must be taken into consideration in order to make sounds that are symphonic and not "sinphonic".

Parameter 1: Attack and Decay. The full ADSR mode on the synthesizer is not needed to create good string sounds. Use the attack and release modes (as well as a full sustain for triggering) according to the phrasing you desire. Pizzicato strings, for example, are very short in attack (staccato), while decay is set according to the musical context. Multiple triggering is nice, but not always available on most units. Try to listen to the string players, classical and pop, and their characteristic phrasings.

Sustaining releases are good when you want to create reverberent-type string ambiences. Vangelis uses a considerable amount of reverb and release to create very nice lush string sonorities in his music. He is a great one to listen to (as well as Larry Fast and Carlos) for ideas in string syn-

thesis.

The attack characteristic depends totally on the type of string sound you need. The Korg Trident Poly-Synth has an interesting "bowing" circuit that retriggers your envelope while you play; its overall action depends on the setting of an associated "sensitivity" control. Remember, a real string section does not have each player playing with the same attack characteristics. If you have several envelopes of volume (VCA) and filter (VCF) used together, you'll hear a very human and intriguing sound. Experiment.

Parameter 2: Waveform. We all know waveforms are the life blood of a synthesizer's imitative abilities. The sawtooth wave is generally most desirable for strings, but a combination of pulse wave (and pulse width modulation) and other waveforms can simulate the resonant idiosyncracies of wood. A real violin is not a keyboard, VCO, LFO, VCA, and VCF, but it does have complex waveforms that can be reproduced electronically. At some point to aid your synthesis, you might want to invest in an oscilloscope; this piece of test equipment enables you to "see" various waveforms and their importance in shaping sound. Also, registration of an instrument (the location of the notes high or low on the staff) determines the waveform that is heard. The higher "harmonic" single violin voices are thin and more akin to a sine or triangle wave, while the low cello or double-bass sounds are darker with a higher harmonic content. Combining these harmonics and waveforms creates the orchestral sound that we identify as strings playing "en ensemble".

Parameter 3: Filter Settings. The amount of low pass filtering determines the brightness of the strings. Most commercial units have a control for timbre to create an overall equalization (EQ) of the sound. The Korg Delta and Trident have bass and treble EQ, while the Crumar Orchestrator has an overall timbre control, the Crumar Performer has a three band tone control, the Moog Opus has a filtering control, the Rolands have brightness or emphasis controls, and so on. Avoid nasal sounding strings, unless you want on oboe to be present in the ranks somewhere; piercing string sounds are also undesirable. Generally a flat EQ setting works best in the studio, with a touch of Harmonizertm or delay to animate the upper end of the sound. Playing live is a different story, depending on room acoustics, delay, reflections, auxiliary signal processing, and the like. Additional chorusing or doubling via electronic means can also be useful at times.

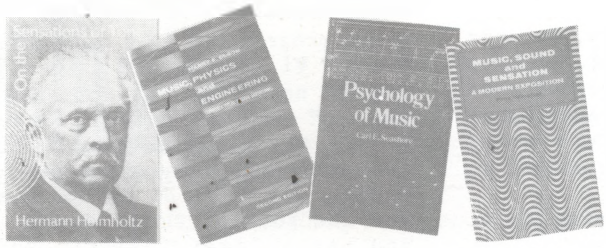
Parameter 4: Playing style. Do not play the string synthesizer like an organ. Try to voice chords openly, allowing rather wide intervals in the notes of your left hand. Block type chords do not sound orchestrated. The interval of a tenth in the left hand is very useful. The lowest note can be thought of as the bass, while the upper tenth (or third an 8va away) acts as a cello-like member of your synthetic orchestra. Chord clusters can work, but remember that simplicity is just as effective. Long sustaining single notes over the measures (ligatures) work just as well in synthetic situations as in the real string world. Animate both your hands (piano exercises and scales will help drastically) as if they are both imitating the bow bouncing or phrasing qualities of the string orchestra. Use the notes that different chords have in common to sustain high string sounds over the harmonies of other instruments. For example, if a chord progression goes G, D, Em, B (G: I V Vi III#3), use a sustaining "b" note over the progression. The "b" is the third of the G triad, the sixth of the D chord (which is slightly dissonant but works), the fifth of the Em chord, and the root of the B chord. If you use a lower register sound such as a cello, use the "G" as a pedal throughout the same progression. Other procedures and examples of string technique will be explained in my forthcoming book, due to be published by Christmas. Right now we are just scratching (add some rosin!) the surface.

Parameter 5: Practice. You and your knowledge of string arrangement are the most important factor in string synthesis. Research and analyze the ranges of cello, violin, viola, etc. What kinds of figures do they play? What is their voicing in the arrangement? What are their timbral characteristics (waveform/filtration)? Hopefully with that knowledge and the help from this column you'll turn into a symphonic keyboard player - with no strings attached.

Well, everfriends, until next issue... e

POLYMART

SCIENCE



The physical and psycho-acoustical background to music is an important part of musical synthesis. Helmholtz's **Sensations of Tone** is, a century after its publication, still the standard text for physiological acoustics. **Psychology of Music** by Carl Seashore, developer of the Seashore Music Test, provides an in-depth analysis of musical style and performance characteristics of many instruments. **Music, Physics, and Engineering** by Harry Olson, who worked on the first RCA synthesizer, is a thorough discussion of the physical properties and design of traditional musical instruments (plus a chapter on electronic music). **Music, Sound and Sensation** by Winckel is much like the Helmholtz work, with a bit less detail and more concentration on psycho-acoustics.

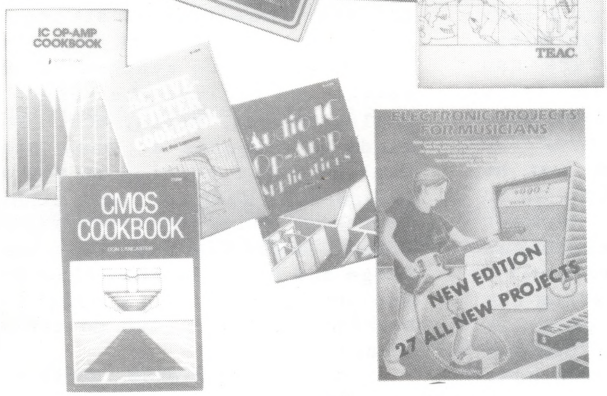
#SENS On The Sensation Of Tone	**$7.50**	**#PSYCH** Psychology of Music	**$5.00**
#MPE Music, Physics & Engineering	**$6.00**	**#MSS** Music, Sound and Sensation	**$3.50**

TECHNIQUE

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 easily. **How to Make Electronic Music** by Drake, Herder and Modugno is a standard introductory text for music systhesis classes, with chapters on equipment, tape technique, composition projects, and more. **Multitrack Primer** by Teac is a step-by-step guide to building, outfitting, and operating your home studio. The Byte book of **Computer Music** describes computer control of electro-mechanical instruments, Fourier analysis, circuits and loads of software. **Home Recording For Musicians** is the original guide to outfitting and operating a budget studio for maximum results, including mixer and audio processing circuits and a demo recording.

#HMEM How to Make Electronic Music	**$3.95**	**#TEAC** Multi-Track Primer	**$4.95**
#BYTE Byte Book of Computer Music	**$10.00**	**#HRFM** Home Recording	**$9.95**

ELECTRONICS



Electronic cookbooks are a great way to stock your library with materials that are not only heavy on theory, definitions, and educational material, but chock full of practical applications as well! These books can easily replace stacks of manufacturers data sheets and applications notes all in an easy to use reference. Walt Jung's **Op-Amp** and Don Lancaster's **Active Filter** books are self-explanatory -- required reading for synthesists! **Audio Op-Amp** is an edited version of OACB, containing only audio applications. Lancaster's **CMOS** book is much more than a digital reference -- phase lock loops, top octave generators, touch switches, and other things you need! **Electronic Projects** discusses electronic construction technique for the novice and provides **27** projects with PC patterns and a demo recording of the effects.

#OACB Op-Amp Cookbook	**$ 15.95**	**#CMCB** CMOS Cookbook	**$ 12.95**
#AFCB Active Filter Cookbook	**$14.95**	**#EPFM** Electronic Projects **2nd edition**	**$14.95**
#AUOA Audio Op-Amp Applications	**$8.95**		

REFERENCE

Often used reference materials to answer the many questions encountered in everyday synthesis(?) ... **The Source** is over 125 pages of patches in universal flow chart notation; the largest publication of its type. **Audio Cyclopedia** has 1760 pages with 3650 entries and hundreds of drawings and schematics to answer any questions about audio. **Polyphony Binders** hold up to 12 issues of any 8½" x 11" publication without punching holes; keeps issues like new for unending service. **Electronic Music Synthesizers** devotes the first half of features and functions of commercial equipment (Moog, Arp, Paia, Oberheim, EML, and RMI); the second section provides schematics and projects for the experimenter.

#SOURCE	The Source	**$4.00**
#CYCLO	Audio Cyclopedia	**$44.95**
#EMS	Music Synthesizers	**$6.95**
#BIND	Binders	**$4.95**

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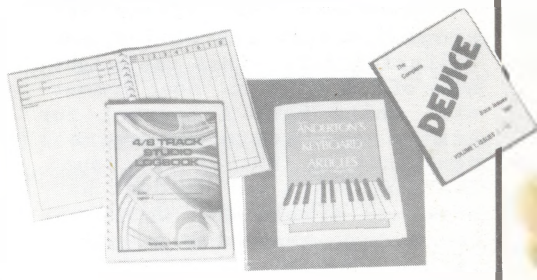
4/8 Track Studio Log Book provides a place to keep all the important information on your tape library. Log in timing, type of tape used, record patches, make notes and use the expanded track sheet to list sequential changes in tape tracks relating to the settings of the index counter. Craig **Anderton's** Contemporary **Keyboard Articles** is a reprint of all the articles from June 1977 through February 1981, covers tips, technique, theory, maintenance, and numerous construction projects. **Device Back Issues** — during the year that this newsletter was published, it featured almost 200 pages of technical information for the guitarist/musician. A wealth of articles on; design, product reviews, and modification and construction projects. Sold in complete set, individual issues not available. Limited number available, order yours now.

