POLYPHONY

Sept./Oct. 1979
$1.50
ISSN:0163-4534

ELECTRONIC MUSIC & HOME RECORDING

BUDGET EQ

LFO PROJECT

COMPOSING ON SYNTHESIZER

JAN HAMMER

PHASE MODULATED SYNC

PATCHES

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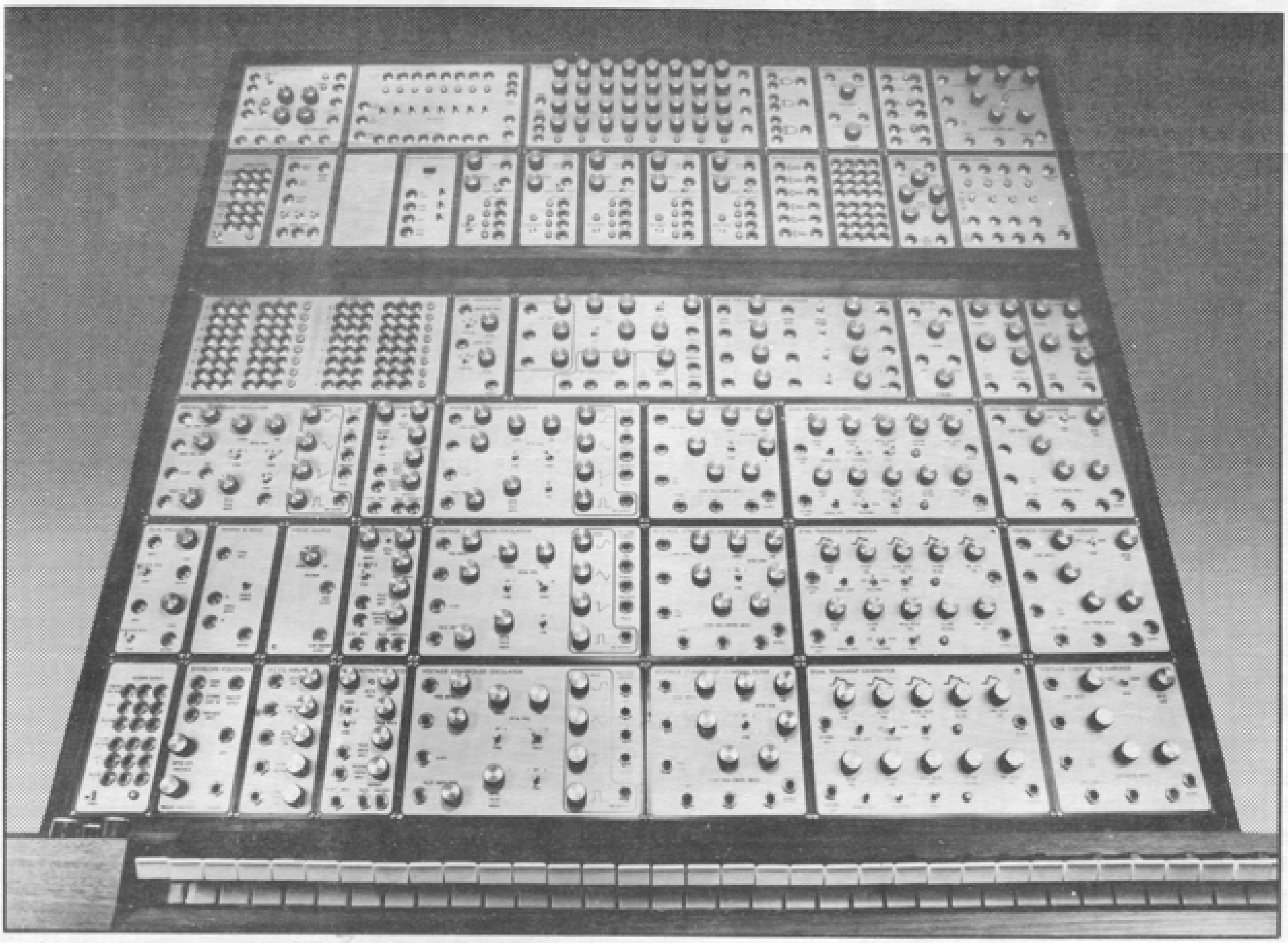

The State of the Art

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For the definitive modular system reference, send $5.00 for the 100 pg. Eμ Systems Technical Catalog, with photos, functional descriptions, and specifications of all Eμ products. (Calif. residents be sure to include sales tax.)

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Not because we say so, but because they say so. . .

"The *'New Yorker'* of audio magazines"

—ESS, Input, Sacramento, CA

Audio Amateur is a magazine that continues a great American tradition—a tradition that loves tinkering and experimentation and embraces rather than eschews technology. Readers of this magazine, I suspect, don't simply discuss the latest heavily advertised "quantum leap" forward. **TAA** subscribers are impressed more by an interesting project they can build from scratch. They love to extract, by modification, the greatest possible perfection from classic and recently introduced audio products.

Like the **New Yorker,** the **Audio Amateur** publishes articles that are measured and thoughtful, articles that are beyond superlatives by the bushel basket found in most of the mass circulated audio magazines. The reasoned tone results in part from the considerable contributions made by English writers, including the late B.J. Webb. Edward T. Dell, Jr., the editor, almost always includes a thoughtful editorial that, alone, is worth the cost of admission. Unlike some of the little audiophile magazines, **TAA** is generally beyond clannish allegiance to a few manufacturers. Articles on projects to construct and modify appeal to the fondness of its readers for a wide range of projects.

Audio Amateur has served up a smorgasbord of projects over its ten year existence. How to properly adapt a Grace arm to an AR turntable, build a record cabinet, modify a Formula-4 tonearm to improve low frequency reproduction, or build a 10 dollar three-element Yagi antenna have all been offered as appetizers, projects that require some familiarity with tools and a few nights of your time. The main course offerings demand various degrees of more sophisticated electronic skill. If you've only assembled a one tube radio (twenty years ago), many of the electronic projects are going to be more than you can chew. Numerous past articles have shown how to improve classic Dynaco products. Recently, Nelson Pass of the Threshold Corp. discussed how to build a 40 watt per channel class A amplifier. Electronic articles typically assume an ability to find the parts necessary to build the projects. Chances are you'll spend some time searching through parts catalogs and local surplus houses before you can begin to wade into the actual construction.

Sophisticated articles that examine specific audio problems but do not involve building projects also abound. Walt Jung, contributing editor, has discussed slewing induced distortion in amplifiers in a series of articles. How we actually perceive sound and how many speakers may be necessary to recreate the closest possible approximation of the live event has also been discussed.

If speaker building is your forte, past articles have dealt with horn loaded and transmission line designs. Instructions on how to build electrostatic transducers from scratch, and box fabrication for sub-woofers with an accompanying active crossover have also been features. It's a measure of **TAA** contributor ingenuity that a complex driver like the Heil air-motion transformer has been built by an amateur — complete instructions on how to build a home version of the large Heil appeared in the magazine in 1977.

An excellent analysis of recently introduced audio kits is a regular feature. Kit reviews are technically very thorough and are often more objective than you find elsewhere. A regular feature, "Audio Aids," offers all kinds of informative hints from readers. A letter section from readers comments on past articles and present concerns and lends a thoughtful and inquiring tone to the magazine. Advertisements, themselves, are often helpful to the reader since many of the ads list parts that are vital for project construction. Most of the better kit manufacturers also advertise in **Audio Amateur.**

If you are already an audio craftsman, or would like to become one, **Audio Amateur** is an excellent touchstone. For less than the price of a good meal and a movie ticket, you can receive four issues a year.

—George Hortin, Staff Writer

INPUT, published by **ESS, Inc.**
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THE AUDIO AMATEUR is the only U.S. publication I know of devoted exclusively to the home builder and experimenter in audio. The major distinction of this magazine is that it is written by doers. Thus its pages contain useful information, not just another collection of mystic reviews. Its information content on construction projects, sources of parts, and basic audio and electroacoustic theory make it one of the outstanding values for the amateur.

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NATIONAL ASSOCIATION OF MUSIC MERCHANTS
MEMBER NAMM

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September/October 1979

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Getting Started in Home Recording
Real-Time Parameter Control of Computer/Synthesizers
Wireless Mic Technology and Application
Problems and Solutions for Live-Plus-Tape Composition
Plus- New Columns, Contests, Projects, Patches, Software, ...

The length of a patch cord is inversely proportional to its need.

EDITORIAL

With the release of this issue, Polyphony will have completed its fourth calendar year of publication-- 16 issues. That's really not bad for a specialized magazine dealing with such new technology. But it is appropriate that we take a moment to reflect on some of Polyphony's original goals, and see how we are doing.

The best single word I can think of to represent the initial intent for Polyphony is -- communications. The electronic arts were, and still are, a very young field. Most of the people involved learned what they needed to know the hard way- scraping together manufacturers spec sheets and users manuals, trial and error experimentation in primitive 'tape oriented' studios or ill-equipped college music labs. It wasn't all easy or fun but, as most of you know, once the bug bites it's hard to get that sonic mirage out of your mind. So we persevered and grew with the industry.

When Polyphony started, it provided a communication link between similar minded people who were far apart and who otherwise might not have been able to learn how others accomplished their music magic. After all, there is no reason for one person to go through the entire experimentation and development facets of creating new techniques or equipment if someone else has already reached a point of functional utility, or even perfection. I like the old adage about 'no need to re-invent the wheel', because it seems like so many people are doing exactly that. The material we cover is not the type of material you can learn of from reading the evening paper or watching the 10 o'clock news. There must be a communications link specifically for our industry in order to maintain educational and technological advancement. Other magazines help the field in general, but seem to tend towards either personality features or highly specialized technical dissertations. There is a definite need for a 'Polyphony' to provide advanced applications for the majority of the 'working' musicians.

The industry is starting to recognize what Polyphony is doing, and is helping support our communications link in many ways. In turn, I have had excellent reports from our advertisers concerning the inquiries and responses of the Polyphony readers. Thank you for supporting them, because it supports us and your communications link, and allows us to expand our communications to include a number of related areas. Be sure to always mention that you are a Polyphony reader when responding to ads and product announcements.

As a publishing or communications oriented company, we see Polyphony as having the capability to do much more than publish one magazine. Communications and education can take on many forms. Concerning the field we are directly involved with- electronic music and home recording- the obvious first branch to explore is a record label. At this point I can almost guarantee that this will happen. Several plans have been in the works for nearly a year, and are preparing to emerge. The Polyphony label is founded on the fact that most larger labels refuse to get involved with any type of experimental music. And then they wonder why people are bored with new music, and why album sales are going down. Obviously, we know something they don't. We know that experimental artists, and even the amateur musicians, all have something to be shared with other people- something that must be shared in order for us all to find new avenues of exploration and expression. Polyphony intends to give the artist the voice he should have had all along.

Other current projects include publishing more books (we've got a couple in the works; if you know anyone writing a book, have them get in touch with us), finding more materials (books, records, tapes, supplies) to make available to you through our Bookpage (we envision Bookpage as becoming a sort of clearinghouse for hard-to-find items dealing with experimental music), and someday perhaps starting some regional electronic music expositions, production of educational and/or artistic video releases, and who knows what else. But....

The search for a means to provide these increased facilities for our use brings us full circle to Polyphony's founding principle.... communications. It doesn't take a dictionary to realize that communications is a 'two-way street'. We need your help. The industry needs your help. Your fellow experimenting musicians need your help. Tell our advertisers that you are a Polyphony reader, and what you like about their equipment, and what new products you need. Tell your local radio stations to play more electronic music; get them to set aside an hour a week for experimental or self produced music. Tell your friends and fellow musicians about Polyphony; talk to music stores, college book stores, record stores, and pro audio stores about who and what Polyphony is - and the possibility of becoming a dealer for the benefit of their other customers. Many of you have expressed an interest in a thicker or more frequent Polyphony. We'd love to do it, but the only way it's feasible is with increased circulation. Tell Polyphony what you are working on, write articles for us, send us patch charts, send us your criticisms. Polyphony was, and still is, an information exchange medium which is heavily based on reader contributions. We have an increasing need for material due to our newer high density format, increased publishing schedule, and possible increase in pages or frequency. DON'T THINK THAT YOUR MATERIAL DOESN'T MATTER. It does. DON'T THINK THAT YOUR MATERIAL ISN'T GOOD ENOUGH. It is. We have an editor; we have an artist. We can make your ideas presentable; if there isn't enough information, we'll get back to you with ideas on how to expand your article. We've developed a useful, accepted format to help spread our thoughts. We want to help you; we want you to help us. We want to .... communicate ....

It's your turn.

Marvin

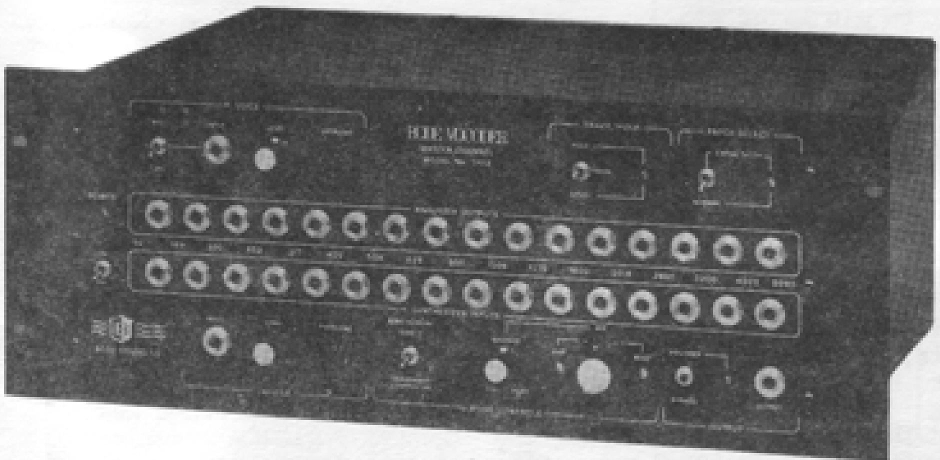
Tell them you saw it in POLYPHONY

INDUSTRY REPORT:

Experimenters Boards

Gentle Electric announces the availability of their Model 101 Pitch and Envelope Follower circuit boards to manufacturers and experimenters. The board sets are fully assembled and tested and include schematics, PC board layouts, suggested interconnections, trim procedure, and owner's manual. Two boards are included in each set. The Amplitude circuit board is 5.125" wide X 6" long with a 22 pin edge connector. The Pitch circuit board is 5.125" wide X 5.25" long, also with a 22 pin edge connector. All I/O and controls for each board are on the edge connectors. The Pitch board contains two potted modules: an exponential current sink (contents shown on schematics), and a fundamental extractor circuit (contents not shown). The circuit boards are designed to stack with the Pitch board on top to allow easy access to all board trimmers (7 total).

Board sets are $275 per set; $250 each for six sets. Fundamental Extractor potted modules (with applications assistance) are $150 each; $100 each for six. Schematics are $200 (applicable toward purchase). For more information, contact: Gentle Electric, 130 Oxford Way, Santa Cruz, CA 95060, Ph. 408-423-1561.



New Vocoder

Bode Sound Co. (1344 Abington Place, No. Tonawanda, NY 14120, Ph. 716-692-1670) has announced the release of their new Model 7702 Vocoder. This vocoder features 16 analyzer and 16 synthesizer channels in the vocoding range, extending from 50 Hz. to 5080 Hz. A patented high frequency section in the non-pitched range from 5080 Hz. to 15,000 Hz. virtually adds the effect of another 6 vocoding channels, thus creating the performance of a 22 channel vocoder. With a response time of only 6 milliseconds, the model 7702 is capable of processing drum sounds, violin pizzicato and the like. The analyzer outputs and synthesizer inputs are accessible through phone jacks, and cross patches for sound scrambling can be made and activated or deactivated by manual or foot control. A Sample/Hold control holds entered vowels when desired. The model 7702 has a built in voiced/ unvoiced sound selector, a noise generator for the "s" sounds, a hiss/ buzz balance control, various mode switches, overload indicators for voice (program) and carrier inputs, a MIC-LINE switch for the voice input with the nominal mic input level being -40dBm (7 mv) and the nominal line input level being 0 dBm (.7 volts). The maximum output level is +15 dBm (4.4 volts), signal to noise ratio is better than 70dB. Input impedances are 20K ohms minimum; output impedance is 600 ohms. The unit is 19" wide, 7" high, 12" deep (without controls), and weighs 17 lbs. Suggested retail price is $5600. A stereo demo cassette is available for the 7702; for more information, contact the manufacturer directly.

New Synthesis Chips

Paia Electronics, Inc. announces their selection by Curtis Electromusic Specialties as exclusive small quantity distributor for CES's new line of integrated circuits for audio processing and music synthesis. The Curtis ICs are wide range, low noise, second generation devices which represent a higher degree of integration than similar devices currently available, requiring minimum external support circuitry.

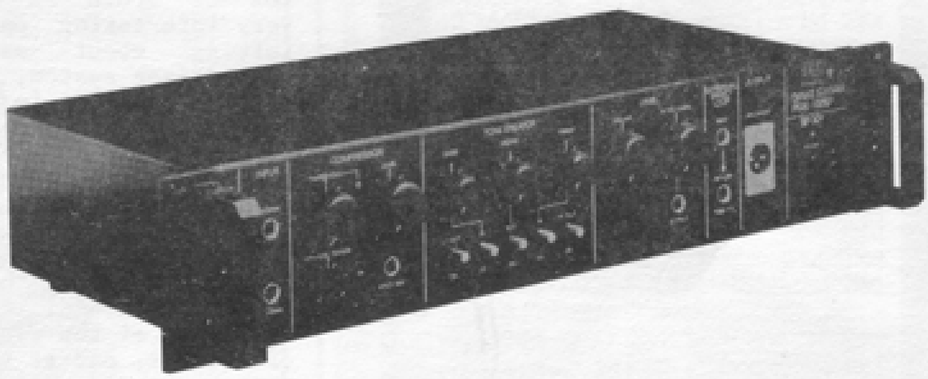
The CEM 3310 is a voltage controlled envelope generator with a typical time control range of 250,000:1. Attack, Decay, and Release times are exponentially related to control voltage input, and control voltage feed-through is for all practical purposes nonexistent.

The CEM 3320 is a high performance voltage controlled four pole filter with on-chip voltage controlled resonance and temperature compensation. The 3320's four independent sections may be interconnected to provide a wide variety of filter responses.

The CEM 3330 Dual Voltage Controlled Amplifier provides such innovative features as simultaneous linear and exponential control response and virtual ground summing nodes so that signal and control mixing can be accomplished within the device itself.

The CEM 3340 Voltage Controlled Oscillator is a full feature device which directly provides ramp, triangle, and voltage controlled pulse waveforms. Both exponential and linear response control voltage inputs are available, and hard and soft sync inputs are provided. The chip is fully temperature compensated so that external tempco resistors are not required.

Small quantity prices are: 3310 and 3330, $7.95 each; 3320, $8.95 each; and 3340, $10.00 each. Complete applications documentation is provided with each chip. A $2.00 shipping and processing charge is required on each order. For more information, contact: Paia Electronics, Inc., PO Box 14359, Oklahoma City, OK 73114.



Rack Equipment System

RolandCorp US is introducing The Roland Rack, a system featuring complimentary products that provide a complete amplification and effects package for the performing and recording musician. Included in the Roland Rack products are two models of instrument preamplifiers, two power amplifiers, and five different signal processing devices; any and all of which can be housed in the rugged Roland Rack case. The system concept allows the musician maximum flexibility in tailoring his amplification and effects set-up to his individual taste. Rather than being limited to the control features found on conventional guitar amplifiers, the musician can build his system by adding the desired effects where he chooses, gradually or all at once.

The Roland Rack products and the Rack itself are built to existing industry standards of 19" in width and 3.5" in height. The Rack itself has 24.5" available rack space, which

continued on page 14....

Letters

CA3080 Input

Having long found the CA3080 OTA to be useful for electronic music, Gary Bannister's "Experimenters Circuits" in your recent issues are certainly on the right track. I would like to mention however one point of possible misunderstanding with regard to the input level of the CA3080. Gary states (Nov/Dec 1978 issue, pg. 19) "The input voltage on pin 2 or 3 cannot exceed 100 millivolts. There seems to be some confusion on this by the way. Some sources recommend no more than 10 mv, others up to 200 mv. My experience is that 100 mv won't hurt."

In fact, this choice can be pinned down quite well, and is a matter of both theoretical and experimental record. The input stage of the CA3080, a standard two transistor differential pair, is inherently non-linear. As soon as you apply any voltage, it starts to bend over at the output. The engineering trade-off is to choose a large enough signal voltage so that signal-to-noise ratio is not terrible, and a small enough voltage that non-linearity does not produce excessive harmonic distortion. This is a very tight corner in fact. Most designers have chosen to limit the input level to ±10 mv or ±20 mv at most. The ±100 mv Gary suggests is totally excessive from a linearity point of view. If you input a triangle wave at this level, it will be rounded beyond a sine wave. Incidentally, about ±80 mv makes a pretty good triangle-to-sine converter.

Therefore, ±100 mv is excessive, but we must always be careful to say what it is excessive for. Gary says "My experience has been that 100 mv won't hurt.", and I suspect that there is a lot of truth in this, and there is a very interesting point here. Gary is talking about using these in a synthesizer system, and the signals we present to a VCA in a synthesizer system are (logically enough) synthesized. They have no prior identity, nothing to be faithful to, making the term "high-fidelity" meaningless here. Such signals are also generally rich in harmonics and highly processed already. While we normally think that a VCA only alters amplitude, it is also permissible to think of the synthesizer as a string of modules, and at the output it doesn't matter who did what. The fact that the VCA altered the harmonics slightly may not be important, or it may even add a useful enrichment to the sound. In short, you may not be breaking the rules because there aren't any rules.

So the 100 mv figure may be realistic for synthesizer VCA's, but it is well to know exactly what is going on if you use it. Don't use much more than that however because the curve saturates at about 400 mv. For a general purpose VCA however, where harmonic distortion may not be so well neglected, you should limit the signal to 10 mv. By the way, we mentioned the "tight corner" that the CA3080 gets us into, and there is a better way. Some newer OTA's and similar chips have a "Gilbert input" which compensates for the inherent non-linearity. Probably the 200 mv figure Gary saw was in reference to one of these newer units, which are far better choices for a general purpose VCA than the CA3080.

Sincerely,
Bernie Hutchins
Electronotes
1 Pheasant Lane
Ithaca, NY 14850

FM Fan

Count my vote with those in favor of a basic programming column. Every issue I see the computer carrying us to new levels of versatility. But just as one must know how basic connections within a synthesizer will resond as opposed to simply following a patch diagram, so it is with computers/synthesizers. I really appreciate the software, but what the program does is not (in the long run) so important as how it does it.

I opted to buy a Pet 2001. Hence I know some BASIC, but nothing about anything concerning machine language, memory locations, etc.. Also, such statements as "...press the second C on the keyboard...", how do I interpret this on my homebrew 5 octave system? You see, with some insights as to (for example) what location $AE actually does, I could modify the listing to run on my custom homebrew system. Such a column would be invaluable to owners of PETs, Apples, Z-80s; in short, any computer based system, and mostly those not using the Paia 8700 or 650X processors.

