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# CONTENTS

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## FEATURES



## COLUMNS



## REGULARS



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ON THE COVER: A Silk Screen Print by: Piet Jan Blauw

# Letters

**C-64 MICRO-DRUMS!**

I was just reading the August '84 issue of Polyphony (excellent, as usual) and thought I'd cast my vote in favor or a C-64 implementation of Micro-Drums (I'm sure many other Polyphony readers are Commodore owners).

Mike Dolan
Bristol, TN

**OOOH...AHHH!**

I have read and thoroughly enjoyed "Electronic Projects for Musicians" and "Home Recording for Musicians". I have also built several of the Craig Anderton PAIA projects, to my complete satisfaction. Now I have an interesting idea for a project. Would it be possible to connect a keyboard with oscillators and chorusing circuit to a speech synthesis chip to produce a keyboard that could sing oooh, ahhh, la, da, etc.? Such a unique keyboard would be very useful for my home studio. Even better, could such a circuit be added to a Stringz 'N Things? Any thoughts or opinions would be greatly appreciated.

Jon Hand
Pulaski, TN

Jon -- Actually, there are several ways to do what you want to do. A speech synthesizer chip would probably not be satisfactory, as these have a kind of "Mr. Roboto" timbre which tends to grate after a while -- certainly not the dulcet tones of a choir. I have had very good results using vocoders with Stringz 'N Thingz to produce vocal/choir effects; in fact, PAIA has just come out with an under $100 vocoder which would probably give the exact effects which you seek. If you want to invest some big bucks, the Emulator II can do fantastic vocal simulations via sampling; Ensoniq has also just released an under-$2000 sampling keyboard, although I have not yet had a chance to check it out. Overall, I think the vocoder/Stringz combination would be your most cost-effective option.

**MORE GUITAR!**

I am an electric guitarist and a very interested reader of Polyphony. Although I realize that the electric guitar has little to do with synthesizers (unless you have a guitar synthesizer), I am annoyed by the fact that most of your projects are written with only the keyboard synthesist in mind. Is it assumed that there are too few guitar playing readers out there to consider?

A super envelope follower project would be nice. It would enable an electric guitar, bass, or microphone to use and control many of your past projects: VCFs, VCLFOs, etc. that are voltage controlled. Also, in future articles where a guitar might be applicable, a mention of some sort as to how the guitar can be applied would be appreciated.

I enjoy Polyphony and plan to continue receiving it. I have learned many things I would not have learned elsewhere.

Tom Carter
San Carlos, CA

Tom -- We are always looking for guitar-oriented articles, but don't receive too many from readers. I would write them myself, except that I already covered the subject of guitar interfacing fairly thoroughly in DEVICE (see Polymart). DEVICE contained an ongoing series on the AMS-100, a trigger-oriented audio modification system for guitar and other electric signal sources (I still use the AMS-100 all the time in my own studio). The AMS-100 interfaces guitar to many existing modules, as well as custom modules. Older issues of Polyphony also contain some AMS-100 add-ons. Until more readers get

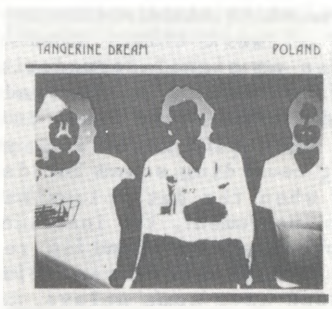
(continued on page 33)

Robert Carlberg's

# re-view

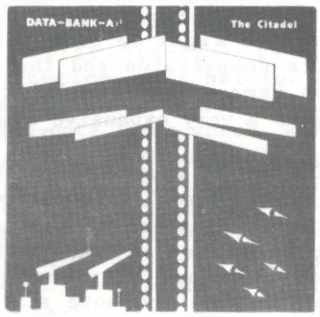
Laurie Anderson **United States** (Warner Bros. 25192-1). A 5-LP set of the complete United States" performance piece, taped live during the Feb '83 debut of the whole thing. This is meant to be a companion to the book of the visuals form the show (reviewed October). It's a fabulous job of live recording, with just the right amount of audience response and hall acoustics -- if you're not already Lauried out.


Tangerine Dream **Poland** (Jive Electro HIP 22). Live albums have been good for the Dream because they don't just repeat their hits. The extended side-long improvisation, which they practically invented, still works due to their intelligent synthesizing and unceasing diversity. Nobody does it better.



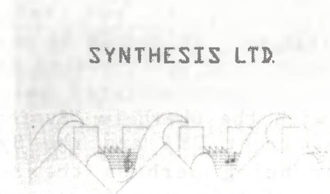

Penguin Cafe Orchestra **Broadcasting from Home** (Editions EG 38). Simon Jeffes' utterly original music is sorta classical, sorta Terry Riley, sorta acoustic Laurie Anderson. It can't be categorized, anticipated or ignored.

David Snow **The Passion and Transfiguration of a Post-Apocalyptic Eunuch** (Opus One 59). Only from a university electronic music studio (in this case Yale) could you get a concept album about eunuchs and cockroaches after a nuclear war. Fragments of jazz, rock, and Zappaesque humor are combined with an unrestrained students' verse.




Data Bank A **The Citadel** (KO2). Rhythm box rock. Last October I described it as similar to Peter Baumann. This time Andy Szava-Kovats' songs are meatier, and his monotone vocals recall Iggy Pop as much as Baumann. $6 from 262 Mammoth Road, Lowell, MA 01854.

Various **Synthesis Ltd.** (ARM 8487). A project of the Electronic Music Club of Saddleback College, although this is not some dry academic noodling. No, here are twelve pop-oriented professional-quality tunes, well recorded and coherent despite the numerous contributors. Saddleback's "best-kept secret" indeed! $5 from the club, 28000 Marguerite Parkway, Mission Viejo, CA 92692.




(continued on page 20)

# MIDI:

## RECORDING MEDIUM OF THE FUTURE

### BY: DAVID ALBIN


MIDI has settled down and is starting to make a real impact in the music world. Some of its impact is for live playing situations, where performers can work with a simplified keyboard setup on-stage and control off-stage synthesizers. Many bands now have "keyboard technicians" who load disks, change channel assignments, and so on for these remote keyboards. But MIDI is of great use in the studio, too. In fact, we are starting to see the development of "MIDI studios" which are in about the same state of sophistication as analog home recording was in the late 70s. The birth of the MIDI studio doesn't mean it's time to trade in your TASCAM or Fostex gear yet, because there are many things that MIDI cannot do. Those who are interested in recording, though, cannot afford to ignore MIDI's implications. This article will describe the concept behind a MIDI studio and the components that make up such a studio.

**The MIDI studio concept.** With a conventional analog studio, sound waves (or electrical impulses, if you're going direct) are recorded on tape. The variations in magnetic flux on the tape are analogous to the sound waves or impulses. Therefore, the analog tape stores the actual sound, and playing back the tape plays back a replica of the original sound.

MIDI recording uses semiconductor memory instead of tape to store sounds. However, to sample sounds and record the actual sounds into memory would use up a prohibitive amount of memory. As a point of comparison, the Emulator II has about a half-megabyte of RAM and stores a little over 17 seconds of sound. If you think of this as one track of a recording, then you would need 4 Megabytes to store 17 seconds of eight tracks of sound. A typical length for a short song would be about 170 seconds (just under three minutes). Therefore, to record eight tracks of even a short song would take about 40 Megabytes of RAM. Although it might be possible to use data compression and the like to save memory, we're still talking about a very expensive system.

The way to solve this problem is to store musical information in "shorthand." Instead of having to use up Megabytes of RAM to store the actual sound, we record <u>data</u> indicating when the note started, what pitch it is, if any pitch changes (modulation) occur during the duration of the note, and when the note stops. This obviously would take up a lot less memory. Again using the Emulator II as a means of comparison, that same half Megabyte which stores 17 seconds of sound can store the <u>information</u> for 90,000 notes!

MIDI, of course, is perfectly suited to be the language that defines notes since it has provisions to communicate note on, note off, pitch, and modulation parameters. This MIDI data is recorded into RAM as data. On playback, this data is played back into any device with a MIDI port. In many ways, what we have done is created a modern player piano -- but one with far greater flexibility, ease of use, and timbral possibilities. It should be noted that the concept of recording data instead of sound existed before MIDI, with the Oberheim "System," the Fairlight CMI and the Synclavier being perhaps the best examples of this type of approach. However, MIDI certainly streamlines matters since it allows for a great degree of compatibility between different makes of equipment.

What makes MIDI so useful for recording is that different data can be assigned to different channel numbers. This is just like assigning sounds to different tracks on a multi-track recorder. These channels can retain independent information, and play back through independent instruments. For example, the data in MIDI channel 1 can play a bass line through a MIDI instrument set to receive on channel 1, the data in MIDI channel 2 could play a brass line through something like a DX-7 set to receive on channel 2, and so on. One other important point that people sometimes overlook is that each channel can contain <u>polyphonic</u> information, so chords can be recorded in a channel just as easily as single notes.

Before describing what makes up a MIDI recording system, let's talk about some of the advantages. By recording in RAM you need not concern yourself with tape problems such as alignment, head cleaning, azimuth adjustments, and so on. Also, long rewind and fast forward times are no longer a problem, which saves a tremendous amount of work time. Finally, there is no signal degradation. Recording a sound on tape adds hiss, distortion, and other problems. With MIDI recording, you are not recording sound but data -- and when the data plays back into an instrument, the instrument will produce its sound in real time. Well, almost real time. MIDI does have some delays, but these are of little consequence unless you are trying to simultaneously play many notes on many instruments.

However, before we write off tape completely, remember that acoustic instruments, the human voice, and many other instruments do not lend themselves to MIDI recording. In practice, although

MIDI recording can stand on its own for some applications it is more commonly used in conjunction with a conventional analog multi-track recorder. The combination of the two is extremely powerful, as we will see later on.

**MIDI studio components.** The most important element is the MIDI sequencer, which is the equivalent of the multi-track recorder. The least expensive way to get started with MIDI sequencing is to hook a MIDI interface to a personal computer like the Commodore 64 or Apple, and record data in the computer's memory. Data can be saved on diskette or cassette, providing the computer has provisions for this. Passport Designs, Sequential Circuits, MusicData, and many others offer this kind of package.

The next step up is a dedicated MIDI sequencer. Probably the unit getting the most attention these days is the Yamaha QX1, which lists for $2,795. It features a built-in 5-1/4" disk drive, eight tracks, and an 80,000 note capacity. The QX1 makes no sound of its own (just like a tape recorder) and is strictly a controller. Of course, Yamaha would like you to control **their** MIDI devices, but you are not limited to using their products.

Several musical instruments also include built-in MIDI sequencers. The Linn 9000, in itself a great drum machine, features a very complete MIDI sequencer. The Emulator II includes an 8 track MIDI sequencer that stores velocity and modulation settings. The Oberheim DSX/OB-8 combination can be retrofitted for MIDI, with DSX data appearing over the OB-8's MIDI port. If you have one of the these instruments, you are already well on your way to having a MIDI studio.

All sequencers are not alike, however. Generally, sequencers are available in 4, 8, and 16 track versions. You should at least be able to punch-in and punch-out, and have optional auto-correction (quantization) to smooth out human timing errors (if desired). Some sequencers store more notes than others, some can record modulation settings and some can't, and some let you bounce data tracks together. Ease of editing notes once they have been recorded .aries greatly as well. The main rule is try before you buy. No one sequencer has everything, so it is up to you to anticipate your needs and choose the device that fits your musical desires.

You will also need several MIDI sound generating devices, and this represents a considerable expenditure. Any keyboard with a MIDI input will do, of course, but don't overlook expander modules and mini-keyboards with MIDI inputs. Casio's CZ-101 lists for under $500 but can provide four independently addressable MIDI single-note lines. The sound quality wouldn't be mistaken for a Prophet, but it's a quick way to add a bass part or whatnot. The Sequential Six-Trak is another good option; it's not too expensive and provides six independent single-note lines which, while not always rich sounding, are flexible and easy with which to work. Korg makes an expander version of the Poly-800, and Roland offers rack-mount MIDI modules. Probably the king of the expanders, though, is the Oberheim Xpander. Not only does it provide up to six individually addressable MIDI voices of superb sound quality, but the MIDI implementation is one of the most universal and complete in the industry. Anyone interested in a MIDI recording setup would be well-advised to look into the Xpander.

**Putting the MIDI recording studio in context.** One of the most exciting things about MIDI recording is that you can usually sync the MIDI sequencer to one track of an analog recorder. This means that the MIDI instruments <u>never have to be recorded on tape</u>. During mixdown, you just run the analog tracks and the MIDI instrument outputs into the master tape. The MIDI tracks are called "virtual tracks" because they never exist on tape, but act as if they do. If you have a sampling instrument like Ensoniq's Mirage or an Emulator II, you can even sample acoustic instruments and sequence them along with the tape tracks.

Have you always wanted to go 24 track but didn't have the money? By synching a 16 track MIDI sequencer to a Fostex B-16, you'll have 31 tracks (32 minus the sync track). That should hold you for a while! This assumes that you have 16 MIDI instruments to sequence, but don't forget that buying 16 high-quality MIDI voices is considerably cheaper than buying a 24 track recorder, let alone a 32 track (and think of all the money you'll save not having to buy 2" tape). Plus you won't need noise reduction on the MIDI instruments. Throw in a SMPL system with chase lock, and you can even lock your MIDI gear to the tape recorder so that you can autolocate the MIDI parts as well as the tape. The mind boggles, as does the bank account. But no matter how you look at it, you will obtain a quality of sound that would have costs hundreds of thousands of dollars a couple of years ago. By that standard, a MIDI/analog recording setup is a bargain, and as with everything else that's computer-based prices will probably decline further in the future.

One other thing about a MIDI studio: you can play tricks you just couldn't play with a normal recording setup. With an analog recorder, if you record a violin line and think it would send better as a cello, you would have to re-record the part. With a MIDI instrument, the note information remains the same and all you would have to do is change patches. Also, MIDI sequencers can typically be slowed down and sped up by outrageous amounts -- much more than the variable speed option on most recorders, and without timbre or frequency shifts! Even the slowest players can play parts in a MIDI studio.

**Acknowledgment.** Much of this article was inspired by a seminar I attended on MIDI and synchronization given by Craig Anderton. When I suggested he write up his talk, he said he didn't have time but was willing to help me with an article provided that I submitted it to <u>Polyphony</u>. For further information, I recommend writing away to manufacturers for literature on MIDI sequencers. And start experimenting! The only way you'll learn how to use MIDI is to start using it. Even if you've been baffled by how it works, everything falls into place once you actually sit down with the gear.

# Practical Circuitry

# Build a Voltage-Controlled State Variable Filter

By: Thomas Henry

Looking for a versatile sound modifier to add to your synthesizer system? Then consider a state variable filter, a filter with a single audio input but three independent outputs (lowpass, bandpass or highpass response). As you might expect, these multiple outputs -- which may be used simultaneously if desired -- offer quite a few more waveshaping options than most other filters! In this installment of "Practical Circuitry" we'll see how to build one of these marvelous devices; and despite the seemingly exotic nature of the state variable filter, the simplicity of the circuit might surprise you. Let's look a little closer at this neat filter.

**What Is A State Variable Filter?** This circuit uses a two pole design, meaning that the response of the lowpass and highpass outputs roll off at a rate of twelve deciBels per octave. The bandpass output obeys a six dB per octave pattern. Although these figures are low compared to the results obtainable with four pole designs, for some applications the two pole filter's gentler rolloff can actually be more musically appropriate. (Editor's note: Certain top-of-the-line synthesizers, such as the Oberheim OB-8, allow for a choice between two and four pole filter response.) Also, a special Q control allows you to add a peak to the critical frequency of the circuit, and this can generate some really wild "wah-wah" effects. Subtle timbre changes are possible too, as you will discover if you build this fascinating circuit.

To be usable within a synthesizer, the state variable filter should be voltage-controlled. Typically you will want a one volt per octave control input; this lets the filter track a VCO so that the resulting waveshape remains unchanged as you play notes up and down the entire keyboard. For dynamic effects, an envelope input should be available too. This allows an ADSR, for example, to modulate the filter, thus generating a timbre which changes with time. Finally, coarse and fine tuning controls are handy to have since they allow you to precisely set the initial cutoff frequency of the filter.

Reliability is always an important aspect to consider in the design of synthesizer modules. This includes, among other aspects, the notion of temperature stability. To make a state variable filter as useful as possible, provision should be made for temperature compensation. This insures that the unit will work predictably and reliably at any temperature. As it turns out, this feature isn't that hard to include in the design.

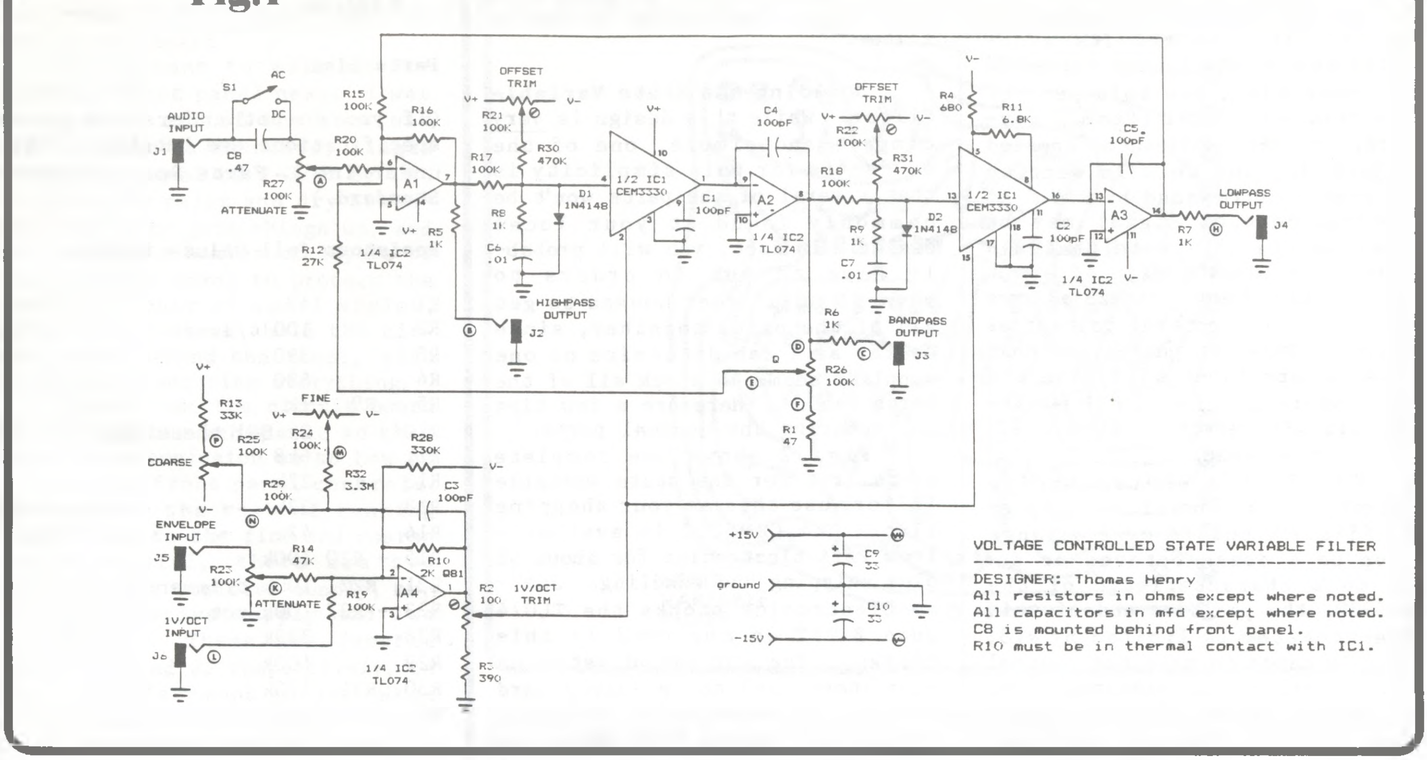
These comments should give you an idea of what a state variable filter is all about and what a good implementation for one should include. Let's look at the schematic for a tested design and check out some of these features in greater detail.

