POLYPHONY T.M.

ELECTRONIC MUSIC & HOME RECORDING

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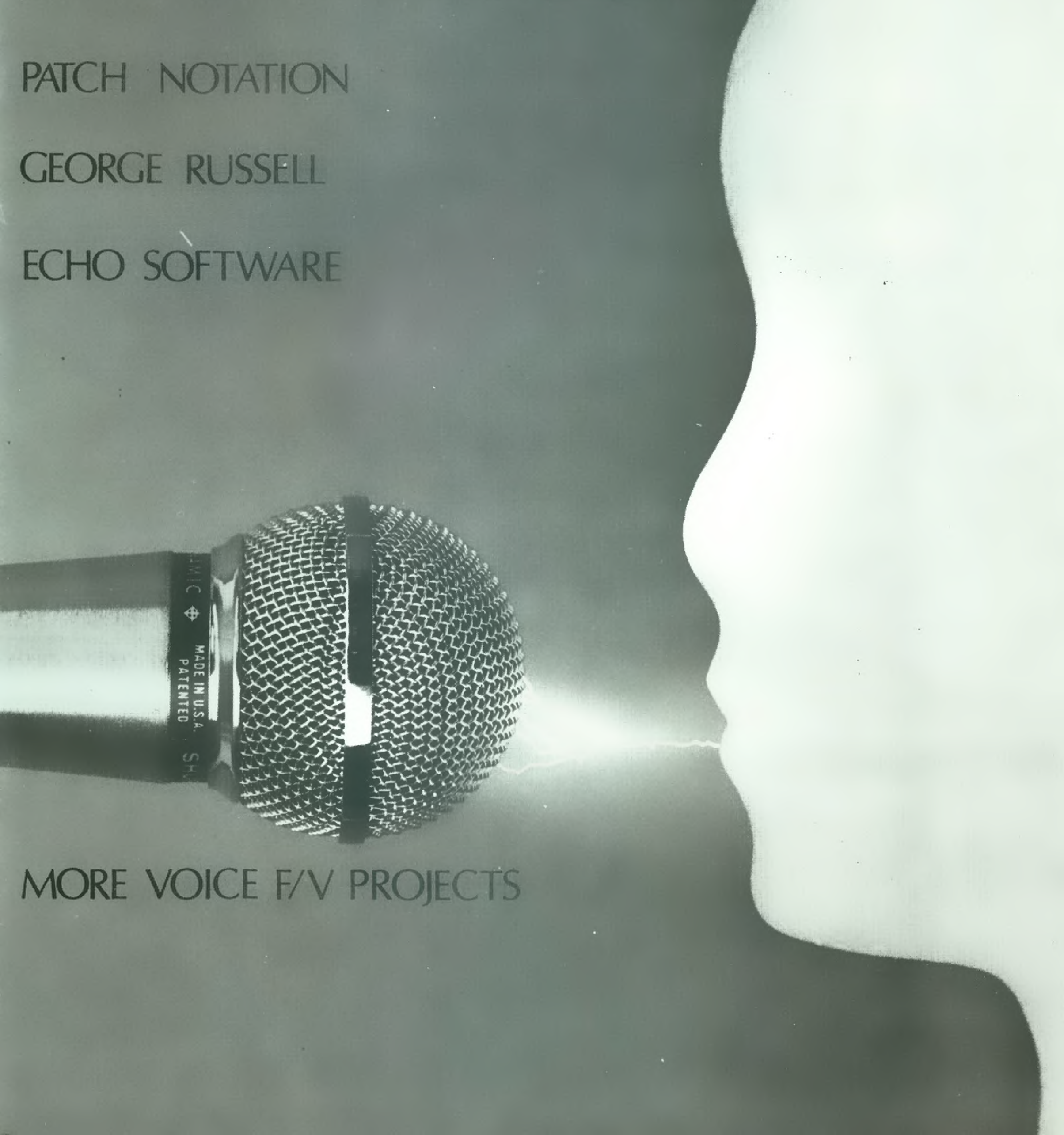
1979

PATCH NOTATION

GEORGE RUSSELL

ECHO SOFTWARE



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Linda Kay Brumfield

CONTRIBUTING EDITORS
Craig Anderton
Gary Bannister
David Ernst
John S. Simonton, Jr.

PRODUCTION
Jarice Kirkendoll
Donald Cooper
Marcina Howard

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LETTERS...

AUTHORS ADDENDA

Upon re-reading my article in the last issue of Polyphony, it occurred to me that several points in the article might be clarified.

(1) There is a hidden assumption in the "rhythm-tuning" method which should be made more explicit. As one can see from the chart, the frequency ratios are all ratios of integers, which means that they are harmonic series intervals. The most accurate method of tuning up the sequence, therefore, is by zero beating the sequencer controlled VCO against harmonics of a reference signal. The intervals thus obtained are not (except the octave) equal to the intervals of equal temperament; hence the cautionary note about quantized sequencer outputs in the article.

As an extension to the above, it can be seen that the "1/4 tone" detunings given for the Oberheim mini-sequencer are not true 1/4 tones. Short of calculating the exact frequency ratio involved and using a frequency counter, the appropriate method would be to approximate the given interval by ear. In my experience, this works well enough.

(2) The method outlined in the article, although generally a solution to rhythm tuning, does in fact have certain limitations. This became obvious to me upon rereading the discussion of sequencer rythmn tuning in The Technique of Electronic Music, by Thomas Wells and Eric Vogel. Wells and Vogel give an example of a rather intricate rhythmn and describe a method for tuning the sequencer pots which requires a voltmeter. While it is possible to calculate a series of frequency ratios using my method, which would theoretically work for their example, it would be totally impractical to tune the resultant pitch sequence. For those whose music is limited to a restricted set of relative note durations (i.e. most pop music), my method works quite well and does not require the purchase of a voltmeter; for those who do not wish to be so restricted, I would suggest the purchase of the Wells-Vogel book and a DVM.

John Duesenberry
Boston School of
Electronic Music

MONKEY BUSINESS

One thing concerning the ape-peggio synthesist on the cover of the Sep/Oct issue: This reminds me of the animal trainer who had his chimp smoke a cigarette to show the audience how 'silly' they look. Come to think of it, a monkey could play better than the present AM fare.

Robert J Abend II
E. Syracuse, NY

LOST GNOME

I've run into a REALLY strange problem, and could use some advice and/or help. Back in late October I received a Gnome for finishing assembly and repairs-- I think. The box contained no letter of explanation and, worst of all, no return address.

The Gnome has been very distinctly customized; the board and controls have been mounted in a custom wood and plexiglass cabinet that more resembles a computer console. Also included was a Godbout power supply. The cabinet has some very nice custom graphics on it-- white on smoked plexiglass.

There was a receipt for the graphics enclosed. An effort to locate the owner by way of the Admad Company (the address on the graphics receipt) has failed. I have no more information except a Sunnyvale, CA postmark. The unit was mailed Parcel Post Uninsured, by the way.

I'm hoping that the readers of Polyphony may be able to help me find the lost Gnome's owner. Any ideas?

Gary Bannister
7208 New Augusta Rd.
Indpls., IN 46268

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for
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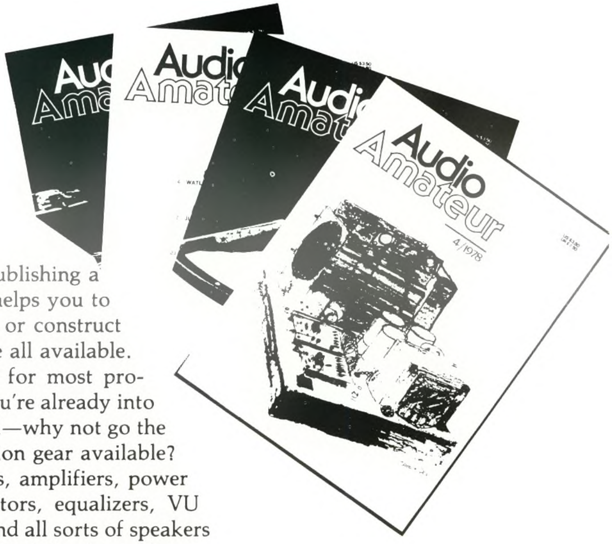

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Great articles out of our past

1970 "Price, Time and Value" surveys nine years of the fortunes of used equipment. An all silicon, complementary output, 20W per channel amplifier, fail-safe overload protected by Reg. Williamson. A high efficiency bookshelf speaker by Peter J. Baxandall. How do update and improve your Dynaco PAT-4 preamp. A visit to the Heath Co.

1971 A superb, simple, high quality preamplifier by Reg Williamson; A 4 + 4 microphone mixer, using four ICs in a compact chassis, with eight inputs and two-channel output. A four channel decoder for adding a new dimension to listening: cost to build: $12.50. Two four-channel encoders, one with microphone preamps, to put four signals on two tape tracks. Three voltage/current regulated power supplies for better power amp performance.

1972 A nine octave graphic equalizer with slide pots by Reg Williamson. A 10½" reel tape transport, a full-range electrostatic loudspeaker and a 900 watt tube amplifier for driving the electrostatic panels directly. A high quality op amp preamp, Heath AR15/AR1500 modifications. A new type A + B, low cost 35W power amp, electronic crossovers for bi- and tri-amplifier operation. All about microphones, and tuning bass speakers for lowest distortion.

1973 Construction: Five transmission line speakers: 8" to 24" drivers, peak reading level meter, dynamic hiss filter, tone arm, disc washer, electrostatic amplifier II, and customized Dyna Mark II and Advent 101 Dolby. How to photograph sound, power doubling, microphones, Jung on IC op amps, Williamson on matching and phono equalization, and much more.

1974 A perfectionist's modification of the Dynaco PAS tube preamp, a mid/high range horn speaker, a wall-mounted speaker system, an IC preamp/console mixer by Dick Kunc, a family of regulated current limited power supplies, a switch & jack panel for home audio, grounding fundamentals, low-level phono/tape preamp with adjustable response, an IC checker, a lab type ± 15V regulated supply. A series on op amps by Walt Jung and kit reports on an electret microphone and a Class A headphone amplifier.

1975 The superb Webb transmission line speaker construction article, how to test loudspeakers, a test bench set of filters, a variable frequency equalizer, building and testing Ampzilla, a power amp clipping indicator, a compact tower omni speaker, controls for two systems in three rooms. A visit to Audio Research Corp., an ultra low distortion oscillator, all about filters by Walt Jung, a universal filter for either audio garbage or crossover applications. An electrostatic speaker and complete schematics for Audio Research Corp.'s SP-3A-1 preamp, Heath's XO-1 and the Marantz electronic crossovers.

1976 Three mixers by Ed Gately, a vacuum system for cleaning discs, a 60W per channel amp for electrostatic speakers, a silent phono base, a perfectionist's tonearm, re-mods for Dyna's PAS preamp, Jung on active filters, a white noise generator/pink filter, A-Z tape recorder set-up procedures by Craig Stark, modifying the Rabco SL-8E, a high efficiency speaker system for Altec's 604-8G, uses for the Signetics Compandor IC, modifying Heath's IM (tube) analyzer, simple mods for Dyna's Stereo 70 amp, a tall mike stand. Kit reports: the Ace preamp, Heath's 200W per channel amp, Aries synthesizer, Heath's IO-4550 oscilloscope.

1977 Walt Jung's landmark series on slewing induced distortion, a wood/paper/epoxy horn, Reg Williamson's Super Quadpod, experiments with passive radiator speakers, a high efficiency electrostatic speaker with matching low-power direct-drive amplifier, modifying the AR turntable for other arms, do-it-yourself Heil air motion loudspeakers, a $10 Yagi FM antenna, Ed Gately's 16-in/two out micromixer, the speaker saver: complete stereo system protection. Audio Research modifies the Dyna Stereo 70; the super output buffer, a 101dB precision attenuator.

1978 Modular equipment packaging, A PAT-5 preamp modification, a radio system for Hospitals, supply regulation for Dyna's Mark III amp, B.J. Webb on phono interfacing and record cleaning, a 24" common bass woofer, a TV sound extractor, modifying the Formula 4 tonearm, a phono disc storage cabinet, Jung on IC audio performance and noise control, a visit to Peter Walker's Quad factory, a small horn enclosure, an audio activated power switch, the Nelson Pass 40W class A amplifier, a thermal primer, a capacitor tester, recording with crossed cardioids. Kit reports: Heath IC 1272 audio generator, Heath's IM5258 harmonic distortion analyzer, Hafler preamp, Dynaco's octave equalizer, West Side Electronics pink noise generator.

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POLYPHONY REVIEWS

Moving Pictures
by Michael Gilbert

Available by mail from Gibex Music, Riverglade 104 E. Hadley Rd, Amherst, MA 01002 for $6.85 (includes postage and handling).

"Moving Pictures" is the first album by Michael Gilbert, whose background and development has been quite varied. His broad tastes in composition and technique are well showcased on this album. I sense that the material used was collected over a period of time, as each composition seems to be a very intense study on the themes and techniques being used in that particular song. Other pieces are equally well presented, but deal with radically different types of expression; if nothing else it just takes a while to shift gears and get your chops down again. So, in this respect, the album is a good 'sampler' or anthology of a number of progressive music styles.

Side 1 starts with the "Moving Pictures Logo", which is unfortunately the shortest cut on the album. The musical and sonic coherency presented in 'Logo' make the piece conclude much sooner than I would expect. Multi-layered sequencing with string and drum parts operate within a very strong tempo context, yet there is little reference to a strict time signature. The longest cut of the album follows, "Ascents". Throughout this work is found a unique and successful blend of some of the newest and most technical musical developments with some of the oldest and most primitive techniques. The theme introduced by the wood flutes at the beginning is carried throughout the development and is frequently countered with electronic themes and intermittent effects. Sometimes the electronics attempt to duplicate the basic theme; sometimes the distractions shake the listener from the hypnosis of the basic theme. With each new conflict, the development becomes more complex and hectic. Despite the periodic resolution with vocal choirs, organs, and synthesized flute voicings, the final resolution could be nothing but a reinstatement of the opening theme. A really well done piece. "Unwinding" closes side one with a study of the 'de-mechanization' of an initially complex structure. The sequenced FM timbres move through a number of changes while accompanied by a noise based rhythm structure. Eventually the sequence slows, and the theme evolves to a pure tone structure at the end. Very relaxing.

I should also mention that the pressing and mix on this album are really superb. The wide dynamic range utilized in many of these compositions is accurately reproduced with little distraction from surface noise. That seems to be an increasingly rare thing these days.

Side two carries a number of shorter studies, each of which is more of a specialized timbre study or 'electronic tone poem'. The opening "Other Voices/ Other Rooms" is a very distant sound, free from tempo or rhythmic indications, and accented by reverb and muting of the major statements. "Steel Clouds" is based primarily on percussion instruments, and uses a number of miked instruments in addition to the processed and synthesized signals. "Winter Light" is a chordal study with little melodic emphasis. What appears to be processed guitar is a major element of this study. "Plant Life" seems more machine-like, with heavy use of sequencers, than one would expect from the title. Nevertheless, the electronic effects are a bit more solid here than in some other cuts, and the music flows very well. Side two closes with "Phase", which is another of those cuts that leaves you wanting more. The stereo bass line serves as an integral part of the melodic development as well as providing a basis for other parts. One of the more interesting techniques used here is the increase in stereo activity during the final fade, which is actually a very long fade. You find your self listening to the music more and more as the music moves further away; thus, the listeners unresolved desire to hear more of the piece.

Much of the technology on this album is very tastefully used, which sets the result far above other music which has used similar timbres, techniques, and mixes. The music is very emotional. "Moving Pictures" is not so much for those who want to explore new horizons of expression as it is for those who are interested in how the technology we have gained so far can be solidly integrated into existing musical heritages.

I Just Got Here Myself
by Pat Gleeson

(This album is currently unreleased, but is expected out soon.)