Right now there is a routine that I would like to write, but don't have the programming chops for. I'm experimenting (as often as finances allow) with the E-Mu music ICs mentioned in a previous issue of Polyphony. Eventually this system will be quite large and make extensive use of FM synthesis techniques. To create a synthesizer voice, the computer would print "Carrier to Modulation Ratio?" whereupon you would input the desired ratio. Then the computer asks "Modulation Index?", and you input some kind of transient generator code. It's set. The computer looks at the C:M ratio and sets the appropriate control voltages at the respective sample and hold outputs. The modulation index controls a VCA which perhaps is an internal part of a dedicated Modulation Oscillator. The output of the Mod. Osc. goes to an AC coupled linear input of the Carrier Oscillator and presto! FM! The advantage is, of course, hundreds of pre-sets without ever pulling a patch cord or twiddling a knob. Each voice also uses a minimum number of modules. FM should be able to run at the same time as Poly 1.0, Seque 1.0, Echo, Shazam, as it could greatly (and easily) expand the sound resources available to use during these routines. It will probably require some very high resolution DACs, though this problem may be circumvented by some use of look-up and transposition tables.

As I said before, I can't write FM. But I believe it would be a tremendously powerful routine in the right circumstances. Programming columns and articles in Polyphony would provide the musician with the skills needed for him to turn such ideas into useful tools for his art. If you know of anyone working in this field, please let me know, or send my name and address along to them, please.

David Mays
212 Elm St.
Hurst, TX 76053

P.S. Oh yes. It seems that EM composition has permeated your typesetting methods in the new style Polyphony. First, pick a word event. This event may sometimes be followed immediately by another event, or by a long space (silence)....

David:

Your ideas for additional coverage of software would undoubtedly be worthwhile for the development of your FM program, and I think we all want more software. But, the results being indicated in Polyphony's current reader survey show that most readers feel that a basic programming or microprocessor fundamentals column would just use up space which needs to be devoted to advanced coverage of existing analog synthesis equipment and techniques. Many readers said they felt programming and processor technique could be found in any number of other magazines and books, and that Polyphony should concentrate on providing the advanced construction, modification, and applications education which has been our major strength in the past. I tend to agree with the survey response. The great flood of computer books and magazines from the past two or three years is just sitting out there waiting to teach us all something. We will continue to run some software and computer interface articles (in fact we already have articles lined up for future issues which present music software for a number of other machines including SWTPC 6800, PET, TRS-80, and others), but Polyphony will probably not become primarilly digitally oriented. Meanwhile, if some of you readers can help David realize his program, contact him directly. AND REMEMBER -- when you guys get the program finished, be sure to send it to Polyphony to be published so others can benefit from it, too!

Marvin

P.S. Concerning our new word processor (our new machine's name is AL) and his printouts-- he is still young and has a lot to learn. But with some upcoming software modifications, ALs printouts should be increasingly better looking and easier to read.

OOPS!

A few bugs jumped into our previous issue at the last moment.

In "The Achilles Heel ..." article, the picture in the left column of page 11 should be exchanged with the one on page 12.

The author of the Hex VCA project on page 30 is Jim Rosen.

Sorry for any inconveniences these mistakes have caused.

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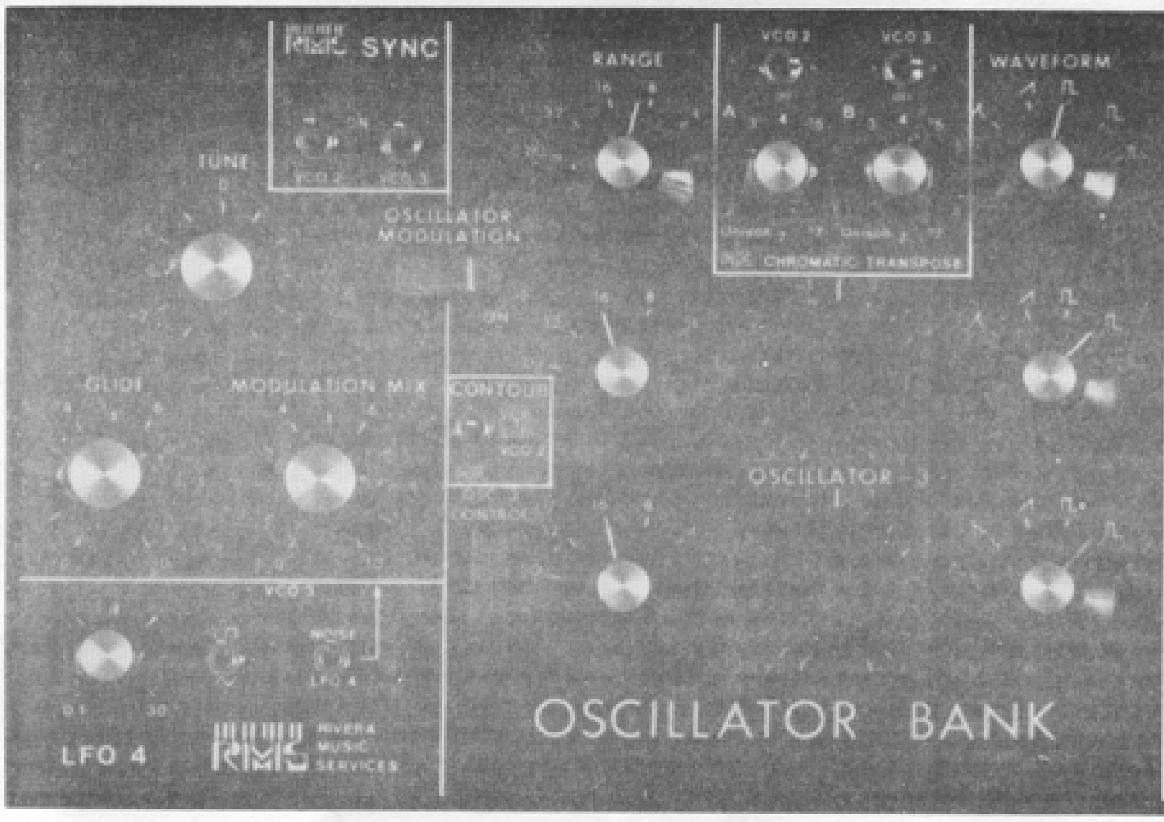


The Synthesizer as a Medium

by Mark Styles

The purpose of this article is to explore the use of analog synthesizers as a compositional tool, and to examine the various relationships that can develop between the performer and the synthesizer. When the concept of the synthesizer as an extension of oneself is adopted, the performer will attain a higher level of awareness in his playing.

I have conducted a series of experiments where individual modules and the patching of the synthesizer itself played a part in the composition of a musical piece. The first half of this article contains some background information and theory while the second part contains specific patches and explanations.

Anyone familiar with synthesizers recognizes the factors which come into play as one goes about operating his or her system. Musical ability, hearing, tactile response, psycho-acoustics, logic, and sonic recall are but a few of the areas that are encompassed. The human ear may be regarded as a microphone or transducer which translates air pressure fluctuations (sound) into nerve impulses. The brain processes this information along with data in its memory, and then sends commands out. Since our brain can only understand electro-chemical impulses, all information recieved and sent must be in this medium. Our senses therefore act as an interface to the outside world.

The synthesizer also acts as an interface; it is the medium which allows us to manipulate and process voltages into meaningful sound. In figure #1 we can get an idea of the various feedback loops that are set up when using a synthesizer. The brain directs the hand to move sliders, pots, etc. The commands sent from the brain are affected by information recieved from the ears, eyes, hands, and information already stored in the brain. Once a person is familiar with the particular equipment involved and the procedures used, his consciousness will transform the synthesizing process into one continuous action.

At this point, the experienced synthesist will realize his knowledge and use of the equipment has transcended to an intuitive level. The achievement of a particular sound is no longer broken down into basic steps; rather it has become a series of programmed muscle movements which are guided by auditory feedback. One's awareness, then, is a whole integrated process; your consciousness is normally unaware of the consecutive steps being carried out by the brain when solving any given problem. Stated more simply, we are normally unaware of our awareness. This is a cybernetic or self-organizing process, and the success in achieving the desired sound depends on the synthesists abilities, habits, experience, mood, and so on.

Another way to view this cybernetic process is that your brain has stored all the commands necessary to achieve the desired results and filed them under one name. It is like a subroutine or a defined function. Your brain goes through all the steps to achieve the end results while consciousness is focused only on the right sound.

When this level of proficiency has been reached the instrument becomes an extension of yourself. While using the instrument, your concentration is focused on the achievement of the end result rather than individual components, patch cords, etc. Your ideas have been translated to electrical information and the synthesizer is being used as an extension of yourself to manipulate and process this information. The resulting sound represents a manifestation of your creative thought processes.

Once you have mastered this creative process, you are no longer dealing with the medium (the synthesizer itself), but rather it's content (your ideas). The synthesizer, in effect, has become a vehicle for the content or message. By forgetting about the mechanics involved, the doors to creativity have been opened and the boundaries expanded simply by the nature of your interaction with the equipment.

I freelance as a composer, recording engineer, and synthesizer programmer and find it humorous when unaquainted musicians first confront and work with synthesizers. They will usually experience what is known as the "Me Against The Machine" syndrome. The novice's conception of the synthesizer usually is a machine which they must battle, coax, intimidate, or whatever to get the desired result. The sound created is a result of the process of a musician interacting with a machine. Until one "gets" this concept he will never realize the full potential of himself and the equipment.

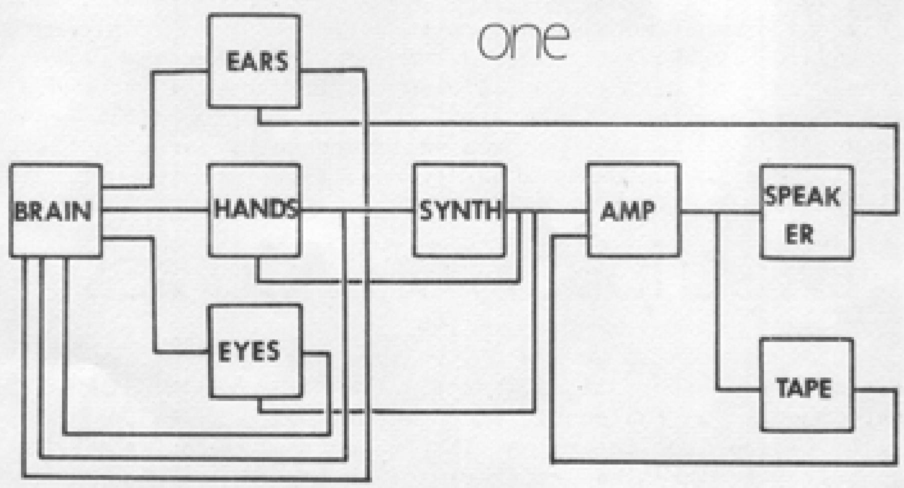
Emotional and Social Consequences

Examining the emotional and social consequences of the creative process can give valuable insight into ourselves. The emotional consequence of playing synthesizers obviously is a sense of satisfaction in the realization of particular sounds. On occasion, I have done pieces which were the result of pure frustration. This 'music as therapy' will assuredly never make it to the top ten, however, it releases the tension for me to move on to new areas of exploration. At other times, my compositions have reflected tranquil and happy states of mind, and on listening to these pieces repeatedly, it becomes an affirmation of myself.

It is interesting to note what different people will find desirable and satisfying in a particular piece of music. Some synthesists will gravitate to specific uses of their equipment. Jan Hammer has an uncanny knack for doing guitar licks. His bending of notes and speed riffs obviously give him a sense of pleasure and release in playing.

The social consequence of a man/machine interaction can reach incredible proportions. While the content or message is spread more or less intact, it is possible for the medium itself to shift radically in the process.

The sounds created by Jimi Hendrix and the Beatles opened new vistas of sonic territory. Their interaction with technology accelerated the pace for exploration in sound producing and modifying equipment. The music of these artists was a reflection of their lifestyle in that it had a certain undefinable freeform style to it. Part of the message was "I'm doing what I want, it doesn't conform to the social norm, and I don't care.".



While a number of musicians followed the footsteps of Hendrix and the Beatles in the quest for sound, many listeners got just the second part of the message. A similar attitude was adopted and applied to other areas of life rather than music (colorful clothes, long hair, a more open viewpoint of life, drug experimentation, etc.). This audience has definitely had some effect upon society although we are still too close to ascertain it's full impact.

I am not implying the next time you adjust your "Initial Decay" it's going to have great social impact upon the world, rather I'm trying to demonstrate the potential for change. If someone can create a sound unique enough, he has the potential to reach and affect literally millions of people.

Intuition Versus Logic

The intuitive process used in tuning a patch is constantly moving in and out of focus as you work. While you are in the intuitive mode, your analytical mind is in the background, functioning to a degree. Each person operates on a unique basis; one individual's intuitive level may be 80% intuitive and 20% analytical, while another person's might be 55% intuitive and 45% analytical.

For example, if you are working on a gong-like chime patch and have not done much exploration in this area, you will probably switch between analytical and intuitive methods. When analytical methods do not achieve results you will turn to intuitive methods or, if desperate enough, even try random ideas. Eventually, when you have reached the desired sound, your brain will have made some record of what you did so the next time the search will be shorter.

If your tape deck suddenly stops working, your consciousness moves from its automatic "get your favorite patch" mode to its analytical, step-by-step, "trace down the problem" mode. Other factors are also present; you might be waiting for the phone to ring (three rooms away with closed doors) or you might be excited or sad. All of your feelings and thoughts will affect your interaction with the synthesizer to a degree. Shifting back and forth between the dozens of appropriate modes results in different levels of consciousness. I have reached an almost blissful state on the fiftieth take of an overdub. After going through boredom, frustration and anger, I would reach an incredible level of sereneness. A totally detached feeling would envelope me as my hands mechanically moved about the keyboard and my mind thought of nothing.

Composing with Machines

What I propose to examine are a few methods of using sequential devices as compositional tools. Naturally the reader is encouraged to use these ideas to the extent he or she sees fit. By understanding these principles and making them part of his own personal knowledge, the composer will be free to draw upon this information and use it appropriately. I am not suggesting that we give up writing music and relegate the task to sequencers, switches, etc., but that we use these devices to implement our ideas and to make them an extension of ourselves.

The equipment I use throughout these experiments consists of an Aries AR-334 analog sequencer module, an AR-335 switches module, an AR-318 sample/hold clock and noise module, an AR-323 dual mixer, an AR-331 preamp and envelope follower, and a digital divider module. Also used were the usual complement of standard synthesizer modules found in a medium sized system. The procedure I followed was to record a sub-audio sawtooth on tape and use it as my click track to clock the sequencer, divider, and switches on subsequent passes. In this way, I could lay up several passes onto tape to check out the various results.

Description of Modules

The sequencer is an 8 step, 2 layer analog sequencer with position gate outputs at each step. Its features include a reset input, a run/enable/step input, and a three position toggle switch to detirmine the mode of operation. The sequencer requires an external clock to operate. Sequences shorter than eight notes can be obtained by patching one of the sequencers gates to the reset input. The run enable and step input permits voltage control of the sequencers start and stop. Normalled to this input is +10 volts. When the sequencer switch is in the RUN position and greater than +3 volts is present at the run enable and step input, the sequencer will step once with each rising edge presented to the R/E/S input. The step position is a spring loaded position which permits the user to manually step the sequencer through its positions. The eight dual pots provide +10 volts out for maximum versatility. To facilitate tuning of the VCOs, the +10 volts should be attenuated.

The switches module consists of two 2 position switches and two 4 position switches. All four switches are bidirectional. The clock of the second 2 position switch is normalled through a divide-by-two to the clock input of the first 2 position switch. The 4 position switch is identical in operation to the 2 position switch except it has four stages. These switches each have their own clock input and their own manual toggle. In addition there is ten volts normalled to the single input of each switch.

The second 4 position switch is a threshold sensitive switch (consecutive window comparators). This switch does not step on the rising edge of a waveform as do the other three, but rather on the threshold levels that are set for each position of the switch. There are four threshold pots which will turn on their respective positions depending on the voltage present at the clock input. By sampling noise and patching the sampled output to the switch clock input, the switch will randomly step among the four stages. By setting each succesive threshold higher, a sawtooth will turn on each position sequentially - A, B, C, D, A, B, C, D, etc. A sine wave, after it has been biased up above 0 volts, will fire C, D, C, B, A, B, C, D, C, B, A, B, etc. The switch will not respond to negative voltages, although they do no harm.

There is also a three position toggle for the threshold switch which allows only one stage on at a time, or each stage to stay on as it's threshold is exceeded, or each stage to turn off as it's threshold is exceeded. This module also contains ten LEDs which display the status of each switch at all times. Both the sequencer and the switches module can be driven by TTL logic without an interface.

The sample/hold module also contains a voltage controlled, syncable low frequency oscillator and white, pink and random noise. By adjusting the output attenuator, the sample/hold can be scaled to unity gain. Thus sequencer permutations can be derived by using two different clocks (one for the sequencer, one for the S/H). This is illustrated in diagram #7 and covered later.

The preamp and envelope follower is a four function module containing a preamp, envelope follower, comparator, and inverter. The comparator provides a gate and trigger output, which makes it very useful for generating timing signals from an external instrument.

The digital divider is a home made module. It was designed by John Ball, a radio astronomer at Harvard, who experiments with synthesizers in his spare time. This project was originally described in Electronotes (EN #34, 8- 1974). The divider utilizes five TTL counter chips and has a number of uses. Dividing an audio frequency produces octaves and other justly intoned intervals which allow one to build a really big sound from just one oscillator. Dividing sub-audio signals can provide various rhythmic divisions.

The divider basically consists of three sections, the first of which is the comparator. The comparator's function is to derive a clean usable pulse from any waveform that is presented to it. The pulse is then fed to the two other sections, both of which are divider circuits. The second section actually consists of nine counters, each with its own output. So by feeding a waveform in, one can obtain simultaneous divisions of 2, 3, 4, 5, 6, 8, 10, 12, and 16. The third section is a counter which will divide by any number from 2 to 256. This is accomplished by eight binary coded toggle switches (weighted 1, 2, 4, 8, 16, 32, 64, and 128). You merely add 1 to the sum of the toggles thrown and this output is available at a separate jack.

Three LEDs have been installed to show the status of the input, monitor, and output sections. A suitable pulse is derived by adjusting the comparator so that the monitor LED matches the input LED. A visual check on the second divider section is provided by the output LED. It's rate will correspond to the toggles switched into the circuit. I have also added a pushbutton which resets all the counters to zero. This will allow me to sync 'divide by' pulses to a tape. As a convenience to use with the rest of my system, I have added an attenuator with a 'multipled' outlet. As mentioned earlier, the divider's TTL logic will drive the Aries switches and sequencer directly. To fire an ADSR, an additional circuit has been added to bring up the voltage level to the ADSRs threshold.

Using The Sequencer and Switches Modules

The consensus seems to be that analog sequencers are obsolete now that digital units are available. Analog equipment still has some merits to it, and when used in conjunction with the switches module, you can obtain some quite interesting results. See figure #2.

This patch takes two eight note sequences and generates a 32 note sequence before repeating. I have depicted the results in both musical notation and in a step representation. Note that the second 2 position switch is normalled to fire at one half the rate of the first 2 position switch. The numbers stand for sequencer step positions, and the letters stand for pitch output (sequencer layer) A or B.

STEP AND PITCH BREAKDOWN

1A, 2B, 3A, 3B, 4A, 5B, 6A, 6B, 7A, 8B,
1A, 1B, 2A, 3B, 4A, 4B, 5A, 6B, 7A, 7B, 8A,
1B, 2A, 2B, 3A, 4B, 5A, 5B, 6A, 7B, 8A, 8B

The first pulse from the LFO steps the sequencer and toggles both switches to position 1. Switch 1 sends the voltage from 1A to the VCO. The next clock pulse steps the sequencer to the second position and toggles switch 1. The voltage at sequencer position 2B is sent to the VCO. On the third clock pulse switch 1 toggles to pass the voltage at 3A, switch 2 is toggled and the +10 volts is disconnected from the R/E/S input. The fourth clock pulse toggles switch 1 and it routes the voltage at 3B to the VCO. Note that the sequencer did not step to a new position because the +10 volts was not present at the R/E/S input. The fifth pulse toggles switches 1 and 2. The +10 volts from switch 2 advances the sequencer and pitch 4A is sent out. The result of this patch is a 32 step repeating sequence.

Bear in mind that the switches will process any signal put through them. For instance, they could be used to select between two different envelopes, audio waveforms, or similar signals. With some audio signals, undesirable transients could be encountered when the switch changes state. These are most apparent when switching between mellow, continuous sounds; an altering of the patch could achieve similar results however.

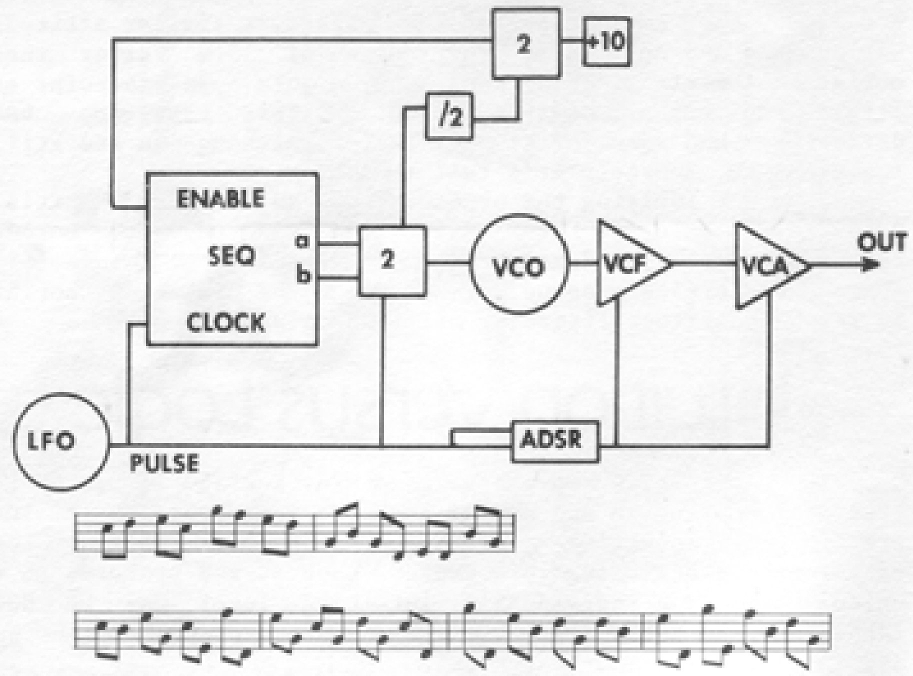
The patch in figure 3 utilizes one 2 way and one 4 way switch. The 4 way switch is used to reset the sequencer position gates, and the 2 way switch is fired once every 22 notes. This gives you a 44 note sequence.

It is possible to program rests into your sequences. There are a number of methods to accomplish this. The patch shown in figure 4 will cycle through every 14 steps with 6 rests occuring. It is permissable to mix position gates by simply 'multing' them together. Two mixes of gates are sent to a switch and then inverted. These inverted gates will then close down the VCA which in turn will stop the audio output.