#SLB	Studio Log Book	**$4.95**
#AA	Craig Anderton's Keyboard Articles	**$5.95**
#DEVICE	Complete set (12) Device Back Issues	**$18.00**



RECORDINGS

#CORDS	Cords - Synergy	**$7.95**
#ERRO	Electronic Realizations - Synergy	**$7.95**
#SEQU	Sequencer - Synergy	**$7.95**
#GAMES	Games - Synergy	**$7.95**
#ELEM	Elements - Mychael Danna	**$6.95**
#CAMT	Craig Anderton Music Tape	**$5.95**
#ARPT	Music For Airports - Brian Eno	**$7.95**
#DELTA	Rainbow Delta - Pat Gleeson	**$7.95**

BACK ISSUES

The wide variety of practical applications and construction projects in past issues makes a binder full of POLYPHONYs a frequently used reference to keep near your synthesizer , home studio or workbench. Most back issues are still available for $2.50 each postpaid. Check the issues you wish to order on this coupon and add the total to your POLYMART order.(other side), or order by volume and issue number (0304,0402, etc.) on the POLYMART order form.

☐ #0204: 4/76: music notation - timing, external inputs for Gnome, Programmable Drums, Equally Tempered D/A, low cost AR project, digitally encoding keyboards, patches, Volume 1 & 2 index.

☐ #0301: 7/77: frequency divider project, random tone generator project, normalizing synthesizer controls, eliminating patch cords, computer control of analog modules, Chord Egg modification, adding pitch bending, patches.

☐ #0302: 11/77: The Sensuous Envelope Follower, digital gates, LED wall art, build a bionic sax, data to music peripheral project, Apple II as a music controller, using the NE566 as a VCO, patches.

☐ #0303: 2/78: computer controled Gnome, using joysticks, build a bionic trumpet, ultra-VCO modifications, voltage control the Mu-Tron Bi-Phase, oral joystick, patches.

☐ #0304: April/May 78: Minimoog modifications, non-keyboard module use, phasing and flanging (theory and circuits), memory expansion for programmable drums, digitally addressed transposer project, polyphonic software (with software transient generators), patches, Volume 3 index.

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

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

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

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

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

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

☐ #0602: July/August 80: Peter Gabriel, digital VCO project, dream modules, optimum level settings, dynamic phrasing, patches.

☐ #0603: Sept/Oct combined with Nov/Dec 80: alternate controllers, add voices to Casio M-10, voltage controlled quadrature oscillator project, cordless patch bay, recording rules, patches.

☐ #0604: January/February 81: Special Construction Edition; Build: Audio Circuit Breaker, Pulse Width Multiplier, Magnetic Harp, 50 Watt/Channel Stereo Power Amp, Quad Sequential Switch, DOD Mods, patches.

☐ #0605: March/April 81: Portable Music Issue, reviews of Remco's FX, E-H Mini-synthesizer, Casio's VL-Tone, plus mods for the M-10, GR-500, mini-amp, and the Korg X-911. Introducing; Practical Circuitry and On Location, new columns.

☐ #0606: May/June 81: Synthesizer: Hardware Mods and Software. Modular Synthesizer Effects, Environmental music, Keyboard assignment for the 8700, new columns; Details, Practical Circuitry, and On Location. Volume 6 index.

☐ #0701: July/August: Guitar Electronics: Modify; Fender Amp, MXR Phase 100, GR-500. Input/Output Structures, $5 Analog Programmer, Sample and Hold technique, Modular Synthesizer Effects, new column: Applied Synthesis, Marketing Your Records.

☐ #0702: Sept./Oct. '81: Harald Bode Interview, Live Plus Tape - New Technique, Xenharmonics, Kraftwerk Live - Review, Psycho-Acoustic Experiments, Practical Circuitry - Super Controller, Applied Synthesis - Brass, Construction Tips for Beginners.

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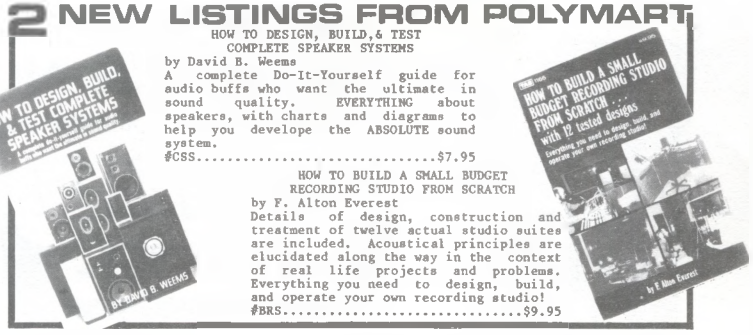

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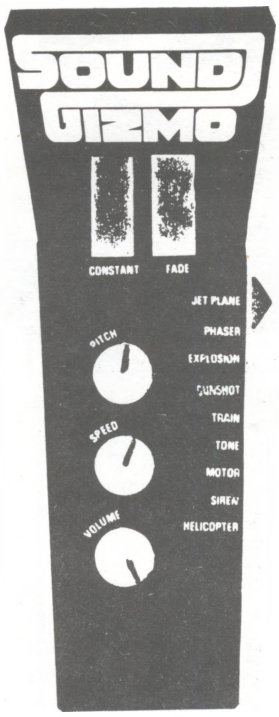
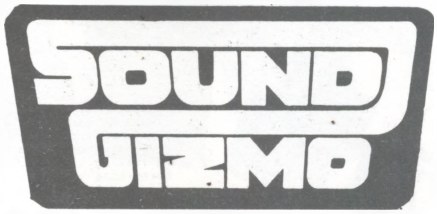
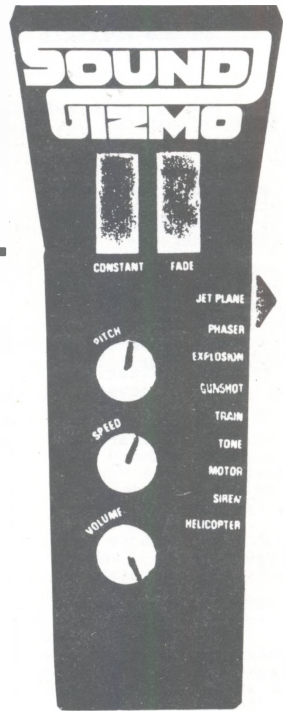


THE POP MUSICIANS' VERY PORTABLE NOISEMAKER

* * *





BY ANDREW GELT

This is my fourth published article on electronic musical toys; these have covered subjects from educational applications to sobriety testing. As my primary interest lies in eclecticism in music, and the composition thereof, perhaps it would be a good idea to stop and examine the value of this new and seemingly frivolous pursuit.

First, large scale integration has allowed a new family of low cost, high performance products. These small sound toys and musical games exhibit a relatively current state of the art in electronics that provides a glimpse into a future of increased miniaturization, as well as decreased expense. And, what will happen when the new VLSI - very large scale integration - technology becomes equally prevalent?

Secondly, "...perhaps the most important aspect of portable music has nothing to do with technology, but instead with the social implications..." "Clearly, portable music isn't just music; it's people, a social thing." (Craig Anderton, "Editor's Note", March/April 1981 issue of Polyphony.) These devices show us that music composition and electronic synthesis is now universally accessible, and for everyone.

As there are several more articles planned on this subject, I hope that these short statements will reaffirm that the value of these new devices is both intrinsic, and indeed, multifaceted.

The "Sound Gizmo" (trademark of Fundimensions) is about the size of a small walkie-talkie and is made of very durable plastic (almost "kid-proof"). The back panel has a large metal insert (with Morse code printed on it) and despite the approximate $20 price, there appears to have been no deliberate attempt to compromise or cut costs.

The controls of actuation are labelled "Constant" and "Fade". "Constant" is merely an on-off switch which allows any of the presets to sound when depressed and held. The "Fade" button, when depressed and held, causes any of the sounds to begin and then fade out entirely. In effect, "Fade" is an envelope generator (controlling a VCA) of the AR type, where A = 0 seconds and R = 1.5 seconds. Both "Constant" and "Fade" can be depressed simultaneously (with "Constant" overriding) to produce a sustained sound, then "Constant" can be released (while still depressing "Fade") to activate a release. In this respect an envelope generation of an "ASR" type can be produced where A is still 0 seconds, S is variable, and R = 1.5 seconds. Perhaps due to the current consumption of the built-in power amp/speaker combination, the "Fade" paddle causes a marked pitch fluctuation in the primary oscillator (when engaged) during the initial portion of the envelope generation. This is not totally disadvantageous, however, as it provides for another different yet totally predictable effect.

Figure 1 shows the modules and controls which are in operation for each of the main preset sounds. Additionally, the product's accompanying pamphlet has instructions on how to produce the following "secondary sounds": Snare Drum, Bird Call, Rocket Launch, Steam Release, Machine Gun, Electronic Shaver, Busy Signal, and Airboat. These are essentially the main preset sounds with different control positions.

As the battle between performance versus modular synthesis continues, the "Sound Gizmo" may be thought of as performance oriented (as opposed to, say, the Remco FX's more modular orientation). With its quickly changeable presets, it can be carried to a rock job and used to make the somewhat impeachably placed sound effects heard in rap music and similar musical styles. This is a significant and usable first.

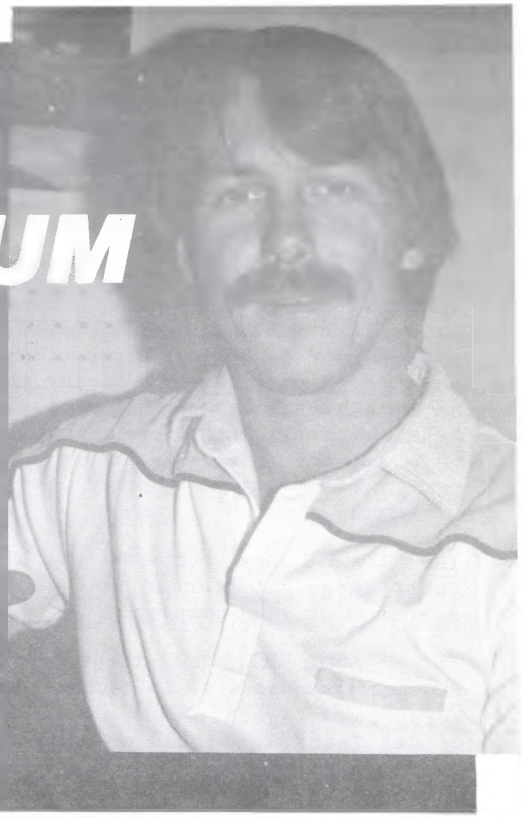
Parents will also be thankful for the inclusion of a volume control.

The "Sound Gizmo" is operated by one nine Volt battery and is an excellent product. And remember - it's priced around $20!

*The sub-audio Square Wave Modulator is used as the sound oscillator and can be attenuated to low audio sound with the "Speed" Control.