This album, finally, provides the electronic music community with a full musical statement from Pat Gleeson. And a good one it is. Unlike the first two albums, 'Just Got Here' contains all original material; Pat Gleeson is every bit as much a composer as the synthesist, arranger and engineer he proved himself to be on the first albums. His compositional tastes run to the grand orchestral statement, similar to Fast or Tomita, but there is less emphasis on searching for new timbres and an effort to extend the boundaries or capabilities of traditional voicings. From this approach, his synthesis shines with a mastery of the physics of acoustic instruments, and a grasp of which synthetic mutations of these families will provide extended, but compatible, instrumentation.

Side one is "Draconian Measures". One of the first obvious achievements is the brass synthesis. Using short enveloped bursts of FM during the attack period of the main envelope, Gleeson has taken the traditional 'horn blip' patch to a more realistic 'splattered' brass which encompases not only tonguing blips, but embouchure adjustment as well. He has also perfected a guitar patch which, in conjunction with sequencing or fingering technique, gives an excellent simulation of finger-picked guitar. The stereo thunder which seperates the two major sections of the piece is well done, and the tympani near the end are more accurate than any I've heard. Overall, the complex filtering and generated harmonic structures render instrumentation which is tremendously real and 'acoustic' sounding. Subtle mixing and good ambiencing (reverb, delay, phase differencing, etc.) are the final touches to the production of a classic classical image. The composition itself is as much a major feature as the technology. Much of the composition and chord structure shows influences of George Martin. The movement of the themes through the ensemble and the balance of concurrent voicings are exceptional.

Side two contains "Rainbow Delta" and "Unacceptable Dance Styles". The first few moments feature a very accurate reproduction of solo cello. That's not easy. Later on this side, solo violin parts can be distinctly heard. There is obviously a lot of work in the strings. "Rainbow Delta" uses sequences of tonal noise bursts as a rhythmic structure and bass line reinforcement near the beginning. This sounds like 'melodic explosions' and is very interesting. Based on the main theme, the strings, flutes, and organ increase the complexity of their interaction amongst sliding atonal effects in an almost free form contrapuntal style until, at the climax, a tight splice is made into a snappy disco-type ensemble. An extensive percussion pattern with a type of temple block pattern and punchy synthesizer bass line provide the basis for the too-short movement. Another tight splice drops you in the middle of a lonely nature setting replete with crickets and creatures which sets the mellower mood for a chamber orchestra with the solo violin and viola voicings mentioned earlier. Another thunderstorm, this time with blowing rain, leads to "Unacceptable Dance Styles" which is a sort of cosmic mambo that contains some of the more original patches of the album. This piece, more than the others, spotlights Gleeson's skill in keyboard technique and improvisation. Near the end, each key modulation is accented by a cymbal crash, sound effect, or cluster sweep to set off the myriad changes and developments the piece is going through. The side closes with a majestic brass/ string recap of the main themes.

I really like this album. It contains a lot of innovative patching based on a solid understanding of acoustics. The composition is ambitious and well arranged. It is a good addition to any library; watch for it.

Timo SS II
by Timo Laine
Lady Records LALP001

Most guitarists who are interested in guitar synthesis are familiar with Timo, his system (one of the most extensive guitar synthesizers around), and his first release on A&M, Symphonic Slam. With his second release, Timo has started his own label, and has developed his proficiency on the synthesizer to a point where it is nearly impossible to tell that the two instruments are related at all. The pitch followers used for each string are very fast and accurate, so nothing is lost during the transition. The amount of power and guts the synthesizer adds to the sound is really amazing. In fact, one of the weaker points of the album is that the synthesizer is mixed back to the point where some of the impact is lessened in comparison to what you hear during a live demonstration. But, for the guitarist thinking of adding synthesis to his system, this album should show the degree of technical development which currently CAN be achieved with guitar synthesizers, as opposed to what you see at your local music store.

The basic ensemble on the album is Timo on guitar and synthesizer, plus a bass player and a drummer. On a few cuts, additional musicians set in on other instruments. Most of the material on the album is commercially oriented, but holds a lot of technical display for those interested. Towards the end of "Cyclops" a patch is used which uses foot controlled pitch bend, or perhaps envelope generators are doing the bends, to generate a sitar type of sound. With the tablas mixed into the background, the effect is very convincing. In "The Nights About To Come", a Vanelli-type strut starts off with a synthesized brass/ sax type chorus accompanying the guitar. "Do Me Slow" shows the effects obtained with polyphonic glide and foot pedal controlled transposition, a technique which helps convince the listener that the instruments are not one. The solo in this piece also shows to good advantage the trackability of the followers where Timo is doing string bends and finger vibrato.

On side two, the muted trumpet patch used on "No One Knows" sounds amazingly like there's a small dance band playing behind the guitar work. The other songs on side two are the strongest commercial contenders, "Freedom", "J J Jane", and "I'm On My Way". Most are based on rock compositions with guitar/ synthesizer solos and lots of studio processing.

If you like rock music, the

continued on page 35 ....

JOT NOTATION...

An Electronic Music Shorthand

BY: JOHN ALLEN MITCHELL

The development of any new technology brings with it the need to develop new systems of notation. Most frequently a particular form of notation is used as a means of recording a process (as with a computer program) but more importantly, notation permits the exchange of ideas -- so necessary for the maturation of any art or science. Time has proven that an art which does not develop a means with which to describe itself, or looses that ability, eventually reaches a type of stasis and becomes a tradition. To remain vital, an art must mature, and as I have just stated, maturity is a result of the free exchange of ideas via some form of notation. The field of electronic music finds no exception to that rule.

While the methods of patch notation are as varied as the composers who use them, they must all have one thing in common -- reproducibility. A score which doesn't repeatedly yield the same result either for the composer or for the performer is worthless. If such is the case, an improvisation would likely serve better, so why bother.

Both performers and composers working with the electronic medium have as yet to settle on any one patch notation, and probably never will, as no one system seems to cover enough ground. Beaver / Krause - type flowchart notation seems to be enjoying immense popularity as of late, due to its clarity, reliability, and ease of adaptation to a multiplicity of systems. This method however, can be lengthy and often times unwieldy. What is needed then, is a means by which electronic music patches can be quickly and concisely notated. What we need is an electronic music shorthand.

The notation introduced here, (which we shall call "JOT" for the sake of brevity) combines certain aspects of standard flowcharting methods with symbols derived from mathematics and welding (yes Virginia, Welding!). Jot presents several distinct advantages:

1. Jot is space saving. A patch recorded in Jot generally requires less space than one in flowchart notation. This is particularly advantageous on already- overcrowded music manuscripts where space is at a premium.

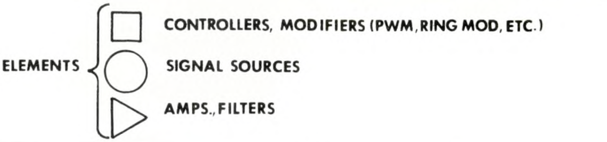
2. Jot can be typed. This comes in handy where a neat copy is required, but a template plus graph paper are unavailable. The fact that it can be printed with but little modification, makes for space saving and efficient means of cataloging patches with the aid of a microcomputer and printer. In this manner, not only could thousands of patches be stored on cassettes, but by employing a modem, they could be "traded" with a friend in Timbuktu via the telephone lines!

3. Jot is readily translated into flowchart notation and back again.

4. Jot can be written from a verbal description. Using conventional notation this task can prove nightmarish. Just how Jot is written and read will be expounded on shortly, but if your skills in flowchart notation are rusty, now would be a good time to dig out whatever information you may have on the subject and review.

WRITING IN JOT

Figure 1 is a list of



PWM	PULSE-WIDTH MODULATOR	○	OSC. (WAVEFORM GOES INSIDE)
LP	LOW-PASS	Ⓝ	NOISE
BP	BAND-PASS	**-1**	INVERTER
HP	HIGH-PASS	**SH**	SAMPLE AND HOLD
BR	BAND-REJECT	**M**	MULTIPLIER (RING.MOD.)
AR	ATK./RLS. GEN	K_g	KBD. GATE
T	ADSR	K_t	KBD. TRIG.
SEQ	SEQUENCER	K_p	KBD. PORTAMENTO
EF	ENVELOPE FOLLOWER	**K**	KBD. CONTROL VOLTAGE
VCA	VOLTAGE CONTROLLED AMP.	**MR**	MIXER
LFO	LOW FREQ. OSC.	**RVB**	REVERB UNIT

FIG.1

symbols and abbreviations we will need to implement the notation. You will notice that some of the abbreviations are already fairly standard. Also note that the geometric shapes used to denote modular functions are identical to those currently found in flowchart notation. While the list includes most of the standard functions indigeneous to contemporary electronic music systems, it is a simple matter to include abbreviations for non-standard or hybrid functions as required. To save time and space, it is recommended that all such added abbreviations be kept to a maximum of three letters.

In figure 2 can be found the various operators along with their verbal interpretations. These operators are needed to show how the modules are grouped and/or interconnected.

STATEMENT	INTERPRETATION
A/B/C	A GOES INTO B WHICH GOES INTO C.
A×B	A AND B.
A(B)	
AB	
AB/C	A AND B GO INTO C.
$A/\underset{D}{\underline{B}}/C$	A AND B GO INTO C. D GOES INTO B.
$A/\overset{\underline{D}}{B}/C$	

FIG. 2

The framework on which the abbreviations and operators "rest" is called a <u>structure</u> (figure 3A). A structure is composed of an <u>element</u> and an <u>indicator line</u>. More often than not, the main element will be an oscillator, although it could also be a filter, VCA, or other module. Inputs to the main element are found on the upper side of the indicator line. The output from the main element is routed into the leftmost element <u>under</u> the indicator line (see figure 3B). As in flowchart notation, signal flow is presumed to be left-to-right. It is important to note that <u>the rightmost element under the indicator line is the last in the system (typically a VCA) and is presumed to terminate into a main amplifier or mixer</u>.

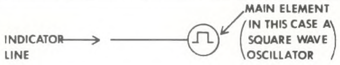


FIG. 3A

JOT STRUCTURE

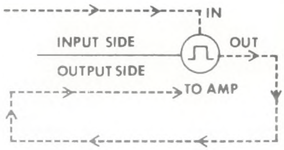
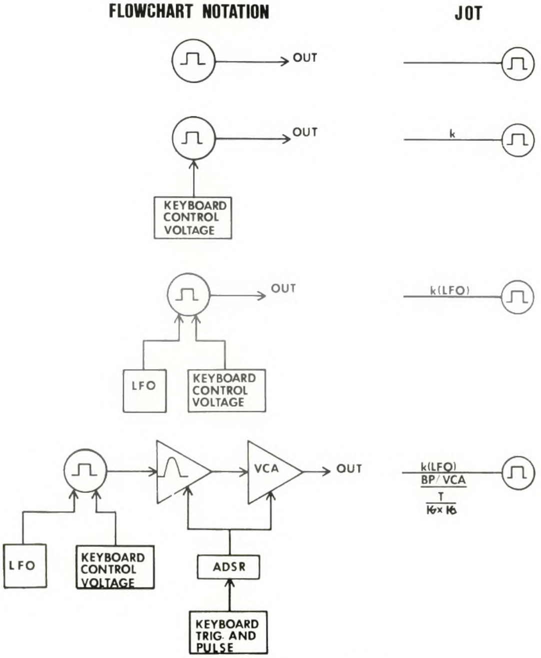


FIG. 3B

SIGNAL FLOW TO AND FROM MAIN ELEMENT

PUTTING IT ALL TOGETHER

Figure 4 shows the evolution of a simple patch in flowchart notation along with its equivalent in English and in Jot. As an exercise you may wish to place a sheet of paper over a figure and practice writing its equivalent. The ease with which jot can be written will be made apparent by covering both the Jot



DESCRIPTION

SQ. WAVE OSC. GOES TO AMP.

KBD. CONTROL VOLTAGE GOES INTO THE OSC.

KBD. AND LFO GO INTO THE OSC.

KBD. AND LFO GO INTO OSC. OSC. GOES INTO BAND-PASS GOES INTO VCA. ADSR GOES INTO BAND-PASS AND VCA. KBD. TRIG AND KBD. GATE GO INTO ADSR.

FIG. 4

serge modular

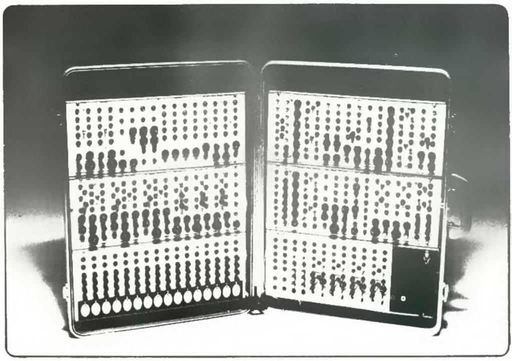


PHOTO BY BILL MATTHIAS

music systems

We'd like you to know about Serge Modular. We make what may be the most precise and versatile synthesizer system in use today. The Serge Modular is available either fully assembled or in money saving kit form. We made it modular, to assure that a musician (or video or computer artist, for that matter) can put modules together to fit his or her specific needs, and also to make room for future expansion of a Serge System.

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Have A Project?

Write It Up

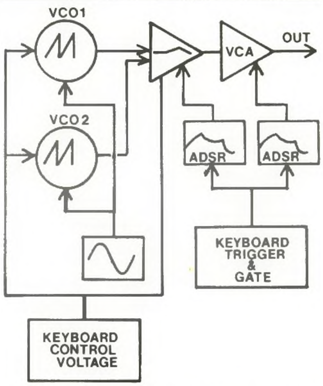
For..... POLYPHONY

.....Polyphony will pay $25 per printed page for articles on items such as equipment modification, circuit design and "build it" projects, software, equipment service and maintenance, interviews, theory tutorials, recording tips and techniques, and other topics of interest to our readers. $10 is paid for each patch chart published - we want patches for any model synthesizer. We also want photo or graphic submissions for use on our cover ....

.....If you have never written an article before, don't hesitate to give it a try. Material need not be polished -- that's our editor's job! Illustrations and schematics can be presented as rough pencil drawings. Photos should be black and white glossies. Text should preferably be typed double spaced ....

.....All material submitted must be original and never before published. Payment is made upon publication. Submissions are returned if you provide a SASE. All materials accepted and associated copyrights become the property of POLYPHONY (unless arranged otherwise) and may not be reproduced without written permission from the publisher. If you'd like to test your ideas first, send 'em in. Or, if you have any questions, let us know ....

JOT...



FROM SYNAPSE VOL. 2#5. USED WITH PERMISSION

FIG. 6A

and flowchart representations and reproducing the patch from its verbal description alone.

One of the reasons for Jot's relative compactness is that the system does not rely on continuous interconnection lines. This feature does away with the "octopus syndrome" (see figure 5) and all but abolishes the confusing crossovers often found in flowcharting.

Examination of figure 6 will reveal some other advantages of Jot. The flowchart presented is from Synapse, volume 2, number 5, page 38. Note that the letter K appears twice in the Jot representation. This indicates that the same keyboard control voltage is presented to the two oscillators AND the low-pass filter as well. If a seperate keyboard was desired for the filter, a subscript would be used. It may already have occured to you that making an addition or deletion to a patch recorded in Jot is often reduced to the simple task of adding or erasing a letter in the graphic structure. For example, the addition of a sequencer to jointly control the filter in the flowchart of figure 6 might be messy. To make the same addition in Jot, we simply add the expression "seq" to the topside of the indicator line as in figure 6C.

Like any other system, Jot is not a panacea for one's every notational ill. settings for each module must still be recorded elsewhere, and the synthesist has

continued on page 36 ....