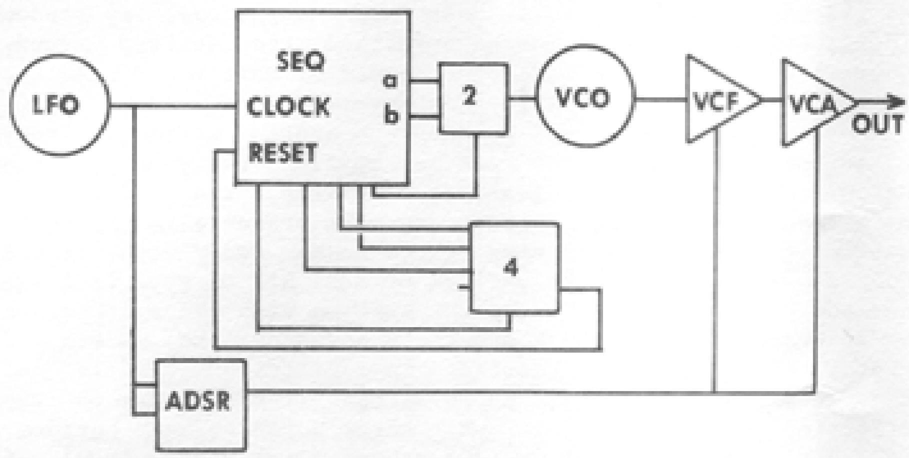
Using The Divider in Conjunction With The Sequencer

By utilizing the patch of figure 5 we are able to come up with an almost limitless variety of sequences. The procedure was to lay down one track of sequenced material, rewind the tape, and then lay down a second pass of the sequencer. The gain, envelope follower, and threshold levels must be properly set so as to derive sharply rising gates as timing signals. The sequencer must be reset to its starting position if you want the two tracks to be in sync. On successive passes you can try different "divide by's" for different effects.

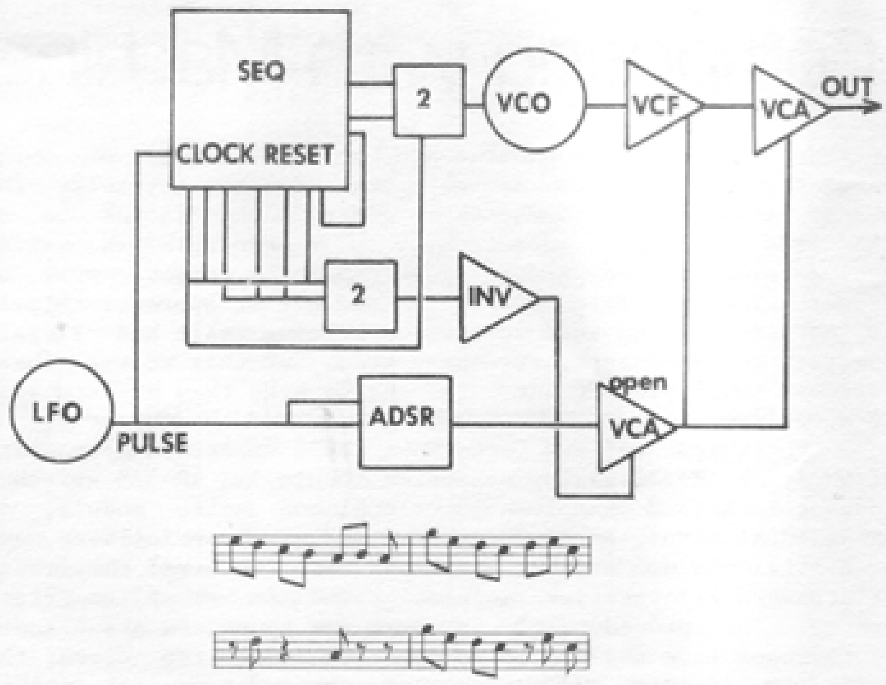
two



three

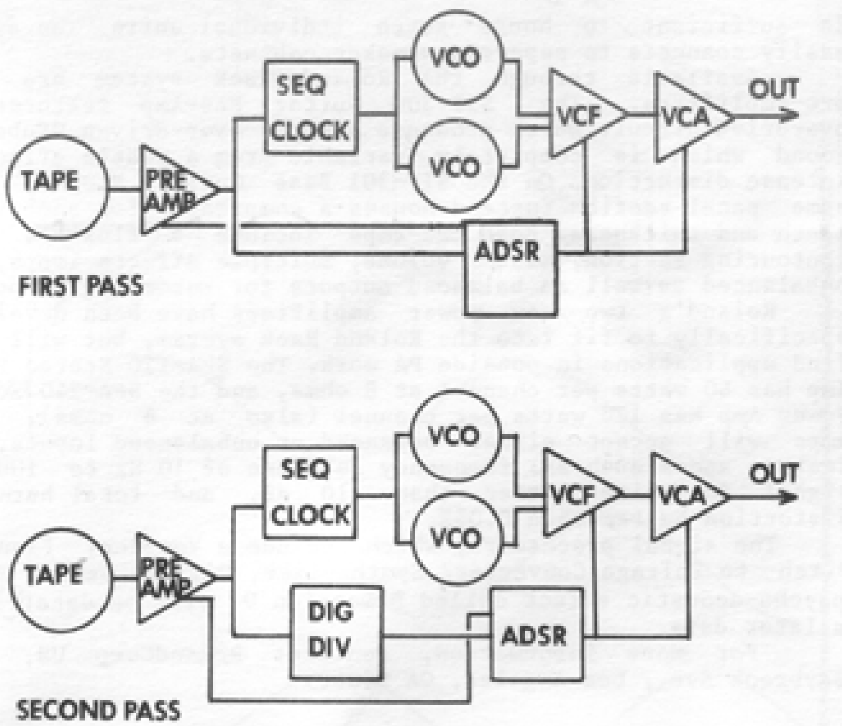


four

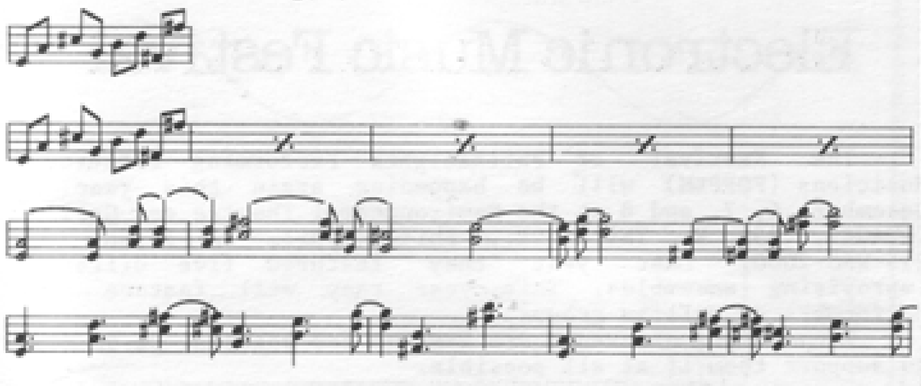


In figure 6 I have notated some of the different permutations. 6A is the original eight note sequence. On pass one, line 6B was recorded. The digital divider was then added to the patch and VCO 2 was tuned a fourth above VCO 1. Divider output 5 was patched to the sequencer clock and preamp. This resulted in the divider stepping the sequencer at one-fifth the speed of the previous track. The resulting line is 6C. The two tracks, represented as lines 6B and 6C will coincide on every 40th pulse recieved from the tape.

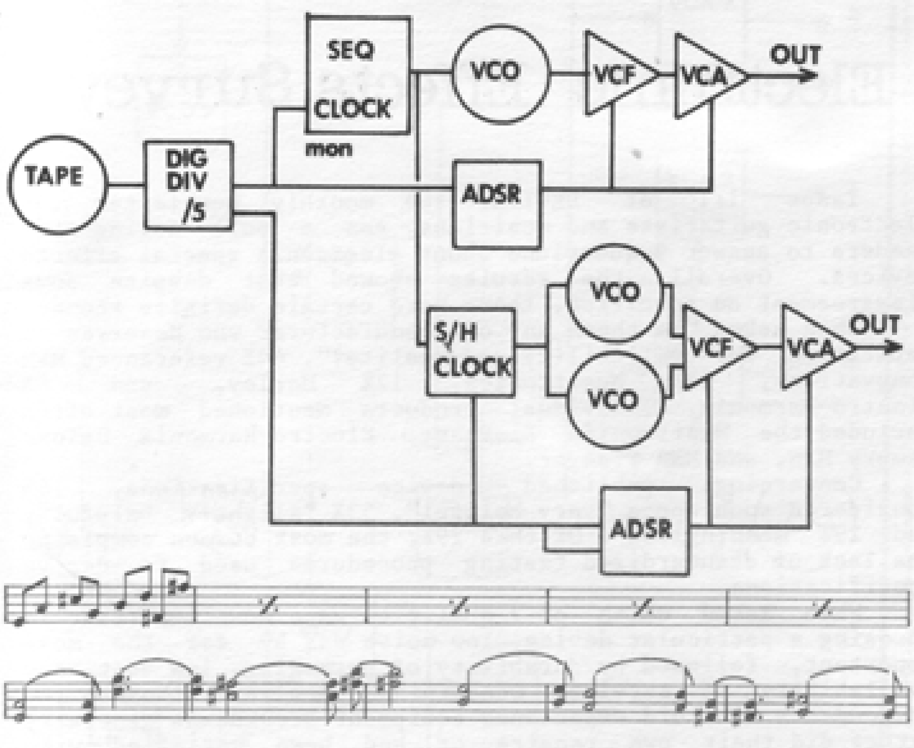
five



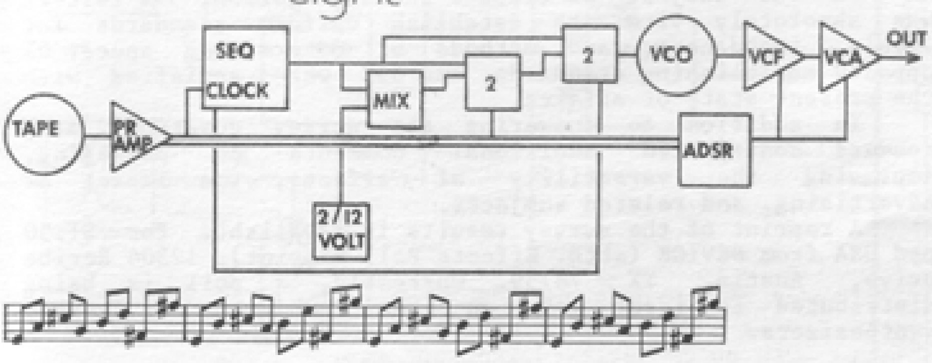
six



seven



eight



On the third pass I changed the divider output to 3. This resulted in line 6D. The sound is now beginning to get thicker; switching envelope and filter settings, plus changing octaves can relieve some of the tension. Now the coinciding point for all three voices will be 120 clock counts (3 X 5 X 8).

Introducing The Sample/Hold

The patch in figure 7 has had a sample/hold added. Note that the digital divide monitor output is the same as the input. The divider's output is 5 and the second VCO is still tuned to a 4th. This results in the musical passage illustrated. Here again the sequencer will count 40 tape counts before resetting. By using different filter and envelope parameters these two lines can set up a lot of activity. The aural effect does not strike one as being the same notes because of the voice difference, the fourths, and the different attack rate.

The Mixer as a Modifier

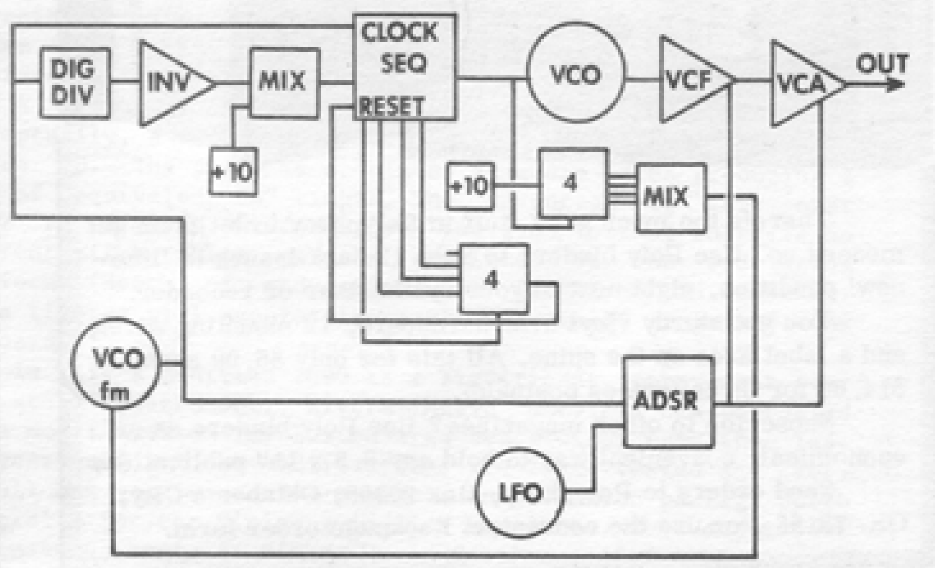
It is possible to use a mixer for automatic transposing as shown in figure 8. First the mixer must be inserted between the sequencer and VCO at unity gain. Next, patch +10 volts to mixer input #2 and adjust the pot to transpose by the desired interval. In the example cited, I added 2/12 of a volt which raises the sequence by a whole note. I find it easier to disconnect the sequencer and substitute a sequencer for this tuning procedure. If both switches are set to the first position the resulting notes will be as illustrated. Every second note of every even measure will be raised by one whole step. By using more mixers, both layers of the sequencer, and the 4 way switch, the permutations become staggering.

Another use of the mixer is to patch the four outputs of the threshold switch into a mixer and then use the mixer output to frequency modulate the VCO clock as shown in figure 9. The threshold switch is illustrated as being clocked by the sequencer pitch output. An interesting alternative would be to trigger the threshold switch with a sample/hold output. I have also inverted and biased the divider output so every 48th note holds the sequencer for one count before moving to the next step.

The mixer may be used with the four way switch as a simple sequencer by patching directly to a VCO. This could be used in conjunction with the sequencer. Another possibility would be to patch the mixer output to the PWM and FM inputs of a voltage controlled clock. This would not only control the speed, but the duration of the sequencer position gates (note duration).

A number of previous examples are called into play in the patch of figure 10. It will shift keys twice, a whole step and an octave. The length is indeterminable because of the sample/hold's function in the circuit. The multimode filter is being opened and closed by three ADSRs. Care must be taken so as not to overload the filter when both ADSRs are present. Certain notes will be accented with an amplitude shift. Gates could be attenuated and used for PWM, VCF or VCA modulation to add more variation.

nine



Conclusion

While this information does present some new avenues for exploration, it is painfully obvious how much knowledge lies out there undiscovered. One of the advantages of modular synthesizers is that it does help in the conceptualization of a patch. By becoming intimately familiar with some of the prnciples here, one will discover that he will begin to use them at an intuitive level.

references

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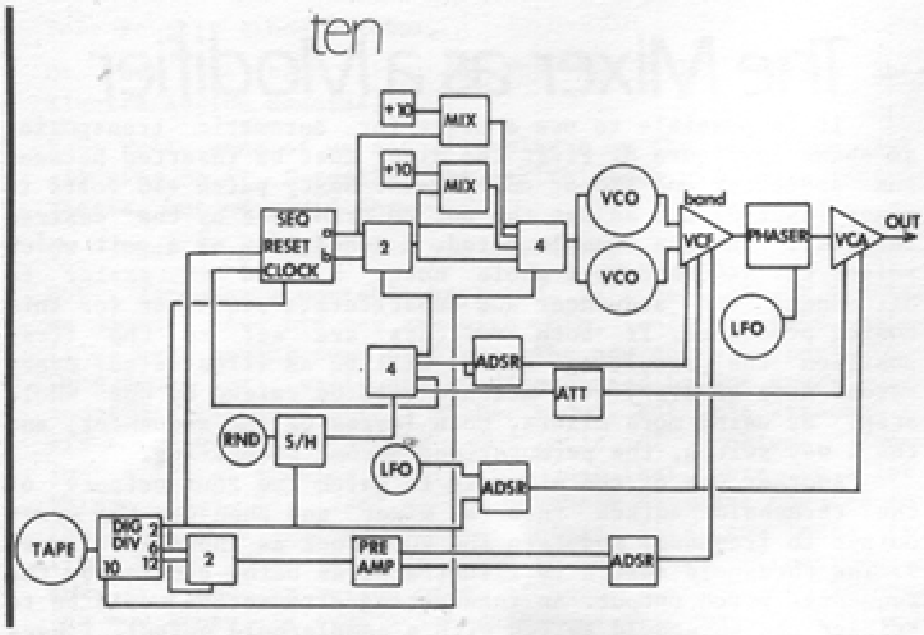
Understanding Media, Marshal McLuhan, McGraw-Hill, NY 1964

Waves And The Ear, by Van Bergeijk, Pierce and David, Double Day Anchor, NY 1960

"The Digital Divider in Electronic Music Synthesis", John Ball, Electronotes #34, Ithaca, NY



INDUSTRY REPORT

....continued from page 7

is sufficient to house seven individual units. The system easily connects to seperate speaker cabinets.

Available through the Roland Rack system are two pre-amplifiers. The SIP-300 Guitar Pre-Amp features an overdrive circuit which produces a warm over-driven tube-amp sound which is completely variable from a subtle effect to intense distortion. On the SIP-301 Bass Guitar Pre-Amp this same panel section instead houses a compressor for additional punch and thickness. Both pre-amps include a flexible tone contouring section, master volume, multiple effects loops, and unbalanced as well as balanced outputs for recording purposes.

Roland's two new power amplifiers have been developed specifically to fit into the Roland Rack system, but will also find applications in outside PA work. The SPA-120 Stereo Power Amp has 60 watts per channel at 8 ohms, and the SPA-240 Stereo Power Amp has 120 watts per channel (also at 8 ohms). Both amps will accept either balanced or unbalanced inputs, and feature specs such as: frequency response of 10 Hz to 100KHz, signal to noise better than 110 dB, and total harmonic distortion better than 0.05%.

The signal processors, which include a Vocoder, Flanger, Pitch to Voltage Converter/ Synthesizer, Digital Delay, and a psycho-acoustic effect called Dimension D, will be detailed at a later date.

For more information, contact: RolandCorp US, 2401 Saybrook Ave., Los Angeles, CA 90040.

Electronic Music Festival

The Festival of Philadelphia Performing Electronic Musicians (FOPPEM) will be happening again this year on December 6, 7, and 8 at the Environmental Theatre and Gallery (ETAGE), 253 N. Third St., Philadelphia, PA 19106, ph. 215-WA3-2080. Last year they featured five different improvising ensembles. This year they will feature solo performers as well as groups.

These types of events are few and far between, so be sure to support them if at all possible.

Electronic Effects Survey

Issue 1:1 of DEVICE, the monthly newsletter ror electronic guitarists and musicians, ran a poll asking its readers to answer 9 questions about electronic special effects devices. Overall, the results showed that despite some disagreement on specifics, there were certain definite trends.

When asked "Is there any one manufacturer who deserves a reputation for reliability and quality?", 44% referenced MXR Innovations, 18% Musitronics, 12% Morley, and 6% Electro-Harmonix. Individual products mentioned most often included the Musitronics Bi-Phase, Electro-Harmonix Deluxe Memory Man, and MXR Flanger.

Concerning published device specifications, 16% considered such specs "very helpful", 53% "slightly helpful", and 29% "meaningless". Of this 29%, the most common complaint was lack of standardized testing procedures used to derive specifications.

When asked which of 7 qualities were most important in choosing a particular device, low noise was by far the most important, followed by durability of packaging, low cost, and availability of service. Concerning service, those few respondents who did experience equipment breakdowns generally either did their own repairs or had been satisfied with manufacturer warranties and servicing.

On the subject of effects standardization, 59% felt it was absolutely vital to establish uniform standards for levels, impedances, and methods of determining specs; 6% opposed establishing standards, and 35% were satisfied with the present state of affairs.

In addition to answering the survey questions, many readers contributed additional comments on packaging, improving the versatility of effects, the impact of advertising, and related subjects.

A reprint of the survey results is available for $1.50 ppd USA from DEVICE (attn. Effects Poll Reprint), 12304 Scribe Drive, Austin, TX 78759. Currently, a poll is being distributed in issue 1:5 on the subject of guitar synthesizers.

Updated Tape

The 'Robert Greenberg' tape reviewed in the May/June 79 issue of Polyphony has been re-released under the name "Music By Robert Banks". The music contained on the tape is mostly as listed in the review, however there are a couple newer cuts used to supplement or replace the original material. A program notes sheet is now available, listing equipment used and personal credits.

The Robert Banks tape is still available from: Future Now, 600 S. Muirfield Rd., Los Angeles, CA 90005, Ph. 213-273-4020.

Synthesizer School

ACCESS - The Synthesizer School is offering in-depth courses in the use of the synthesizer and home recording techniques. The school is designed to offer maximum hands-on experience with instructor guidance and back-up paced at the student's own speed. The instruction equipment ranges from simple monophonic to large polyphonic systems, and includes the use of sequencers and vocoders. ACCESS aims to bring the exciting experience of electronic music to a larger group of users.

For more information, contact: Roger Clay, ACCESS- The Synthesizer School, 12304 Scribe Dr., Austin, TX 78759, Ph. 512-836-3069.

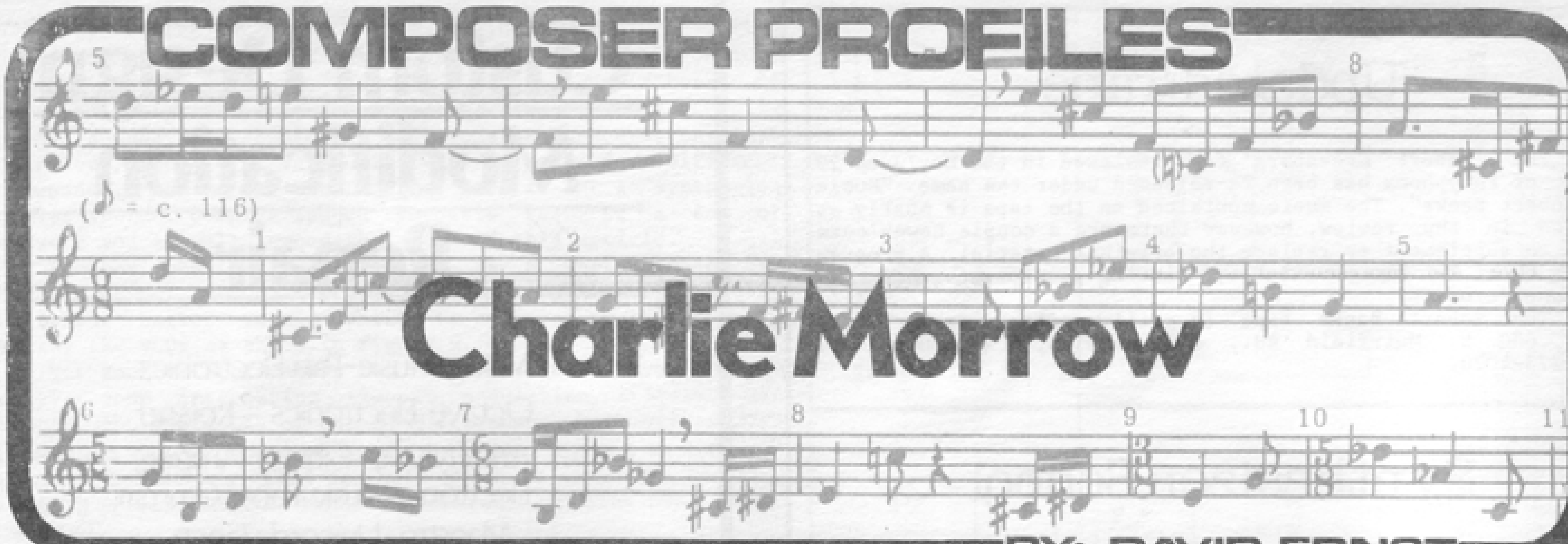
Mono/Stereo Console

The Model 2000 Mixing Console, producing simultaneous mono and stereo outputs that allow mono and stereo formats to operate independently of each other, is being introduced by Audy Instruments of Salem, MA. The unit also provides separate monitor and effects sends. Available in 12 or 16 channel versions (stackable up to 32 channels), the console utilizes high speed, low noise IC technology to reduce TIM distortion to .03% and enhance sound quality.

