Polyphony

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ELECTRONICS - MUSIC - HOME RECORDING

February 1983

TWO NEW SYNTHESIZER MODULES:

SHEPARD FUNCTION GENERATOR

&

DYNAMIC TOUCH CONTROLLER

NEW AGE MUSIC

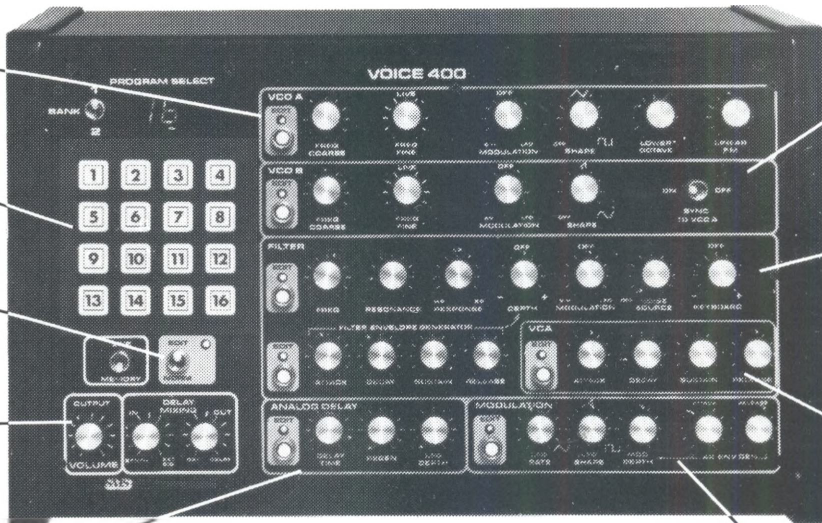
SYNTHESIZED CHOIRS

PRO-ONE DYNAMICS MODIFICATION

ISSN: 0163-4534

THE VOICE 400

The Fastest, Most Versatile and Musical Synthesizer Voice Available



Oscillator A
continuous waveshaping, variable pulse width, modulation by S/H or LFO, lower octave, linear F.M.

Keypad and Bank Switch
Selects one of thirty-two presets.

Operating Mode Switches
control Live, Memory and Edit functions.

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

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

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

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

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

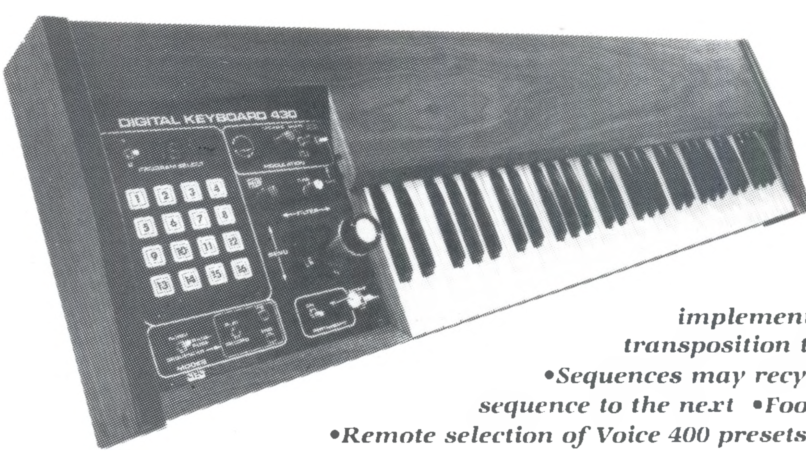
Modulation
wide range Low Frequency Oscillator with continuous waveshaping of three waveforms. LFO may be modulated by the Attack Release envelope generator.

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

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

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

#430 Digital Keyboard Sequencer



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

SMS

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

STAFF

PUBLISHER
John S. Simonton, Jr.

EDITOR
Craig Anderton

MANAGING EDITOR
Linda Kay Brumfield

TECHNICAL ILLLUSTRATOR
Caroline Wood

CIRCULATION
Ramona French
Peggy Walker

BOOKEEPING
Cathi Boggs

PRINT PRODUCTION
Phuong Nguyen
SEMCO Color Press

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PO Box 20305
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VOLUME 8, NUMBER 2
FEBRUARY, 1983

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Cover Art By: Bobby Beausoleil

Editor's Notes

Our readership includes those who like more musical, and those who like more technical, articles. As you may have noticed, alternate issues tend to emphasize one more than the other. Last issue was more "musical", covering lots of information on recording, reviews, and so on. This issue, the pendulum swings towards a more technical orientation, and we have quite a line-up for all you synthesizer fans out there.

First off, there are two excellent articles on adding dynamics to electronic instruments. Steve Wood tells us how to retrofit Sequential Circuits' Pro-One with PAIA's new Veloci-Touch circuit; the same principles described in his article also apply to retrofitting other keyboards. Bobby Beausoleil writes about a fascinating new controller which also provides dynamics, but in a different way. His DTC is intended to be designed into a variety of systems, from processors to instruments. It's a pretty universal device, and what's more, it's inexpensive and fun to use.

And now, for those of you who have been wondering whatever happened to John Simonton, Lab Notes is back! John is certainly making a splashy re-entry with his Shepard Function Generator module. This is the first module I've seen which is specifically designed to generate "infinity" functions, but it also looks like a winner for panning and synchrosonic applications.

These three articles are excellent arguments against those who complain that there are "no new modules". In fact, these articles, taken together, show that we indeed have many more frontiers to explore when it comes to musical electronics.

We've got some fine applications articles as well. Thomas Henry tells us all about how to devise a patch-over scheme for small synthesizers, which should help those of you who are trying to resolve the playing live vs. playing in the studio dichotomy. On a related subject, Charles Lauria discusses bus distribution in modular systems. His approaches are equally valid for signal processing systems which use voltage controlled modules.

David Vosh tells us how to go beyond the ADSR, and Jack Orman, a popular writer for DEVICE, describes a new AMS-100 module. And if you get tired of building, you can learn how to synthesize choirs by following the latest installment in Bill Rhodes "Applied Synthesis" series. Or, if you're looking for some new kinds of music as well as new kinds of technology, check out the article on New Age music by Don Schwartz. There's a whole new synthesis of acoustic and electronic sounds happening, and Don introduces us to this increasingly popular musical style.

Of course, we have the regular record reviews by Robert Carlberg, Current Events, some very interesting letters, and reviews of Mattel's Synsonic Drums and MXR's Dual Limiter.

All in all, this is quite an issue. So, let's give a thank-you to the above people who were willing to share the results of their work with fellow readers. Polyphony has always depended on its readers to come up with innovative, interesting, creative articles...and you've never let us down.

Next issue? Well, we have a bunch of interesting mod articles on file, plus a great article on applying the latest digital filter components. And of course, we've got a few surprises up our sleeves as well. See you then.

* * * *

While we're on the subject of thank-yous, I greatly appreciate all the kind words and letters of support about Polyphony. It's not easy editing a magazine with such a diverse readership, but your feedback certainly helps make the task a lot easier. Keep those cards and letters coming -- don't just wait for a reader survey. If you really like an article, drop us a postcard. It only costs you 13 cents, and the result will be a magazine which more closely fulfills your desires.

Finally, I'd like to call your attention to some people who are less visible than me in Polyphony's operation, but are very much deserving of your compliments. Caroline Wood is the person responsible for the spiffy looking diagrams we've been running lately. She's fast and accurate, which is about all anybody could ever ask for from a technical illustrator...aside from a good sense of aesthetics, which she also has. Linda Kay Brumfield is the glue which holds this magazine together; she's the one who makes sure the magazine gets laid out, bound properly, and mailed. She also deals with advertisers, and in fact, does just about everything else required to get this show on the road. She works very hard for this publication, and is, simply stated, invaluable.

John Simonton is an editor's dream publisher. He gives complete editorial leeway, and when he does have comments, they're always of a helpful nature. Then there are the folks in the front office who handle missing issues, complaints, deal with the mail, and so on. And there's Vesta Copestakes, who has produced a string of beautiful covers despite a heavy schedule as head of a respected ad agency (this issue she has taken a well-deserved vacation, but she'll be back).

Add it all together, and you've got a magazine. We're all real happy with the way this publication is going, and your cards and letters seem to indicate you're pretty pleased too. Great! Look for more exciting articles and ideas in the months and years ahead.

Craig

Craig Anderton

Letters

DIGITAL CLOCK TIP

In Jacques Boileau's article in the March/April Polyphony, you may eliminate the 4013 when building the clock/modulation circuit by using the 4046's phase comparator #1 as an inverter (actually, it's an EX-OR gate). Connect pin 14 to V+ (pin 16), and pin 3 to the VCO output at pin 4. The output of comparator #1, at pin 2, is the inversion of the pin 4 output. This method was used in the ARP Omni chorus/phaser circuit.

Jim Rumberg
Johnson Creek, WI

Jim - Thanks for the tip about the 4046. However, bear in mind that the current capacity of the 4046's phase comparator is limited. Thus, if you're driving a delay with a lot of buckets, or trying for very short delays, you may need something with more ooomph.

PUBLISH OR PERISH

I want to release a single this winter, produced and played by myself. I've got the music together, found a good studio, got the copyright forms filled out, but I'm stuck as to how I get the song published. Do you have a book with this information, or can you start me in the right direction? Any info would be valuable.

Donnie Bedford
Houston, TX 77035

Donnie - Help is no further than the center page of this magazine, where you can order "How to Make and Sell Your Own Record" by Diane Sward Rapaport. It gives lots of information you'll need, including the basics of setting up a publishing company. I also recommend the book "This Business of Music" by Sidney Shemel and William Krasilovsky (Billboard Publications, New York, 1977). Good luck!

FLUTE CONTROLLER?

I am a flutist interested in electronic music and would like to make a flute controller with four independent controls (something like a "flute Lyricon"). Are there any circuit diagrams available for these kinds of controls? Or, is it possible to get the circuit diagrams of the "Wind Synth Driver" (by Computone)? I will appreciate you or your readers can do for me concerning this.

Pedro Eustache
17 Rue Chauvelot
75015 Paris
France

Pedro - I don't play flute, but it would seem to me that a flute produces a simple enough waveform so that any pitch-to-voltage converter designed for guitar (Roland, Gentle Electric, etc.) would work well with flute. This would allow you to use your standard flute with synthesizer modules, and not have to invest in an extra controller. You could also tap an envelope follower off the audio output for dynamic control. Then again, maybe other flute players out there have some other solution...so if you have any ideas, write Pedro at the address above.

SINGAPORE SPEAKS

First of all, let me tell you that Polyphony is one of the best magazines that I've ever subscribed to. And what makes it most interesting is that most of your articles are written by the readers themselves; I think it's a fantastic way of running a magazine.

How about some more articles on multi-track recording, particularly how to get the most out of a TEAC Portastudio. I was wondering if you or any Polyphony readers could turn me on to articles written about the Portastudio which may have appeared in other publications.

Finally, I'd like to say that the "On Location" pieces are extremely fascinating and I hope you maintain this as a regular feature of the magazine.

Once again, thanks for a magnificent publication.

Soo Khian Teck
5, Lorong 4,
Realty Park
Singapore 1954

Soo - Thanks for the nice words. Concerning the Portastudio, I can't recall any articles on getting the most out of this unit. Perhaps some readers would care to write to you directly if they have any "hot tips".

Re "On Location": This was always one of my favorite features too. Unfortunately, the reader's survey indicated that most readers didn't agree. I think part of the problem is that is has been very difficult to get articles from foreign readers, so the column never quite had the orientation I had hoped for. But let's give it one more try, okay? Any readers living outside the US are invited to write a short article on the electronic music scene in your country. How easy is it to find parts? Are there any popular electronic music artists in your country? What kind of equipment is available? What kind of problems do you run into (customs etc.)? Are there many do-it-yourselfers? What kind of studios are there? And so on. Don't worry about your English, that's what editors are for. I'd really like to make this column go, but this will only happen if the readers co-operate. So get out those pens and pencils and tell us what's happening in your country. I'm particularly interested in seeing information from Australia, Brazil, Switzerland, France, Germany, and the Orient.

SENSITIZING THE AMS-100

I'd like to make my AMS-100 modules more sensitive, so that they work on 0 to +5V control voltages instead of 0 to +10V. What should I do?

Glynn Black
Palmetto, GA

Glynn - Simply double the value of the feedback resistor in any control voltage summing stage to twice its original value; or, decrease the value of the control voltage mixer channel attenuating resistor by one half.

Letters

PC BOARD TIP

I'd like to share a construction tip that I think you and your readers might be interested in. For many years, I have tried to make a good PC board. Finally I have found a way to do that thanks to one of your suggestions and my own haphazard experimentation.

I start with a clean copper clad board (obviously). Then I lay out the holes for the components with a direct-etch dry transfer similar to the lettering you mentioned in "Electronic Projects for Musicians" for labelling projects. This transfer is available from Radio Shack (#276-1577). It has many different configurations for most any component plus straight lines and various sizes of holes for resistors etc. I only use the holes on these sheets; the lines are very difficult to use.

Once I have the holes laid out, I use Testor's model paint to "connect the dots", so to speak, and complete the circuit.

After removing the board from the etchant, I found the fastest and cleanest way to remove the paint and transfers is to use fingernail polish remover. The brand is Cutex Oily. I just use a cotton ball soaked with the polish remover and wipe the paint and transfers right off. This leaves you with a good quality board with no broken copper circuitry.

Again I'd like to thank you for bridging the gap between electronics and music for amateurs. And just in case you're thinking of publishing schematics for a small chorus, attack delay unit, or maybe even a wireless, I'm anxiously waiting.

Scott Hampton
Greenwood, IN

MG-1 REVISITED

I've read the review of the MG-1 Synthesizer in the May/Aug '82 Polyphony, and as owner of both the MG-1 and the repair manual, the article contains some

continued on page 38 ....

Robert Carlberg's re-view

For anyone who's interested, I'd like to state a few policies on my reviews. First, if I don't like a record I say so. I'm learning to be more subtle, but I don't intend to get into the "you scratch my back and I'll scratch yours" syndrome. I don't want to write glowing reviews just to get free records, and unfortunately much of the industry seems to work that way. For this reason I am somewhat suspicious of any record which resorts to printing favorable reviews in their ads or on the cover.

Second, although my tastes in music are varied, I don't expect everyone to agree with me. I'm trying to describe as well as react to the music, so that you might see something interesting even if I miss it. Bear with me; it's a temptation to put down things I don't understand and I'm constantly catching myself doing it.

Third, even within the confines of specialized category such as electronic music, there are more releases than a body can get to (or that a magazine can print - Ed.). I buy what records I can afford in the styles I like best, reviewing them along with whatever promos I get in the mail. If you don't see what you'd like to see reviewed, either get after the distributor to include Polyphony or turn me on to it.

Last, the span of time from the writing to the printing of this column can be quite long. If the material seems kind of out-of-date or if you're waiting for mention of your release, my apologies of behalf of the magazine. (Note: Music indicated with an asterisk is available through New Music Distribution Service).

Wendy Carlos **Tron Soundtrack** (CBS 37782). Incidental music to an incidental film. The parts that sound like a real orchestra and chorus really are an orchestra and chorus -- new vistas for Carlos.

National Health **D.S. Al Coda** (Europa 2008). Dedicated to the late Alan Gowen, featuring nine of his recent compositions which he had been arranging for large band. The 4-member Health is joined by seven additional musicians in a joyous celebration of one of English Jazz's finest.

Artless Time **Homages** (Random Access Music 801; E.P.). Ensemble tuned percussion music which develops a sort of dreamy, laid-back feel. Not as rigid as Gamelan or Steve Reich.

King Crimson **Beat** (Warner 23692-1); Adrian Belew **Lone Rhino** (Island 9751). Both are full of fabulous guitar effects, and Belew's has some of the nicest rhythm tracks I've ever heard. Unfortunately both also share Belew's 3-note singing and ludicrous lyrics.

Michael Rother **Fernwarme** (Polydor 2372-111). Guitarist/synthesist Rother has never been a flashy player, demonstrating instead that it is more important to know when NOT to play. Rock-steady Jaki Liebezeit as usual provides drums for the dignified tune-weaving, with rather more emphasis on synthesizers this time out.

Emerald Web **Aqua Regia** (cassette). Kat and Bob concentrate on synthesizers here instead of the flute/synthesizer meditation music for which they are known. Some great tones result, a little darker than usual but still beautiful. In places it sounds more like Michael Gilbert than Web. $8.98 from 29 Canterbury Lane, Unionville CT 06085.

Music and Rhythm (PVC 201). A compilation of 12 Western artists using ethnic influences with nine authentic ethnic artists from Africa, the Near and the Far East. While the ethnic artists aren't the best of their lot, the Western artists almost certainly are: Peter Gabriel, Morris Pert, Jon Hassell, Rico, David Byrne, Vic Coppersmith-Heaven, etc.

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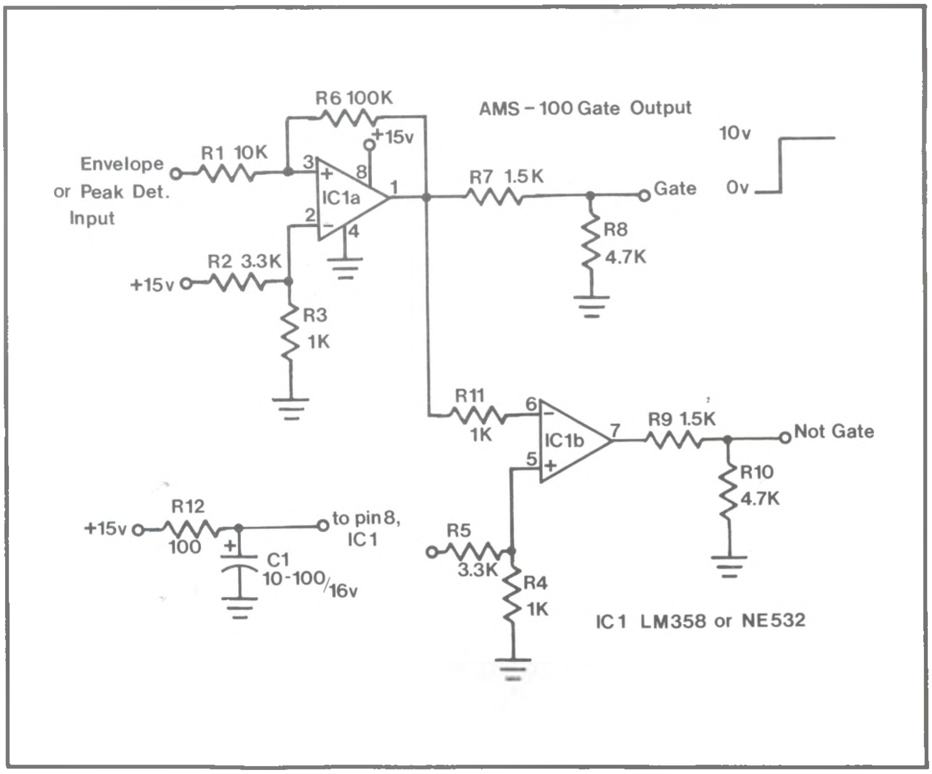
AMS-100
Gate Output

By: Jack Orman



There seems to be a reasonable amount of interest in additional modules to round out the AMS-100 series, first presented in DEVICE. I have the majority of modules built and mounted in my pedalboard, in addition to a few compatible circuits of my own design. Hopefully by presenting some of my ideas, other experimenters will be stimulated into writing about their own developments for the AMS-100 series.

Some of the modules that I wanted to use with my AMS-100 needed a gate type input rather than the trigger which was already available; therefore, I came up with this circuit, which works off of the envelope or peak detector outputs to produce the gate as well as an inverted gate.

Since conventional op amps will not switch completely to ground when operated from a single supply, a special device is required. The LM358 (or NE532) is specially designed to work on a single supply voltage and will give an output which swings to ground. This chip is inexpensive and available from a number of suppliers.

IC1A is a non-inverting comparator with hysteresis that produces a positive output (near V+) when the input signal exceeds a threshold set by the voltage reference on the inverting input, and the positive feedback through R6. This "on" threshold is about 3.5V in my prototype. For the comparator to switch off, the input must fall below the "off" threshold, which is somewhat lower than the turn-on point (about 2.3V to turn off). This difference in switching points provides some measure of immunity from ripple in the envelope.

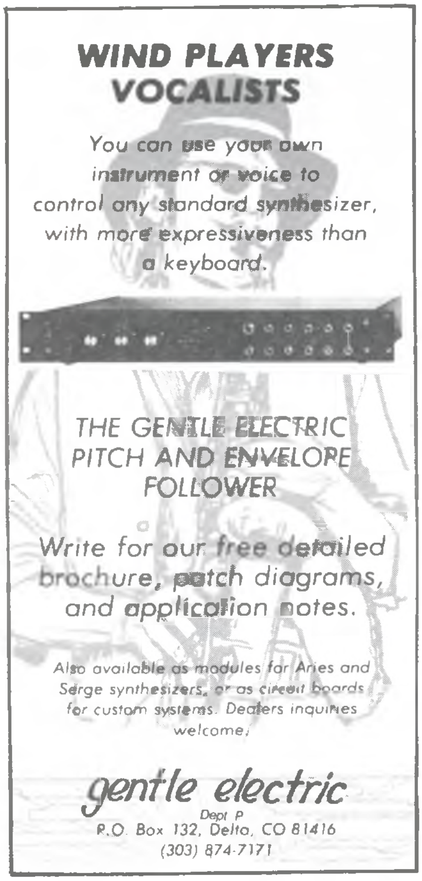
IC1B is a simple comparator since the noise immunity provided by the hysteresis is not needed. The inverting configuration is used here to provide the complementary inverted gate output.

Since the output is somewhat larger than the 10V required by AMS-100 modules, a resistor divider is used on each output to trim the gate signals to the proper levels (pairs R7/R8 and R9/R10). R12 and C1 provide power supply decoupling.

PARTS LIST

All resistors 1/4 Watt, 5%.

R1	10k
R2, R5	3.3k
R3, R4, R11	1k
R6	100k
R7, R9	1.5k
R8, R10	4.7k
R12	100 Ohm
C1	10 to 100 uF @ 16V
IC1	LM358 or NE532



re-view

Don Slepian **The Rhythm of Life** (cassette); **New Dawn** (cassette). Waterfalls of synthesized background provide a shifting basis from which Slepian improvises. Somewhere between minimalism and meditative, with a dose of guitar, flute, and a Gamelan patch to add interest. **Largos for Learning, Loving, and Living** (cassette). Slepian breathes life into several Baroque chestnuts through use of delicate and beautiful synthesizer sounds. Even Pachelbel's Canon responds to a lovely Mormon Tabernacle patch. All cassettes are $8.95 each from Don and Judy Records, PO Box 836, Edison, NJ 08818.

David Borden **Music for Amplified Keyboard Instruments** (Red 002). To tell the truth, I think Borden is overrated. Consisting almost entirely of 32nd notes, his compositions here and with Mother Mallard tend to anaesthetize through sheer exhaustion.*

Paul Woznicki **Woz** (Ulterior 1000). Simple synthesizer, Rhodes and rhythm box constructions, but like Ghostwriters, includes enough surprising choices and courageous faux pas to set it above the din.*

Alireza Mashayeki **A. Mashayeki** (Retro 180); **Alireza Mashayeki** (Retro 280). Computer music utilizing the by-now familiar elongated bell-tones. An Iranian, Mashayeki includes some traditional elements of his homeland music to modulate the randomness somewhat.*

David Rosenboom **Future Travel** (Street 002). Longtime BSEM staff member is understandably enamored of technological ways of doing things, and these seven tracks exemplify bio-feedback in a computerized synthesizer system. The addition of piano, violin, ethnic percussion and his wife's voice through a harmonizer further offset the influence of the computer.*

Drahcir Ztiworoh **Eros in Arabia** (Ethnotech 777). Richard Horowitz plays Prophet 5, oblique flute and on one track, piano. The Prophet is set up with percussive voices and played very fast, giving almost a Gamelan effect. The piano track is a similarly percussive tribute to Conlon Nancarrow.*

John Holland **Music from a Small Planet** (American Sound 1001). Endlessly overdubbed sequencer patterns form dense sheets of sound. Background music for the terminally neurotic.*

Eric Ross **Electronic Etudes (Op. 8)** (Doria 103). "Contemporary classical/avant-garde" compositions for piano, bassoon, clarinet or voice with Moog III. The liner notes try hard to explain them.*

Frank, Olan **A.C.A. Recording Award** (CRI 419). Two contemporary classical compositions by each composer, one by Olan for clarinet and tape. An acquired taste but near the top of their genre.
McLean, Korte, Hanlon **Extended Saxophone** (CRI 431). Contemporary avant-garde featuring three composers and a saxophonist from the University of Texas at Austin. Tape manipulations produce some very non-saxophonic sounds. Hanlon uses a Frippertronics loop.

Hiller, Yttrehus, Maslanka **Portfolio, Quintet, Three Pieces** (CRI 438). An aleatory piece for singers plus tape, a quintet characterized by turmoil and three catfights between clarinet and piano.

Todd, Semegen, Wells, Greenwald **Satan's Sermon and other Electronic Fantasies** (CRI 443). Synclavier and Buchla synthesizers employed by four composers with academic backgrounds. Predictably unpredictable rimg-mod bell tones.

David Behrman, Paul De Marinis, Anne Klingensmith **She's-A-Wild** (Record 8101-54; single). Three distinctly different novelty tunes featuring Anne's singing and a recurring theme of cannibalism. Good clean fun.*

Peter Gabriel **Security** (Geffen 2011). When Peter Gabriel and Genesis parted ways, it was like cutting up a flatworm - Gabriel grew a new band and Genesis grew a new singer. Some say, as the former head of the band, Gabriel took the brains with him. Full of considered lyrics, exotic synthesis and elaborate processing, these tunes do little to contradict that view.

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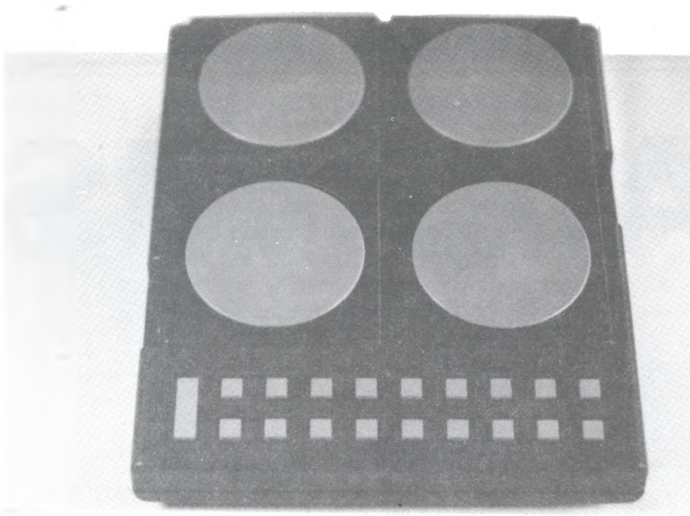
SYNSONICS DRUM

By: Robert Carlberg



Is it a toy or a musical instrument? Do you buy it in the supermarket kiddie department or at the keyboard store?

Well, a little of both. Mattel has marketing outlets that you might not normally shop, but the Synsonics Drums is far from a toy.

At $150 list, the Synsonics Drums is a good value for a programmable rhythm box, even without considering some of its unique features. The layout and capabilities are a little different from your standard Dr. Rhythm-type programmable rhythm box, however, so it bears a little investigation before purchase to make sure it's what you want.

The Synsonics is housed in a molded plastic shell about the size and shape of a hardback dictionary. Four hard rubber drumpads atop the unit act as triggers, and an array of buttons (in the same hard rubber pressure sensitive material) lies across the front of it. Each of the four drumpads controls one voice: Tom Tom 1, Tom Tom 2, Snare, and Cymbal. Here is the first of the disadvantages - there is also a bass drum, but it is set to once a bar. The overall tempo can be varied with two of the front-mounted buttons, and it can be defeated with a third, but it is not programmable. Accordingly, it is not recorded when storing a rhythm, but comes on automatically when playing back unless defeated.

An LED near each drumpad indicates when each voice has been triggered, with one more in the center for the bass drum. This is your basic timekeeping reference, and watching it helps you time out the programs. It flashes 16 times before the program repeats itself, although there are many more than 16 "slots" which can be filled. The length of each sequence (there are three separate memories) is set to sixteen measures, and each program will repeat indefinitely until stopped.