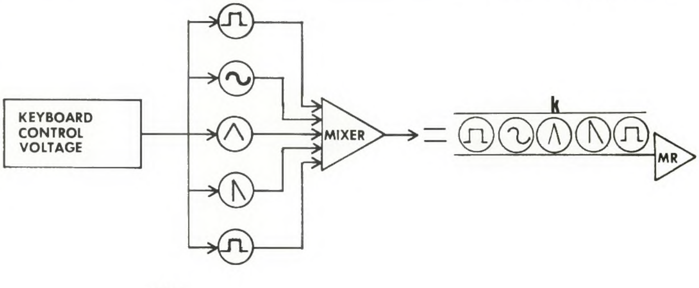
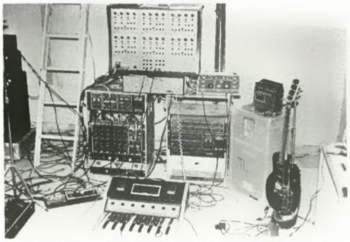


FIG.5



In Canada, Lady Records are distributed by MetroDisc. Lady Records are available in the U. S. by mail order only. The Timo SS II albums are $6. 95 each postpaid. Use the handy coupon or contact: Lady Records, POBox 1463, Newport Beach, CA 92663.

With a hexaphonic 360 Systems pitch to voltage converter, six Oberheim expander modules, and a Moog Taurus bass synthesizer all used in conjunction with multi-processed guitar, Timo Laine's instrument represents one of the most advanced guitar synthesizers available today. Throw in some punchy back-up musicians, exciting studio effects, and solid rock composition and you have an album which is providing one of the strongest forces in electronic guitar work today.

$6.95 from: LADY RECORDS, Box 1463, Newport Beach, CA 92663

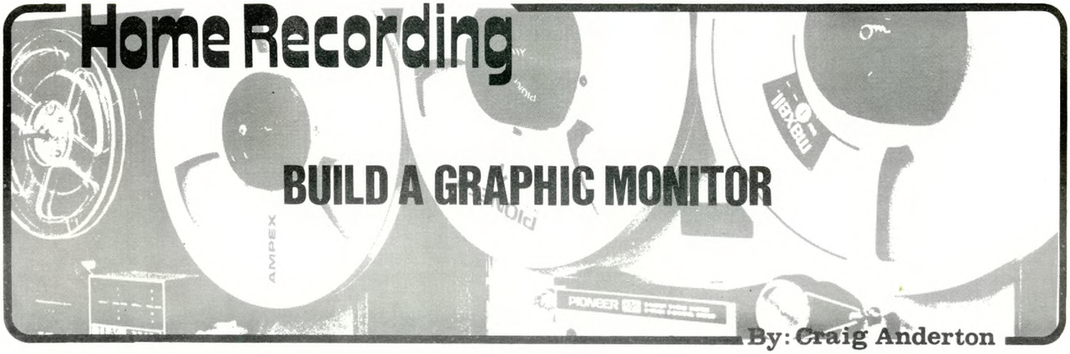
PLEASE RUSH ______ TIMO SS II Albums at $6. 95 ea. $______ enclosed.

name: ______________________________

address: ______________________________

City: ______________ state: ______________ zip: ________

LADY RECORDS, Box 1463, Newport Beach, CA 92663



Home Recording

BUILD A GRAPHIC MONITOR

By: Craig Anderton

You've all probably heard of a graphic equalizer; well, this project is very similar, since it takes an input signal and divides that signal up into a certain number of bands. However, instead of re-combining these bands into an output mixer, we drive a separate LED with each output (see figure 1). With this device, we can check out the overall energy distribution of a recording with regards to frequency response. During bassy sections, the lower LEDs will light; during trebly parts, the upper LEDs will light. It's really interesting to see this monitor in action---you can actually relate specific LEDs to specific instruments, not just frequency ranges. When a bass part slides down from the top of the string to the bottom, you can see a corresponding change in the illumination of the LEDs.

First, credit where credit is due. To avoid the hassle of calculating the cap values for the various filter sections, I took the values given for "Room Equalizing Instrument" filters described in National's Audio Handbook. If you're interested in using this circuit strictly for equalization as opposed to visual monitoring, I suggest you refer to the Handbook for details.

HOW IT WORKS

Figure 2 shows the complete schematic for one filter section. The other sections are identical save for the capacitor values. The op amp is a 741---it's cheap, and the noise doesn't matter in this application; you could also use two 4136 quad op amps since there are 8 filter sections all told. The circuits are basically non-critical...I wired them all up on a piece of perf board, which worked just fine. C1 and C2 determine the resonant frequency of the filter. R2 is a bias control for the LED, which we'll get into later under calibration.

This circuit is designed to work from a regulated bipolar 9V DC supply. Suitable power supplies are made by Bill Godbout Electronics (project #13A-9) and PAIA (kit #4771) if you don't already have a ± 9V power supply handy.

CALIBRATION

Connect all the inputs of the filters together in parallel, and then connect these to the sensitivity control as shown in figure 1. Hook the sensitivity control up to a loudspeaker output. The Graphic Monitor is designed to be used with loudspeakers; if you wish to use the monitor with low level or high impedance output signals, you'll need to add some kind of buffered gain stage in front of the sensitivity control.

Apply power to the monitor with the sensitivity control turned all the way down. Some of the LEDs may light up, and then again, they may not. Rotate one of the filter trimpots....at one extreme of the rotation the LED should be lit, and at the other extreme it should be off. Set the trimpot so that the LED is just barely glowing at as faint a level as possible (it helps to turn down the room lights for this!). Adjust the other trimpots so that the other LEDs glow with the same brightness; the monitor is now ready for use. By the way, if you can use matched LEDs for the display, so much the better.

Put some program material through the speakers, preferably something with a wide range of frequency response dynamics. Adjust the sensitivity control so that the LEDs don't saturate to full "on" during loud passages, but are still sensitive to low-level passages of music. It may take a bit of fiddling to determine the optimum setting, just aim for a display that is at its very brightest during the loudest parts of a piece of music.

APPLICATIONS

OK, SO NOW I'VE GOT IT BUILT. WHAT GOOD IS IT? There are several uses for the Graphic Monitor...here are some of them. If you can come up with any more, let Polyphony know about it.

1. Mixing level monitor. If you are mixing a bunch of tunes for an album all on

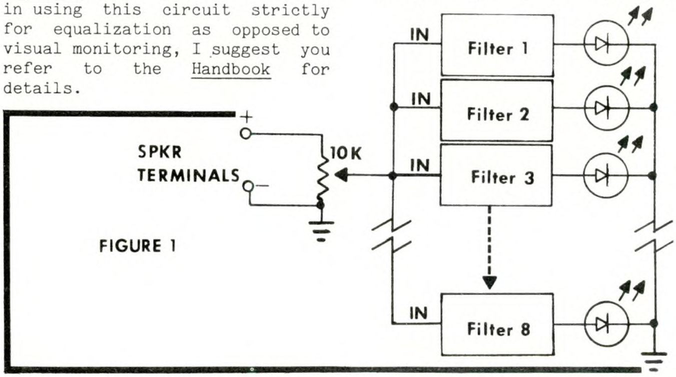


FIGURE 1

different days, it's an advantage to be able to listen to the monitor speakers at similar overall listening levels. By having the Graphic Monitor hooked up to the speakers at all times, you can tell if you are monitoring at a consistant level by aiming for a constant average brightness in the display. Don't touch the sensitivity control once you have it calibrated to your "optimized" listening level.

2. To keep frequency response balancing under control. I've put on a few albums and analyzed them through the Monitor. On one, I noticed a tendency for the high frequency content to increase as the album progressed from cut to cut. Since I believe this was an album where the tunes were pretty much mixed in order of appearance on the disc, the increase in high frequencies might have been due to listener ear fatigue setting in, and a turning up of the highs to compensate. Normally when using the Graphic Monitor, the highs will be like a frosting or topping--not very sustained, but dancing into view from time to time. If you start laying in too many highs, or too many boomy lows, you'll be able to see the results on the Monitor.

3. For analyzing instrument timbres. Try putting a sine wave through the Monitor, then a triangle wave, then a sawtooth wave. You'll note that the increased energy contributed by the sawtooth wave's higher harmonics cause the upper Monitor lights to glow more brightly. This can be real useful---for example, want to find out how many highs are really in that percussion instrument? Feed it into the monitor and find out.

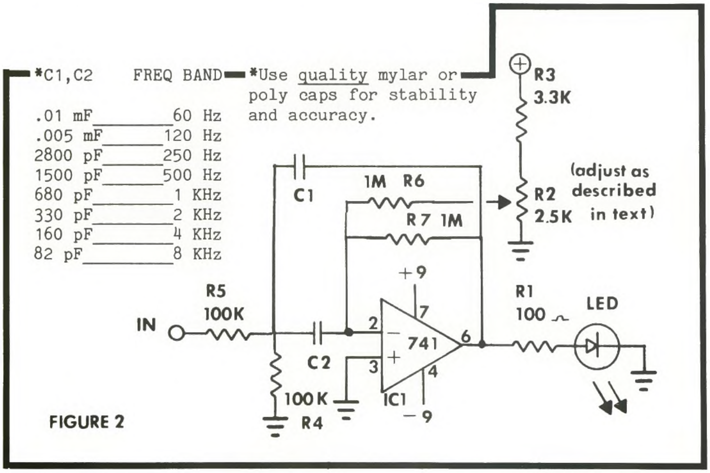
4. It's a great toy. Yes, the patterns are interesting and fun to watch---but I couldn't help noticing that in the majority of cases interesting patterns went with interesting music. It helps to keep your music varied with regard to frequency balance as we discussed last issue, and this device is really an invaluable aid towards that goal.

MODIFICATIONS

If you want, you can add filters that are spaced in between the current ones to get a half-octave response. In theory the filters should be sharper, but considering that this is a relatively relaxed application the same filter structures we've been using for the 8 step version are acceptable.

Aside from that, there's not much more to talk about. Enjoy your monitor; whether used as an interesting visual toy or as a cheapo subsititute for a spectrum analyzer,, you'll be glad you put one together.

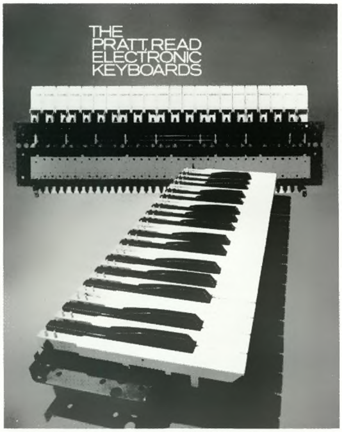
P.S. If you'd like to see someone offer this circuit as a kit of parts, or perhaps just offer the circuit board, tell Polyphony about it. If there are enough responses, I'll start to work putting a board layout together. ●

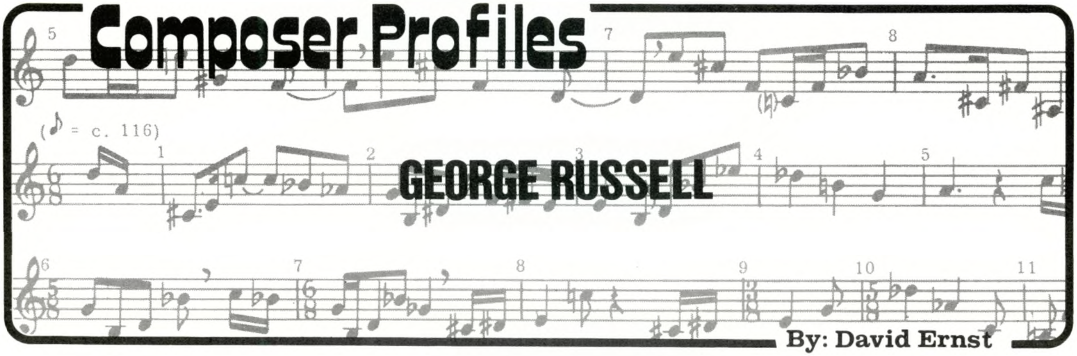


*C1,C2	FREQ BAND
.01 mF	60 Hz
.005 mF	120 Hz
2800 pF	250 Hz
1500 pF	500 Hz
680 pF	1 KHz
330 pF	2 KHz
160 pF	4 KHz
82 pF	8 KHz

*Use quality mylar or poly caps for stability and accuracy.

FIGURE 2





Composer Profiles

GEORGE RUSSELL

By: David Ernst

George Russell (b. 1923), composer-pianist-theorist, is perhaps known for his work in the field of jazz. His book, The Lydian Chromatic Concept of Tonal Organization (1953), has exerted a profound influence on jazz musicians for the past two decades. In 1969 Russell received both a National Endowment of the Arts grant and a Guggenheim Fellowship, enabling him to spend some time in Scandinavia to teach, to perform, and to compose for radio and television. During this period he produced some electronic compositions that display a refined approach concerning the treatment and mixing of sounds, as well as the combination of prerecorded tape with jazz ensemble. Russell is presently teaching at the New England Conservatory of Music in Boston.

The "Electronic Organ Sonata No. 1" (1968) is Russell's first electronic work, whereby he taped an improvisation on a church organ in Oslo, Norway. This organ tape was then manipulated and transformed at the electronic studio of Radio Sweden in Stockholm, and the result is a musique concrete composition for solo tape.

Although obvious jazz elements rarely appear, e.g., a few 'blues' note patterns, the improvisational character of this piece clearly reflects Russell's affinity with the jazz medium. His compositional aesthetic in the "Organ Sonata" is similar to that of Ilhan Mimaroglu in his "Piano Music for Performer and Composer" (1966-67). Both compositions were realized from electro - mechanical transformations of prerecorded tapes of improvised keyboard music, and both are solo tape pieces. Russell, however, did not splice and cut-up the organ sounds to the extent that Mimaroglu did with the piano material. For Russell, splicing was used primarily for editing. Therefore, the original organ sonority is always present and never destroyed by extensive tape modifications as in parts of Mimaroglu's piece. Russell treats the unmodified organ timbre as the principal voice, and this is usually louder than any accompanying modified organ material. The result is not unlike that of a piece for organ and tape.

Before we consider the electronic modifications that Russell applied to the organ tape, we should first mention the inherent timbral resources of the organ. These include a wide range of predetermined timbres (referred to as 'registration' by organists) that may be used either individually or in combination. The resultant effect is analogous to filtering, providing Russell with a variety of preselected organ sonorities.

In addition to the many available organ registrations Russell also employs tape transposition, delay, and reversal, along with filtering, ring-, and amplitude modulation. The modulation signals are often subaudio frequencies, and all modulated material undergoes extensive filtering---especially high-pass. Sometimes white noise is mixed with the filtered sounds to create thicker textures, but the original organ timbre usually remains in the foreground.

Among the most outstanding features of the "Organ Sonata" is its clear presentation and articulation, since large chordal masses (reminiscent of the orchestral music of Ligeti and Penderecki) permeate the entire work. Frequent overdubs also contribute significantly to a rather complex texture, but Russell successfully avoids timbral and textural ambiguity by discrete application of modulation and filtering techniques to create subtle delineations. By working in this manner Russell was able to superpose melodic, harmonic, rhythmic, timbral, and textural elements in a polyphonic fashion. The following illustrations show how some of these procedures may be applied to a small synthesizer system.

FIG. 1

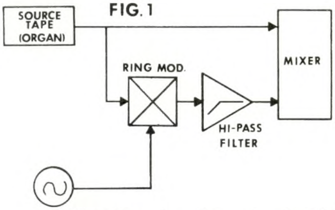


Figure 1 is a basic configuration that permits a prerecorded source tape (in this instance organ) to be mixed with a modulated-filtered version of itself. In the "Organ Sonata" Russell generally mixes the source tape at a higher amplitude level than that of the modified material. Figure 2 illustrates the inclusion of slight tape delay only to the modified sounds. Finally, figure 3 demonstrates the addition of amplitude modulation via a low-frequency oscillator to the modified-delayed material. In each instance the resultant texture becomes more complex, but each level of complexity is accompanied by an additional modification to insure timbral and textural clarity.

Like the "Electronic Organ Sonata", Russell's "Electronic

Sonata for Souls Loved by Nature" (1969) was also realized at the electronic studio of Radio Sweden. The latter is for prerecorded tape and jazz sextet consisting of tenor sax, trumpet, electric guitar, piano, bass, and drums. It is divided into two parts, each with numerous subdivisions.