Providing input preamps with a dual LED system, the Audy Series 2000 maintains headroom of 24 dB throughout. Other standard features include: individual channel and output patch points, transformerless balanced inputs and outputs, 3 band EQ with switchable midrange center frequency, switchable pre and post monitor and effects sends, soloing for any input or output, phantom power for mics, and a work lamp socket.

The Audy Series 2000 Mixing console is priced at approximately $2450 for 12 channels, and $2995 for 16 channels, including an Anvil flight case for either model. Literature is available on request from: Audy Instruments, David Tkachuk, Shetland Industrial Park, PO Box 2054, Salem, MA 01970, Ph. 617-744-5320.

COMPOSER PROFILES



Charlie Morrow

BY: DAVID ERNST

Charlie Morrow (b. 1942) is a composer, performer, and founder of the New Wilderness Foundation-- an organization established in 1978 that includes an Ocarina Orchestra and Audio Graphics. The ocarina is a clay whistle or flute formed in the shape of an animal, originally used in religious and ritual ceremonies of civilizations that date from the 13th century BC, Shang Dynasty, China. Charlie describes the Ocarina Orchestra as an ensemble that "produces a music built around breathing patterns, vocal ranges, natural tuning, and resonance which come out of instruments which are close but not <u>exact</u> copies of each other... it is a sound pool and meeting ground for noise and music exchange...". Audio Graphics is an affiliation of the New Wilderness Foundation to produce performance oriented and cross cultural events and publications, e.g. sound environments, music, storytelling, poetry, and drama.

Although Charlie Morrow sometimes employs electronic devices in his music we will concern ourselves with his compositional ideas and vocal treatment, enabling us to formulate electronic applications derived from Charlie's masterful handling of natural sounds. His involvement with the psychic and physical meanings of sound has prompted Charlie to work with breathing, counting, the linguistics of non-human species, and the oral poetry and music of tribal people. The essence of Charlie's compositional aesthetic is revealed in the following statement. "Fortunately the future is over and less of us are junkies to the 'new'... Recent years have also seen growth of a world art community and blurring of the lines between high, folk, ethnic, commercial and idiosyncratic art."

Upon realizing that nothing is 'new', contemporary artists frequently pattern their work on preexisting forms. Some are able to use this information as a point of departure and subsequently develop a personal mode of expression, whereas others choose simply to imitate the work of their predecessors. No value judgement is intended here for both methods have a place in our culture, but two noteworthy features of Charlie's aesthetic and music indicate the manner by which he transcends this apparent conflict. His awareness of a universal music, reflected by the previous comment on a world art community and blurred stylistic lines, sets the stage for a completely uninhibited sound environment-- where not only all sounds may exist, but also any elements of all musical styles are perceived as complementary subsets of a homogeneous texture. Secondly, rather than baseing his music on preexisting musical forms Charlie uses nature as a model; more will be said about this shortly.

Now that we have a general idea of the nature of Charlie's music we will trace the development of his compositional process. Charlie's prime interest is chanting music, and he explains: "In my search for a rhetoric of chanting music, I was reliving the ancient process of anthropomorphism, looking for the animal world in me and for myself in the natural universe." This suggests an underlying principle of universality throughout the natural world, and it has led Charlie to investigate and to employ sound processes that exist in nature (bottom of the waters, outer reaches of space, etc.) as models for composing and performing. Notice that Charlie's method is more concerned with the internal rhythm or flow of a river than with the actual sound of water; he is searching for a universal structural principle in nature.

The next step for Charlie was to delve further into the sonorous laws of nature, and he discovered that "Much of the sound world is concerned with signalling. Much of the soundmaking is repetitive biological process that is audible: simple sounds in simple relationships, often pulse series." This is reminiscent of Alvin Lucier's "Vespers", a live performance piece involving sondols to generate electronic clicks or pulses to stimulate the process of echolocation, i.e. the sensory perception of dolphins and bats.

Although pulse series account for many animal sounds (crickets, etc.), human soundmaking is closely associated with breathing which, in turn, is related to body functions. In fact, human sounds may be classified as discontinuous because of man's dependence on breathing, whereas non-human sounds may also be continuous (wind, oscillators, etc.). Comparison and analysis of continuous and discontinuous sound led Charlie to number counting and numerically organized pieces.

These compositions are performed by counting number patterns while inhaling and exhaling. Charlie's preliminary work often involves studying his breathing on tape to find a symmetry along his breathing and other bodily functions. This provides insight into the basis of diverse number systems, e.g. binary, decimal, etc. Before we consider musical examples and attempt to integrate Charlie's number system with synthesizers and computers, it is appropriate to be aware of his description of his pieces. "...the counting pieces that I have been composing over the years are ways of thinking about time on the pulse level and on the various levels of structure that characterize musical architecture."

The counting pieces of Charlie Morrow are notated as numeric patterns, often in a physical arrangement of a triangle. The numbers may be read on one tone or associated with pitches, or they can be played on instruments. In addition, the numbers may be replaced with silence or with any other sound to expand the modes of organizing time that each counting piece describes. Figure 1 contains three possible versions of "COUNTING", a simple pattern that displays the organizational power of

MARY ELLEN MORROW

numeric sequences as a structural principle for music.

The numbers are read (or played) horizontally, and each space is a beat. Realization of such patterns rests with the performer, but the sonorous capacity increases with the use of synthesizer and tape recorders. Addition of tape echo, delay, flanging, etc. will enhance both live and prerecorded versions of counting pieces, especially if each modification is designated a discrete loudspeaker or channel. Furthermore, each number may be assigned to a specific pitch, timbre, amplitude, etc. Since these pieces are organized rhythmically as a series of equal pulses, superposed versions may be delineated further by ascribing different tempi to each loudspeaker or channel; and employment of sequences and automated percussion is ideal for this purpose. Since any number patterns are admissible, I have written a few number generating programs in BASIC (see figure 2) that may be adapted to other counting pieces.

```
1
1 12
1 12 123
1 12 123 1234 etc.

1
121
12321
1234321 etc.

12345678
1234567
123456 8
12345 78
1234 678
123 5678
12 45678
1 345678
 2345678 etc.
```

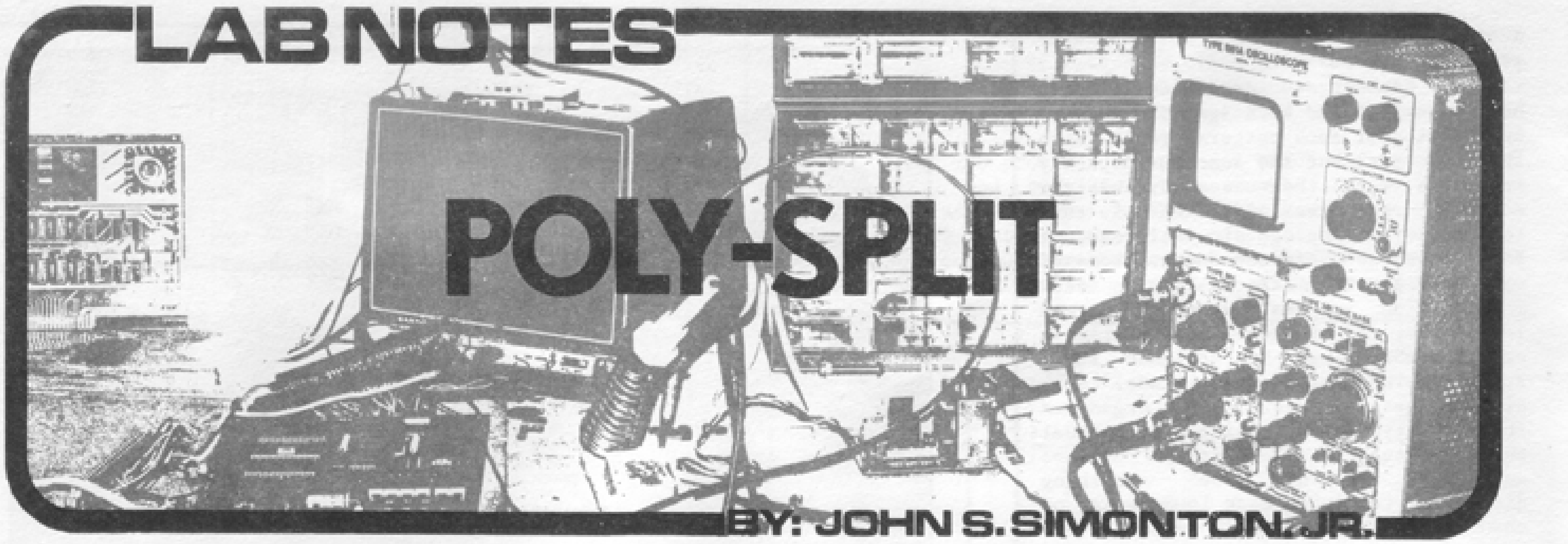
Figure 1. "COUNTING"
(from The Book Of Numbers)

```
 5  REM FIBONACCI SERIES
10  A = 1
20  B = 1
30  PRINT A,B
40  A = A + B
50  B = A + B
60  GO TO 25
70  END

 5  REM 8 AND 2 COUNT
10  N = 0
20  P = N + 8
30  PRINT N,P
40  N = P + 2
50  GO TO 20
60  END

 5  REM FACTORIALS
10  PRINT "N", "N!"
20  N = 1
30  F = 1
40  PRINT N
50  FOR N = 2 TO 7
    ("7" may be any integer)
60  F = F * N
70  PRINT N,F
80  NEXT N
90  END

5  REM PYTHAGOREAN TRIPLES
6  REM (X^2 + Y^2 = Z^2)
7  REM X<Y
8  REM X,Y,Z ARE VARIABLES
   WHERE Y=Z-1 AND X=Z-2
10  PRINT "XX..+", "YY..=", "ZZ"
20  FOR Z = 5 TO 16
30  FOR Y = 2 TO 15
40  FOR X = 1 TO 14
50  IF X > Y THEN 80
60  IF (X * X) + (Y * Y) = (Z * Z)
    THEN PRINT (X*X), (Y*Y), (Z*Z)
70  NEXT X
80  NEXT Y
90  NEXT Z
99  END
```

Figure 2
Number Generating Programs

As mentioned earlier, Charlie's interest in anthropomorphism has led him to investigate the rhetoric of non-human communication. In "TOADFISH LANGUAGE" he constructs a number language based upon recordings and research of Dr. Howard Winn and colleagues at the Graduate School of Oceanography in Kingston, Rhode Island. Each number represents different fish who call and answer by playing a boatwhistle. In this instance, the numbers are neither spoken nor sung, but are replaced by the sound of boatwhistles. Whereas "COUNTING" demonstrated how abstract numeric patterns could be used to structure a piece, "TOADFISH LANGUAGE" treats numeric repetitions as a number language analog of fish language. A sample calling pattern for "TOADFISH LANGUAGE" is given

continued on page 40....



LAB NOTES

POLY-SPLIT

BY: JOHN S. SIMONTON, JR.

Many times we've talked about how the personality of our computer based equipment is a function of the operating system software that we happen to be running at the moment. Let's play some head games with the gear and feed it some code that will give it a split personality.

POLY-SPLIT does just that; it gives us two complete polyphonic synthesizer systems under the control of one keyboard. Play a chord or note on the lower keys and they are always assigned to a lower group of outputs. Play on higher keys and the result is assigned to another group.

Before we get into the listing of this program and its operation and use, we need to keep one fact clearly in mind; POLY-SPLIT is simply an extension of the polyphonic personality offered in MUS 1.0. All of the options offered by that code (STGs, dynamic output refresh, etc.) are provided by this one also. Since many of MUS 1.0's subroutines are used by POLY-SPLIT you must have this PROM or its equivalent available, and the variables that you manually initialize for MUS 1.0 (see LAB NOTES: MUS 1.0, April/ May 1978 Polyphony) must be set for POLY-SPLIT also.

In addition to OUTS, CTRL, etc. which MUS 1.0 used there is a new variable which is unique to POLY-SPLIT; OUT2 ($BF). This is the variable that tells the program how many channels are to be set aside for use exclusively by AGO keys below the split point. Notice specifically that if MUS 1.0's STG option is selected, the number entered into this variable must include those channels which will be producing envelope transients. (i.e. The number entered for OUT2 will always be an even number when STGs are being used.)

For example, if you have hardware (QuASH, etc.) for eight channels, this number is entered into the normal MUS 1.0 location for it; OUTS ($EA). If you want to split these into three channels for low keys and five for high keys, you would set OUT2 ($BF) to contain 03.

The program appears at the end of this column and is loaded starting at location $000 in the same way that we've loaded programs in the past. If you're the careful sort, you will also save the program on tape as soon as it's loaded so that if there's a problem it won't wipe out all of your work.

When the program has been loaded, preset the MUS 1.0 variables according to your preferences and application, and set the low channels variable (OUT2) as discussed above.

Run the program from location $000. With POLY-SPLIT running, keys 0 and 1 on the command keyboard retain the functions that they had under MUS 1.0. Key 0 clears and mutes the system; key 1 causes all of the channels to produce a note corresponding to middle C on the AGO keyboard.

A use for command key 2 has now been added; it provides a means of changing the split point while you're playing. Touch this pad and, as long as it's held down, any key on the AGO keyboard that you press will become the new split point. Now while playing, any key below the split point will be assigned to the channels that you've set aside for them, while keys greater or equal to the split point will be assigned to the remaining channels.

```
                  :*********************************
                  :*                               *
                  :*           POLY-SPLIT          *
                  :*                               *
                  :*   A PROGRAM FOR POLYPHONIC    *
                  :*        SPLIT KEYBOARD         *
                  :*                               *
                  :*              BY               *
                  :*         JOHN SIMONTON         *
                  :*                               *
                  :* (C) 1979 - PAIA ELECTRONICS   *
                  :*                               *
                  :*********************************
                  KTBL .DL 00E0
                  NTBL .DL 00D8
                  HKEY .DL 00A2
                  SPLT .DL 00A1
                  OUT2 .DL 00EC
                  OUTT .DL 00EB
                  OUTS .DL 00EA
                  TRGN .DL 0DC3
                  INIT .DL 0D21
                  NOTE .DL 0D2B
                  POLY .DL 0D71
                  DECD .DL 0F00
                  :
                  :FIRST, SYSTEM THINGS ARE DISPOSED OF.  THE SYSTEM IS
                  :INITIALIZED USING MUS 1.0'S "INIT" ROUTINE, THEN THE
                  :QUASH CHANNELS ARE REFRESHED AND THE AGO KEYBOARD
                  :SCANNED ALSO USING ROUTINES FROM MUS 1.0
                  :FINALLY, THE PIEBUG ROUTINE "DECODE" IS USED TO READ THE
                  :COMMAND KEYBOARD AND ANY COMMANDS ARE EXECUTED.
                  :0-SYSTEM CLEAR AND RE-INIT; 1-TUNE ALL CHANNELS;
                  :2-SET SPLIT POINT, ANY AGO KEY PRESSED BECOMES SPLIT
                  :
                       .OR 1000
                  :
1000-  A5 EB      STAR LDA *OUTT     :GET THE # OF RESERVED LOW CHANS
1002-  85 EC           STA *OUT2     :SAVE PERMANENTLY
1004-  A2 07      POSP LDX 07        :SET UP A POINTER/COUNTER
1006-  A9 00      SLP9 LDA 00        :AND GET READY TO ZERO STUFF
1008-  95 A2           STA *HKEY,X   :ZERO THE TEMPORARY BUFFER
100A-  CA              DEX           :AND POINT TO THE NEXT
100B-  10 F9           BPL SLP9      :IF SOME ARE LEFT, LOOP
100D-  20 21 0D        JSR INIT      :MUS 1.0 - INTIALIZE SYSTEM
1010-  20 2B 0D   SLP6 JSR NOTE      :MUS 1.0 - REFRESH AND READ AGO KBD
1013-  20 00 0F        JSR DECD      :PIEBUG - READ COMMAND KEYBOARD
1016-  F0 EC           BEQ POSP      :IF COMMAND = 0, BRANCH TO RE-INIT
1018-  C9 01           CMP 01        :IS COMMAND = 1?
101A-  D0 07           BNE NTST      :NO, BRANCH TO NEXT TEST
101C-  A9 2E           LDA 2E        :WILL BECOME MIDDLE C
101E-  20 23 0D        JSR INIT+02   :USE PART OF MUS 1.0 INITIALIZE
1021-  F0 ED           BEQ SLP6      :BRANCH ALWAYS
1023-  C9 02      NTST CMP 02        :IS COMMAND = 2?
1025-  D0 08           BNE SPLI      :NO, BRANCH TO POLY-SPLIT PROGRAM
1027-  A5 E7           LDA *KTBL+07  :GET THE LOWEST KEY DOWN
1029-  F0 E5           BEQ SLP6      :IF NONE ARE DOWN, LOOP
102B-  85 A1           STA *SPLT     :SAVE THE KEY AS THE SPLIT POINT
102D-  D0 E1           BNE SLP6      :BRANCH ALWAYS
```

```
                :
                :NOW THE SPLIT PROGRAM. AT THIS POINT A LIST OF THE
                :AGO KEYS WHICH THE MUS 1.0 SUBROUTINE "LOOK" FOUND TO
                :BE PRESSED HAS BEEN COMPILED AND SAVED IN THE INPUT BUFFER
                :AREA "KTBL".  WE BEGIN BY REMOVING FROM THE INPUT BUFFER
                :ALL THOSE KEYS WHICH ARE ABOVE THE SPLIT POINT AND
                :TRANSFERING THEM TO THE TEMPORARY BUFFER AREA "HKEY".
                :
102F-  A0 07    SPLI LDY 07       :SET UP POINTER TO HIGH BUFFER
1031-  A2 07         LDX 07       :AND ONE TO INPUT BUFFER
1033-  B5 E0    SLP0 LDA *KTBL,X  :GET THE KEY
1035-  F0 0F         BEQ SNX1     :IF ZERO, GO TO NEXT
1037-  C5 A1         CMP *SPLT    :GREATER THAN SPLIT POINT?
1039-  90 08         BCC SNX0     :IF NOT GREATER, BRANCH
103B-  99 A2 00      STA HKEY,Y   :GREATER, SAVE IN HIGH BUFFER
103E-  88            DEY          :POINT TO NEXT HIGH KEY BUFFER
103F-  A9 00         LDA 00       :PREPARE AND
1041-  95 E0         STA *KTBL,X  :ZERO THIS KEY
1043-  CA       SNX0 DEX          :POINT TO NEXT KEY
1044-  10 ED         BPL SLP0     :IF SOME LEFT, LOOP
                :
                :NEXT THE NUMBER OF CHANNELS AVAILABLE FOR LOW KEY USE
                :IS TRANSFERRED TO THE TEMPORARY COUNTER "OUTT" AND THE
                :MUS 1.0 ALLOCATION PROGRAM POLY IS CALLED TO ASSIGN LOW
                :KEYS TO LOW CHANNELS.
                :
1046-  A5 EC    SNX1 LDA *OUT2    :GET THE NUMBER OF LOW CHANS AVAILABLE
1048-  85 EB         STA *OUTT    :AND PUT IT IN THE TEMPORARY COUNTER
104A-  20 75 0D      JSR POLY+04  :AND CALL THE MAIN PORTION OF POLY
                :
                :NOW THAT THE LOW KEYS HAVE BEEN ALLOCATED TO LOW CHANNELS,
                :THE HIGH KEYS ARE TAKEN FROM "HKEY" AND PLACED BACK IN THE
                :INPUT BUFFER (KEYS ALREADY ALLOCATED ARE REMOVED FROM THE
                :INPUT BUFFER). SIMULTANEOUSLY THE LOW CHANNELS ARE MOVED
                :TO HKEY AND ALL LOW CHANNELS IN THE OUTPUT BUFFER
                :ARE MARKED AS "IN USE" SO THAT THEY WILL BE IGNORED
                :WHEN HIGH KEYS ARE ALLOCATED.
                :
104D-  A4 EC         LDY *OUT2    :A COUNTER TO MOVE ONLY THE LOW CHANNELS
104F-  A2 07         LDX 07       :AND A POINTER/COUNTER
1051-  B5 A2    SLP1 LDA *HKEY,X  :GET THE HIGH KEY FROM TEMP BUFFER
1053-  95 E0         STA *KTBL,X  :PUT IT IN THE INPUT BUFFER
1055-  88            DEY          :ONE LESS LOW CHANNEL  TO DO
1056-  30 08         BMI SNX2     :ALL LOW CHANNELS DONE, BRANCH
1058-  B5 D8         LDA *NTBL,X  :GET THE LOW NOTE
105A-  95 A2         STA *HKEY,X  :PUT IT IN TEMPORARY BUFFER
105C-  09 40         ORA 40       :THEN SET THE TRIGGER TO MARK NOTE
105E-  95 D8         STA *NTBL,X  :AND REPLACE THE NOTE
1060-  CA       SNX2 DEX          :ONE LESS CHANNEL, POINT TO NEXT
1061-  10 EE         BPL SLP1     :IF SOME LEFT, LOOP
                :
                :NOW POLY IS CALLED AGAIN, THIS TIME TO ALLOCATE HIGH CHANNELS
                :
1063-  38            SEC          :PREPARE FOR SUBTRACTION
1064-  A9 10         LDA 10       :16 CHANNELS SUPPORTED BY MUS1
1066-  E5 EC         SBC *OUT2    :LESS THE LOW RESERVED CHANNELS
1068-  AA            TAX          :RESULT IS POINTER
1069-  38            SEC          :ANOTHER SUBTRACTION - PREPARE
106A-  A5 EA         LDA *OUTS    :TOTAL HARDWARE CHANNELS
106C-  E5 EC         SBC *OUT2    :LESS LOW RESERVED CHANNELS
106E-  85 EB         STA *OUTT    :BECOMES CHANNELS LEFT TO ALLOCATE
1070-  20 77 0D      JSR POLY+06  :CALL MAJOR PORTION OF POLY
                :
                :FINALLY, THE REAL STATE OF THE LOW CHANNELS IS RESTORED
                :TO THE OUTPUT BUFFER. SIMULTANEOUSLY THE TEMPORARY BUFFER
                :IS ZERO'D FOR THE NEXT PASS.
                :
1073-  A4 EC         LDY *OUT2    :NUMBER OF LOW CHANNELS FOR COUNTER
1075-  A2 07         LDX 07       :POINTER/COUNTER
1077-  88       SLP2 DEY          :ONE LESS LOW CHANNEL
1078-  30 04         BMI SNX3     :AND IF ALL DONE, SKIP NEXT TRANSFER
107A-  B5 A2         LDA *HKEY,X  :GET THE REAL CHANNEL STATE
107C-  95 D8         STA *NTBL,X  :PLACE IN OUTPUT BUFFER
107E-  A9 00    SNX3 LDA 00       :NOW GET READY AND
1080-  95 A2         STA *HKEY,X  :ZERO THIS TEMPORARY BUFFER LOCATION
1082-  CA            DEX          :ONE LESS TEMP BUFFER LOCATION
1083-  10 F2         BPL SLP2     :IF SOME REMAIN, LOOP
1085-  30 89         BMI SLP6     :BRANCH ALWAYS TO CONTINUE
                :
                END  .EN
```

```
***********************************
*             NOTES:              *
*                                 *
*  DUMP PROGRAM FROM 0000-0090    *
*                                 *
*  SET THESE LOCATIONS:           *
*                                 *
*     $0E8  CTRL $40  DYNAMIC     *
*     $0E9  ODLY $20  DELAY       *
*     $0EA  OUTS $XX  TOT CHANS   *
*     $0EB  OUTT $XX  LOW CHANS   *
*                                 *
*  COLD START - $0000             *
*  WARM START - $0004             *
*                                 *
*                                 *
*  NOTE THE FOLLOWING THINGS:     *
*                                 *
* 1) THE PROGRAM IS RELOCATABLE;  *
*    IT MAY BE LOADED AND RUN IN  *
*    ANY NON-CONFLICTING MEMRORY  *
*    SPACE                        *
*                                 *
* 2) CALLING POLY TWICE IS NOT    *
*    EXTRA EFFICIENT.  TIME RE-   *
*    QUIREMENTS DICTATE MEDIUM    *
*    TEMPO KNOB SETTING - ABOUT   *
*    10 MS/SCAN                   *
*                                 *
* 3) AS SOON AS THE PROGRAM IS    *
*    RUNNING, TOUCH COMMAND PAD   *
*    2 AND THE KEY WHICH IS TO    *
*    BE THE SPLIT POINT.  THEN 1  *
*    TO TUNE AND FINALLY 0        *
*    BEFORE PLAYING               *
*                                 *
***********************************
POLY-SPLIT 8.8
```