The 12 left-hand pushbuttons are also triggers for the voices - but unlike the drumpads they are quantized. The first button plays two beats per measure, the second four, and the third eight. In addition, pushing more than one button at a time is programmed to provide additional variety: depressing buttons one and two gives a 1-2-1-2 pattern slightly offset from the downbeat, buttons two and three give a double off-beat, and buttons one and three a single off-beat. Pressing all three buttons provides a sort of "lub-dub" beat they call "shuffle". It should be noted that these buttons are pressure sensitive, not mechanical, and that it takes a fair amount of pressure to hold them down. The drumpads won't even trigger on normal finger pressure - you have to strike them hard with sticks or your knuckles. One advantage here: the harder you strike the pads, the louder they sound (luckily, the case construction looks sturdy enough to stand up to the beating). Disadvantage: when you record them into memory instead of playing them in real time, they play back at a constant volume.

There is an oversize "accent" button on the left which shortens the envelope on the cymbal, creating a high-hat sound. This is, thankfully, recorded when storing. On the left side is a plastic "thumbwheel" control, which sets the basic pitch for Tom Tom 1. Both Toms are made up of a descending tone (like you hear from Synares) and Tom 1 may be tuned anywhere from a high-pitched squeal to about equal to the bass drum.

A similar thumbwheel on the right side acts as an on-off/volume control. An AC adapter port on the left, stereo RCA outs, and a 1/4" headphone jack on the right round out the basic controls.

Recording a sequence with the unit is where most of the strange features come in. First, on the plus side, depressing the "record" button does nothing at all - until you also depress one of the cymbal pushbuttons, which double as memory addresses. This prevents accidental erasure of one of your stored programs. Once you have depressed both keys, an LED momentarily lights to show which memory is being erased. Three seconds later, all five LEDs blink and the bass drum starts its rhythmic pulsing. Anything you hit after that - whether quantized pushbuttons or unquantized drumpads - will repeat after the first 16 notes go by.

But wait! The unit is still recording. Anything you hit the second or third time through will be recorded too, allowing you to layer more activity than you can play in real time. This can be somewhat disconcerting if you miss the five-LED flash which indicates the start of a new cycle. The unit will continue recording until the "stop" button is depressed. If you make a mistake - ooops, the whole memory must be erased and you must start over. There are no single-voice editing facilities - perhaps the unit's most serious drawback. But with a little practice mistakes on the pressure pads can be reduced, and you soon develop a "feel" for the instrument. Of course, if you simply miss a note you can fill it in next time the cycle repeats. One other disadvantage: all three memories are erased every time you turn the unit off.

In closing, manual triggering is the real strength of this unit. With its pre-programmed sequence length, unprogrammable bass drum and single memory playback, it is not as flexible as, say, a Dr. Rhythm for basic drum tracks. But used manually in conjunction with a Dr. Rhythm-type unit, or even a factory programmed rhythm box such as those made by Roland or Korg, it should provide a pleasing, non-mechanical variety to the overall sound.

Applied Synthesis:

By: Bill Rhodes

The Synthetic Choir

One of the most dramatic sounds is that of the human voice en ensemble. For years, the Mellotron and vocoder have been the best choice for economically and effectively emulating a vocal choir. The Mellotron voice is good (it better be, because the sound is genuine voices) but the tape sustain problems and attack idiosyncracies make it difficult to use. The vocoder produces a much more synthetic voice timbre, but at times is awkward to use because the breath unintentionally actuates the synthesizer mechanisms. The Chamberlin (another tape machine keyboard) is very nice but cost-prohibitive for most musicians. Another machine, the Roland String/Vocoder, is excellent if you wish to use a machine suited especially for voices (male and female) and strings. The price is competitive and the resulting sounds greatly resemble the human voice. However, if you do not want to buy any of the previously mentioned keyboards you can use the following information to duplicate the desired sounds on your keyboard synthesizer.

Let's dissect the qualities of the human voice in order to synthesize them correctly.

Waveform. A voice can change waveform depending on whether it's voicing vowel or consonant sounds. The "ah" and "oo" sounds used for background ensemble effects connote waveforms which have the least amount of harmonic content. The higher registration female voices center around a sine or triangle wave. They can be quite piercing, so try to equalize the higher octave voicings so that they aurally palatable. The alto and tenor male voices will contain more nasal characteristics of perhaps a pulse nature. But remember, these are approximations - female and male voices are a combination of many complex waveforms resulting in a complex composite. As we've explained earlier with string and organ synthesis, we need to be aware of this combination of many waveforms to produce the desired sound; on the other hand, you can get by with simpler voicings when you're only playing in specific registrations or footages.

The lower baritone and bass male voice has a rumbly or harsher overtone sound compared to the alto or soprano voices. The lower rate of vibrato and "darker" timbre suggest more harmonic waveform content than the higher voices. Very low male voices exhibit a cello-like quality and therefore assume a more pulse-like and sawtooth waveform. The chart summarizes the information.

We can use the organ as an analogy for the voices - add square wave upon square wave, footage upon footage, and you get a full timbre and complex voicing. The same principle applies to voice, but the envelope and vibrato dynamics must be different for truly vocal-sounding effects. In fact, certain pipe and electronic organs have a stop or tab called

Voices	contrabasso	basso	tenor	alto	mezzo-soprano	soprano	falsetto or "castrati"
Charac-teristic	cello-like	deep	horn-like	hollow	hollow	flute-like	sine or triangle
Waveform							or
	add delayed vibrato if needed						
Vibrato	low rate	low rate	moderate variable rates rather shallow depth			high rate	very high rate or non-existent

"vox humana". This human voice stop is a rough approximation of the desired sound, but many important parameters were generally not variable (envelope, vibrato, individual LFO modulation of each registration of voices, chorusing, etc.). With synthesis techniques, of course, we can get a much better approximation.

Envelope. Use a "breathy" or slow attack to approximate the embouchure of the human voice. This technique works well for sustained background voices. For a more pizzicato (scat-like) voice decrease the attack time and hope you can make your keyboard talk! A relatively long release time will blend polyphonic voices together so that they sound as if they are "singing" in a reverberent room. Full volume sustain should provide adequate keying with ease of playability.

Modulation. Since vocalists produce different small pitch changes and different delays of vibrato attack, this parameter is most difficult to deal with when synthesizing voices. Some people, for example, will start their vibrato slow and low in depth, and gradually increase the intensity according to the phrasing of the musical passage. Others start an instantaneous vibrato, while still others sing without vibrato. If we could have separate modulation (LFO) control on each voice of a polyphonic synthesizer, it would be much easier to duplicate these nuances; otherwise, use a delayed triangle wave vibrato (of a somewhat "warbly" nature) to imitate the voices from tenor on up. Delaying the entrance of the vibrato slightly will "feather" the voices and allow them to sound more genuine. However, be very careful when adding vibrato in the bass and baritone range because it will sound peculiar if it is too intense.

Chorusing. As mentioned in the "Keyboards Imitating Keyboards" installment (Polyphony May/August 1982 issue), slightly detuned oscillators produce a richer sound. This ensemble or chorus effect may also be realized by modulating the pulse width of an audio oscillator. The harmonic shift caused by the pulse modulation circuit (which is in a dynamic state of overtone stretching and compression) creates rich doubling sounds. The pulse wave can change dynamically to square wave and back again because of this modulation. Be careful, though, to carefully watch the rate and depth control settings. Pulse width modulation in certain keyboard registrations will seem out of tune if not controlled. Some of the more sophisticated poly synths have automatic pulse width modulation (i.e. the Polysix and Mono/-Poly by Korg are two examples).

Another way to assume the quality of pulse width modulation is to modulate the pulse width by the envelope generator waveform as it's attacking. Each consecutive attack can modify the waveform characteristic, thus producing a chorus sound of a less automatic nature when compared to straight pulse width modulation techniques.

I hope this article has shed some light on this particular area of synthesis. Voice is one of the most tedious sounds to electronically synthesize because of its many dynamic colorations and anomalies. As I always say with electronic music, however, experiment with your endeavors and don't be frustrated if you cannot get results right away - with practice, you will. See you next time, Ever-friends.

Veloci-Touch/Pro-One Interface

By Steve Wood

What's a Veloci-Touch? It's a new PAIA kit which magically turns any voltage controlled synthesizer into a new machine, by providing three different control voltage outputs (velocity, pressure, and A/R) from the keyboard dynamics. Each of these may be applied to any control voltage parameter.

While a velocity sensing keyboard is hardly a new concept, Veloci-Touch (VT for short) is unique in that it may retrofit existing synthesizers. It's easy, and versatile (both in terms of interface and application); it's fun, and best of all, it's inexpensive.

This article describes how to install Veloci-Touch in Sequential Circuits' Pro-One. While the details won't be the same for other synths, the general installation procedure will be. You don't have to worry about whether your synth responds linearly or exponentially, is monophonic or polyphonic, and the case bottom can be metal or wood -- either is cool. About the biggest compatibility problem you might run into (although it's easily surmounted) will be the logic sense of the instrument's gate output. For example, if you're adding VT to your mini moog, you will have to invert the Moog's gate output (S trigger) so that it goes from low to high when activated. This can be done with the simple addition of a single NPN transistor and a couple of resistors. For details, see the VT assembly and using manual (calibration section).

With traditional velocity sensing keyboards, each key usually has its own discrete output. While there are a variety of ways to do this, existing methods are all pretty costly and complex. The VT doesn't use info from each key, but rather, monitors the entire keyboard using one sensor. A piezo electric transducer (or "bender") mounts between the bottom of the keyboard frame and the case bottom, at the center of the keyboard length, towards the front edge. Any pressure change on the bender flexes it and causes voltage transients to appear across its terminals; these reflect both the applied pressure and the velocity. The VT processes these voltages to provide three control voltage outputs. These outputs may drive any voltage controlled parameters. Since the Pro-One provides us with plenty of voltage controlled parameters, to be as versatile as possible, we have added three rotary switches (not included in the VT kit) between the VT outputs and the control inputs of the Pro-One. This lets us apply any one of the three outputs to any one of four or five control inputs.

To prevent the possibility of shorting two outputs together, we've added a 1k resistor to each output (see figure 1; figure 2 shows how we mounted the resistors on the rotary switches). Otherwise, the outputs of two op amps within the VT could be connected directly together, which is not a good idea.

In figure 2, note that only one half of S1 and S3 is being used. Lug 7 is used like a free lug on a terminal strip, since it's assumed that there are no other connections to this lug. If you decide to use this side of the switch for something else, remove both the wire and resistor lead from lug 7, and connect the wire and resistor back together. Insulate the connection with tape or heat shrink tubing.

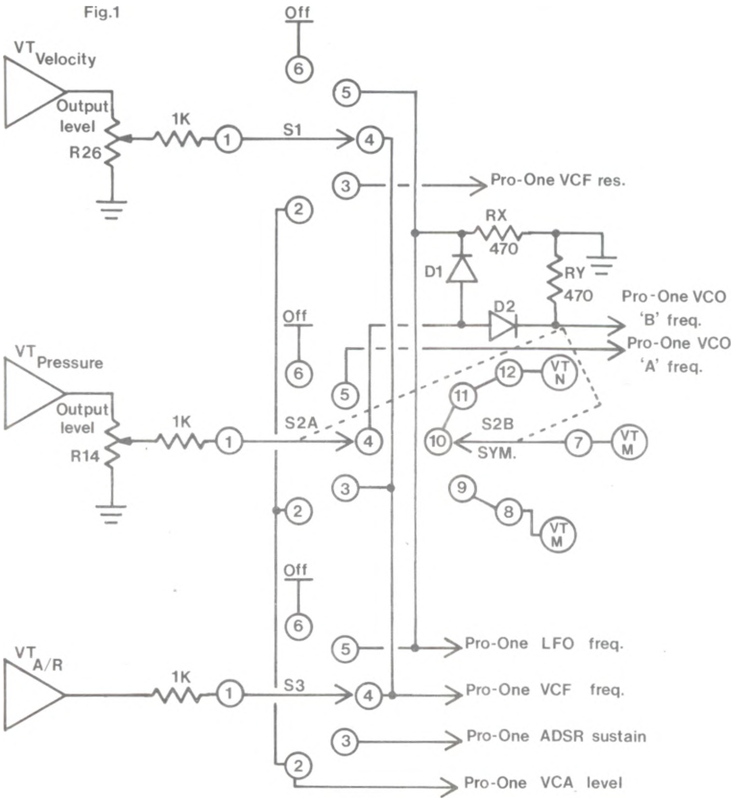
There are also a few more resistors (10% tolerance) required to pad control voltages at the Pro-One parameter control inputs, along with a couple of small signal diodes. So, you do need a few parts which don't come with the kit. However, if you are willing to assign each VT output to controlling one parameter (or two parameters at once), you can do without the rotary switches altogether.

You will need to drill some holes in the case top of your Pro-One for mounting the rotary switches and VT controls. In our unit, we mounted all seven controls and it didn't seem to make any difference in the structural strength of the case. We've included a printed version of the control panel graphics which serves as a drilling template and, if you wish, graphics for your project. Installing a thin piece of Plexiglas[R] or some other kind of thin plastic over the printed cutout adds a nice touch. Plexiglas is not hard to work with. It cuts fairly easily and is especially easy to drill. However, be sure to hold the plastic in a vise of some kind while cutting or drilling. It's kind of soft, so you have to be careful where the vise holds the plastic, and you must have good control over your tools. Also, you'll need pots and switches with mounting shafts and bushings which are long enough to accommodate both the case piece and plastic face plate.

One more thing: A hole must be drilled in the Pro-One PC board to facilitate mounting the VT circuit board, but be careful about where the drill bit goes (watch out for things like circuit board traces and fingers). Also remember that the Pro-One has a two-sided PC board, and has conducting foil on both sides.

Details. All you need for this modification is an assembled VT PC board. You won't need the front panel or output jacks J1-J5. As you build the kit, do not follow the point-to-point wiring schedule in

Fig.1



the VT assembly manual, since the wire lengths will have to be longer for this project. You will need some additional stranded/insulated hookup wire (#22 or so), the usual tools, and a drill with 1/8", 1/4", and 7/16" bits.

You will also need the following parts: Resistors, (3) 100k, (1) 68k, (1) 220k, (1) 4.7k, (2) 470 Ohm, (3) 1k; (2) 1N914 or similar signal diodes; (3) 5 position rotary switches; (1) 2 X 10 X 1/16 inch piece of Plexiglas (if you're doing that part); (2) 5/16" spacers; (2) #8-32 X 1/2" machine screws; (1) #8-32 X 3/4" machine screw; (1) #4-40 X 3/4" machine screw; and (1) #4-40 nut.

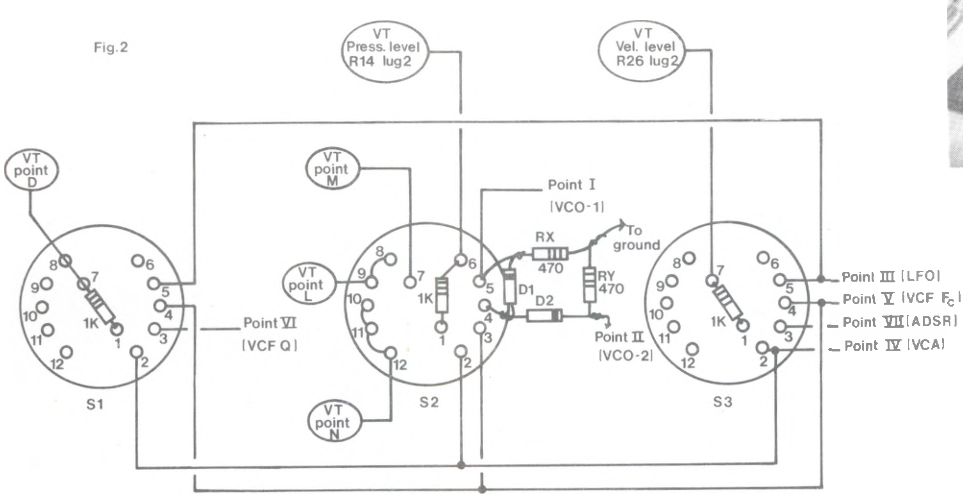
Disassembly. Here's Sequential Circuits' instructions for disassembling the Pro-One. Remember, though, while it may snap apart like Tinker Toys, it's still delicate machinery. Take your time and don't rush.

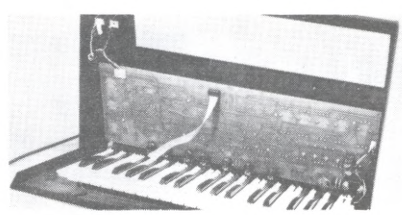
1. Switch power off.
2. Unplug power cord and disconnect all cables on back panel.
3. Remove wooden side panels (2 screws on each side).
4. Remove three screws along front edge. (Some units may also have one #8-32 screw near the power switch to be removed.)
5. Carefully slide top panel assembly forward. When front edge is clear of keys, lift it up just enough to allow clearance fór your hand.
6. Reach in and disconnect AC power connector running from back panel to underside of printed circuit board, at right. (When reconnecting, orient connector so the side through which you can see the metal contacts is visible.)
7. Also disconnect keyboard cable from PCB. (When reconnecting, the keyboard cable should be twisted so that the ribbon crosses over the board. If correct, the numbers 9-16 stamped on the connector will run along the PCB edge.) The top panel assembly will now lift away.
8. Pull off all knobs.
9. Disconnect wheel connector at left of PCB. (When reconnecting, orient connector so that the side in which you can see the metal contacts is visible.)
10. Remove 9 screws holding PCB to front panel (there are two screws near the transformer).
11. The PCB should now lift away from the top panel. If not, check that the GATE LED is not stuck or cemented.

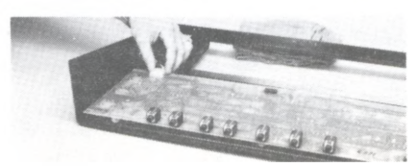
Be careful not to bump any of the calibration trimmers out of adjustment. Most of them are located near the left hand end, and the center of the PCB.

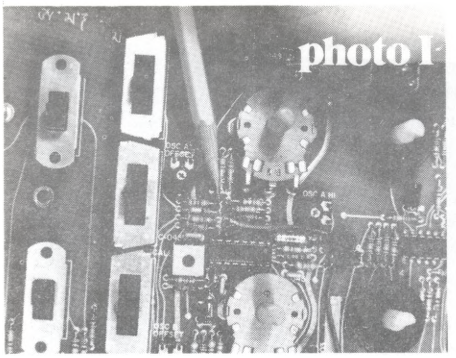
WARNING: LETHAL VOLTAGE IS PRESENT IN THE POWER SUPPLY AREA ON THE PCB (in earlier models) OR ON THE BACK PANEL (in later models).

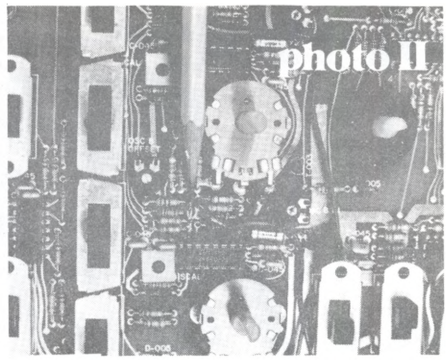
Fig.2

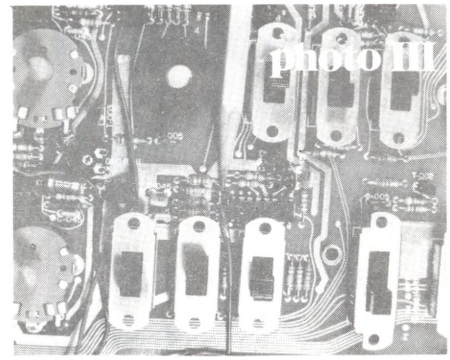












Pro-One PC board resistor installation. We'll be soldering resistors to the Pro-One PC board at those points where the VT outputs connect. This is tricky, since the circuit board layout is quite tight and the traces and pads are small. So, beware of solder bridges. Use a soldering iron with the smallest tip possible. It helps to tin the leads of the resistors and cut them to about 3/8" before installing them. After installing each resistor, tin and trim the lead on the other end, then connect a piece of insulated wire to this end. Insulate the connection with heat shrink tubing or tape, and dress the wire down neatly among the components on the circuit board. Bring all of the wires to a point near the lower left hand corner of the board by the LFO controls, and leave enough wire length to extend off the edge of the board at this point, plus about four extra inches. The accompanying photos should help you out considerably as you perform this modification.

First, install a 100k resistor at the frequency control input for OSC A (see photo I). This point is located at the left hand end of the board near integrated circuit U102, a CEM3340. Just above the chip is a cluster of resistors which all buss together, and connect to pin 15 of U102. These include R121 through R127 and R131 through R133. Note that R134 (located adjacent to U102) is not included in this cluster, so do not make any connection to R134. Solder the new resistor in place, and connect an insulated wire as described above. Use plastic model cement to stick the components and wire to the surface of the board at strategic points for the best mechanical stability.

Next, install a second 100k resistor at the frequency control input for OSC B (see photo II). The circuitry for this VCO is arranged in the same way as the first. The chip is U103, located just below U102, and the resistor cluster we are interested in this time includes R142 through R147 and R155. R156 is not included in this cluster. Install the resistor and wire, being careful not to make a connection to anything that is not part of the summing buss which connects to pin 15 of U103.

Install a third 100k resistor at the LFO frequency control input, as shown in photo III. The chip is U106, and the resistors on the summing buss are R190, R191, and R195, located just to the left of the IC. The connection to the sum pin (pin 15) should be made on the other side of the board, so look closely before soldering.

Select a 4.7k resistor; this solders to the VCA control input. This one is a tight squeeze. The resistor connects to the emitter of transistor Q104 (also labelled T-003), located near the center of the board just about the Glide Rate control. The emitter of the transistor connects to 3.9k resistor R1143. This junction is where we connect our 4.7k resistor. Try to solder it to R1143's lead rather than to the lead of the transistor. You should probably heatsink the transistor lead (see photo IV).

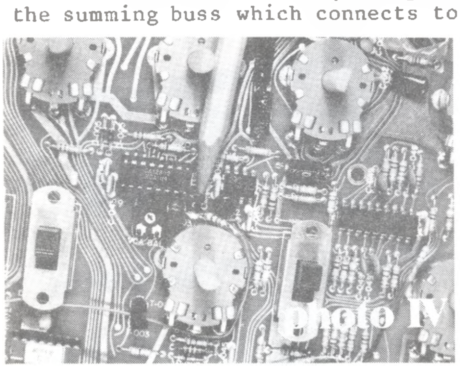
Now install a 68k resistor at the VCF corner frequency control input (see photo V).. This input is at pin 9 of integrated circuit U115, an LM348. The most convenient place to solder to is the summing buss for R1170 through R1173, located just above U115. They buss together near the chip and connect to pin 9 via a foil trace on the top side of the board.

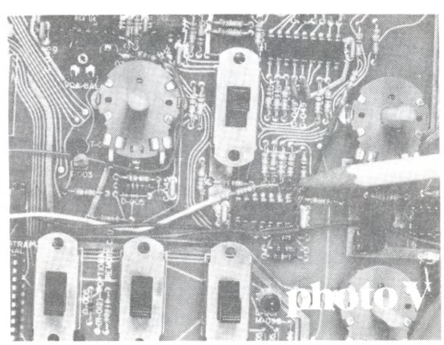
Next, install a 220k resistor at the VCF resonance control input, as shown in photo VI. The VCF chip is a CEM3320 and the resonance control input is pin 9. You can connect the 220k resistor to the lower end of R1157, a 200k resistor located just to the right of the VCF chip. This is another tight squeeze, but the only alternative is the pin on the chip and it's best to avoid putting the iron to that.

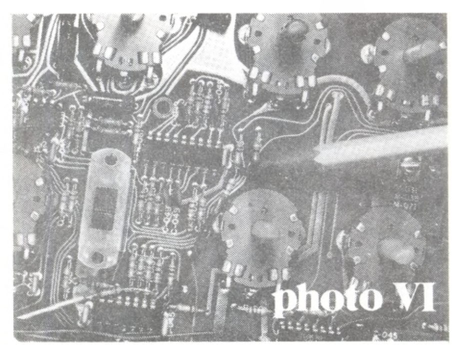
Now we need to connect two more wires, and then we'll be through making connections to the Pro-One PC board.

Connect a wire to the center lug (wiper) of VCA sustain control pot R1204, located in the lower right hand corner of the board, the third pot in the row of five pots near the lower edge. See photo VII.

Connect another wire to the HOT lug of gate







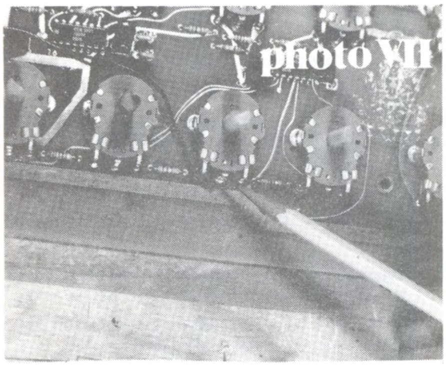


photo VII

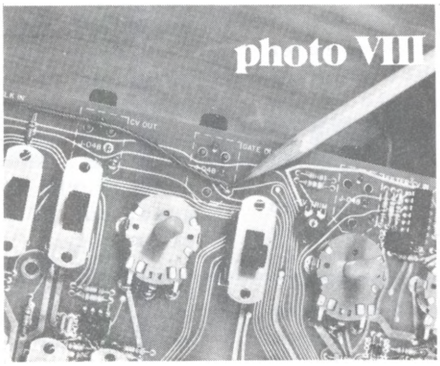


photo VIII

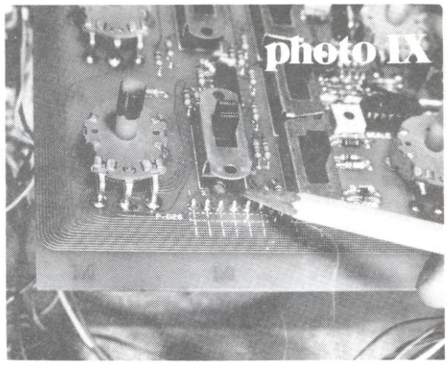


photo IX

output jack J5 (or J-048), as shown in photo VIII. Dress both of these wires down neatly and bring them out at the lower edge of the card, like the previous wires we've installed.

Now we can set the Pro-One PC board aside and prepare the rotary switches.

Rotary switch preparation. Referring back to figure 2, connect a 1k resistor to the common lug of each rotary switch. If you are using double pole switches like the ones shown in the photos, you can use lugs on the unused side of each switch to mount both the other end of the resistor and the wire which connects to it. Note that both sides of S2 are used, and the 1k resistor is mounted slightly differently on this switch.

Using a piece of bare wire or component lead clippings, connect lugs 8 and 9 of S2 together. Solder the connections. In the same manner, connect lugs 10, 11, and 12 of S2 together. Now make the following wiring connections between the three switches:

From	To	To
S1 lug 2	S2 lug 2	S3 lug 2
S1 lug 4	S2 lug 3	S3 lug 4
S1 lug 5	S3 lug 5	

Take the two signal diodes and twist the leads from the anode (unbanded) leads together. Connect these leads to lug 4 of rotary switch S2, and solder them in place. Heat sink the diodes while soldering by attaching an alligator clip to the lead being soldered.

As shown in figure 2, connect the cathode (banded) end of one of the diodes to lug 5 of S2. For convenience in later steps, we'll call this diode D1. Also connect one lead of a 470 Ohm resistor to lug 5 of S2. Solder both leads to the lug. We will label this resistor Rx.

Connect another 470 Ohm resistor to the free end of Rx. Twist the leads together and solder. We'll call this second 470 Ohm resistor Ry, and the junction between Rx and Ry will be connected to ground. Similarly, the junction we are about to make between Ry and diode D2 will be where one of the wires from the Pro-One board will connect.

Note: Diode D1 sends modulation signals to both VCOs when rotary switch S2 is in the VCO 2 position. If you want VCO 2 to be modulated without affecting VCO 1, remove D1.

Take the free end of Ry, and connect it to the still free cathode end of diode D2 by twisting the leads together. You can solder this connection later when you connect the wire to it, instead of subjecting the diode to soldering heat twice.