**A Practical Design.** Refer to **Fig.** 1. The mathematical derivation of a state variable filter isn't too difficult, but in keeping with the title of this column, we'll skip over it. If you're interested in seeing exactly how one of these puppies works, be sure to refer to Bernie Hutchins' excellent manual, Laboratory Problems and Examples in Active, Voltage-Controlled, and Delay Line Networks, published in 1978 by Electronotes (see DataBank, p.37). This provides a good mathematical treatment, an explanation of how to add voltage-control to state variable filters, and several practical circuits as well. For our purposes, however, we'll simply consider a state variable filter to be composed of two integrators and a summer (mixer) and assume that hooking these three circuits together correctly does indeed lead to a state variable design.

In **Fig.** 1, the summer is easy to find; it consists of amplifier A1 and associated components. Notice that it sums the input signal, a return path from the Q control, and the "loop" path from the lowpass output. The integrators are composed of op amps, capacitors, and transconductors. For example, the first integrator (reading the schematic from left to right) is composed of A2 and C4, with one-half of the CEM3330 VCA chip acting as the transconductor. (A3, C5 and the other half of the CEM3330 form the other integrator.) The transconductor's purpose is to allow a varying amount of current to flow into the integrator while under voltage control. Ultimately, the control voltage determines the cutoff frequency of the entire filter.

We've been discussing the circuit in rather broad terms so far; let's get more specific and examine some of the actual components comprising the state variable filter. The signal to be filtered is injected into the circuit at

**Fig.1**




J1. Switch S1 lets you choose AC or DC coupling. When filtering audio signals, you can flip S1 to the AC position and C8 will block any undesired offsets. On the other hand, you may have occasion to process DC control signals (this can produce some fascinating control voltage waveshapes); leave S1 in the DC position for this application.

R27 lets you attenuate the input as needed. The state variable filter is designed to handle a standard 10V p-p signal with very low distortion, but if the input rises above this (perhaps when filtering a mix of several "hot" signals), then R27 can tame the input accordingly.

The lowpass, bandpass, and highpass outputs are available at J4, J3, and J2 respectively. R7, R6, and R5 trim the output impedances of these three outputs to a standard value of 1K.

All voltage controlled state variable filters need some sort of transconductor, or voltage-controlled resistor. In the past, FETs, 3080s and some of the newer transconductance op amps (like the LM13600 and CA3280) have been used for this purpose. None of these methods really appealed to me, so I scoured the literature for a better and simpler way to implement the transconductor. After much research, I finally arrived at the CEM3330. This excellent IC offers a number of advantages over the methods described above. First, the chip includes two identical VCAs in one package (and the state variable filter needs two transconductors). Secondly, the CEM3330 has noise and dynamic range characteristics which are clearly superior to FETs or 3080s. Lastly, and perhaps most importantly, the chip is already set up to generate an exponential response, thus obviating the need for matched pair transistors and other exotic components.

Let's look at the basic power requirements of the CEM3330. Pin 10 connects directly to the positive supply. An on-board Zener regulator simplifies the task of generating a suitable negative supply voltage at pin 5; R4 limits the current going to the Zener diode. Pin 18 connects to ground. (Note that balance pins 3 and 17 are also grounded since the balance feature is not needed in this application.)

A special feature of this chip is the way in which the operating mode for the amplifiers may be selected. I decided on Class A operation for this circuit, and so set R11 at 6.8k, as recommended in the spec sheet for the CEM3330. This resistor connects pin 8 to pin 5 (which as we have seen, is the negative supply pin for the chip).

So far we have looked at features that both VCAs within the CEM3330 share in common. Let's now consider some aspects of the individual VCAs. To this end, we'll examine the half of the chip associated with op amp A2 in **Fig. 1**, keeping in mind that the other half works in a similar fashion. R17 is the input resistor for the VCA, and converts the input voltage to a current in a range with which the CEM3330 can work efficiently. R8 and C6 (along with C1) compensate the gain cell, and D1 prevents latch-up during unusual conditions. R30 in tandem with R21 allows for nulling out input offsets, about which more will be said later.

But what about the voltage control aspects of the VCA? Well, refuting Murphy's law, things really do work out well here. Although the CEM3330 contains all sorts of neat logarithmic converters and whatnot, we can bypass the entire lot, thus greatly simplifying the design and adding to the temperature stability of the circuit as well. The normal linear and exponential control inputs (at pins 7 and 6, respectively) are ignored entirely and the control voltage is instead applied directly to pin 2. This is a really slick approach; note that pin 2 is even ganged with pin 15, the control input for the other transconductor. What could be easier!

Let's move on and now consider the control-voltage summer for the circuit. A4 and its associated components handle this job. An exponential, one volt per octave control signal (from a keyboard, for example) can be applied to jack J6, and this is sent to the summer unimpeded by R19. An envelope-control signal may be injected via J5. Note that R14 gives this input a gain of about two, while attenuator R23 allows you to tame the control voltage as desired. This duo guarantees that the envelope input will give the broadest range of control possible, allowing anything from small to monstrous sweeps.

R25 is the coarse tuning control. This control will sweep the filter's cutoff over a range of up to a dozen octaves or so. Due to a characteristic of the CEM3330, the positive end of this potentiometer is limited by R13, and this tends to make the control more useful in its midrange. For similar reasons, R28 adds in a fixed negative offset to the summer network. To make smaller adjustments possible, R24 and R32 implement a fine tuning control. This pot covers a musical interval of about a fifth.

The summer mixes all of these just-mentioned signals, and attenuates the result to a suitable range via R10, R2 and R3. R2 may be adjusted to give the filter a precise one volt per octave exponential response.

The CEM3330 is temperature compensated for second order effects, which are the most troublesome. To compensate for the remaining first order effects, we must apply a little ingenuity. Notice that R10 is actually a thermistor which has the characteristic of changing resistance in a manner opposite, but proportional, to the undesired changes going on in the exponential circuitry internal to the CEM3330. Thus, undesired changes with respect to temperature are automatically concealed by this simple mechanism. R10 is a standard Tel Labs Q81 thermistor, with a value of 2k and a temperature coefficient of +3600 ppm/degree. In order for it to do its duty, it must be in thermal contact with the CEM3330.

As you can tell, this circuit has a number of professional features, and yet the design isn't too outlandish. In fact, you may be surprised to learn that the entire circuit can be built with just two chips! Let's dig in now and see exactly how to construct and adjust the state variable filter.

**Building the State Variable Filter.** While this design is very compact and simple, one of the tradeoffs for this simplicity is that several of the parts won't be commonly found at your local dealer. In fact, you will probably have to put in orders to several mail order houses to get all of the parts together, since as far as I can determine no one supplier seems to stock all of the parts needed. Here are a few tips on procuring the unusual parts.

**Fig. 2** shows the complete parts list for the state variable filter; use this as your shopping list. The CEM3330 is available from PAiA Electronics for about $8 plus shipping and handling. Jameco Electronics stocks the TL074 quad BIFET op amp used in this design. The 100 pF polystyrene capacitors used to be fairly hard to find, but fortunately PGS Electronics now stocks them for about a quarter apiece. The 2k Q81 thermistor is made by Tel Labs; if you can't locate a 2k thermistor, you can always use the more readily available 1k type, but then you will have to halve the values of R19, R14, R29, R32 and R28 to compensate. The 1k Q81 thermistor is available from PGS Electronics if you elect to go this route. (See Databank for suppliers' addresses.) Write to these places for catalogs and ordering information (and make sure you mention Polyphony!). The remaining parts are easy to find at local electronics stores.

Now here's a word to the wise. If the parts list specifies a certain type of capacitor, then use that type only! On the other hand, if no type is mentioned, then use whatever you have handy. Similarly, IC2 must be a BIFET type op amp package; don't even consider using a standard bipolar op amp in this circuit!

After collecting all of the components, you are ready to start building. Whatever mode of construction you choose, be certain you apply neat and orderly techniques since any exponential circuit that covers a ten octave range or better is subject to a number of stray capacitance problems. All in all, the best route is to go with a printed circuit board. To simplify the task of putting a board together, **Fig. 3** shows the artwork for a test design. (If you still harbor the feeling that a state variable filter is "complicated," notice how compact and simple the circuit board is!) **Fig. 4** shows the accompanying parts placement guide for the circuit board.

If you're using a circuit board, simply load the board using the parts list and the parts placement diagram as your guides. Don't forget to install the jumpers (denoted by the letter "J"), and be sure to use sockets for the ICs. Also, don't install R10 just yet; this occurs in a later step.

## Fig. 2

**Parts List**

(If you do not understand parts specifications see Databank, p.@@, under **Int'l. Parts Specification Standard.**)

**Resistors** (all values in Ohms)

| | |
|---|---|
| R1 | 47 |
| R2 | 100 trimmer |
| R3 | 390 |
| R4 | 680 |
| R5 - R9 | 1k |
| R10 | 2k Q81 thermistor |
| R11 | 6k8 |
| R12 | 27k |
| R13 | 33k |
| R14 | 47k |
| R15 - R20 | 100k |
| R21, R22 | 100k trimmer |
| R23 - R27 | 100k pot |
| R28 | 330k |
| R29 | 100k |
| R30, R31 | 470k |
| R32 | 3M3 |

**Capacitors**

| | |
|---|---|
| C1 - C3 | 100p |
| C4, C5 | 100p polystyrene |
| C6, C7 | 10n (0.01 uF) |
| C8 | 470n mylar (0.47 uF) |
| C9, C10 | 33u electrolytic |

**Semiconductors**

| | |
|---|---|
| D1, D2 | 1N4148 or equiv. diode |
| IC1 | CEM3330 VCA chip |
| IC2 | TL074 quad op amp |

**Hardware**

| | |
|---|---|
| J1 - J6 | 1/4" phone jacks |
| Misc. | Sockets, panel, wire, knobs, hardware, solder, etc. |

And by the way, C8 mounts behind the front panel, rather than on the circuit board.

You'll want to make an attractive front panel next. I was able to fit all of the controls easily behind a standard, single width rack panel of 1-3/4" by 19". I used epoxy paint and dry transfer letters to jazz things up, and then applied a number of layers of clear plastic spray to protect the finish. A pair of small angles, formed of aluminum, hold the circuit board behind the panel, with #4 hardware securing everything.

Notice that the circuit board is keyed to the schematic so that there is no confusion as to how to hook up the front panel controls. To simplify the task of running power lines to the fine and coarse tuning control, there are two holes for the positive supply, two for the negative supply and two for ground. These are denoted +15V, -15V and G, respectively, on the parts placement guide. When comparing the schematic to the parts placement guide, keep in mind that V+ is +15V and V- is -15V.

When you're done wiring up the front panel to the circuit board (and don't forget C8 now, which connects between J1 and R27), consideration should next be given to R10, the thermistor. Refer to **Fig. 5**; note how the thermistor mounts directly on top of the CEM3330, thus insuring good thermal tracking. Two holes, located at either end of the CEM3330, are provided on the circuit board to facilitate this operation. To minimize confusion, they are marked by asterisks on the parts placement guide. Before soldering R10 in place, smear some silicone heatsink grease on top of the CEM3330, and then press the thermistor into it. Solder this in place, and you've completed construciton of the voltage-controlled state variable filter.

**Fig.3**

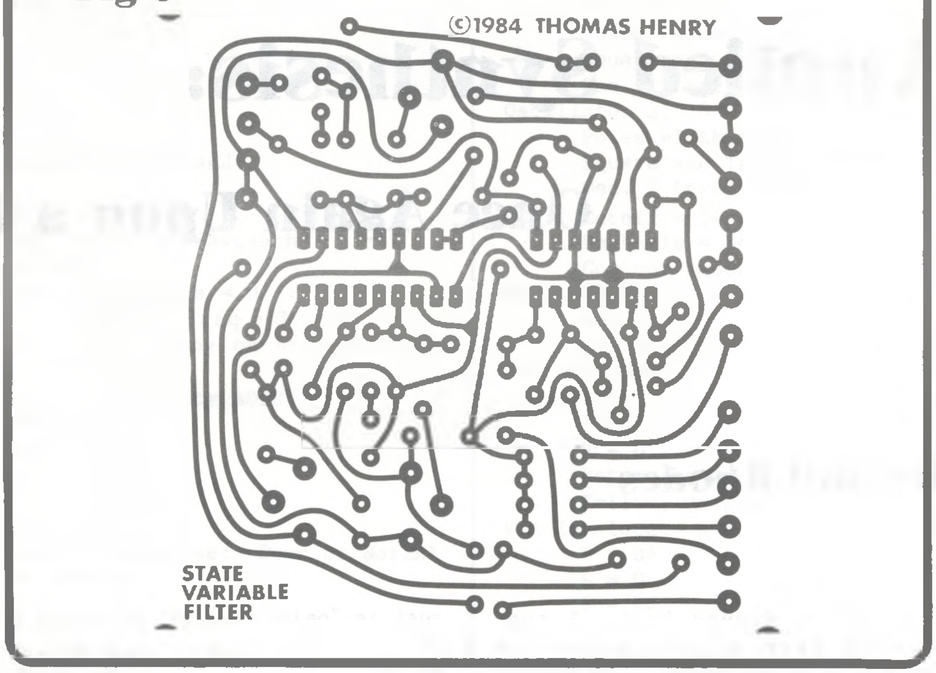
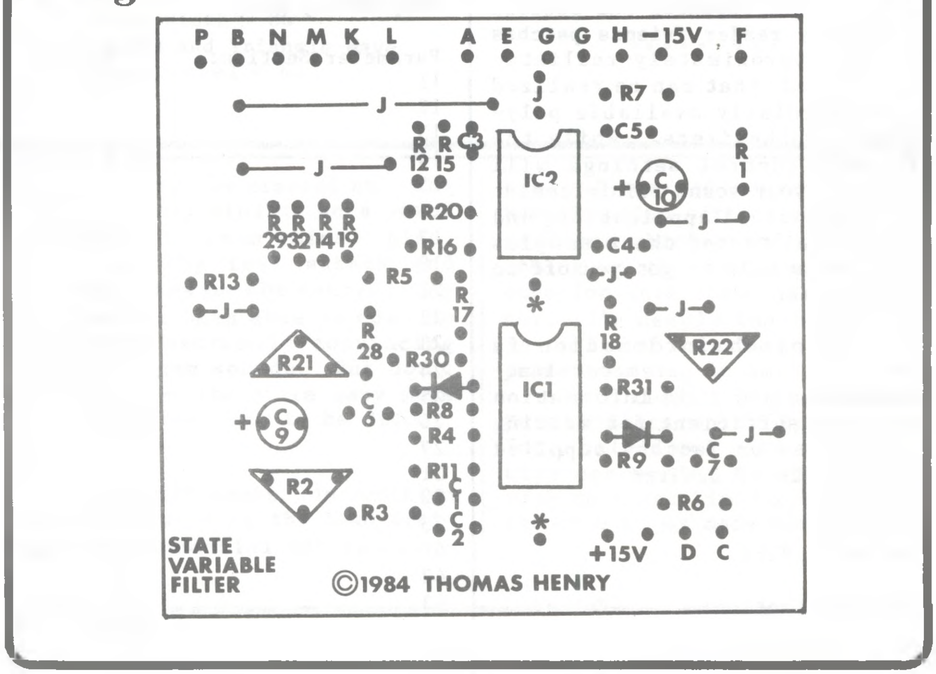



**Fig. 4**




**Adjusting and Using the Filter.** To tweak volts per octave trimmer R2, simply connect a control voltage from your keyboard to the 1V/Octave input at J6, and then run a patch cord from the lowpass output (at J4) to an audio amplifier. Set the amplifier for a comfortable listening level. Turn down the audio input attenuator, R27, completely. Now turn up the Q control, R26, until the filter starts oscillating. Adjust the coarse and fine tuning controls, R25 and R24, until the oscillation is around 500 Hz or so. Now tweak the filter as you would any VCO until a perfect 1V/Octave response is obtained.

Adjusting the offset trimmers, R21 and R22, is slightly more difficult. I found that these trimmers simply didn't behave the way I thought they would, and I'm still a little perplexed by it all. In theory, the filter is rapidly swept across its range while R21 and R22 are adjusted for minimum deflections at the lowpass output. However, for one reason or another, this method fell flat for me, with the output assuming all sorts of unusual DC values depending on the settings of the tuning controls. I finally arrived at the following intuitive method which has given me good results. While monitoring the lowpass output on an oscilloscope, turn up the Q control, R26, until the filter oscillates. Now sweep the filter across its entire range while watching the sine wave for

**(continued on page 13)**

# Applied Synthesis:

## Once Again Upon a Piano

### By: Bill Rhodes

A few issues back, I suggested that if one needed an acoustic piano sound, one should use a piano instead of a synthesizer. Well, after all the nasty mail, I have repented and decided to give my reader friends patches for an "approximately-realistic piano sound" that can be realized on commercially available polyphonic synthesizers. While the ultimate control settings will depend on your sound reinforcement system, musical application, and (of course) taste, these samples should be a help to get you off to a good start.

The patch information is given in terms of parameter function and value; this information should be sufficient for writing out patches on factory supplied patch sheets if desired.