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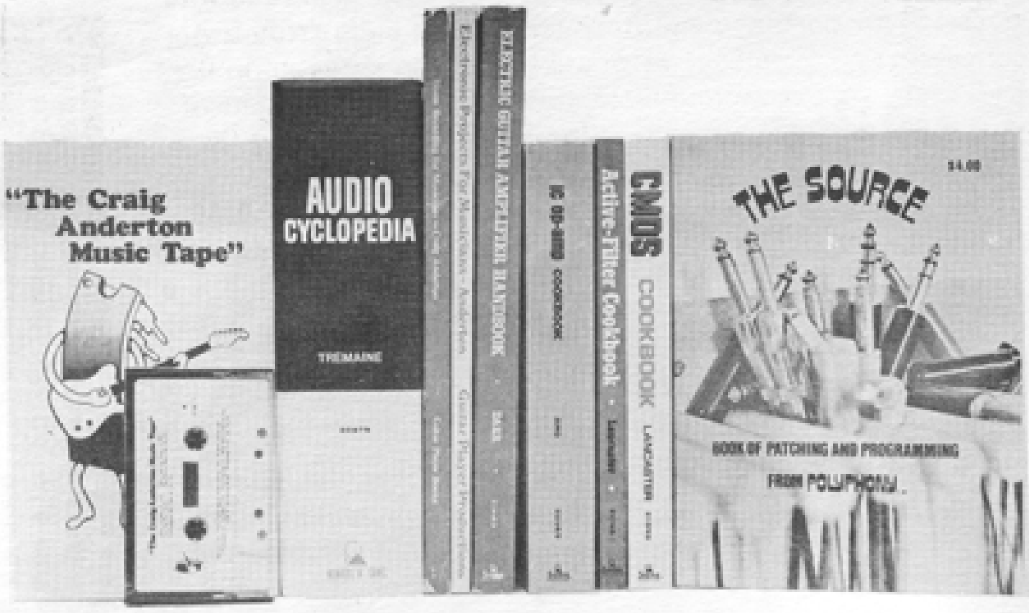

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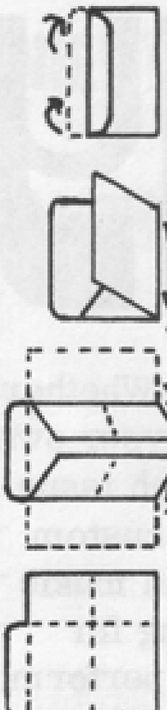

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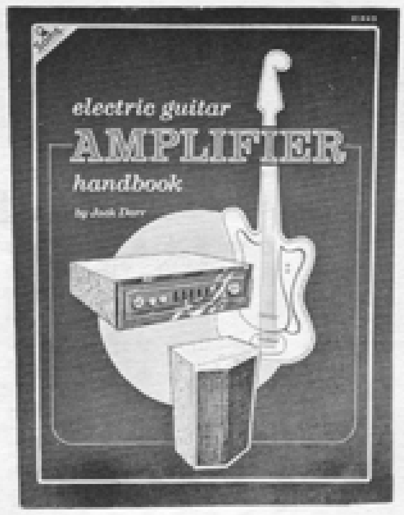

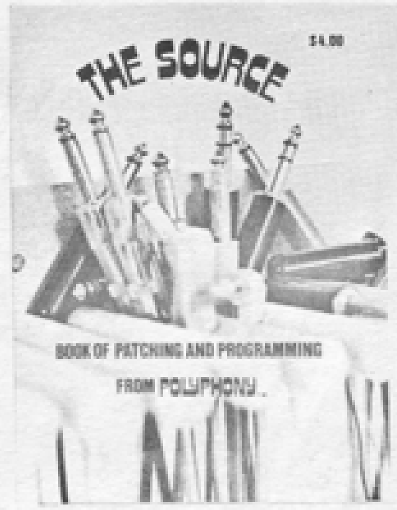

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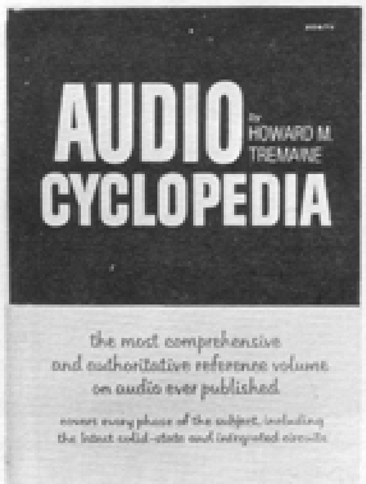

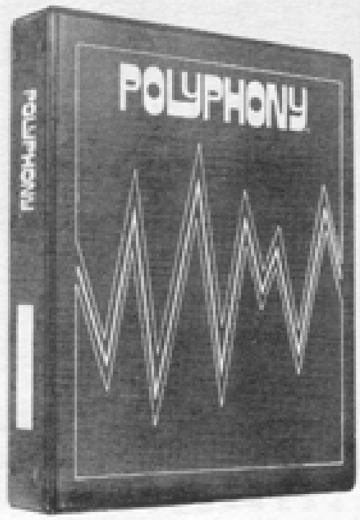

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☐ #0101: 1975: patch cord racks, time constants in envelope generators and glide circuits, patches.

☐ #0201: 1/76: glide footswitch, low-cost preamp, sequencer game, LFO trigger generator project, patches.

☐ #0202: 2/76: foot pedal projects and patches, thunder and explosion patches, "Wa/Anti-Wa" pedal project, sequencer game, top octave generator projects, patches.

☐ #0203: 3/76: "A Time Trip - Ready:" (a future fantasy), envelope generator modifications, VC clock project, music notation- pitch, "fine-tune" controls, sound systems, computer drum projects, patches.

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

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

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

☐ #0303: February 78: computer controlled Gnome, using joysticks, build a bionic trumpet, octave controllers for bionic sax and trumpet, ultra-VCO modifications, voltage control the Mu-Tron Bi-Phase, oral joystick, patches.

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

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

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

☐ #0403: November/December 78: multi-purpose keyboard software, Sohler keyboard and notation system, voice frequency to voltage converter project, proposals for tape exchange, VCA project, sequencer software- part two, frequency balancing in recording, Barton and Priscilla McLean.

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

☐ #0501 May/June 79: using click tracks, PET music software, clockable sample/hold and noise source project, voice processing patches, VCF circuits, profile of John Cage, linear DAC.

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

The Use of Sophisticated Modifications in Creative Synthesis

by Ken Perrin

All musical instruments evolve and electronic music instruments evolve very quickly. Only 15 or so years ago, Moog and Buchla were developing the first voltage control instruments and since then we've seen not only refinements to the basic set of modules but also conceptual changes in the original instruments resulting in portable synthesizers, polyphonic synthesizers, digitally controlled synthesizers, and now digital synthesizers. Unfortunately, due to the high initial costs of these instruments, many musicians cannot afford to take advantage of these technological advances.

With this in mind, Rivera Music Services of Boston has developed modification packages for existing instruments. The modifications add features and functions to standard instruments to increase their potential and facilitate their use. One modification package was developed especially for the Arp 2600 and a modified 2600 is the instrument we describe here. At the outset, here is a list of the complete modification package for the 2600.

1. Independent gate inputs to both envelope generators
2. Voltage control of internal clock, with input attenuator
3. Voltage control switching of electronic switch
4. Slow random voltage with variable bandwidth
5. Three - position switches for normalled connections on selected inputs
6. Monitor speaker disconnect function for "headphone only" monitoring
7. Final output disconnect switch
8. Voltage controlled resonance on VCF with modulation input attenuator
9. Manual and voltage control of pulse width on VCO 1 with associated input attenuator
10. Reset sync on VCOs
11. Foot pedal signal attenuator with +10 volts normalled to input
12. Twelve-position chromatic transpose switches for VCO 2 and VCO 3 with assignment switches
13. Comparator with voltage controlled threshold
14. RMS Phase Modulated Sync for VCOs

Looking at the list of modifications, the first ten features offered are features which Arp could have built into the instrument in the beginning but did not, due either to increased manufacturing costs or lack of foresight. Whatever the reason, these shortcomings have been frustrating 2600 owners for a good many years. For the most part, these modifications and their uses are obvious. Likewise, the addition of a voltage pedal to a synthesizer is not revolutionary, it's just an extremely convenient and useful controller. Two of the more sophisticated and far-reaching modifications are the Phase Modulated Sync and the Comparator. Let's take a look at the basic operation and application of these new features.

Comparators

Although all analog synthesizers have comparators in them, only a few manufacturers offer a non-dedicated, user accessable comparator module. The RMS Comparator is a handy module and can be used in a variety of ways. All comparators have two inputs, a signal input and a threshold input. When the instantaneous amplitude at the signal input exceeds the threshold level, a gate goes high at the output and remains high until the signal amplitude falls below the threshold level. This particular comparator has a voltage controllable threshold. Figure 1 should make its operation clear.

One way the RMS Comparator can be used is to generate pulse waves. A triangle wave patched into the comparator produces a pulse output. Changing the threshold of the comparator produces a change in the pulse width. Some interesting timbral effects can be realized by mixing two pulses at exactly the same frequency and voltage controlling the width of each pulse with a different control voltage. The patch in figure 2 can accomplish this, and will produce the wave forms shown. (For the sake of clarity we chose not to show the modulated pulse from the VCO in the wave form diagram. In the patch, the width of pulse D would also be modulated causing composite pulse E to be even more complex.)

The comparator is often used to generate timing signals. In processing acoustic or electric instruments, the comparator will derive a gate from the envelope follower's output. This gate could initiate an envelope generator which could, in turn, alter the timbral or amplitude envelope of the original instrument. Another possibility is to use the module to derive gates from recorded clock tracks. Tone bursts recorded on tape and patched into the envelope follower and then into the comparator, would produce gates at the comparator's output. These gates could be used for stepping sequencers, firing envelope generators, triggering sample and holds and the like. This method insures synchronized events throughout each track of a multi-track recording.

Still another possibility for the comparator is to create a gate delay. Lagging a gate "rounds off" the rising and falling edges. Patching the lagged gate into the comparator and setting the threshold fairly high results in a gate output from the comparator whose onset is some time later than the onset of the original gate. Varying the comparator's threshold varies the length of the delay.

The comparator could also be used to create a gate for the second voice of the keyboard. To accomplish this, patch the 2nd voice control voltage output not only to a 2nd VCO but also to the comparator's signal input. Patch the keyboard control voltage into the 1st VCO and also to the comparator's modulation input. Set the threshold manually to less than 1/12 volt. To accomplish this, depress any keyboard key and turn the comparator's threshold down just past the point where the comparator's output gate goes low. This sets the threshold to less than 1/12 volt. When a single key is depressed the threshold will be greater than the input signal (by less than 1/12 volts) and there will be no gate at the output. When two keys are simultaneously depressed the upper voice voltage at the signal input will be greater than the lower voice voltage at the threshold modulation input and a gate will go high at the output. This gate could be patched into a second envelope generator (assuming the envelope generators have been suitably modified) and, when two notes are played, each could have its own envelope.

One final suggested application is to use the comparator to split the keyboard. We assume here that the electronic switch has been modified to accept an external trigger. The patch is shown in figure 3. Set the comparator's threshold to just less than 2 volts. Any keyboard note lower than the middle C (the key equivalent to 2 volts on an ARP 2600) produces no gate at the comparator's output. The voltage from middle C or above is greater than the threshold voltage so a gate is present at the comparator's output. This gate is patched into both inputs of the ring modulator which is capacitively coupled. The coupling differentiates the gate's rising and falling edges into positive and negative voltage spikes respectively. The modulator multiplies the positive spikes together resulting in a positive spike at the rising edge, and also multiplies the negative spikes together resulting in a positive spike at the falling edge. These spikes act as triggers which toggle the electronic switch on both the rising and falling edges of the comparator's gate.

PhaseModulatedSync

The modification which proved extremely useful in creating many and varied effects is the Phase Modulated Sync. Briefly, sync is a connection between two VCOs in which one VCO, the master, forces the second VCO, the slave, to have the same fundamental frequency as the master, or an integral multiple thereof. Typical reset sync achieves this by interrupting the oscillation of the slave VCO and re-setting it to the beginning of a cycle each time the master begins a new cycle. Visually, the effect of reset sync is plotted in figure 4.

The RMS Phase Modulated Sync is similar but it has some additional features. First, a toggle switch selects whether the slave VCO resets to the rising or falling slope of the master VCO. Second, an "Initial Phase" control sets a threshold which detirmines the voltage level on the master waveform at which the slaved VCO will reset. The two VCOs need not begin their cycles simultaneously and this feature allows the slave VCO to reset when the master is, for example, one third of the way through one cycle. Third, there is an input to allow voltage control of the threshold and consequently of the phase relationship between the two VCOs. This input has an associated input attenuator and when syncing, for example, from the rising slope of a sawtooth, the voltage at the phase modulation input and the voltage set on the initial phase control are summed together allowing voltage control of the phase of the master and slave over a full 360 degrees. This feature can provide some rich and interesting timbres which could otherwise be difficult to synthesize.

Figure 5 is a diagram of Phase Modulated Sync. The sawtooth is the master waveform, the dotted line through the master is the modulated threshold, the vertical dotted lines are the reset sync points, and the higher frequency sawtooth-looking waveform is the synced sawtooth from the slave. Note that during the third cycle of the master, we flipped the toggle switch and reset the slave to the master's falling edge.

Comparator

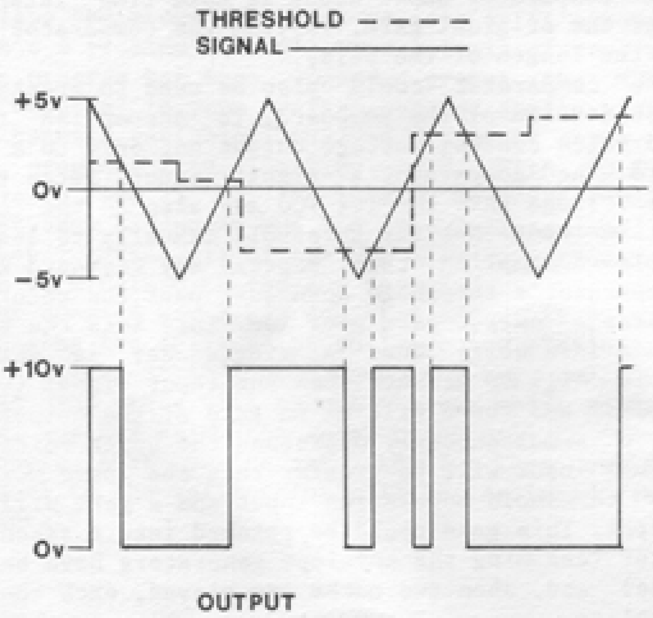

FIGURE 1

The use of two slaved VCOs and a resonant low pass filter with a fixed cut-off frequency often provides all the tools necessary to synthesize the complex formants used to produce vowel sounds. This can be accomplished with the patch in figure 6.

The first VCO provides the basic pitch of each vowel sound. VCO 2 and VCO 3 and the resonant filter provide the three significant formants of each of the vowel sounds. Tuning VCO 2 and VCO 3 to the appropriate formant frequencies and syncing them to VCO 1 creates waveshapes with strong harmonic emphasis at each of the two main vocal formants. Tuning the

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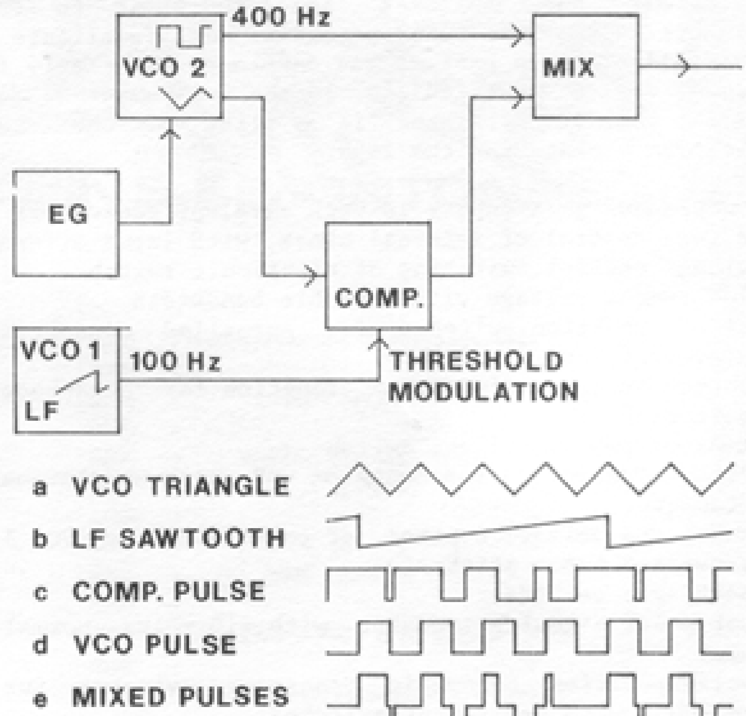

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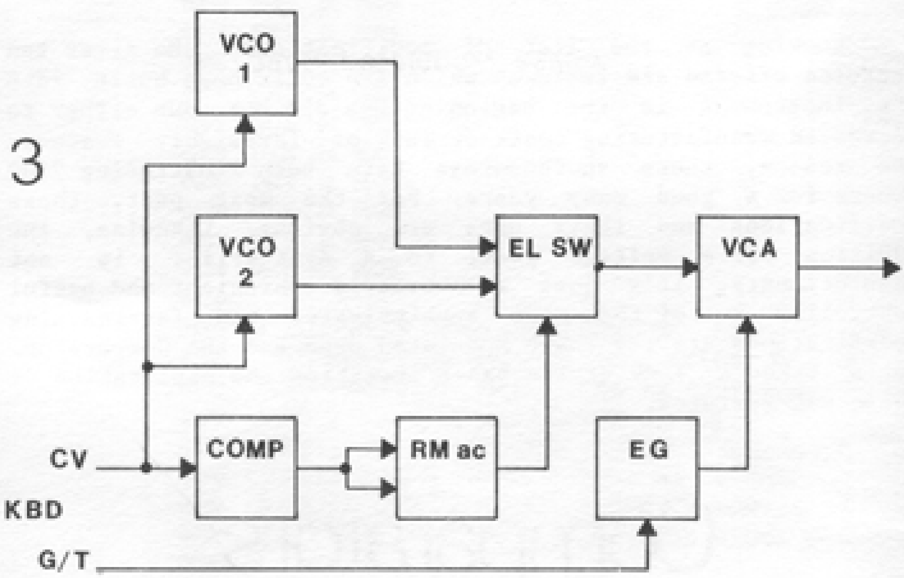

RESET SYNC

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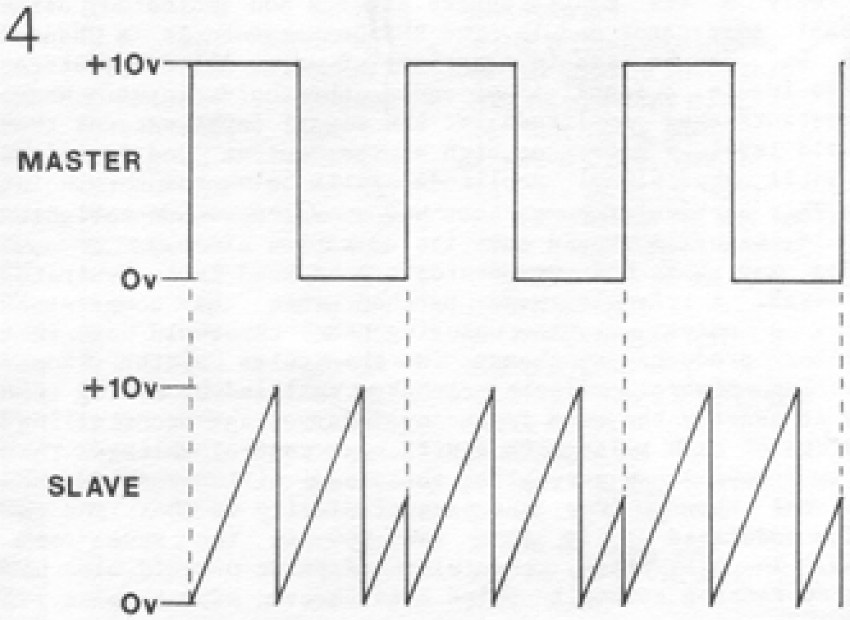

resonant filter to the appropriate frequency creates the third formant. The frequencies to which the VCOs and VCF are tuned are selected from the following chart showing the formant structure of English vowel sounds. The chart, and additional information on this topic is taken from Music, Sound, and Sensation, an excellent text by Fritz Winckel, published by Dover Publications, Inc., New York, 1967.