Connecting the wires. Now we can connect the wires from the Pro-One board to the rotary switches. The first resistor you installed (at the frequency control input of OSC A), is associated with photo I, so we'll call this wire "the wire from point I". The wire from the second resistor installed in photo II we'll call "the wire from point II", and so on. Okay? Here's the wiring schedule:

From	To
Point I	S2, lug 5 (OSC A connection)
Point II	Junction Ry/D2 at S2 (OSC B connection)
Point III	S1, lug 5 (LFO connection)
Point IV	S1, lug 2 (VCA connection)
Point V	S1, lug 4 (VCF Fc connection)
Point VI	S3, lug 2 (VCF resonance connection)
Point VII	S3, lug 1 (VCA-ADSR sustain connection)

Yes, there is one more wire left, but it goes to the gate input on the VT board. We'll connect it later.

Connect the center lug (wiper) of VT VELOCITY output LEVEL control R26, to lug 7 of rotary switch S3.

Connect the center lug of PRESSURE LEVEL control R14 to lug 6 of rotary switch S2.

Final installation. Now it's time to mount the VT board to the Pro-One board, and then we'll install the whole mess back in the Pro-One case top, and complete the wiring.

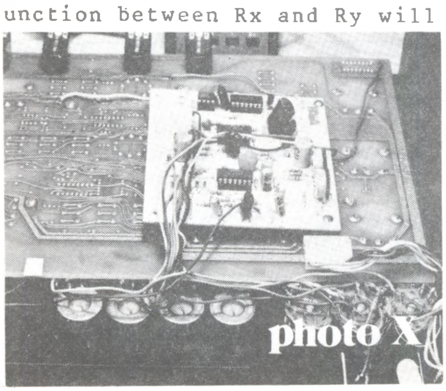


photo X

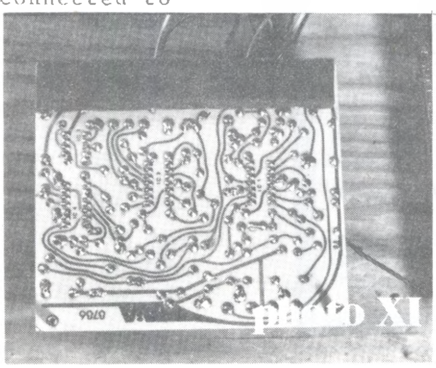


photo XI

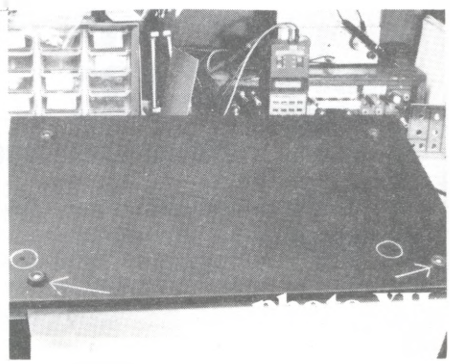


photo XII

VT/PRO-ONE...

First we have to make one of the mounting holes on the VT board a little larger so that it will accommodate a #8 screw. Ream out the hole in the corner of the board near the PAIA logo and the V+ power input to about 5/16" or so. Put the #8 screw in this hole from the component side of the board and slip the 5/16" spacer onto the shaft from the foil side. Crimp the spacer slightly with wire cutters to hold it in place on the screw. Then in the same manner (without the hole reaming), install a #4-40 X 3/4" screw in the mounting hold near the two trimmer pots on the VT board.

The next step must be done very slowly and carefully. Drill a 1/8" hole in the Pro-One as shown in photo IX. This hole must be drilled from the bottom side of the circuit board, at one end of the LFO WHEEL/DIRECT switch on the lower left hand corner of the board, near the front edge. There is a foil trace running across that area on the bottom side of the board; the hole will be immediately adjacent to this trace. Don't let the drill bit or chuck hit the trace! On our model, the bit cut into the phenolic part of the switch just a little, but that's all right as long as you don't go too deep and hit the contacts. The VT board will be mounted as shown in photo X. Hold it in place on the Pro-One board and insert the #8 screw into its mounting hole to make sure you've got the right place to drill for the other (#4-40) mounting screw.

Locate the wire from point VIII (VCF Fc), and connect it to point "G" on the VT board.

Install a piece of foam household insulation tape on the bottom side of the VT board as shown in photo XI, to insulate the unsupported end of the board from the Pro-One board.

Mount the VT circuit board on the Pro-One board as shown in photo X, by pushing the shaft of the smaller, #4-40 screw through the new hole drilled near the LFO switch on the Pro-One board, and installing a #4-40 nut on the shaft from the component side of the Pro-One board. It's kind of hard to grab, but if you are patient things will work out all right.

Make the power supply wiring connections between the two circuit boards as shown in photo X. The pencil in this photo points to the ground connection. Be sure to make nice, neat connections on the Pro-One board so that no shorts occur, and no connections come loose.

Now it's time to install the Pro-One board, with its piggy-backed VT board, into the case top. It will go a little easier if you slip the control pots and rotary switches for the VT into their respective holes first, and then set the board down into place. Also, to make things easier place some kind of blocks or books at each end of the case top to hold it up off the work surface. This allows room for the control shafts to protrude through the holes in the top of the case. Thread the other mounting screw for the VT board (#8) into place first, and then add all the other mounting screws which hold the Pro-One board in place.

Install the transducer in the case bottom, under the front edge of the keyboard frame (installing the transducer is described fully in the VT assembly manual if you encounter trouble). You can try installing it with the keyboard mounted "as is", but for best results drill new holes to mount the rubber feet, and ream out the old holes so the screws can pass freely through them. You may have to add a flat washer to each screw to keep the head from going up through the hole. New mounting locations for the feet are shown in photo XII, and the circled holes are the old ones. Ream these holes out to about 1/4". When the keyboard is installed, the screw in these holes will not be tightened all the way; they will be tightened just enough to pull the keyboard down, and squeeze the foam spacers down at each end. This is detailed in the manual which comes with the VT kit.

Once you've got the keyboard and transducer installed, plug everything in (transducer, keyboard cable, power supply, modulation wheels -- refer back to the disassembly steps), power up the system, and calibrate the VT as outlined in its manual. Once you've buttoned everything up you're back in business, but with one important difference: you have more expressive power than you ever had before! The VT manual tells how to adjust the keyboard sensitivity by adjusting the two front mounting screws.

That's it. Wail on, fellow keyboard players!

An Overview Of NEW AGE MUSIC

By: Don Schwartz

The 70s saw the unheralded emergence of a new genre of music most frequently called "New Age Music". What is this kind of music? To begin with, there is no one kind of "New Age" music. There are many diverse styles, all of which have an underlying theme of mystical spirituality, some being more explicitly spiritual than others. The "spirituality" of New Age music is more universal, rather than being associated with any particular religious or mystical tradition.

New Age music has emerged directly from the need for a music which is central to life, rather than peripheral. Music for healing, relaxation, transformation, stress reduction, meditation, etc. describes the public interest in seeking out new roles for music. This music both expresses and supports a "New Age" philosophy which maintains a positive vision of life as whole and interconnected. Any musical expression from a solo flute to a rock band, from a choir to the sounds of nature may serve. What makes a given music "New Age" is the artist's consciousness and intentions, as well as the music's impact on the listener.

In terms of fully understanding New Age music, note that there is a certain undefinable feeling which is "New Age", much like there are characteristic feelings to rock, reggae, or the blues. Although it's easy to use words like "calm", "peaceful" or "inspiring", the New Age feeling is essentially a subjective experience. The best way to understand this is to listen to the music, and eventually, you will discover that feeling. From that point on you will develop your own inner criteria for discerning what has the "New Age feel" and what doesn't. You are now hooked! Whatever your listening tastes, you will probably incorporate some forms of New Age music because of its abilities to soothe, relax, focus the mind, create an agreeable ambience, and calm the soul.

The roots of New Age music are inextricably linked with those of music itself. Yet, to understand the music's emergence today, we must examine the contemporary social forces which nurtured this development. Music was in the heart of the cultural explosion of the 60s. Thrown by the shattering forces of liberation, the seeds of countless new styles of music flew far and wide. Some flourished a short season, while others took deep root to become part of today's musical landscapes. It is nearly impossible to recreate with words the excitement -- musical and otherwise -- of those times, the rich creativity which startled and amazed, which shocked and dismayed. It is not so difficult, however, to remember what happened afterwards, as the war continued, as the smart hippies went to the country, the corporations, or Canada; as the ignorant ones destroyed themselves, as our culture divided, and as the global problems of our world became horribly clear.

Harnessing the creative energies of the 60s cultural explosion, a new musical production and distribution establishment arose. This system, reflecting the sobriety, fear, and anger in the nation, lacked the openness and vulnerability which had led to the emergence of rock in the 50s, and its explosion in the 60s. High end business consciousness took over, and the categories of commercially acceptable music shrank so that only a few artists and titles would receive the support necessary for meaningful exposure. Desperate business practices brought down the quality of record pressings and tape reproductions. Radio returned to rigid programming formats. No longer would an epic 20 minute piece like Yes' "Close to the Edge" be put on rotation at commercial stations. The hard-life street music of the Eagles and Rolling Stones became preeminent.

Although this reactionary movement squelched the dissemination of new music, it could not destroy it. Technology and talent joined together to give birth to the independent production and distribution system. This system had more successes in production than in distribution, but as we will see later on in this article, it just may be that the development of the New Age music market can serve as a useful distribution model for other types of music.

Religion, psychology, and social consciousness were no less affected by the 60s than music. The "Human Potential" movement, focussing on mobilization of individual potential, grew into a significant social force. The participants in this movement began to cultivate humanitarian values in the context of a universal, transcendental world-view. A few of these people happened to be musicians who choose music's ability to communicate as the primary force for sharing their personal transformations. So, it was in the early 70s that the New Age music movement met the independent production movement.

Of all the musical genres which have developed outside the mainstream of today's music, New Age music began as the most isolated. Unless you happened to be related to one of the musicians, or had some contact with the human potential movement, you simply did not hear this music. It all began with a spiritual in-group scene of listeners searching for those sounds which reflected their feelings, beliefs, and experiences. Three of the most prominent -- and now classic -- early titles where "Tibetan Bells" by Henry Wolff and Nancy Hennings, "Music for Zen Meditation" by Tony Scott, and "Inside" by Paul Horn. It is noteworthy that also in the early 70s, Polyphony's editor, Craig Anderton, did electronic and synthesizer work with classical guitarist Linda Cohen on her two early albums, "Leda" and "Lake of Light", both of which were clearly New Age in their

NEW AGE MUSIC ....

sound and impact. These two albums were ten years ahead of their time, especially in the artful blending of acoustic and electronic instruments.

Keyboardist Steven Halpern is generally acknowledged as the pioneer in the successful marketing of New Age music. Focussing on the relaxing, de-stressing effects of this music, Halpern found a sound appealing to many people searching for an intelligent, graceful alternative to "MOR" easy listening music. He targeted his market accurately and applied good business practices to promotion and distribution, and most importantly, Steven Halpern was willing to travel and perform wherever opportunities presented themselves.

Iasos (pronounced Ya-sos) is another early pioneer who explored a highly etheric, celestial sound. His *Crystal*White*Fire*Light* is among the most sensational pieces of music I have ever heard. In performance, Iasos presents a visual experience I've yet to see excelled. Two other significant artists are Georgia Kelly and Joel Andrews, both harpists.

As the New Age movement and independent music production system grew, so did New Age music. Interestingly, and somewhat ironically, it was Brian Eno's "Ambient One: Music for Airports" which introduced a wider listening audience to an appealing sound carrying a "New Age" feel. A highly significant yet unacknowledged breakthrough occurred with the release of the film "Chariots of Fire", and Vangelis' soundtrack of that film. In the vernacular of the entertainment industry, both film and album were "monsters". This was quite unexpected. The listening public, at last, presented uncontestable evidence that a New Age sound, when composed and exposed well, can be commercially successful. Vangelis' breakthrough also points to the potential that film holds in contextualizing and exposing New Age music. While radio continues to be either rigid, reactionary, or chaotic, film remains a more open medium.

"Chariots of Fire" is not the only evidence of a New Age market. Many popular artists create music which represents positive feeling-tones, and listener impact, characteristic of New Age music; the list includes Chick Corea, Jean Luc Ponty, Yes, Stevie Wonder, Todd Rundgren, Dan Fogelberg, George Harrison, John McLaughlin, Jon Anderson, and many others. George Harrison's "My Sweet Lord" was the fifteenth most popular song for the decade of the seventies.

Today, a clear New Age music territory is discernible. The scene is now growing at a rapid pace, with nearly 500 titles and several new releases every month. There are about a half-dozen regional distributors, and another two dozen smaller ones. Everybody sells retail -- artist, record company, and distributor. Artists sell their tapes and records at performances, growth centers, seminars, conferences, and the like. The primary retail outlets have been health food stores, spiritual bookstores, and growth centers. This is now changing to include actual record stores. On both coasts you can find "New Age", "Meditation", or "Electronic" music sections in major record outlets. Typical artists found in these sections include Kitaro, Deuter, Eno, Vangelis, Steven Halpern, Iasos, and Schawkie Roth. Over the last few years, there has also been an increasing employment of electronic and synthesizer sounds. At this point, as a matter of fact, these instruments are dominating the sound. The future of New Age music will most likely involve various blendings of acoustic, vocal, electronic, and natural sounds.

In addition to the emergence of New Age music into the traditional music market, it is receiving more and more airplay. New Age music shows are popping up in listener-sponsored and college radio stations across the country. The longest running program of this type is "Music from the Hearts of Space", produced and broadcast throughout Northern California by Stephen Hill and Anna Turner. This program will likely be the first of its kind to receive national syndication.

New Age music represents not just a new type of music, but a way of looking at music as an integral part of life, rather than simple entertainment or diversion. It will be interesting to see how this new form of music progresses in the months and years ahead.

(Editor's note: Two years ago Don Schwartz began The New Age Music Network to bring cohesiveness and cooperation to this scattered musical movement. The Network has placed every artist, record company, distributor, related business, and Network member on a computer diskette. This information is available to subscribers of what is now called The Music Network, currently administered by Augie Blume. For more information, write: The Music Network, PO Box 190, San Anselmo, CA 94960; or call 415-457-0215.)

REVIEW MXR Dual Limiter

By: Craig Anderton

One of the most useful signal processors for any recording studio is a limiter. It helps you control dynamics, create sustain with percussive instruments, bring out harmonics and overtones, simplify level-setting, and so on. When it comes to choosing a limiter, however, the options can be bewildering. Not only are there lots of available limiters, but there is a certain mystique about a limiter's sound; some engineers will only use limiter A on vocals, limiter B on drums, limiter C on guitars, and so forth. The reason for this is that limiters often affect a signal in a relatively obvious manner. Thus, differences between units tend to show up as audible differences on tape.

Recently I've had the opportunity to use the MXR Dual Limiter in my studio, and overall, I think it is probably one of the best "general purpose" limiters available. The cost is quite reasonable ($450 list; the lowest price I've seen on it is from Dickstein Distributing, who sells it for $338). It is remarkably non-critical to use, and includes a number of convenience features.

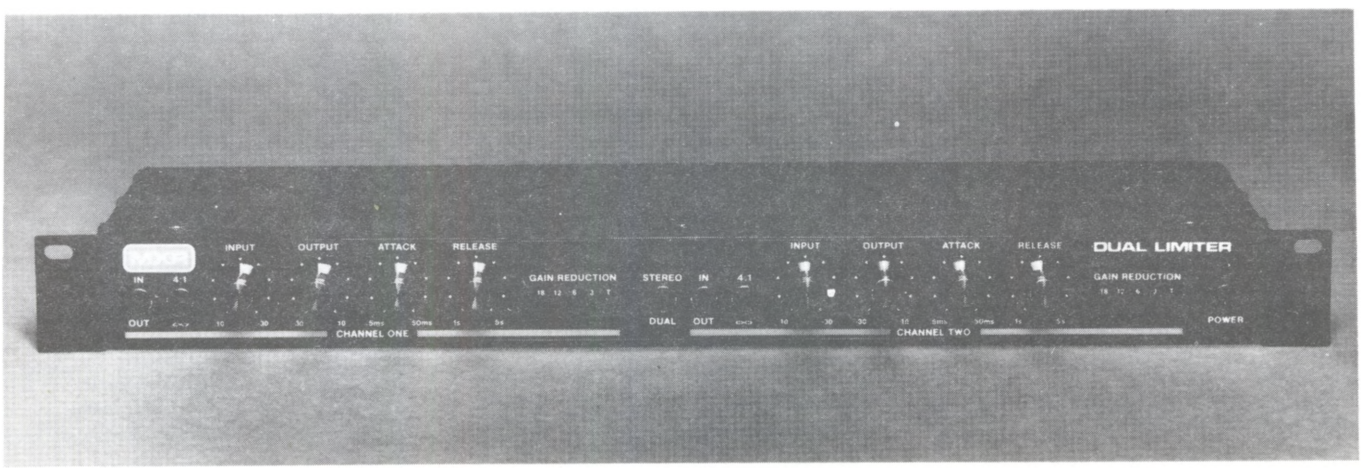
General description. The Dual Limiter (DL for short) includes two limiters in a single-space rack mount package. Each limiter may operate independently, or both limiters may be ganged together for stereo applications (as chosen by a front panel pushbutton switch). When ganged, the two channels work in a related manner to avoid shifting of the stereo image.

There are two pushbuttons for each channel: limiter in/out (bypass), and infinite limiting/4:1 limiting. When the latter switch is in the infinite position, a signal is clamped to a particular limit point which will not be exceeded no matter how much the input signal exceeds this limit point. In the 4:1 position, the slope is gentler; rather than a tight clamping action, a 4 dB change in input level creates a 1 dB change in output level. This position is useful when you want to restrict dynamics in a less drastic fashion.

There are four controls for each section: input, output, attack, and release. The input control sets the minimum level of signal required for limiting. When fully clockwise, limiting occurs with signals above +10 dB, and when fully counterclockwise, limiting occurs with signals above -30 dB. The output control adjusts the overall output of the unit. When fully clockwise, a limited signal will hit +10 dB, and when fully counterclockwise, the limited signal's level will be -30 dB. Note that the above is true only when the infinite limit position is selected. In the 4:1 position, the output levels vary in a manner related to the amount of limiting.

The attack control determines how long it takes the limiter to react to changes in input level. Interestingly, the DL includes internal circuitry which fine tunes the setting selected by this control. For example, a rapid change in input signal level shortens the attack time -- exactly the effect you want. Thus, if the attack control is set fully counterclockwise (shortest attack time), the attack time is 500 microseconds for large changes in level and 10 ms for small level changes. This helps greatly towards achieving a natural sound. When fully clockwise, the attack time is about 50 ms for any change in level.

As it says in the manual, "The release control varies the time it takes for the VCA gain to recover towards unity gain in order to respond to further changes in input signal level." When fully counterclockwise, the release time is 100 ms; when fully clockwise, the release time is 5 seconds.



There is also a five LED display for each channel showing the amount of gain reduction. The more the signal is being limited, the greater the number of LEDs lit. This display is operative whether or not the limiter is bypassed. This is a nice touch, as it lets you get a rough control setting before punching the limiter into the circuit. All LEDs dim when maximum limiting is being approached (essentially an overload condition which requires turning down the input level control).

Re jacks on the back, there are balanced 1/4" phone and XLR input connectors for both channel inputs (the input impedance is about 20k). The 1/4" phone jack may be used with unbalanced lines simply by plugging a standard mono cord into this input; when used with a stereo cord, it becomes balanced. The outputs have unbalanced 1/4" phone and XLR connectors, however, when the unit is bypassed the XLR connectors retain their balanced characteristic. The output impedance is about 200 Ohms.

There are also two jacks (labelled "detector loop") which let you insert signal processors in the control path which affects the VCAs in each limiter channel. These option lets you do some neat tricks. For example, if you patch in a bandpass filter designed to boost high frequencies in the range of about 2 kHz to 10 kHz, the higher frequencies will go into limiting before the lower frequencies. This works well for de-essing vocals.

On the other hand, many musicians complain of a less "bright" sound when a signal is being limited. By inserting a low pass filter in these jacks, the higher frequencies are less limited than lower frequencies, which produces a brighter sound.

You can even use these jacks to give tremolo type effects by patching a 0 to +10V triangle wave LFO output into the detector loop input jack. While this jack appears to be AC coupled, and therefore will not react like a "real" DC coupled VCA or work with slow LFO speeds, you can still obtain a nice pulsating effect (an attenuator might be needed in some cases after the LFO). For a synchro-sonic tremolo effect, I tried plugging the output of a drum track into the detector loop input while a guitar was going through the limiter itself. The result was a guitar sound which pulsated in sync with the drums.

Here's another interesting patch which involved plugging a guitar into the limiter. By inserting a high pass filter (controlled by an inverted envelope follower tapping off the guitar) into the detector loop, I was able to obtain "backwards tape" effects. This occurred because upon first striking the guitar, the cutoff frequency of the filter was low and therefore passed all of the signal through to the limiter's VCA, thus giving full limiting. As the guitar signal decayed, the filter cutoff frequency would move upward, thus presenting less and signal to the VCA and producing less limiting. Eventually, the guitar signal passed at full strength.

I even tried patching the second limiter into the first limiter's detector loop. This appeared to really "squash" the dynamic range of the signal -- try it with cymbals when you want a real sustained cymbal sound.

You can also use these jacks to obtain limiting slopes of 1.6:1 and 2:1. Simply connect the audio output from a limiter into its associated detector loop input jack. When infinite slope is selected on the front panel, the slope becomes 2:1. With the 4:1 slope selected, the new slope is now 1.6:1.

There's one other nice feature worth mentioning: a built-in mute circuit shuts off the output briefly upon turn-on/turn-off, or if AC power is interrupted. Thus, if someone kicks the cord out of its socket by accident, your speakers will not become sacrificial victims in the process.

Those who are familiar with my style of reviewing are probably expecting me to mention those things which I didn't like, and possible improvements which could be made. Well, there really weren't any aspects of the DL's operation I didn't like. As for improvements, about the only thing I would like to see is a patch point which lets you access the VCA and which conforms to the industry standard 0 to +10V control range. That way, by building some special purpose circuits to stick in the detector loop, you could make the DL do other tricks (such as noise gating). And of course, having two extra VCAs around would be helpful even if you weren't using the DL for limiting.

Technical description. The DL VCAs are pulse-width modulated at a very high frequency (250 kHz); to obtain limiting effects, the pulse width varies with respect to the input signal and therefore controls the gain. Note that like an analog delay line, having these high clock frequencies requires input filters to prevent aliasing and output filters to prevent clock feedthrough. I was initially skeptical about how much this approach affected the sound, but the sound quality is certainly just fine. And while I tried to get it to alias by feeding through signals with lots of harmonics, I couldn't get the unit to misbehave. The four pole output filters also must be doing their job, as I couldn't detect any residual clock frequencies at the output. The rest of the circuitry is relatively straightforward, and is explained more fully in the brief but exceptionally readable and useful instruction booklet.

Overall evaluation. The MXR Dual Limiter is a fine, effective product which is amazingly simple to use. I haven't seen many reviews on it, probably because limiters tend to be pretty ho-hum and it's easy to think that this is "just another limiter". But it seems to me that MXR did its homework, and rather than repeating the standard limiter design (whose heritage lies in broadcasting applications), came up with a limiter designed from the ground up for musical applications in smaller studios. If you've been looking for an inexpensive, general purpose limiter with excellent performance and some thoughtful convenience features, look into the MXR Dual Limiter.

CURRENT EVENTS

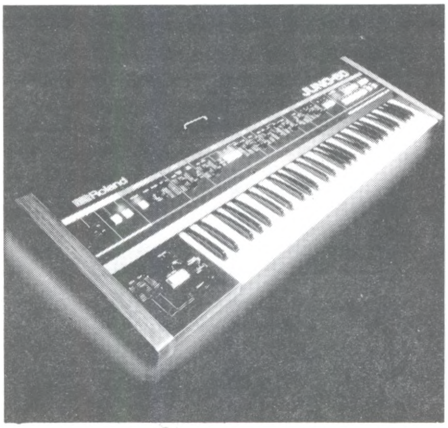
'Tell Them You Saw It In Polyphony'

New Keyboards. Yamaha (PO Box 6600, Buena Park, CA 90622) has announced the CP35 electronic piano with velocity sensing on all 73 notes. The CP35 also includes a built-in flanger, stereo tremolo effect, and equalizer. $1995 list.

Even more interesting is Yamaha's MP-1 Mini-Printer, a portable keyboard (it weighs less than 4 lbs.) with a computerized music printout system. After practicing a song or composing a new melody, you simply use the "Easy Print" function to obtain a refined printed score. The MP-1 uses a miniature ballpoint pen stylus to print out melody lines on a 2.5" wide paper roll, and may be powered from household current, batteries, or an automobile cigarette lighter. List price is under $1000.

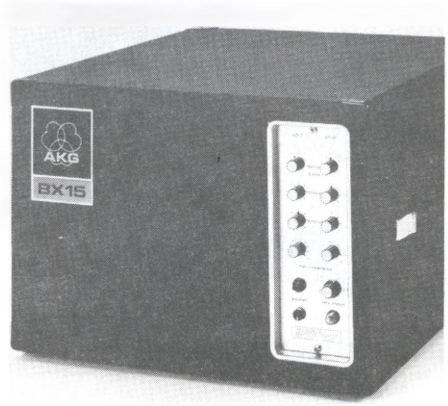
Korg (89 Frost St., Westbury, NY 11590) has announced the BPX-3 bass pedals, featuring a 13 note pedalboard, 4 preset bass voices, bass/treble EQ, and attack/sustain controls. List price is $595.

Roland (1022 South La Cienega Blvd., Los Angeles, CA 90035) continues its onslaught of press releases with information on the EP-6060 combo piano. This 18 lb. instrument includes two mixable/-detunable voices, octave and decay controls, 6 band equalizer, transposer, and arpeggiator. $895 list.



The JUNO-60 is a programmable poly synth with 5 octave keyboard, 56 programs organized as 7 banks of 8 programs, arpeggiator, transpose, chorus, etc. There's also a Digital Communication Bus designed to interface to the Roland MC-4 microcomposer, digital sequencers, and similar devices for programmed performances. $1795 list.

Also from Roland: The OP-8 Interface ($750 list) which lets you link together the JP-8 and MC-4 Microcomposer. An OC-8 Interface Modification for the JP-8 ($350 list) is also required. Music Technology Inc. (105 Fifth Avenue, Garden City Park, NY 11040) is shipping the Synergy, an all digital synthesizer with programmable keyboard response and built in 4 track recorder with overdub capabilities. Audio learning courses, text manuals, and video software are all available for the instrument.



New Reverb. AKG (77 Selleck St., Stamford, CT 06902) has introduced the BX-15E portable reverb unit. It features independent decay time selection (1.5 top 3.5 seconds), high and low freq EQ, and reverb/dry mixing for each of the two channels. Claimed to be impervious to acoustic feedback and mechanical vibration, the BX-15E does not require a special base or separate room for installation and weighs in at 47 lbs.

More Yamaha News. Yamaha now offers several products for the commercial sound market, including mixers, stage monitor speaker systems, and commercial power amplifiers.

Syntheswiss. The Swiss Society for Computer Music (Sommerau, CH-8618 Oetwil Am See, Switzerland) is looking to contact all persons and institutions interested in the development of computer technology in all styles of music.

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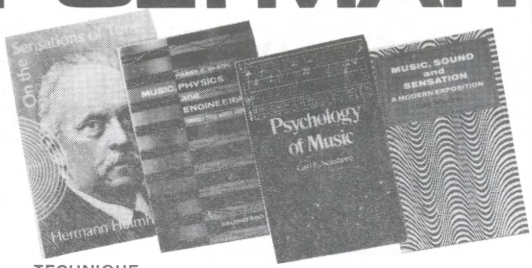
POLYMART



SCIENCE

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.