**Poly 61** by Korg

| Parameter Section: | Value: |
|---|---|
| 11 | 8' |
| 12 | 3 |
| 13 | 3 |
| 21 | 4' |
| 22 | 2 |
| 23 | 1 |
| 24 | 1 |
| 31 | 7 |
| 32 | 0 |
| 33 | 1 |
| 34 | 3 |
| 41 | 0 |
| 42 | 14 |
| 43 | 0 |
| 44 | 12 - 14 |

(Note: The exact value for the above depends upon the desired sustain "pedal" characteristics.)

| | |
|---|---|
| 51 | 1 |
| NO MG | No Value |

**Poly 800** by Korg

| Parameter Section: | Value: |
|---|---|
| 11 | 3 |
| 12 | 2 |
| 13 | 1 |
| 14 | 1 |
| 15 | 1 |
| 16 | 1 |
| 17 | Volume |
| 18 | 1 |
| 21 | off |
| 22 | off |
| 23 | off |
| 24 | off |
| 25 | off |
| 26 | off |
| 27 | off |
| 31 | 0 |
| 32 | 0 |
| 33 | 0 |
| 41 | 56 - 66 |
| 42 | 0 |
| 43 | 1 |
| 44 | 2 |
| 45 | 0 |
| 46 | 2 |
| 47 | 0 |
| 48 | off |
| 51 | 0 |
| 52 | 19 |
| 53 | 22 |
| 54 | 25 |
| 55 | 0 |
| 56 | 21 - 26 |
| 61 | 0 |
| 62 | 0 |
| 63 | 0 |
| 64 | 0 |
| 65 | 0 |
| 66 | 0 |
| 71 | 0 |
| 72 | 0 |
| 73 | 0 |
| 74 | 0 |
| 75 | 0 |
| 76 | 0 |
| NO MG | No Value |

**Poly-6** by Korg

Waveform: Sawtooth
PWM: Off
Sub Osc: Off
All MG: Off
VCF: Cutoff = 5.5
 Resonance = 0
 E.G. = 0
 KBD Track = 0 (vary for best results)
Envelope: A = 0
 D = 7.5
 S = 0
 R = 6 (vary for best results)
No Chorus

**JX3P with PG200** (Edit Section) by Roland

DCO: 1
Range: 16'
Waveform: Sawtooth
All modulation: Off
No HPF
VCF: Cutoff = 3
 Resonance = 0
 Pitch Follow = 100%
 + ADSR Polarity
VCA Mode: Env.
Envelope: A = 0
 D = 3
 S = 1 to 2
 R = 4

**Jupiter 6** by Roland

(Note: Factory Preset (A-1) is a very good piano sound, especially in the lower registration. The low end of the piano is duplicated rather well by the use of two oscillators in different octaves. However, the top two octaves, when played alone without the left hand, leave a lot to be desired because the timbre in the upper end seems tinny and thin. You can achieve a good sound by MIDIing two synths together; find one that sounds good on the bottom registration and another that is good in the upper octaves. Following is an alternate sample sound.)

VCO - 1 Waveform: Sawtooth and
    Square 8'
VCO - 2 Waveform: Triangle and
    Sawtooth
VCF: Cutoff = 3 to 5
    Resonance = 0
    KYBD = 5
ENV, LFO: Off
ENV 1: A = 0
    D = 5
    S = 0
    R = 1
ENV 2: A = 0
    D = 4
    S = 4
    R = 3 to 5

**Juno 60** by Roland

No LFO
DCO Waveform: Sawtooth
No HPF
VCF: Resonance = 0
    Cutoff Freq = 3 to 3.5
    (depending on brightness)
    ENV = 3
    + Polarity ADSR
VCA: +5
ENV: A = 0
    D = 5
    S = 2
    R = 5
Octave Transpose: Down or Normal
No Chorus

**Memory Moog** by Moog

Oscillator 1: 8'
    Sync On
    Pulse Width = 86
    Square and Triangle Wave
    Osc. 1 Level = 44
Oscillator 2: 8'
    Pulse Width = 54
    Square and Triangle Wave
    Osc. 2 Level = 23
Oscillator 3: 8'
    Pulse Width = 50
    Square and Triangle Wave
    Osc. 3 Level = 64
VCF: Cutoff = 54
    Emphasis = 34
    Contour Amount = 0
Modulation: LFO on Triangle (Rate
    = 20)
    Destination = PW1, 2, 3
VCF Envelope: A = 0
    D = 100
    S = 0
    R = 6
    Return to Zero, Keyboard
    Follow, Release On
VCA Envelope: A = 7
    D = 88
    S = 0
    R = 46

There's your collection of piano patches...with no strings attached! Good luck with them.

•••••••••

## Practical Circuitry...

signs of flattening on either peak. Move back and forth between R21 and R22, adjusting either or both until the sine wave remains pure across the entire audio spectrum and centered about ground. This seems to set the optimum point for the filter and gives fine audible results.

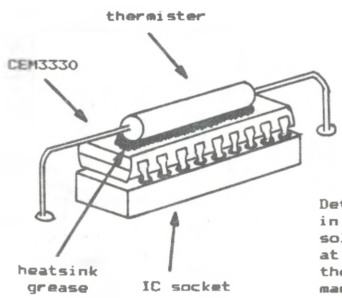
If anyone comes up with a better method for adjusting the DC feedthrough, or can explain why the first method gives weird results, please write to me in care of Polyphony so that I can pass the information along. And by the way, the DC feedthrough is very small anyway. In fact, for most non-critical applications, R21, R22, R30 and R31 could probably be removed altogether with your ears being none the wiser!

At this point you're all set to start using your new state variable filter. If this module is new to you (as it was to me), then I predict you'll be quite surprised by the quality of the sounds possible with it. One of my favorite effects is generated by inputting a square wave and listening to the highpass output, with the Q really cranked up. The result is very similar to a vowel uttered by a human voice. Also, by using very fast attack and release times on the controlling ADSR, I have been able to come up with some extremely convincing Hammond organ sounds. But don't stop there -- there are many more sounds just waiting to be discovered.

**Acknowledgments.** In addition to writing some of the best articles on exponential voltage-controlled current sources, Bernie Hutchins also offered several suggestions which eased the task of coming up with the control structure for this state variable filter. I'm deeply indebted to him for all of his help over the years.

Doug Curtis, the president of Curtis Electronics Specialties Inc., helped out with a number of tips concerning the CEM3330. I wish to thank him for his patient response to my many questions.

**Fig.5**




Detail showing how tne thermister is mounted in thermal contact with the CEM3330. Note that solder holes are provided on the circuit board, at either end of the socket, for installing the thermister in this fashion. The two holes are marked on the parts placement guide with asterisks.

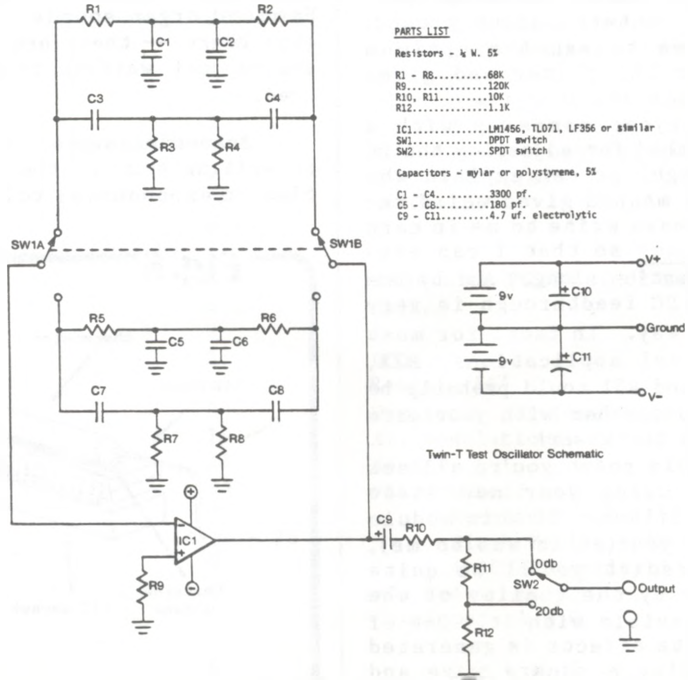
# TWIN-T TEST OSCILLATOR

By: Jack Orman

A good, stable signal source is invaluable on the workbench or in the studio for setting levels, so I designed and built a simple and inexpensive sine wave oscillator to fulfill these functions. Basically, IC1 is a "Twin-T" oscillator with two switch selectable frequencies. The "Twin-T" network of resistors and capacitors in the feedback loop of the op amp determines the sine wave generator's frequency. The network consisting of R1 - R4 and C1 - C4, when selected by the DPDT switch SW1, gives the oscillator a frequency of 700 Hz while R5 - R8 and C5 - C8 yield a 13 kHz signal. The exact frequency and purity of the sine wave depends on closely matching the components; 5% resistors and capacitors gave excellent results for me. Resistors R10 - R12 form a divider network on the output to attenuate the high level signal. SW2 selects between the normal output or one that is attenuated by 20 dB. This switch is also necessary when using the oscillator to test the calibration of a tape deck.

The procedure to test a tape deck for proper bias is as follows. Turn on the oscillator, set SW1 to 700 Hz, and set SW2 to the 0 dB position. Connect the output of the test oscillator to the line input of one channel of the tape machine. Adjust the deck's input level for a 0 VU reading. Now flip SW2 to the -20 dB position and record about 15 seconds of the tone. With SW1 set to 13k Hz and SW2 back on 0 dB, again adjust the input level for a 0 VU reading. Set SW2 to -20 dB and record 15 seconds of the tone. Connect the line out of the tape deck to the input of another unit that has a VU meter on it, such as a mixer or another tape deck. Rewind the tape and play it back. When the 700 Hz tone comes on, adjust the second machine's input level for a 0 VU reading. Let the tape play and notice the difference between the 700 Hz tone level (0 VU) and that of the 13k Hz tone (this should also be 0 VU but may vary). A difference of plus or minus 2 dB is allowable. If the 13k Hz tone level is significantly different from that of the 700 Hz tone, the bias level of the recorder needs to be adjusted to even out this difference. Repeat this calibration procedure for each channel of the recorder. Note that any form of noise reduction must be off when this test is made or the results will be invalid.

I mounted this circuit in a small plastic box and keep it handy in the studio for when I need to do any testing or calibration. Although this unit is not as versatile as a complete function generator, its low cost and usefulness make it a welcome addition to my workbench and studio.




PARTS LIST

Resistors - ¼ W. 5%

R1 - R8..........68K
R9...............120K
R10, R11.........10K
R12..............1.1K

IC1..............LM1456, TL071, LF356 or similar
SW1..............DPDT switch
SW2..............SPDT switch

Capacitors - mylar or polystyrene, 5%

C1 - C4..........3300 pf.
C5 - C8..........180 pf.
C9 - C11.........4.7 uf. electrolytic

Twin-T Test Oscillator Schematic

# This is the last issue of Polyphony.

EDITORIAL

In the 10 years that Polyphony has been in publication it has been supported by a devoted following of readers who were on the forefront of knowledge in the somewhat esoteric field of electronics and computers as they were applied to the production of music. Now that field has expanded into a major market area. Electronics and computer technology in the music industry is the norm rather than the rarity it once was, yet we find that musicians are asking for still more knowledge. There is an expanding base of equipment to choose from but a void in the area of information on how to use this new technology. We know of no one else that would be more qualified to fill the information void than Craig Anderton and the folks that have produced Polyphony.

Beginning with the next bi-monthly issue Polyphony merges with and becomes ELECTRONIC MUSICIAN Magazine. With this name change comes a change and expansion it the area of coverage. Emphasis will include multi-track recording techniques, MIDI applications, the application of Electronic Effects, Video Production basics, computer applications and more. Along with this expanded editorial coverage and name change comes a change in the look; Electronic Musician will be completely typeset, professionally produced and includes computer generated graphics and illustrations. We're going for expanded newstand sales, added subscriptions and we will be distributed and represented at the June NAMM Show.

We've taken in to consideration all of the things that you, our readers, have been asking for over the past year. We hope that you enjoy ELECTRONIC MUSICIAN as much as we think you will and that its expanded coverage will meet all of your expectations.

Cordially,

Linda Simonton
Managing Editor

ELECTRONIC MUSICIAN

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# ON LOCATION: Musicom 84, Holland

## By: Craig Anderton

For me, Musicom 84 (held November 9 - 11 at the Rotterdam Hilton in Holland) was the electronic music event of 1984. Of course, there was the standard fare of exhibitors showing their latest products, just like a NAMM or AES show. But what really set Musicom apart were the special exhibits, workshops, concerts, and lectures. Not only could you catch up on what various manufacturers were up to, there was a valuable opportunity to interact and learn from independent researchers, studio owners, academics, and hobbyists in order to obtain a complete electronic music experience.

The credit for Musicom's unique character lies with its originator, Felix Visser, a prominent figure in the Dutch electronic music hardware scene (see accompanying story). His vision was to bring people together in this field, since individuals cannot unravel the secrets of sound by themselves; he feels they need interaction and stimulation from a variety of sources. As Felix said, "The tools of (electronic music) have to be improved, and if events like Musicom don't happen, these tools will not be improved. We need the right input from the right people. Perhaps a common language will evolve that will allow us to make real progress in finding out the secrets of sound. I want all this electronic stuff, the bits and pieces, to evolve into a real instrument...and besides, I want to be around people with ears."

I particularly liked Musicom's unusually non-commercial nature. The best exhibit space wasn't auctioned off to the highest corporate bidder; it was donated to avant-garde artist Piet Jan Blauw, whose work combines visual arts, performance arts, and electronic music. He considers the artist to be as important as the art, and uses electronics to help integrate himself into the artistic process. For example, one of his "performances" is to wear clothes with built-in speakers, sound generators, vocoders, signal sources, and microphones (to pick up external sounds), then simply walk around and see what happens. His artwork ranges from robots to video, but perhaps his most popular work at Musicom was a "keyboard radio." Each key of a three-octave keyboard connects to an individual radio, which can be tuned by the player. As you press down the keys, you get dialogue, music, static, strange radio noises, and so on -- sort of a sampling device for the electromagnetic spectrum. Packaged in blue plexiglass, people simply couldn't resist coming up and pressing the keys to see what would happen next.

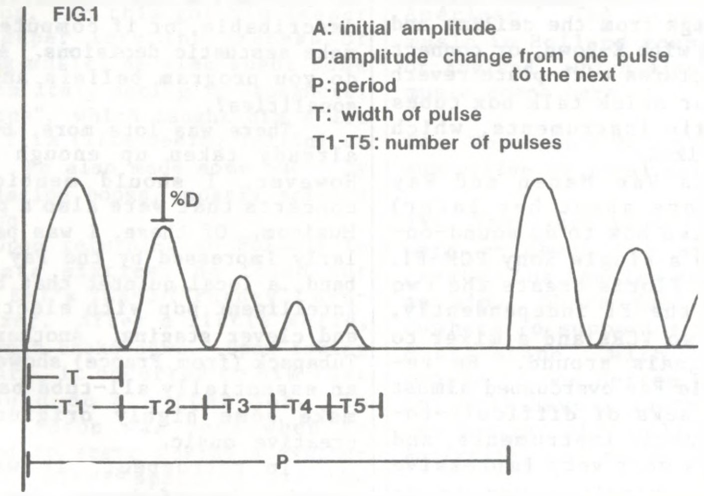
Another interesting exhibit was put on by Steim (address: Groenburgwal 25, 1011 HR Amsterdam), an experimental Dutch studio founded in 1969 with a subsidy from the Ministry of Cultural Affairs and the city of Amsterdam. Steim was originally founded as an open studio for composers, but has grown into a multi-studio research center devoted to music, visual/-theatrical disciplines, and electronics. Musicom donated five rooms at the Hilton for Steim to show their work, and for many attendees these rooms were the best part of the show.

One of the Steim highlights was a remote MIDI controller for the Yamaha DX-7, designed by Michel Waisvisz. This novel controller consists of two hand-sized plates, each of which straps on to one hand. Each plate has a matrix of switches that covers one octave, plus mercury switches so that changing the hand position changes the octave range. There are also patch advance buttons plus an infrared sensor in each hand controller. The latter controls dynamics in such a way that varying the distance between the two hands makes the sound louder of softer. It's difficult to convey the drama created by someone playing in this manner, but I think you get the idea...spreading the hands apart in an expansive gesture increases the volume, while turning the hands rapidly can give dramatic octave jumps. At any rate, normal remote keyboards look somewhat less exciting after seeing the Steim hand controllers in action.

But that wasn't all by any means. Steim also showed a "museum piece" of an early programmable drum machine, consisting of a trap set with solenoids and controllers. Another room, dramatically darkened, contained a number of tuned metal ribbons, connected to contact microphones, that were occasionally struck with solenoids. The sound was shimmering, eerie, and very complex. There was also a "low-tech percussion controller" consisting of an 8 track tape loop upon which had been recorded such musique concrete sounds as cymbals rolling across the floor, percussive sounds, and so on. These passed through VCAs that were triggered by drum pads. As pointed out by the designer, this approach combines musique concrete and emulation effects but at a pretty low price (assuming you have a multitrack recorder handy, which most studios do).

Steim also made much use of the "Vosim" generator, which is a type of burst generator (see **Fig. 1**). Simply stated, a trigger establishes the basic frequency; this trigger initiates a series of bursts with variable amplitude, period, percentage of amplitude change, and number of pulses. After Musicom, I was able to play with the Vosim generator at the Sweelinck Conservatory studio, as well as hear tapes made with the device. While not necessarily a "new" signal generator, it was new

FIG.1




to me and I was quite impressed. With a single Vosim you can make traditional oscillator sounds, hard sync effects, sounds which resemble traditional filtering effects, and much more. Most of the versions I saw used software-generated waveforms, although it seems that you could make a decent approximation with some Curtis and/or SSM chips. Oh yes, and Vosim is great for percussive effects -- every drum machine should have one.

Not all of the Steim exhibits were ultra-technical, however. The "Interactive Frog Pond" (by Felix Hess) was whimsical, but highly intriguing: It consisted of a large number of self-contained units about the size of a transistor radio, designed to mimic the effects of frogs interacting with each other. For example, if you walked into the room and made noise, the boxes would stop making sounds. Then if you were quiet, eventually one box would start up, followed by more boxes interacting with the sounds from that box. The interactivity included the rate of repetition of the sound, volume level, and other factors. One of the fun stories that came out of Musicom occurred after the show closed one day and the room was closed up. Because there had been a lot of noise, the "frogs" were not making any sound and so the Steim people forgot to turn off the batteries. Several hours later, a security guard heard strange sounds coming from behind the door. He decided to investigate, but of course, upon opening the door and making some noise the "frogs" stopped chirping. Further confused, he closed the door again only to hear the sounds emanating a while later, which once more shut up when he went to see what was going on. (I suppose this was the "how to drive a night watchman crazy" portion of the exhibit.)

Of course, there were plenty of exhibitors showing their wares; Linn introduced the Linn 9000 to the general public for the first time at Musicom, and it is a very impressive drum machine. Actually, it's more than a drum machine since it features a sophisticated MIDI sequencer (conceptually sort of like combining an Oberheim DMX and DSX in one package). Drum and keyboard sequences are stored on a micro-floppy, and you can sample some of your own percussion effects in addition to using the existing drum sounds. The drum pads on the box also have dynamic response -- a nice touch (pun intended). It's in the same high price range as other Linn products, but for professionals it should be a very popular item.

Many American and Japanese products were also present, mostly distributed by European distribution companies...but we've covered most of these in previous On Location columns dealing with NAMM shows. There were also a number of European products that I had never seen before. Generally, they're pretty similar to what we have here (MIDI interfaces, synthesizers, a zillion MIDI sequencers, etc.), but it's always instructive to remember that the USA and Japan are not the only game in town when it comes to electronic music instruments.