As seen in the "Organ Sonata", Russell obtained a high degree of timbral unity by restricting the sound source to the organ. All sounds were derived from this sole source to generate a rather limited sonorous environment. Although this form of timbral unity is still present in "Souls Loved by Nature", Russell extends the concept to include stylistic unity. He refers to the prerecorded tape as being "pan-stylistic"---made up of different tapes of music electronically modified. The tape serves as a background for the jazz ensemble, which also plays in a pan-stylistic fashion. Russell's compositional aesthetic here parallels that of Karlheinz Stockhausen as found in "Telemusik" (1966) and "Hymnen" (1967). Both composers are striving toward a form of music that is not restricted to any one culture, but rather a universal music.

The tape for Part I of the "Electronic Sonata for Souls Loved by Nature" is characterized by slowly evolving textures like those prevalent in the tape pieces of Pierre Henry ("Le Voyage") and Iannis Xenakis ("Bohor I"). Russell's tape is continuous, and it is composed both of electronic and instrumental sources. Sustained chordal textures are enhanced via tape reversal and transposition, much overdubbing, ring modulation of melodic material, and ring modulation to produce metallic sounds. Since one of the functions of the tape is to provide an accompaniment for the instruments, Russell utilizes both extreme high and low frequencies to complement the instrumental registers.

Another function of the tape is to serve as a transition between the subdivisions of Part I, defining the overall structural framework of this composition. Figure 4 depicts the relation between tape and jazz ensemble, wherein brief sections are delineated by various instrumental combinations playing in conjunction with a continuous tape. Sectional distinctions are enhanced by the pan-stylistic performance of the jazz sextet. Conversely, these distinctions are frequently blurred because of soft instrumental entrances that blend with the tape to the extent that timbral and textural changes are not immediately perceptible.

Part II of "Souls Loved by Nature" displays more sectional contrasts than did the first part due to the inclusion of more ring modulated instrumental sounds (on tape), and increased rhythmic variety. In addition, the beginning of the tape contains African men speaking and chanting, and since this texture appears only once strong sectional divisions arise. Finally, the tape of Part II is not as continuous as that of Part I so that sectional contrasts are even more apparent.

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FIG.2

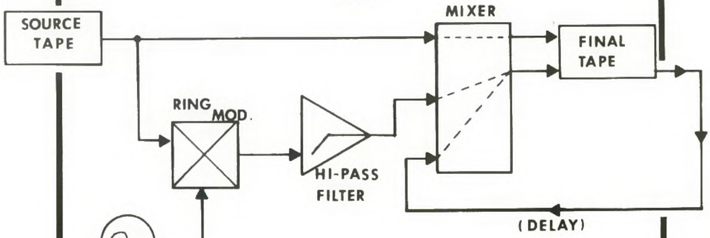


FIG.3

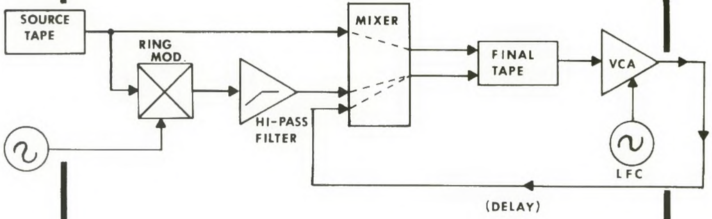


FIG.4

INDUSTRY NOTES

Tell them you saw it in POLYPHONY

VERSATILE RHYTHM

RolandCorp US has introduced two unique computer rhythm units which provide accent and variation to overcome the mechanical rhythm machine feel common to most electronic drummers. The CR-68 and CR-78 models' unique feature accentuates and syncopates different voices to add real feeling to the rhythms. The units also contain an automatic/manual variation mode or "fill" which automatically provides one of 11 rhythm variations to occur every 2, 4, 8, 12, or 16 measures. In a manual format, these fills enter the beat only when triggered by a footswitch or panel button.

Rhythms provided include 20 selections, such as two seperate disco beats and many rock variations. Volume and brilliance controls allow modification of the final drum mix, with capacilities to add and delete some instruments for further variation. The innovative CR-78 also includes a programmer which allows the musician to pre-program four rhythm tracks of up to 11 instruments each by selecting the memory and voice and then tapping its part out on a special programming pad. The CR-78 features the additional voice mixing capabilities as well as a fade in/out device. Both units provide sync connections for use in conjunction with synthesizers, sequencers, and other automated devices.

The CR-78 has a suggested list of $795 and the CR-68 is $495. For more information contact RolandCorp US, 2401 Saybrook Ave., Los Angeles, CA 90040, phone (213) 685-5141.

JOB OPENINGS

Sequential Circuits (1172G Aster Ave., Sunnyvale, CA 94086, 408-296-3116) is looking for technicians to work on their Prophet synthesizers. If you have a background of music, electronics, and work with microprocessors (especially the Z-80), it would be well worth your time to contact Dave Smith. For more information see the ad in Equipment Exchange.

SUMMER CLASSES

The New England Conservatory of Music will hold a Summer Session from June 25 through August 3, 1979 featuring workshops, courses, and master classes. Among the numerous offerings of special interest is the Electronic Music Workshop with Robert Ceely, June 25 - 29.

The workshop will be divided into two sessions . Mornings will include lectures and demonstrations of the hardware and software of electronic music, and discussions of how and why synthesizers work, and the theory behind their sounds. The afternoon sessions will allow students to get hands-on experience with the Arp, Moog, Buchla, and EML synthesizers. Studio technique covering multi channel recording, recording with sync, overdubbing, reverb, advanced sequencing techniques, and unusual patch configurations will also be explored and demonstrated.

For more information, contact Robert L. Annis, Director of Summer School, New England Conservatory of Music, 290 Huntington Ave., Boston, MA 02115, (617) 262-1120 extension 270.

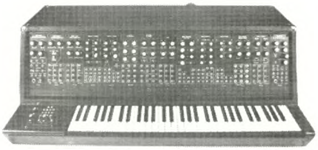
LIVE McLEAN

"The McLean Mix" (Barton and Priscilla McLean) mentioned in the Composer Profiles column last issue are making a number of live performances which you may wish to catch: North Texas State University- April 19, University of Illinois at Urbana- April 23, Indiana University at South Bend- April 25, University of Akron- April 27, Columbia University in New York- May 2, The Kitchen in NYC- early May, and the National Gallery of Art in Washington, DC- May 6.

SYNTHESIS WORKSHOP

Again this summer there will be an electronic music workshop at Fort Lewis College, Durango, Colorado from June 11 through 22. Three labs will be presented, each with different sets of synthesizer gear. Minimal information was available at press time. For more information contact Carroll Carruth at Arizona Western College, PO Box 929, Yuma, AZ 85364, phone 726-1000.

NEW MODULES, NEW SYSTEMS



The Aries Music Keyboard System III is a modular keyboard-controlled synthesizer featuring normal pre-patching that can easily be overridden when other patches are desired for use in live performance. Designed for playing lead lines, the synthesizer also provides a "natural filter patch" that adds realistic quality to the timbre, with brightness controlled by a foot pedal. Other capabilities of the system include low pass and multi-mode VCFs, VC phasing, VC stereo panning, stereo reverb, 5 octave keyboard, pre-amp and envelope follower, balanced modulator, 3 audio VCOs.

This and other new Aries systems feature the latest Series III modules which incorporate electronic music integrated circuits for easy assembly.

Features of the new modules include voltage controlled envelope generator parameters, phase locked sync on the VCOs, two modes of pulse width modulation, VCQ and notch/peak response switch on the multi-mode filter, and much more.

The Aries System III is priced at $2625 in kit form, and $3460 wired. A new catalog is available which describes the new modules and systems by contacting: Aries Music, Inc., PO Box 3065, Shetland Industrial Park, Salem, MA 01970, phone (617) 744-2400.

CREATIVE RECORDERS

TEAC is introducing a new four channel open reel deck with Simul-Sync to succeed its famous 3340 series which is being put to rest this year. The new A-3440 is a three head, three motor, two speed (7.5 and 15ips) system which features all new transport design, and new circuitry and logic systems.

In the new 3440, the Simul-Sync is tied directly to the 'Function Select' switch for each channel either for recording or overdubbing. Rather than having Record Mode switches, Sync select switches, and Monitor select switches which must be enabled for each channel, the new Function Select switches route and select all the appropriate signal paths automatically. Other features include a manual cue lever for fast search, cueing and editing, four VU meters, mic/line input selectors, four front panel mic inputs, and independent output level controls for each channel. Also included is a headphone monitor mixdown section which allows you to hear your material mixed with variable output level. Optional pro curve dbx is available, as is remote control.

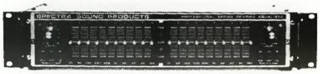
The A-3440 has specs which include .04% wow and flutter, 65 db S/N ratio, and 35 to 22K Hz frequency response at 15 ips. The suggested retail price is $1500.

* * *

The model 124 Syncaset is a special purpose cassette recorder which features Teac's Simul-Sync for increased user flexibility in multi-track recording. Features include a DC servo motor, Dolby noise reduction, memory rewind, bias and EQ switches, and a unique cross feed switch which allows blending of the layered recording for playback. Specs include wow and flutter of .07%, frequency response of 30 to 16KHz, S/N of 55dB without Dolby. For the amateur recordist who doesn't want to jump all the way to a 4 track recorder, this looks like a good place to start. The 124 will be available in April at a suggested retail of $449.

For more information on either of these recorders contact TEAC Corporation of America, 7733 Telegraph Rd., Montebello, CA 90640, phone (213) 726-0303.

SUPER EQ



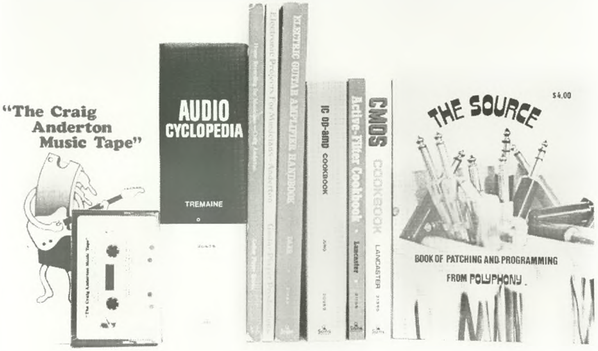
The Spectra Sound 1000B graphic equalizer incorporates the latest bi-fet circuits to provide wide bandwidth (20 to 20KHz ± .5dB), low noise (-100dBm), high slew (13 volts/microsecond), and low distortion (IM and THD less than .008%). LED overload indicators are provided for each channel as well as output level controls. Active balanced inputs and outputs with line drivers are available. For more information contact Spectra Sound Products, Inc., 2245 South West Temple, Salt Lake City, UT 84115.

MUSIC ICs

A kit for a complete synthesizer voice, based on six custom SSM electronic music ICs, is available to experimenters from E-mu Systems (417 Broadway,

continued on page 36 ....

POLYPHONY 'Bookpage'



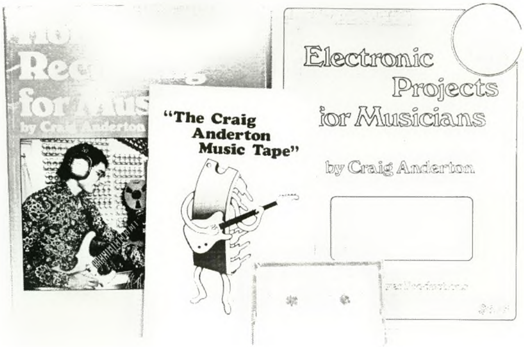
BOOKPAGE is provided to help Polyphony readers find the special publications their projects require. The items we currently stock are a must for every active music experimenter's lab or studio bookshelf. We are looking for additional items which may interest you. Let us know of books (and publisher, if you know it), records, tapes, or anything else which would be useful to you or others. Circuit boards for Polyphony projects? Patch chart pads? Some specialized electornic supplies?

TO ORDER: Use the convenient fold-up in the center of this issue or, if someone beat you to it, the order form at the bottom of the next page. We cannot invoice; payment must be included with any orders. As a bonus, we pay postage on shipments within the U.S.. Foreign customers must enclose $1.00 per item to help defray additional costs. ALL foreign orders MUST be paid in U.S. FUNDS, preferably drawn on a U.S. bank. Better yet, send certified check or money order.

Craig has prepared two fine books to serve as continuous reference manuals in building, and using musical electronics. Electronic Projects for Musicians is a perfect introductory manual for the musician with no previous experience in electronics. Home Recording for Musicians outlines the selection and operation of recording equipment for the musician with BIG ideas and small budgets. The Craig Anderton Music Tape is a collection of original compositions recorded by Craig while he was writing HRFM.

#EPFM	$7.95
#HRFM	$9.95
#CAMT	$5.95

Order CAMT with either book and deduct $1.00!



Howard Sams' cook books are an excellent 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. "Audio IC Op Amp Applications" is a smaller version of the Op Amp Cookbook. This edition contains only the sections which are pertinent to audio circuitry.

#OACB Op-Amp Cookbook	$12.95
#AFCB Active Filter Cookbook	$14.95
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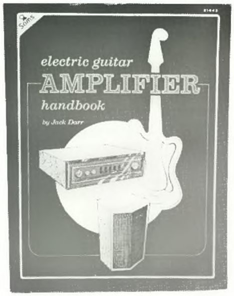

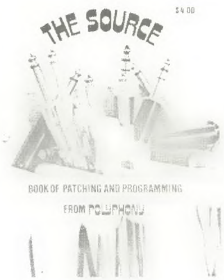

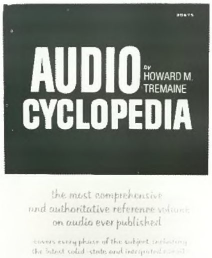

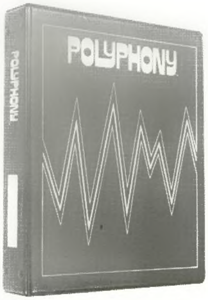

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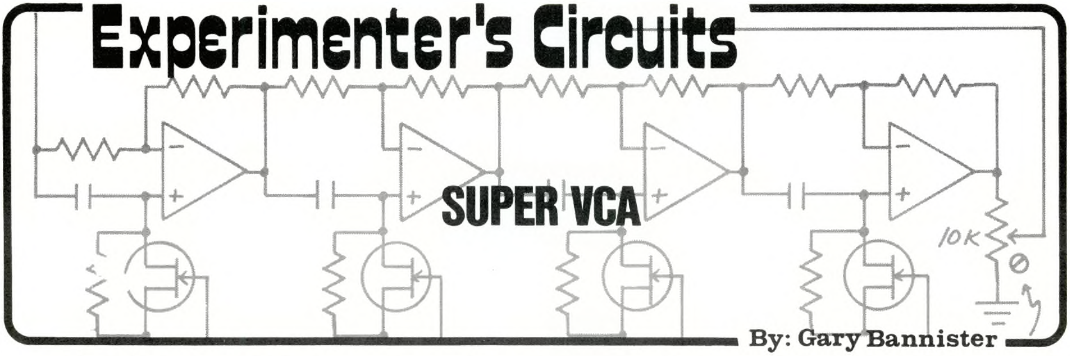


Experimenter's Circuits

SUPER VCA

By: Gary Bannister

If you built the simple VCA from last time, you may have found some alignment problems. More than likely you found that it didn't get a gain of one. That is, for a one-half volt input signal and a control of 5 volts, you didn't get a one-half volt output signal. You also probably found that the unit popped as the control voltage was changed quickly. If anything, this may be the most aggravating problem of all. Fortunately, it is also quite easy to get rid of.

In the updated schematic of figure 1, note the addition of the 100K trim pot and the 100K resistor to pin 2 of the 3080. The two ends of the trimmer are tied to the two supply voltages respectively. This trim control is used the same as all the others. While entering a quickly changing control voltage, monitor the output of the module with a scope and/or by ear. Adjust the control for minimum audible pop.