#SENS	ON THE SENSATIONS OF TONE	$8.95
#MPE	MUSIC PHYSICS AND ENGINEERING	$6.50
#PSYCH	PHYCHOLOGY OF MUSIC	$6.00
#MSS	MUSIC, SOUND AND SENSATION	$4.50

TECHNIQUE

Synthesists must be well versed in a number of techniques and principles. "How To" and project oriented books are a great way to pick up these skills. **MULTITRACK PRIMER** by TEAC is a step-by-step guide to building, outfitting and operating your home studio. The Byte Book of **COMPUTER MUSIC** describes computer control of electro-mechanical instruments, Fourier analysis, circuits and loads of software. **HOME RECORDING FOR MUSICIANS** is Craig Anderton's original guide to outfitting and operating a budget studio for maximum results, includes mixer and other audio processing circuits and a sound sheet demo recording.

#TEAC	TEAC MULTITRACK PRIMER	$4.95
#BYTE	BYTE BOOK OF COMPUTER MUSIC	$10.00
#HRFM	HOME RECORDING FOR MUSICIANS	$11.95

ELECTRONICS

Electronic Cookbooks are a great way to stock your library with materials that are not only heavy on theory, definitions and educational material but chock full of practical applications as well. These books can easily replace stacks of manufacturers data sheets and applications notes all in an easy to use reference. Walt Jung's **OP-AMP** and Don Lancaster's **ACTIVE FILTER** Cookbooks are self-explanatory — required reading for synthesists! **AUDIO OP AMP APPLICATIONS** is an edited version of the Op Amp Cookbook by Walter Jung, containing only audio applications. Lancaster's **CMOS** book is much more than a digital reference — phase lock loops, top octave generators, touch switches, and other things you need. **ELECTRONIC PROJECTS 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.

#OACB	OP-AMP COOKBOOK	$15.95
#AFCB	ACTIVE FILTER COOKBOOK	$14.95
#AUOA	AUDIO OP-AMP APPLICATIONS	$8.95
#CMCB	CMOS COOKBOOK	$12.95
#EPFM	ELECTRONIC PROJECTS FOR MUSICIANS	$14.95
#EMCR	ELECTRONIC MUSIC CIRCUITS	$16.95

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. **AUDIO CYCLOPEDIA** has 1760 pages with 3650 entries and hundreds of dawings and schematics to answer any question about ratio. Hardbound. **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.

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

"HOW TO" GUIDES

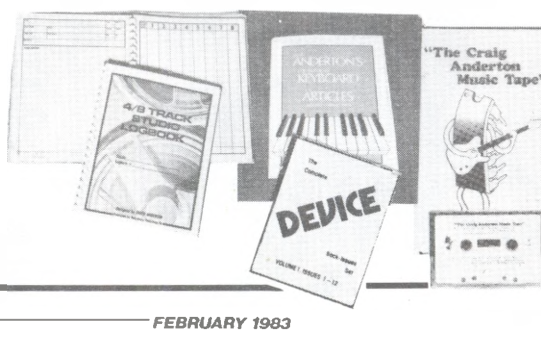
HOW TO BUILD A SMALL BUDGET RECORDING STUDIO FROM SCRATCH by F. Alton Everest covers twelve tested designs. **HOW TO DESIGN TEST AND BUILD COMPLETE SPEAKER SYSTEMS** BY David Weems is a do-it-yourself guide for the ultimate in sound quality. **HOW TO MAKE AND SELL YOUR OWN RECORD** — With the major lables having severe economic problems, many insiders feel that the future of the music industry may be in independent record production. Learn the ropes with Diane Sward Rapaport's indispensable handbook of how to get started. **MAKING MONEY MAKING MUSIC** by James Dearing — Everyong dreams of being at the top, but there's an enormous amount of "middle money" out there for the taking. This is not a book about how to become a Millionaire Rock Star, but the strategies revealed will give you the knowledge you need to keep afloat if you decide to pursue a recording contract. A fresh and practical approach to staying alive in the music business. From the publishers of Writer's Digest.

#BRS	BUDGET RECORDING STUIDO	$9.95
#CSS	COMPLETE SPEAKER SYSTEMS	$7.95
#MASR	MAKE & SELL YOUR OWN RECORD	$11.95
#MMM	MAKING MONEY MAKING MUSIC	$12.95

& MORE FROM POLYPHONY

4/8TRACK STUDIO LOG BOOK designed by Craig Anderton provides a place to keep all the important information on your tape library. Log in timing, type of tape used, record patches, make notes and use the expanded track sheet to list sequential changes in tape tracks relating to the settings of the indes counter. Craig Anderton's **CONTEMPORARY KEYBOARD ARTICLES** is a collected reprint of all the articles from June 1977 through February 1981, covers tips, technique, theory, maintenance, and numerous construction projects. **DEVICE BACK ISSUES** — during the year that this newsletter was published, it featured almost 200 pages of technical information for the guitarist/musician. A wealth of articles on design, product reviews, and modification and construction projects. Sold in complete set, individual issues not available. Limited number available. **CRAIG ANDERTON MUSIC TAPE** — Delightful listening plus a booklet explaining how the effects were achieved.

#SLB	STUDIO LOG BOOK	$4.95
#AA	CRAIG ANDERTON'S CONTEMPORARY KEYBOARD ARTICLES	$5.95
#DEVICE	COMPLETE SET (12) DEVICE BACK ISSUES	$18.00
#CAMT	CRAIG ANDERTON MUSIC TAPE	$5.95



BACK ISSUES

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

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

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

☐ #0302: 11//7: The Sensuous Envelope Follower, digital gates, LED wall art, build a bionic sax, data to music peripheral project, Apple II as a music controller, using the NE566 as a VCO, patches.

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

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

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

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

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

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

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

☐ #0601: May/June 80: Gary Numan, Microcomputers in Real Time Audio, Build a Digital Audio Delay Line, writing Documentation, Richard Hayman Composer/Performer Home Recording: Applying Harmonizing and Pitch Transposing Techniques by: Craig Anderton.

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

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

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

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

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

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

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

☐ #0703: Nov./Dec.'81: Dave Rossum interview, Applied Synthesis: Strings,Details: Series-parallel/Sum-Difference. The Sound Gizmo and Pro-One Reviews, Practical Circuitry: VCO Deluxe.

☐ #0704 Jan./Feb.'82: Bob Moog interview, Chip Power - STK-050/070, Simple Square Wave Shaper, Tape Timer Ruler, Practical Circuitry: VCAs made simple, Details: Gozinda & Gozouta Revisited, Korg Trident & Casiotone 202 Reviews.

☐ #0705 Mar./Apr.'82: Electronic Music Math, Analog Delay Clock / Modulation; Frequency Domain Modifiers; Screen-Wave for the TRS-80; Touch Switches Revisited; Practical Circuitry: ADSR the Easy Way; Getting the most out of a Cheapo (Guitar).

☐ #0706 May/August '82: Anatomy of a Private record, Don Slepian Interview, Understanding Digital Synthesizers: A Digital Filter, Syn-Bow Review, Optical Audio, Profiles of SSM 2033 & 2044, The PAL Filter, Bill Rhodes Applied synthesis: Bells, Pipe Organ, Harpsichord, Electronic piano; The Realistic MG-1 Reviewed.

☐ #0801 Sept/Oct.'82: Ambience in Electronic Music,Tone Bypass for Fender Amps, 8 Track Reviews, Parametric EQ Tips, Solo/Cut Circuit for TASCAM Model 3, The SSM 2011, Tube Preamp, Snare + Drum Voice Circuit, Triple Pick-up Switcher, Simulated Stereo, When Quality Reocrd Mfg. Counts, Independent Record Mfg. Convention report.

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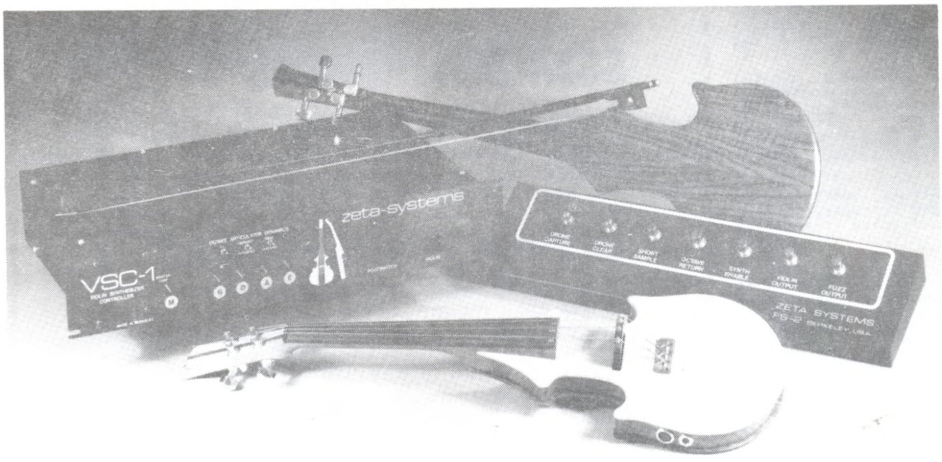
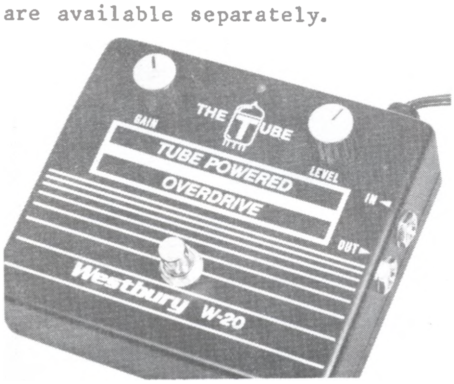


Electronic Violin. Zeta Systems (1122 University Ave., Berkeley, CA 94702) has introduced the VSC 1 Violin Synthesizer Controller. Four independent pickups mount in the custom built solid body violin (one pickup per string) to produce a true quad signal which is free of IM distortion. The VCS 1 is not a P/V converter; it employs four laser trimmed position transducers mounted in the fingerboard which, in conjunction with the pickups, generate immediate and accurate gates, triggers, and pitch/amplitude control voltages for each string. The VSC 1 is designed to work in conjunction with existing polyphonic synthesizers. Full system retail is $5,000; violins are available separately.



Tube Fuzz. Unicord (89 Frost St., Westbury, NY 11590) has announced The "Tube", an overdrive circuit based on a 12AX7 tube instead of solid state circuitry.

New Multiple FX Unit. MXR (740 Driving Park Road, Rochester, NY 14613) has introduced the "Omni". This 3.5" high rack mount unit includes compression, fuzz, EQ, flanging, echo, and an FX loop. Also included is a remote footswitch. $725 list.

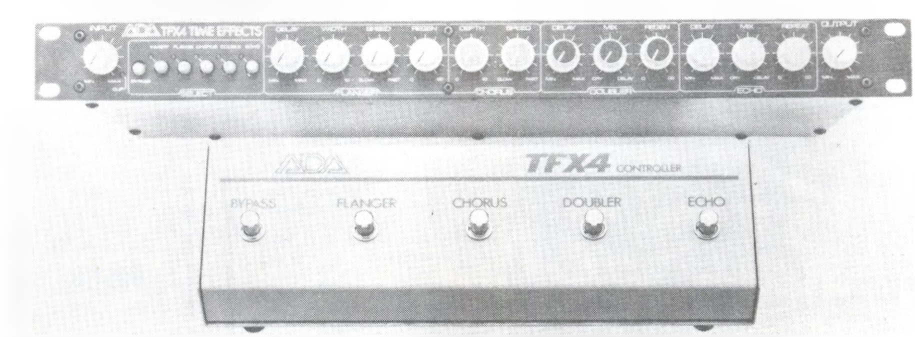
Music Game. Notable Software (PO Box 1556, Dept. PY, Philadelphia, PA 19105) announces "Note Trespassing", a musical game for the Apple II. Hi-res graphics display notes which move across the musical staff. The object is to eliminate all the notes, by matching them with their letter names (creating its pitch through the Apple speaker), before they can reach the end of the row. Options include choice of clefs, three levels of difficulty, and muted response. $25 list.

News from National. National Semiconductor (2900 Semiconductor Drive, Santa Clara, CA 95051) has introduced the LP365, a quad comparator which improves on the popular LM339. The LP365 features programmable current consumption, as well as complete isolation from the analog power supply since the output transistors have open collectors, and their emitters tie to a common "digital ground" pin. This allows for easy interfacing between analog and digital circuits, as well as keeping digital noise out of the analog circuitry.

Also new from National is their 1982 Linear Data Book with over 2000 pages. For a copy, send a check for $12.00 (US price only; Californians add sales tax), payable to NS Publications, to the address given above.

Synchro-sonic Syntauri. Syntauri (3506 Waverly St., Palo Alto, CA 94306) now offers sync to tape and drum interfacing capabilities. The drum interface supports the Linn, Oberheim DMX, Roland TR-808, and Roland TR-606 drum units. The sync to tape option lets each stereo track recorded on the synthesizer to be dumped to its own track on a multi-track tape deck, permitting per-track post processing and full range of mixing/remixing options.

Multi-purpose Delay. A/DA (2316 Fourth St., Berkeley, CA 94710) has announced the TFX4 Time Effects delay unit which includes four groups of controls covering flanging, chorusing, doubling, and echo functions. Each group may be individually preset, and switched in via front panel switches or footswitches. Thus, with one push of a switch the same unit switches from, say, flanging to doubling. Additional controls include phase inversion and input/output level controls. List price is $499.95.



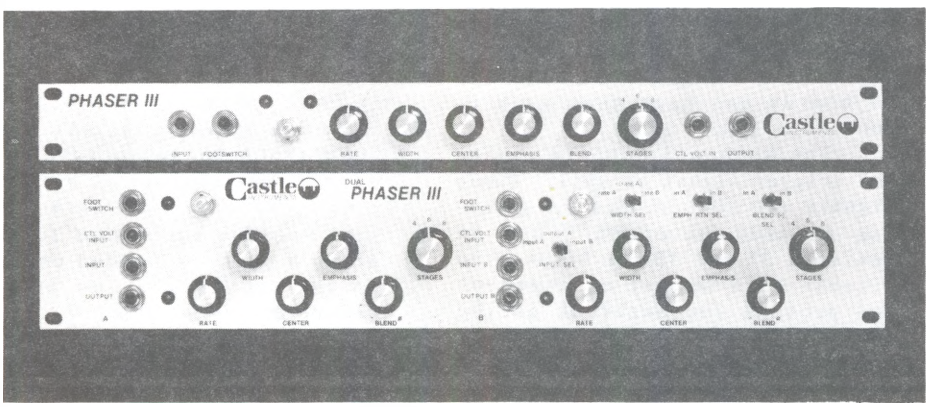
New EQs. Teac (7733 Telegraph Road, Montebello, CA 90640) has introduced the PE-40 four channel, four band per channel parametric EQ. In addition to the parametric EQ, each channel has three pushbutton selectable filters (two high pass and one low pass) along with a bypass switch. Claimed specs are 83 dB S/N with under 0.15% THD.

Furman Sound (30 Rich St., Greenbrae, CA 94904) has introduced the SG-10 sweep graphic equalizer. Unlike other graphics, the SG-10's center frequencies can be adjusted over a four octave range; it may also work as a 10 band mono graphic or 5 band stereo graphic. Other features include center detent on all slide pots, overload indicator for each channel, and balanced/unbalanced inputs and outputs. List price is $495.

Phasing In. Castle Instruments (2 Carteret Ct., Madison, NJ 07940) has introduced the Phaser III in rack mount and dual rack mount configurations. Both feature switch selectable 4/6/8 stages, optional control voltage inputs, FET switching, and balanced inputs/outputs. The dual rack mount model incorporates two phasers in one package, with four crosslinking switches which interconnect the LFOs, feedback paths, and inputs and outputs of the two phasers. Options include

Surplus Parts. If you're into strange surplus bargains, telephones, video, home security, electronic odds and ends, and a catalog displaying a healthy level of lunacy, write to ETCO Electronics (Dept. 554, Box 796, Plattsburgh, NY 12901) for their listings.

New IC Makes Waves. The CY360 Intelligent Waveform Synthesizer from Cybernetic Music Systems (445-203 So. San Antonio Rd., Los Altos, CA 94022) is a 40 pin LSI device which provides flexible generation of sine, triangle, and square waves, as well as computed waveforms and user defined functions. The initiation and termination of these waves can be under external control, or under control of a program stored in the CY360. It accepts binary commands from a keyboard or computer through a parallel TTL interface which also enables simply RS-232 and IEE-488 interface designs. The CY360 drives standard 8 bit D/A converters and includes a 16 bit mode of operation. The cost is $75 in hundreds.



PATCH-OVER SCHEME FOR A SMALL SYNTHESIZER SYSTEM

By: Thomas Henry

For the past year, this column has been very microcosmic in nature. We have talked about many individual circuits, techniques, and modules. For the first time we are going to go macrocosmic and describe a complete structure or system. In this installment, we will cover a master patch-over hardware system for a small synthesizer.

First off, what is patch-over hardware and what is it good for? A patch-over scheme is a means whereby one can have his or her synthesizer pre-patched or "normalized" for some standard arrangement of circuits. For example, many synthesizer voices start with a VCO which is fed to a VCF and finally goes to a VCA. In a patch-over system, this arrangement would be available automatically to the user without the use of any patch cords. What sets this arrangement apart from a strictly normalized scheme, however, is the ability to override the internal signal routing with outboard patch cords. A patch-over arrangement allows you to have the best of both worlds. Most of the time you won't have to use any cords to arrive at your final patch. But for those times when you are coming up with an "unusual" arrangement, you may override the internal patching and achieve any result available to a studio type synthesizer.

Patch-over schemes are most suited for smaller (non-studio type) synthesizers. This is especially true if you play in a band and expect to take the critter on the road with you.

I wanted my first synthesizer to be small and portable. Since it was a DIY project (naturally), I was free to choose any arrangement I wanted. I decided to build it in a standard rack mount enclosure and follow the Electronotes standards for voltages, impedances, etc. By doing so I ended up with a portable system which was not only very usable on stage, but was suitable for use in a home recording studio as well. And it ended up being compatible with all sorts of other stuff including an E-mu synthesizer, a Gentle Electric Pitch Follower, tape recorders, and so on.

To get our bearings, let me describe the modules in this system. For sound sources I had two VCOs and one noise source. Then there was one lowpass VCF, one VCA, one ADSR, one AD (attack-decay), one keyboard and interface, one sample-and-hold, one LFO, and the power supply. The whole synthesizer (less the keyboard) fit in a rack case 19" wide by 28" high.

Figure 1 shows the patch-over scheme I used. Consider the audio trail first. The basic arrangement is VCO to VCF to VCA. But notice that this trail can be broken, if desired, by inserting plugs into J10 or J8, allowing you to add additional audio processing if desired.

Since several of the modules are controlled by the keyboard, there are a number of 1V/octave lines as well. Both VCOs are controlled by this voltage, as is the VCF. It is important to have this sort of input on the VCF to allow "tracking", thus providing a consistent waveform from the VCF output over the VCO's entire range. Note, however, that J4, J5, and J11 allow you to override the 1V/octave inputs. This is especially handy if you have two keyboards and want them each to control a VCO.

The keyboard outputs a gate and trigger signal as well. These signals are both needed to generate an ADSR waveform. But if desired, J1 and J2 can be used to disable this normal arrangement and provide for external gating and triggering of the ADSR. I commonly use this arrangement to allow the LFO to trigger the ADSR for repeating envelope effects.

The AD unit also requires a trigger signal. As in the case of the ADSR, this trigger input may be overridden via J3.

The envelopes are commonly used to modulate the VCF and VCA. Note that a switch is associated with both of these modules (S2 and S1, respectively). These switches allow you to choose which envelope generator will control which module. If you don't want any envelope control, J7 and J6 allow you to substitute some other signal.

This, then, is the patch-over scheme I elected to use in my first system. There are a few more tricks we can employ, and we'll discuss that in a moment. First, however, I should say that figure 1 only shows the controls and inputs on the synthesizer which are affected by the patch-over scheme. Of course, the synthesizer has many more inputs and outputs, but they weren't shown on the diagram to keep things simple.

After giving this arrangement much thought, I decided that the other inputs and outputs were less "standard", and no single patch-over scheme seemed appropriate for them. As it turned out, with the system described in figure 1, I could do just about any patch I wanted with under 10 cords!

Perhaps one of the smartest things I did was to add a patch-over mixer to one of the VCOs. Figure 2 shows the arrangement. The VCO I built was the predecessor to the "VCO Deluxe" described in this column some months back. It had many outputs, all the way from sines to sub-octave square waves. Having lots of outputs is great, but the drawback is the increase of cords needed to accomplish a patch. I added the mixer to the VCO module, as shown in figure 2.

Without any patch cords inserted, the mixer allows you to

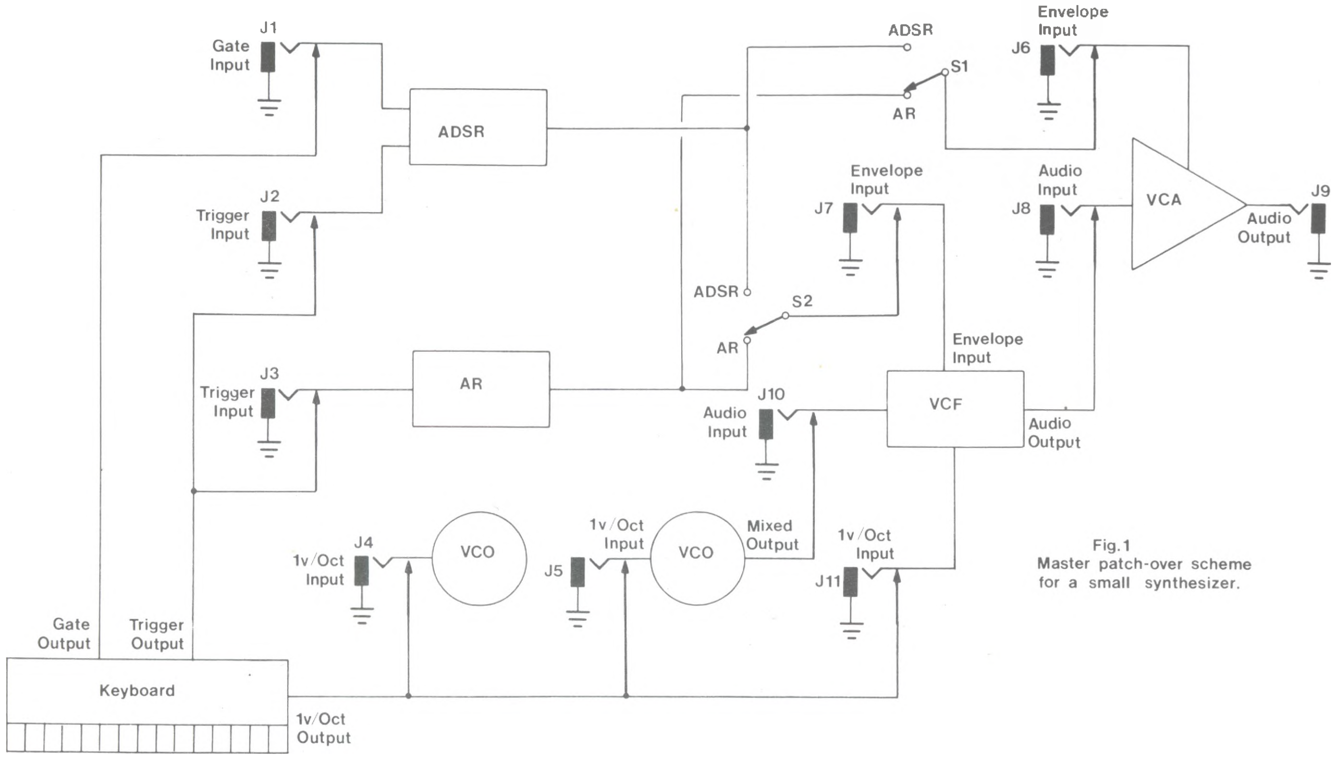


Fig. 1
Master patch-over scheme for a small synthesizer.

combine the VCO outputs to a single mixed output, which can then go on to the VCF. However, suppose you want to use the mixer for something else (like mixing some control signals). You can do that too! Simply insert plugs into the inputs (J1 and J2 in the figure), and away you go. Note that the VCO outputs are still available from the straight output jacks (J3 and J4 in the figure). My VCO had five outputs, but I went ahead and made the mixer a six-in, one-out arrangement and kept one of the inputs "uncommitted".

Figure 2 shows the mixer as being audio in nature, but actually I designed the thing to be DC coupled so that I could mix control signals as well. However, experience has shown that I tend to use the mixer with the VCOs for audio mixing more often than not.

Figure 3 shows the finishing touches for the master patch-over scheme. This is a very simple arrangement, so not much need be said. Essentially, when using the sample-and-hold, I found that I

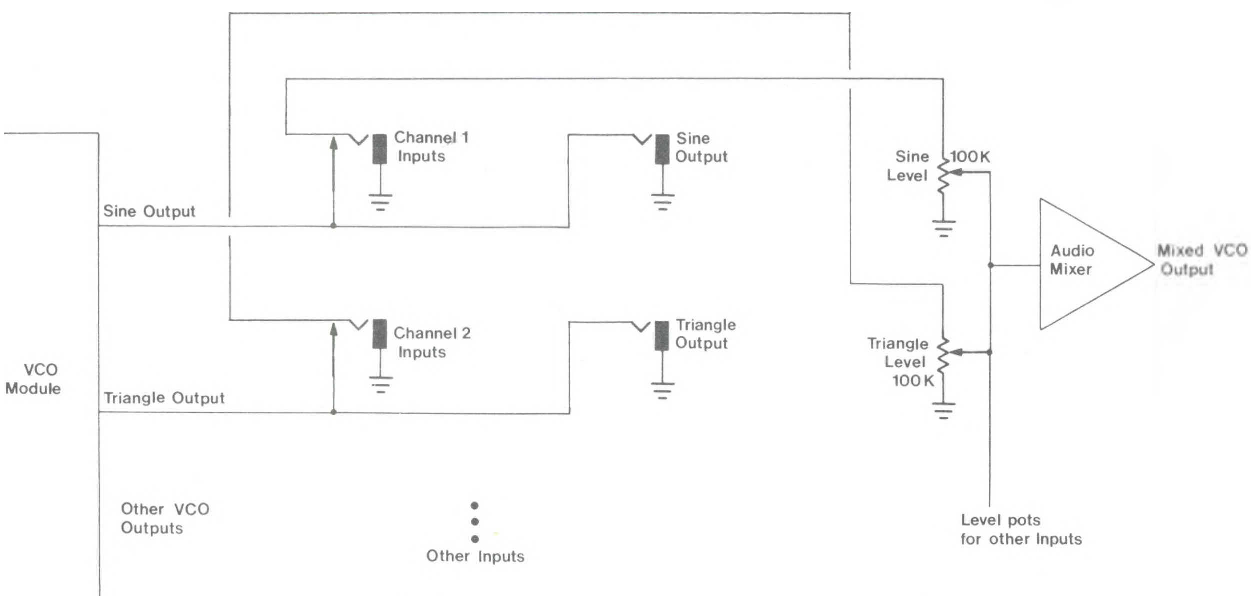


Fig. 2
Patch-over mixer for a VCO.