	PRIMARY OSCILLATOR (triangle wave?)	SAWTOOTH MODULATOR	SQUARE MODULATOR	SINE MODULATOR	SPEED CONTROL	PITCH CONTROL	WHITE NOISE	PINK NOISE	VOLUME
JET PLANE	●					●		●	●
PHASER	●	●			●	●			●
EXPLOSION								●	●
GUN SHOT							●		●
TRAIN			●		●		●		●
TONE	●					●			●
MOTOR			*		●			●	●
SIREN	●			●	●	●			●
HELICOPTER	●	●			●	●	●		●

INTERVIEW: DAVE ROSSUM



This month's interview is with Dave Rossum of E-mu Systems, Inc. Jay sent the following introductory comments along with the interview:

"My first encounter with Dave Rossum was over the telephone some nine years ago, when he prevented me from purchasing sight-unseen thousands of dollars worth of E-mu equipment. Dave instead encouraged me to purchase a few submodules and test them out. I did, and I was hooked.

Dave is an original. His E-mu modular system has set the standard by which all other modular systems can be judged. Brilliant and witty, Dave is a new breed of electronic music pioneer whose contribution to the design of equipment for other manufacturers may have shaped the nature of the synthesizer as much as did Robert Moog's original thinking."

by Jay Lee

Jay Lee: Isn't it true that your formal training was as a biologist?

Dave Rossum: Yes, that's pretty much true. I did my undergraduate work at Cal Tech in Pasadena starting out as a chemist, then changed to physics, and then (about the end of my sophomore year) changed again to biology because it seemed that the neatest people were in biology. In the end it didn't really matter what I studied; I guess my career shows that. I graduated in 1970 with a Bachelor of Science in biology, and went to University of California at Santa Cruz for graduate school.

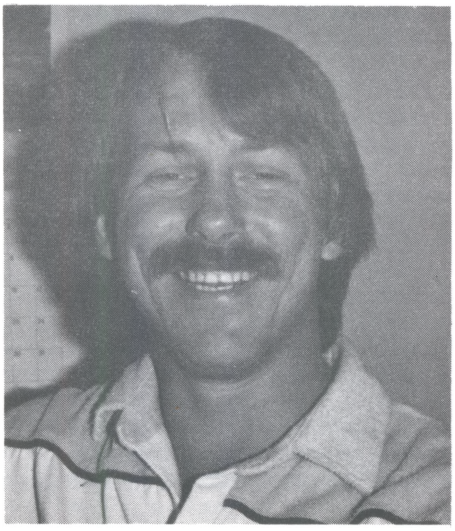
JL: Who, or what, changed the direction of your career?

DR: At U. C. Santa Cruz I worked on the biochemistry of ribosomes, and my adviser was a man named Harry Noller. One day I came into the lab and he asked "Have you ever seen a Moog synthesizer?", and I said no. He mentioned that they were unpacking a Moog 12 (predecessor of the Moog 15) over at the music department, and would probably need some help. As people there started playing it, I realized that I intuitively knew as much about it as anybody else in the room...it just seemed natural to me. I helped one of the professors there, Eric Regener, in teaching students how to use the instrument as well as encouraging them to experiment with the synthesizer. When people ask me how biology relates to electronic music, I simply reply that the whole world is negative feedback.

JL: Your modular E-mu system is considered by many to be the Rolls-Royce of analog/digital music equipment. How would you describe your system, or modular systems in general?

DR: Well, in fact you're wrong...many people say that Rolls-Royce is the E-mu of the automotive industry (laughs).

When we decided to design the modular system, we had to make a decision concerning the level of the instrument...would we make a cheap one, a fine one, or what. We decided to make the best we could possibly make; any time we found a component, a design, or anything that would make it better, we put it in unless it was just totally cost-prohibitive. And this attitude kind of shows. When we go around to demonstrate modular systems, we take prototype #1 - most people simply won't believe that the instrument is 10 years old. We clean the dirt off and wash it up before we show it, but that's about all we do.



The reason we can do this is because we use such parts as expensive controls and knobs. We pay maybe $1.50 a piece for the knobs on the synthesizer, because we want them to last. That kind of caring attitude comes through particularly with the panel components; we want everything to last 10 or 20 years, down to the 1/8 inch panels that won't bend when you push on them.

In terms of design, I've looked at a lot of other circuit designs, and I think that our approach is substantially more sophisticated than anyone else's. But the perfectionist attitude also has some drawbacks: We often get criticized because we update our circuitry, and someone will buy an E-mu system only to have us come up with a replacement submodule a year later that performs better than the original. We usually keep it pin-for-pin compatible to allow for upgrading, but

very often people do get upset because our instruments are constantly evolving and improving. I don't know what to do about that problem - if we find a better way to do something, then that's the way it has to be done. Even our decisions as to what products to build show this influence. We built modular systems first because with a big enough modular system, you can do anything. We have a huge variety of modules - if we think of something that can't be done with our system, we'll add the modules necessary to do it. For example, we have five different kinds of filters.

JL: How did your partnership with Scott Wedge get started, and who does what in your company?

HOME OF E-mu.

DR: After Harry Noller and Eric Regener introduced me to synthesizers, I got together with a group of Cal Tech people and we spent the summer of 1971 building a synthesizer. Towards the end of that summer, Scott Wedge showed up at our door in Santa Cruz - we had a sign out front that said "Starships and Synthesizers, since 1984". I'd known Scott since junior high school, although after I went away to college, we didn't see each other that much. Scott had hurt his back jumping out of airplanes and didn't have anything to do, so he came over to see what I was up to. Just as in my case, he became intrigued with synthesizers. After that summer, all the Cal Tech folks went back to various places, but Scott, myself, and my girlfriend at the time, Paula Butler, continued working in our spare time building a synthesizer. We called it the E-mu-25 (named after our favorite drug), and spent much time improving it and learning more about the best way to do things. Eventually we sold that first unit, and used the money we got from it to build a second one. We sold the second one, and used that money to start designing a modular system. By now it was November 1972, and we decided to form a real company. We became a general partnership, filed our papers, opened our bank accounts, and E-mu systems was formed. Back when it started, Scott worked full-time at E-mu. I worked at a little company called Santa Clara Systems as an engineer (which enabled me to earn enough money to buy parts), and would then spend another eight hours working at E-mu. Paula provided life support and cheerful company for us all; she also had a nine-to-five job, so it was just the three of us working at first and supporting ourselves with other jobs.

We bounced ideas off of each other, and nobody wore a single hat back then. When E-mu became incorporated in 1979, we tossed a coin and Scott lost, so he became president and I got to be vice-president and chief engineer. Since that time he's been taking over the financial and management end of things, but also attends to a number of engineering functions - he's great at digital and software stuff, and can hold his own in analog. I help out with management too sometimes, so we still share tasks, but Scott's less involved in the engineering than I am now.

JL: Your company started out humbly, coupled with a friendly philosophy. Has that changed over the last eight years, and if so, why?

DR: Well, I think we're all a little less humble than when we started; I don't think we're any less friendly, although changes in our lifestyles may make it appear that way to some poeple. I've sort of become a bit of a recluse. It's hard to get to talk to me, I don't come out and meet everybody that comes in the door anymore, and that's a change in my own lifestyle. In the beginning synthesizers were my life: I'd spend 16 hours a day on them, and also, my time was less structured. So, if somebody walked in the door, I'd chew the fat with them for an hour or two; now it matters whether or not something gets done by next week, so I don't have that freedom anymore. I also have more hobbies now that also take time. I mountain climb and teach scuba diving, and my life is real full so I can't allow myself so much time to relax with people. That doesn't mean I'm any less friendly, all it means is that you have to come scuba diving with me rather than talk synthesizers.

As far as the company philosophy goes, we very definitely hold an attitude that is lacking in other companies, namely, even if you don't want to buy something from us we'll tell you about what we do and try to help you out. It's not so much a selfless attitude as the fact that when we do people a favor, they'll talk to their friends or when they are ready to buy, they'll come back because they have a good feeling about us. We're still a small company, and I hope it stays that way. All of our employees are friends; we do other things than electronic music together. That's real important to me - I want to keep the company so there isn't any "elite" management department or so that engineering doesn't hide from the production people and so on. Internally, that's important to me...more important than the size of the company.

INTERVIEW: DAVE ROSSUM

JL: You were involved in the design of the first electronic music chips (SSM). To what extent were you involved in this?

DR: The story goes something like this. Back in 1975, Ron Dow dropped by E-mu and had this idea for a VCA design on a custom chip. He wanted us to kick in the $1000 or so it took to integrate it, because he didn't have the money. It sounded interesting, but then he said the power supplies couldn't be any higher than ±12V. Since the E-mu system was based on ±15 and I didn't want to change the whole system standards for one IC, the idea no longer made sense to me. So we turned Ron down and he then got connected with SSM, and they were the ones to integrate and start selling the SSM2000. About a year later I ran into Ron again, and he mentioned an improved VCA they were working on with ±15V supplies. With my major objection gone, I became interested again. I contributed some ideas to the SSM2010 (low distortion VCA); then Ron started frequenting E-mu and used our lab to develop the chips - first the 2020 voltage controlled amplifier, then the 2030 voltage controlled oscillator (which was a very hard project and took about a year), the 2040 voltage controlled filter which went pretty quickly as did the 2050 transient generator, and we've kept up the collaboration since then. My involvement varies from chip to chip; with the 2020 I helped Ron design the control stage, while he did the VCA cell entirely by himself. The oscillator was definitely a joint effort, while the 2040 and 2050 were largely my designs. All of the SSM chip vary like that. We've pretty well decided that we're co-designers on what we do, because you just can't separate where an idea came from. Ron might suggest something to me for a filter design, but I'll end up using the idea with something like a transient generator instead.

JL: Your influence in the design of both the Prophet 5 and the equipment of Oberheim Electronics have been grossly overlooked. Would you care to comment on what you've done for both companies?

DR: Let me talk first about Tom Oberheim, whom I met at an AES convention in the spring of 1974. We both liked each other immediately and started talking about ideas. He was talking about his phase shifter, and said something to the effect that he knew it could be done with a 3080 transconductance amp, but didn't know exactly how to do it. So I scribbled out a circuit and said here's how you do it. He said "I bet that's patentable", and I said "I'm sure it is". He asked how would I feel about patenting it and sharing the rights under the patent? I thought it was a great idea, and that was the beginning of our association. Incidentally, I don't think he ever used that design in anything he built.

Towards the fall of that same year he visited E-mu to see our prototype polyphonic keyboard, and decided that he liked that design too. He felt that it too was patentable, and wanted to use it in a product. So we worked out an arrangement whereby he'd pay us royalties, and could use the circuit, and we would share in the benefits of the patent. The arrangement worked well in the sense that it was profitable for E-mu; we were essentially putting an investment into Tom's company with our work, and the payments (in the form of royalties) came later. I know Tom was happy with the arrangement too.