The lack (or excess) of gain is an easily cured problem, too. Since the gain of a 3080 circuit is dependant on the resistor marked Rgain, it is a simple matter to make this resistor variable. While we're here, lets also look at some other refinements on this circuit.

There is no lower limit on the frequency response of this circuit, providing that we do not voltage controlled rate and depth! The 'super VCA' schematic is shown in figure 2.

When we try to apply all of these requirements into one module, we definitely come up with a multipurpose VCA. It is capable of accepting both audio and control voltages, converting audio to control voltages, and in special cases letting you use certain control signals as audio!

Just a few notes on this circuit:

1) We haven't discussed suitable control voltage summers, but we will. Read ahead, select the appropriate summer, then proceed to the following calibration procedures.

2) The three .47uf capacitors are there to keep DC out of the audio line. For some types of equipment, these may be enlarged or left out entirely. Leaving them out will improve the low end of the system (GREATLY) but will leave you susceptible to use any coupling capacitors. This being the case, we can conceivably voltage control the amplitude of control voltages. This would allow for delayed or time-varying vibrato, 'echo' envelopes, and possibly chromatic sequencer effects.

If we make the gain of this

FIG.2

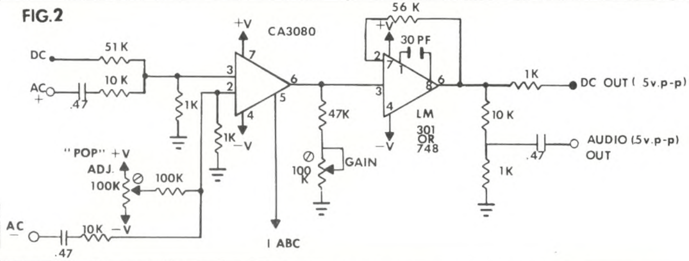


FIG.1

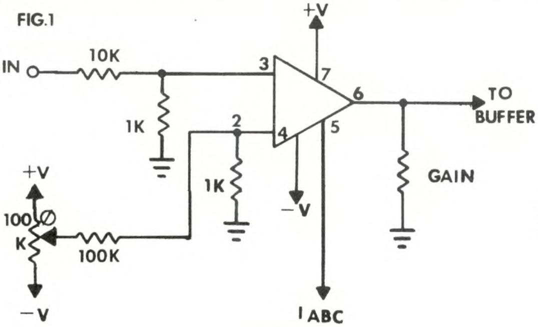
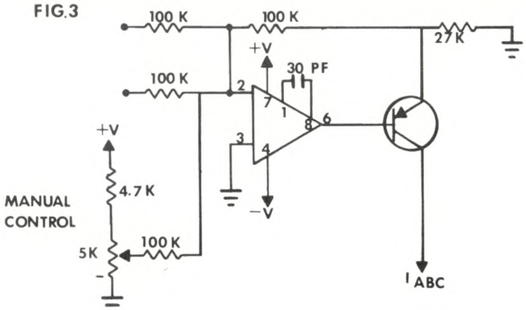


FIG.3



circuit greater than one, say to ten, then we could use our audio oscillators as control voltage sources, giving us vibrato with DC offsets if you use outside sources.

3) Notice that there are TWO AC inputs labeled + and -. This is a true differential amplifier. The audio signal may be inverted or not. This can make for some special effects with filters, phase shifters, and flangers. So there you have a super VCA.

CONTROL SUMMERS

Now let's get some IABC sources. The first would most logically be a linear VC current source as shown in figure 3. This should be quite familiar to those of you with PAIA equipment. The only change is the MANUAL control pot. This is front panel mounted, if you desire to use it. It will allow you to turn the VCA on without using a patchcord; handy when testing a patch or for a manual volume control. You've already seen the second current source; the single transistor setup is shown in figure 4. Basically, the same as the PAIA source, it simply lacks the accuracy of its counterpart. However, there's nothing wrong with it in noncritical applications, and it's certainly cheap and small.

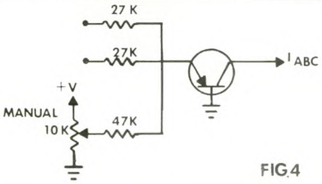


FIG.4

The last current source is a radical change from what we've had so far. Both of the above have been LINEAR response sources. For more realistic percussion sounds, and a few other uses, we need an EXPONENTIAL current source.

It has been stated many times that the ear is more sensitive to pitch than to volume, so we can get by with a little bit less precision in a VCA than in VCOs. The circuit in figure 5, although a true exponential converter, is not usually of sufficient accuracy to be used in a VCO. It probably can be aligned, but we won't bother with it here. Like all exponential converters, this one is touchy and difficult to align properly. However, in this use (a VCA) it really doesn't make too much difference, hence the noticeable lack of trim pots.

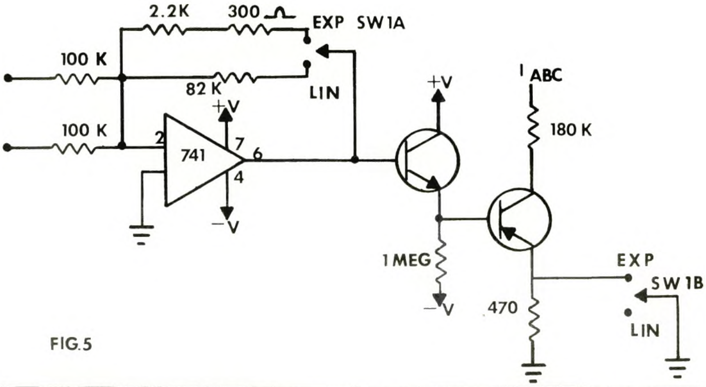


FIG.5

It may be necessary to find two suitable transistors by trial-and-error. This is the only place I've found that matched transistors could be an absolute necessity. Theoretically, the base-emitter voltage drops of the two transistors (about .6 volt) cancel since the two transistors are of opposite polarity. In practice, I could not find two transistors that EXACTLY cancelled. For this reason, in the exponential mode, you may find some more or less small signal feed-through even when the VCA is off. You should select transistors to minimize this feed through. Put the transistors in sockets and plug in and out until you find a suitable pair. Incidentally, you should find no problems in the linear mode.

As was stated, this circuit is touchy. Play with the values of two feedback resistors. Adjust the 82K for linear response and the 300 ohm (don't bother the 2.2K unless necessary) for exponential response. Generally, get the linear response working, and adjust the exponential to match.

CALIBRATION

The actual procedure differs somewhat depending on whether you want audio fidelity or control voltage fidelity. First, put in a measured 5-volt control source. If you want maximum audio gain, put in an audio source (oscillator) and adjust the GAIN trim pot for the same output as you have input, voltage-wise. If you wish greater DC accuracy, substitute a DC input voltage instead of the audio. With 5-volts control, adjust for unity gain.

When using this VCA with DC voltage, it can be thought of as a voltage controlled potentiometer. If you think of it this way, you can do chromatic transposition (like with a sequencer or keyboard) by voltage control for systems with linear oscillators. Indeed, it is so but we may need a little more accuracy, yet.

The inaccuracies stem from possible DC offset in the 301 op amp. For maximum DC accuracy, we will need to zero the 301--an easy job as you can see in figure 6. To adjust the offset, run an oscillator at some frequency, and plug in the output of the VCA. Adjust the zero control, and continue to plug and unplug the VCA until plugging it in produces no change in pitch--absolutely zero output. The 'pop' control will have some effect as well, but if it is properly adjusted it should be unnoticable.

continued on page 34 ....

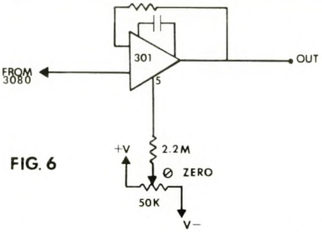


FIG. 6

NAME THAT TONE

- AGAIN!

In the July/August 1978 issue of Polyphony, Jerry Von Loh presented a feature article dealing with formulae which could be used to convert specific frequencies into musical note names and vice versa. Polyphony recieved several responses to the original article with suggestions for streamlining the approach offered by the author. The first deals with modifying the equations for use with existing systems of notation numbering rather than "defining" yet another system. The second deals with using the equations on programmable calculators since there are (most likely) a number of notes or frequencies to be determined at one sitting. Since there seemed to be a good deal of interest in this article, we decided to publish the responses so everyone could benefit. As always, further comments are welcome. - M.J.

USING EXISTING NOTATION SYSTEMS

Lucius Day
Lakewood, CO

Referring to "Name That Tone" in the July/August issue, I was disappointed that you let the author further cloud the notation problem by introducing another system of notation.

The A to A octave system is out of line with your C to C octaves of "Fundamental Music Notation" (Polyphony issue 3/76) and out of step with the more common usage. For example, let's take C7 - Von Loh defines C7 as the top of the piano; Polyphony defines C8 as the top. This is bad.

Furthermore, by using such a cumbersome system Von Loh has developed a most cumbersome and awkward set of equations. If he had considered the accepted notation of "key numbers" as used by piano tuners, his equations would have become much more elegant and simple. For example: let's define "n" as the "number" of the key on a standard 88 key piano such that

$n = 1$ = the lowest A
$n = 88$ = the highest C
$n = 49$ = reference note A

If F_n = the frequency of the n^{th} note we can write

$$F_n = K(2^{n/12})$$

where K is yet to be defined for $n = 49$ and for standard AFM 440 pitch as in

$$F_{49} = K_{440}(2^{49/12}) = 440 \text{ Hz}$$

Final solution yields

PIANO KEYBOARD CHART

Mathematical notation (with traditional notation in parenthesis). Read left to right.

							Start		**A0(A1)**	**A♯0(A♯2)**	**B0(B3)**
C1(C4)	**C♯1(C♯5)**	**D1(D6)**	**D♯1(D♯7)**	**E1(E8)**	**F1(F9)**	**F♯1(F♯10)**	**G1(G11)**	**G♯1(G♯12)**	**A1(A13)**	**A♯1(A♯14)**	**B1(B15)**
C2(C16)	**C♯2(C♯17)**	**D2(D18)**	**D♯2(D♯19)**	**E2(E20)**	**F2(F21)**	**F♯2(F♯22)**	**G2(G23)**	**G♯2(G♯24)**	**A2(A25)**	**A♯2(A♯26)**	**B2(B27)**
C3(C28)	**C♯3(C♯29)**	**D3(D30)**	**D♯3(D♯31)**	**E3(E32)**	**F3(F33)**	**F♯3(F♯34)**	**G3(G35)**	**G♯3(G♯36)**	**A3(A37)**	**A♯3(A♯38)**	**B3(B39)**
C4(C40)	**C♯4(C♯41)**	**D4(D42)**	**D♯4(D♯43)**	**E4(E44)**	**F4(F45)**	**F♯4(F♯46)**	**G4(G47)**	**G♯4(G♯48)**	**A4(A49)**	**A♯4(A♯50)**	**B4(B51)**
C5(C52)	**C♯5(C♯53)**	**D5(D54)**	**D♯5(D♯55)**	**E5(E56)**	**F5(F57)**	**F♯5(F♯58)**	**G5(G59)**	**G♯5(G♯60)**	**A5(A61)**	**A♯5(A♯62)**	**B5(B63)**
C6(C64)	**C♯6(C♯65)**	**D6(D66)**	**D♯6(D♯67)**	**E6(E68)**	**F6(F69)**	**F♯6(F♯70)**	**G6(G71)**	**G♯6(G♯72)**	**A6(A73)**	**A♯6(A♯74)**	**B6(B75)**
C7(C76)	**C♯7(C♯77)**	**D7(D78)**	**D♯7(D♯79)**	**E7(E80)**	**F7(F81)**	**F♯7(F♯82)**	**G7(G83)**	**G♯7(G♯84)**	**A7(A85)**	**A♯7(A♯86)**	**B7(B87)**

C8(C88) = last note on right.

THE PIANO TECHNICIAN'S JOURNAL · *February, 1978*

$$K_{440} = \frac{440}{2^{49/12}} = 25.956544$$

Therefor, in the 440 tuning system, the final equation for note and frequency conversion is

$$F_n = (25.956544)2^{n/12}$$

but K can be immediately evaluated for <u>any</u> reference note and any arbitrary pitch, such as

$$A_{49} = 435 \text{ Hz}$$
or
$$C_{40} = 256 \text{ Hz}$$

Note that there are no "octave numbers" with which you need to deal. And, of course, the equation can be solved for "n" for finding notes from frequencies:

$$n = 12 \ \frac{\log (F_n/k)}{\log 2}$$

This seems to me to be a lot more straightforward.

We wish to thank <u>The Piano Technicians Journal</u> (PO Box 1813, Seattle, WA 98111) for letting us reprint the piano keyboard chart from their February 1978 issue.

NOTE/FREQUENCY CONVERSIONS ON PROGRAMMABLE CALCULATORS

Terry Fitzpatrick

By now, all you techno-freaks have had a chance to try Jerry Von Loh's formulae for converting frequencies to notes and octaves, or vice versa (Polyphony, July/August 1978). Great, huh? Did one time through the calculations answer all your questions? No? Bet your fingers got tired. How would you like to eliminate about 80% of these keystrokes while greatly reducing the chances for error? Well, for those of you with programmable calculators (who can afford not to have one at today's prices), here's a program incorporating Jerry's formulae. Using his "Note Equivalency Charts", enter the frequency, press a button, and see the octave number displayed. Press another button and get the note number. Or knowing the octave and note numbers, enter them and press a

```
000 46      060 00
001 16      061 98
002 43      062 81
003 01      063 46
004 07      064 19
005 55      065 02
006 02      066 07
007 07      067 93
008 93      068 05
009 05      069 65
010 95      070 02
011 28      071 45
012 55      072 43
013 02      073 01
014 28      074 09
015 95      075 65
016 22      076 53
017 37      077 02
018 22      078 35
019 37      079 01
020 57      080 02
021 00      081 54
022 37      082 45
023 22      083 43
024 57      084 01
025 42      085 08
026 01      086 95
027 09      087 57
028 98      088 03
029 81      089 98
030 46      090 81
031 17      091 46
032 43      092 11
033 01      093 42
034 07      094 01
035 55      095 09
036 53      096 98
037 02      097 81
038 07      098 46
039 93      099 12
040 05      100 42
041 65      101 01
042 02      102 08
043 45      103 98
044 43      104 81
045 01      105 46
046 09      106 14
047 54      107 42
048 95      108 01
049 28      109 07
050 55      110 98
051 53      111 81
052 02      112 46
053 35      113 15
054 01      114 25
055 02      115 47
056 54      116 22
057 28      117 57
058 95      118 81
059 57      119 00
```

```
STEP #    CODE          COMMENTS

000       LBL A'        This routine
          RCL 17        finds the
005       / 27.5        octave number.
010       = log
          / 2 log
015       =

          INV D.MS      INTeger
          INV D.MS      truncation.
020       fix 0
          D.MS
          INV fix
025       STO 19
          prt HLT

030       LBL B'        This routine
          RCL 17        finds the note
035       / (27.5       number to be
041       X 2           used with the
          y^x RCL 19)     equivalency
048       = log         charts.
050       /(2 y^1/x
054       12)
          log =
          fix 0
061       prt HLT

063       LBL D'        This routine
065       27.5 X        finds the
070       2             frequency to
          y^x RCL 19    three decimal
075       X (2y^1/x     places.
          12)

082       y^x RCL 18
          = fix 3
089       prt HLT

091       LBL A         Stores entered
          STO 19        octave number
          prt HLT
NOTE:  no  re-entry needed unless
changing octaves

098       LBL B         Stores or
          STO 18        changes note
          prt HLT       number.

105       LBL D         Stores
          STO 17        entered
          prt HLT       frequency.

112       LBL E         Initializes
          CLR CMs       system.
          INV fix
          HLT
```

Above - Documented Program

Left - Machine Code Printout

....more....
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NAME THAT TONE

.... from previous page ....

button to obtain the frequency.