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Expanding Envelopes

By: David M. Vosh

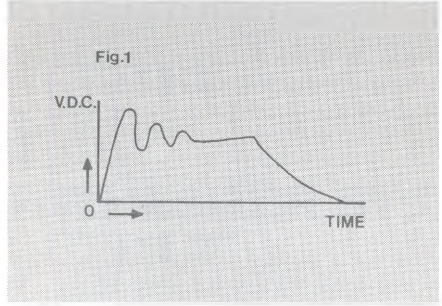


Fig.1

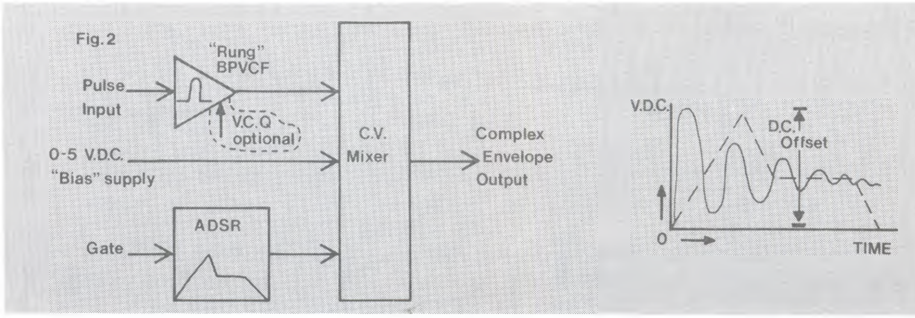


Fig.2

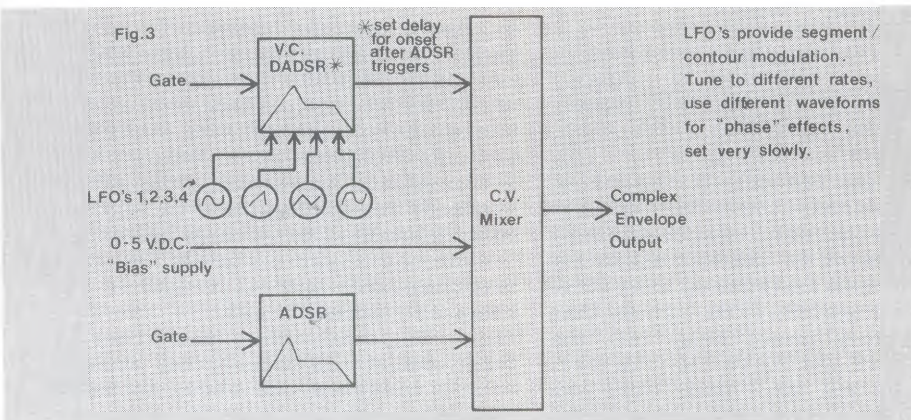


Fig.3

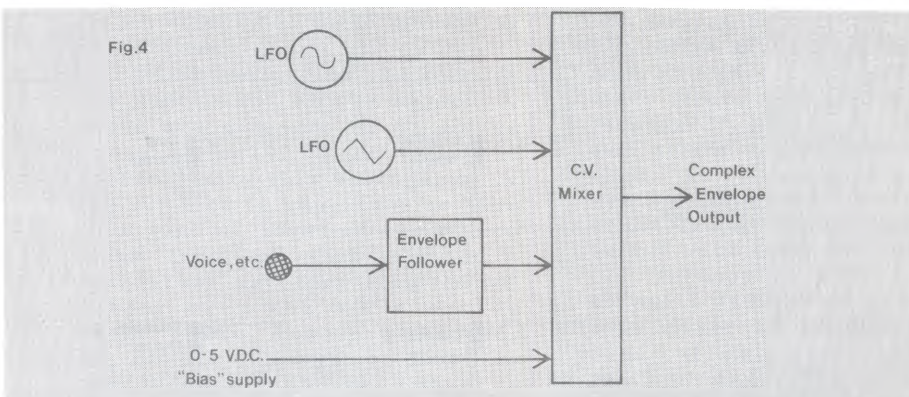


Fig.4

With analog synthesizers, it seems that the ADSR has become "standard" with the AD running a close second. Most of the emerging digital synthesizers have the capability of more complex envelopes, but what many people don't know is that it's fairly easy to develop more complex envelopes on our analog machines! All you need are the usual modular synthesis components and an inclination to explore.

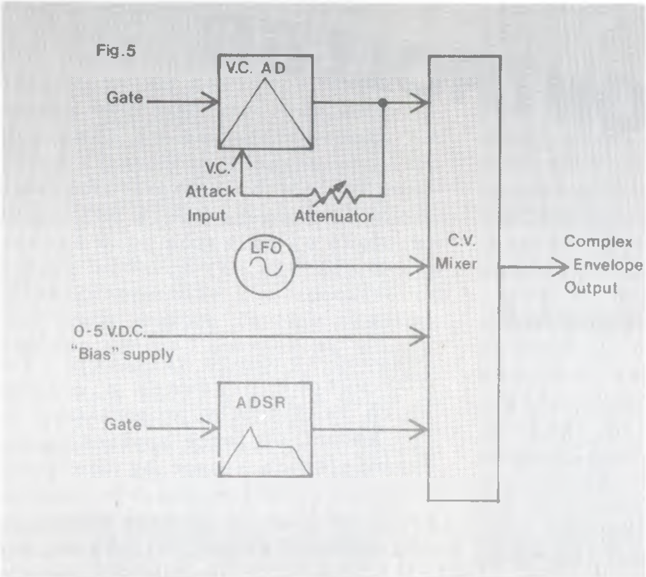
For the "hardware artists" out there, I should mention this subject has been addressed in "Electronotes" in recent years. An "Envelope/Transient Generator" was presented in EN #86, which is based on a 2nd order filter section and produces envelopes as shown in figure 1. An "Articulating Contour Generator" was given in EN #110 which produces triggered, 8 segment envelopes with two levels of smoothening (provided by integrator circuits) and some other enhancements. Both of these circuits are quite interesting, and Electronotes should be commended for their efforts to expand the hardware horizons.

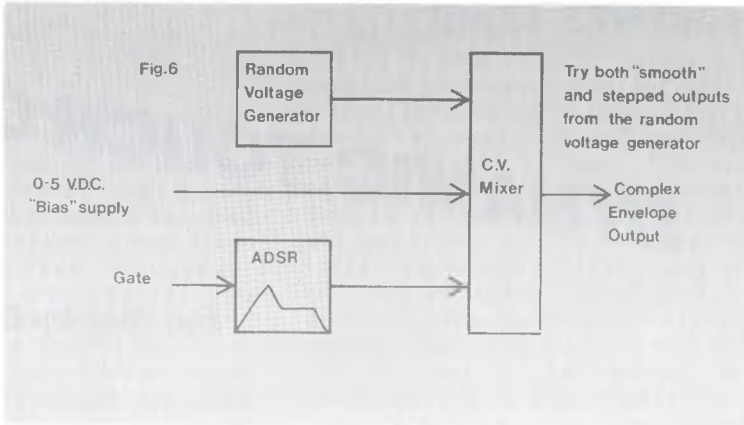
However, if your synthesizer has a state-variable VCF with voltage controlled Q, you can set up a patch to envelopes like the "Envelope/Transient Generator" so you don't always need dedicated hardware before you start experimenting.

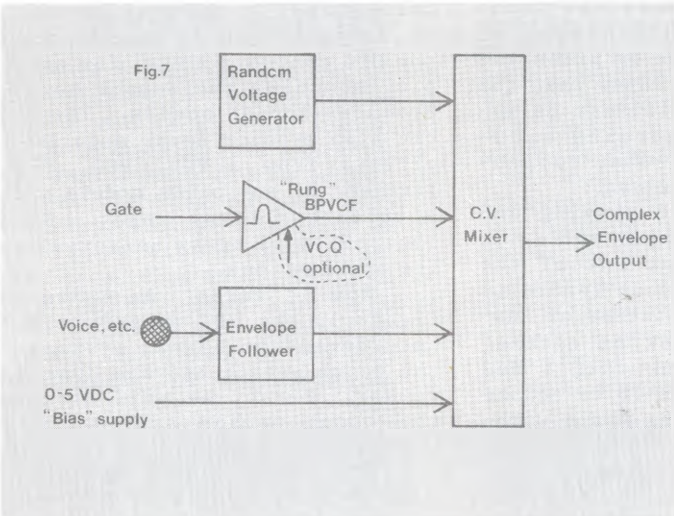
Figures 2 - 8 show patches which produce complex envelopes, and are starting points for further experimentation. The graph in figure 2 shows the typical parameters involved; you can "tune up" the envelope by adjusting levels at the CV mixer. If the mixer has inversion capabilities, that can also be useful.

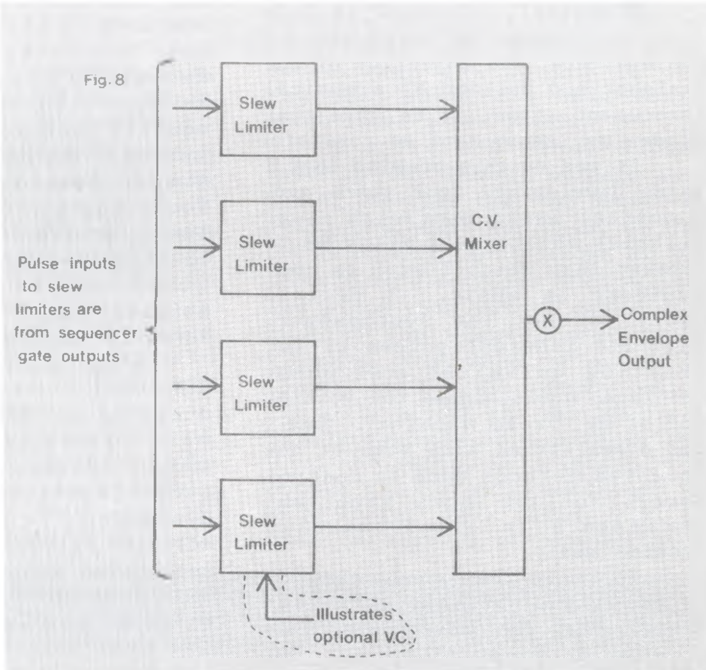
Analog sequencers can also generate complex envelopes, as has been mentioned before in this and other publications. Figure 8a shows a variation on this technique, where pulse outputs from the sequencer feed slew limiters before being summed at the CV mixer. Inserting an additional slew limiter at point is also most useful. Figure 8b shows a possible result of the complex envelope generator process. Slew limiters with separate attack and decay controls are helpful, and voltage controlled inputs add further possibilities.

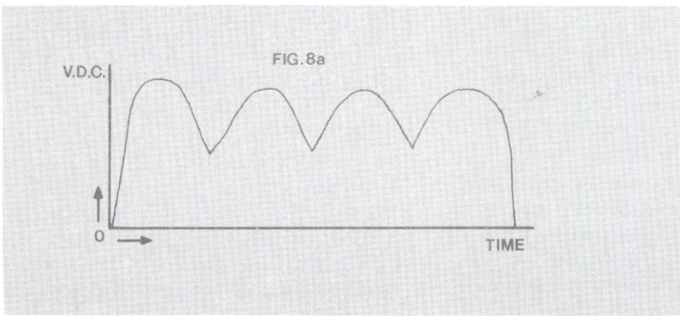
An additional consideration when using a patch such as figure 8a is the gate output spacing. In figure 8b outputs from stages 1, 4, 7, and 10 feed the slew limiters, with the 13th stage gate going to the sequencer's "stop"

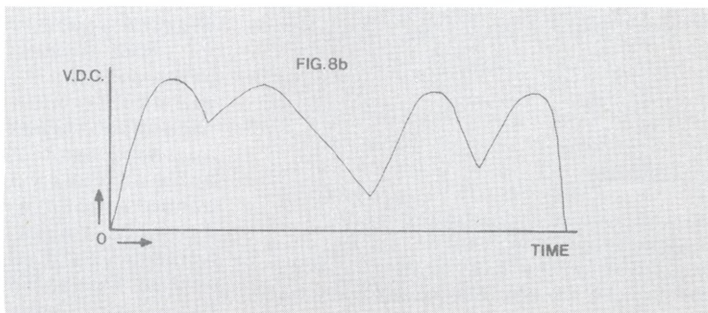












input. Other spacings produce quite a variety of envelopes; figure 8c is an example of an "irregular" envelope. The amount of slew limiting should be adjusted to taste.

Three other points are worth mentioning. First, you don't need a lot of slew limiters - only two or three stages will still give good results. Second, you don't need an overly complex sequencer; even a 4 stage version provides a good balance between musical usefulness and complexity. Third, we should mention possible applications. I particularly like using these complex envelopes to control depth of modulation (via VCA) in FM patches, and heavily attenuated the CV to provide small frequency variations in multiple VCO patches. Most signal processors seem to respond well to these envelopes as well.

I hope this article will stimulate experimentation in this area by others interested in creative synthesis. Naturally, if you turn up anything interesting, you should let everyone know via an article in Polyphony.

Build the DYNAMIC TOUCH CONTROLLER

By: Bobby Beausoleil

A musical instrument is like a window between the performer and the audience. The objective of the designer and builder of a musical instrument, then, is to make that window as transparent as possible.

It was on this premise that I began developing, four years ago, a musical instrument with capabilities which would allow me to express that which I wish to communicate in music. I decided early on that the synthesizer, with its inherent potential for creating unlimited instrument voicings, would handle the voicing end of my "Dream Machine". But I was dissatisfied with some of the accepted methods used to control several of the most important parameters in electronically created music.

It is in the broad category of amplitude control, strangely enough, that the conventional synthesizer is the most lacking. Oh, we have all the volume we need. And low noise VCAs, and sophisticated transient generators. There is one parameter within this category, however, which has sadly been overlooked in the development of electronic music instruments: dynamics. By this I am referring to the kind of playing dynamics which occur (or should occur) in response to the manner in which a musician plays his or her instrument. It is not the musicians who lack dynamics; it's just that the vast majority of electronic musical instruments lack any facilities for interpreting the musician's playing dynamics.

Conventional amplitude control generally involves a transient generator (TG) and VCA. The TG responds to a gate or trigger signal, usually derived from some sort of mechanical switch (as with keyboard controlled synthesizers), and the TG voltage output controls the VCA gain by a pre-determined amount. Since the output of the TG is a static waveform, the VCA responds in the same manner from one note to the next. Hence, the only way a synthesist can add dynamic variety in this instance is with a volume pedal, or some equally laborious and inadequate method. So, dynamics are often simply overlooked in favor of focussing attention on pitch and timbre control, in hopes that the lack of dynamics will not be noticed, or, in recording situations, can be somewhat compensated for during mixdown.

After years of playing and listening to electronic music, it occurred to me that one of the main reasons the music tended to sound "electronic" instead of "natural" was the lack of natural dynamics. To think that I had actually allowed myself to become accustomed to hearing music without dynamics! Upon realizing this, I made up my mind that my "Dream Machine" would incorporate the facilities necessary to interpret the natural dynamics of my playing style.

The search for a device to provide this function led to two remarkable instruments, the Blacet "Syn-Bow" (see Polyphony, May/August), and PAIA's "The Drum". Both of these instruments use a piezo-electric transducer as a dynamic control element; the transducer drives a transient generator which, in both cases, has a fixed percussive attack and a variable decay rate - no variable attack, sustain, or release controls are provided. For these instruments, this is not a severe limitation, but I wanted more flexibility.

The Dynamic Touch Controller (DTC) is an electronic realization of a mechanical device. It is designed to mimic the dynamic properties of traditional musical instruments, such as the acoustic piano and other instruments which respond to touch velocity. The DTC output produces a voltage which is directly proportional to the amount of force applied to the dynamic/touch sensor by the musician. You may alter the raw dynamic voltage to produce virtually any desired transient shape, and this can then be applied to the control inputs of voltage-controlled modules for the creation and/or manipulation of sound. When the DTC controls a VCA, for example, touching the sensor harder produces louder sounds. The DTC can also produce novel modulation effects from a VCF, VCO, PWM, flanger, or any other module with a voltage control input. This controller is truly universal in its applications.

How it works. As shown in figure 1, the DTC comprises two interdependent sensor systems - the touch sensor and the dynamic sensor. First, we will discuss the touch sensor system built around IC1.

The touch sensor is really just a touch switch of the simplest possible kind. It tells the rest of the system (and any external modules) when the sensor is initially touched, the duration of the touch, and when the touch ends. Placing a finger on the contacts of the touch sensor provides enough continuity to cause IC1A to switch to a high state at the output, producing a gate signal. The remaining three sections of IC1 derive a logical inversion of the gate, a short duration (just under 1 ms.) trigger pulse, and its logical inversion. These two pairs of complementary outputs provide the logic required within the DTC system to develop several types of dynamic control signals. The gate and trigger also feed outputs intended to activate external devices; R11 - R14 form voltage dividers to attenuate these signals to the standard +5V required for compatibility with most synthesizer equipment - assuming, of course, that the DTC unit is powered by the recommended ±15V supply.

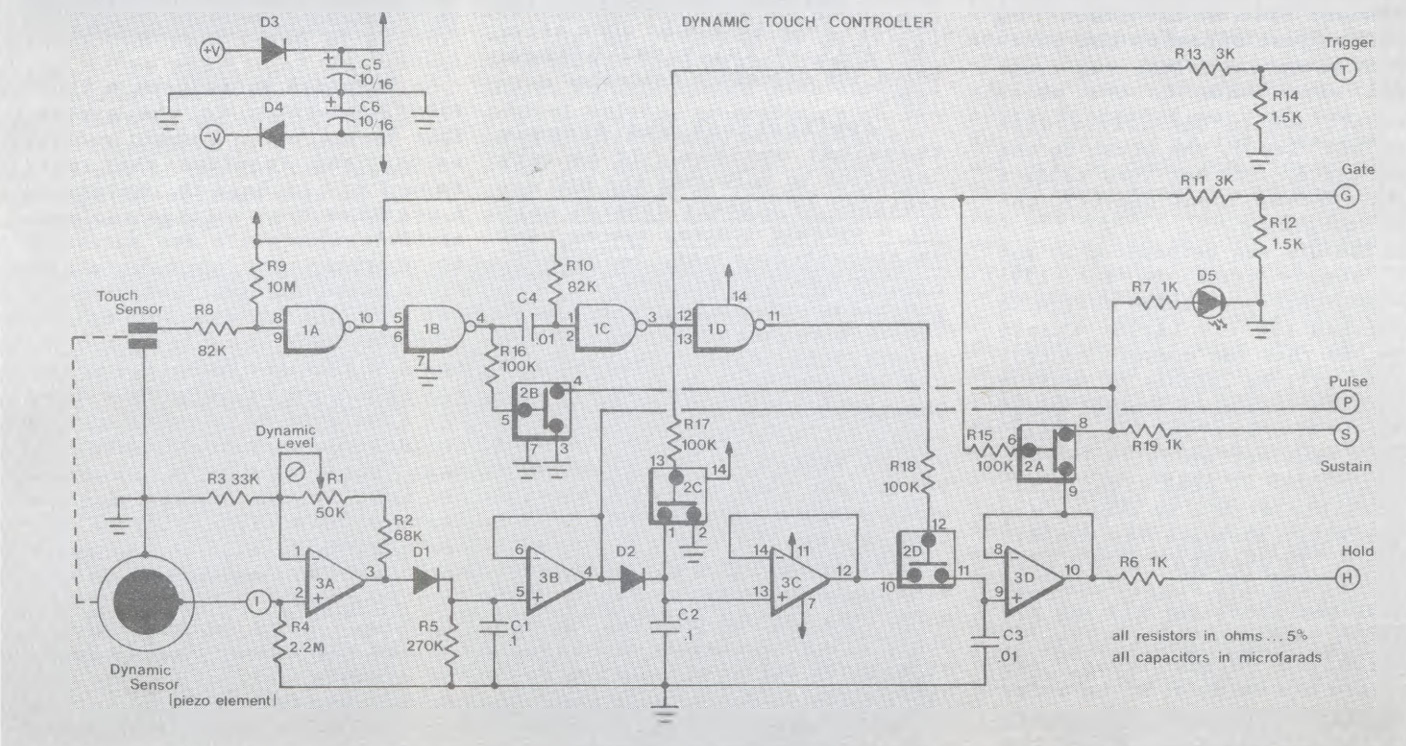
The purpose of the dynamic sensor is to tell the external system how hard the combined sensor unit is being touched. A piezo transducer element provides this intelligence to the DTC system. The piezo element is sandwiched with adhesive to the component side of the circuit board directly opposite the contacts of the touch sensor; in this fashion, the two sensors operate as a unit.

Tapping or touching the piezo transducer emits a single pulse at an amplitude which reflects the amount of stimulation. Preamp IC3A buffers and amplifies this signal. Trimmer R1 provides a means of adjusting the dynamic sensor sensitivity to accommodate various personal playing styles and control signal requirements, as it governs the output level of all three dynamic control outputs. A simple filter follows the preamp and selectively lags the pulse to extend the decay time somewhat, while preserving the characteristic fast attack of the pulse. D1 serves the dual purpose of preventing premature discharge of C1 so that it will discharge at the rate prescribed by R5, while eliminating the negative voltage transition of the original pulse signal. The output of the buffering stage for this filter is tapped off and sent to an output pad (P), which will henceforth be referred to as the dynamic pulse output.

For a unit with full-on dynamic control possibilities, we need more than the percussive type envelope provided for above. So, the DTC includes a sample-and-hold circuit which derives a sustained dynamic signal from the dynamic pulse. There are actually two sample-and-holds used for this function, each operating in a complementary fashion and run in series. The first S/H is under control of the touch sensor's trigger signal; when the sensor unit is touched, the trigger pulse at IC1C's output closes IC2C (1/4 of a quad bilateral switch), which instantly dumps the voltage stored by C2 to ground. This simultaneously resets the S/H, whereupon it is prepared to receive and hold the next dynamic pulse, which occurs precisely at the same time as the trigger pulse. The second S/H works in a manner exactly opposite the first, because it is under control of the inverted trigger (IC1D's output). Normally, switch IC2D is closed and passes the held dynamic voltage to its output. For the duration of the trigger pulse, however, the switch is open; the previous voltage sample is held momentarily by C3 until the reset period is over, whereupon it resumes the function of doing nothing (other than following IC3C's output). This second S/H prevents the abrupt transition to zero Volts which occurs during the reset period from being passed to the output, thereby eliminating the possibility that the transition will cause an audible pop in the module it controls (this could be especially true if the module under control is a filter). The end result of all this is that we now have a voltage proportional to the amount of force applied to the dynamic sensor, as we do in the case of the dynamic pulse output, except that this new dynamic hold output (H) will remain at a constant level (even when the sensor is not touched) until the next touch establishes a new level.

A third and final output sustains the dynamic voltage only during the time that the sensor is being touched. For this, we use the remaining two sections of IC2 to form an electronic DPST switch, and place this switch under control of the complementary gate signals. Now, when the sensor is touched the dynamic sustain output (S) will reflect the voltage level of the dynamic hold output, and when the sensor is released, the output will drop immediately to zero Volts. This output will be used to develop attack-sustain-

Fig.1



release transients - which, of course, will be dynamic.

Finally, an LED monitor provides a visual indication of the dynamic output level when the sensor is touched. This is tremendously helpful when learning how to gauge touch velocity to obtain a desired response. As with any sophisticated musical instrument, some degree of practice is required for best musical results with the DTC.

Construction. By using a printed circuit board, the touch sensor pattern may be etched on the same board that the electronic components mount on, thus making the DTC a self-contained unit. For remote mounting of the sensor, which may be best in some applications, the sensor portion of the board may be cut off, and connections made via a two-conductor shielded cable. See the end of this article for information on parts kits for the DTC, as well as sources for obtaining the piezo element for those who prefer building from scratch.

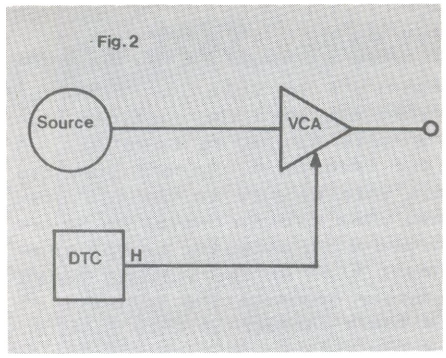
In mounting the piezo element to the board, note that the brass side of the disk is the side which goes against the board. If this is reversed, the positive excursion of the pulse will follow the negative excursion, resulting in a delayed response which can be very annoying. Use a rubber compound adhesive to affix the disk to the component side of the board in a position just opposite the touch switch pattern. Solder two insulated wire leads to the piezo element, one to the silver disk (the "hot" lead), the other to the brass plate. The hot lead solders to the preamp input pad (I), the other to ground.

The DTC can be powered by any dual supply from ±9V to ±15V; however, this particular design is optimized for the latter supply range, so that the dynamic control outputs will be capable of peaking at 10V (the standard control range in many synthesizer systems). Even if your system falls in this category, the 8V peak signal that is possible with a 9V supply may be perfectly adequate for dynamic control. To operate the DTC with a ±9 Volt supply, simply use 3.6K or 3.9K resistors for R12 and R14 to bring the gate and trigger up to standard output level, and decrease the value of R7 in order to obtain sufficient brightness from the LED monitor. After the unit is calibrated, use a cut-and-try procedure to determine the value of R7, as the maximum brightness of the LED is also relative to the setting of R1. Since the DTC may be powered by a ±9V supply, battery power is possible. This may be best in some applications, as it cuts down on some of the wiring traffic and makes remote installations a lot easier.

Calibration. Calibration of the DTC involves adjusting R1 to obtain best results in response to an individual user's playing style. After applying power to the unit, read the dynamic hold output with a VOM and adjust the trimmer until the maximum output level (corresponding to maximum touch velocity) is within the limits prescribed by the control inputs of your system. After establishing this setting, any further adjustment of the trimmer for personal playing style should result in a maximum output level that does not exceed these limits. It may be that the initial setting will be just right for your playing style, but if in practice the response is too sensitive then back off on the gain a bit. If you are using more than one DTC in a system (highly recommended!), take extra care in calibrating each unit so that each unit responds identically to an equal force of touch. Again, it may be necessary to decrease the value of R7 for optimum LED monitor response. This should be done after the unit is fully calibrated, using the procedure described earlier.

Applications and dynamic transient options. It is very important to note that the DTC was designed as a basic control unit for a dynamic control system, and as such it does not, by itself, provide useful waveforms for the generation of musical envelopes. The development of dynamic transients, therefore, involves the use of outboard devices. The thought behind not including a transient generator in the DTC design proper is rooted in the philosophy that the unit should offer maximum versatility; rather than being limited to one TG, several dynamic TGs may be driven simultaneously by a single DTC unit. The options provided for in the DTC design suggest a variety of methods for implementing dynamic control in a music system.

The diagram in figure 2 shows what amounts to a touch operated



volume control. Its main benefit is that making fast level changes by touch is far superior to making volume changes with a pot. This method may process any program material in which the musical transient is already shaped. One application is processing prerecorded tapes during mixdown to compensate for a static quality in the original program material. Tapping the sensor in time with the music creates subtle, or dramatic, dynamic alterations.

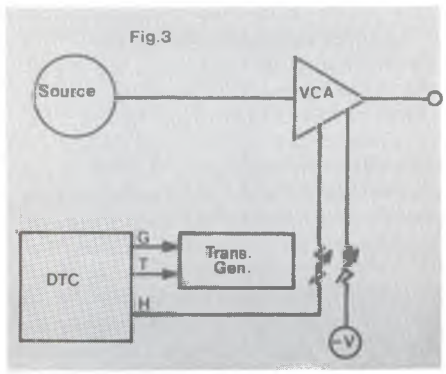


Figure 3 introduces a transient generator to the system. This is the basic dynamic control patch, and requires the least amount of hardware to implement. A negative bias voltage and some careful adjustments are necessary to properly set up this patch. First patch the gate and/or trigger outputs of the DTC to the transient generator, and patch the audio source and TG output to the VCA in the normal way. Touching the sensor at this point should produce the predictable ADSR dynamic response with which we are all familiar. Before applying the dynamic hold signal to the VCA control input, give the sensor a hard tap - as hard as you anticipate touching the sensor when playing. Now, without having touched the sensor again, apply the dynamic hold signal to the input. Gradually feed in the negative voltage until the sound which appears at the output of the

VCA just disappears. The next time you touch the sensor, the velocity of the touch will determine the output level of the VCA, while the controls of the TG will determine the musical envelope. The pot in the dynamic hold path varies the amount of dynamic signal in the VCA's control voltage summer. Juggling back and forth between this control and the negative bias control lets you adjust the dynamic range of the system.

A variation on this patch involves replacing the VCA with a VCF. This is similar to controlling the filter with a keyboard voltage, but instead of tracking pitch the filter will track dynamics. You may even do both simultaneously. Or, for that matter, you may use the DTC to control both the VCA and the VCF, producing simultaneous amplitude and timbre variations in response to playing dynamics. Optionally, the dynamic control voltage may be inverted before delivery to the VCF, resulting in a brighter sounding timbre at lower amplitude levels and a muted sound at greater amplitude levels. And there are many other possibilities. In short, the DTC is as useful for timbre control as it is for amplitude control.

Figure 4 shows my favorite system. It requires an additional VCA (with DC response so that it may process control voltages), but it has many advantages. (For a suitable VCA, see Thomas Henry's "Practical Circuitry" column in the Jan/Feb 1982 issue of Polyphony.) The additional VCA works in conjunction with the DTC to operate as an automatic, dynamically responsive level control on the output of the TG; the resulting control voltage may be used anywhere you want to use it. In this circuit, the dynamic range control is reduced to a single pot. At one extreme, it offers the full dynamic range; rotating the control gradually compresses the dynamic range until at the other extreme of rotation, the output is a static waveform.