While several computer-based products were designed for the Commodore-64, products were also available for Sinclair computers (ZX-81 and the newer Spectrum). Many Europeans seem to be Fairlight fanatics; but of course, the Fairlight is beyond the financial reach of all but the most affluent musicians. As a result, there are a number of low-cost sampling programs designed for use with home computers. While obviously not capable of matching the specs of something like a Fairlight or Emulator, these devices provide a lot of fun and can give musically useful results.

One of the best of these was the DS:3 sampling add-on for the Apple II or IIe, demonstrated by John Molloy of Mainframe (an English synth duo). The basic package (about $300 in England) consists of a plug-in card, software, and cabling. The sampling system (30 kHz, 8 bit) allows for four-voice, polyphonic, splittable operation from either the Apple keyboard or optional five-octave AGO keyboard (add about $575 for the latter). The DS:3 also accepts and supplies synchronizing signals for drum machines, synths, etc. There are two modes for waveform editing, aural or via a monitor display. Although the maximum sampling time at full fidelity is only 1 second (2 seconds at reduced fidelity) and there is no looping, according to John improvements are in the works. Finally, there's a sequencer that makes good use of the Apple's capabilities. Mainframe has a record out, "Into Trouble with the Noise of Art" (recorded on an old four-track in their bedroom studio) which not only convincingly demos the system but is also pretty entertaining in its own right. For more information contact Greengate, 24 Missden Drive, Hemel Hempstead, Herts, England HP3 8QR.

Another sampling device, the Action Replay, is a Sinclair Spectrum add-on and is manufactured by Ricol Electronics (48 Southport Road, Omskirk, Lancs, England L39 1QR). In addition to providing sampling with looping, it can also do delay and primitive harmonization. The sound quality is similar to something like an Instant Replay; cost is approximately $250.

There were also a large number of high-tech products there -- in addition to the Fairlight, people had a chance to check out the new Emulator II, Kurzweil, PPG, Synclavier, and Synergy. You could also look at some of the many European music magazines, and I was quite surprised that the much smaller European population base could support such a diversity of magazines in the same genre as Polyphony. This is probably because the European magazines are really several magazines rolled into one; you don't have the extreme specialization as experienced in this country, where you have separate magazines for keyboard players, guitarists, recording, high-end recording, etc. Instead, there's basically one or two magazines for each country which contain a variety of musically-related articles.

Musicom also featured numerous lectures and workshops, including two by yours truly ("The Synchronized Electronic Music Studio" and "Getting the Most Out of Signal Processors"). The former was an overview of SMPTE, MIDI, and synchro-sonic techniques; the latter, a grab bag of tips and techniques for synchronizing and generally utilizing signal processors to the fullest extent. Unlike American conventions, except for a couple of cases all the lecturers were invited by Musicom and were not representing any kind of corporate interest. In other words, instead of having a salesperson or engineer hyping a company's latest product or product, all the lectures and workshops were given by individuals who work in the field -- authors, studio owners, and other professionals. Of particular interest was a lecture by Terry Fryer on the economics of running a small commercial studio, and the use of synchronization when doing sound work for TV commercials. Terry is one of those people who is not well-known to the general public, yet if you have a TV, you've heard his work on dozens (if not hundreds) of commercials and spots.

Swiss composer Bruno Spoerri gave a workshop that went heavily into musique concrete and electromechanical devices. After Musicom I travelled down to Switzerland to do an interview with him, which will appear in an upcoming issue of Polyphony. I learned a lot, and so did the audience: After all, not too many people suspend piano strings from the ceiling and drive them with E-Bows, or connect metal sculptures with plate reverb drivers...or stick talk box tubes into acoustic instruments, which are then miked.

Floris Van Manen and Fay Lovsky (more about her later) demonstrated how to do sound-on-sound with a single Sony PCM-F1. Basically, Floris treats the two halves of the F1 independently, and uses two VCRs and a mixer to bounce signals around. He recorded while Fay overdubbed almost a dozen tracks of difficult-to-record acoustic instruments, and the results were very impressive -- finally, here's a way to do inexpensive digital overdubbing while maintaining extremely high sound quality. (By the way, PCM recording is very big in Europe; every studio I visited, and most individuals, had some kind of PCM setup. It definitely inspired me to get one, because as more than one person noted, you save enough on tape costs so that the unit pays for itself in a short period of time.)

There were several seminars geared mainly to the Dutch market about promoting music, doing film work, and so on. Klaus Netzle (Polyphony, Re:View, October 1984) gave a talk on applications of music computers and there were also talks about FM synthesis, demos of various pieces of equipment, and so on.

Several people invited by Steim gave lectures on the final day of the show. Nicholas Collins, an independent recording artist from this country, talked about using home computers for automated mixdown; Richard Teitelbaum described a computer-controlled, electro-mechanically driven, acoustic piano player system called the Digital Piano. The basic premise is that music played on one piano is read into computer memory, where it can be stored, processed, and/or sent to two Marantz Pianocorder systems. Luc Steels (University of Brussels) gave a talk about the use of Artificial Intelligence in composition. He feels there are many similarities between problem solving and musical composition; his goal is to create instruments that truly extend the capacity of the performer because they actually bring in musical knowledge and experience. This naturally leads to other questions, such as whether aesthetics are compositionally describable, or if computers can make aesthetic decisions. And how do you program beliefs and personalities?

There was lots more, but I've already taken up enough space. However, I should mention the concerts that were also a part of Musicom. Of these, I was particularly impressed by the Fay Lovsky band, a local quintet that blended intelligent pop with electronics and clever staging. Another band, Tubapack (from France) showed that an essentially all-tuba band can make some highly original and creative music.

In retrospect, it was the combination of demos, lectures, workshops, and concerts that gave Musicom its unique flavor. The theme of the show was not hardware; it was sound, concepts, creativity. I hope there will be another one -- the day after I hear the dates, I'll be making plane reservations.

## *The Man Behind Musicom: Felix Visser*

Felix Visser (b. 1943) is well-known in the Dutch music scene. His company, Synton, has made cost-effective modular synthesizers and top-of-the-line vocoders for several years; he also distributes high-tech gear from a number of countries in Holland. What's less well-known is his musical background as a drummer (he started at age 15) and composer. In fact, just before Musicom he finished scoring the movie "Voro Nova" using nothing but a Fairlight (one of his favorite instruments). Interestingly, he sampled sounds from the movie itself and, by various permutations, turned them into the soundtrack.

Felix started playing in bands regularly at age 19, performing "typical nightclub jazz stuff in obscure places in Germany". He became disillusioned with the long hours, mercenary club owners, and general sleaziness of the music scene at that time; one day in the mid-sixties, he had enough and simply stopped playing drums. Next he started experimenting with tape and musique concrete techniques. An opportunity to do some film music came up, and this got him deeper and deeper into the process of making music; he also spent some time in Italy acting in "spaghetti Westerns", which taught him a lot about film (like seeing Fellini at work). He also made some bucks by translating books, mostly science fiction.

Upon returning to Holland in the late sixties, he found out about EML's VCS-3 ("Putney") synthesizer. After a little while, the synth broke down. Upon taking it apart, it occurred to him that he could do a better job. In 1972, Synton was born when he decided to start making a synthesizer. After spending all the money from a bank loan and still not having a complete synthesizer, working more or less out of desperation he made a phase shifter (the effect was pretty hot at that time), and the company was off and running. Synton then started making vocoders, which were very well received, and the company was well established.

As Felix started traveling to the states to show his products, he eventually got asked about distributing products in Holland; Synton now distributes E-mu, Synergy, Fairlight, Quantec, TC Electronics, Kurzweil, Linn, and Sound Workshop in the Benelux countries. Felix has some interesting thoughts about the current state of the electronic music scene, as evidenced in the following interview.

**CA:** What's the main difference between the American and European music scenes?

**FV:** The United States is a one-language country. In Europe, the situation is completely different and only a few Americans appreciate that. If you drive your car for about an hour and a half, you're in a different country and people don't understand you any more. Sometimes there are even different dialects in the same country.

As a result, no one is interested in Dutch pop music in other countries, which limits the market and therefore the sales. Germany is different; there are about 60 million people there, so their home market is strong enough to maintain home-made products and the people who make them. In most Dutch homes, you'll find 90% of the pop records are English or American. The money spent on music in Holland goes to the USA or England, which also limits the music scene here.

**CA:** Why aren't Dutch people more supportive of local acts?

**FV:** There are 14 million people here -- about the same as the greater Los Angeles area, but not as hip. There just aren't the numbers to support a vital local music scene. With 250 million people in the United States, you not only have a large home base but you can also sell to England, Australia, and so on.

How can we evolve if there's nothing to support? That's why we asked people like you here; that's why I started Musicom...to aid evolution.

**CA:** What equipment creates the most interest here?

**FV:** Cheap instruments -- little Rolands, Korgs, etc. For me, though, the hottest item is the Fairlight. Not in terms of sales -- I'm lucky to sell two or three a year -- but in terms of interest; it's a fascinating instrument. The company is also very supportive. When they make promises, they live up to them. The Fairlight is also extremely reliable.

What makes the instrument fascinating to me is the "organic" quality of the sound. For an analogy, I consider analog synthesizer sounds of the past 15 years as like walking in an area with huge apartment buildings -- after a quick scan, you've absorbed the essence of the area. Especially preset synthesizers, which are kind of like junk food, as opposed to the modular types which are more like gourmet food. On the other hand sampled sounds, because they are related to real life and musique concrete techniques, have an inherent fascination. You can close your eyes to what's around you, but you can't close your ears. The Fairlight takes those natural sounds, magnifies them, makes them useful.

**CA:** What does this mean musically?

**FV:** It means we have different raw materials to work with for sound. It formed the basis of instruments like the Emulator; the E-mu people are very clever, and built on the strongest concept of the Fairlight -- the ability to do instant musique concrete with sampling techniques.

**CA:** Do you think analog synthesis is a dead issue, and that sampled sounds will be around for a while? Or are we just entering into another fad?

**FV:** Analog synthesizers might go the same way as the electronic organ, which is not even a standard instrument. Acoustic sounds have a special quality, and sampling instruments can actually use those special sounds. I'm still working on why sampled sounds are so appealing to people...

**CA:** Perhaps it's because it takes the synthesizer concept of being an "audio erector set" one step further. With analog synths you only had a couple of oscillators to modify; with sampling machines you've got the whole world to play with...

**FV:** Yes. Well, there's no question that sampling devices are very much taking things in new directions.

**CA:** As a drummer, how do you like the new Linn 9000?

**FV:** I'm very much impressed. I think they really listened to what people wanted...they paid attention to details, like high-hat decay. I'm not sure about the sequencer though; I think the drum computer is strong enough to stand by itself, but I suppose it is handy to have a good MIDI sequencer.

What I admire about Roger Linn is he doesn't shoot off his mouth a year in advance. They talk about the product when it's ready. He only panicked once, I think -- when Tom Oberheim came out with the DMX -- and when he announced the LinnDrum, it obsoleted the LM-1. Like the same problem with E-mu announcing the Emulator II before it was ready.

**CA:** That's a common problem, though, especially in the computer business. The technology moves so fast the companies can't keep up with it, although they think they can. And these instruments are essentially computers with keyboards or pushbuttons attached...

**FV:** What I like about the computer in musical instruments is that people with little musical education, but specific ideas about how something should sound, can make music. This will lead to a new type of music, because these people will not be hindered by tradition or technique. That intrigues me. Of course, the thing is abused -- wallpaper music, for example -- but then again, a lot of instruments are being abused.

**CA:** So what happens when people start getting bored with sampling devices?

**FV:** Maybe in another five years that will happen, but I'm not sure. If we do get bored, I think the next thing will be alternative tuning. I was amazed when I spoke to Wendy Carlos last time I was in New York; I heard her next album -- the one after Digital Moonscapes -- and that was fascinating. It's something old, but also very new.

**CA:** Let's get back to the difference between the American and Dutch music scenes. What are some of the economic considerations?

**FV:** One problem in the United States is that it is a very competitive environment from an economic standpoint...artists have to work harder to prove themselves, because if they don't, there are a zillion other people ready to take their place. That's different here; people are spoiled because of the social security (welfare). If you're a musician and lazy, and don't feel like working, there's no need to work hard because the state won't let you starve. I think this affects the quality of the end product, because you hear music from people who are bored stiff. They can afford to be bored stiff, because if they're fired from their jobs, they don't have to worry. I think the artistic as well as technical level is much higher in the states...

**CA:** It is a very competitive scene, you have to produce to survive...

**FV:** That was taken out of Europe a long time ago. But I don't like the way it is exaggerated in American high schools. You can see competitiveness taught to people as soon as they're born.

**CA:** Competition leads to other problems too. People keep reinventing things because there isn't a framework for co-operation. That's why I was so impressed with Musicom; the only problem was that the high school kid playing with ICs is the one who should be at Musicom, but it costs a lot of bucks to fly to Holland.

**FV:** That's one reason we're making tapes of the various workshops -- hopefully that will get the word out further. At least, though, there is a communication happening. Eventually people will become more interested. And the personal contacts are very valuable, especially what happens at conventions.

**CA:** What do you plan to work with in the future?

**FV:** I like what Floris Van Manen is working on, the concept of being aware of the sounds around you...the fact that you can't shut your ears but you can close your eyes. That is such an important statement; well, you heard what he did with the PCM sound-on-sound seminar. To me, the point is not that he can make a good recording -- the point is that he's working on a very high level of awareness of what makes up good sound. I don't think my interest is so much music; music is beautiful, but I can do without it. Yet I couldn't do without sound, and working with sound is where my future lies.



## re-view

Arnold Mathes **Concentrated Void** (AM #13; cassette). Few sequencer patterns are sustainable over 15 minutes, though by improvising over them Mathes creates a tolerable, perhaps even meditative boredom. $5 postpaid from 12750 Homecrest Avenue, Brooklyn, NY 11235.

**(continued on page 24 )**

# Optical Disk Drum Machine

**By: James A. Lisowski**

My article "Optical Audio" (Polyphony 5/82) presented an overview of optical processing techniques that can greatly expand upon current electronic music technology. This article continues in the same vein with a timely example: the Optical Disk based Drum Machine.

I'm sure that you are familiar with conventional Electronic Drum Machines and Rhythm Sequencers (as well as with their limitations). The prodigy of fast microprocessors, cheap memory, precise Analog to Digital to Analog conversion systems and a willing market, these systems provide the studio recordist or practicing musician with an on-beat, properly mic'ed, "drummer in a can." The first models had only a limited kit of synthetic voices and limited set of preset rhythms. The next step was "programmability", which placed the musician at the controls of a sequencer-based percussive synthesizer. Once microprocessors and digital recording became "inexpensive," manufacturers recorded the best percussion instruments, under the most ideal recording conditions, and squeezed the sound into electronic circuits that could give decent frequency response and dynamic range at an "affordable" price. Even with their limitations -- frequency and tuning range, accent quantization, number of voices, sample duration, and programming -- early units were good enough to make the listener wonder if that new song was backed by a real drummer or a machine. Sometimes it's obvious; the performance is too perfect (or limited) to be a human. Far from being an endangered species, the accomplished modern drummer is often the best qualified person to program a drum unit or suggest improvements, as would anyone else using a tool of the trade. Indeed, a drum programmer is a "drummer" with a new set of sticks, as well as tricks. (Even so, I've yet to see someone toss and twirl a pair of drum machines in mid beat. Any takers?)

**How digital drum machines work.** Price, not technology, impedes advancement in the digital drum field. Sure, we'll see improvements in price/performance ratios due to competitive marketing and the manufacturing learning curve, but the component "guts" define the base price. Quality source recording facilities and talent is a constant. The drum unit itself consists of a control microprocessor(s) that sequences the drum voices. The voices are "digitized" percussion sounds that are encoded into electronic ROM or PROM data storage devices. These "digital sound samples" are read sequentially and then sent to a fast D/A converter (and associated circuitry) to reproduce the drum voice. If the "playback" speed (clock rate) that is moving the samples out of the ROM is faster than the "recording speed" (sample rate) at which the sound was originally recorded, the result is a higher pitched sound of shorter duration. Conversely, slower playback clocking makes a sound lower in pitch and longer in duration than the original, thus allowing for "tuning" a single drum sound. Ideally, an infinite number of samples should be taken and played back. In the real world, however, a fixed number of samples can be stretched or compressed only so far before the sound loses its identity or is obscured by the artifacts of the process.

To accurately capture the exact waveform, samples are usually taken at a rate of two or more times the highest frequency sound to be sampled. Many closely spaced (toward infinitely small) samples can record and reproduce a waveform to any desired degree of fidelity, but the number of samples stored and the sampling rate increase with greater degrees of accuracy. (Special data encoding methods, such as Linear Predictive Coding used in speech synthesis, can help use memory more efficiently.) So, real world tradeoffs are made in order to keep the product's price from rising toward infinity. Often this means a medium sample rate and low pass filtering (to prevent sampling errors or aliasing), and a small number of samples (sound durations of a second or two) so that the voice will fit in a memory device. The manufacturer limits the number of different voices (each voice uses more memory) and number of data bits in the digital word (resolution) that describes the analog voltage level of each sample point on the recorded waveform. More bits give higher fidelity, better signal-to-noise ratios, and greater dynamic (amplitude) range (and with percussive sounds, wide dynamic range is a very important characteristic) but of course add to the cost. Adaptive companding or nonlinear D/As can give higher specs without adding bits, but these are still not perfect. Digital drums also cut memory needs by using a programmable amplifier playback level control. This provides several "accent" settings from a single voice sample, but otherwise does not change waveform content (as would happen if you hit a real drum harder or softer).

The central bottleneck to all

these considerations is MEMORY. Simply put, a large collection of high fidelity, large dynamic range, and long duration sound samples requires massive amounts of memory. Other things being about equal, a lower priced mass memory device makes for a substantially lowered drum price. Now, if you are familiar with OPTICAL memory devices such as industrial computer data systems, video disks and the compact audio disk (CD), you should start to "see" the connection!

**The optical drum machine.** Most digital drum machines use electronic ROM memory chips that store up to several hundred thousand bytes (data words) of sound sample. Played end to end, the sounds last only seconds. Now, consider that optical compact audio disks and video disks hold several million to several billion bytes of high quality audio, position, timing, and sometimes control data (or computer programs) and can play for hours. Optical disks can be manufactured from low cost materials and encoded with a one step mechanical or photographic exposure method. Add to this the mass produced consumer optical disk players that have comparable audio specs and lower prices than many digital drum machines and what do you get? The Optical Disk-based Digital Drum Machine!