The program was written for my "antique" Texas Instruments SR-52 with printer, but can easily be modified to run on other machines, including RPN types, since the coding is straightforward. Only three memories are needed. If your machine doesn't have enough program memory, try doing the data storage manually. The Initialization can also be done manually, as can fix point formatting; also you can leave out the print codes. If your calculator has an integer truncation key (INT), that will also save a few steps. No attempt has been made to optimize the program since it is relatively short and already runs faster than you can write down the numbers.

Let's run a few examples. Enter the program, press E.

1. Calculate the frequency of B-flat 6?

   Enter octave number (6), press A.
   Enter note number from equivalency chart (1), press B.
   Get frequency by pressing D', see "1864.655" displayed.

2. What is the frequency of E6 ?

   Same octave number as last calculation, so no entry required.
   Enter note number (7), press B.
   Get frequency by pressing D', see "2637.020" displayed.

3. What note corresponds to a frequency of 60 Hz ?

   Enter frequency (60), press D.
   Get octave number by pressing A', see "1" displayed.
   Get note number by pressing B', see "2" displayed.
   Using the charts, 60 Hz = note B in the first octave.

This program makes the formulae a joy to use. For you digitizers, it should now be a simple thing to translate a score into frequencies. For the rest of us, all the good points in Jerry's article will apply. However, for example #3 above, instead of 40 or more keystrokes, you will only need 7. That's about an 80% reduction in effort and a similar reduction in time and error opportunity.

As always, Polyphony welcomes additional feedback from other readers who have comments about any of our articles. We enjoy doing these types of "reader rebuttal" articles. It really helps keep the feeling of communication.

RUSSELL...

.... continued from page 15

These electronic works of George Russell reveal the question concerning the role of technological advances that all contemporary artists (not only musicians) must answer. Russell's compositional aesthetic, based upon his extensive use of conventional instruments and musique concrete techniques, is apparent when he says that "man, in the face of encroaching technology, must confront technology and attempt to humanize it..."

DISCOGRAPHY

"Electronic Organ Sonata No. 1". Flying Dutchman FDS-122; Sonet SLP-1409.
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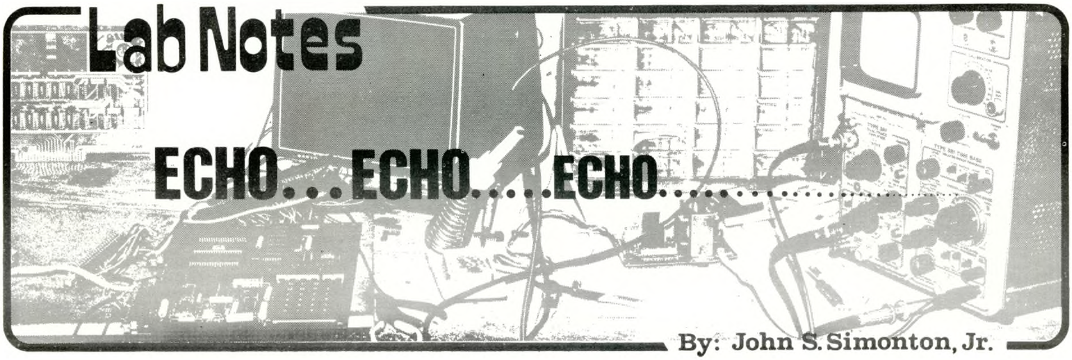
Lab Notes

ECHO...ECHO....ECHO.......



By: John S. Simonton, Jr.

A couple of issues ago, I said that we were going to look at a D/A that would allow those of you with exponential response synthesis equipment to begin playing with the computer software we have been discussing here. Then SEQUE ran longer than I thought it would, and we ran into logistics problems and .... In any case, it's not ready yet. Next time for sure.

Meantime, I've got some quickie code that I think you'll like, It's a program we call ECHO. I'll bet you think that ECHO echoes. It does.

It works in conjunction with an allocation algorithm (POLY from MUS 1 in this case, though something like Bob Yannes' SHAZAM could also be patched in to use this) and "follows" whatever data is being produced from QuASH channel #1, delaying it for a controllable period of time before playing it from a second channel, delaying again before playing on a third channel, and so on.

A convenient conceptual handle that may help you understand the "how-it-works" of ECHO might be a clock face. With only a second hand.

The numbers around the clock face represent memory locations and the second hand represents a pointer to these memory locations which, as it sweeps past each number, writes whatever note happens to be coming out of QuASH channel #1. This is really a funny clock, though, because in addition to the single second hand it has many minute hands that rotate at the same rate as the second hand. If the second hand is a "writing pointer", these funny minute hands are "reading pointers". Within some restrictions that we'll discuss shortly, we can have as many reading pointers as we like; the important feature is that each of these fast minute hands correspond to an additional QuASH channel.

Now as the clock runs, the writing pointer scans merrily through memory, writing the note that's in channel #1. In step behind it are the reading pointers, and as they point to successive memory locations they read them and place the result in the QuASH channel to which they correspond. Presto, echo.

In computerese, this kind of procedure is called a queue.

ECHO has a variety of software control features, and since I don't really know which of them are more important, we'll just plunge into the middle.

While ECHO always pulls the note that it's going to echo from channel #1, the first channel that the echo effect appears on doesn't have to be channel #2. Why? So that some channels can be set aside for polyphonic work while others are producing the echo.

Here's how. One piece of data that every polyphonic allocation subroutine must have is the number of output channels available for its use. POLY established the precedential name OUTS for this datum and set its location in a Paia 8700 as $EA.

Previously, we've always set this variable to represent the number of QuASH channels that were hardware supported. In a system which had a single QuASH, OUTS was set to contain $04 so that all available outputs were used for polyphonic allocation.

But OUTS may be set equal (may I please start saying "equal" instead of "contains"? It's not strictly true, but much less cumbersome.) to a number <u>less</u> than the number of hardware supported channels and the result will be to reserve some channels. In a system with two QuASH (for example) OUTS could be set equal to $05 and the result would be that the upper 3 channels (6 - 8) will not have keyboard activations directly assigned to them. POLY (or whatever) doesn't know they're there.

So we can use them for other things. Like echo channels.

ECHO, in its turn, must know how many channels it has to work with. The location labeled ECCO ($BB) serves this function, and in most cases will be set equal to the number of remaining channels.

To give a final example; if we make OUTS equal to $03 and ECCO equal to $05, we've produced a system which has 3 polyphonic channels (the first three) with channels 4 through 8 echoing, in sequence, the notes that appear on polyphonic channel #1.

I would be less than candid if I didn't forewarn you that successful use of a system which combines both polyphonic and echo channels requires a thorough understanding of the allocation algorithm being used as well as a certain manual and mental dexterity. It's best to start playing with a configuration which has only one channel available to POLY and the remainder used as echo channels. With practice, you can progress from there.

DELAY CONTROLS

As you certainly know by now, all timing in our system references back to the scan rate of the keyboard, and ECHO has associated with it a variable labeled EDLY ($BC) which regulates how fast (in terms of keyboard scans) the hands in our clock analogy (the reading and writing pointers) advance from one memory location to the next, which in turn contributes to how long the echo delay is.

If we set EDLY equal to $01, the echoing routine is invoked after every keyboard scan (which is variable, but typically will be every 10 to 50 milliseconds). Making EDLY equal to $02 means that the routine is used on alternate scans which, if everything else is equal, will produce an echo delay twice as long.

Notice that this affects only the ECHO and does nothing to alter POLY's allocating channels after every keyboard scan. This is important because when changing the value of EDLY you should be aware that if you skip more than about 8 scans before invoking ECHO, it may miss some keyboard activity in a fast riff. The notes will still play through the polyphonic channels, but won't be echoed.

A second variable also interacts with EDLY to detirmine the echo delay. OFST ($BD) controls the offset between the pointers into the echo queue. Going back to the clock metaphore, it detirmines how "far apart" the hands on the clock are. The farther apart they are (the bigger the number in OFST), the greater will be the echo delay.

Like EDLY, there are some caveats that go with OFST. The echo buffer (queue) area of memory is 64 bytes on page 1. You don't want to come up with too many pointers (controlled by ECCO, remember) that are too far apart or they will represent a memory area larger than that set aside. The result of that is far from disastrous, but it will cause things like the high order channels echoing much sooner than you expected, as the reading pointers for those channels "wrap around" past the writing pointer. But, as we've decided here in the past, the difference between noise and a neat effect is often nothing more than a creative mind.

Control of the time delay involved in the echo is important for reasons that you might not first think about, because like any device (or now software) that messes with the subjective flow of time, echo offers a variety of totally different effects depending on how long a time we are talking about.

For example, if the delay is very short, as when both EDLY and OFST are set to $01, the effect will not even be percieved as an echo, but rather as a "thickening" of the voice (voice doubling, actually). It's a lot like phasing or flanging, except that with those techniques the predominant effect is frequently that the subjective flow of time is cyclicly changing.

Longer delays (EDLY = $01 and OFST = $08) produce the types of effects which give ECHO its name. Echoplex type echoing. There is a major difference, though, in that with conventional echo devices you can only echo in a voice that is essentially the same as the starting voice. Here, the echoes can be anything, and there's no way to appreciate the power that this implies without working with it.

When the delays get very long (EDLY = $02 and OFST = $10) you find yourself playing with an instrument that allows you to play rounds with yourself. Also, of course, in different voices.

Because the character of the instrument is so greatly influenced by delay times, and because the different characters can so frequently be used in the same musical performance, we've added a means of quickly switching from one set of operating parameters to another. Four of these presets are provided by pads 0-3 on the command keyboard. Touching one of these pads causes ECHO to get the requested set of parameters from a table that lives in memory $9A - $A9 and place them in the locations referenced by the rest of the program. The pre-sets that are in place in the listing which follows are:

COMMAND KEY	POLY CHANS	ECHO CHANS	TIME DELAY (KBD SCNS)
0	1	7	1
1	1	7	8
2	1	3	16
3	1	3	32

Notice a couple of things here. First, if you're using a system with only a single QuASH (a P4700/J or its equivalent) it doesn't matter that there are more echo channels than there are hardware channels; the last four iterations simply won't have the hardware to voice them. Secondly, observe that when we got to longer delays we cut back on the number of echo channels so as to circumvent the "too many channels too far apart" problem that we looked at earlier.

You can substitute your own presets for those shown simply by altering or replacing the values shown. Here is a map of locations that will make that a little easier:

	PRESET #			
	0	1	2	3
OUTS	$9A	$9E	$A2	$A6
ECCO	$9B	$9F	$A3	$A7
EDLY	$9C	$A0	$A4	$A8
OFST	$9D	$A1	$A5	$A9

With some experimentation you will find echo presets which seem to complement each other particularly well. You will inevitably get to where you use a specific set of presets for each particular song, not only changing presets throughout the song but within a riff or phrase. This can create some neat effects such as having an initially long delay set and, in the middle of the echo chain, hit a faster preset to initiate a burst of echoes. Or, have one preset for the "voice doubling" characteristics we discussed. Then you can switch between echoes for special effects and doubling for use on bass lines or solos.

Actually, there is a lot of power hidden in this program that can be liberated with innovative patching, voicing, and mixing. How about having a chain of voices which are all related but slightly different, such as having higher Q on the filters as the echo is passed on. Or changing envelope times so the first echoes have sharp attacks and delays and later voices have increasingly softer envelopes. Here's a good one- progressively detune each voice so you get a spiraling echo, or the echoes sequence upscale (or downscale). Completely different voices can be used, and this technique really works well on the long delays for doing rounds.

Just playing with the mixing or panning of the normal echo voices can entertain you for hours. Have the echoes pan across the stereo field, or bounce back and forth. Or have the echoes begin to fade out, but set the last or next to last voice at a higher level.

You can also use a multi- voice setup with only a few of the outputs driving voices. Set up the computer to provide (for example) one poly voice and seven echo voices, but only use channels 1, 4, 5, and 8 to drive oscillators. Work with various combinations here; each is a completely different rhythm and could easily provide a rhythmic basis for a whole piece.

Well, by now you are probably ready to dig into the program, so here is the listing.

LOADING THE PROGRAM

As with other programs that we've examined in the past, ECHO may be hand-loaded using the 8700 computer's monitor, but first set the monitor stack pointer:

```
0-E-D-DISP-F-F-ENT
```

and the user's stack pointer and status register:

```
0-F-E-DISP-F-F-ENT-0-0-ENT
```

and then load the program:

```
0-0-0-DISP-2-0-ENT-2-1-ENT-8-D-ENT- (etc.)
```

and don't forget this data base information:

```
088-  20  21  0D  4C  C0  FF  C9  07
090-  D0  05  A0  5C  20  52  0D  4C
098-  10  10  01  07  01  01  01  07
0A0-  01  08  01  03  02  08  01  03
0A8-  02  10
0B8-  FF  FF  01  03  02  04
0E8-  40  20  01
```

After loading (and before running) the program and data should be dumped to tape (from location $000 to $0EC) using this sequence:

```
0-0-0-0-0-0-E-C-0-1-D-D-TAPE
```