Using a voltage controlled transient generator (with external control inputs to each of the ADSR parameters) provides some very useful and often exotic transient shapes - simply route the ADSR output (or the output of an external TG) to one or more of the parameter control inputs. Or, for something really different, one or more of the ADSR control inputs may be fed by the DTC's dynamic hold output; this could result, for example, in the attack time varying in response to playing dynamics. Then again, replacing the TG in this system with another control voltage generator, such as an LFO, offers many more possibilities for using the DTC - controlling the amount of vibrato modulation to a VCO, for example.

Fig.4

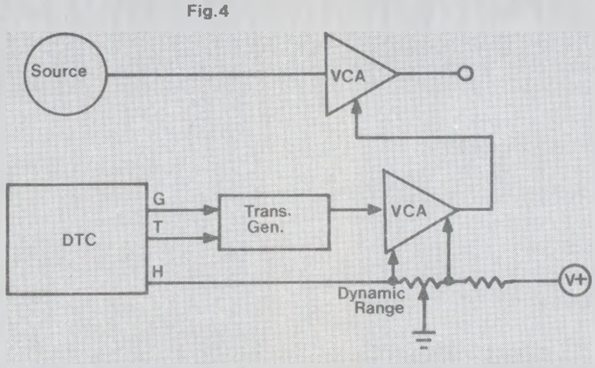
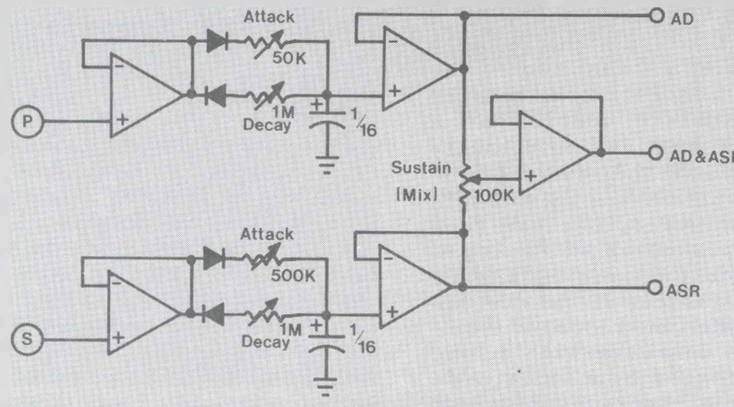


Fig.5



This seems to be an appropriate time to mention that the DTC will, to a limited degree, respond to pressure. Response to pressure is limited to the extent that easing off on the pressure applied to the sensor will not decrease the output from the DTC, as this would require resetting the S/H. But increasing the pressure applied to the sensor will result in an increased voltage output, without retriggering the TG. This feature is of great benefit in amplitude and timbre control, as well as pitch control (such as using the pressure sensitivity to increase the depth of vibrato while holding a note).

For those of you who want lots of dynamic transient generators at minimum cost, try the circuit in figure 5. This simple AD + ASR type transient generator will do most everything the circuits described earlier can do, but it will do it for less money and only a small time investment. The circuit consists of a pair of lag processors run in parallel; one processes the DTC's dynamic pulse output, while the other processes the dynamic sustain output. The controls provide a way to independently vary the rate of ascending and descending voltages, like conventional transient generators. The attack phase of the AD section is limited to the duration of the dynamic pulse, but attack times close to a half-second are possible (use the ASR section when longer attack times are desired). Other than that, the circuit works just the same as other TGs of this type, except that the input dynamics are retained at the outputs. Each TG section has its own output, and a third output offers a continuously variable mix of the two - the associated control works so much like the sustain control of a conventional ADSR it could easily be called that. The advantages of having two attack controls should not be overlooked, because they offer the potential of having a lot more control over this most important parameter of the musical envelope. For instance, a transient can be developed with a long attack preceeded by a short percussive burst - perhaps just the thing to help with realistic brass voicings.

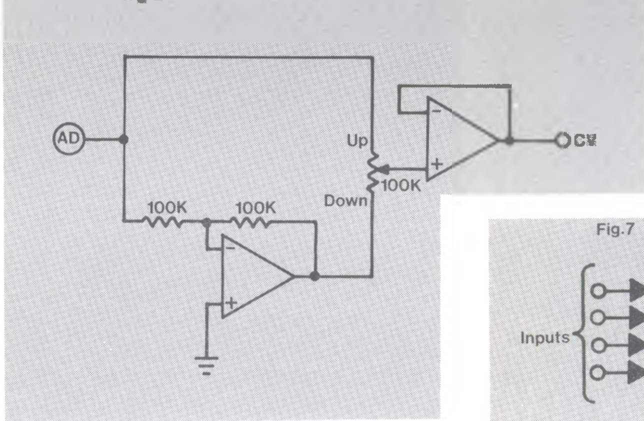
Figure 6 shows a related circuit. This little gem offers the potential of using the DTC as a pitch bender device for keyboard synthesizers which haven't got one. The circuit is used in connection with the AD output of the circuit in figure 5, giving control over the rate and shape of the bend. The pot selects direction of the bend and governs the maximum depth of modulation in both directions. That's all there is to it; just plug the thing into your VCO modulation input and go for it.

Both of the circuits in figures 5 and 6 may be built in a matter of minutes on perfboard using the two 4136 quad op amps. This leaves one op amp left over which might be used to build an inverter for the inverted dynamic filter control method discussed earlier in this article. The result will be an extremely versatile control system, to say the least.

System packaging. If one DTC module can perform wonders, multiple DTCs can work miracles. I firmly believe that the DTC shines best in multiples. After all, each of us has more than one finger to a hand, so why not use them all? If two or more fingers control as many DTCs, it is possible to achieve an absolutely unprecedented degree of musical expressiveness. Now, that statement may seem overblown, but it's true. A multiple DTC system offers a greater potential for complex musical articulation than any method so far devised for manual control over amplitude and timbre in an electronic music system.

Immediately there is a tendency to think in terms of using multiple DTCs to develop dynamic

Fig.6

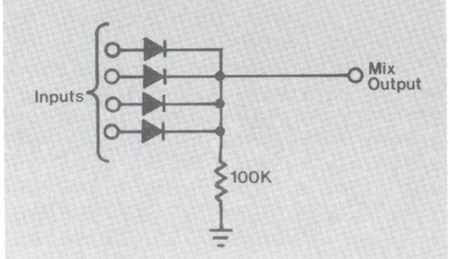


keyboard systems. Certainly this is a valid approach, and I would encourage any such ambition on the part of the reader. The boards that I have designed for the DTC measure a little more than 1" in width; the units may easily be stacked side by side and on top of one another to form a wide variety of custom keyboards. The DTC gate outputs can be matrixed to a digital encoder and converted from digital to an analog voltage for pitch control, if desired; such a system will lend itself equally well to both polyphonic and monophonic applications. And there is nothing to dictate the necessity of arranging the DTCs in the conventional AGO keyboard configuration. Other schemes are possible; perhaps something along the lines of the Buchla synthesizer's touch plate scheme would be a more appropriate keyboard application of the DTC system. The point is: It's the musician's choice.

My hope is that the DTC will inspire a departure from "keyboard thinking"; the keyboard-oriented musician might be better off to look into PAIA's recently announced keyboard retrofit. Although I have not seen the guts of the thing, I do know that it uses the piezo transducer and boasts some dynamic control options similar to those of the DTC. This would probably be the most cost-effective approach for keyboard musicians.

My personal preference is to control pitch independently of amplitude and dynamics in the manner of the vast majority of traditional musical instruments. Excluding percussion instruments, only keyboards control pitch and amplitude with the same device. Is it any wonder, then, that keyboard synthesizer imitations of traditional instruments so often sound like they were played on

Fig.7



keyboards? Part of the philosophy behind the design of the DTC is that a successful voice imitation depends just as much on how it is played as on the nature of the sound itself, and most musical instruments control amplitude independently of pitch.

I hope to prepare a future article covering the design for a dedicated pitch controller that I believe to be the ideal mate for the DTC system. For the meantime, you might want to try using a keyboard exclusively for pitch control while using the DTC to handle all dynamic-related functions. Other possible methods of dedicated pitch control will present themselves as you experiment with this concept.

Remember, only a few DTC units are required for a full-blown system since all you need is enough units for one hand. Four DTCs is plenty. This will leave the thumb free to perform other functions, such as octave switching - reminiscent of the technique employed by woodwind players for the selection of register.

Whatever your preference, using more than one DTC is bound to present some situations in which having a mono mix for each DTC function will help to minimize hardware requirements. The simple diode mixer in figure 7 will do the job. Any like outputs from multiple DTC units can be combined with this mixer, but avoid mixing unlike outputs (i.e., a trigger with a dynamic sustain). Also, there is not much use in mixing the dynamic hold outputs because the mix will only reflect the highest level output whether that output belongs to the last DTC activated or not. This mixer will not cause interaction between units, maintaining the integrity of the individual outputs. And you may extend the pattern indefinitely to combine as many DTC units as you wish, even if your intention is to build a large keyboard system.

Conclusion. The Dynamic Touch Controller is a new type of device for music synthesis, and some time may be required for some of the concepts behind the development of the device to take hold on a broad scale. The best way to know if this method of control is right for you is to give it a try. The DTC invites experimentation, both from a construction standpoint and a user's standpoint.

So, experiment! I am sure you will be pleasantly surprised, if not utterly amazed, by what real dynamic control can do for your music.

PARTS LIST

Resistors

R1	50k trimpot		
R2	68k	R8	82k
R3	33k	R9	10M
R4	2.2M	R10	82k
R5	270k	R11, R13	3k
R6, R7,		R12, R14	1.5k
R19	1k	R15-R18	100k

Capacitors

C1, C2	0.1 uF mylar
C3, C4	0.01 mylar
C5, C6	10 uF, 15V

Semiconductors

IC1	CD4011 quad gate
IC2	CD4016/CD4066 switch
IC3	4136 quad op amp
D1-D4	1N914 or equiv diode
D5	LED

Other Components

DS1	Dynamic piezo sensor
Misc.	Circuit board, solder, wire, etc.

PARTS KIT AVAILABLE: A kit of parts is available for the DTC including circuit board, dynamic transducer, and all electronic components which mount on the board (builder provides power supply). $29.95 postpaid from Beausoleil Enterprises, PO Box 1033, Grover City, CA 93433. Four kits are available for $100 postpaid; write the company for quotes on larger quantity purchases. The dynamic transducer is available separately for $4.95 postpaid. California residents include appropriate sales tax on any purchases.

PolyTest: DTC

We built the DTC using a circuit board provided by the author. Construction went fairly easily; the piezo transducer was held on with hot glue, which seemed to work just fine.

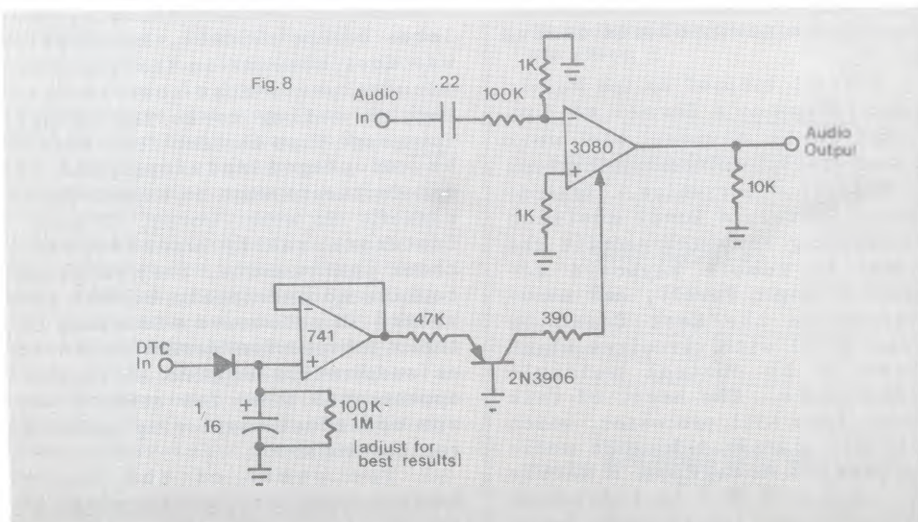
Bobby is right when he says that the DTC does not, by itself, provide many useful waveforms. In fact, after first building the DTC, the results seemed somewhat disappointing. The gate output occasionally picked up hum and was therefore a bit inconsistent. Striking the sensor squarely (and cleaning the contacts when they got dirty) corrected this anomaly 99% of the time, and for the remaining 1% of worst-case conditions, a little lag processing or low pass filtering will remove any residual hum from the gate output. Still, even though the DTC was now working up to specs, outboard units were clearly required to make best use of the other outputs.

We then tried the evaluation circuit shown in figure 8. This lets the dynamic peak open up a VCA, and pass an audio tone. The DTC proved to be a very reliable, and amazingly consistent, dynamic sensor. It seemed far less critical than other dynamic sensors, and if you hit it repeatedly with the same force, the amplitude would remain the same rather than varying slightly. It was surprisingly simple to get a feel for the dynamics, and thereby achieve satisfying and highly consistent results from the device.

There is one caution, though. Note that the touch sensor includes a ground trace, which means that you must touch ground to make this unit work. Those of you who have touched grounded hot guitar strings and an improperly grounded amp are familiar with some of the shock hazards possible with this kind of arrangement. While the odds of encountering a hazardous condition are not great, it is possible so make sure that all equipment being used with the DTC is properly grounded. This will minimize any chance for shock hazard. Note that most touch circuits published in hobbyist magazines and the like have a similar potential for shock hazard, so this is by no means a problem which is unique to the DTC.

Using some of the outboard circuitry recommended in the article convincingly demonstrated the musical usefulness of the DTC. Overall, the DTC is yet another indication that synthesis still has many new worlds to conquer. Adding dynamic control really does make a difference, and while the article gives a reasonable number of applications, there are certainly others just waiting to be discovered.

If you're even remotely into alternate controllers or electronic percussion circuits, check out the DTC - building one doesn't require a lot of effort or money, and the excellent results belie its simplicity.



ON LOCATION:
Northern California

By: CRAIG ANDERTON

Northern California (specifically the San Francisco Bay Area) has a lot of activity in the musical electronics field, what with CES and SSM (music IC manufacturers), Sequential Circuits, Syntauri, E-mu, SMS , Gleeson, A/DA, etc. With the NAMM show coming up in January, I thought it would be a good idea to visit some of the local companies and get an advance look at products scheduled for introduction at the show, as well as talk about the future of the industry with some of my friends in the business. I also wanted to check out the industry reaction to the MIDI (Musical Instrument Digital Interface) specification worked out by several companies in the industry.

As it turned out, there were far too many people to visit in the allotted week. So, I hope to take another trip before too long and talk to all the people I missed this time around, such as Doug Curtis (of CES), Serge synthesizers, Furman, Donald Buchla, and so on.

Most of the companies had been working on top secret projects, but as the NAMM show looms it becomes harder and harder to keep a secret. Luckily, it seems I came along at just the right time -- early enough to get some advance word out to Polyphony readers, but late enough so that I could actually see and evaluate some of the new products coming out.

I first stopped to see Marvin Jones, Polyphony's former editor. He currently divides his work between digital telecommunications and musical electronics. Regarding the latter, a local musician is retaining Marvin to outfit the "Probe" (a remote keyboard designed by Roger Powell, and manufactured on the East Coast by Jeremy Hill) with circuitry which allows it to control multiple synthesizers. The heart of this circuit is a 6511 processor, which sends all signals through a modified RS-232 interface. Marvin also mentioned that he had talked to Roger recently, and that Roger was using only an S-100 computer with Casheab music boards and a Polysix on his tour with Utopia. That's just one more sign that huge keyboard stacks are soon going to be a thing of the past (if they aren't already).

After a bit, Kirk Austin, who wrote the "$5 Analog Programmer" article in the July/August '81 issue, got together over dinner with Marvin and myself. These days, he's working heavily with speech synthesis, but most of the conversation concerned computers in music, and the impact of cheap memory. We discussed an article in Electronic Engineering Times about an experimental process which claimed to be capable of economically putting megabytes of non-volatile storage on a chip. If it pans out as expected, that will change not only synthesizers, but probably the way we record, play back, and even buy music as well.

We also talked a bit about the MIDI specification. Marvin's fist comment was "I'm amazed that five companies in the music industry got together on anything", a comment which was echoed by several other people during my trip. Having tried to establish standard input and output characteristics for signal processors a few years back, I knew exactly what he meant. This led to further discussion of how the industry relates to its clients, and Marvin was most adamant in stating that manufacturers must realize that they're making tools for an art form, and that the art form should be encouraged and supported if manufacturers want to expand their market. In other words, if manufacturers really want to sell their instruments, they're going to have to once again become involved in extensive education of their prospective market, as well as cooperative support and development with those few artists who are creatively exploring advanced realizations.

The state of the music business was also on the minds of the people at A/DA, the next day's first stop. There I met with Dave Tarnowski (president) and Mike Maia (marketing).

First, I looked at A/DA's new digital delay which covers 1.2 seconds for $800. It's in the same price range as other delays, but includes a number of interesting convenience features. The final prototype, which was virtually ready to go into production, was working away when we walked by; considering the price/performance ratio, it should do well in the marketplace.

We talked about the MIDI specification, and Dave -- like many other people I talked to -- wondered why an existing computer standard (such as one of the networks, or IEEE-833 General Purpose Instrument Buss) hadn't been chosen instead. These were questions I brought up at my meeting with Sequential Circuits, which we'll cover later on.

Eventually our conversation turned political and global. Dave had just returned from the Far East, where he had seen some of the latest examples of Japanese high technology; for example, he had visited a semiconductor factory where 6 people, aided by 400 robots, controlled a 10,000 square foot facility. He also mentioned that they did their chip testing with low level E-Beams, not wired connections. "It's so far ahead of (prominent American semi manufacturer) I just don't think we're going to catch up", he said at one point. I asked if he felt the musical electronics business would come to be dominated by Far Eastern manufacturers, but he indicated that could take a while. He sees Far Eastern manufacturers as taking over what are essentially mature industries, and driving prices down to the point where the US is non-competitive through aggressive pricing, low wage scales, and automation. The musical electronics industry is not really a mature industry, and Dave believed that there is still a significant

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ON LOCATION:

market for American-made equipment. This opinion was dramatically confirmed later on in the trip.

Dave also expressed concern with the way many companies focus on short term gains. He feels that one of the reasons why IBM has retained its dominance in the computer industry, despite both foreign and domestic attempts to change that, is because they have a massive budget for research and development. He stressed that it's not enough to simply create a market, because you're opening yourself up to having that market taken over by someone else: you have to continually innovate and explore new paths in order to hold on to a market.

From A/DA, it was about an hour's drive down south to GPI, which publishes Guitar Player, Keyboard, and Frets magazines. Everyone had just finished sending the February issues of GP and Keyboard off to the printers, and was appropriately frazzled. As I was staying with Jim Aikin, who in addition to writing record reviews and other articles for Keyboard has also written some very well-received articles for Polyphony, we took off for the nearest Italian restaurant to talk over the latest industry gossip.

Then it was time to look over Jim's modular Serge system and 8 track studio, based on a TEAC Model 38. For those of you who aren't familiar with the Serge synthesizer, it's a very advanced, and sound-oriented (as opposed to keyboard-oriented) synthesizer capable of producing extremely rich timbres. Several of the Serge modules are unique, such as their Wave Multiplier module, which appeared to be some type of precision controlled distortion device. Suffice it to say that after seeing Jim's Serge, I decided that I simply had to visit the Serge company on my next Bay Area trip.

In case you wondered what kind of music a music reviewer makes, Jim's songs use the Serge to great advantage with thick, FM/ring modulator type sonorities interspersed with more conventional synthesizer sounds. It's good to know that at least some reviewers experience the process of making tapes and playing them for people first hand...maybe that's why he writes such empathetic reviews.

From there, the evening dissipated in a haze of beer and discussions of science fiction, aliens, the art of writing fiction, future types of music, and much more. That's one thing that I really like about electronic musicians; everyone I've ever met usually has a myriad other interests as well.

The next morning was spent going over my latest column with Guitar Player, and playing with Yamaha's new MP-1 which had sent in for review. This little keyboard not only included features such as automatic accompaniment, but also had a ball-point pen based printer which printed out lead lines on a roll of paper. That may sound flaky, but the quality of the printout was quite good -- sharps, flat, nice G clefs, and the like. It wasn't the fastest printer in the world, but for a portable unit, what do you want? I'm not sure exactly what the intended market is (probably educational), but the implications of this type of technology are clear: keyboards will only get less expensive, more compact, offer more features, and rely more on digital technology in the years ahead.

That afternoon was scheduled for Sequential Circuits, located in a 33,000 square foot facility towards the north end of San Jose. Originally I had gone to talk about the MIDI specification, so that's where I started. I met with engineer Chet Wood and asked first about the origin of MIDI, which was apparently hatched by Dave Smith (Sequential's president) around August 1981. He presented a paper at the Fall AES on a standard digital interface, and then Sequential sent out a batch of questionnaires to various manufacturers concerning the proposed standard. Eventually, about two dozen people from a variety of manufacturers got together at the 1982 Winter NAMM show to hammer out a workable standard. Some companies wanted a parallel interface, some serial, some an RS-232 type standard interface, and so on. Then there were other factors, such as ground loop problems (opto-isolators were finally incorporated into the design to help solve this particular problem). Roland presented an alternative proposal which flagged the status bytes separately from the data

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Letters

.........continued from page 6

errors. Following are some corrections and additional information.

The MG-1 keyboard has two busses, however, both are used for "oscillators". The first develops the usual control voltage using the typical string of 100 Ohm resistors. The second acts as the summer for square waves generated by a top octave divider and flip-flops, like an electronic organ. Also, the keyboard trigger is developed from the second or polyphonic buss (called trigger, but more properly a gate). The second buss is the source of teh polyphony mixed in at the output mixer, hence can be used without using the 1st or 2nd monophonic oscillators at all. The polyphony control adjusts the frequency of the main oscillator driving the top octave divider, and has nothing to do with the two tone generators at all.

Two MG-1 features are derived using a single CD4046 phase locked loop. The VCO portion drives the top octave divider, while the phase comparator (an EX-OR gate) generates the "belltone". The square wave outputs of TGs 1 and 2 are EX-ORed, and if the frequencies of the two are not locked and an octave or more apart, this can create some rich harmonics with phasing effects built in.

Of course, due to the simplicity of design a lot more features can be easily added. For example, the summing buss for all the tones is fed by large value resistors, so that adding oscillators is simple. The keyboard voltage is generated by op amp summers, the driver to the VCOs being an LF353 which can source 10 - 20 mA (depending on polarity) easily to drive any sort of VCO you wish. The power supply regulator chips are 78MXX and 79MXX series, but can be replaced with higher power 78XX and 79XX series for more power (however, the power transformer may need to be replaced -- but there is plenty of room for a larger unit).

Finally, the "computer ports" are a bit of a disappointment.

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bytes to simplify receiver routines; this concept was incorporated into MIDI as well. Finally, the various parties agreed that the RS-232 standard was too slow, that parallel presented various stumbling blocks, existing interfaces (such as the GPIB interface) were usually too expensive, and so on. As a result, Sequential Circuits and Roland (serving as a liaison with Korg, Yamaha, and Kawai) worked out a final interface specification which would let their various keyboards, sequencers, and drum units all work together coherently. This means that keyboards equipped with a MIDI interface could be slaved together (therefore allowing you to play several synthesizers from one keyboard), and you could at the same time synchronize the whole setup to any drum unit equipped with a MIDI interface jack.

It remains to be seen whether the industry as a whole will make equipment compatible with these specifications, but for now, MIDI is off to a good start and shows that competitors can work together for the good of their customers. Now if they could just standardize synthesizer nomenclature...

I then met with Stanley Jungleib, who had written the article on the Pro-One digital interface in the November/December '81 issue. He is currently working on an article describing the MIDI interface in depth so that others may design equipment compatible with the standard. Polyphony will be publishing the entire article as soon as Stanley sends it to us.

While at Sequential, I took a look at their Pro-FX system. It's quite costly ($1000 for the mainframe, $300 and up for the various modules), but then again, it's intended for high level applications, not the garage band down the street. They were currently finishing up their spring reverb module, but the big news was that Sequential was readying a DDL for introduction later this year (whole systems may not be released until May '83). I also asked about their flanger, which features a very wide sweep range and true stereo effects. While other musicians I talked to had said that the Transpose/Sync module works incredibly well, unfortunately I didn't have a chance to play through these processors. However, Bruce Bowers (who is overseeing the Pro-FX project) and Steve Salyer (marketing) made sure that a review unit would be sent to me soon so that I could write up an evaluation for Polyphony. Incidentally, readers who have enjoyed Lee Powell's articles in Device, Electronotes, and Polyphony will be pleased to know that he is heavily involved in the design of these various modules. I think the days are just about gone when engineers only designed and musicians only played; there seem to be more and more musician/engineers all the time, which to me is a very healthy sign.

It was while walking towards another section of the building that I saw an unfamiliar keyboard. The people I was with kind of hemmed and hawed until Dave Smith came by and offered to give me a demo of the new machine, scheduled to be introduced at the 1983 Winter NAMM show and named the Prophet-600. Clearly designed to compete head on with the influx of inexpensive Poly synths, the Prophet-600 features 100 user-definable programs (accessible via a keypad towards the left of the unit), five octave keyboard, two oscillators per note, built-in arpeggiation and sequencing capabilities, true polyphonic glide, six voices, and a very competitive $1995 list price. One thing that immediately struck me was that the Prophet-600 truly felt like a synthesizer, rather than a sophisticated organ (which is the case with some inexpensive Polys), and also, the sound was extremely clean -- no noise or artifacts that I could hear at all. The Prophet-600 convinced me that there is plenty of life left in at least some U.S. companies.

After Sequential, it was time to see what was cooking over at Solid State Microtechnology for Music, creators of the SSM line of music chips. SSM provides the 2044 filter chips for the Korg Polysix (among other products) and when I arrived, they had just finished torturing some poor Polysix to see if they could make their chips fail or fall out of spec (no problems ever did show up, incidentally). They were also working on an evaluation kit for their new chips which will include a PC board, the ICs, and some assorted hard-to-find components.

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They expect the kit to list for around $135; when it becomes available, you'll see ads for it in Polyphony.

When I asked "what's new", Ron Dow (author of the SSM2011 mic preamp article in the Sept/Oct '82 issue) showed me his design for the SSM2015, a new mic preamp chip. They are aiming for a noise spec so low I won't even mention it, but even if they miss it by 100% they will still have the lowest noise monolithic preamp chip on the market. It's amazing what kind of performance you can wring out of custom ICs these days, and SSM is working hard on a bunch of new products. Also in the works is a quad VCA (which seems like just the thing to design into a Poly synth) called the SSM2024, and the SSM2031, a high frequency voltage-to-frequency converter in a minidip package. It has a linear current input and offers a 10,000:1 sweep, high accuracy up to 1 MHz, and operation up to at least 10 MHz.

Eventually we left the office, and the rest of the evening turned to technical talk, sushi, and sake at the local sushi bar. This was my first time checking out sushi, and all I can say is that if you like seafood, eating at a sushi bar is one great way to spend an evening. Later on I discovered that Ron had a real affinity for video games, and after saving the earth from alien invaders, we called it a day.

Ron had mentioned during the evening that E-mu was working on a product which would interest me...and that's all he would say. So the next day, we took the hour's drive down to Santa Cruz to see what was happening (like the Prophet-600, this project was so secret I knew little about it until I actually arrived at E-mu). What I saw blew me away: the prototype "Drumulator", a drum machine employing digitally recorded drum sounds which will list for -- this is not a misprint -- $995. Considering the price, you might expect some real limitations but the only ones that I think people will object to is the lack of a crash cymbal sound, and the fact that the drums aren't tunable. But the Drumulator more than offsets any potential objections not just with a low price, but with some really neat features. These include excellent drum sounds, easy programming, programmable dynamics, and perhaps best of all, programmable mix. The mix can even change during the middle of a piece, so based on what I heard the Drumulator potentially has more of a "flesh and blood" sound to it than some more expensive units. All drum sounds may be taken out individually for processing, and there are standard features such as tape interface, operation from external sync tracks, and so on. Additionally, there are many convenience programming features such as copying repetitive segments, editing, insert and delete options for programmed segments, and lots more...all indicated in a very intelligent manner on two 7 segment readouts. Everybody at E-mu was grinning like the cat who ate the canary as they showed me their new toy -- they expect that the Drumulator will be a big success, and I don't see how it could be anything but. In fact, I think that the Drumulator will do to digital sound drum units what the Polysix did to polyphonic synthesizers: crack the market open, lower prices, and give more capabilities to musicians who want to make big sounds with small budgets. With MXR coming out with a full-feature digital drum unit listing in the under $2000 range, and Roland reportedly coming out with some kind of new programmable drum, it should be a very interesting NAMM show for drummers.