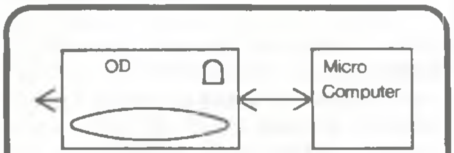


FIG.1

**Fig. 1**, the "basic" optical disk drum machine, might be a special device and disk format or it might be a modified CD player scanning a "drum sound" disk or a commercial "album" containing interesting drum sounds. A single read head picks off the encoded information from the single disk, while other circuitry decodes the data and translates it to audio. An internal or external microcomputer directs the head and disk to the desired drum sound or riff for selection. With a "drum" disk, a snare track/program, bell program, five-minute "Rock 1" or "Waltz" beat could be selected. The disk might be one instrument and one beat, or several programs of the same rhythm -- each at a different tempo, "tuned pitch" or accent style. It might also have one beat with selectable combinations of instruments on different "tracks". Unfortunately, this simple approach still has definite confines, especially with an unconverted CD player and album disk. (For example, the pitch, tempo and pattern of the drum beat is fixed.) Nonetheless, this "basic" configuration takes a significant lead over electronic ROM drum machines, even with the use of off-the-shelf CD player technology.

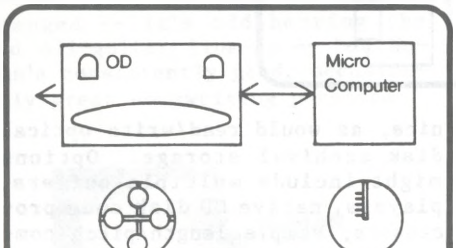


FIG.2

**Fig. 2** shows the next step up. This unit contains either multiple read heads, scanned read light beams, multiplexed sensors, multiple fixed space or programmably directed fiber optic read bundles. This allows continuous sequences because one "head" reaches the end of the pattern. (In the "basic" model, a sequence would end sometime, though the gaps could be minimal or timed to be between beats.) The "multi-head" version could produce continuously increasing or decreasing tempos by interleaving tracks of different rhythm speeds on-the-fly. Since each head presents a different voice source, you can mix instruments in any combination, taking each source to a separate volume control/filter output jack. (This can be done in the "basic" version with "stereo," "quad," etc., but there might be some channel leakage.) The multi unit is an improvement on the basic model, but it requires complicated source disk mastering and read head movement logic. Heads, mounted at 90 degrees to each other and moving on a radius from the disk center should give good program selection with modest head control logic; but then again, if four heads are better than one, how about four disks?

Several disk players, each having its own head and program disk, could have one central controlling computer directing the motions and mix of the entire group. This version might be composed of common CD players plus external controllers, or an all-in-one box that contains several disk transport/decode units (eliminating redundant amplifiers,

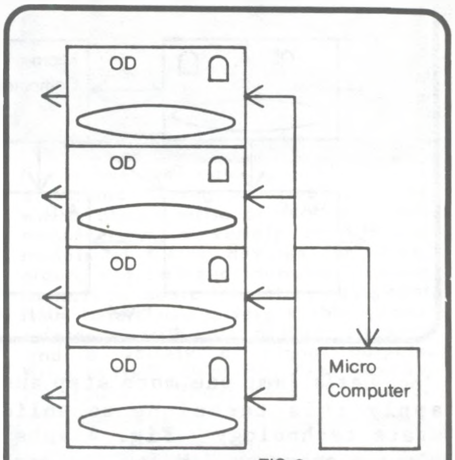


FIG.3

etc.), and custom controller (**Fig. 3**). The benefits should be obvious: Long (virtually infinite) program lengths, very many rhythm and voice choices, multiple outputs, accents, tempos and pitches, all without greatly complicated disk programming or head configurations. The main controller might even be a MIDI controller. This system could also make musical sounds other than percussion -- besides a drum disk you might want a flute, guitar, synthesizer, choral voice, or other disk, all working together or doing occasional fills. Why not have several "string" disks of "real" violins that can be doubled or omitted in real time? What a great way to conduct your own quartet or orchestra! The "master control" might be a CD disk bearing computer control or MIDI instructions that direct the other disk players (or other electronic instruments, effects, recording equipment or lighting) to produce "canned" or sequence programmed backing or lead lines. ("Jam Disk of the Month", anyone?)

Then again, why have all of those real disk players when one player can simulate several "virtual players"? Just treat the optical disk as sound sample mass storage -- access the sample, then dispense with the disk. For example, you could record a drum track from disk to one track of a multitrack recorder, change the disk, lay down another track, change disks again, etc. In this case tape loops or the variable speed record/playback features of the tape machine can do the multiple instrument, beat, tempo, pitch and programming sequence tricks.

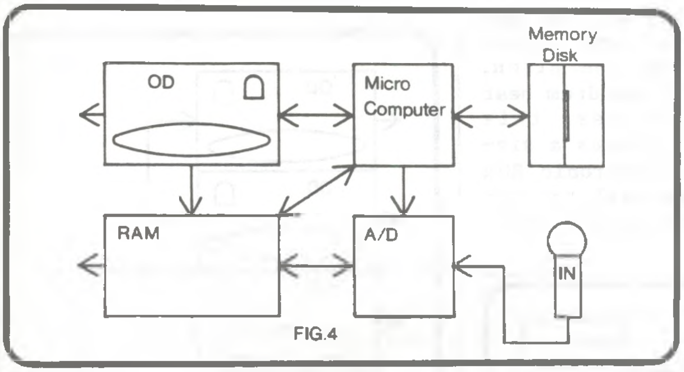


FIG.4

Let's jump one more step and apply this technique to solid state technology. **Fig. 4** substitutes computer RAM for the magnetic tape recorder in the previous example. The procedure is this: Select the drum sound sample (for example, a four beat measure, fill, or entire song) position from the optical disk, "down load" it into the RAM buffer memory and then play out the buffer as the final composition. This setup is capable of all the tricks of digital recording and processing, and also, the sound can be stretched past the boundaries of the optical disk source material in several ways.

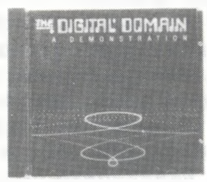
As with tape, this device could sequentially record several sounds or tracks (off several disks) and compile a "master mix" from component parts, with total control over their individual amplitude and tone attributes. Any number of voices could be overlayed to develop a thicker sound. Change the clock speed to (or from) RAM to transpose the disk sound up or down. The same tactic could provide smooth, programmed tempo changes from one (or a range of) disk-based rhythms. You can even play the RAM sound image backwards, create multiple copy "echoes", or do digital flanging. Digital editing could provide gap free loops, bridges, assembled multi-position re-arrangements, or cut out unwanted beats or instruments from the source material. With digital waveform subtraction you could even extract a desired sound from a commercial album disk, even if several other instruments were playing at the same time. At this point all you need is an A/D front end and data port to digitally record your own (non-disk based) sound sources, click tracks, or MIDI instructions; then overlay and edit them. Magnetic tape/disk data save and recall would be nice, as would read/write optical disk archival storage. Options might include multiple buffers, players, native CD disk code processors, sample length/pitch compensation, resynchronizers, timecoders and loop-through sweeteners. This combination of digital computer processing, optical mass storage and large, cheap RAM memory is a capable instrument ready for today's modern musician's hand and spirit.

Now, I'm not suggesting this as the perfect robot band. It should be used as a tool in the same way that programmable drum machines and bass lines are being used now: for recording, practice and inspiration when flesh and blood musicians are not available, or for composers who have arranging skills aplenty and playing skills alack. Imagination and experience will still be needed, and talented musicians will still have to create the disk samples. Sharp technicians will need to nurse new prototypes and make old circuits jump through different hoops. But remember that high technology doesn't have to mean high price. For instance, in the networked, separate CD player configuration, you could rent additional players and disks from a music store on demand (just like calling in session musicians and instruments) instead of buying the entire unit up front.

Although you won't have to be an electro-optical engineer to use an Optical Drum Machine, this chapter still has numerous pages to be written by enterprising home experimenters and major manufacturers. Make those disks and computers sing with your musical ideas! A good starting point to learn about the basics of digital recording circuits and optical disk technology is: "The Sony Book of Digital Audio Technology," #1451B, from Tab Books Inc., Blue Ridge Summit, PA 17214. (This book is also part of the Sony Digital Audio Club package.)

# re-view

Seth Bovey **Wolf Tones with Black Sound** (cassette). Six tracks (1/2 C-60) featuring rhythm box, guitar, bass, a small synthesizer whose keyboard doesn't seem to track quite right, and one spoken vocal, all by Seth (another uses a guest vocalist). Two of the pieces are quite striking, utilizing taped clock sounds in one case and Gregorian chant in the other. $5 from Seth, Route 4 Box 280, College Station, TX 77840.

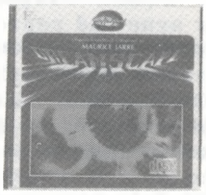



Peter Schafer **Peter Schafer's FARN-Werke** (FARN 18002). Peter continues to amaze, turning out mature accomplished neo-classical pieces using only a PPG 2.2, Roland TR-808, and Tascam 38. He's wise beyond his 23 years. Distributed by Eurock, PO Box 13718, Portland, OR 97213.

**The Digital Domain** (Elektra 9 60303-2; compact disc). As explained in the accompanying booklet, this CD was designed not only as a demonstration of digital recording but also digital synthesizers. Accordingly, over half the disc is devoted to various transformations and compositions utilizing the one-of-a-kind digital synthesizer at Stanford University's Center for Computer Research in Music and Acoustics. Some "old friends" of Polyphony (Michael McNabb, Stuart Dempster, Richard Waters) are featured.

It's a good demonstration of the capabilities of both digitals -- and consequently won't (and can't) be released in any other format.

Maurice Jarre **Dreamscape Soundtrack** (Sonic Atmospheres 302). Well, Keyboard's Jim Aikin liked it. Like many soundtracks the music doesn't really stand on its own, and the synthesis is nice but nothing special. I think Craig Huxley (the performer) was punching a time clock on this one.



The Art of Noise **Close-Up** (ZTT 12Ztps001; 12" single). Yes-producer Trevor Horn's studio tomfoolery gives us yet a third (or is it fourth?) version of "Close (To The Edit)", a sort of loose Linndrum rhythmic framework into which he fits different parts each time. Can he make a career of it?

Jean-Michel Jarre **Zoolook** (Dreyfus 18118). Jarre has been gradually shedding his goody two shoes image, becoming more experimental and adventurous with each release. Teamed here with such sound-pioneers as Adrian Belew, Marcus Miller and on one track Laurie Anderson, and making it all make perfect sense, you'd hardly recognize the author of "Oxygene". Unlike that other Jarre, the synthesis is pretty impressive too.


Gary Numan **The Plan** (Beggars Banquet 55); **Berserker** (Numa 1001). Two albums, one unreleased tracks from 1978 (before the


"First Album") and the other late 1984, illustrate just how little Numan's songs have changed in 6 years. The instrumentation has changed -- it's odd hearing the old all-guitar line-up -- but Numan's consistently good, occasionally great songwriting persists.

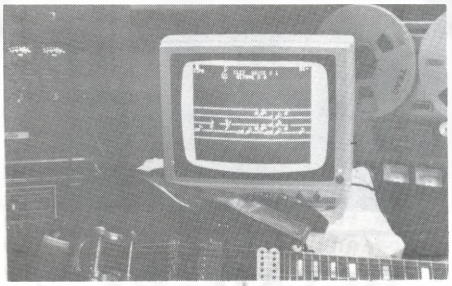

Moebius and Beerbohm **Double Cut** (Sky 091). Like their previous collaboration (reviewed 9/82), these are pretty simple patterns played by human sequencers who go on long after you wished they had quit. I don't know why I keep buying them.

# CURRENT EVENTS

*'Tell Them You Saw It In Polyphony'*

All prices quoted are suggested retail prices, as supplied by the manufacturers.

**Updated software. Studio 64** for the Commodore-64 (C-64) has been re-designed to include higher resolution graphics and simplified control. It retains compatibility with another En-Tech program, **Add Mus'In**, which adds music to other programs and can play music developed with the older version. Retail price (diskette version only) is $39.95. En-Tech, 10733 Chiquita, Studio City, CA 91604.

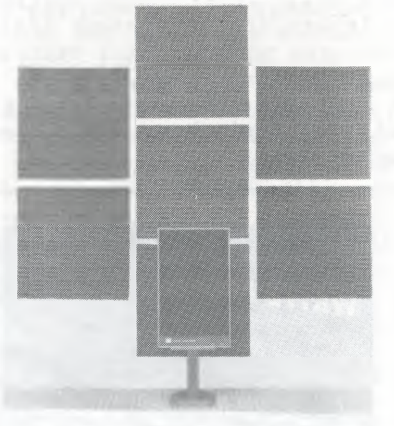


**Keyboard Controlled Sequencer** ($100) allows real or step time note entry, editing, auto-correct, transpose, 3300 note capability, inversion, and independent real-time control of up to 35 sequences on up to 16 MIDI instruments. Includes sync to drum machine. Runs on the C-64 and handles SCI, Passport, Yamaha, or Korg interfaces. **DX7 Patch Librarian** ($75) provides disk storage and patch editing/printing. All operator related patch parameters are displayed on one screen, with other parameters displayed on a second screen. Changes to any parameter made on the computer are sent immediately to the DX7. Dr. T's Music Software, 24 Lexington St., Watertown, MA 02172. Tel. 617/926-3564.

**Help in Canada.** For synthesizer service, custom work, and a Canadian source for SSM music ICs, contact Music Technologies Group, 10204 - 107 Avenue, Edmonton, Alberta T5H 4A5.

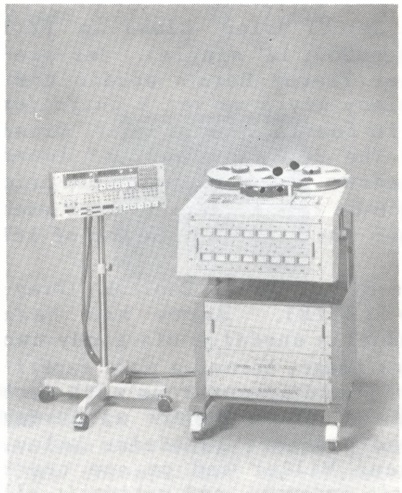
**Voltage controlled active filter module.** The Kyodo VCF-410 provides high pass, low pass, and broadband responses and can be configured as a Butterworth, Chebychev, or Bessel filter. Control range is 1000:1. Sabor Corporation, 2908 Oregon Ct., Suite H-1, Torrance, CA 90503.

**Home studio acoustic foam.** Sonex, used to control sound in pro audio applications, has now been packaged in 24" square, 2" thick panels for home studio use. Sonex can greatly reduce standing waves, slap echoes, and resonances caused by hard walls, windows, and other hard room surfaces. Packaged four panels to a box; $39.95 per box. Distributed by Illbruck USA, 3800 Washington Avenue North, Minneapolis, MN 55412. Tel. 612/521-3555.



**Helpful hints on how to hype.** "How to Promote Your Rock Band", by Grey Smith, is claimed to answer many important questions for aspiring musicians (including effective group management, contracts, auditions, demo tapes, copyrights, publicity, and showmanship). Available only by direct mail for $25. GS/GROUP, 27252 Pinocha, Mission Viejo, CA 92692. Tel. 714/495-3372.

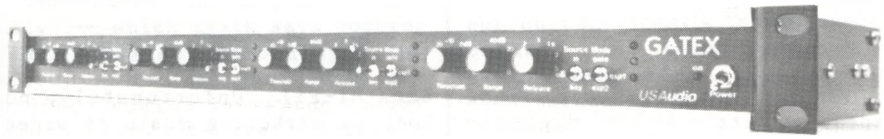
**Sweet 16.** The MS-16 one-inch 16 track tape recorder includes SMPTE control capability. Head bumps have been minimized to under 1 dB, and equal response is claimed for sync and repro heads. Other features include DC-coupled amplifiers with differential pair FETs, +4 dBm XLR and -10 dBV RCA outputs, and separate low frequency compensation adjustments for record/sync and playback heads. TASCAM, 7733 Telegraph Road, Montebello, CA 90640. Tel. 213/726-0303.



**New LSI Soundmaker.** Phillips has shown a prototype stereo sound generator IC scheduled for introduction in 1985. Conceptually similar to the Commodore SID chip, there are six eight-octave (30 Hz to 7.81 kHz) tone generators, with resolution to 256 tones per octave (perfect for microtonal work). It interfaces with most 8 or 16 microprocessors; each output (six stereo pairs = 12 channels) has 16 selectable output levels. The chip also has noise capabilities, runs from a single +5V supply, and draws less than 70 mA.

**Noise gate/expander.** The Gatex provides four channels of noise gating and expansion for $399 list. It includes the Valley

People TA-104 gain cell to minimize noise and distortion. USAudio, PO Box 40878, Nashville, TN 37204. Tel. 615/297-1098.



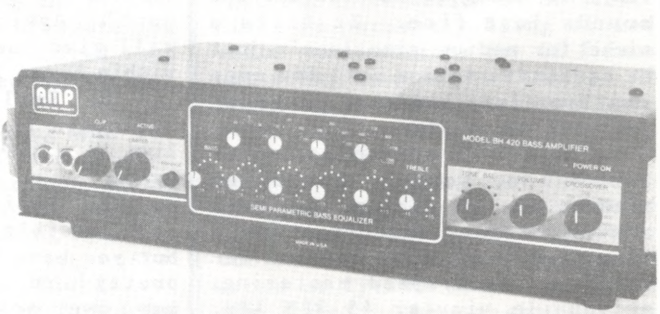

**New music software. Guitar Master** ($49.95) is a C-64 compatible diskette that teaches tuning, chords, progressions, chord analysis, transposing, picking and strumming patterns, scales, etc. through 64 different lessons and an accompanying 78 page manual. Manufactured by Mastersoft; distributed by Maverick Publications, Drawer 5007, Bend, OR 97708. Tel. 503/382-6978.

**Yamaha enters home computer biz.** The CX5M Music Computer ($469) is a Z-80 based, MSX-standard compatible computer, designed with musical applications in mind. It includes a built-in FM synthesizer with 46 pre-programmed voices; these sounds can be recorded and played back using the CX5M's memory, for storage and playback of up to 2000 notes. There's also an auto accompaniment section with bass, rhythm, and chords.

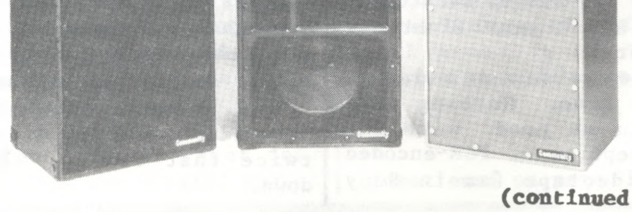

An optional **keyboard (mini** type, $100; standard size, $200) turns the computer into a live performance synthesizer with programmable split and MIDI. Available software cartridges ($50 each) include the FM Music Composer for computer-aided music composition and orchestration; the FM Voicing Program allows for creation of new voices, as well as modifying the FM sound synthesizer's existing voices. A DX7 Voicing Program allows for easy DX7 programming, with on-screen displays of all voicing parameters. FM Music Macro accesses the synthesizer via MSX BASIC for voice selection, music composition, and automatic performance.

Regarding other computer functions, the CX5M runs other MSX software cartridges (word processing, etc.). Graphics capabilities include 16 colors, sprites, and plotting functions. Yamaha, 6600 Orangethorpe Ave., Buena Park, CA 90620. Tel. 714/522-9011.