FORMANT STRUCTURE OF ENGLISH VOWEL SOUNDS

U(who'd)	340 Hz	1 KHz	2.25 KHz
O(obey)	400 Hz	700 Hz	2.4 KHz
ɔ(hawed)	600 Hz	900 Hz	2.5 KHz
ə(hod)	750 Hz	1.2 KHz	2.5 KHz
æ(had)	690 Hz	1.7 KHz	2.5 KHz
I(hid)	410 Hz	2.1 KHz	2.6 KHz
i(heed)	290 Hz	2.3 KHz	3 KHz
ʊ(hood)	450 Hz	1.1 KHz	2.3 KHz
ɜ(heard)	470 Hz	1.4 KHz	2.5 KHz
ʌ(hudson)	640 Hz	1.25 KHz	2.2 KHz

But many of the sounds commonly called vowel sounds are actually sequences of vowel sounds called dipthongs. Pronounce the English pronoun "I" very slowly and you will hear and feel in your mouth that it is in fact two sounds, ɔ (as in hawed) followed by i (as in heed). In going from one sound to the next, you can feel your throat close and the back of your tongue move to the roof of your mouth. There are a couple of different ways to synthesize a dipthong. The simplest, though not the most elegant, is to add an attenuated offset to our previous patch. As figure 7 shows, we've also added a VCA and an envelope generator for articulation. Also note that we've simplified the patch by choosing not to change the filter frequency. The exact frequency of the third formant is not crucial to the intelligibility of the vowel sound.

Tune the VCOs and VCF to the frequencies necessary to produce the formants of the first vowel sound. To achieve the shift to the second set of formants, fully open attenuator #1 so it passes the full +10 volts. Then tune attenuators #2 and #3 so that the appropriate pitch change will occur at each VCO. So, with the first attenuator closed we have the first vowel sound and opening this attenuator fully slides the VCOs upwards stopping at the frequencies which will produce the second vowel sound. This timbral change should occur within the temporal space of one amplitude envelope. Using an envelope with a moderately long actack and release generally yields the best results.

A more elegant and versatile patch to synthesize dipthongs uses an analog sequencer. The principle is the same as in the previous patch but the manner of execution is easier. Assuming we want to synthesize the dipthong "I", the patch is displayed in figure 8.

Consulting the chart, we find that the significant formants of the first sound occur at 600 Hz, 900 Hz, and 2.5 KHz. The significant formants of the second sound occur at 290 Hz, 2.3 KHz, and 3 KHz. We are again going to leave the highest formant unchanged and at approximately 2.75 KHz. The frequencies written above the VCOs indicate the initial frequency setting. In order to tune the patch, step the sequencer to the first position. Tune VCO 1 to approximately middle C. Increase the voltage at sequencer pot A until VCO 2 produces a frequency of 600 Hz (between D and D#-- a 9th above middle C). Turn the layer B tuning pot for the first stage to 0 volts. Next step the sequencer to the second position. Tune pot A to 0 volts and increase the voltage on pot B until VCO 3 produces a frequency of 2.3 KHz (approximately 3 octaves above VCO 2). Sync both VCOs to VCO 1 The keyboard control voltage controls VCO 1 detirmining the pitch of the voice. The pitch range of the voice is best when it's at least an octave lower than the lowest formant. The keyboard gate initiates the

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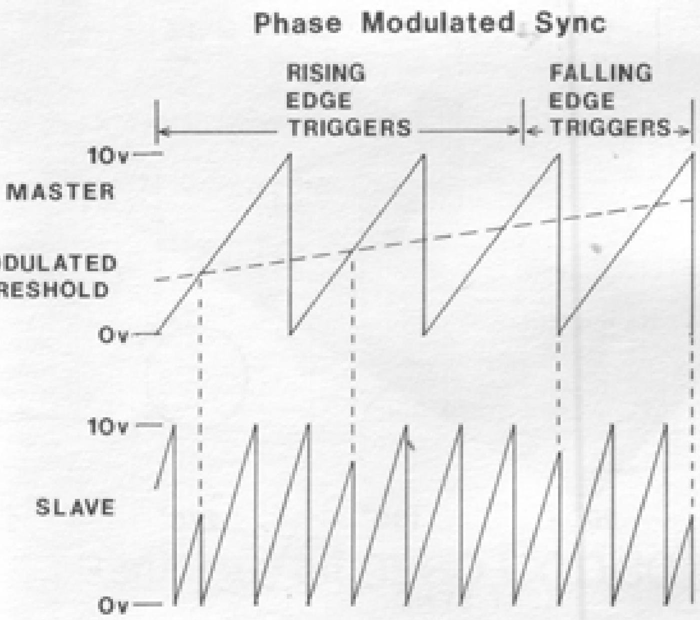

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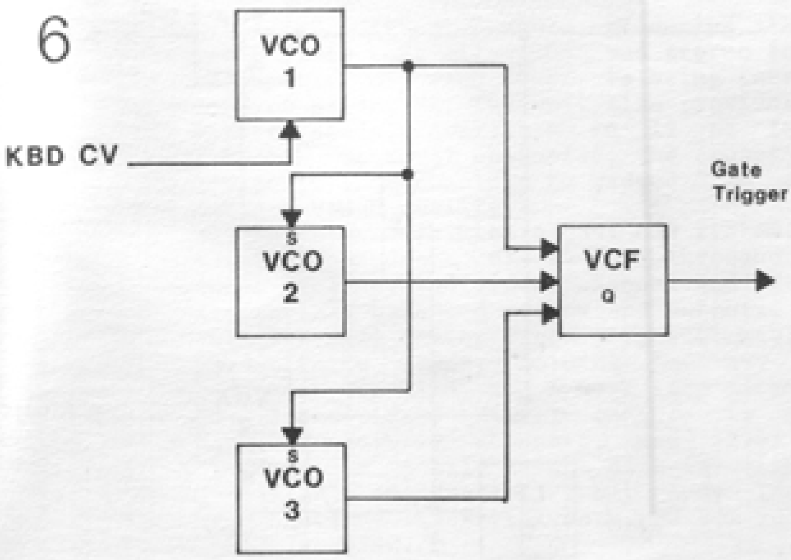

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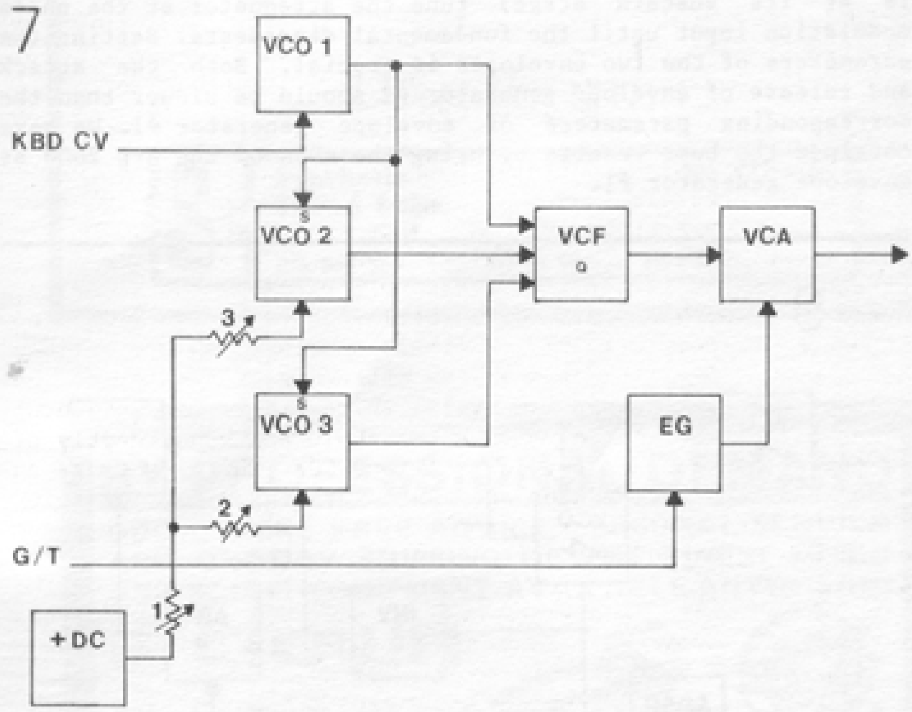

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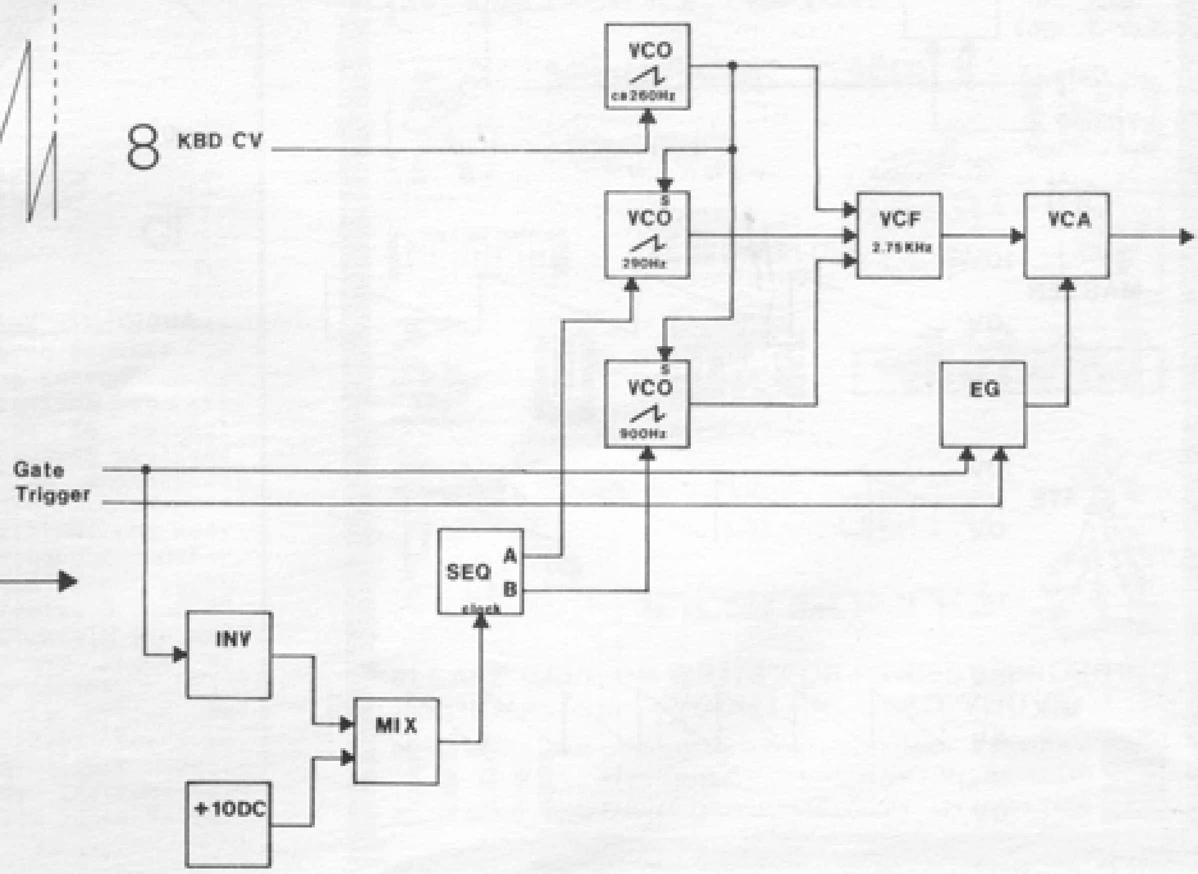

envelope, causing the sound to be articulated. The falling edge of the gate steps the sequencer to the next position and, as the envelope releases, the second vowel sound is heard. Of course the patch could be expanded to create more complex speech sounds. We could add additional sequencers, envelope generators, modifiers, variously filtered noise for different consonants, and so on. But the goal here is not to synthesize complex speech. Complex speech synthesis requires a lot of hardware and is well beyond the scope of this article.

Another possible voice patch is a variation on the standard OH BOY and OH YEAH patches which Arp has previously published. Better vocal quality is achieved in the patch by syncing the VCOs as shown in figure 9.

Yet another application for Phase Modulated Sync is to achieve some interesting timbral modulation effects. For example, two sawtooths of the same frequency which are synced 180 degrees out of phase as shown in figure 10, and then mixed together produce a third sawtooth which is an octave higher. In the frequency domain, the odd numbered harmonics have been cancelled.

This feature can be used to approximate the onset behavior of a violin. According to Winckel (op. cit.), between 40 mS and 100 mS after a violin sound has initiated, the second harmonic (the octave) predominates. By using an envelope generator to modulate the phase angle between the master and slave VCOs, we can approximate this phenomenon. A suitable patch is illustrated in figure 11.

The VCOs are tuned to unison and synced from the rising slope of a sawtooth to an initial phase difference of 0 degrees. While holding a key down (so the envelope generator is at its sustain stage) tune the attenuator at the phase modulation input until the fundamental disappears. Setting the parameters of the two envelopes is crucial. Both the attack and release of envelope generator #2 should be slower than the corresponding parameters of envelope generator #1. We have obtained the best results by using the ADSR of the Arp 2600 as envelope generator #1.

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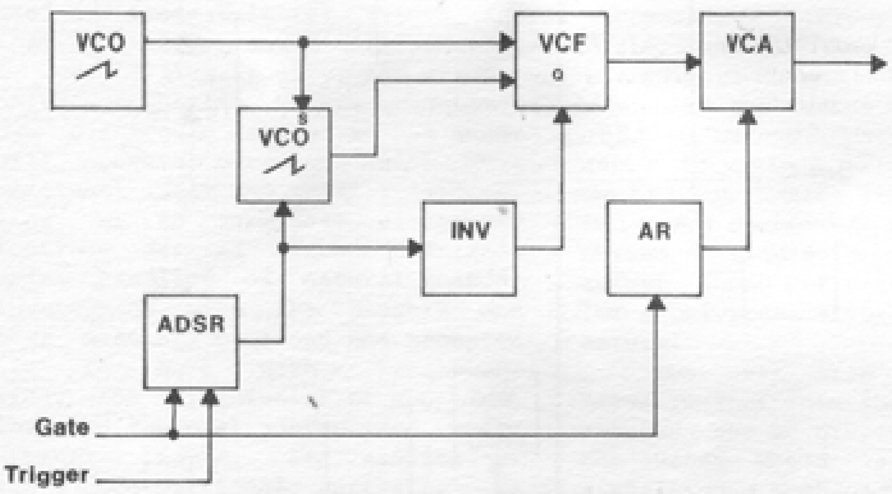

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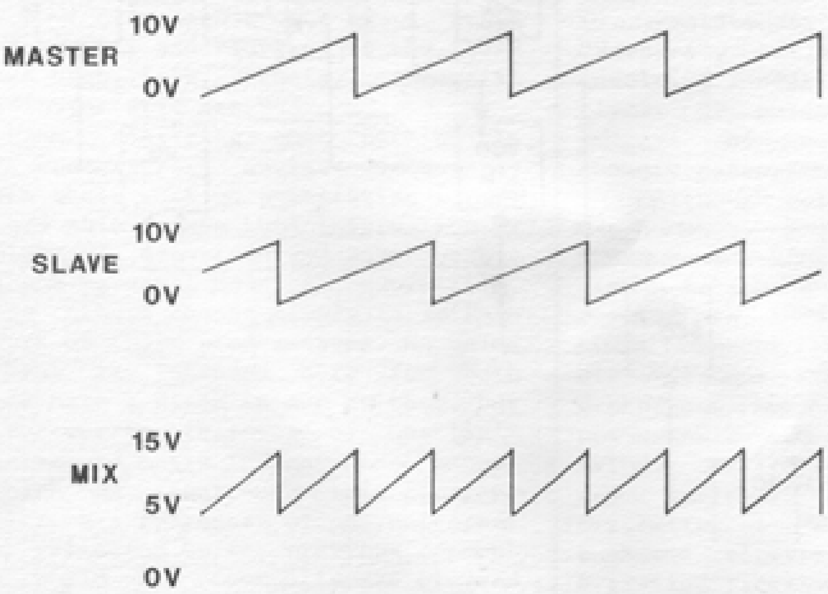

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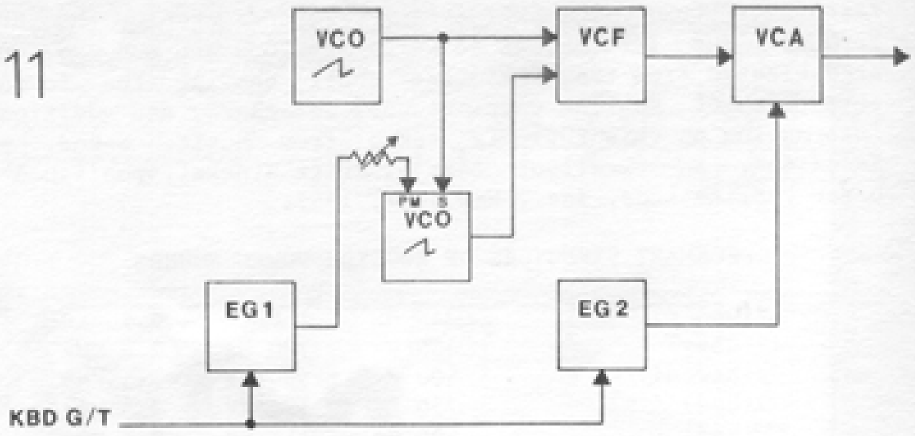

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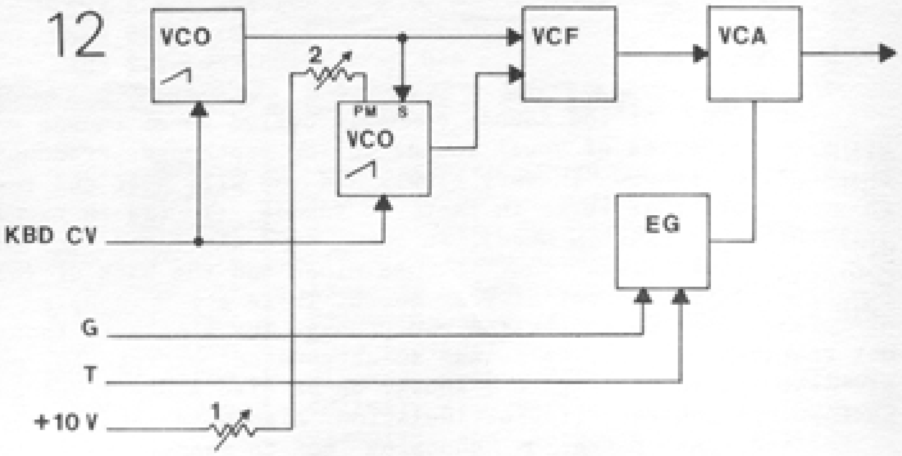

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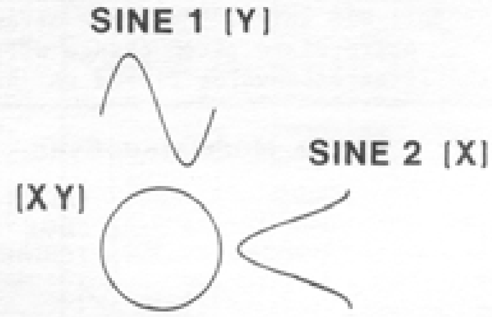

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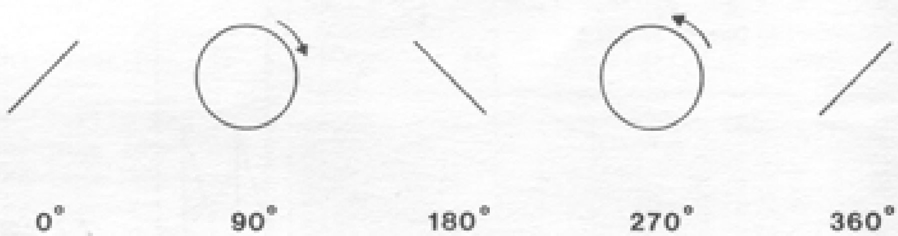

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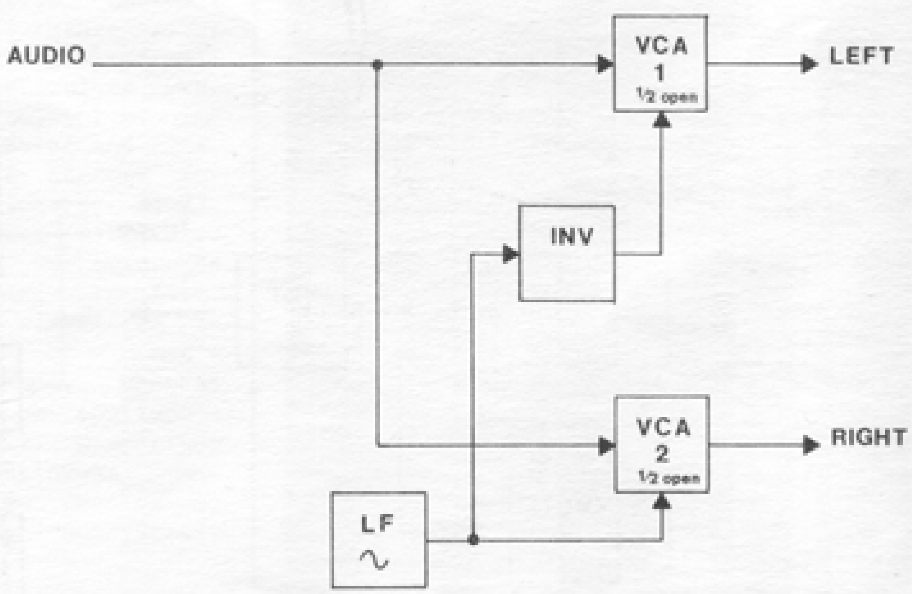

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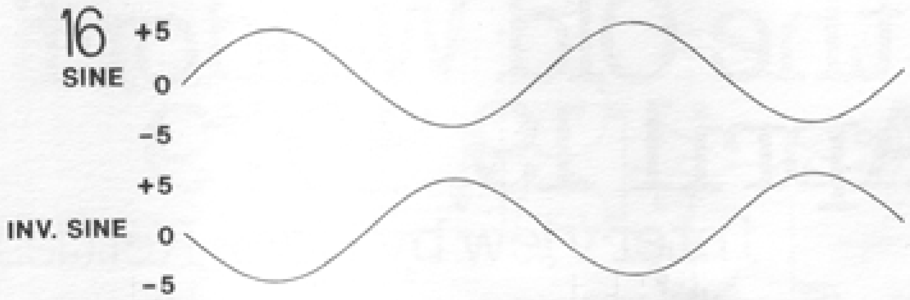