When this tape is loaded in the future, it should be loaded from $000 to $0EC so that the presets will be loaded along with the program.

```
                0010  :*******************************
                0020  :*                             *
                0030  :*          ECHO 0.31          *
                0040  :*                             *
                0050  :*   POLYPHONIC VOICE QUEUING  *
                0060  :*                             *
                0070  :*             BY              *
                0080  :*       JOHN SIMONTON         *
                0090  :*                             *
                0100  :*(C) 1979 PAIA ELECTRONICS, INC*
                0110  :*     ALL RIGHTS RESERVED     *
                0120  :*                             *
                0130  :*******************************
                0140  :
                0490  :
                0500  :INITIALIZE SYSTEM, CLEAR OUTPUT BUFFERS AND ECHO BUFFER
                0510  :
000-  20 21 0D  0520  STAR JSR INIT     :CALL MUS1 INITIALIZATION
003-  A2 FF     0530       LDX 0FF      :PREPARE TO SET STACK POINTER
005-  9A        0540       TXS          :SET STACK TO TOP OF PAGE
006-  A9 00     0550  EBZR LDA 00       :PREPARE TO ZERO OUT ECHO BUFFER
008-  A2 3F     0560       LDX 3F       :POINTER TO END OF ECHO BUFFER
00A-  9D 00 02  0570  ILP  STA EBUF,X   :ZERO ECHO BUFFER LOCATION
00D-  CA        0580       DEX          :POINT TO NEXT LOCATION
00E-  10 FA     0590       BPL ILP      :NOT DONE YET, LOOP
010-  20 71 0D  0600  ECHO JSR POLY     :CALL MUS1 POLYPHONIC ALLOCATION
                0630  :
                0640  :DETERMINE ADDRESS OF THE FIRST CHANNEL AVAILABLE
                0650  :FOR ECHO USE
                0660  :
013-  A0 0F     0670       LDY 0F       :OFFSET TO FIRST OUT-BUF LOCATION
015-  A6 EA     0680       LDX *OUTS    :NUMBER OF POLYPHONIC CHANNELS
017-  88        0690  LP0  DEY          :POINT TO NEXT OUTPUT CHANNEL
018-  CA        0700       DEX          :ONE LESS POLY CHANNEL
019-  D0 FC     0710       BNE LP0      :ALL POLY CHANS NOT USED, LOOP
01B-  84 EB     0720       STY *OUTT    :SAVE FIRST ECHO POINTER FOR LATER
                0730  :
                0740  :ADVANCE ECHO BUFFER POINTER AND ADJUST IF NECESSARY
                0750  :
01D-  A6 BE     0760       LDX *EPNT    :GET CURRENT ECHO BUFFER POINTER
01F-  C6 EC     0770       DEC *CNTR    :DECREMENT TIMER
021-  D0 09     0780       BNE GETN     :TIME NOT UP, BRANCH
023-  A5 BC     0790       LDA *EDLY    :TIME UP, RE-INIT TIMER VALUE
025-  85 EC     0800       STA *CNTR    :RE-INITIALIZE TIMER
027-  CA        0810       DEX          :POINT TO NEXT
028-  10 02     0820       BPL GETN     :BRANCH IF STILL WITHIN BUFFER AREA
02A-  A2 3F     0830       LDX 3F       :OTHERWISE, RE-INIT POINTER
02C-  86 BE     0840  GETN STX *EPNT    :SAVE NEW POINTER
                0850  :
                0860  :PUT CURRENT CHANNEL 1 NOTE IN ECHO BUFFER AND
                0870  :PREPARE ECHO CHANNEL COUNTER
                0880  :
02E-  A5 DF     0890       LDA *CHN1    :GET CHANNEL 1 NOTE
030-  9D 00 02  0900       STA EBUF,X   :SAVE IN ECHO BUFFER
033-  A5 BB     0910       LDA *ECCO    :GET NUMBER OF ECHO CHANNELS
035-  85 BA     0920       STA *TEMP    :SAVE AS COUNTER
                0930  :
                0940  :CALCULATE SUCCESSIVE ECHO BUFFER LOCATIONS AND
                0950  :ADJUST AS NECESSARY
                0960  :
037-  8A        0970  LP1  TXA          :ECHO BUFFER POINTER TO ACCUMULATOR
038-  18        0980       CLC          :PREPARE FOR ADDITTION
039-  65 BD     0990       ADC *OFST    :CALCULATE NEXT LOCATION
03B-  C9 40     1000       CMP 40       :STILL WITHIN ECHO BUFFER?
03D--  90 03    1010       BCC SAVE     :YES, BRANCH TO CONTINUE
03F-  38        1020       SEC          :NO, SET CARRY FOR SUBTRACTION
040-  E9 40     1030       SBC 40       :AND ADJUST POINTER
042-  AA        1040  SAVE TAX          :PUT POINTER IN PLACE
                1050  :
                1060  :THEN PULL NOTES FROM ROTATED ECHO BUFFER LOCATIONS
                1070  :AND PLACE IN ECHO CHANNELS OF OUTPUT BUFFER (NTBL)
                1080  :
043-  BD 00 02  1090       LDA EBUF,X   :GET NOTE FROM ECHO BUFFER
046-  99 D0 00  1100       STA NTBL,Y   :PLACE TO OUTPUT CHANNEL
049-  88        1110       DEY          :POINT TO NEXT OUTPUT CHANNEL
04A-  C6 BA     1120       DEC *TEMP    :ONE LESS ECHO CHANNEL
04C-  D0 E9     1130       BNE LP1      :BUT SOME LEFT, LOOP
                1140  :
                1150  :NOTES ARE PLAYED BY CALLING THE QUASH DRIVER (NOTE).
                1160  :FINALLY, ECHO OUTPUT CHANNELS ARE CLEARED SO AS NOT
                1170  :TO CONFUSE POLY WHEN CALLED
                1180  :
04E-  20 2B 0D  1190       JSR NOTE     :CALL MUS1 QUASH DRIVERS, ETC.
051-  A4 EB     1200       LDY *OUTT    :GET FIRST ECHO CHANNEL POINTER
053-  A6 BB     1210       LDX *ECCO    :GET # OF ECHO CHANNELS
055-  A9 00     1220       LDA 00       :PREPARE TO ZERO
057-  99 D0 00  1230  LP2  STA NTBL,Y   :ZERO ECHO OUTPUT CHANNEL
05A-  88        1240       DEY          :POINT TO NEXT OUTPUT
05B-  CA        1250       DEX          :ONE LESS ECHO CHANNEL
05C-  D0 F9     1260       BNE LP2      :SOME LEFT, LOOP
                1270  :
                1280  :READ COMMANDS. 0-3; PRESETS, 4-INITIALIZE SYSTEM
                1290  :5-CLEAR ECHO, 6-BREAK, 7-TUNE
                1300  :
05E-  20 00 0F  1310       JSR DECD     :READ COMMAND KEYBOARD
061-  C9 04     1320       CMP 04       :IS COMMAND A PRE-SET?
063-  10 1B     1330       BPL NEXT     :NO, BRANCH FOR NEXT TEST
                1340  :
                1350  :THE COMMAND IS TO CALL UP A PRE-SET.  AFTER CALUCLATING
                1360  :THE BASE ADDRESS OF THE PRE-SETS CALLED FOR, THE PRESET
                1370  :VALUES ARE TRANSFERED TO THEIR RESPECTIVE LOCATIONS
                1380  :AS ACTIVE PARAMETERS. NOTE THAT THE NUMBER OF
                1390  :CHANNELS ALLOCATED TO POLY USAGE (OUTS - $00EA) IS IN
                1400  :NON-CONTIGUOUS LOCATION AND MUST BE HANDLED SEPARATELY
                1410  :NOTE THAT THE CONTIGUOUS LOCATION *TEMP IS USED AS A
                1420  :DUMMY VARIABLE AT THIS POINT
                1430  :
065-  8C 20 08  1440       STY DISP     :SHOW PRESET
068-  A9 FF     1450       LDA 0FF      :ONE LESS THAN PRESETS BASE ADDRESS
06A-  18        1460  LP3  CLC          :PREPARE FOR CALCULATION
06B-  69 04     1470       ADC 04       :THERE ARE 4 PRESET VARIABLES
06D-  88        1480       DEY          :POINT TO NEXT PRESET BASE
06E-  10 FA     1490       BPL LP3      :IF NOT THIS PRESET, LOOP
070-  AA        1500       TAX          :PUT POINTER CALCULATED TO X
071-  A0 03     1510       LDY 03       :4 PRESETS, WILL COUNT TO -1
073-  B5 9A     1520  LP4  LDA *PRST,X  :GET PRE-SET DATA
075-  99 BA 00  1530       STA TEMP,Y   :AND PLACE AS ACTIVE PARAMETER
078-  CA        1540       DEX          :POINT TO NEXT PRESET DATA
079-  88        1550       DEY          :AND NEXT ACTIVE PARAMETER
07A-  10 F7     1560       BPL LP4      :IF NOT YET DONE, LOOP
07C-  85 EA     1570       STA *OUTS    :SAVE THE MAVERICK PARAMETER
07E-  30 90     1580       BMI ECHO     :BRANCH ALWAYS
                1590  :
080-  F0 7E     1600  NEXT BEQ STAR     :COMMAND IS FOR CLEAR, BRANCH
082-  C9 06     1610       CMP 06       :IS COMMAND 5 (CLEAR ECHO) OR 6 (BRK)?
084-  30 80     1620       BMI EBZR     :COMMAND IS CLEAR ECHO, BRANCH
086-  D0 06     1630       BNE NXT0     :COMMAND IS NOT BRK, BRANCH
088-  20 21 0D  1640       JSR INIT     :SHUT DOWN SYNTHESIZER
08B-  4C C0 FF  1650       JMP BRAK     :AND RETURN TO MONITOR
08E-  C9 07     1660  NXT0 CMP 07       :IS COMMAND TUNE?
090-  D0 05     1670       BNE BRDG     :A BRANCH TOO FAR
092-  A0 5C     1680       LDY 5C       :PREPARE TO TUNE TO MIDDLE C
094-  20 52 0D  1690       JSR FILL     :SEE MUS 1.0 DOCUMENTATION
097-  4C 10 10  1700  BRDG JMP ECHO     :PLAY ON AND ON AND ON
                1710  :
                1720  :SET-UP VARIABLES FOR MUS1
                1730       .OR 10BA     :INITIAL PRE-SET
                1740       .HS 01030204
                1750       .OR 10E8     :SYSTEM CONTROL AND QUASH DELAY
                1760       .HS 402001   :AND OUTS
                1770  :AND PRESETS
                1780       .OR 109A
                1790       .HS 01070101
                1800       .HS 01070108
                1810       .HS 01030208
                1820       .HS 01030210
                1830  :
                1840  END  .EN
```

add to your:

VOICE F to V CONVERTER

BY: JOHN BLACET

In the last issue of Polyphony, we covered the construction of a F/V converter for voice application. In this article we will add a circuit that allows amplitude to voltage conversion from the same microphone. We will also look at a control voltage processor that has four touch switch controlled channels, with a control voltage attenuator for each. With these added features, the unit becomes a complete voice to control voltage processor. It is capable of a wide variety of uses on a synthesizer or with voltage controlled effects.

THE CIRCUITRY

The amplitude to voltage converter, shown in Fig.1, is simply an envelope follower. This consists of a two op amp full wave rectifier and a low pass filter. This can be connected directly to the output of the mike preamp discussed in the previous article. You may wish to adjust R1 (filter time constant) or R2 (gain) for optimum results in your application. The outputs of the two types of conversion (F/V and A/V) could lie in the same area, or the amplitude functional range could occur before the frequency function's. Controlling R2 gives you this option.

Construction is pretty straightforward. The op amp is not critical. The 4136 is a quad 741 type. Power supplies can range from ±9 to ±15 volts. Bypassing is assumed.

The control voltage processor, shown in Fig.2, has four channels, each identical. The control voltage level is adjusted with the 100K pot and selected via the analog switch section and its associated touch switch. The op amp section buffers the output.

The touch switch is a simple set-reset latch or flip-flop that has its inputs held high with 10M resistors. When either touch switch gap is bridged by a finger, that input goes low and latches. The output goes high when the on switch is touched and visa-versa. The LED indicates the status of the channel. This is a handy circuit. It would be easy to replace the analog switches with ordinary toggles and save some board space, but touch switches are fun--and fast enough to be played!

The construction of the touch switch itself is easily accomplished on a PC board. The touch plates should be about 1/4-1/2" across with as narrow a gap as possible. Position the "ON" switch above the "OFF" in the manner of regular switches. Leave enough room between switches so that the finger will not overlap. Run the inputs up the board and the ground line along the bottom.

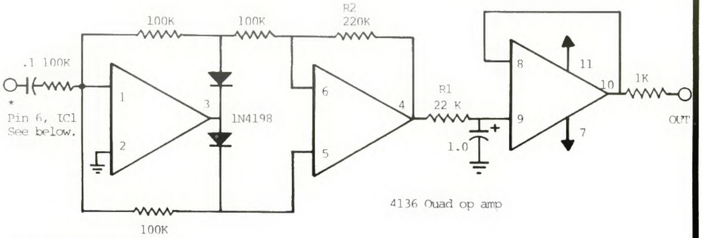
After etching, solder plate each pad with a soldering iron. Using fine sand paper or steel wool, polish the solder bumps until they are relatively even. Then paint the PC board with an opaque color everywhere except the actual touch pads and the in-out connector area. This procedure produces optimum results. However, much cruder arrangements will work.

MODIFICATIONS

There are no particular limits as to the number of channels you can connect to one control voltage source. In this particular case, two pair are used--one pair for the "frequency to voltage" output and another for the "amplitude to voltage" output.

This control voltage processor can be used for other applications. The supply voltages for the 4066 analog switch must cover the input voltage range, however, and the total must not exceed 15 volts. This can be ±7.5 volts, or +15 volts to ground, or any other combination. Since the 4066 uses very little current, simple resistive dividers in the 10K ohm range can supply the proper voltages from the main supply.

For use as an audio mixer with touch switching, leave the pot inputs seperate and place a



* See previous article.

Fig. 1 FWR and filter for amplitude to voltage conversion.

capacitor on each input. Use a single op amp summing section with 100K input and feedback resistors in place of the four voltage followers.

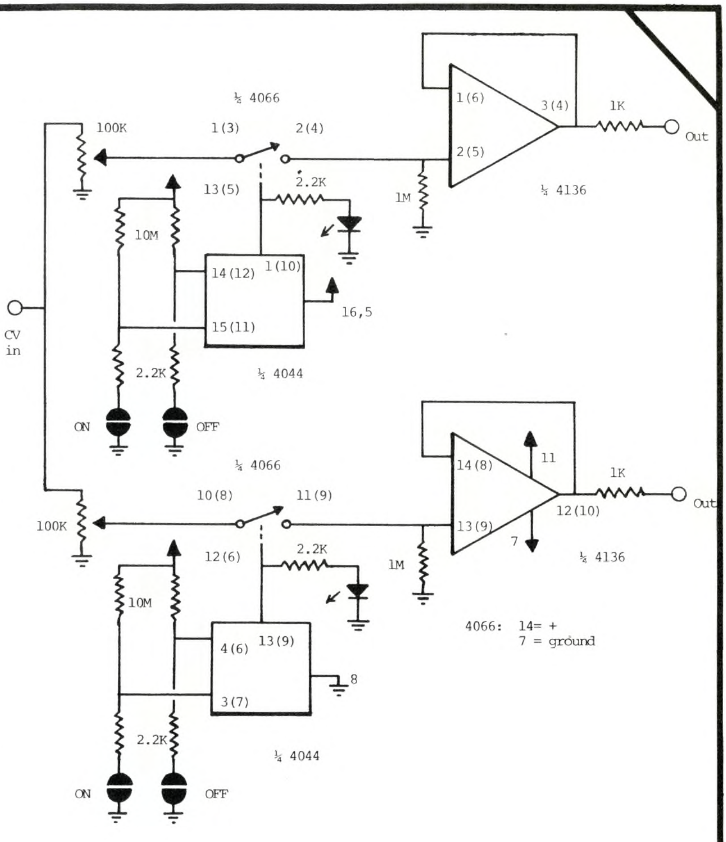
THE APPLICATIONS

The patches show two types of uses for the Voice to Control Voltage Processor. In Fig.3 a drum sound is changed in pitch and held until the next input. This is a nice way to change a percussive background you may be playing along with.

In Fig.4 both sections of the processor are used, one for each modifier. This is a very general patch with practically any instrument at the input and any two or more voltage controlled modifiers. With a little practice, you can keep the "frequency to voltage" controlled modifier static (by maintaining a steady pitch) and move the "amplitude to voltage" controlled modifier. With both voltages varying, a nice subtle control can be added to your instrument. The touch controls allow quick switching of modifiers in and out of the system. Many other patches for this module will become apparent as you use it in your system.

Jean Michel Jarre's new album, EQUINOXE, (Polydor PD-1-6175), has a novel sounding effect about half way through band 4. This sounds suspiciously like the analyzer outputs of a vocoder controlling VCOs and VCFs. It is not so much a vocal sound as a vocally derived one.

Our goal at the start was to add some control flexibility to the synthesizer. Now we have two parameters of the voice as controlling elements and this should be a significant step in the right direction ●



4066: 14= +
7 = ground

Fig. 2 Control voltage processor. One-half shown. (Numbers in parenthesis are for the identical second half.)

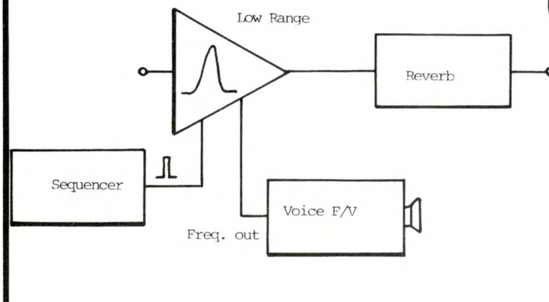


Fig. 3 Drum patch.

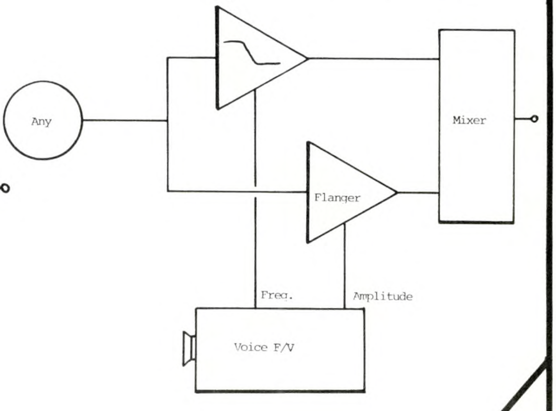


Fig. 4 Multiple effect patch.