**Back to bassics.** The BH-420 bass amp head ($799) specs out at 400 Watts at 0.05% THD with only 2.75 microvolts of input noise. Includes limiter/compressor, bass/treble EQ, and four bands of quasi-parametric EQ. AMP, 9829 Independence Ave., Chatsworth, CA 91311. Tel. 818/709-0518.




**Speaker.** The RS325i 3-way loudspeaker ($549) handles 400 Watts continuous from 60 Hz to 18 kHz from what is said to be a "lightweight and portable" cabinet. Community Light & Sound, 333 E. Fifth St., Chester, PA 19013.



**(continued on page 31)**

# CDs

**By: Robert Carlberg**

Compact Discs (CDs), those iridescent little 4-3/4" digitally-encoded discs, have lately been the subject of much hoopla. As usual, a lot of it is pure bunk. Some of it isn't though, and that's what's interesting.

It may be a little while before CDs become the domain of "Electronic Music and Home Recording" (however, see related story by James Lisowski in this issue), but as a subscriber to Polyphony you're probably a fan of high technology. So, here is one person's view of what CDs mean to us.

**I. The hype.** The press is filled with advertising hype and press agent feeds masquerading as reporting. Many fantastic claims have been advanced:

"CDs will soon eliminate the LP." Baloney. (1) Most of us couldn't afford to replace our records with CDs even if they were available (2) which they're not and (3) probably never will be. Vinyl is cheap to manufacture and sounds just fine. It fills a niche for medium size labels just as cassettes fill a need for even smaller distributions. Augment yes, but not replace.

"The CD is an audio revolution." No, sorry. Evolution without the "r". Things have been moving in this direction for a long time, what with digital recordings, half-speed mastering, audiophile vinyls, 45 RPM LPs, etc. The next "revolution" will be even more amazing -- for example, speaker design hasn't changed much since the '50s.

"Playing time of 75 minutes uninterrupted." Possible, but rarely achieved. Most CDs are re-released LPs, with their 20 minute per side limitations.

"The biggest advance in audio since stereo." A judgement call. CDs may affect the way music is recorded the way stereo eventually did, but it hasn't yet. Right now that's puffery.

"The new audio standard." This one is true. Nothing else sounds quite as good, with the possible exception of PCM-encoded audio on videotape (as in Sony Beta Hi-Fi). Unfortunately, nobody is marketing music on videotape, unless it happens to be the soundtrack to a movie.

**II. The real ad & disadvantages.** For the record (pardon the pun), here is my own list of the true advantages and disadvantages of CDs, arranged in rough order of importance (first the advantages):

1. **Wider dynamic range:** In excess of 90 dB compared to the LP's 60 dB.
2. **No surface noise:** Approached by audiophile LPs but only really achieved by the CD.
3. **Higher signal-to-noise ratio:** With a louder maximum and no noise floor, it follows.
4. **No wear:** Since the disc is read by a laser, nothing actually touches the disc. Practically speaking, a good turntable comes darn close though.
5. **No wow and flutter:** Speed information is encoded on the disc itself, making variations "below the limit of test equipment". Again, a good turntable will give the same performance, within human limits.
6. **Small size:** A marginal advantage in my book (but if you live in a tiny apartment it might matter).
7. **Resistance to skipping:** It's possible to make a CD skip, but you have to thump the player pretty hard. This is an improvement over most turntables, but I stopped dancing to records a long time ago.

Disadvantages:

1. **Susceptibility to damage:** Like an LP, a CD can warp, scratch, or get too dirty to play. They must be handled with the same care as records (by the edges with clean hands) and stored like records (in their cases away from sunlight). The plastic covers are scratched more easily than your better quality LP jackets.
2. **Cost:** Presently ranges from $10 to $20 per disc, roughly twice that of LPs. It'll come down.
3. **Documentation:** Because of the smaller package, some manufacturers do not print all the same information as on record covers.
4. **Packaging:** The plastic packets used for most releases are impossible to open without a scissors -- you gotta wait until you get home.
5. **Visual information:** You can't see where the pickup is on the disc (they're read from the bottom). Some players make up for this with an elapsed time display.

A couple of comments on the above. The first advantage is a theoretical one; aside from a few old classical "spectaculars", not much music reaches these limits. Those that do tend to do so at the expense of "musicality", since we're not accustomed to being startled. This may change (both statements, actually) as CDs become the accepted medium to release new music. Right now, for the majority of existing music and especially re-issued analog recordings, it's academic at best.

**III. Players.** There are a number of different players available. They all will play any disc, which is quite a feat when you consider all the compatibility problems with Beta/VHS, MIDI, SQ/QD quad, etc. It happened because this technology was developed by two industrial giants (Phillips and Sony) and licensed to other users, not unlike the basic cassette rights which recently lapsed on Phillips. The technology had a remarkably short gestation -- a prototype was first shown in 1979 and the first player was marketed in 1983.

Yet the players have emerged more-or-less full grown. It can be safely said that any CD player sounds better than any turntable. And while some differences in sound quality have been rumored, if you stick to reputable brands you won't be disappointed. Features are the main difference between models and makers.

For instance, the first player I took home could only access tracks by the beginning of the track. For pop this might have been okay, but you couldn't get to the digitally-recorded cannons at the end of the 1812 Overture except by sitting through all 15 minutes of it -- a fatal flaw if you're planning to impress guests. That may be one reason why the decks were being given away for under $200 -- less than the cost of a decent cassette deck!

Prices range from under $200 to over $2000, so there's plenty of room to experiment. The upper levels include some pretty bizarre features, such as the ability to load in 51 discs and play any track from any disc in any order. What am I, a radio station?

More common features, besides elapsed time displays and fast-forward/fast-reverse which I decided I couldn't live without, include track order programmability (in case your new $20 CD has a few stinkers on it), wireless remote control (in case you're catatonic), repeat function (a CD will theoretically play forever), and time-remaining display for those of you who like to time your cookies by Van Halen. So-called "second generation" players sample at twice or even four times the normal 44.1 kHz rate for improved resolution. The improvement is reportedly minor. Some players also utilize a 3-beam laser pickup for improved tracking -- though single beam pickups will track just fine under normal circumstances.

Last, they have just introduced portable (Walkman-type) and automotive players. These seem to completely ignore the top three advantages and top two disadvantages of CDs, but anything that promotes consumer acceptance of the new format is bound to benefit the price and availability of CDs and related equipment, so I'm all for it.

**IV. Garbage in, garbage out.** A CD is only as good as the original recording. It is exactly as good as the original recording, but no better. For this reason, the glut of old analog LPs which are being re-released in the CD format is truly surprising. In order to fully take advantage of the digital playback capabilities, a digital original recording is necessary. Unfortunately, almost all rock, a lot of jazz, and many classical CDs are not digital originals. Some labels are getting tricky about admitting this -- they print "digital" across the CD to alert you to the CD format, but don't mention it was recorded in somebody's basement on a 3340. Or the record was "mastered digitally" -- which again says nothing of the original recording. The Society of Professional Audio Recording Studios is trying to get manufacturers to use a 3-letter code to identify recording, mixing, and mastering processes. In this scheme full digital would be designated DDD, analog recordings digitally mastered would be AAD, and so forth. Industry acceptance of this standard is slow in coming, but it should be showing up on more and more CDs.

Of course, many old analog recordings transfer very nicely to the CD. Since they usually go back to the original master tape and eliminate all stages of degradation in between, some dramatic improvements over the LP versions are possible. Abbey Road by the Beatles was "digitally remixed" for Japanese audiophile LP release, and these same digital tapes were used for the CD release. It is obviously cleaner and closer to the original. On the other hand, Jethro Tull's Aqualung is reportedly worse than the LP (possibly a ham-fisted engineer?) (Editor's note: Or possible a very good LP mastering engineer who wasn't consulted when the album was mastered for CD from the original tapes.)

Generally speaking the releases so far, which number something over 2000 and growing daily, are those platinum sellers which stand a chance of making back the still-expensive pressing costs. You've got your Pink Floyds and Billy Joels and 1812 Overtures and Pachelbel's Canons. As costs go down and sales go up, the variety will improve; which for eccentric tastes like mine can't happen soon enough. A couple labels, like Verve, are compiling material from more than one LP to release 60+ minute CDs, which is a good sign.

Some of the most popular electronic music, and that which translates best to CD, has been released. Jarre, Vangelis, Tomita, Kitaro, Deuter, and Tangerine Dream are but a few, and more will be out before you read this. Check your stores -- daily if necessary.

**V. The future.** What does the future hold for digital audio? In terms of software, many artists will design their releases specifically for the CD format, putting out longer and more dynamic pieces. Only then will you see the limits tested in what can be put on CD. Today's "sonic spectaculars" will sound pathetic in comparison, since none of them even scratches the surface of the potential.

In hardware, advances in playback equipment are due and will be forced somewhat by the CD explosion (many apartment dwellers can be expected to be evicted). But more radical advances are also on the drawing boards. Digital tapes are being developed which will likely obsolete the automotive disc players, due to improved shock handling. However, ease of access will probably keep discs in the forefront for home units, especially considering that CDs can be used as a storage medium for computers, and disc "read only" memories are in the prototype stage. CDs have an enormous storage capability -- the equivalent of 1000 floppy disks, or an entire set of encyclopedias on one disc. At present these would be "read only" until the "write once" and "eraseable" discs move out of the lab and into the home (Editor's note: Some people think that the introduction of eraseable discs is purposely being slowed while manufacturers try to figure out a way to prevent the type of pirating problems that have happened with cassettes and records). Other reference works which might be called up on your computer screen from a CD include translation dictionaries, phone directories -- how about the printed score or lyrics to the music on the disc? The capability is there.

Interactive computer games or educational programs are another possibility. You make a choice and the computer zips ahead to the appropriate "next question" on the CD. You could even hear the questions at the same time -- in digital fidelity. Think of the musical possibilities! You decide what music is played and the screen takes it apart for you.

These functions are a couple of years away yet, and new players will probably be needed with greater accessing capabilities.

**(continued on page 31 )**

# Video Focus: The Forgotten Technology

## By: Don Slepian

Editor's note: This new column, by noted synthesist Don Slepian (interviewed in the May 1982 Polyphony), introduces the world of music-oriented video experimentation. Don has been doing electronic music since 1970 and computer graphics and video since 1976; he has twice been sponsored by the French Ministry of Culture to perform live videosynthesis in Paris and La Rochelle, and has also given performances in New York and Philadelphia as well. His specialty is creating visual accompaniment to electronic music, both on tape and in performance.

In this column, Don takes a hands-on, practical, friendly approach that's tailored for those with more creativity than cash. For example, how would you like to shoot your first video project for under $300? Keep reading, because this first column will tell you how. We're very pleased that Don is writing for us -- if there are specific subjects you would like to see covered in future columns, write him c/o Polyphony.

Welcome to Video Focus! In this initial column, I'll take two steps back and give you an overview, and future columns tackle specific subjects. My goal is to increase the information available to the home video enthusiast. In the process, I will shed some light on the field of videosynthesis as seen from the viewpoint of the Electronic Musician.

Let's first talk about hardware, using audio as a basis for comparison. A VHS or BETA home-type VCR (Video Cassette Recorder) is the video equivalent of a $25 mono cassette recorder, a consumer color camera is like a $5 crystal microphone, and an inexpensive personal computer is like a PAiA Gnome synthesizer. Now before you get depressed, you should spend a few hours with the Gnome, cheap cassette, and crystal mic. Go to the bottom of a good reverberant concrete stairwell, turn on the Gnome, loosen your voice, mind, and creative energies, and amaze yourself with the beautiful, powerful, complex soundscape you can create using these inexpensive little tools. This column is dedicated to those people who would achieve the same experience in video.

The next step up in equipment from the home consumer type is "industrial", which roughly corresponds to "semi-pro" in audio. An editing 3/4" tape VTR (Video Tape Recorder; approximately $5000) is the equivalent of a 7-1/2 ips 1/4 track audio tape deck. A super-low cost graphics system ($8000) is like a Korg Poly 61 or Roland Juno 106, and an industrial color camera ($2000) is like a decent Shure mic.

MTV videos, and other "slick" video productions, are made in the far different world of broadcast video equipment. Broadcast video recorders generally use 1 inch tape, and are placed in rooms with a great deal of support equipment. These rooms, called editing suites, contain several VTRs, a video mixer (called a switcher), several processing and enhancement devices, and many monitors. Rather than having a big pane of glass looking out onto a room full of musicians, most editing suites are places where clients bring pre-recorded video tapes ready to be combined and finished. A typical three-machine 1" editing suite could easily cost over $500,000. The computer graphics systems used in this setting are priced from $40,000 to $100,000. There is a good choice of broadcast cameras in the $60,000 range. Continuing with our audio analogy, a 24-track studio with a 32-voice Synclavier, the new Fairlight CMI, and a PAiA Gnome sitting side by side along with Neumann mics would make up a single moderate purchase in a video studio budget. Since all highly advanced video equipment becomes obsolete every few years, it's easy to see that professional video is a very tough, capital intensive business.

Video people also have "toys" which correspond to flangers, pitch transposers, digital reverbs and all the other signal processors used to enhance audio. These are called digital effects (never "toys"), go under names like "Mirage" and "ADO" (Ampex Digital Optics), and are in evidence every time you see a can of dog food flip, tumble, spin, and roll-over in TV commercials. These effects are actually special purpose powerful computers, which allow the operator to alter perspectives and reposition video images. They range in price from $20,000 to $250,000. I have yet to see one packaged in a "stomp box" for floor use, however, I predict these studio machines will soon be used in live music performances. During sports events they are often used to create transitions between scenes.

A standard budget for music videos is $50,000, with most of that money going for rental of the video studio. Few video companies have found this field to be profitable, and many production people work in this field to add to their sample-reels and challenge their creativity. "Creativity" is the key word here, for just like electronic music all the equipment in the world will not substitute for creativity. Oftentimes the lack of money or equipment serves to foster new solutions.

If, like most of us, these huge sums of money all lie outside the scope of your budget and/or reach of your VISA card, don't despair -- read on. Videosynthesis is just as absurd and esoteric a hobby as electronic music (and you'll endure the same scorn and abuse from your friends and relatives), but you can get started for less than $300. Now $300 doesn't even get you a decent VCR, so I am suggesting an indirect approach. There is a very powerful, appropriate technology that is being forgotten and discarded today, a technology that gives you immense image-gathering powers. It is lighter than a $30,000 Betacam (broadcast quality remote camera/recorder), yields higher resolution than an $80,000 1" VTR, and edits faster and easier than a $250,000 video editing system. It is Super 8mm film. Just as electronic music can be made without synthesizers by gathering acoustic sounds and using tape transformations and studio techniques, non-broadcast music videos can be made economically using 8mm film.

Look through the papers for someone's used Super 8mm movie camera, projector, and screen. You or someone you know may even have an old camera and projector hidden away. It's desirable to find a movie camera that can advance one frame at a time. Read some books and talk to some filmmakers to find out what will best serve your needs. Buy film, a splicing block, and a good supply of quick-splice tape at your local photo shop. Unlike video, 8mm film can be edited at the kitchen table using a bright table lamp, splicing outfit, and some sort of filing system to keep track of your many little snippets of film. Shoot images that might go with your music, and create a short film with no less than 20 splices. Don't even try to sync it up exactly with the music; just let the visuals and music complement each other.

If you don't already have a home computer, look in the papers and at garage sales for an abandoned Timex Sinclair or VIC 20. These typically cost $50 or less. The computer can be used to make titles, cartoons, and graphics. To transfer the computer graphics from the video monitor or television to film is easy. Carefully wash the glass front of the tube, and set your camera up on a tripod. Put the movie camera in single frame mode, turn out all room lights to eliminate reflections on the front of the tube, and take exposures of 1/8 second or longer. If the movie camera doesn't have this capability, follow the same procedure with a 35mm still camera and then shoot movies from the prints.

Once you have made the perfect movie for your music, you can take what money you have left and have your movie transferred to VHS or BETA at one of the many photo stores that offer this service. The trick is getting them to transfer your music onto the same videotape at the same time as the movie, so see if it's possible for you to be there for the transfer. The alternative is to try to do the transfer yourself. You will be projecting the film to form a small, bright image, carefully balancing the color controls on the video camera for proper color balance, and controlling three machines: projector, tape recorder, and VCR, all of which require some time to come up to speed. You must mix your music "bright" (i.e. extra treble) and greatly limit its dynamic range, creating a special tape optimized for the transfer. This is because the audio facility of a standard (not "hi-fi") VHS or BETA VCR is inferior to a single cassette track with no noise reduction on non-premium tape! Practice, practice, and at last you will have your first music video.

To learn more about the field, check out some of the available publications. In the pro video field one of the best general magazines is Videography. Subscriptions are $15.75 per year, available from Videography at 475 Park Avenue South, New York, NY 10016. Video Systems is available on request to qualified applicants; write to them for a free subscription application at PO Box 12912, Overland Park, KS 66212. Also of interest is Computer Pictures Magazine, published bimonthly for $15.00 per year by Back Stage Publications, Inc., 330 West 42nd St., New York, NY 10036. A new educational non-profit association is the National Association of Videographers (NAV). Write to them for membership information at 120 West Second St., 2000 Hulman Bldg., Dayton, OH 45402.

## CDs

(continued from page 29)

In the meantime the audio potential alone is enough to keep the industry hopping. Mics, mixing boards, multi-track digital recorders, noise-free effects boxes, splicing and editing facilities are all due for some accelerated evolution. The new standard is out -- and the world will just have to adjust to it.

In conclusion, I won't be retiring my trusty old turntable to the attic quite yet. It would be foolish to replace the majority of my LPs since the advantages of the CD format aren't manifested in old analog recordings. There will be new LPs for a number of years to come, too.

But when new music is released simultaneously in LP and CD formats, and especially if it's a digital original recording, you can bet I'll be first in line for that little silver disc with the magic sound.

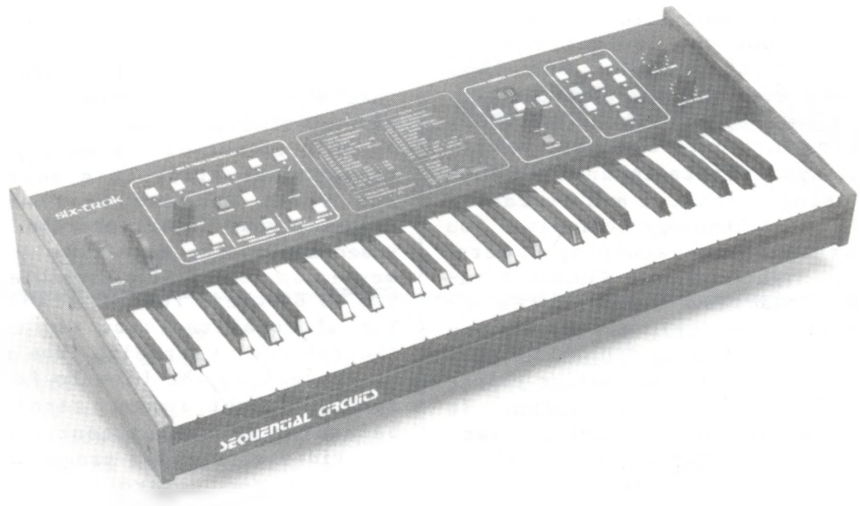
## CURRENT EVENTS

**Headset mic.** The PH20 is a head-worn microphone designed for hands-free vocals. It features light weight, comfortable fit, PS10 in-line power supply (1.4V calculator-type battery), and slender design for best visual effect. Also available: PH21 head-worn microphone (less PS10 power supply), and PH22 head-worn mike that connects directly to Telex wireless microphone systems. Telex, 9600 Aldrich Avenue So., Minneapolis, MN 55420. Tel. 612/884-4051.