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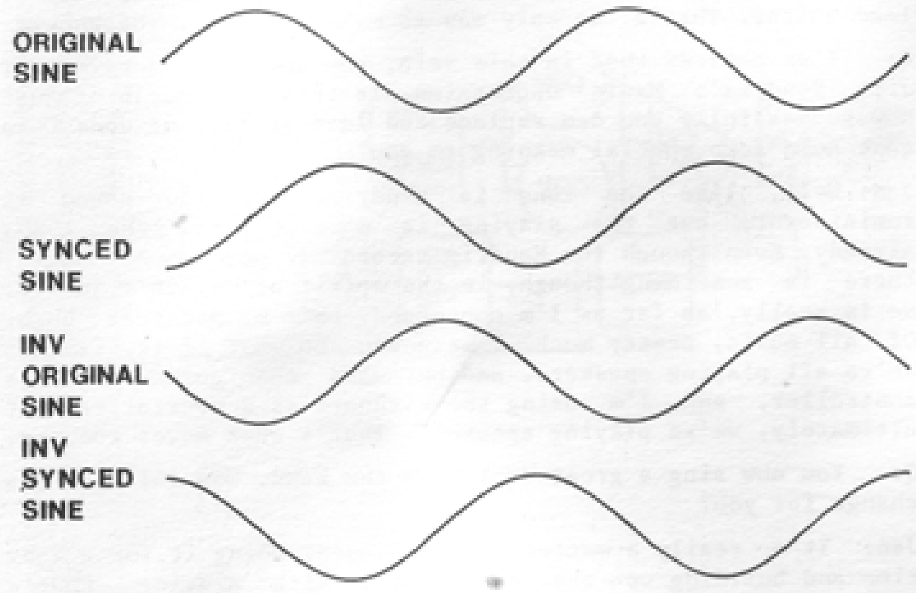

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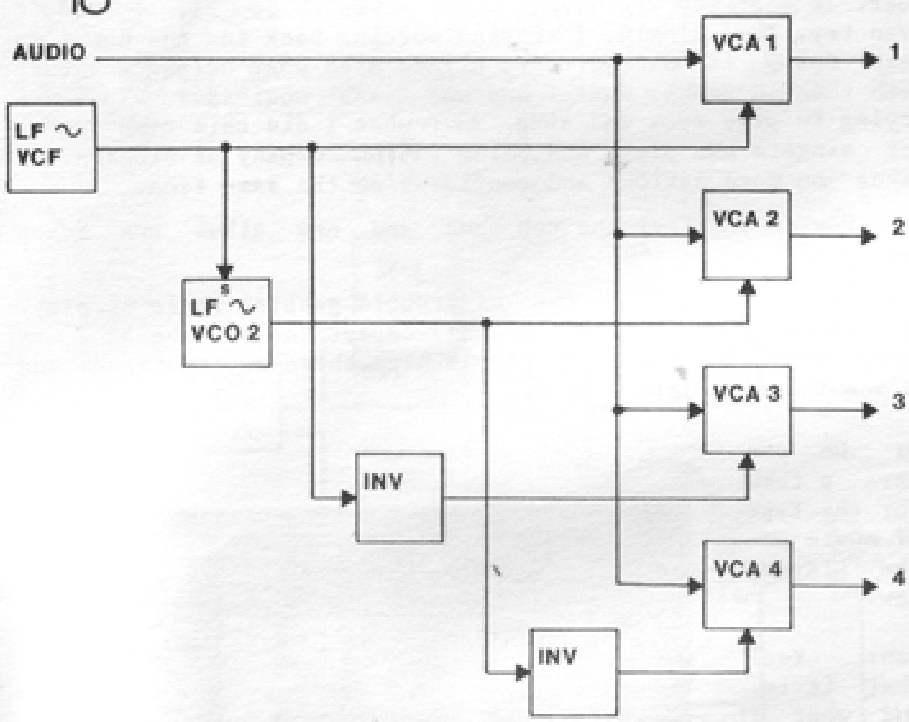

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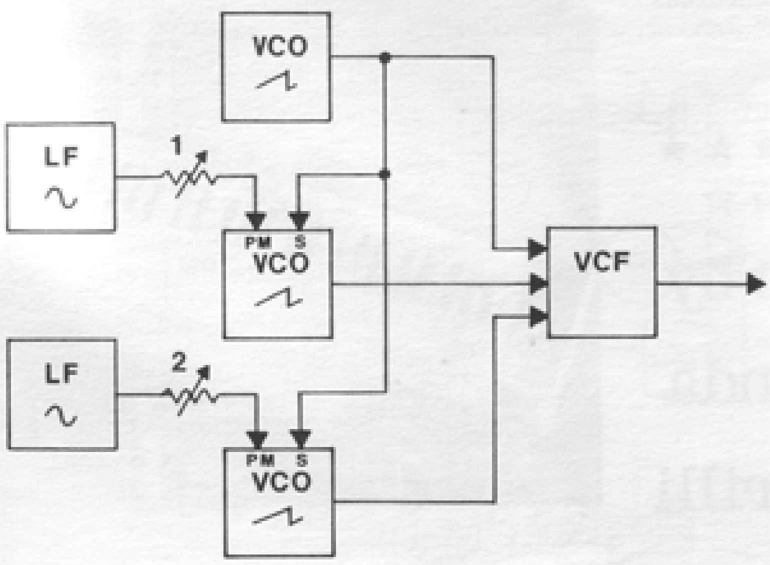

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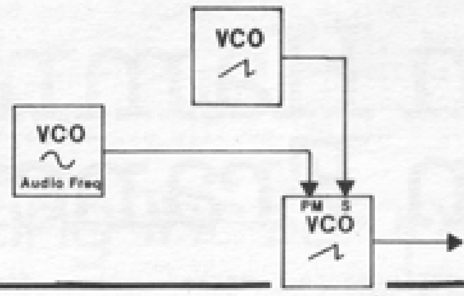


Another possible use for this effect in some keyboard applications is represented in figure 12. The offset controls the phase angle and it is pre-tuned so that, as attenuator #1 is completely opened, the phase of VCO 2 is shifted so the fundamental again disappears and the second harmonic is accentuated. This attenuator is then "played" to produce timbral changes which enhance the melodic changes as played on the keyboard. On notes of particular musical emphasis, the stressed pitches of each musical phrase, the pot can be fully opened causing a frequency doubling which adds additional sonic interest and further accentuates these notes.

By inverting the synced sawtooth before mixing, the resultant waveform is a square wave. In the frequency domain, the even numbered harmonics have been cancelled. Modulating the amplitude of either the master or slave will modulate tha amount of even-harmonic cancellation in the mixed waveform. Modulating the phase relationship between the master and slave results in pulse width modulation of the resulting waveform.

Another application for the Phase Modulated Sync is to generate stable Lissajous patterns for laser deflection and oscilloscope art. At the basis of both of these is the assumption that one will deflect a beam of light horizontally (on the X axis) and the second will deflect the same beam vertically (on the Y axis). Shown in figure 13 is a simple patch in which sine #1 deflects the beam horizontally and sine #2 (90 degrees out of phase) deflects the beam vertically. The resulting XY display is a circle.

By modulating the phase relationship from 0 to 360 degrees, we achieve the displays shown in figure 14. Notice that after we pass the 180 degree phase angle, the beam changes its direction of rotation. Using a low frequency sine to modulate the phase results in a continuous and repeating change in the pattern. Of course, using more complex wafeforms to deflect the beam make more interesting visual patterns.

An application which is closely related to creating visual Lissajou patterns is using two synced VCOs to produce the control signals for quad panning. to begin with, look at the patch in figure 15 for simple stereo panning effects. (On the Arp 2600, the second VCA would be replaced by the ring modulator, direct coupled.) The relationship between the sine and the inverted sine should be obvious; as the sine "opens" VCA 1, the inverted sine "closes" VCA 2 and vice versa.

For quad panning, we require 4 VCAs as in figure 16. The original sine and inverted sine control VCA 1 and VCA 3. The remaining 2 VCAs are controlled by a second sine tuned in unison with the original sine and synced 90 degrees out of phase. This sine controls the second VCA, and the inversion of this sine controls the fourth VCA. The panning is accomplished, then, with these tour signals.

The panning occurs in a circular fashion among 4 speakers. A complete patch is presented in figure 18. Changing the phase relationship between the first and second sines such that the second sine preceeds the first sine by 90 degrees reverses the direction of rotation in the four speakers.

Synthesists who have experimented with chorus effects can appreciate the following patch. The phase modulation used in figure 19 is a simulation of the various cancellations and reinforcements which occur in multiple unison voicings.

A patch of similar nature but which produces, at least perceptually, a radically different result is offered in figure 20. The phase is modulated at audio rates producing a kind of "equivalent FM" timbre. The advantage of this over other FM timbres achieved on the 2600 is that there is no apparent pitch shift as the amplitude of the modulating waveform (depth of modulation) is increased. In addition, these timbres will accurately track the full range of the keyboard.

In all, a modified 2600 is a significant improvement over the stock instrument. Rivera Music Services has installed these modifications for Boston area synthesists over the last few years and all the modifications are field tested and carry a full one year warranty. A second modification package built especially for the Minimoog is also available. For additional information, contact: Rivera Music Services, 48 Brighton Ave., Boston, MA 02134, Ph. 617-782-6554.

Jan Hammer at the Old Waldorf San Francisco, April 19, 1979

interview by Buzz Kettles

This performance marked the first San Francisco appearance of Jan Hammer with his new band "HAMMER". A short interview was conducted with Jan and his new band members backstage between sets.

The new band consists of Jan featured on lead synthesizer, and occasional rhythm keyboards and background vocals; Bob Christianson on lead vocals and rhythm keyboards; Colin Hodgkinson on lead vocals and bass; and Greg Geya Carter on drums.

Q: How long has the 'HAMMER' band been together?

Bob: Roughly since October.

Q: Did the band play on the new album (Black Sheep, Electra/Asylum Records)?

Bob: Well, none of us actually played on the new album. But we all sing on it. Actually, the entire album was done, including vocals, but we redid them.

Q: What stage setup are you currently using?

Bob: Jan uses a portable keyboard. It's basically a Minimoog. He doesn't have his new axe with him, unfortunately because it's been having some problems. The new one's keyboard itself is a Probe (originally designed/specified by/for Roger Powell, sometimes termed a Powell Probe. -ed.), but it's attached to a six voice Oberheim. But it's got lot's of problems. There's just a lot of growing pains. It's just that there are a lot of things that he (Jan) wanted Oberheim Expander Modules to do that they have just never done. Things like low note rule, like Moog. This is no problem in polyphonic situations, but when it's used in unison mode, it shows up. This new keyboard, the new Probe, is even a lot different from Roger Powell's. In unison, it bends normally, but in polyphonic mode, it's got a different type of bending where you can pick any note and bend that note. It also has a full set of Expander Module controls, and the whole thing is multiplexed down through a very small cable. The current portable keyboard weighs quite a bit, but this new one weighs only eight pounds. It's like a feather. It's made totally out of plexiglass.

At this point, Jan joined the interview.

Jan: What's really bugging me is that I can't even play it because it's busted. I played it for about two weeks, and it's still got growing pains right now and a few things died, and so I can't even play it. But I TALK about it every day... people ask me about it all the time.

Q: Bob said that it had the same controls as found on an Expander Module (Oberheim SEM-1), is this true?

Jan: Yes, but it's more than that actually; it's additive. In other words, I can get pulse as well as sawtooth from each oscillator. So it really gets complex. On the Expanders, you have a choice of either/or. They're controlled by a pot which to the left is sawtooth and to the right is pulse. And the center is off. One oscillator cannot put out both. The new keyboard has switches to add them in or out. I was playing it for about two weeks total, I think. And I've been just blitzed out every night, and then it started wandering and there's all kinds of things, 'cause the clock's really... it's the crucial thing (the multiplexing clock, I assume -ed.). If it doesn't read in the signals right , well... it's not really roadworthy yet, the vibrations are still totally wasting it. Things come loose and so we have to 'bulletproof' it. I HAVE to be able to MOVE!

Q: It seems like being able to move around has changed your style of playing in a way that's quite unprecedented for a keyboard player. Was this a hard transition to make?

Jan: That's it! That's the secret! Really! It makes the music feel different, sound different. It's very physical. I didn't know that. When I was growing up, I would play the piano and all that, and I was probably really just following what my parents told me to do. And it's taken me this long to find out just really what I liked to do. The freedom is amazing. And it's not going to take long. Keyboard Liberation Front has definitely started. Every town I go to, there are crowds of them (keyboard players) with their mouths open... That's it, within two years. The real difference is not just like Gary Wright does it, and Roger Powell does it, but I'm talking about using it as a lead instrument, taking over the role of lead guitar. That's the only way to make any dent, you know?

Q: I've noticed that in this vein, you now do a version of Jimi Hendrix's Manic Depression. Is this an example of just how successfully you can replace the lead guitar, or does this tune hold some special meaning to you?

Jan: Well, like the tune is Hendrix, and the sound is reminiscent, but the playing is more towards the 1980s already. Even though the Hendrix recordings may sound dated, there is something though, in the spirit of it, it's just... He is really, as far as I'm concerned, he's my greatest hero. Of all music, pretty much. I mean you see what it is, is that we're all playing speakers, and he used the guitar as his controller, and I'm using the keyboard as a controller. But ultimately, we're playing speakers. That's what moves the air.

Q: You now sing a great deal with the band. Was this a big change for you?

Jan: It's really a matter of confidence. Doing it for a long time and building up the confidence. With singing, that's really important. I sang when I was a child, and then you go through the funny time and all of a sudden, your voice changes. And you say, "Oh, Oh." And you don't even talk for a year. It's pretty wierd. And until I was 20 or 21, I didn't even try. Then slowly, I started working back in. And now I've been doing it onstage every night. Also what helped was that with the old group, what I had was jazz musicians who were trying to play rock and sing. Now, what I did this time was to get singers who play. And being in the company of singers, it makes you more serious and confident at the same time.

Q: How do you feel the new band and new album are being received?

Jan: I still get a lot of trouble getting radio airplay, because of my past. They (the DJs) expect fusion. The band is doing great. But the radio people have these expectations, and it's not fusion jazz anymore.

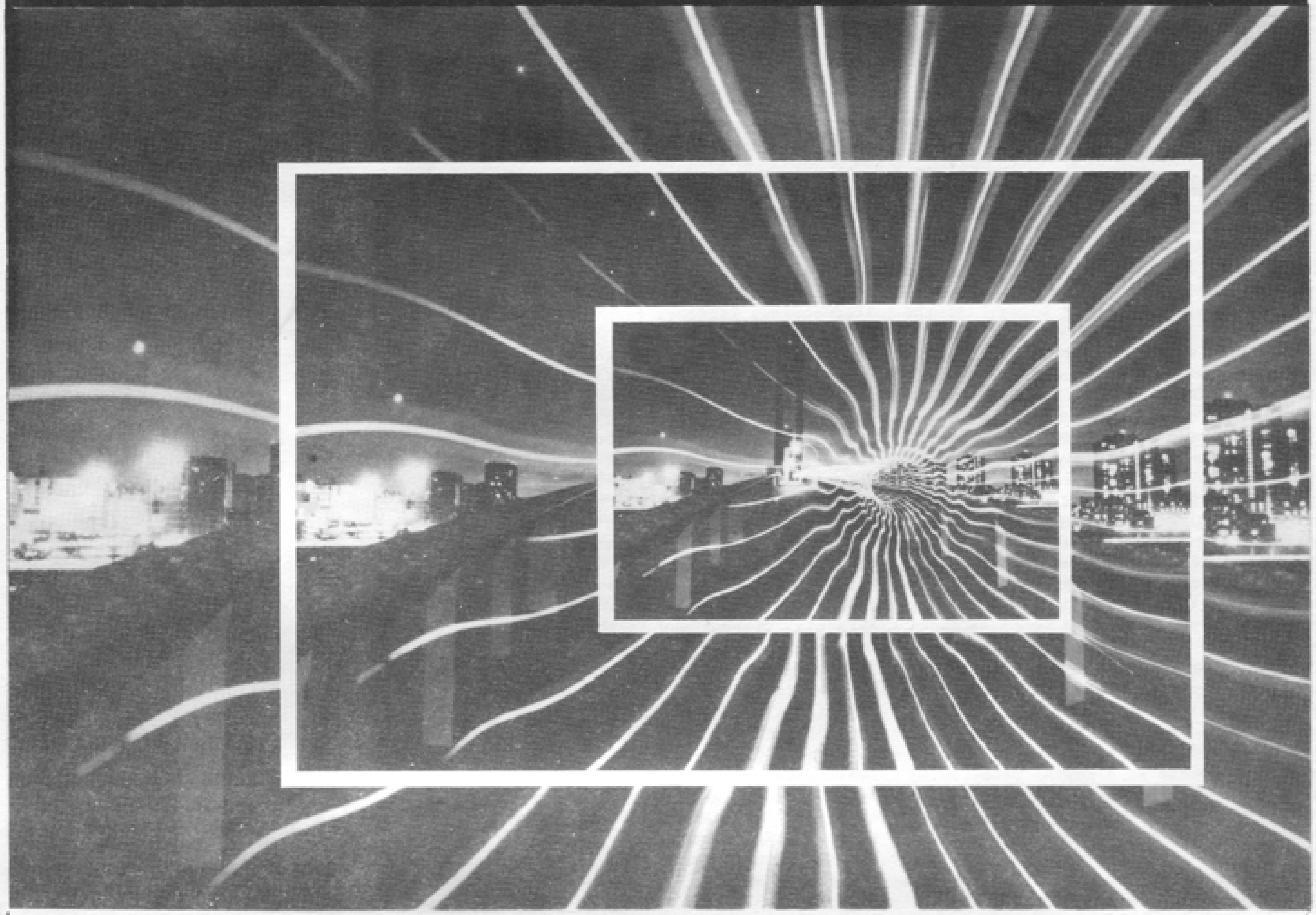
Q: Do you have a term for the type of music you are playing now?

Jan: Yes! What it is, and what I like to call it is ...... "Adventurous Rock" !

★★★★
☆☆

pic by Linda Zerilli

"My turntable was turned on all night playing "Games."



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HOME RECORDING

Budget Studio Equalizer

BY: CRAIG ANDERTON

I recently decided that it was time to upgrade my studio mixer from 6 input mono to 16 input stereo. One of the most important aspects of a mixer is the equalizer circuit employed for each channel; in my particular case, I had several considerations to take into account since there were certain constraints on the mixer. These were:

-- The mixer had to be physically small, owing to the small amount of space allotted to my studio. As a result, the idea of using parametric or graphic equalizers, with their multitude of knobs and switches, was simply out of the question. What I really needed was an equalizer that only required two knobs and a bypass switch.

-- The equalizer had to be as quiet as possible. Since the least number of active stages contributes the least amount of noise, I figured that a single op amp equalizer would be ideal.

-- Low cost was paramount; when you're building 16 of anything, budgets tend to escalate at dizzying pace. So, I set a top limit of $10 per equalizer.

-- The equalizer had to be flexible, preferably with a choice of responses.

With a set of requirements like these, I figured that the best I could do was the old bass/treble, boost/cut circuit that is traditional on budget consoles. But somehow, that didn't sound too exciting -- or anywhere near as versatile as I wanted. There had to be a better way....

In addition to the above considerations, I had also been doing a lot of thinking about consoles recently, and came to certain conclusions about the "ideal" console equalizer. First of all, there has been a trend towards including increasingly sophisticated equalizer modules on all consoles. While this is great for "cost-is-no-object" situations, I've found that very rarely do you really need all that versatility on all console channels, unless the original tracks are so poorly recorded that they need an incredible amount of tailoring just to sound decent. So, I think it's better to have some very sophisticated outboard units for when the need arises, but otherwise, console equalizers can be relatively simple... boost and cut in a variety of bands will really suffice for most situations requiring equalization.

I finally figured out that the "ideal" studio equalizer, in addition to meeting all the size and cost constraints above, should be capable of low frequency shelving at 3 different frequencies, bandpass response at octave intervals throughout the audio range, and high frequency shelving at 3 different frequencies. This should take care of most of the equalization needs I encounter during day-to-day recording.

Well, that's quite a list of requirements. I had almost given up on the idea of something really cheap but really flexible, until I ran across some circuits in Walt Jung's Op Amp Cookbook. This is an excellent book to turn to when you're looking for inspiration, because Mr. Jung throws a bunch of circuits at you that can be used directly, without any further tinkering or modifications. However, the real advantage to this book is that he gives enough information so that you can use his circuits as a point of departure, and alter them to fit your own needs.

One of the circuits he mentions (which has been around for years, actually) is a single op-amp resonant equalizer, which he points out could be turned into a more universal equalizer block by switching various capacitors in and out to achieve low pass and high pass responses... however, he doesn't tell you how to do this, or how to tailor the frequencies for best results in a studio. So, I took the basic idea and ended up with the permutation shown in the nearby schematic.

About the Circuit

S1 is a 2 pole, 12 throw mechanical rotary switch. I know -- it would have been much hipper to use CMOS multiplexers, and maybe one of John's neat little digitizer pots. But remember, low cost and simplicity were of the essence... so I stuck with the good old rotaty switch. Besides, mechanical switches contribute no noise, distortion, or crosstalk.

The 12 positions select the following responses, with R6 offering the boost/cut function:

Position 1: Lowpass starting at 70 Hz
Position 2: Lowpass starting at 140 Hz
Position 3: Lowpass starting at 360 Hz
Position 4: Bandpass centered at 100 Hz
Position 5: Bandpass centered at 200 Hz
Position 6: Bandpass centered at 500 Hz
Position 7: Bandpass centered at 1 KHz
Position 8: Bandpass centered at 2 KHz
Position 9: Bandpass centered at 4 KHz
Position 10: Highpass starting at 2.4KHz
Position 11: Highpass starting at 3.6KHz
Position 12: Highpass starting at 7.2KHz

Of course, with R6 in the cut position, positions 1 - 3 become low cut, positions 4 - 9 become notches, and positions 10 - 12 become high cut.

I gave the pinout for IC1 as a 741 type op amp. Actually the source impedance is so low that a 741 will give reasonable noise performance, although you could substitute a spiffier bipolar op amp for better performance; or, you could use a 4136 and assemble a quad equalizer module.

One point that must be considered is that the input impedance of this module is around 2K ohms, the minimum possible that can still be driven to full output by just about any op amp (except the micropower types). As a result, there should be some kind of buffering stage preceeding this equalizer. Usually this is not a problem, since professional tape deck outputs can drive a 600 ohm line if you're feeding the deck output directly into the equalizer. Alternately, if the first stage you go through in a console is a mic preamp, it will also have enough drive to properly feed the equalizer.

The original circuit given in Jung's book has a 10K input impedance, but requires the use of a 500K pot. Since I had a bunch of 100K pots handy, I figured that I didn't really need a higher input impedance anyway ... and besides, the low source resistance does keep the noise down.

S2 cuts the equalizer in and out. A bypass switch is an absolute necessity for equalizers so that you can compare equalized and non-equalized settings, so don't be tempted to eliminate it to cut costs.