The involvement with the Prophet 5 was very similar. Dave Smith had come to us for design help with other projects; when he decided to get into a synthesizer, he came over and basically started picking our brains from the beginning, which was an intelligent thing to do. We did some specific circuit designs for the Prophet 5, reviewed virtually everything in it, and gave him access to lots of E-mu documenation. Again, we had a royalty arrangement, the product went over well, and we made a lot of money...but of course had the Prophet not gone over, we wouldn't have gotten anything for our time so essentially, we again invested in the company. Still, royalties are not a trouble-free way to conduct business. One reason why we have a bit of trouble right now in our relationship with Oberheim and Sequential Circuits - I can't say I'm close friends with, or see a lot of, either company - is because of this arrangement where they're still paying us royalties for things we did a long time ago. Sometimes people forget how much they appreciated you a long time ago. So we've learned that royalties aren't always best, not so much because they're not financially lucrative, but because they destroy friendships - and as far I'm concerned, friendships are probably more important than the money itself.

JL: Who or what was your greatest influence in designing electronic music equipment?

DR: I can't say there was one greatest influence, but here are a few.

While we were designing E-mu's Universal Active Filter for the modular system, we'd scrape around and see what other people were doing - I'm not proud, I'll steal anybody's idea whenever they've got a good one (laughs) - and came across the ARP multi-mode filter spec sheet. The specifications were pretty amazing; I worked and worked and worked to get a circuit that would meet those specifications, analyzed the thing, and found out all kinds of obscure information about parts. For example, the final design had to be re-designed because of the capacitance from one trace to another on the circuit board, which affected the performance of the whole thing. I was just totally beat designing that thing...I'd dream about it every night.

When I finally got my hands on an ARP multi-mode resonant

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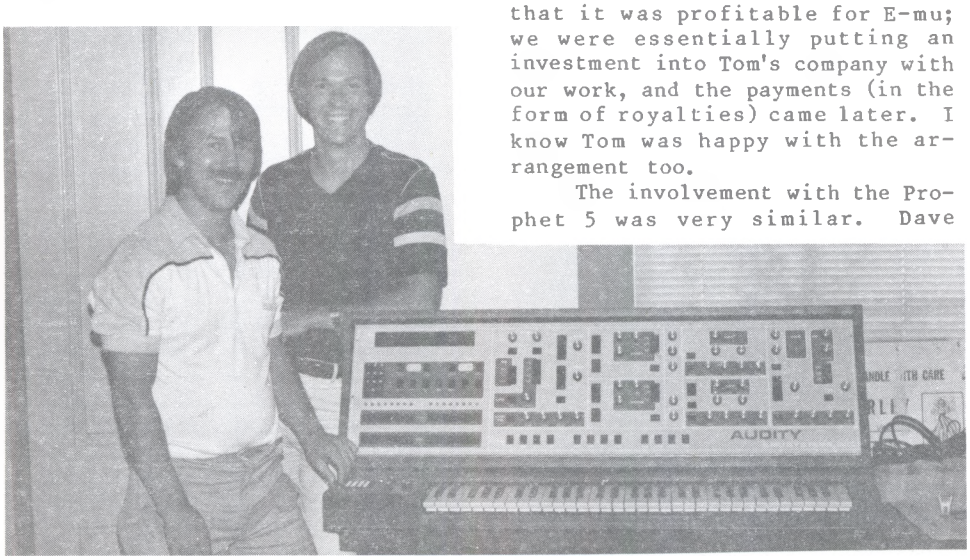

Dave Rossum and Scott Wedge with Audity

ON LOCATION:

by: Karen Petersen

70th AES CONVENTION NEW YORK CITY

One's first Audio Engineering Society convention is like one's first love: there may be others later on, but you never forget the impact of the first. This year the 70th AES show was held at the Waldorf-Astoria in New York City; the four days of workshops and exhibits encompassed a convention comprehensive enough to suit everyone's taste, and then some.

My first reaction upon hearing it would be held at the Waldorf, one of New York's swankiest and most expensive hotels, was to picture swarms of either frizzy haired, wild-eyed inventors or long haired, t-shirted, pale, I-haven't-had-a-wink-of-sleep-in-200-hours studio types earnestly discussing VCOs and low pass filters among the potted palms and oriental rugs. But actually, even though there was some of that, put us all together and we were a pretty conservative looking lot...just a friendly bunch of people, from all over the world, out to share ideas and enjoy the plethora of events and stimulating minds.

Upon registration you received a schedule of events and exhibitors, along with a map to help you find your way around (realize that the AES convention is no street fair - it's four full floors of a very BIG hotel). You also got these plastic shopping bags to pick up the various spec sheets and free gifts that the exhibitors had on hand. The funny thing about having those bags was that when you walked into the first floor of exhibitions, your immediate reaction to the immensity of it all was to rush around with them as if you were in one of those 10 minute, all-you-can-grab supermarket races. Of course, after about five minutes of that madness you saw how hopeless it was and ended up saying to yourself: "Hey, wait a minute! There's four floors of this stuff! Time to slow down and develop a plan of action."

In my case that meant heading through the wonderful maze of Neve, MCI, Trident, and SSL consoles tyring to decide which one to sit down at and try first. (Yes folks, these were not just there for display. The AES convention was a hands-on event.) Then, having had my fill of flying faders, pre- and post-sends, and the like, it was over to the EMT stereo reverb generator, which was an incredible machine. The different reverb parameters and various settings each produced a unique sound, and if any readers can get to a demonstration of this machine I strongly recommend it.

One of the big hits of the show was the Sony Digital Audio Disc System. It was based on a linear 16 bit quantization scheme per channel for encoding, with a sampling frequency of 44.1 kHz. It utilizes a laser stylus and a 4-3/4" disc, which offers 60 to 80 minutes of two channel recording per side or 30 to 40 minutes of four channel. This was quite a machine to see, if only because of the disc's tiny size, and naturally its presence helped to intensify the digital/analog recording battles that were going on constantly at the convention. (Hint: for a taste of the fray, the musicians in the audience can compare the advantages and disadvantages of the New England Digital synth vs. the new McLeyvier synth.)

In retrospect however, one of the nicer highlights of the convention was the many lectures and workshops that you could attend (albeit for a price). It was there that you could meet the Pickerings and the Bodes of the audio age, and a lot of very interesting discussions developed as a result. I attended the more musically oriented lectures, such as the ones on "Sound Modification Devices" and "Electronic Music and Musical Instruments".

At the former, I not only heard about the history of the subject (from Harald Bode) and generalized effects systems (Michael Beigel), but also demonstrations of effects hooked up in series, in parallel and so forth (Craig Anderton), and applications of delay lines (Richard Factor and Anthony Agnello). The "Electronic Music" lectures covered everything from fiber-optic guitars to synchro-sonic recording. Basically, the fiber-optic guitar (developed by George A. Bowley) utilizes the principle of modulation of light within optical fibers, strung on the guitar just like regular strings, optically transmitted to an amplifying device. With this new technology, still in the prototype stage, problems such as hum pick-up and induced noise would be eliminated. Synchro-sonic recording, aside from being a philosophy of music, is a process that optionally ties all instruments, processors, and rhythmic events (vibrato, tremolo, envelope generator sweeps, etc.) into one rhythmic framework, traceable to one master clock. Craig Anderton, who presented the paper, gave some very convincing examples from the last 30 years of popular music to illustrate the evolution of this percussive principle, and closed out the presentation with excerpts from his upcoming synchro-sonically recorded album. For those who wish to read further about the Fiber-Optic guitar or Synchro-Sonic recording, preprints are available (preprints #1828 and #1821 respectively) from the AES. For more information on these and other preprints, write the AES at 60 East 42nd St., New York, NY 10165. Other papers included "Anomalies in the Violin-String-Length/Frequency Relationship" from Norman C. Pickering, "A Proposal for a Standard Electronic Music Synthesizer Interface" by David Smith and Steve Salani from Sequential Circuits, and "Modular Software Design for a Synthesizer System" by Ellen Lapham from Syntauri.

Another interesting workshop was chaired by John Simonton (PAIA Electronics), with Hal Chamberlin, Larry Fast, Craig Anderton, David Smith, and David McLey participating in a panel discussion loosely devoted to the future of synthesis. Early in the workshop any barriers between the panel and audience more or less disappeared, as comments bounced back and forth between both elements. As a result, in addition to comments from the panel we also heard the views of many musicians, including composers such as Laurie Spiegel, Michael Gilbert, and Suzanne Ciani. Topics not only covered the inevitable analog/digital discussions, but also whether there are ways to make electronic music more widely distributed to the general public, and the state of the music business in general.

Continued on page 35.....

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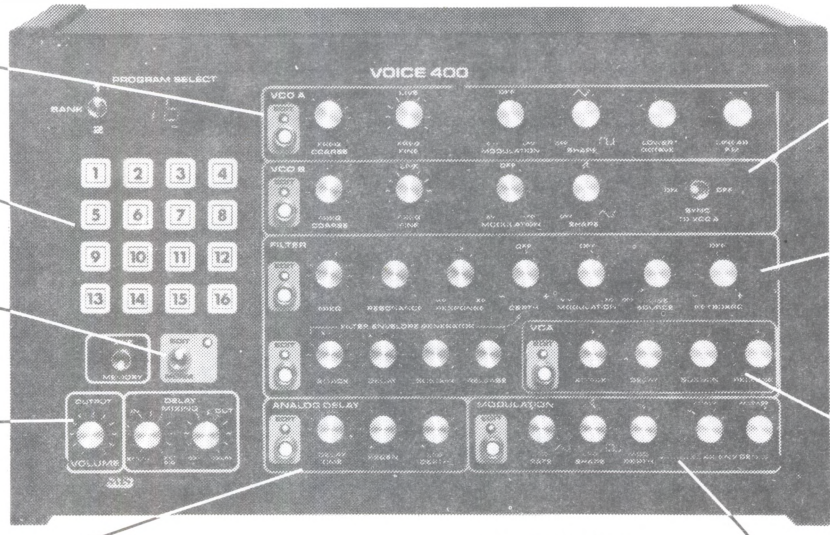
Output Section
mixes your external signal into the delay, mixes Dry/Delay, and output volume control.



Oscillator B
continuous waveshaping from saw to sine, AR envelope generator or LFO modulation, hard sync to VCO A.

Filter
High pass, Low pass, Band pass all modes are 24db/oct. Controls include Resonance, Response (continuously variable) ± ADSR modulation, S/H or LFO mod, Noise source, Keyboard tracking.

Voltage Controlled Amplifier
has its own ADSR and features low noise and wide dynamic range.

Analog Delay
wide range low noise delay line operates from flanging to multiple repeats. Regeneration and LFO depth control will create a wide range of effects.