# REVIEW:

# Sequential Circuits 610 MIDI Six-Trak

## By: Chuck Pogan




The Sequential Circuits Six-Trak synth is a polyphonic keyboard with the unique ability to store two multi-timbre real-time sequences of six different tracks; total on-board memory is 800 notes. The Six-Trak also expands to over 4000 note memory when used with a Commodore 64 and Sequential's own "Sequencer 64" module. In fact, SCI's compositional package, which includes their "Drumtraks" as well, gives the user more songwriting power than ever available before in so reasonably priced a system.

To keep costs down, the Six-Trak's front panel -- like many other synthesizers -- is devoid of any individually assigned knobs for any of the modules. Instead, rubber "soft-touch" buttons are used that serve multiple functions depending on the selected mode. Parameter controls must be punched up numerically, and only when the "par" (for parameter) button is activated. Then, a two digit number (00 - 37) is entered via the "select" (0 - 9) keys on the right. At this point, select the value button to the right of the parameter button and the value of that function will be visible in the LED display. Value percentages are not given on a 0 - 100% basis but rather are adjustable in steps of 0 - 15, 0 - 31, or other spans. The current value can be changed by the "value" rotary pot (one of only 5 pots on the front panel). To get to or change another parameter, simply enter that double-digit number and the LED shifts back to "current parameter" automatically. You can edit patches as you like, and save them in any one of the 100 user definable presets.

Some of the sound producing capabilities are unique for a low cost synth, such as 6 note polyphony, VCO assignable to its own ADSR, polyglide, filter inversion and poly or unison modes. This all adds up to a lot of diversity and some of the unison patches sound impressive. The poly mode, however, is not quite what you may expect, being a little thin at times. If you are not planning to use the 610 primarily for sequencing or expanding to MIDI, you may want to check out SCI's thicker-sounding Prophet 600 or other polyphonic keyboard.

But as mentioned before, sequencing is the name of the game and at this the 610 cleans house. The multitrack on-board sequencer is a snap to program; if you have your timing and riffability together, you'll produce superior results. To sequence, enter the patch and select sequence A or B. Set the speed knob straight up, and press "track record". The desired tracks (1 - 6) are then selected for recording. If playing polyphonically, choose two or more tracks as needed for however many notes will be played simultaneously. Recording doesn't start until you do, so there's no need to play "Beat the Clock."

When you start playing, the lit track's LED will blink. When you want to end your riff, press the same button that got you started recording in time with the beat (the one you hear in your head, unless you're also using a drum machine). An external footswitch can be used instead of the button, thus making the job a little easier.

Overdubbing is also a piece of cake, and gives you the feeling that you're working with a no-hassle 6 track tape recorder. However, on this multi-tracker there is no noise build-up when new tracks are recorded. Just select the program you want to overdub and pretty much repeat what you did for your first track, only select a different track (1 - 6) for recording the new material. Don't worry about when to stop recording overdubs, because the 610 will record as long as the first entered sequence lasted. Also, it won't record overdubs until the first sequence plays all the way through once, so this gives you a chance to practice what you want to play before committing your part to the sequence. By the time you're done you can have up to a 6-track, multi-timbre sequence in real time. You can now erase tracks, change volume on each track, change patches, and speed up or slow down the tempo (the sequence replay speed). This by itself makes the 610 an instrument to be reckoned with, but there's more.

The assignable arpeggiator has a couple of new twists previously available only on "higher-end "instruments (like the OB-8): It will not only arpeggiate up and down note patterns, but up and down in the order in which the notes were pressed. This really expands the flexibility of the arpeggiator's role. Another nice touch is that you can latch an arpeggio with one patch, then solo over it with a totally different sound. How many other low-cost synths can do that?

Last and certainly not least is the devastating "Stack Mode." With this function up to six completely different patches can be ganged-up on ONE key. Talk about fat sound! With the press of a key one can have, for instance, bass, strings, trumpet, organ and electric piano playing at the same time. There are two positions in which to save the bombastic stacks which could, depending on the individual frequency of selected patches, be tuned to six-note one key chords. These two buttons can make you mad with power!!

The 610 contains many "hidden functions" that are selected thru a combination of track record, control record, and keypad buttons. Such functions include sequence erase, program dumps, MIDI modes, external MIDI clock control, MIDI control over patch changes, modulation and pitch changes thru MIDI, and keyboard enable/disable. You will find yourself using most of these hidden goodies as your system grows; expansion and accessibility are the keys to the 610's success. The 610 is a software-based instrument and already new EPROM updates are available at SCI approved repair shops. Another currently available mod is an audio channel splitter that expands the single audio output to 6 separate outputs. This mod would be most useful for multitracking (tape that is), if you want to create stereo imaging or treat separate tracks with effects devices.

**Conclusions.** As mentioned before, the poly sound of the 610 is not as high quality as, say, Roland's Juno 106, JX3P, or SCI's Prophet 600. The high end can sound a little honky-tonk and unless two tracks are used on the sequencer at once, it can get a little thin. The filters sound buzzy when the "Q" goes down but for all the other cool stuff the 610 does for the money these drawbacks can be overlooked.

Besides, you can just plug in another 610 with a MIDI cable and double your pleasure, double your fun. The cost of two Six-Traks is still a bargain in the polyworld and the textures that can be realized with a slave/master setup are something else.

The future looks bright for the 610. The software heart ensures that it can be updated periodically so that you won't find it suddenly obsolete. And when you hook it up to a Commodore 64, add the Sequencer 64 module, and throw in the SCI Drumtraks, you'll understand why everyone's talking about MIDI.

■■■■■■

## Letters

the hint and start sending in guitar-related articles, hopefully the AMS-100 series will keep you occupied.

**TIPS**

Here are some tips which might be of interest to Polyphony readers. I was looking for a way to limit the travel of a slide pot on my MXR Drum Computer, and hit upon using electrical tape. Simply cut off a piece and stick it down across the slide pot slot, just at the point where you want the slider to stop. Use two pieces if you want to limit both the maximum and minimum slide pot excursions.

Here's a way to start the Drum Computer, stop it, and restart it exactly on cue. First, get yourself a 2-conductor, 1/4" phone plug but don't connect anything to it. Next, start the Drum Computer. At the point where you want to pause, insert the plug into the "From Tape" jack (on the MXR rear panel). The DC is now in infinite pause mode. To de-activate, pull the plug out of the "From Tape" jack and the DC will carry on.

Finally, here's a Dynamic Noise Enhancer. String together a bunch of Radio Shack molded plastic adapters -- you know, phone to phono, phono to mini, etc. If you're lucky, you might chance upon what I call a "Variable Continuity Connection" that entirely disrupts the ground signal. The result, as you can well imagine, is a positively geometric increase in Noise Potential (I've actually managed to obtain an incredible 00 dB S/N ratio).

Doug Bierer
San Francisco, CA

**WHITHER GIZMO?**

A letter in the March/April 1982 issue of Polyphony references a device called the "Gizmo". From the context, I assume that it is intended as an automated bowing device for guitar. Could you give me more information on the Gizmo? I have been working on such a device and would like to correspond with the originators of the idea.

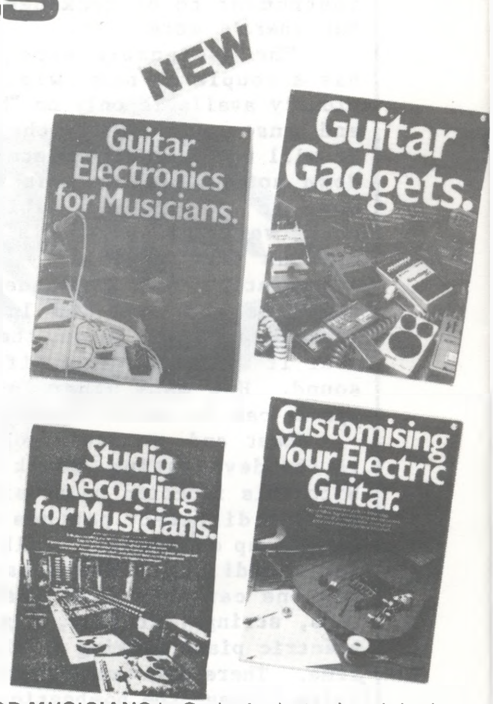
Robert E. Monroe
Pleasant Ridge, MI

Robert -- The Gizmo was the result of a collaboration between Kevin Godley and Lol Creme (formerly of the rock band 10 CC), and was supposed to be distributed by an offshot of Musitronics (known mostly for the Mutron III envelope-controlled filter). However, the company was plagued with financial problems, and the Gizmo with technical problems. The Gizmo used a small electric motor to drive several small wheels which rotated against the guitar string, thus providing the bowing effect. Unfortunately, it was impossible to find a material for the wheels which was soft enough to not damage the strings, yet firm enough to not deteriorate with repeated use. Also, the motor generated electrical interference and "hash" which often came through the pickups.

If you want to hear the Gizmo in action, Godley & Creme cut a three-record set called "Consequences" (Mercury SRM-3-1700; sometimes available in cut-out bins) which uses lots of Gizmo. However, the Gizmo is currently a dead issue; with the exception of the E-Bow, which works on a different principle, the field of mechanical bowing devices for guitar is wide open.

# POLYMART BOOKS

## NEW BOOKS!




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

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**#MMM MAKING MONEY MAKING MUSIC $12.95**

## REFERENCE


Often used reference materials to answer the many questions encountered in everyday synthesis. **THE SOURCE** Book of Patching and Programming from Polyphony has over 125 pages of patches in universal flow chart notation; the largest publication of its type.

**ELECTRONIC MUSIC SYNTHESIZERS** by Delton Horn devotes the first half to descriptions and functions of commercial electronic music synthesizers (Moog, Arp, PAIA, Oberheim, EML, and RMI); the second section provides schematics and projects for the experimenter.

| | | |
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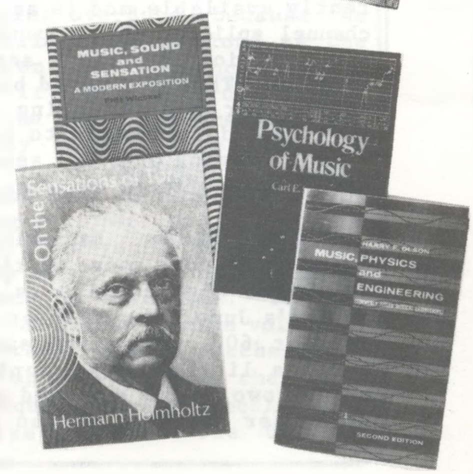
| **#SOURCE** | **THE SOURCE** | **$4.00** |
| **#CYCLO** | **AUDIO CYCLOPEDIA (hardbound)** | **$44.95** |
| **#EMS** | **ELECTRONIC MUSIC SYNTHESIZERS** | **$6.95** |

## SCIENCE OF SOUND




The physical and psycho-acoustical background to music is an important part of musical synthesis. Helmholtz's **SENSATION OF TONE** is, a century after its publication, still the standard text for the 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 Helmotz work, with a bit less detail and more concentration on psycho-acoustics.

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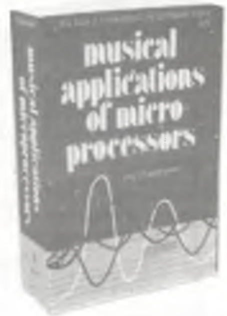
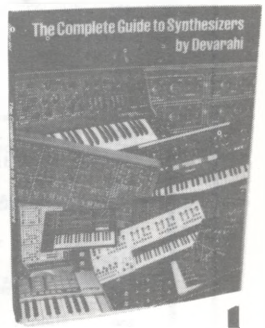
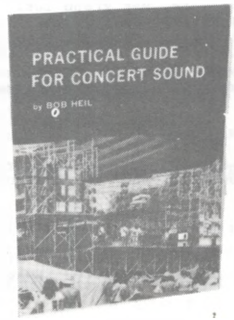



**THE BEGINNERS BOOK OF ELECTRONIC MUSIC** by Delton T. Horn — A fascinating introduction to synthesized sound using build-your-own circuits or commercial equipment.
**#BEM Beginners Book of Electronic Music** .................................... $12.95

**MUSICAL APPLICATIONS OF MICRO-PROCESSORS** by Hal Chamberlain — If you only have space in your library for one book on music synthesis, this is the book. The easily read text is entertaining and enlightening and teaches both basics and advanced theory without plowing through pages of equations. Liberal examples illustrate theory and practice of both digital and analog signal generation and processing.
**#MAM MUSICAL APPLICATIONS OF MICRO-PROCESSORS** **$21.95**




**THE COMPLETE GUIDE TO SYNTHESIZERS** by Devarahi — An extraordinarily thorough treatment of the subject of analog synthesizers. Covers everything clearly, and the suggested experiments are excellent. Well organized and leaves nothing out.
**#CGS THE COMPLETE GUIDE TO SYNTHESIZERS** **$20.95**




**PRACTICAL GUIDE FOR CONCERT SOUND** by Bob Heil — Finally, a manual that explains in very simple language those many "magic terms" you've heard for years and never really understood. It's the first book written especially for musicians, roadies, and sound technicians who want to KNOW and UNDERSTAND what their sound system is all about.
**#PGC PRACTICAL GUIDE FOR CONCERT SOUND** **$10.00**


## 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** Cookbooks are self-explanatory — required reading for synthesists! **ELECTRONIC PROJECTS FOR MUSICIANS** by Craig Anderton is almost in a class by itself. It discusses electronic construction technique for the novice and provides 27 projects with printed circuit board patterns and a demo recording of the effects. Even if you're an old hand at musical electronics, you'll appreciate that all of these processors, from Tube sound Fuzz to Phase shifter are compatible and work together without creating noise, signal loss, bandwidth compression or any of the problems common to interconnecting effects from different manufacturers. There's even a complete chapter on how to modify and combine effects to produce your own custom pedalboard. **ELECTRONIC MUSIC CIRCUITS** by Barry Klein covers synthesizer system design, power supplies, control voltage generators, VCOs, Filters, analog multipliers and more. Lots of schematics and data sheets on the most popular music oriented ICs. An excellent technical reference.

| | | | | | |
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| #EPFM | ELECTRONIC PROJECTS FOR MUSICIANS | $14.95 | #OACB | OP-AMP COOKBOOK | $15.95 |
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# DATA BANK

**A/D** - Analog-to-Digital Converter.

**AC or DC coupling** - DC coupling allows DC (such as control voltages) and AC (such as audio) signals to pass whereas AC coupling blocks DC signals and only allows AC signals to pass.

**Audio Taper Pot** - A potentiometer whose resistence changes at a logarithmic rather than linear rate.

**Curtis Electromusic Specialties** - 110 Highland Ave., Los Gatos, CA 95030.

**D/A** - Digital-to-Analog Converter.

**DPDT** - Double-Pole-Double-Throw switch.

**Electronotes** - 1 Pheasant Lane, Ithaca, NY 14850.

**Jameco Electronics** - 1355 Shoreway Road, Belmont, CA 94002.

**International parts specification standard:** This standard avoids the unnecessary repetition of zeroes, decimal points, and stating Ohms or Farads where it is implicitly understood. It is widely accepted in the international community and **Polyphony** would like to bring it home. Following are some examples.

| USA | Int'l | where |
|---|---|---|
| 1k | 1k | k = $10^3$ |
| 1.5k | 1k5 | M = $10^6$ |
| 2.2M | 2M2 | u = $10^{-6}$ Farads |
| 1uF | 1u | n = $10^{-9}$ Farads |
| 0.01uF | 10n | p = $10^{-12}$ Farads |
| 3300pF | 3n3 | |
| 0.0022uF | 2n2 | |

**PAiA Electronics** - 1020 W. Wilshire Blvd., Oklahoma City, OK 73116.

**PCM-F1** - A device made by Sony which converts audio to digital signals suitable for recording on a VCR, thereby yielding exceptionally high fidelity (low hiss, wide dynamic range, unmeasurable wow and flutter).

**PGS Electronics** - Route 25, Box 304, Terre Haute, IN 47802.

**pot** - Potentiometer.

**p-p** - Peak to Peak; a voltage measurement that represents the potential difference between the most positive and most negative excursions of a waveform.

**ppm** - Parts Per Million.

**RAM** - Random Access Memory.

**ROM** - Read Only Memory.

**SSM** - Solid State Micro Technology for Music, 2076B Walsh Ave., Santa Clara, CA 95050.

**Tel Labs** - P.O. Box 375, 154 Harvey Road, Londonderry, NH 03053.

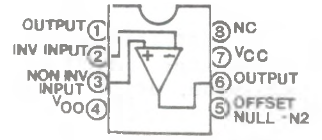
**thermistor** - A device that changes resistance in correspondence to temperature changes. The temperature coefficient specifies the rate and amount of change for that particular device.

**VCA** - Voltage-Controlled Amplifier.

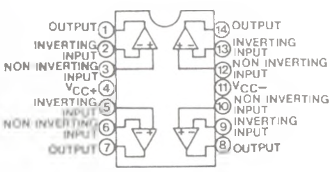
**VCO** - Voltage-Controlled Oscillator.

**50240 based keyboard** - The 50240 is a Top-Octave Divider IC which outputs square waves corresponding in frequency to an equally-tempered scale when input with a high-frequency square wave input clock. Used in many organ products of the 1970s.

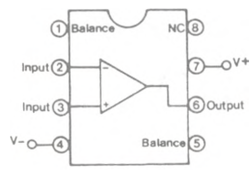
Class B is defined at an idle current of 1uA injected into Pin #8 through Ridle; Class A is at an idle current of 100uA.