By the way, Dave Rossum asked me to spread the rumor that E-mu has some engineering positions which need to be filled; Polyphony readers with substantial digital chops might want to send their resumes to E-mu.

I had hoped to visit the folks at Syntauri on my trip, but some scheduling conflicts cropped up which made this an impossibility. Nonetheless, I talked to them on the phone and found out some interesting things. First of all, although initially some people didn't know whether to treat their products as computers or musical instruments, Syntauri has built a broad user base and is doing quite well (substantial recent financing from Steve Wozniak certainly won't hurt any future plans, either). Their most recent products include software which essentially turns the Mountain Hardware/Apple system into a 16 track digital recorder so that you may layer synthesized

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This is mostly just a trigger function. No provision is made for complete control of other parameters.

Steve Tomporowski
West Haven, CT

COST-EFFECTIVE PHASE CHECKER?

Re Craig's Phase Checker review in the May/August issue: At $450 is the Phase Checker "well worth the money"? You can check the phase of a speaker by touching a battery to the input and watching which way the cone moves. You can test mics by listening to them on headphones, one in each ear. Come on, for $450 I could get something useful!

Robert Carlberg
Seattle, WA

Robert - Sure, you can check woofers with a battery, but what about horns, piezo tweeters, and other transducers which don't move in an obvious manner? And you can test mics for being in or out of phase with respect to each other, but how do you check a single mic's polarity?

The neat thing about the Phase Checker, which if it wasn't clear in the review should be further explained, is that this one device lets you test tape recorder paths for phase (not all tape decks are non-inverting), headphones, effects boxes, balanced cable wiring, stacks of speakers, etc. Sure, if you only have one or two speakers to test then $450 is a lot. But many Polyphony readers are involved with larger studios and/or PA companies. I'm sure the idea of taking the cover off each and every speaker baffle (of which there can be dozens in a PA installation), disconnecting each woofer lead, and connecting a battery would drive them up the wall...let alone trying to check the compression tweeters and horns with a battery.

I know of no other device

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lines, as well as sync track software which is compatible with the Roland TR-808/606, TB-303, Oberheim, Linn, Korg KPR-77, etc. (it sure is reassuring to see people designing for compatibility). They will also be showing a polyphonic music printout option, which supports 8 voices, for the first time at the January NAMM show. This program also provides sync to tape; you may define the number of oscillators, and there are many subtleties in the printout program (for example, you can transpose automatically, print out certain voices selectively, and so on).

Finally, a user's group has been formed for the alphaSyntauri. Membership is $10; for information on the user's group, write SUN (Syntauri User's Newsletter), PO Box 909, Redlands, CA 92373.

I then worked my way over to CompuPro's Christmas party, which I mention only because the people at CompuPro manufacture the computer on which I edit this magazine, and certainly deserve credit for that! They've got new products too, including a 16032 based CPU board and another CPU board based on the Intel iAPX-286. Mark Garetz, CompuPro's general manager, is an avid music buff (and of course, Polyphony reader) who is starting to make some noises about coming out with music boards for S-100 machines. I hope the company stops growing long enough for him to turn these ideas into reality, because I think he would come up with some really excellent designs.

By this point, I was starting to get pretty tired but one more stop remained: the San Francisco Civic Center, to meet up again with Marvin and attend the Peter Gabriel concert. Larry Fast, who has contributed articles to Polyphony in the past and is an old friend of Marvin's, is Gabriel's keyboard player and was able to get us tickets for an otherwise sold-out show. Larry's using a modular Moog system, Fairlight CMI, micromoog with PAIA 8700 computer for sequencing, and a Prophet-5 hooked up to another 8700. Unlike some musicians who look upon the road as a chance to practice the art of dissipation, Larry likes to work on software in his spare time; in fact, this tour's project was to get the 8700/Prophet-5 combination alive and cooking, which he managed to do just before the tour came to a close.

About the concert itself, it was excellent -- state-of-the-art. The first thing I noticed was no microphones and no cords; each musician wore a wireless headset/microphone, and all instruments were wireless as well. Jerry Marotta, the drummer, used a Linn drum machine to augment his already formidable talents. Tony Levin, who also plays with King Crimson, alternated between standard electric bass, Chapman "Stick", and a Moog Source for keyboard bass parts. David Rhoades, their guitarist, played Stratocaster and had an effects rack which included an MXR graphic EQ, MXR Delay System II, and Eventide Harmonizer™. Peter Gabriel, frontman, songwriter, and the focus of much of the evening, was the quintessential professional. His range of emotion (and technical mastery) bore little resemblance to the typical "rock group lead singer". From a spellbinding introduction, to the last encore, the band was tight, hot, moving, and showed technical and musical depth. I've heard rumors that Gabriel is getting tired of touring; while I hope this wasn't his last tour for the sake of concert goers, I know all too well about the grind of the road, and can certainly see that after a lengthy career with Genesis and another as a solo artist, the record-tour-record cycle might be getting a bit stale. But there was certainly no staleness in the performance, and whatever Peter Gabriel does in the future, I'm sure it will be worth checking out.

We went backstage for a bit and chatted with the group members, but it was getting late -- Marvin and I had to make it back to his place, and the band was scheduled to play their last gig of the tour the next day in San Jose. Before saying farewell, though, Larry mentioned that a movie for which he had done the soundtrack (called "The Jupiter Menace") would be out soon, along with the soundtrack album (as this article goes to print, the album has just been released). You can also expect a new Synergy album before too long, once Larry nails down his latest recording contract. And knowing Larry, he will probably pop up on a bunch of

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available at the time of writing the review which lets you check the phase of virtually any signal generating or processing device which you can find in the typical, and not so typical, studio or PA setup. At a going rate of $100/hour for a studio (sometimes more, sometimes less), you only have to save four and a half hours for the Phase Checker to justify its list price. Of course, in practice you can usually buy it for considerably less.

So, I stand by the opinions presented in the review, including the one that $450 is fairly hefty price to pay. But until someone can come up with something better or cheaper, the Phase Checker is a valuable -- and reliable -- piece of equipment for those involved with the making of music on a professional level.

re-view

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Miles Davis **We Want Miles** (Columbia 38005). From the 1949 big band, the 1956 quintet and the 1964 quintet to the present septet, Miles' real strength has been as a chemist, putting together just the right sound for the times. He's been hot for nearly 40 years and that's cookin'.

Stitching Small Tears (cassette). A 90 minute compilation of 12 Vancouver B.C. electronic musicians. Styles vary widely from rhythmic electropop to hardcore tape manipulation (with emphasis on the former), but quality remains excellent throughout, including packaging. $10 (Canadian funds) postpaid to U.S. from C.L.E.M./Alien Soundtracks, PO Box 86010, N. Vancouver, B.C. "7L 4J5, Canada.

Port Said **Through Veils** (cassette). Expanding on their single

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other people's albums as well in his capacity as a studio musician.

Conclusions. If there's one thing Polyphony readers should know, it's that the people I was with during the week of my trip were some of the nicest people you would ever want to meet. There was little evidence of the dog-eat-dog mentality so prevalent in other businesses. On the contrary, one company even went so far as to warn a competitor that they were coming out with a product which would probably reduce their competitor's sales, and that the competitor would be well advised to enhance their existing product in order to maintain their share of the market. That kind of gentlemanly behavior is hard enough to find anywhere nowadays, let alone in an industry which, frankly, has suffered from the recession and foreign competition almost as much as steel, automobiles, or other U.S. industries.

So there you have it -- a week in the life of a Polyphony editor. It was great to get away from the word processor and deal with some real humans, and it was even better to realize what a special field we're in, and that our fellow travelling companions are the kind of people you'd just love to spend an evening with talking over the future of music. If there's any overriding conclusion I came to on this trip, it's that ingenuity is alive and well in the music industry, and that surely, the best is yet to come.

re-view

(reviewed May/Aug '82), side A contains the live Voyage 1 of "Indian Ocean" plus another two live tracks from 1981. Side B has a trio of 1982 collaborations with guitarist Cliff Cultreri, showing an eagerness to work with different contexts. They are definitely a group to watch. Write 132 West 24th St., New York, NY 10011.

Rudiger Lorenz **Wonderflower** (cassette); **Silver Steps** (cassette). Some of the best music on record isn't on records. Like his first effort (see May/Aug '82), these two tapes feature real-time recordings, large self-built modular synthesizer, and dedications to his friend Sabine (whose legs adorn "Steps"). $5 each postpaid (really?) from Lorenz c/o Syntape, Binger Str. 6, D-6507 Ingelheim, West Germany -- and ask for a photo of his studio.

Brand X **Is There Anything About?** (Passport 5006). Brand X is a chance to blow off steam for some of England's top session musicians. It's loose, relaxed, mostly instrumental jazz-rock with no holds barred, and this one is a bit more so than their last couple of albums. I wonder if Dave Stewart minds their borrowing his Prophet without asking.

Gary Numan **I, Assassin** (Atco 90014-1). Numan gets the beat, puts together a hot rhythm section (drums, percussionist, and fretless bass), and comes up with a very danceable album. Inspiring synthesis and nasal vocals sign it as his own.

Blade Runner (Warner Bros. 23748-1). Due to some contractual screw-up, this is not the original synthesizer score by Vangelis, which is a shame because it was one of the best things he's done. Instead, here is the official album of "orchestral adaptations" of his music - "Switched-Off Vangelis", as it were.

Spyro Gyra **Incognito** (MCA 5368). There are several synthesizers listed, but they must be traveling incognito. Mostly this album is Jay Beckenstein's punchy sax lines over Hollywood soup. Unfailingly professional.

Eberhard Schoener **Video Magic** (Harvest 12171). Schoener has done a couple of records integrating elements like Gregorian chant or Balinese gamelan into electronic music. This one, merging with popular rock group The Police, seems to be straining to find middle ground and goes off in all directions.

Conrad Schnitzler **Conal** (Uniton 002). After several undistinguished albums (see May/June '81, Jan/Feb '82, and Sept/Oct '82), the king of non-keyboard synthesis is back to his old tricks. Multiple layers of "VCS noises" - oscillator swoops, birdy tweets, and raindrop sequencing - are mixed in and out for an abstract and totally electronic result.

AD INDEX

Lab Notes:

Shepard Functions

By: John S. Simonton, Jr.

The first audio illusion I ever heard was the Shepard Tone. Maybe you know it by the more descriptive term "barber pole" tone. It got that name because, like the stripes on a barberpole, it seems to defy the old saw "what goes up must come down". The effect is that of a continuously rising (or falling) tone which never resolves.

How the Shepard Tone works. There is nothing very mysterious about the Shepard Tone, as disconcerting as it can be at first, and if you've worked with synthesizers for a while you can figure out pretty quickly what's going on. The spectrum of the tone consists of a large number of octavely related components, all stepping up-scale together. The harmonics at the high and low ends of the spectrum have relatively low amplitudes, while harmonics in the middle of the tone are at maximum amplitude.

Imagine for a moment that you are following the lowest harmonic that makes up the Tone. At first the amplitude of this component is so low that it is, for all practical purposes, inaudible. But as it steps up-scale its level increases, peaking when its frequency corresponds to a point midway between the high and low limits. After peaking, the amplitude decreases as the frequency continues to step higher until finally, at the upper frequency limit, the harmonic is again inaudible.

When the harmonic reaches the high frequency limit it disappears, only to be replaced by a new harmonic at the lower limit.

Since the eight or ten harmonics which make up the Tone are all rising in a "staggered" progression, each in turn starting over again as it reaches its upper frequency limit, the overall effect is that of a tone which is constantly increasing in pitch while not actually getting any "higher" (or lower if the tones are all falling). It's an interesting illusion.

Shepard's original work used a computer program written by Max Matthews, but the same type of effect can be accomplished using analog synthesis equipment controlled by a gadget which, for lack of a better name, we may as well call a "Shepard Function Generator".

The Shepard Function Generator. Thinking about what happens with the frequency and amplitude of each harmonic of a Shepard Tone makes it easier to understand the composite sound. The frequency increases constantly and linearly from low to high, until the higher limit is reached. At this point, it begins again at the lower limit. This is a ramp function. The amplitude of any harmonic increases from the lower limit until it reaches the middle frequency, and then decreases as it approaches the upper limit. This can be a triangle function.

So, we need a gizmo which will produce a bunch of ramp waveforms (eight is a convenient number), and an equal number of triangle waves. The ramps and triangles both must have fairly precise phase relationships to one another, as summarized in the diagram in figure 1.

In the interest of conserving drawing time and space, I have shown only four of the eight functions pairs that our device will generate. I'm sure that you can see the pattern, and that the missing odd functions fit between the even functions that are shown. Notice that each function pair in the complete series is 45 degrees (pi/4 radians) out of phase with each of its neighbors. The even pairs shown are 90 degrees out of phase with one another.

Now, there are almost certainly lots of possible analog ways to generate these function

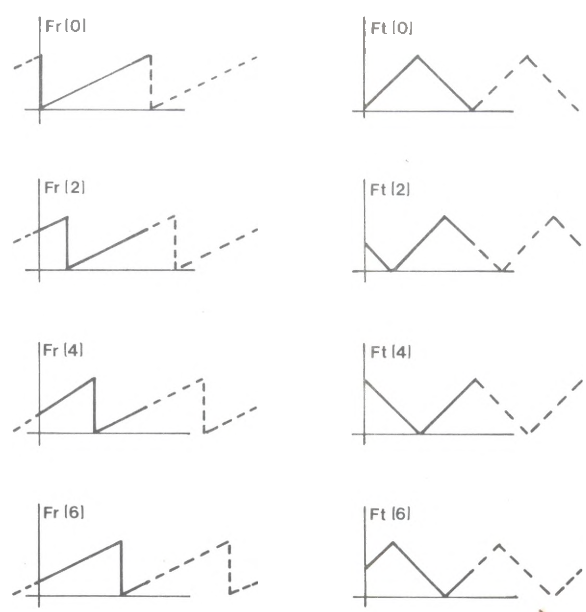


figure 1

pairs. But the simplest circuits I can think of to do this are too simple (for example, they wouldn't be able to generate the functions over a very wide frequency range), and what complicated approaches come to mind are very complicated. But I know of some workable digital approaches and I'd like to show you one, a discrete logic machine.

The day will soon be here when we wouldn't even discuss a logic machine approach to this problem. We would just truck ourselves down to our local electronics store and pick up a blister packed Single Chip Data Processor to be programmed on our trusty Home Data System.

No doubt, but let's look at a way to do essentially the same thing with counters, DACs, MUXes, and such...things we can get today. Figure 2 shows some components which we'll use in the Shepard Function Generator (as you probably realize, a counter connected directly to a DAC generates ramps). If the counter is counting up, upward sloping ramps come out. Having the counter count down, or inverting the counter output before it gets to the DAC, produces downward ramps (see figure 3).

Consider this: A triangle may be thought of as a ramp which changes its mind halfway up. If we replace the inverters in the figure above with Exclusive-OR gates, we can produce a single logic input that when high, causes the DAC to produce an upward ramp and when low, causes a downward ramp. By using the most significant bit of the counter as well as the control signal to the EX-ORs, the digital input to the DAC will count up until the MSB goes high, then it will count down -- in other words, a triangle function (see figure 4).

If we're interested in generating only a single function pair, it's a simple matter to pick up a new Least Significant Bit on the counter and use it to effectively switch back and forth between the circuitry of figures 3 and 4, producing first a small section of the ramp, and then a small section of the triangle. This new LSB also switches between two sample-and-hold circuits to de-multiplex the composite output of the DAC. Figure 5 shows how a little more logic gives Fr(0) and Ft(0).

I'm sure that we're together so far, and to make sure that we stay together I should mention a useful way to think of the eight Most Significant Bits of the counters. Think of them as phase, summarized in Table 1 below.

TABLE 1

Counter Output binary	hex	Equivalent Phase
00000000	$00	0 degrees
00100000	$20	45 degrees
01000000	$40	90 degrees
01100000	$60	135 degrees
10000000	$80	180 degrees
10100000	$A0	225 degrees
11000000	$C0	270 degrees
11100000	$E0	315 degrees

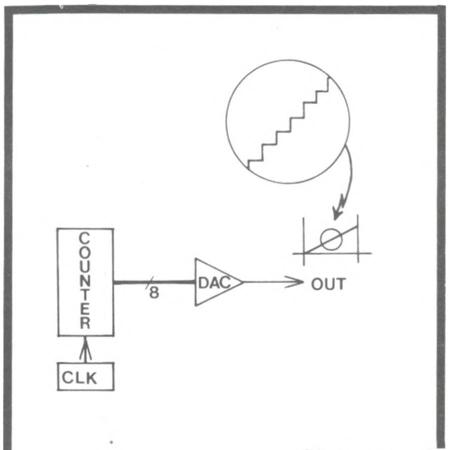


figure 2

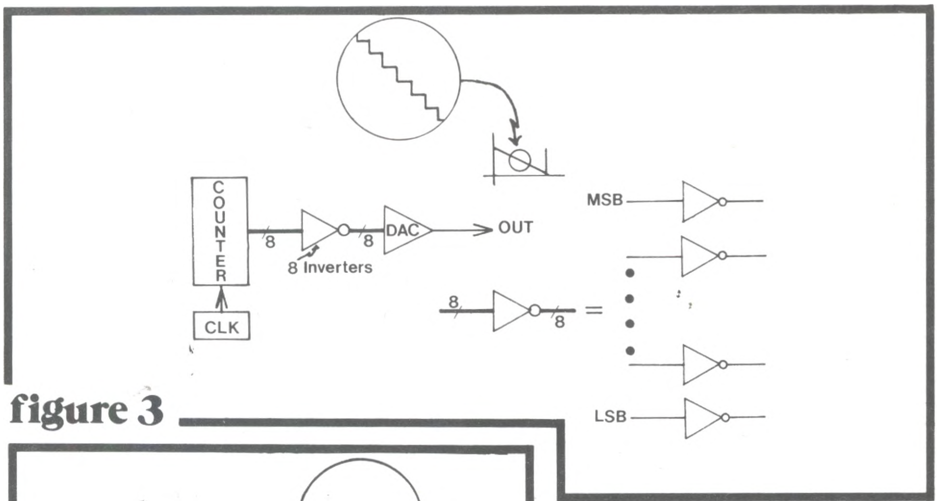


figure 3

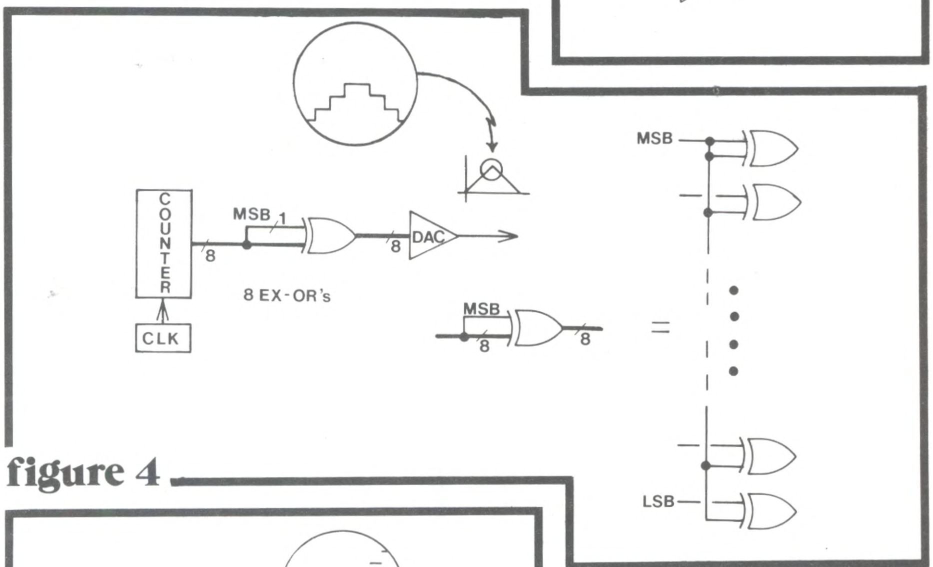


figure 4

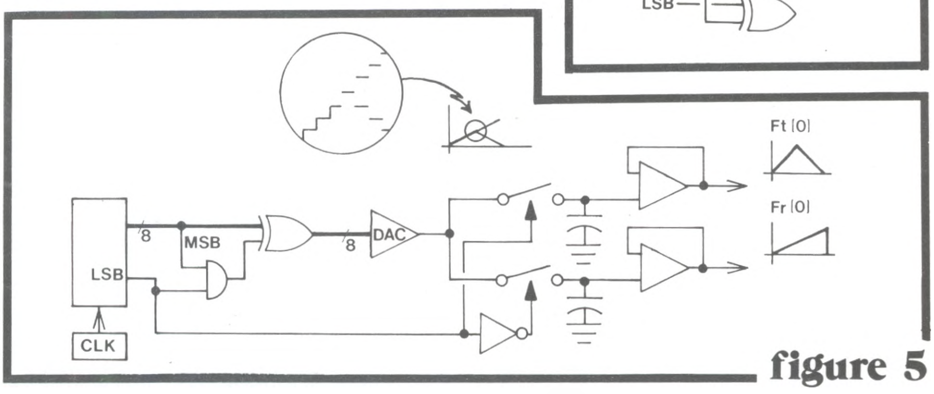


figure 5

If you're more comfortable with a graphic representation, see figure 6. The benefit of thinking of the counter data in this way is that phase shifts are produced by simple additions. For example, to shift the phase of the waveforms produced by the counter and DAC by 45 degrees, simply add $20 to the output of the counter. This is a pretty handy thing to know, particularly when it just happens that we are looking for a way to generate eight sets of functions which are 45 degrees apart.

Figure 7 shows a block diagram of the complete Eight Phase Shepard Function Generator which results when we include an added IC to calculate digital phase offsets and de-multiplex the output with 8/1 analog switch ICs; figure 8 shows the schematic for the complete Shepard Function Generator. In the same way that the circuit of figure 5 alternately generated pieces of Fr(0) and Ft(0), the Shepard Function Generator sequentially puts out pieces of Fr(0), Ft(0), Fr(1), Ft(1), Fr(2).......Ft(6), Fr(7), Ft(7).

Details: Starting from the Most Significant end of the counter, the first eight bits of the counter serve the same functions that they did in the warm-ups. And we've decided to think of that function as phase. Unlike the previous sketches, these phase bits are broken down into two groups: the three Most Significant and the next five. If you don't see the significance of this grouping, review the binary representation in Table 1. To produce 45 degree phase shifts, the three Most Significant Bits are the only ones which change.

Below the eight phase bits, you'll see another grouping of three bits. Think of these as "offset" bits and notice that they are what's added to the three Most Significant Bits by the adder. And note that the offset bits also serve as address bits for the De-Muxes so that any given phase offset always gets strobed into the same sample-and-hold.

The next "Less Significant" bit can be thought of as switching back and forth between ramps and triangles, as in figure 5. And since figure 7 is a block diagram of our working Shepard Function Generator, and not just theoretical like the previous figures, the Least Significant Bit of the counter serves as a strobe which allows time for the DAC to settle

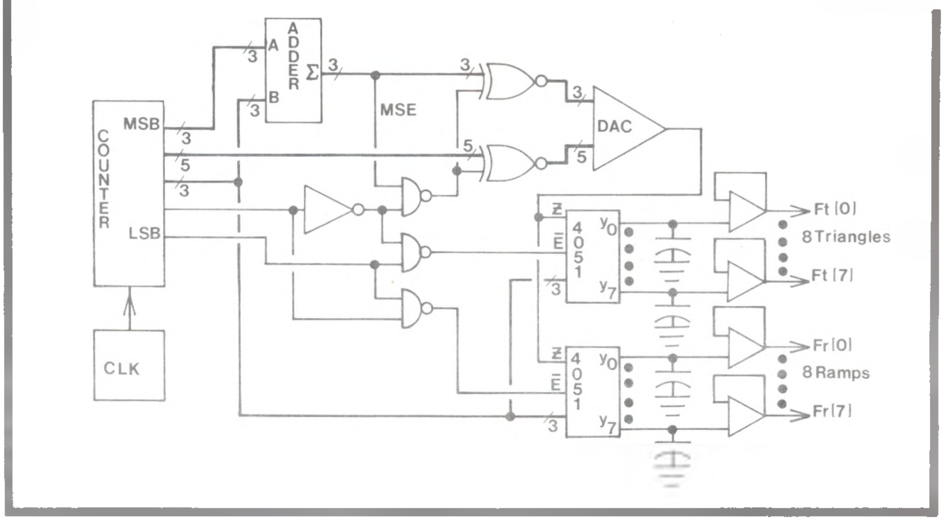


figure 6

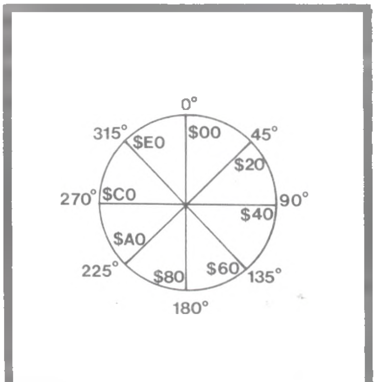
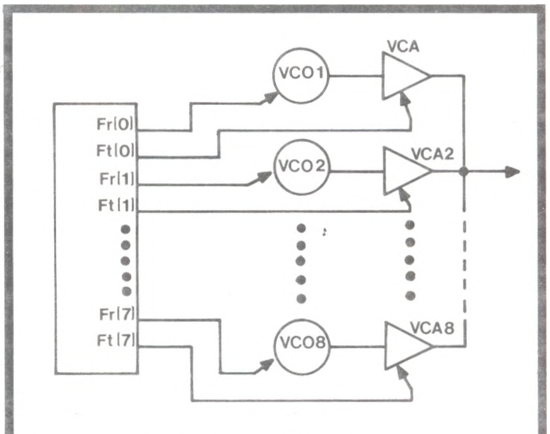
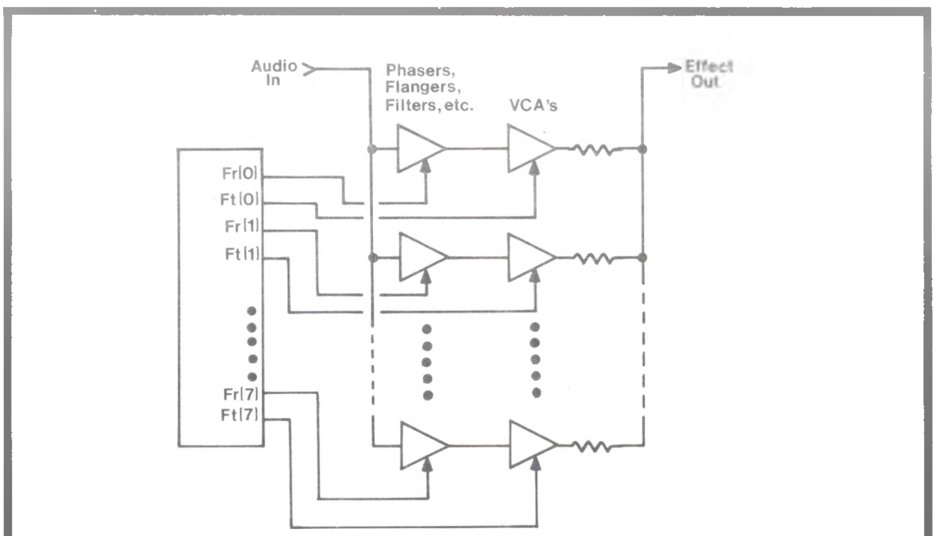


figure 7



VCO's and VCA's controlled by SFG produces shepard tones.