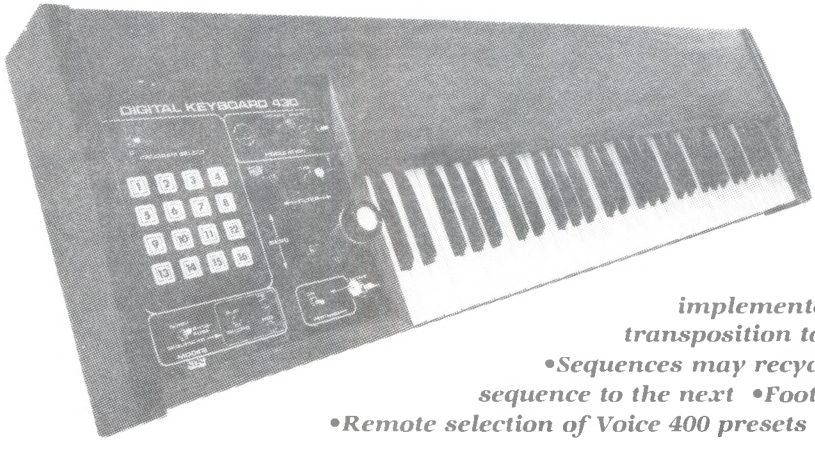
Modulation
wide range Low Frequency Oscillator with continuous waveshaping of three waveforms. LFO may be modulated by the Attack Release envelope generator.

The Voice 400 answers the need for a programmable synthesizer that's versatile enough to be all these things:

* A complete keyboard instrument with the optional SMS Model 430 Digital Keyboard.
* An expander for your present synthesizer whether mono or poly.
* An acoustic instrument-controlled synthesizer when used with a pitch-to-voltage converter.
* A voice for a sequencer or computer.
* A complete synthesizer for wind or string controllers.
* A programmable filter and/or a programmable delay line.

Quality components have been carefully selected for this instrument. Great effort has been taken to insure that the effects of temperature and vibration are minimal. When you're ready to play, the Voice will be too . . . accurately, and every time.

#430 Digital Keyboard Sequencer



•16 sequences of up to 64 notes each programmable from the keyboard—1000 notes total •3 axis joystick for pitch bend, filter and mod depth •New Note assignment for ultimate lead "feel" •Built in LFO with two waveshapes and rate LED •Digital circuitry for drift-free performance •Portamento •3 position octave switch (digitally implemented) •Full length 61 note keyboard •Instant transposition to any key for both keyboard and sequences •Sequences may recycle, play once and end or advance from one sequence to the next •Footswitch input to start and resyne sequences •Remote selection of Voice 400 presets •Fine cabinetry with genuine walnut finish

SMS

P.O. Box 40267 San Francisco Ca., 94140 Tel (415) 824-4837
East Coast Office: 8 Tyler, Norwell Ma., 02061 (617) 659-2618

VCO

NOTES

A1 - A4 = ¼ 4136, IC2

Q1, Q2 = 2N4124

D1, D2 = 1N4148

All capacitors in uF, except where noted.

For dual unit, repeat all parts except for R31, R29, C10, C7, and IC3. IC3 is a dual flip-flop, and the other half may be used independently.

Power to IC2: V+ = pin 11
V- = pin 7

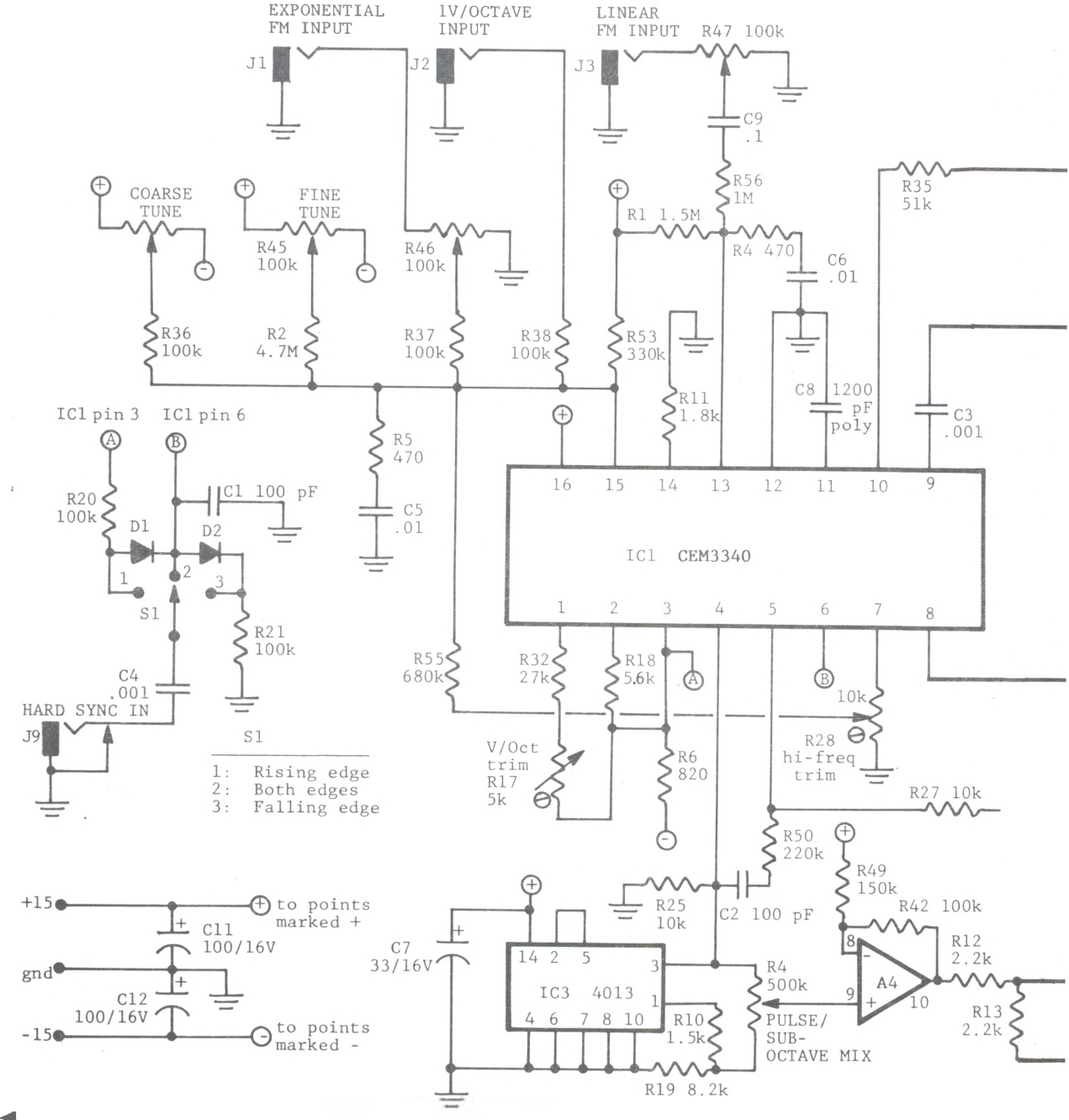


figure1

DELUXE

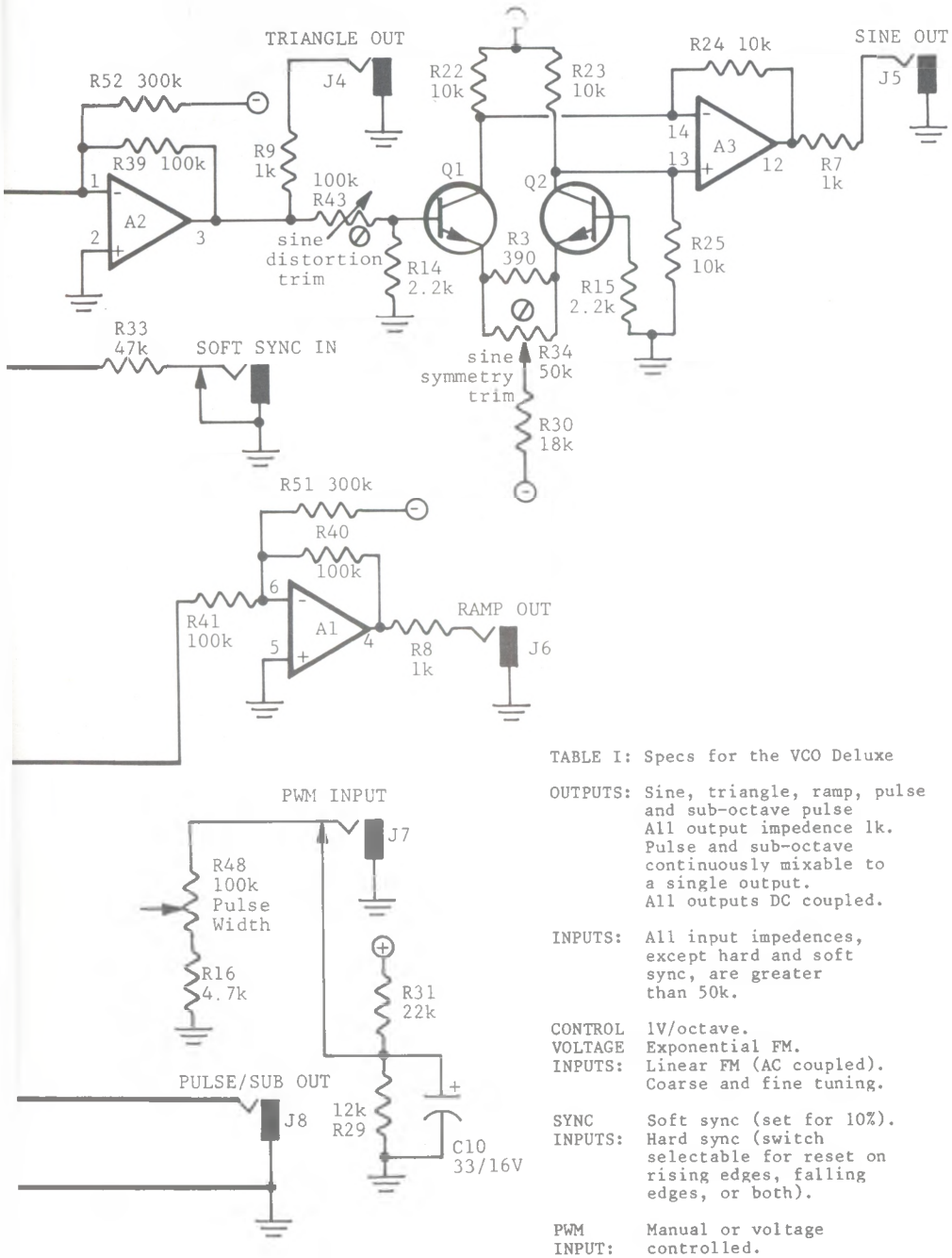


TABLE I: Specs for the VCO Deluxe

OUTPUTS:	Sine, triangle, ramp, pulse and sub-octave pulse All output impedence 1k. Pulse and sub-octave continuously mixable to a single output. All outputs DC coupled.
INPUTS:	All input impedences, except hard and soft sync, are greater than 50k.
CONTROL VOLTAGE INPUTS:	1V/octave. Exponential FM. Linear FM (AC coupled). Coarse and fine tuning.
SYNC INPUTS:	Soft sync (set for 10%). Hard sync (switch selectable for reset on rising edges, falling edges, or both).
PWM INPUT:	Manual or voltage controlled.