SUPER VCA...

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WHY EXPONENTIAL?

Before I go, just a short dissertation on WHY an exponential VCA. It is a known fact that the amplitude response of our ears is logarithmic, that is, we hear loudness in an exponential way. Without getting into decibels, LLs, and phons, let us say that for a sound to SEEM twice as loud it must have about 10 TIMES more power (electrically).

In practical use, let us design a patch. We have a certain percussion sound that we wish to 'crescendo' to a climax, so we derive a patch as shown in figure 7. This will do the job, but it's somewhat wasteful of equipment. We can do the whole thing with the single exponential VCA depicted in figure 8. This is a pretty standard patch, but takes on new light when seen in the exponential mode. In this manner, the last two events will seem to increase in loudness by the same amount as the first two, something that can't be done in the linear mode.

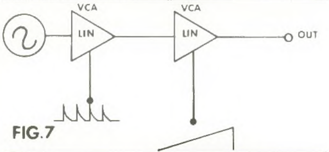

FIG.7

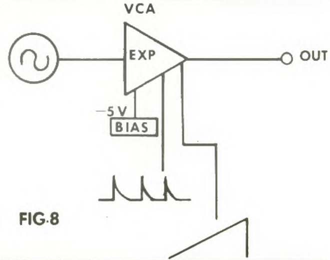

FIG.8

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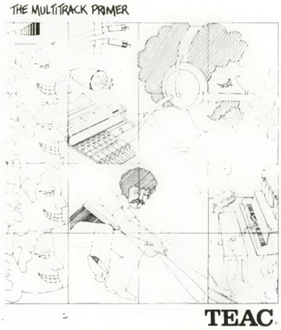
Introduction to the Physics and Psychophysics of Music; Juan Roederer; Springer-Verlag New York, Inc.; New York, New York

REVIEWS...

....continued from page 7

album should be interesting; if you are interested in guitar synthesis- even more so. This album is doing very well in Canada, and is just beginning distribution here, so it may be hard to find. If so, check out the ad which appears elsewhere in this issue for ordering information.



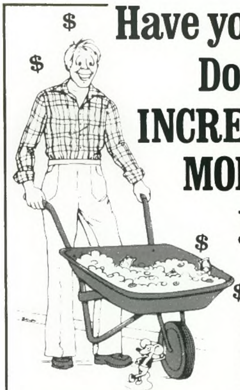
The Multitrack Primer
by Dick Rosmini

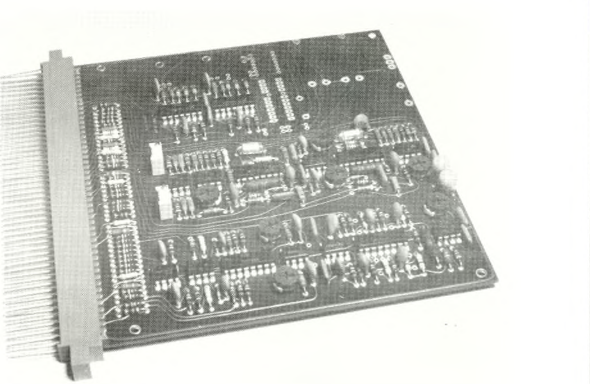
Available at local Teac dealers for $4.95, or directly from Polyphony Bookpage for the same price (postage included).

"The Multitrack Primer" has been published by Teac primarilly as an introductory guide to those Teac customers who are just beginning the multitrack game. Naturally, most of the examples and wiring diagrams in the book use Teac devices as examples. But, nevertheless, there is a great deal of useful information in here for home recordists using any type of gear, and at most any degree of proficiency.

The natural beginning is setting up the equipment, interconnections, and room layouts for various types of instrumentation. Following this, a basic course in technology is presented (dB, impedances, nanowebers, sync functions, and acoustics) to provide a firm foundation for the first experiments with the system. Quite a bit of space is devoted to preparing the home studio acoustically for best results with miked recordings. With quite a bit of detail, the author covers the techniques for building a 'room within a room' for maximum isolation and minimum coloration by the original room. He even provides construction alternatives for those of you who are renting and don't want to drill or nail your walls to death. Nice touch. Another large section is devoted to mikes. Techniques are discussed for measuring the polar response of microphones, and then how to use these response variations to your advantage when miking guitars, vocals, and drums. Many people getting started with recording seem to have lots of questions about miking technique. You've probably heard many people say that the best way to learn about miking technique is to start working with it and learn from experience. That's mostly true, but this book should at least provide a good introduction and foundation for the more advanced texts or experience.

Overall, this is a good place to get the basics. There are a lot of diagrams and a lot of basic, easy-to-understand explanations of technical details. The primary area in which the book is lacking is





....continued from page 17

Santa Cruz, CA 95060). The complete board contains 2 VCOs, 1 VCF, 2 VCAs, and 2 VC Transient (Envelope) Generators. E-mu, manufacturers of modular electronic music systems, have designed the board for hobbyists and also for OEMs interested in evaluating the ICs.

The $100 kit includes the 6 custom ICs (list price $52.50), a 6" X 6 3/8" circuit board, 3 special temperature compensating resistors and 2 polystyrene capacitors. The VCF section can be wired to provide 24db per octave low pass or high pass functions, or an all pass filter for phase shifter effects. The user must obtain standard ICs, resistors, capacitors, and miscellaneous components. All inputs and outputs are wired to a 100 pin .125" center edge connector (S-100 size, but not S-100 plug compatible). The board can be powered by ± 15 volts @ 100 mA, or the on board power supply can be added to interface directly to 110 VAC.

The kit, including documentation (schematics , assembly diagrams, parts list, IC spec sheets) is available from E-mu Systems for $100 postpaid. E-mu does not sell miscellaneous parts, but supplies the kit with all parts required for $250, and a fully assembled and tested board for $450. The documentation alone is $2.00, the double sided board alone is $50, the SSM 2020 Dual VCA and SSM 2050 VCTG ICs are available for $7.50 each, and the SSM 2030 VCO and SSM 2040 VCF ICs are $10.00 each. For more information or orders, contact E-mu Systems at the above address.

REVIEWS...

experiments with the recorded signal after (or while) the initial tracks are down. Little is said about mixing, submixing, noise reduction, line level processing, tape manipulation via speed changes or splicing, punch-ins, or even maintenance procedures for day - to - day operation. Perhaps they have other books to cover these concepts, or intend for the user to acquire this from his equipment instruction manuals. Fortunately, there are other 'home recording' books readily available which cover these additional topics, but which are a bit light on the basic set-up procedures covered in "The Multitrack Primer". Used in conjunction with the other books, this should provide answers to nearly all your questions about getting your first home multitrack studio set up and into production.

JOT...

....continued from page 10

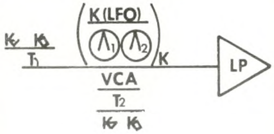


FIG.6B

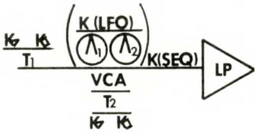


FIG.6C

to be able to decide on a "main element" (which is usually the module with the most input and output connections). He must also be able to distinguish between signal and control sources. For this reason it is recommended that beginners in electronic music become proficient in flowcharting before tackling Jot. Most magazines that deal with electronic music have carried articles explaining flowchart notation at some time or other and should be a great help in this area.

With moderate practice anyone can become proficient in Jot. Once mastered, it can render great savings in time and writing space, and may help to increase one's understanding of system interconnection. Of course, further refinements would be welcome and should be encouraged. If you have ideas, send them to Polyphony for future recaps of this article, or as a letter to the editor. Let us all use each others ideas. Perhaps the most exciting of these would be a Braille adaptation of Jot for sightless musicians, who may now find the rambling lines of flowchart notation impossible to keep track of. Whatever the modification, it is hoped that this system will keep the musicians in the electronic medium thinking, talking - exchanging ideas! We are still pioneers.

Equipment Exchange

A place for our readers to offer for sale or trade equipment related to music and electronics. Keep listings as brief as possible and enclose $1.00 for each listing. Persons responding to ads should write directly to the other party. DO NOT write to POLYPHONY. Polyphony is not responsible for any claims made in ads or results of transactions. We reserve the right to refuse or edit any ads submitted.

Technicians Wanted: to work on state of the art Prophet synthesizers. Analog and digital experience req'd, microprocessor and music background helpful. We are a small, rapidly growing company with a good working environment. Write or call: Sequential Circuits, Inc., 1172G Aster Ave., Sunnyvale, CA 94086, 408-296-3116.

For Sale: Synthi AKS with touch sensitive keyboard, joystick, and pin matrix patching. recently recalibrated. $1900. Brian Horner, 2316 1/2 Fair Park, Los Angeles, CA 90041, phone (work) 213-685-5141.

For Sale: (3) 2720-2A VCO's modified for greater range and linearity, and lower distortion. Includes range controls. Two units have waveform switches. Also: 2720-14 Sine/PWM. All units exceed factory specs. Sold as set only, $100. Gary Bannister, 7208 New Augusta Rd., Indianapolis, IN 46268. Please write before sending money.

For Sale: Crumar Organizer w/legs, like new, $950. Paia Gnome, perfect condition, $90. Would like to trade for Apple II or sell. Bill Boydstun, 2207 N.W. 20, Oklahoma City, OK.

For Sale: P4700J partially assembled and working. Heavy duty hardware, shielded patch cords, 8700 computer with cassette interface, software; tested and certified working by Paia techs. Sold to best offer due to disabling injury. Leo Hasting, 124 N. Fairview, Lansing, MI 48912, 517-371-5760.

For Sale: MXR Phase 90, $90; MXR distortion, $50; Phlanger, $60; Gnome, $40. Everything works perfectly. Dan Collins, Maple Lane, Blandford, MA 01008, 413-848-2875.

For Sale: Oz and Gnome, assembled and factory tested. 2 months old, will sell together or seperately. Will deal on used Stringz 'n' Thingz. Robert Hudler, P.O. Box 487, Fairfax, VA.

For Sale: 2720-4 Function Generator, perfect condition. $20 postpaid. Thomas L. Henry, 3801 Cornerwood Ln., Apt. 14, Charlotte, NC 28211

For Sale or Trade: 2700 modules in custom case, 4700 modules in 4761's. 8700 Keyboard with built-in D/A and glide. Also for sale, Paia Synthespin. Prefer sale as set, but will sell modules. SASE for prices to: Brian Parks, 342 Highland Ave., Johnstown, PA 15902

For Sale: 2720R plus -9, 4710, 4720, 4730, 4740; assembled in custom cabinet, many accessories. Meets or exceeds factory specs. $500 value, sell for $350 or best offer. Robert Smith, 6229 Osler St., San Diego, CA 92111, 714-560-6835.

For Sale: 2720R, fully assembled. 3 years old, but like new. Steve Carmincke, 9905 Lancet Ln., Oklahoma City, OK 73120, 405-751-0607.

For Sale: 1551 Stereo Option for Paia Stringz. Assembled but not tested. $40. Gerard Pardeilhan, 1600 Baker St., San Francisco, CA 94115

For Sale: P4700J completely assembled, factory calibrated. $800. Rene Curbelo, phone 305-595-8300.

For Sale: Paia 2720R synthesizer, 2720 VCO, and OZ mini-organ. All or part, make offer. Chris Barrish, 22230 Euclid Ave. #401, Euclid, OH 44117, (216) 259-5905 or 531-5332.

For Sale: P4700J Computer Synthesizer. New, never used, all parts working. D/A and Quash need assembly finished. $750 firm. Also 3 watt blocks, mixer, modulator, and sequencer in road case. Some need work, $50. Also have unassembled keyboard and road case, $60. Frank Rogala, 4698 Barnum, Hastings, MI 49058. Phone 616-367-3951.

For Sale: 2720-1, (2) -2A, -3B, -3L, -4, -5, -7, and -8 with glide in road case, -14, 4710, 4740, 4720, 4780 with custom panel, 4730 (unassembled), 1702, 3712 assembled board, and (2) 4770. All work, needs case. $350 plus shipping. David Shade, 45B Parkway Apts., Cherry Hill, NJ 08034, (609) 795-1829 or 677-1570.

Local Happenings

The following people wish to contact other synthesists and electronic music enthusiasts in their area to organize ensembles and exchange information. This column can also be used to list local concerts, meetings, or workshops. Soon, we hope to feature concert listings and an event calendar in this column.

Frank Sole
30 Britten Rd.
Green Village, NJ 07935

Wilfried Oswald
506 Chelsea Ave.
Winnipeg, Manitoba
Canada R2K 1A3
(204) 668-9812

INDEX-VOLUME 4

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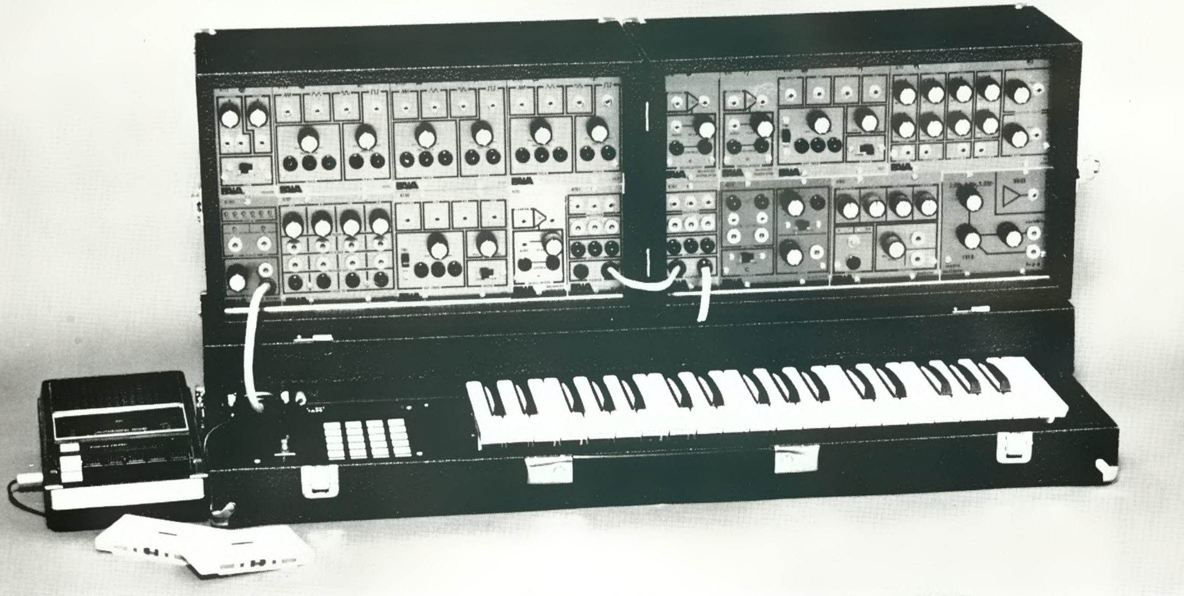

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System firmware includes: PIEBUG - system monitor; POT- SHOT - cassette interface and MUS 1. 0 synthesizer operating system.

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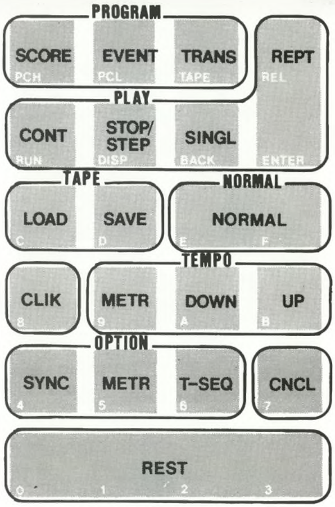

Typical control panel configuration

( ) Sounds intriguing, but I need a <u>lot more information</u>. Please send the most recent edition of your "Friendly Stories About Computers/Synthesizers" . . . . $3. 00 postpaid

( ) Please also send complete instruction manual set for the P-4700/J . . . . $10. 00 (refundable upon purchase of P-4700/J Kit)

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