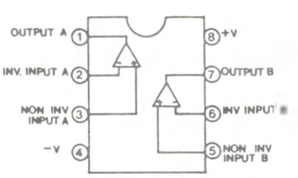

TL071 Low-Noise JFET-Input Op Amp



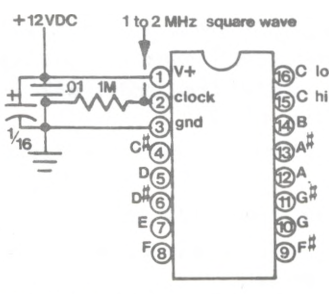

TL074
Low-Noise JFET-Input Op Amp



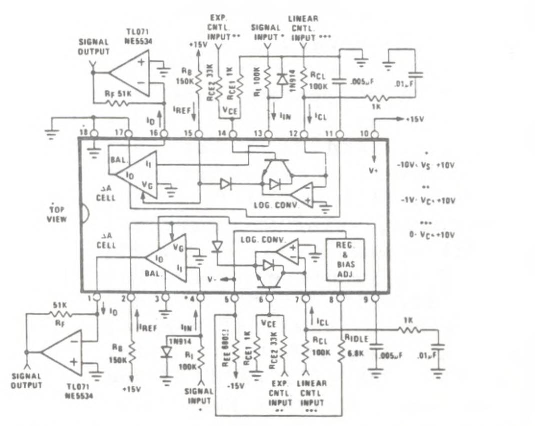

LF356
Wide Band Monolithic
JFET-Input Op Amp




LM1458
Dual Op Amp




50240 Top-Octave Divider




CEM3330 Dual Voltage Controlled Amplifier

# JUSTLY TUNED GUITAR: A Quick and Dirty (but workable) Solution

By: David B. Doty

In the April 84 issue of **Polyphony**, Vanessa Else presented an elegant method for obtaining a justly tuned scale with absolute accuracy. For those who have a 50240-based keyboard instrument to retrofit, or are willing to build one, this is an excellent way of getting started with Just Intonation. For those who lack such an instrument, are intimidated by the thought of building a device with twenty-five ICs, or simply yearn for the joy of bashing out a few justly tuned power chords, here's a method of converting a guitar or bass to a reasonably accurate form of Just Intonation, requiring only a few hand tools and a few evenings of labor. Caution: The techniques that follow are not meant to be applied to any instrument where resale value is a consideration. If, at some point, you decide that you are committed to Just Intonation, and want to have a good guitar refretted, you should have this done by a professional luthier. In the meantime, you can use the method outlined below to convert a pawnshop special into a valuable new musical resource.

The first step in refretting a guitar is to remove the old frets. This is easily done, using either a pair of right-angled flush cutters or a small, sharp screwdriver. The frets should be pried loose beginning at either edge of the fingerboard, using a rocking motion and working gradually toward the center. Care must be taken not to peel the veneer of the fingerboard from the neck during this process. The next step is to fill the slots in the fingerboard, using a commercial wood-filler such as Fix or Plastic Wood. Some luthiers insist that the correct way to fill the slots is to glue in small slivers of wood, and that if this is not done, the neck will warp. I and my colleagues in Other Music have refretted six guitars to date, using only Plastic Wood to fill the slots, and have never observed any warping. Use your own judgment. After filling the slots, sand the fingerboard back to its original smoothness. Initially use a coarse grade of sandpaper (such as 120) to remove most of the excess filler, then use a succession of finer grades to remove scratches and smooth the surface. 600 grit is a good choice for the final work. At this point, check that there are no notches left unfilled where the old fret lines meet the edges of the fingerboard. If such notches remain, they should be carefully filled and resanded, or they may cause much frustration during the following phases of our task.

If we knew the exact positions at which to place our new frets, and had the proper tools and skills, we could replace our old frets with a new set of permanent metal ones. Unfortunately, it is very difficult to calculate the exact position of the fret for any given interval on any given guitar neck. The laws of physics indicate that the pitch of a string of any given diameter at any given tension is inversely proportional to its length. Therefore, to raise the pitch of a string by a perfect fifth (3/2), it is necessary to reduce its length to 2/3. If we could stop a string by grasping it from both sides, without displacing it from a straight line, this rule would be sufficient to predict the stopping points. However, when we stop a guitar string by pressing it against the fingerboard, we not only decrease its sounding length, but also increase its tension. How much the tension increases depends upon the height of the action and the exact stopping point, among other factors, making a mathematical formula for fret placement beyond our reach. The solution to this difficulty is to use moveable frets and to determine their placement by ear (or with the aid of a tuner calibrated in cents, if one is available). **Fig. 1** illustrates the technique for tying a fret knot. The proper material for tied frets is monofilament nylon fishing line. Al-

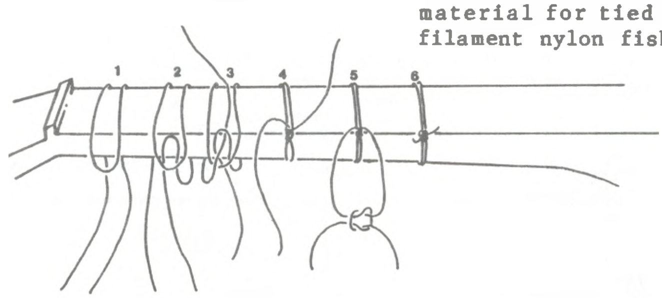



Fig. 1: Tying a Fret Knot

1) Beginning with about 18" of monofilament, lay a loop across the fingerboard (under the strings, of course).
2) With one of the free ends, make a second loop, and draw it through the first.
3) Take the other end over the first loop and through the second.
4) Grasp both free ends firmly and pull the knot tight.
5) When the fret is in its proper place, tie a square knot to hold it there.
6) Trim the ends.

most any weight from 20 to 60 lb. will work satisfactorily, depending on the action of the particular instrument. If nylon rather that metal frets strike you as odd, take note that this practice has historical precedence. The lutes and viols of the Renaissance and Middle Ages had tied frets, and many ethnic and folk lutes and guitars still use this traditional technique.

In order to get a reasonably useful just scale without an unmanageable number of frets, it is a good idea to discard the usual E-A-D-G-B-E guitar tuning in favor of one that repeats in octaves, i.e. a tuning of alternating fourths and fifths or fifths and fourths. Then it is simply a matter of locating the positions for frets which yield the low-number-ratio intervals we desire. For best results, initially tie your frets an inch or so closer to the nut than you expect the final position to be, then slide them down the neck to the desired location. The increased tension due to the taper of the neck will help to insure a tight knot. However, note that if you have left any unfilled notches at the edges of the fingerboard, your new frets will stick inextricably at the old position.

**Fig.** 2 shows the fret positions for a useful scale. I find a fretting such as this suitable for melodic playing, but something different would need to be devised for rhythm guitar-style chording. Note that this fretting doesn't extend up the neck beyond the major sixth. This is because the curve of the neck where it joins the body of the guitar prevents the tied frets from staying in place beyond this point, unless one notches the neck to keep them from slipping. (Once the exact position for any fret has been determined, notches at the edges of the fingerboard will make it easier to keep the fret in place.) On the portion of the neck closer to the nut, it is possible to place frets as close together as a syntonic comma (81/80). It is not particularly difficult to play single-note lines with frets spaced this closely, but barring across several strings may prove impossible.

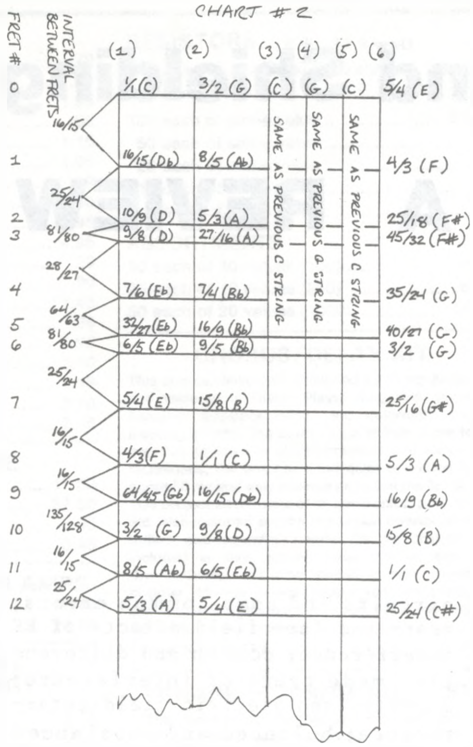



Fig. 2: A usable fret pattern, based on an open C tuning.
Note that this drawing is not precisely scaled, and should not be used as a guide for exact fret placement.

While using a tuner is the easiest way to determine fret placement, one can do a decent job checking by ear. Each new fret usually can be tested by several intervals. **Fig. 3** shows the checks to be used with the fretting in **Fig. 2.** Best results may be obtained on electric guitars with good sustain. Even distortion can be used to good advantage, as it highlights the beating that characterizes mistuned consonances. This process of tuning by ear can do much to refine one's discrimination among intervals, a desirable skill for any musician.

Good luck with your tunings, and feel free to write in to Polyphony with any reactions to your experiments.

OPEN STRINGS

| STRING # | HARMONIC | | STRING # | HARMONIC |
|---|---|---|---|---|
| 1 | 3 | = | 2 | 2 |
| 1 | 2 | = | 3 | 1 |
| 2 | 4 | = | 3 | 3 |
| 3 | 3 | = | 4 | 2 |
| 3 | 2 | = | 5 | 1 |
| 4 | 4 | = | 5 | 3 |
| 1 | 5 | = | 6 | 1 |
| 5 | 5 | = | 6 | 4 |

FRETS

| NEW FRET | STRING # | INTERVAL | CHECK | STRING # |
|---|---|---|---|---|
| 8 | 2 | 1/1 | 0 | 3 |
| 10 | 1 | 1/1 | 0 | 2 |
| 3 | 3 | 1/1 | 10 | 2 |
| 7 | 5 | 1/1 | 0 | 6 |
| 2 | 2 | 3/2 | 7 | 3 |
| 5 | 2 | 3/2 | 8 | 3 |
| 6 | 6 | 1/1 | 10 | 5 |
| 6 | 6 | 2/1 | 0 | 4 |
| 1 | 2 | 3/2 | 6 | 3 |
| 9 | 2 | 1/1 | 2 | 3 |
| 11 | 1 | 1/1 | 1 | 2 |
| 12 | 4 | 1/1 | 0 | 6 |

Fig. 3: A guide to the cross-checks used in ear-tuning the fretting given in Chart #2.
For best results, place the frets in the order given. The open strings can be tuned accurately by matching harmonics. Note: fret #4 is a special case; it cannot be matched to any other fret by a unison, octave, or fifth. The best available check is the seventh harmonic of string one, which is one octave higher than fret #4, on string five.

# Grounding and Shielding Seminar : A REVIEW

BY: VANESSA ELSE

Proper grounding and shielding is becoming increasingly important in this age of low level signals, nanosecond speed logic circuits, and electromagnetic pollution. To learn more about the subject, I recently attended an instructional seminar on grounding and shielding electronic instrumentation in Palo Alto, California at the Holiday Inn. This one-day, eight hour course was geared toward professional electrical engineers as continuing education instruction and is offered by the Department of Electrical Engineering at the University of Missouri-Rolla. The instructor was their faculty member Tom Van Doren who has 21 years of teaching and industrial experience in electronic data acquisition, microwave communication systems, semiconductor processing and electronic circuit design. Polyphony readers who are professional engineers may be interested in this course, particularly if their company has had any grounding and shielding problems crop up in their designs.

The course covered a wide variety of typical problems, and gave me a variety of topics to consider in my work with printed circuit board layouts and studio wiring. Both analog and digital grounding problems were discussed. Among some of the topics were: electrical wiring in buildings; improper wirings and the problems caused by such wiring errors (both what to be aware of and what you should look for); ideas on printed circuit board grounding; proper methods for measuring electrical circuits and some common errors; near- and far- field effects of RF interference; common and differential mode types of interference; magnetic and electric field interference; balanced and unbalanced circuits; methods of identifying noise sources; and shielding with different types of materials.

Those in attendance seemed pleased by the content. Although the course was very general, many of the attendees had specific questions related to their own fields of interest and these were either answered during the session or, if they were more involved, taken care of during the many breaks throughout the day. Mr. Van Doren frequently encouraged participation by presenting problems for the engineers to solve. In addition, he conducted several enlightening demonstrations with the visual aid of an oscilloscope.

Those interested in taking this course should have some familiarity with electrical engineering concepts and an understanding of the mathematics involved. For background before taking the course, or for general background on the subject, I would recommend Noise Reduction Techniques in Electronic Systems by Henry Ott (Wiley, New York, 1976) which will give you a familiarity with some of the material covered.

The cost of this one-day course is $285 which includes: a detailed set of course notes in a workbook, a book entitled Low Level Measurements for Effective Low Current, Low Voltage, and High Impedance Measurements published by Keithley Instruments, Inc., lunch, and coffee, tea or soda during the several breaks taken throughout the day. This course has been on the road to Illinois, Texas, New York, Massachusetts, California, Arizona, and Colorado with upcoming dates planned for Florida some time later this year. It is also available for in-house training, with length and content adjusted to suit special needs (the cost for the eight-hour in-house program is $3,800 at the time of this writing). This course is scheduled to be presented at:

| | |
|---|---|
| Washington, D.C. | May 6, 1985 |
| Philadelphia, PA | May 8, 1985 |
| Patterson, NJ | May 10, 1985 |
| Dallas, TX | May 14, 1985 |
| Houston, TX | May 20, 1985 |
| San Diego, CA | June 4, 1985 |
| Los Angeles, CA | June 10, 1985 |
| San Jose, CA | June 17, 1985 |

Also, Dr. Van Doren will be teaching an extensive three day course on Grounding and Shielding at Rolla, MO (April 22, 23, 24 -- $660) and another 3-day course on Interfacing Computers to Electronic Instrumentation (April 1, 2, 3 -- $660) at Rolla. Additional information may be obtained by writing to: Mr. Bill Kratzer, Engineering·Continuing Education, University of Missouri-Rolla, Rolla, MO 65401.

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SERGE Music System MODULES WANTED! Please call or write Andrej Zdravid, 37 Dolores Terrace, San Francisco, CA 94110 ( 415) 626-4386; or call Woody (505) 473-0614.

FOR SALE: TEAC 3340-S $500, BiAMP 1283 12 input mixer $500, Electro-Voice CS15 Condensor Mic $150, Neuman BS45A Phantom Power $50. Call Phil Cibley (212) 986-2219.

ARP 2600 Synthesizer, excellent condition - $700. Call (904) 222-9242.

INTERNATIONAL ELECTRONIC MUSIC ASSOCIATION. Join Charter Members Craig Anderton, Klaus Schulze, and Synergy in promoting the Art. Dues $12 annually include Syne Magazine. P.O. Box 176-P, Salamanaca, NY 14779.

## AD INDEX



## ADD A NOISE GENERATOR TO YOUR ORGANTUA

**by: Ronald L. Oberholtzer**

After my ORGANTUA gave me many hours of pleasure, I decided to try adding a noise generator to simulate the wind noise present in some musical instruments (such as pipe organ). I knew it was a success after a concert for my organ club; for the last number I asked them to tell me what instrument I was imitating, and before the number was finished they were yelling "**Calliope.**" The noise generator effectively imitated the steam in the whistles.

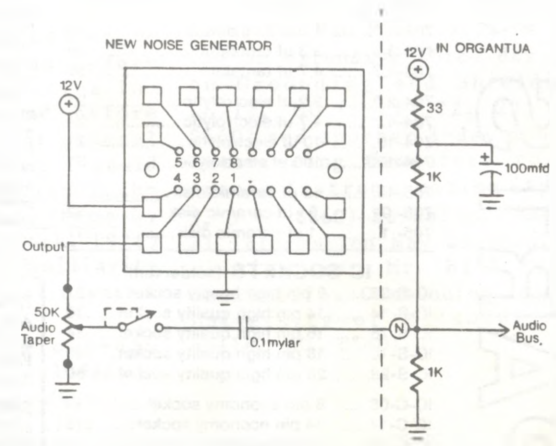
The associated diagram and parts list are self-explanatory, and mounting the components is left to the builder's creativity. Here's my calliope patch for Organtua; play thirds with your right (or depending on your preference, left) hand.

Clock: VAR
Attack: 8 o'clock
Modulation: OFF
Range: Next to Top, All Ranks
Octave Jump: On
Noise: 11 o'clock

**Parts List** (all stock numbers are for Radio Shack parts)

IC board (#276-024)
8-Pin IC socket (#276-1995)
Noise generator IC (S2688 or MM5837)
100K audio taper pot with switch (#271-1722) and knob
0.1 uF (100 nF) mylar capacitor (#272-1069)
Solder, wire, mounting hardware

(Ed. note: **Organtua** is available for $299.00 plus shipping and handling from PAIA Electronics, Inc., P.O. Box 14359, Oklahoma City, OK 73114.)

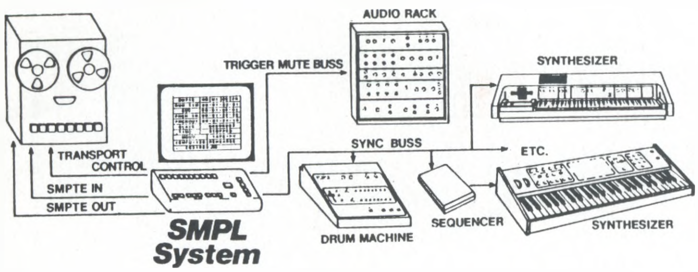


# AT LAST! COMPUTER AUTOMATION FOR THE SMALL STUDIO

### THE SMPL SYSTEM BREAKS THE PRICE BARRIER FOR SMPTE TIME CODE

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

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




### IT'S THE ENGINEER YOU ALWAYS WANTED

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

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

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

### SYNC-LOCK THE NEW GENERATION OF INSTRUMENT/RECORDERS

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

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

### MORE, HIGHER QUALITY "TRACKS"

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

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

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

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

### A VERY HUMAN INTERFACE

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

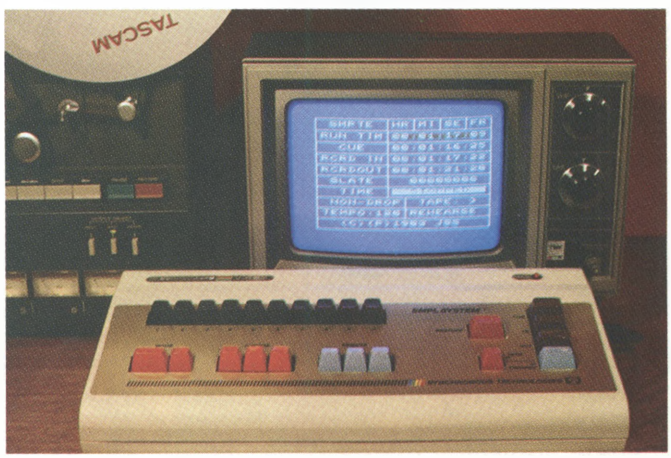
### IT'S A COMPLETE, LOW COST SYSTEM

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

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

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

**CALL OR WRITE FOR THE NAME OF YOUR NEAREST DEALER.**



No representation that SMPL is a product of Commodore Business Machines, Inc. or an affiliated or related company is intended; nor is there any representation that there is any source of origin of Commodore Computers other than Commodore Business Machines, Inc. or its affiliated or related companies.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# *Reflection Perfection...*

. . .whether you're mixing down from multi-track or producing audio-for-video. TOA's new **ME Studio Monitor** outputs a crystal-clear mirror image of **any** input.

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each one easily handles the wide dynamic range & precise acoustic demands of digital and advanced analog sound.

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