It's time to start building again! This issue we're going to work on a VCO with lots of features; in fact, I call it "VCO Deluxe". Since this is a fairly big project I won't be able to dwell on any point in great detail. However, since the circuit is based on the CEM3340 VCO chip, and since that chip is supplied with an excellent data sheet, I don't think you'll have any trouble filling in the details.

There are a number of features in this circuit that pop up again and again, so to save space, Table I summarizes these common features in the form of a "spec sheet". These specs will make many facets of the design easier to understand.

We can breeze through the circuit description now, especially since the CEM3340 does most of the hard work for us. Refer to the schematic in figure 1. J1 and J2 are the two exponential control voltage inputs. The voltage applied to J1 can be attenuated by R46, while the voltage at J2 is left alone and is simply the 1 Volt per octave input.

J3 and its associated attenuator R47 form the input for linear frequency modulation. This would commonly be used for vibrato (FM by an LFO), or for creating gong sounds (FM by another VCO). This input is AC coupled which makes it very easy to use, since you don't have to worry about any DC offsets shifting the fundamental pitch.

R44 and R45 are the coarse and fine tuning controls. R44 covers a very wide range; in the farthest counterclockwise position, oscillation is well below 0.5 Hz. At the other extreme of rotation, the oscillation is around 35 kHz. The fine tuning control has a much more restricted range and covers a musical interval of about a fifth.

We've now overed the entire frequency control input structure comprising two tuning controls, the exponential FM input, the 1V/octave input, and the linear FM input. Now let's take a look at the sync inputs.

The soft sync signal is injected at J10 and should be some sort of rectangular pulse. This

sync input responds only to negative going triggers. Resistor R33 is installed to limit the amount of synching to about 10%. This seems about right to my ear, because more synching causes greater distortion and starts to sound like hard sync, while less sync gives inferior phase locking. However, feel free to experiment with R33.

The hard sync signal is injected at J9. Now to understand S1 and its associated circuitry, we must be aware that the CEM3340 hard sync input responds to either positive or negative going pulses. This being the case, it is easy to see that if the switch is in position 1, only positive sync pulses are transferred to the sync input at pin 6. Negative pulses are blocked by D1. Likewise, when the switch is in position 3, only negative pulses are transmitted to pin 6. When the switch is in position 2, both positive and negative pulses are coupled to the sync input of the chip. The upshot is that we have three types of sync sounds. For more details on this sync-switch structure, see my recent article in Electronotes2.

The pulse width modulation input is at J7. In the absence of any plug being inserted in J7, pot R48 manually controls the pulse width via a fixed voltage supplied by voltage divider R31 and R29. However, inserting a plug into J7 removes the fixed bias voltage and allows for voltage controlled pulse width modulation.

Well, we've covered all the input considerations; now it's time to talk about the outputs. Remember, our "standard" is to have 10V p-p outputs, centered about ground, with a 1k output impedance. With that in mind, let's look at the triangle wave output first.

The triangle is available at pin 10 of the CEM3340, however it is a non-standard voltage. A2 level shifts and amplifies this signal by a factor of 2, making it standard.

The triangle wave also feeds the sine converter formed by Q1 and Q2. If you're used to the old 3080 type sine converters, this one will give you quite a surprise. Distortion is incredibly low, even with garden variety transistors. What makes this converter superior to the 3080 type is the inclusion of feedback via R3, which practically nulls out the "pip" on the extreme end points of the sine wave3. Trimmer R43 minimizes the odd harmonic distortion, while trimmer R34 adjusts the even order harmonic distortion.

The ramp wave makes its exit via J6. It should be obvious how A1 level shifts the ramp output at pin 8 of the CEM3340.

The pulse wave is available at pin 4 of the CEM3340. R25 is a pull-down resistor for the emitter of the internal transistor. The pulse developed across this resistor then splits off in two directions. First it feeds pot R54, which is the Pulse/Sub-octave blend control. Then the pulse goes to the clock input of a CMOS 4013, configured as a binary divider. The output of the binary divider goes to a voltage divider comprising R10 and R19. This resistor string chops the sub-octave wave down to about a 13V level, which is the level of the original pulse. This means that both sides of R54, the blend control, see signals of the same amplitude, allowing smooth blend transitions. The wiper of R54 is buffered, amplified, and level shifted by A4, which is configured in a rather unusual way. If you imagine that R49 goes to ground, it is easy to see that A4 becomes just an ordinary non-inverting amplifier. However, pulling R49 to the positive supply instead of ground level shifts the input signal and at the same time retains the non-inverting amplifier characteristic4. The result is an output that is 20V p-p centered about ground, which is then chopped down to 10V p-p by divider R12 and R13. This combination of 2.2k resistors gives approximately a 1k output impedance at the same time.

So, by utilizing this somewhat strange configuration, we can see that rotating R54 in one direction gives full pulse wave, with no sub-octave. Rotating it in the other direction adds an increasing amount of sub-octave wave, until the pot has been rotated all the way, at which point the output is full sub-octave. And we've kept a 10V p-p signal the whole way around the rotation!

Calibration. Tweaking the module isn't all that hard, but it does take patience. The spec sheet gives the details on how to tune up trimmers R17 (the V/octave trim) and R28 (the high frequency error trim), so I won't say anything about them1. The sine trimmers, R43 and R34, are best adjusted with an oscilloscope, with your ear serving as the final arbiter. While watching the waveform, tweak R43 until a nicely rounded sinusoid is formed. Then adjust R34 for a symmetric waveform. While performing both of these trims, listen to the sine wave on an amplifier and speaker (at around 500 Hz), and note the relationship between waveshape and timbre. Believe it or not, your ear will allow you to tweak up the sine wave for very low distortion. However, if you have a distortion analyzer...

Well, it's too bad that there's not enough space to discuss every point of this VCO, for I think it's a really good one. However, if you take the time to study the CEM3340 data sheet, I'm certain that this will clear up many of the fine points. In the meantime, enjoy the VCO Deluxe and we'll discuss something new in this space next issue.

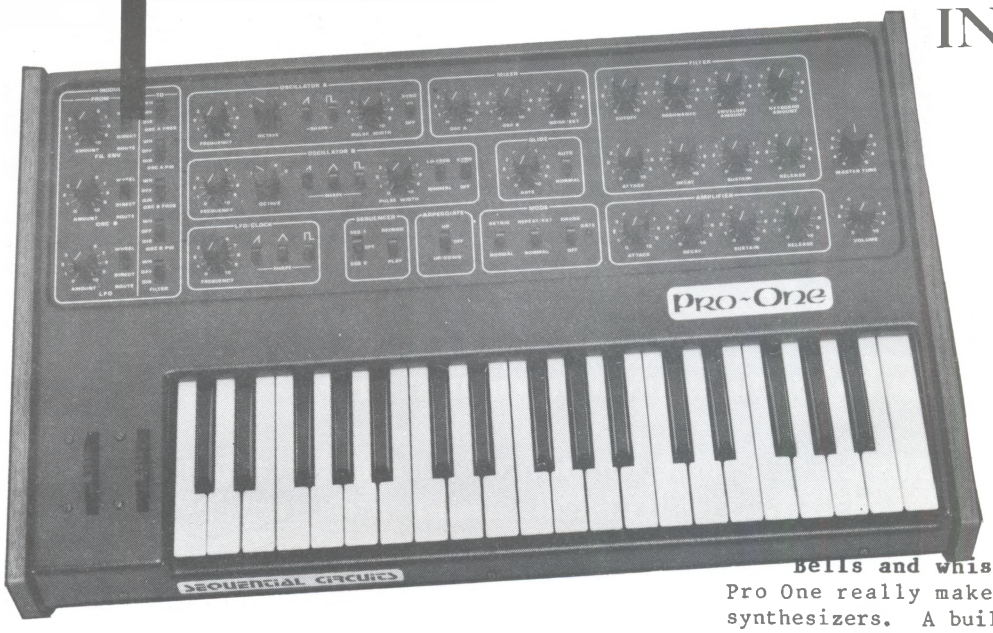
References

1. The CEM3340 and spec sheet are both available from PAIA Electronics.
2. T. Henry, "Two Hints on Using the CEM3340 VCO IC", ELECTRONOTES, Volume 13, Number 121, January 1981, pp. 13-16.
3. For more information on discrete triangle to sine converters, refer to W. Jung, "Triangle to Sine Wave Converter", IC Op Amp Cookbook, Howard W. Sams and Co., Indianapolis, 1979, pp. 386-388.
4. This method presupposes that you are using a well regulated power supply. e

INSTRUMENT

THE

by



The Pro-One is a small monophonic performance synthesizer based on circuitry made famous by its poplyphonic parents, the Prophet-5 and Prophet-10. The Pro incorporates many features that make it a real value in its price range. These include a 40 note sequencer function (true sequencing, not FM wave-shaping) with the ability to add rests in the program, an arpeggiator, and an external audio input for instrument treatments and processing.

Like most mono synths, the Pro has two oscillators with a variety of waveforms (these include sawtooth, square, and modulated pulse). The two oscillators can be offset to any interval using rotary pots, and can also be set at one to three octave spreads using the preset octave control. A master tune control, on the far right of the instrument, allows quick and easy tuning to other instruments. Also included is a sync switch for creation of complex waveforms, and each oscillator also has its own volume level pot. Other oscillator related functions consist of a two position glide switch that gives you a choice of auto-glide (continuous until the VCA dies out), or regular (normal) glide that stops when your finger leaves the key.

The 24 dB/octave low pass filter is par for the course, but the Pro's filter has its very own ADSR along with four controls (resonance, cutoff, keyboard amount, and amount of envelope generator modulation). The VCA also has its own ADSR generator and a bypass switch. The keyboard has provisions for single or multiple triggering, plus a repeat function whose rate is controlled by the LFO speed.

One outstanding feature of the Pro is the control voltage routing system. All oscillators, VCA, and VCF functions can be routed to a thumbwheel (with variable depths of change) or directly to the ADSRs. The oscillators can also modulate each other for an extra LFO, or for a ring modulation effect (a low frequency switch converts oscillator B to the sub-audible range to obtain the LFO function). There is also a thumbwheel for pitch-bend with a center notch for easy return to the original note.