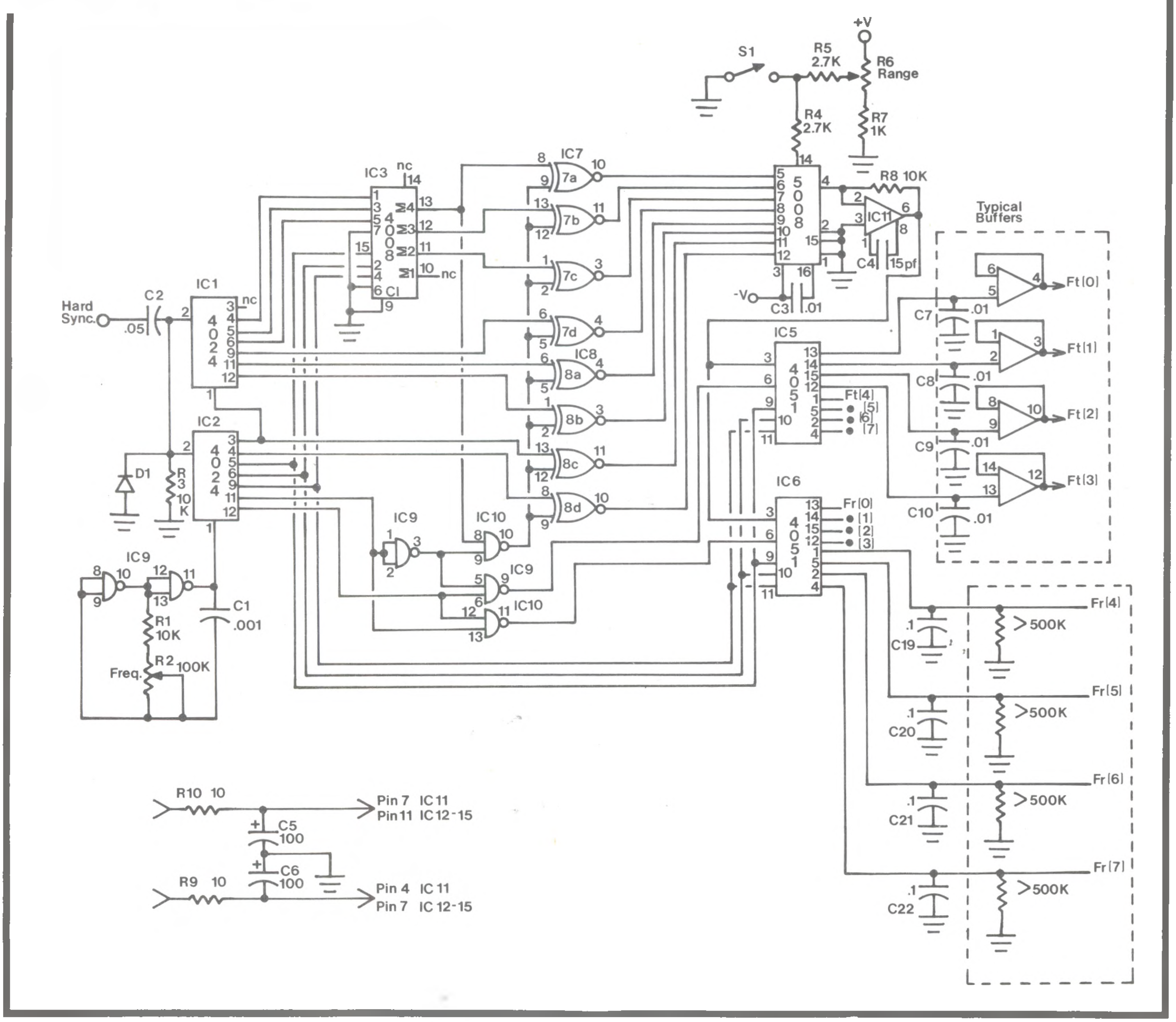
figure 8



Signal processing elements may be substituted for VCO's to produce interesting effects.

figure 9

figure 10



before selecting a sample-and-hold.

So, the Shepard Function Generator which we've developed isn't simple (though I would like to think it has a certain elegance) and when you consider that we'll also need eight VCOs and eight VCAs to produce the Tone (see figure 9), you might question if it's really worth the hassle.

But Shepard Tone generation is not the only application for this circuit; recently the same principles have been applied to other areas of electronic music. For example, the Barberpole Phaser invented by Harald Bode is a signal processing device which substitute phase change components applied to an external signal source for the frequency components of the Tone. The characteristic of multiple phase shifters are controlled by Shepard Functions so that the phasing effect doesn't simply swing back and forth, like we're used to hearing, but rather sweeps up and down eternally. It's really a most unusual effect and if it has occurred to you that the same principle might also work with other processing elements (such as filters, maybe) you're on the right track.

Figure 10 is the configuration such approaches would customarily take, with the ramp functions controlling the parameter being modified (phase, corner frequency, time delay, etc.) and the triangles controlling VCAs to fade the output of each modifier in and out.

You may have noticed that we're using gobs of equipment...lots of phasers or flangers or whatever. Chances are that you don't have eight flangers laying around. Even if you us the least expensive modifiers available (PAIA's EKx module series, for example) you will still have some bucks tied up in repetitive elements. For those who lament the fact that there don't seem to be any new effects, this one qualifies. It's unique all right, but worth the cost...?

Wait. We're being prejudiced by what we've seen so far (always a danger). We're thinking of the Shepard Function Generator only as a way to generate monophonic, noncyclic illusions by always using all 16 output functions to control 16 corresponding processing elements. But that's where we're getting of the track: You don't have to use all the outputs all the time, and the results don't have to be monophonic.

Now, there's no doubt that eight phase Shepard Functions are the absolute minimum number of components which will still preserve the "barberpole" illusion, but there are other times when sets of phase synchronized functions are useful. Is it obvious that any pair of triangles 180 degrees apart -- Ft(0) and Ft(4), for instance -- may be used with a pair of VCAs to give automatic stereo panning? Or that four triangles 90 degrees apart provide quad panning? With the arrangement shown in figure 11, the apparent "revolution" of the sound source is clockwise. To reverse the apparent direction, reverse either pair of corner sources.

Various combinations of triangles with unequal phase relationships may be used to produce effects which don't just swing round and round, but rush out of one of the "corners", swing around in front of (or behind) you to disappear into the other corner. When you start adding effects into this setup (such as phase shifters) under control of the ramps, as shown in figure 12, the sound really begins to move around you in some strange ways.

A nice thing about this is that the effects devices don't all have to be the same to produce interesting results. In fact, some of the most interesting results come from using completely different effects (such as phaser and echo) in opposite corners with only VCA processing on the other corners. While you might be hesitant to rush out and buy eight VCOs just to get a tone, you probably have enough modules or effects to get started. Voltage control is obviously preferable, but even effects which have only manual control are useful. Among other things, be sure to try synchronizing the frequency of the effects oscillator to the frequency of the Shepard Function Generator.

I think you get the idea: Play. Try different effects and different functions applied to different effects. Try controlling the VCAs with the ramps and the effects with the triangles -- try leaving out the VCAs altogether. Not all of the results will be particularly pleasant, but you will surely also find some that are unique beyond words. While many of these effects are somewhat less spectacular when done in stereo, they are still very effective.

This is getting long just when I could go on forever; but it has to end as soon as I draw your attention to the hard sync input. A positive pulse applied to this input resets the counter chain and causes all functions to start from the same known point. This feature will be particularly useful to us as Craig Anderton introduces us to Synchro-Sonic techniques in future issues of Polyphony.

No new effects indeed!

The following is available from PAIA ELECTRONICS, INC., P.O. Box 14359, Oklahoma City, OK 73116, (405) 843-9626:

Experimenter's kit of circuit board and electronic components (does not include hardware or jacks. specify No. EK-9 Shepard Function Generator Experimenter's Kit. $24.95 postpaid.

Etched, drilled and legended circuit board alone for the Shepard Function Generator Specify No. EK-9pc, $4.95 postpaid.

Visa & Master Card accepted ($10 minimum charge) or include check or money order with order.

PRACTICAL CIRCUITRY

CONTINUED FROM PAGE 27

tended to sample the noise source under LFO control quite frequently. So the scheme reflects this fact. Sometimes, however, it is fun to sample the VCO ramp wave output, in which case I can simply patch the VCO into J1. It's as simple as that!

So there you have it: my "top secret" patch-over scheme. I want to emphasize that my small synthesizer was completely homebrew, so I had total freedom to use any arrangement I wanted. You have that freedom too. If after looking these figures over you see something you want or don't want, get into the DIY spirit and do it! One tip, though. It took me several days of thought, rough sketches, and the like to arrive at this scheme. You should do the same. Mull it over, contemplate, meditate, or whatever. Some arrangements just don't suggest themselves in one sitting.

As mentioned in the opening paragraph, we often get trapped into thinking microcosmically, and many times trying to get the "big picture" is very difficult. I hope, then, that this installment of "Practical Circuitry" has given you an idea of one possible system design. If you have some ideas for a master system, be sure to write me c/o Polyphony.

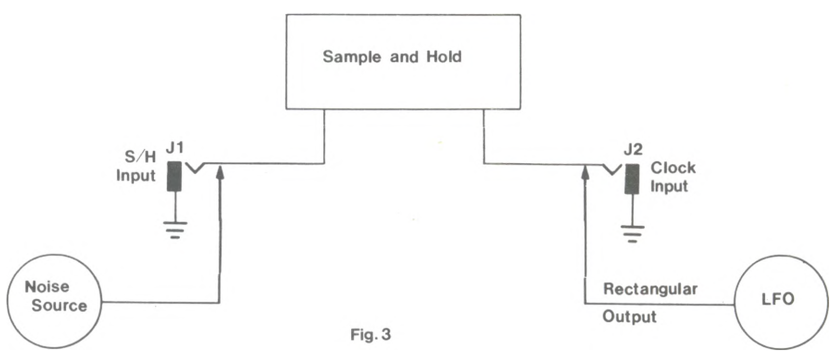


Fig. 3
Sample and hold patch-over scheme.

Bus Distribution Modules For Modular Synthesizers

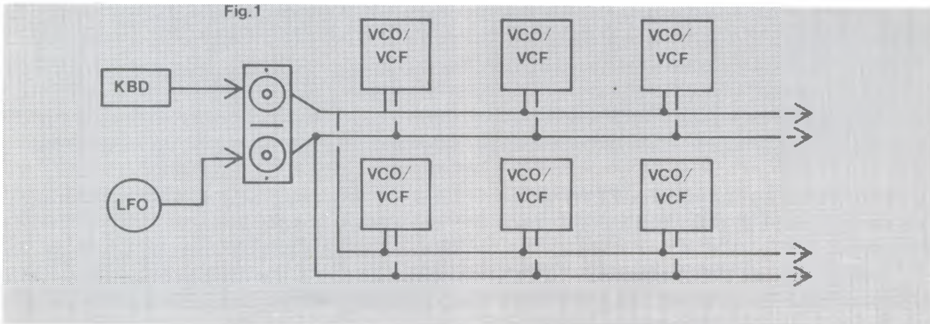
By: Charles Lauria

Back in the days when my modular synthesizer included a "one-of-each" assortment of modules, patches were a cinch to cook up. But as it grew, patching started to get a lot more complicated. The number of patch cords I used doubled and even tripled as I began to discover the beauty of using multiple VCOs, VCFs, and ADSRs within a patch.

Gary Bannister offered some super solutions to this patch cord problem in his article "Eliminating Patch Cords Without Eliminating Capability" in Polyphony Vol. 3, No. 1, July 1977. And although Gary's ideas are extremely useful and simple to build, a couple of points went unmentioned in his article, which spurred me to take a slightly different course. The best part is that both approaches are compatible, and I would advise anyone interested in this article to refer to his as well. Perhaps you'll come up with a personal combination of your own that would suit your particular system.

Providing a 2nd control voltage for modulation. One function Gary's article left out was a way to easily provide a second variable control voltage to all of the VCOs/VCFs for modulation. After all, if you're controlling multiple VCOs from one keyboard control voltage (CV) through a transposer, and you wanted frequency modulation, chances are you'd often want all of the VCOs to be modulated together while still maintaining your transposed intervals during the modulation.

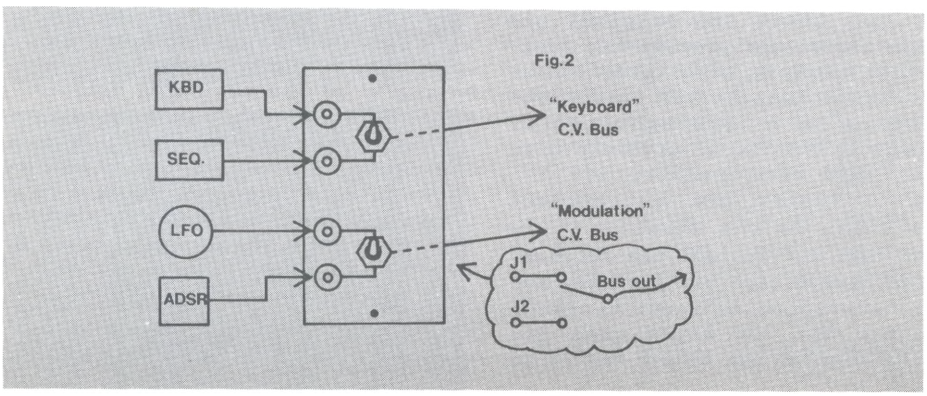
The simplest way to do this uses only one standard size PAIA single-width panel to control any number of VCOs and/or VCFs within the same cabinet (if there's another cabinet involved, you'll have to provide a jumper between the two for the control voltage). I did this by finding room on the individual modules for the transposing pots so that my "Bus Distribution Module" merely has to provide an input to the bus lines.



There are several ways to find room on the panel for extra pots. For example, on the PAIA 4720 VCO, you could replace the two existing pots with two dual pots to add the necessary two transposing pots to the module, or you could use miniature pots and squeeze them into a vacant spot on the panel. The choice is up to you; as for me, I completely redesigned the panel layout/graphics of the modules in question to accommodate these modifications and made new front panels - but that's another story.

The essence of the Bus Distribution Module (BDM for short) is simply an input jack for each of the bus lines you need. In the case of the VCOs/VCFs I wanted two bus lines: a "keyboard" CV bus to go to the first CV input of all VCOs/VCFs in my system, and a "modulation" CV bus to go to the second CV input of all VCOs/VCFs. (As you will see, these bus lines don't have to be used exclusively for the keyboard and modulation control voltages. They could both be used for modulation/special effects or whatever you want. My labels reflect the most common application.) Figure 1 shows the basic BDM setup. On my module, however, I decided to have two switch-selectable input jacks for each of the bus lines so that either of two control voltages could be selected for each of the first two control voltage inputs of all the VCOs/VCFs.

In the configuration show in figure 2, the frequency of all the VCOs/VCFs could be set by either the keyboard or the sequencer, and could be modulated by either an LFO or an ADSR. In practice, this switching capability allows control of the synthesizer to be dominated by either of the two completely different sources. For example, you could be playing the lead voice of a composition with the keyboard and LFO in control, and then switch to a sequencer/-ADSR-controlled bridge or riff at some point in the piece, and then



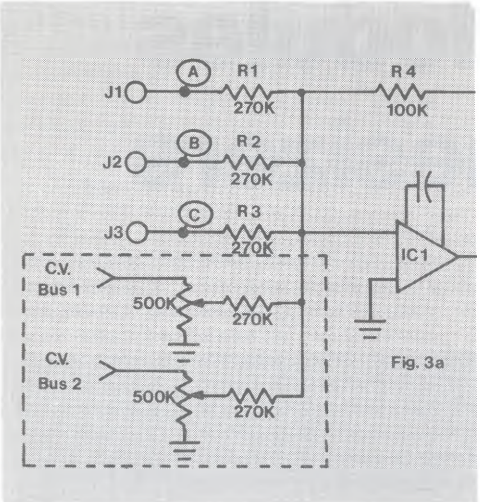


Fig. 3a

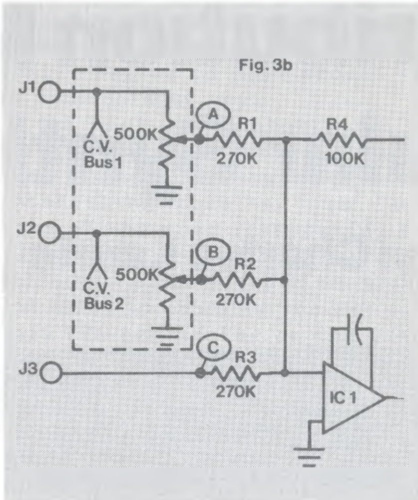


Fig. 3b

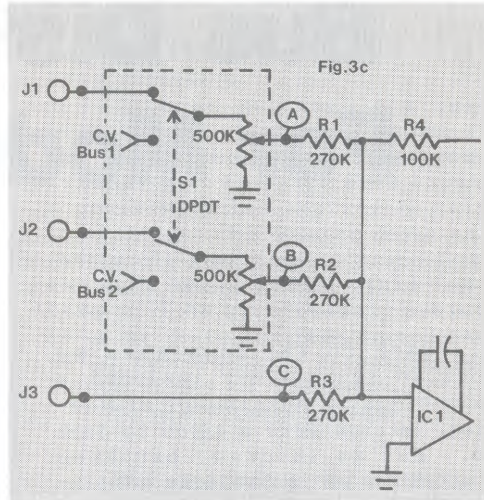


Fig.3c

switch back to keyboard/LFO control...all with the flick of two switches.

Adding attenuators. With those modules controlled by the bus lines, we will need some sort of front panel attenuator for individual transposition of the original control voltages. In the case of the 4730 VCF, I didn't need an attenuator because the initial frequency control serves the purpose nicely (although it controls the sum of all three control voltages rather than each one individually). However, with the VCOs, I chose to put two CV attenuators on the front panel, a separate one for each bus line going to the module. The reason for this extravagance is because of the critical nature of pitch, and the fact that you will often want to transpose the modulation voltage to match the transpositions of the keyboard control voltage.

For the VCO attenuators I used 500k pots with the high end connected to the CV source, the low end connected to ground, and the wiper connected to the CV input resistor. What do you do with the original pin jacks on the module? You can do one of three things:

1. Leave the original pin jacks intact and provide two additional summing resistors for the two new attenuated CV bus inputs (figure 3a - although five CV inputs to the VCO is probably more than you'd ever need);

2. Connect the unattenuated bus lines to the first two pin jack lugs and install the attenuators between each pin jack lug and its corresponding input resistor (figure 3b), in which case you must remember not to use the first two pin jacks and bus lines simultaneously; or

3. Install a DPDT switch on the module to select either the two bus lines or the first two pin jack inputs. This was what I did (see figure 3c).

Dealing with the filters is much easier, since we don't need additional attenuators. See figure 4 for the modification.

When the switch is in the "normal" position, all three pin jacks will be operational in both the VCF and VCO modifications, and in the VCO version, the two attenuators will be operational whether the switch is selecting the bus lines or the pin jacks. Therefore, even in the normal mode, you'll get the added advantage of having two built-in transposers on each VCO in your system!

Trigger bus distribution. Trigger bus distribution is another point which was not mentioned in Gary's article. The principles are the same - but there are some differences in how the trigger source is selected on the AD/ADSR panel. Notice that with the VCO/VCF modification, selecting the CV bus lines automatically disconnected the pin jack inputs from the module because of the way the double throw switch works. With the ADSR/AD modification, the "trigger select" switch will be a DPDT center-off type, for the reasons presented below.

Rather than the switch merely turning the trigger buss system on

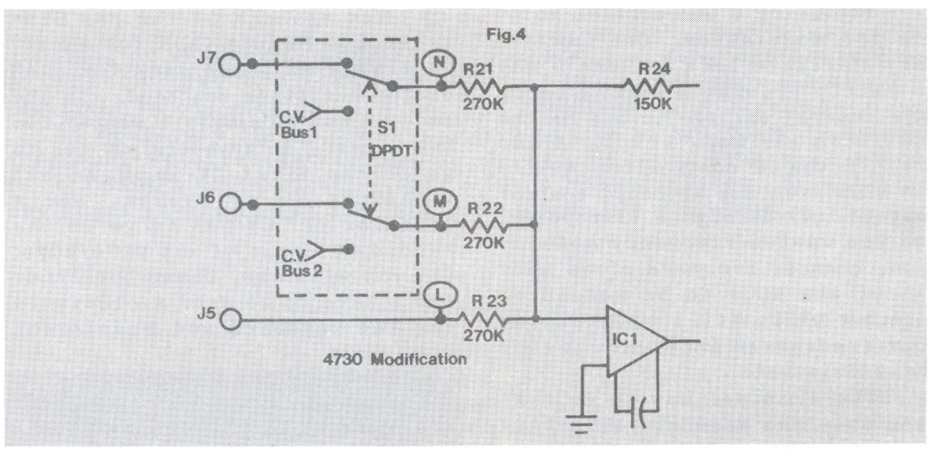


Fig.4

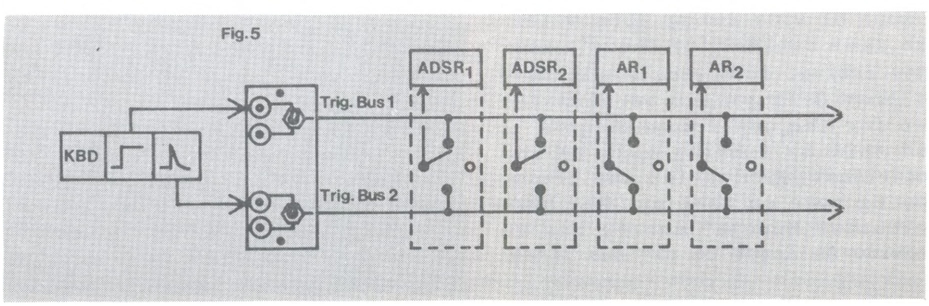


Fig.5

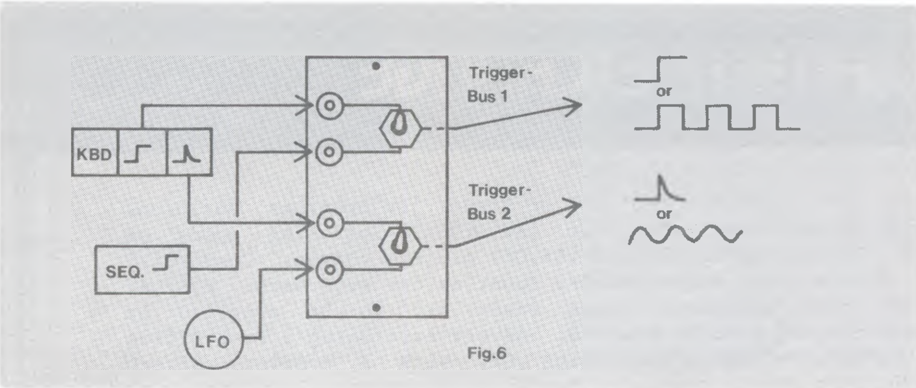


Fig.6

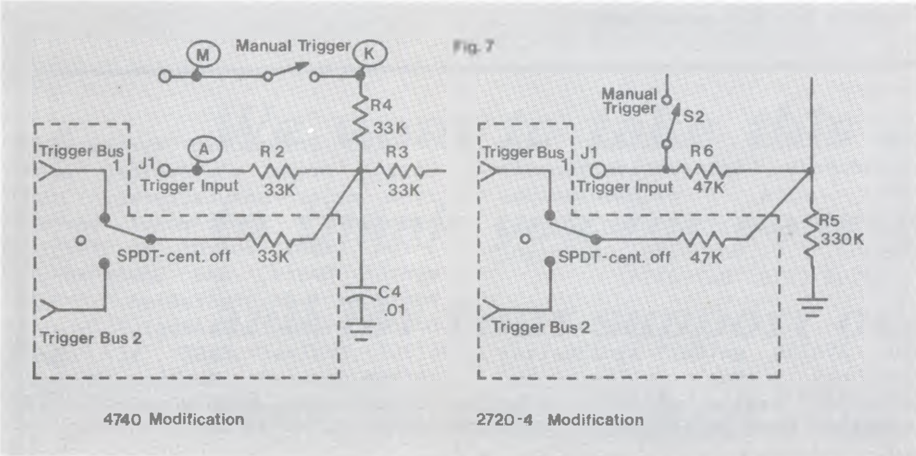


Fig. 7

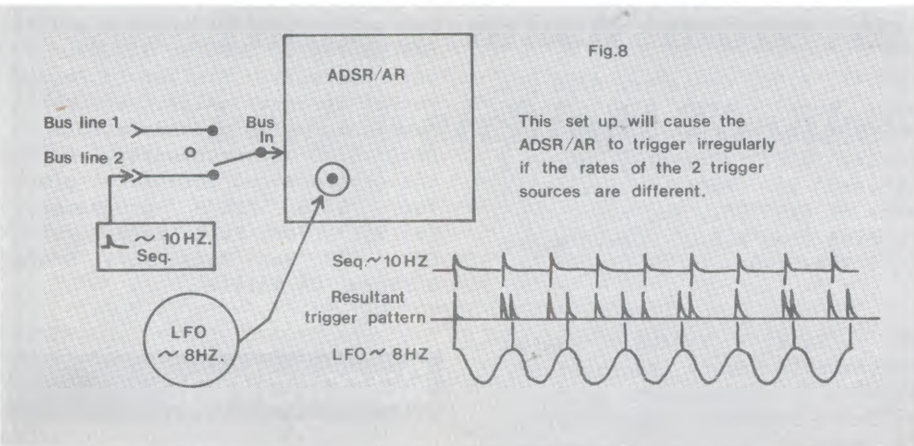


Fig.8

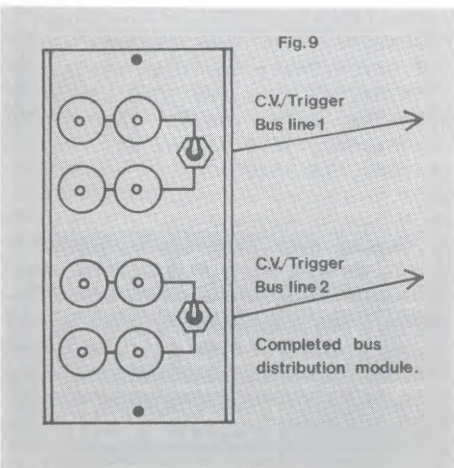


Fig.9

or off as with the CV bus switch, I found that it would be useful to be able to select one of two trigger bus lines to fire each ADSR/-AR. In this way, for example, certain ADSRs/ARs can be triggered from one source. say the step trigger from the keyboard, while others on the bus are simultaneously triggered by other sources (LFO, pulse trigger from the keyboard, etc.) as shown in figure 5. But wait...note that the Bus Distribution panel still has dual inputs itself, allowing each of the two trigger bus lines to carry one of two triggers for a total of four types of triggers available at the ADSRs/ARs (figure 6).

Figure 7 shows the way the bus connects to the modules. When the switch is thrown in either direction, one of the trigger bus lines will serve that module. But when the switch is in its center position, both trigger bus lines will be disconnected from that module. As you can see from the diagram, the trigger input pin jack will remain operational regardless of what position the switch is in. For this reason, I decided it would be best to add a resistor in series with the trigger bus line at the module, forming a sort of "trigger summing network" similar to the control voltage summing networks in VCOs and VCFs. In this way, two triggers can be summed and used simultaneously to trigger and ADSR/AR, permitting, for example, irregular triggering of the ADSRs (see figure 8), "pick and repeat" banjo envelopes, and other effects used in creating complex sounds.

Final comments. When using the Bus Distribution Module there will be times when it would be handy to have access to the original control voltage or trigger for use with other modules not hooked into the bus system. For example, when using the keyboard pulse trigger on the bus line, you may want the same keyboard pulse trigger to also step the sequencer which is not on the trigger bus line. Rather than using a multiple jack panel to branch off this extra trigger, I added a second pin jack in parallel with each of the original input jacks on the Bus Module. This isn't really necessary but is sure is a blessing when doing gigantic patches. Figure 9 shows the final version of the Bus Distribution Module as it appears in my system. I have two of these modules, one for my VCO/VCF bank and one for all my ADSR/ARs.

I hope you enjoy constructing and using this module in your system. And if you should run into any problems with it, contact me c/o Polyphony. By the way, all of the parts can be easily obtained - I found sub-mini toggle type switches at Radio Shack but I'm sure that most electronics parts stores will have what you need. Good luck!

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