Bells and whistles. Computer functions in the Pro One really make it stand out from other mono synthesizers. A built-in digital sequencer has a 40 note memory which can store two individual sequences at once. These can be selected at any time by simply flipping a 3 position switch from "off" to "seq #1" or "seq #2". If nothing is stored in a memory, seq #1 gives an uphill cascade of 20 notes and seq #2 gives a downhill riff. Sequence speed is controlled by the LFO rate or by the gate input on the rear panel. The programming method is quite unusual; to memorize a sequence, you select either the seq #1 or seq #2 mode. Next to this is a record/play switch. You put this switch in the record position and enter the notes via the keyboard. In order to get a rest in the sequence, the record switch is flipped from record to play and back to record again to get one rest. All notes entered are of equal duration, so it doesn't matter how fast or how slow you program the sequence. To play back the sequence, you switch the record/play switch to "play", the three position switch to "off", and then select either seq #1 or seq #2 with the same three-position switch. All signal processing pots (VCO, VCA, VCF) are still active, so treating a recorded sequence is no problem. The sequence can also be transposed at any time by hitting a different key (the default setting is low C).

Another neat function is the arpeggiator switch. Notes can be arpped up or up and down by holding down any number of keys. The Pro One will sequence each of the keys held down at the rate set by the LFO speed. These notes can be "latched" by the computer so that you can take your hand(s) off of the keyboard. The arpeggiated notes cannot be transposed like the sequencer, but new notes can be added at any time by holding down extra keys. Unlike the sequencer, the arpped riff cannot be stored in memory; once the "latch" is turned off, you lose the arpeggio completely.

Those jacks on the back. The Pro has back panel jacks that are a definite plus. Control voltage in, out, filter in and gate in/out are to be expected nowadays, but the Pro also owns an audio input jack that can be used to process other instruments through its VCA and VCF or can trigger a sequence stored in memory. This lends itself to many unusual alternate controllers, such as voice,

REVIEW

PRO-ONE

Chuck Pogan

guitar, other synths, or almost any signal. An audio sensitivity pot is located on the front panel and also doubles as the noise source when this function is not being used. One such unusual controller is what I like to call the PAST system (Primitive, Analog, Sequential Triggering system). It consists of nothing more than a cheapo microphone in a coffee can. By playing the can like a bongo, a successive note store in the memory can be triggered every time it is struck. This allows for real time sequence manipulation, and is a lot of fun for budding or non-keyboardists who want to hear flashy riffs but don't have the dexterity to play them "keyboard-style".

Pro One Cons. The Pro, for all its obvious trick functions, has a few shortcomings. Unless the unit has been warmed up for a considerable time, the sequencer, while loading, may stutter a note even though the selected key is depressed only once. When this happens, the memory will record the same note two times (or more). Perhaps this is just a problem with my particular unit; knowing the standards of excellence Sequential is famous for, this is probably the case.

Also, I wish that the audio input sensitivity pot and noise source were two different controls instead of sharing two functions at once. This would have allowed the Pro to double as a percussion synth using alternate controllers. As it stands now, white noise cannot be added when using the audio input jack.

Finally, the unit's construction is not exactly heavy-duty. The front panel is of thin plastic and the pot and switch quality is only average. If you intend to carry the Pro around, then an anvil case is highly recommended or else handle it with kid gloves.

Pro One Pros. The most important aspect of any musical instrument is the quality of sound and playability. The Pro stands out in these respects. Like most synths, the bass register is very good, but the Pro really shines as a high-end screamer (something not normally found on an inexpensive synth). The oscillators are very stable, and the front panel is laid out logically and intelligently.

Overall evaluation. The Pro One is a very well designed synth in general. Even without the computer functions, it still would be an excellent performance device for the money. But considering that it does so much more than many other units in its ballpark, I feel the Pro should be one of your first considerations when shopping for a mono synth. e

For the Pro-One
A DIGITAL INTERFACE

by GREG ARMBRUSTER

Since Polyphony deals with a more educated and experimentally-oriented readership, the following information might be very useful to those working with the Sequential Circuits Pro-One. For example, I think it would be possible to create an "orchestration" machine using a home computer as the sequencer/controller for several Pro-Ones. Each Pro-One could be patched for different instrumental timbres and the entire array conducted by the computer. The more clever programmers could add the visual notation as part of the computer's duties, but it's the sound that would be impressive. Besides, the cost of five Pro-Ones is much cheaper than a Prophet-5.

Adding a digital interface for external computer control requires both some simple hardware modifications to the Pro-One and the creation of software for your system. If you doubt your technical ability to install the interface, please see an SCI Authorized Service Center or contact our Service Department. However, **please note that we must leave all questions of programming your system to you.** Now for the modification...

1. Remove the front panel, and then remove the 8021 microcomputer (U113) from its socket. Store it in conductive foam and keep it in a safe place; avoid excessive handling of this chip. Note that with the 8021 removed, the Pro-One's keyboard cannot operate.

2. Jumper pin 2 of the 8021's socket to J101-7. Pads are provided on the printed circuit board for this purpose.

3. Locate the area between the trigger switch and the repeat/external switch (in the mode section). You'll see a diode, a resistor, and between these two parts, space for a resistor (this is labelled optional CPU). Add a 10k Ohm, 1/4 Watt, 5% resistor in the space between the existing diode and resistor.

4. The accompanying table lists pin assignments for J101. Mate J101 with a 14 wire ribbon cable terminated in a DIP plug.

DIGITAL INTERFACE PINS (J101)

Pin	Signal	Pin	Signal
1	D5	6	D0
2	D4	7	Gate
3	D3	8	DAC enable
4	D2	9-14	Ground
5	D1		

The interface is compatible with standard TTL logic, where less than 0.8V signifies 0 and greater than 2.5V signifies 1. It accepts six data bits (D0 through D5). This allows the numbers 0 through 63 (decimal) to convert to over five octaves of "keyboard" control when latched to the internal digital-to-analog converter (DAC) by the positive going DAC Enable (DAC EN) pulse. The Gate is exercised via a bit at pin 7.

To play notes on the Pro-One, the system driving it will have to do something like the following (the way in which these events occur is not as important as their sequence). At the start, the

Continued on page 35.....

INTERVIEW: DAVE ROSSUM

.....continued from page 25

filter, it didn't come anywhere near to the specs they had listed on the spec sheet! Sometimes believing that it's possible to do something can inspire you to exceed what really is possible.

Anyother great influence on me was Joe Parmalee, for whom I worked at Santa Clara Systems. Joe was an absolute perfectionist; if there was any conceivable way to improve a circuit, he'd make me do it. That was real good training, as it taught me to be very thorough and do my best work.

Other influences exist all around us. We call up a lot of people and ask them what they think, and remain open to what they say. So, anybody can influence us and they won't even know when they've done it. I don't always like an idea when I hear it, but after it sets for a few days I may decide that it's all right after all.

JL: Voltage control is an important aspect of the modular system. What is its future, if any, with the advent of digital synthesis?

DR: Well, I think there is a lot of misunderstanding about digital synthesis - what it really means and what it's all about.

I can remember Scott coming in once and pointing out an ad to me that said in capital letters, "Digital is Better". Scott says this is what people think, but we both know it's BS. A lousy digital synthesizer is a hell of a lot worse than a good analog synthesizer, but somehow people got this brainwashed idea that digital is better. Actually, digital is different, not necessarily better. But let's look first at analogies between digital and analog systems - and there always are analogies, since both technologies are trying to accomplish the same goal.

Voltage control is considered a big thing, but as most circuit designers know it's really current control. There's nothing magic about voltage control; what's magic is that you can take a handful of modules and because any output can control any input, the number of different patches - the number of ways of making sounds - goes up exponentially with the number of inputs and outputs. So with a relatively small number of identical modules, absolutely incredible things can be done...and that's what's so neat about voltage control. Look at the little pre-patched synthesizers: sure they're voltage controlled inside, but who cares? They don't make much use of the voltage control. A modular system does a lot more.

Looking at digital systems, the analogy to the patch is the algorithm - the way that you push numbers around inside the computer. For example, Fourier synthesis is one algorithm, FM synthesis is another algorithm, a digital filter is yet another algorithm, and you can combine algorithms just like you would combine patches. Hence, voltage control would be analogous to a variable algorithm digital synthesizer, where it didn't just compute in one way, but you could actually go in there and write the microprograms and get the thing to compute the sound in whatever way you wanted. The problem with that approach is you have to be an incredible computer jock - writing microcode is hard stuff - and second, you have to find a general purpose microcoded computer that's fast enough to generate sound in real time. That's a total bitch; it just can't be done economically right now. So, I would say that the analogy of voltage control is still a goal in digital synthesis, and it's not here yet.

When you look around at digital synthesizers, you hear some pretty incredible sounds, but they're still really limited. When we get into that multiple algorithm instrument, you'll really have some power. On the other hand, I don't think the analog instruments are ever going to be outdated or obsolete; they do what they do, and what they do is pretty neat. I think that eventually we we will translate the concept of voltage into variable algorithm in the digital machines, but that's in the future.

JL: E-mu's Emulator (Ed. note: a keyboard instrument that reproduces, rather than synthesizes, the sounds of other instruments and sounds - similar to the idea of a solid-state Mellotron) is a radical departure from your other work. What prompted you to take this direction?

DR: In the sense of the product concept, I guess it is a pretty radical departure. But in the sense of the inside design, it's amazingly similar to what we've done before.

At a certain point our financial picture made it clear to us that we needed to get into some higher volume work if we were going to stay alive. Scott and I were getting tired of starving to death and not having any extra money to play with, when we could have worked in a silicon gulch industry and made at least five times times what we're earning right now. So to some extent, it was a financial decision that made us go in the direction of a high volume instrument. The actual idea came from the fact that I was kind of amazed that something like the Fairlight could sell for $35,000 - or whatever it is - since I knew that it could be done a lot cheaper with a proper design. I think I've been quoted before as saying that anybody can design digital, and I kind of put my money where my mouth was and designed the thing. I think it's a really neat instrument. It does something the electronic music community wants, and it does it well and simply. The hardware in there is exceptionally flexible, so despite the instrument's apparent simplicity, it's pretty incredible what it can do. I could spend an entire interview talking about the Emulator...but that's not really what we're supposed to be doing here, so let's continue.

JL: What do you see in the future for electronic music, and are you working towards that end?

DR: I see a lot of different things in the future of electronic music. You can look at the Casio machines, and that's one future of electronic music - getting it into every home, making it cheaper, and kind of throwing away the fact that it wants to be high fidelity. That's one future of electronic music, but I don't really feel I'm working towards that end so much. In another sense, I see making the existing concepts - programmable synthesizers - more affordable. We're also working towards more utility for a given amount of money, which shows in an instrument like the Emulator. In the future you're going to see more and more digital instruments, not because digital is inherently better but because digital technology is going to be getting cheaper and cheaper, which will make it more cost-competitive with the analog stuff. We're definitely working towards that.

I think that E-mu will always be at the forefront of research, too; I'm a curious person and I've got some pretty off-the-wall ideas that I'm working on right now. But I don't think you'll see E-mu come out with any more real expensive dream machines, that's more for beginning companies. When you're first starting out in electronic music you're kind of idealistic and willing to work for nothing, and that's when you can go work for an artist or people who want special gadgets. You work for little money, spending lots of their money, to make incredible machines. As you go on in life you tend to think more of your own needs and want to produce more volume items...and end up making Emulators and things like that. Again, fine instruments; but more for the masses.

JL: Besides doing design work for Oberheim and Sequential, have you done design work for others?

DR: Yes, I have done design work for a lot of different people, some of them not in the electronic music community. I'm still closely associated with SSM, and consult on almost all of their designs. I've done consulting for my previous employer, Santa Clara Systems, as well as some digital memory testers and VCAs for National Semiconductor, some synthesizer design for a company called GEM which makes electronic organs, Lyricon, Octave Electronics, and I'm also doing some work with a few other companies whose names I unfortunately can't mention due to marketing considerations. E-mu is actually two companies; there's E-mu Systems Incorporated which makes the modulars and Emulators and so on, and E-mu Design which is the consulting and designing service. There we do virtually anything anybody wants us to do as consultants. And yes, we are available.

JL: Would you care to share your secret of accomplishing so much work...does it have something to do with your style of work? You're certainly not a nine-to-fiver, but more of a marathon type. How does this help you work?

DR: Well, I think that the thing I can share with people is a concept called intention. Basically, when you have an intention to do something, you direct your entire attention - all of your consciousness is just focussed - on what you want to do. You have to have really clear goals, go one step at a time, and keep a truly open mind - don't bang your head against the wall trying to get through, be aware enough to realize that you can often walk around the wall. I'm kind of a tornado around work, I leave little piles of mess behind me. My intention is more to get the job done than clean up after myself, and the people that work with me realize that.

The other thing that's real important to me is being completely wherever I am. When I'm at work, I'm completely here. I don't think at all of what I'm going to do tonight, or what I'm going to do tomorrow: I think about exactly what it is I'm trying to accomplish. On the other hand when I go away, like when I spent last week getting certified as a scuba diving instructor, I didn't think about E-mu for a second. That intensity is what helps me to do as much as I do.

Well, I can see we're running out of time so that should just about do it for the interview. Thanks for the chance to say some things, and for the good questions.

JL: On behalf of the readers of Polyphony, thank you.

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the Pro-One INTERFACE

continued from.....page 33

sequence). At the start, the analog Keyboard Control Voltage (KYBD CV) output from the DAC (which controls the oscillators and filter) is unknown. Nothing is now heard from the Pro-One because the envelope generators are not being gated. First the system somehow decides what key (out of 63) it wants to play. It then sets up the key number in binary form on D0-D5. (This might be done with a six-bit output port.) Next DAC EN must be forced high, latching the key data. For practical purposes a delay of 20 microseconds will allow the DAC time to settle (worst-case). At this point, GATE can go high, triggering the envelope generators. When the note is to be turned off, it is likewise advisable to turn the GATE off (low) before again strobing the DAC.

To return the Pro-One to normal operation, unplug your system from J101, remove the 10k resistor mentioned previously in step 3, cut the jumper installed in step 2, and carefully re-install the 8021 (U113).

The technical staff of sequential circuits would, of course, be interested in the results of any successful experiments using the digital interface. Write to Sequential Circuits, 3051 North 1st Street, San Jose, CA 95134; or call (408) 946-5240. e

On Location: 70th AES

.....continued from page 26.

Other than the exhibits, presentations of papers, and workshops, in the evening many of us spent long hours in bars and smoky hotel rooms talking about the future of audio and music, while swapping tapes back and forth on the nearest available Walkman. The conversation turned to such topics as stereo implants, humans and machines, new musical concepts - crazy, visionary things that one gets to thinking about after ingesting a few drinks or other chemicals - you all know the feeling.

Those four days were really some of the best days I've had in a long while. But then again, remember what I said about first loves. e

EQUIPMENT EXCHANGE

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Equipment

SEQUENTIAL CIRCUITS model 800 Digital Sequencer $400; Modular Moog Analog Sequencer, Sequential Switch, Multiple, Interface with power supply in studio cabinet $600; RMI Keyboard Computer Digital synthesizer with all voice cards $1500. Duane Decker (801) 969-8731.

SYNTHESIZER CALIBRATORS, kits or assembled. 1/3 octave pink noise test tapes with warble tones. TEAC 2A upgrade mods. Write to: ARCAS Engineering, RFD #3, Burthill Rd., Winchester, NH 03470.

Music equipment

EXPANDED P-4700/J factory assembled, 1K RAM, MUS-1, 4 VCO, 2 VCF, 3 ADSR, 1 REVERB, 1 LFO, 1 Mixer, 2 Balanced Modulators, 2 Transposers, 1 multiple/attenuator panel, 2 pedals, 2 extra wing cabinets, Realistic cassette for software, cords, manuals, excellent condition $1400 invested $700.00 or offer. Must sell! Ron Slabe, 26351 Lakeshore Blvd., Euclid, OH 44132 (216) 261-1072.

YAMAHA CS-50 4-note touch responsive synthesizer, noiseless, $1200. PAIA 4700/J including all components minus QuASH and one filter, plus 2720-4 AR and 4740 ADSR. VCA's hum, otherwise great shape, $400. Stephen Mizer, 122 Lillard, Belton, MO 64012.(816) 331-3072.

FOR SALE: PAIA modules, all operable and calibrated: 2720-1, '-3B, '-3L, '-4, '-5, '-7, $10 each; 2720-2A (needs repair) $5.00; 4710, $15; 4720 $25; 4740 $17; 4770 $11; 8780 $25. Direct inquiries to: D. R. Cool, 7421 Troy Manor Rd., Dayton, OH 45424.

PAIA Gnome and Oz $60. Needs work. Jim hardy, 1676 E. 4th, St. Paul, MN 55106 (612) 771-7995.

PAIA 1550 Stringz 'n' Thingz with 1551 Stereo option. Excellent condition, professionally assembled, factory adjusted. MUST SELL, $525.00 or offer. Dave Garfield, 1105 Clayton Lane #105, Austin, TX 78723 (512) 458-1731.

OBERHEIM Four Voice Polyohonic Synthesizer, with programmer, 1-1/2 years old, like new. $2975.00 or best offer, Craig Markley, 292 Hane Ave., Marion, OH 43302 (614) 387-6632.

PAIA P-4700/J, assembled, plus extra 4761, (3) 4770, 4712, 4710, 2720-11, 2720-12, 4740, EK-2, 1500 Phlanger, cords, manuals, software. Excellent condition. $1100 new, $600 takes all. Robert Sorel, 393 Ninth Avenue, Woonsocket, RI 02895 (401) 767-3349.

ELECTRO-HARMONIX
Mini-Synthesizers, 2 octave board, 5 octave range, self-contained Speaker/Amp. pitch bend, phase control, weights 2 lbs. $150.00. Runs off 2 nine volt batteries or E.H. AC adaptor, $15.00. 1 year warranty. Money order, cashier's check or cash o.d. Sound Electra, P. O. Box 17206, Minneapolis, MN 55417 (612) 727-1482.

Supplies

BLANK AUDIO TAPE: AMPEX AND SCOTCH (reel-reel). Also empty reels and boxes. Send for information:
RECORDING SUPPLIES
1058 Oakview Drive
Cleveland, Ohio 44143
(216) 442-2467

Literature

PROGRESSIVE, Experimental, Electronic, Avante-Garde, and Microtonal Music is covered in depth in SURFACE NOISE magazine. $2.00 for current issue or $8.00 for one year subscription. SURFACE NOISE, 428 Citrus Road, Melbourne, FL 32935

Recordings

TECHNO DE-FACTO the new 60 minute cassette by Walt Whitney, 14 Avante/Rock songs guaranteed to entertain. Send $5.00 ppd. to: Sub Sound, P.O. Box 2411, Overland, MO 63114. Money back guarantee.

BEACON STAR - synthesizer/space music on cassette tape. $5.00 check/money order to: Chuck Larrieu, Box 294 Corte Madera, CA 94925. "it's good stuff!"

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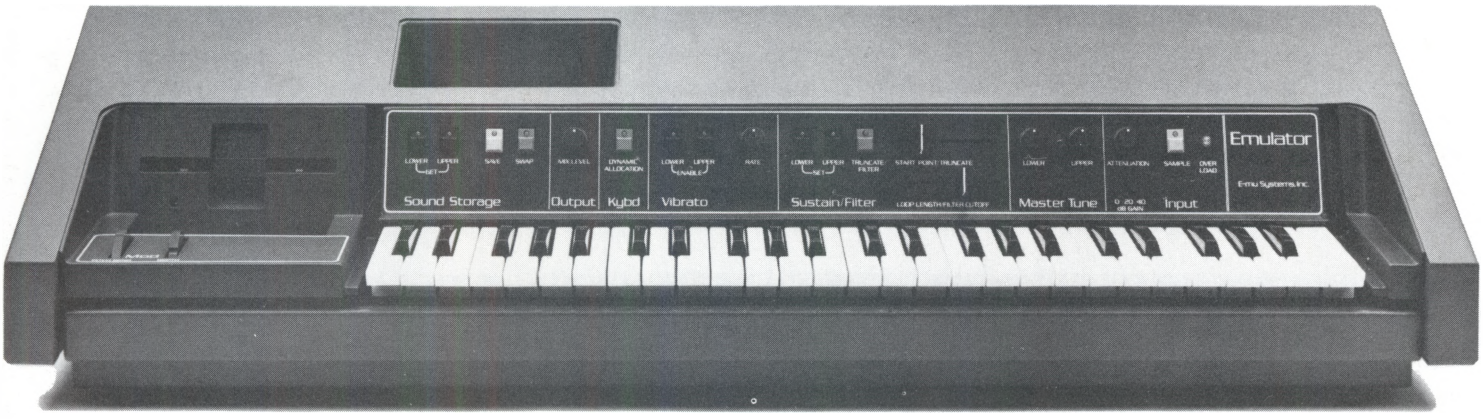
"Any sufficiently advanced technology is indistinguishable from magic."
—Arthur C. Clarke



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