The 10 uf coupling capacitor is required to keep DC level shifts out of the output; the 10K resistor ties the free end of this cap to ground so that you don't get "pops" when you switch the equalizer in and out. The input is direct-coupled to preserve low frequency response. However, if the stage preceeding the equalizer has any DC offset, you'll need to capacitively couple the input. I'd suggest a 10 uf capacitor for best results.

Although you can use as little as a ±9 volt supply, it's worth going for the extra headroom and lower distortion obtainable with a ±15 or ±18 volt supply.

Using the Equalizer

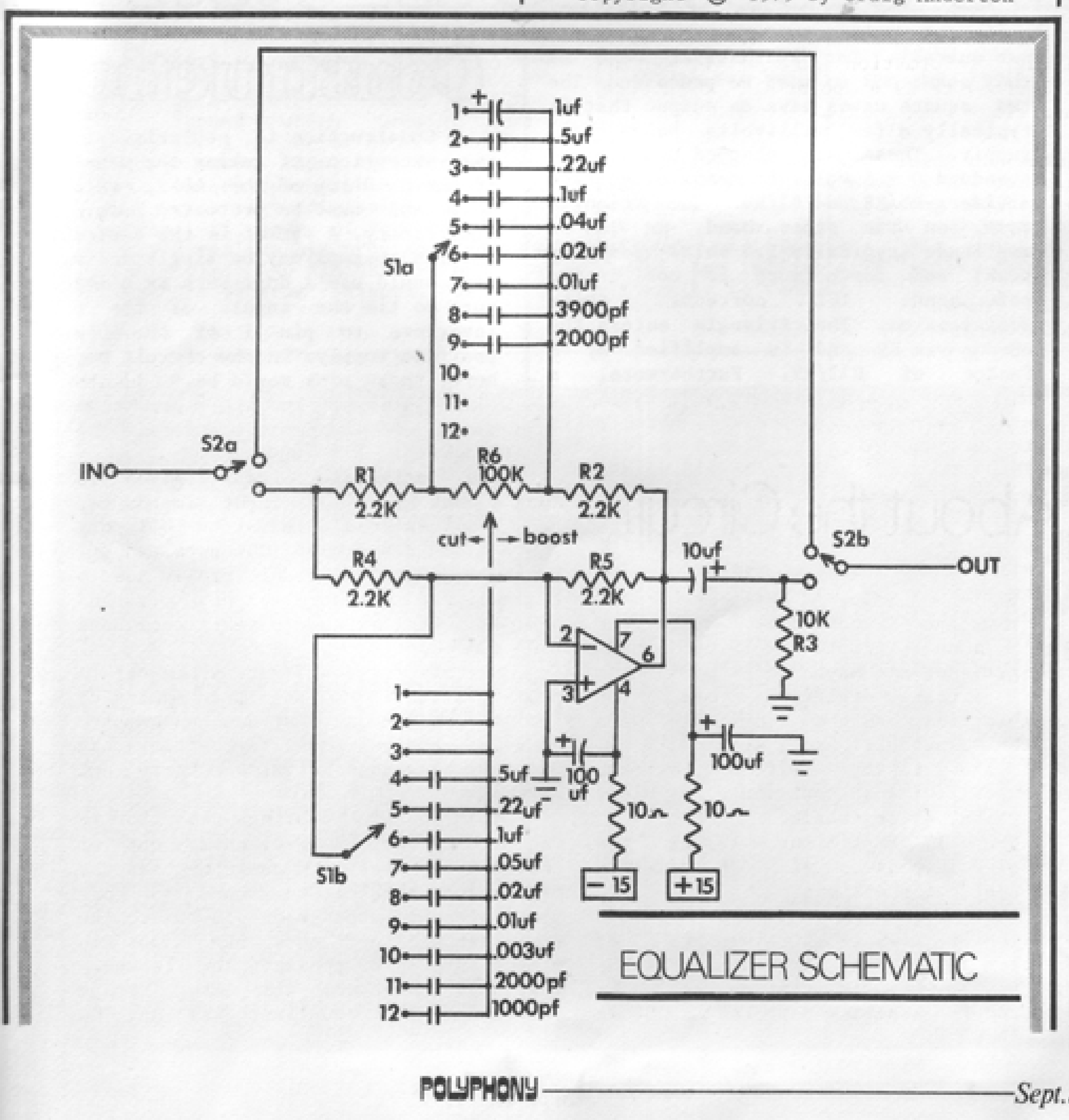
It will take you a while to get acquainted with this thing, despite its deceptive simplicity. For one thing, it is capable of about ±15 dB of boost/cut; so, with large amounts of boost you'll also be upping the volume level considerably. Conversely, when you're notching out 15 dB of signal you'll find that the overall sound will be much, much thinner; this will necessitate a volume boost elsewhere in the system to compensate. The basic point of all this is to adjust the channel fader control to compensate for changes brought about by changing the equalizer's boost/cut control, or you will have an unrealistic picture of what the equalizer is really doing.

The bandpass responses are very broad, which I prefer for a "general purpose" unit (remember, I have outboard stuff to take care of the times when I need really sharp peaks). You'll find that this equalizer works very well with voice, guitar, keyboards, etc. to shape the overall "character" of the sound.

Another point of interest is the 100 Hz bandpass position. Now, most people would question the validity of having a 100 Hz bandpass response, since the low pass sections can take care of the bass. However, you'll notice that with most inexpensive multitrack recorders (and some expensive ones, too) that there will be a hump in the response around 100 Hz. While this is normally not too objectionable, when you do a lot of ping-ponging from track to track you can build up quite a big response anomaly at 100 Hz, giving an unnaturally bassy effect. This hump will typically be about 4 or 5 dB, so one bounce and you've got at least an 8 dB 100 Hz peak...2 bounces and you've got a 12 dB peak! By notching out some of the response at 100 Hz as you record to compensate for the peaking effect on playback, you can bring the response anomaly under control.

There's not much else to say, except that this tone block has many applications beyond the home recording studio. In addition to retrofitting old consoles that only offer the bass/treble, boost/cut function, you can build this circuit into musical instruments to give them some on-board equalization facilities. This is particularly effective with older instruments that have simple tone controls, such as the old "hi-cut" types found on guitars and carried over to other instruments.

So there it is....an equalizer that is simple, inexpensive, effective, versatile, and easy to build. Once you get the hang of it, you'll wonder how you got along without it!



EQUALIZER SCHEMATIC

Build This!

COMPLEMENTARY OUTPUT LFO

BY THOMAS HENRY

Here is an LFO (low frequency oscillator) with complementary triangle and complementary square wave outputs. The module is inexpensive and easy to build and, in addition, lends itself well to modification for those special LFO applications you have in mind. This LFO has a very wide range, and the fact that it is especially weighted toward the lower end (two cycles per minute) makes it perfect for long sweeping voltage controlled filter and phaser effects. Finally, thereq are LED monitors for you blinky light fans.

Specifications

Supply Voltage:	±9 volts
Current Required:	1mA (-), 12mA (+)
Frequency Range:	.03 Hz to 25 Hz
Output Voltage:	0 - 5 volts

How It Works

Three sections of a 4049 hex inverter form the basic oscillator block. This means of generating a triangle and a square wave from CMOS type digital circuits has been presented in several sources, and is currently being used in some music applications (notes 1, 2, 3). R2 is the rate or frequency control, and offers continuous variation from .03 Hz to 25 Hz. There are three outputs from the basic oscillator, a pair of complementary square waves and a triangle. These outputs are not suitable for synthesizer work as they stand and so must be processed. The two square waves have an output that is typically a few millivolts below full supply. These are chopped down to the standard 0 - 5 volts by means of voltage dividers R5/R6 and R7/R8. The triangle wave, on the other hand, is low in amplitude (typically 2.5 volts peak-to-peak) and furthermore is not ground referenced. IC1 corrects these deficiencies. The triangle enters the op-amp via R9 and is amplified by a factor of R12/R9. Furthermore, a negative voltage is summed into the stage through R10 and R11, which pulls the wave down to a ground reference. This is an inverting amplifier configuration, hence we sum in a negative voltage which, when inverted, yields a positive output. R11 is a trimmer which allows precise adjustment of the ground reference. The output of the amp goes to D1, an LED, for visual display of the frequency, and to trimmer R15 which cuts it down to a precise 5 volt maximum.

The triangle wave output next feeds to IC2 which serves a similar purpose as IC1 in that the signal is mixed with a negative voltage, and the resultant is inverted, yielding a positive signal 180 degrees out of phase with the input. The output once again is fed to an LED. We now have two complementary triangle waves to go with the two complementary square waves, all 0 - 5 volts.

Note that the two op-amps also allow us a considerably greater fan-out, or ability to drive more modules, than the 4049 would allow by itself.

Construction

Construction is non-critical with the exception of taking the usual care in the handling of the 4049, which is CMOS and must be protected from static electricity. A socket is the best way to go. The op-amps may be single units, or you could use a dual such as a 1458. Be sure to tie the inputs of the unused inverters to pin 1 of the 4049, the positive supply. In the circuit as shown here, these pins would be 9, 11, and 14. Finally, if you go the printed circuit route, be sure to mount trimpots R11, R15, and R16 where they are easily accessable. The whole circuit with front panel facilities consisting of four jacks, two LEDs, and one rate control will fit comfortably behind a panel as small as 2" X 4".

Power Supply

The output of the basic oscillator block is a function of the supply voltage, so to expect results similar to mine, you must use a similar power supply. The system was designed around a ±9 volt supply, so if yours is much different than this (say, by a volt or two) you should install two 9.1 volt zeners and two limiting resistors to bring the supply down. Current drain is 1 mA on the negative side and 12 mA on the positive supply.

Calibration

Hook up the LFO to the power supply. Set the rate control to a low frequency and monitor the square wave at J1. Verify that it gives about 0 - 5 volts of output. Repeat for the other square wave at J2, and also check to see that the outputs at J1 and J2 are 180 degrees out of phase with each other. Next, monitor J3 with a VTVM or an oscilloscope. Adjust trimmer R15 until you read 0 - 5 volts peak to peak. Next, ground reference this wave by adjusting trimmer R11. Now monitor J4 and adjust trimmer R16 for a ground referenced wave. The output amplitude should still be 0 - 5 volts peak to peak. Verify that the LEDs flash.

These measurements all have a certain amount of error to be taken into consideration, due to the tolerances of the resistors and supply line differences. If you cannot get the triangle waves referenced to ground, reduce the values of R10 and R17 to 270K and 220K respectively and this should take care of the problem.

Going Further

It is almost as easy to build this unit in a dual configuration (since each LFO only requires three inverters and a 4049 has six), using just two chips--the 4049 and a quad op-amp such as an LM324 or a 4136. Applications include summing together two waves of differing frequencies for an irregular control voltage pattern.

Conclusion

The LFO can be used for all the standard applications, and the multitude of outputs allows great versatility. Square waves are nice for generating "trills" on a VCO, and triangle waves are useful for sweeping filters or phasers. Complementary outputs allow you to sweep two paralleled filters in opposite directions for a thicker sound. Give this circuit a try and I think you'll agree it offers a lot of options at an extremely low cost.

Notes

1) Theory of the use of digital circuits for oscillator applications is covered concisely in Linear Applications, Handbook 2, National Semiconductor, Application Note AN-88.

2) Some practical considerations of the basic circuit are taken up in Popular Electronics, August 1974, page 61, "A Guide To CMOS Operation", Walter G. Jung.

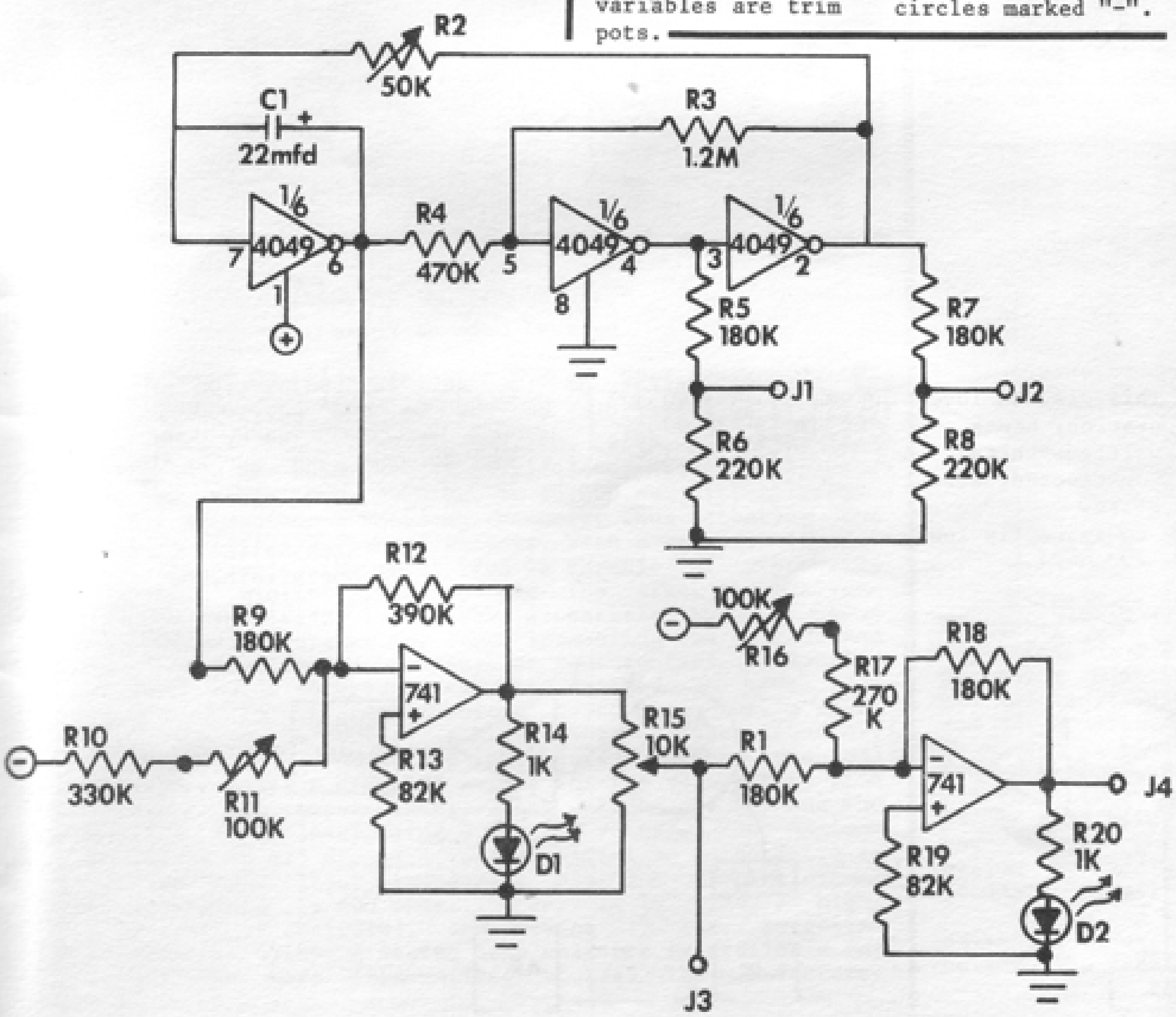
3) Craig Anderton made use of this basic oscillator block in a tremelo circuit in Contemporary Keyboard, August 1979, page 69, "Electronic Projects, Tremelo Part 1".

schematic

Inverters: 1/6 4049
Op Amps: 741, etc.

R2 is the frequency control; all other variables are trim pots.

Power Supply: ±9 V. Besides powering the op-amps, also supply +9 to pin 1 of the 4049, and -9 to all circles marked "-".

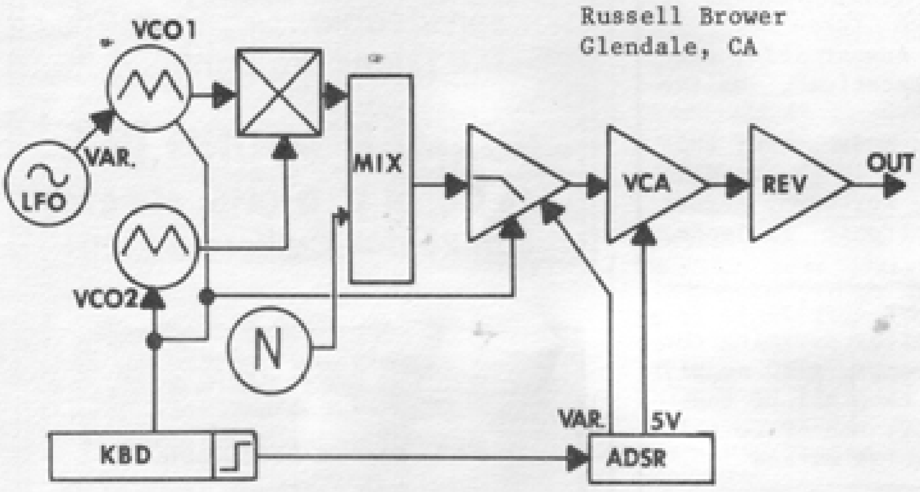


PATCHES!

Human Whistle

VCO 1 - 52%   Adjust these finely for high, resonant
VCO 2 - 50%   whistle tone in keyboard midrange.
Mixer - Tone 50%, Noise 25%, Full output
VCF - High range, Keyboard track mode
      Freq. 65%, "Q" 80%
LFO - approx. 7.5 Hz, very slight output level
ADSR - A 25%, D 45%, S 70%, R 20%, output 30%
Reverb - medium level
Keyboard - midrange, glide 20%
* Fine-tune all settings for realistic sound.
* Play within 1.5 octaves at high end of keyboard.
* Be sure to allow the synthesizer time to "inhale" during long passages!

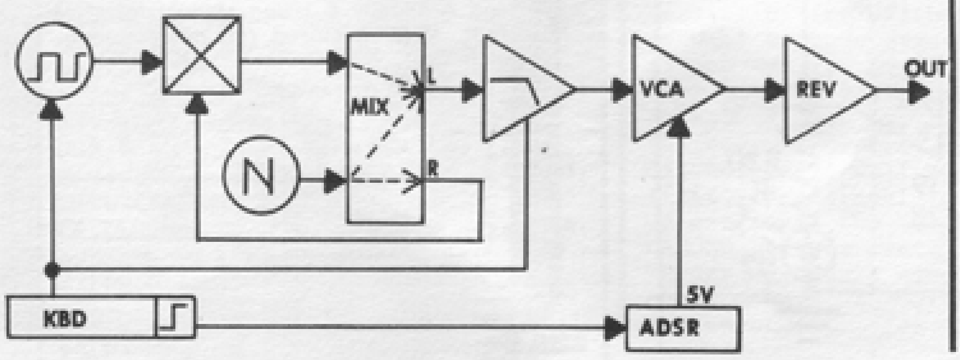
Russell Brower
Glendale, CA



Thunder and Lightning

VCO - Init. PW 20%, Init. pitch 0%
Bal. Mod. - Sensitivity at full for overload
Mixer - Mix equally with panning of signals as shown
VCF - High range, Keyboard track mode, Init. freq. 50%, "Q" 0%
ADSR - A 0% for crash or 40% for roll, D 50%, S 60%, R 100%
Reverb - 50% mix with normal signal
* If possible, use Seque 1.0's event mode to produce a trigger pulse sequence to the ADSR transposable from the keyboard for a rolling thunder.
* To produce a crack of lightning, set ADSR to fastest attack, overload VCA input sensitivity, hit high end of keyboard, fade down VCA sensitivity and slowly move down keyboard.
* This sounds best from large speakers due to extremely low (even subsonic) frequency distribution.

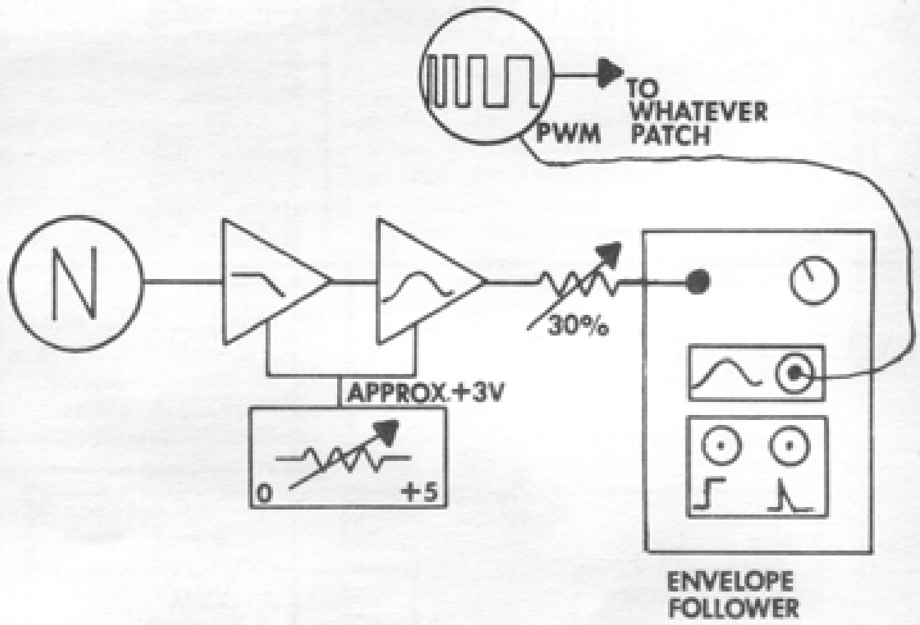
Russell Brower
Glendale, CA



Pseudo Chorusing

Both filters are used in series to avoid upper frequency modulation of the VCO by the noise (which will sound like distortion). The attenuator is at 30%, as is the threshold/sensitivity of the envelope follower. The "Q" of the filters are initially set at midrange. Use for any patch where you are using a pulse wave. Set initial pulse width to 50%. Vary all controls for altered effects. Sounds best when used in dense multi-track work, with all the voices being independently randomly phase modulated.

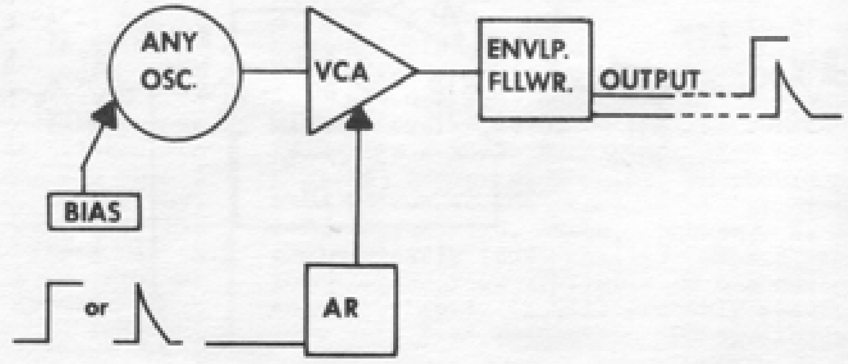
Tim Fluharty
Farmington, NM



Delayed Gate or Pulse

The Attack control of the AR, in conjunction with the threshold or sensitivity control of the envelope follower, detirmines the delay of the trigger. The trigger or gate output of the envelope follower is now used as the delayed output.

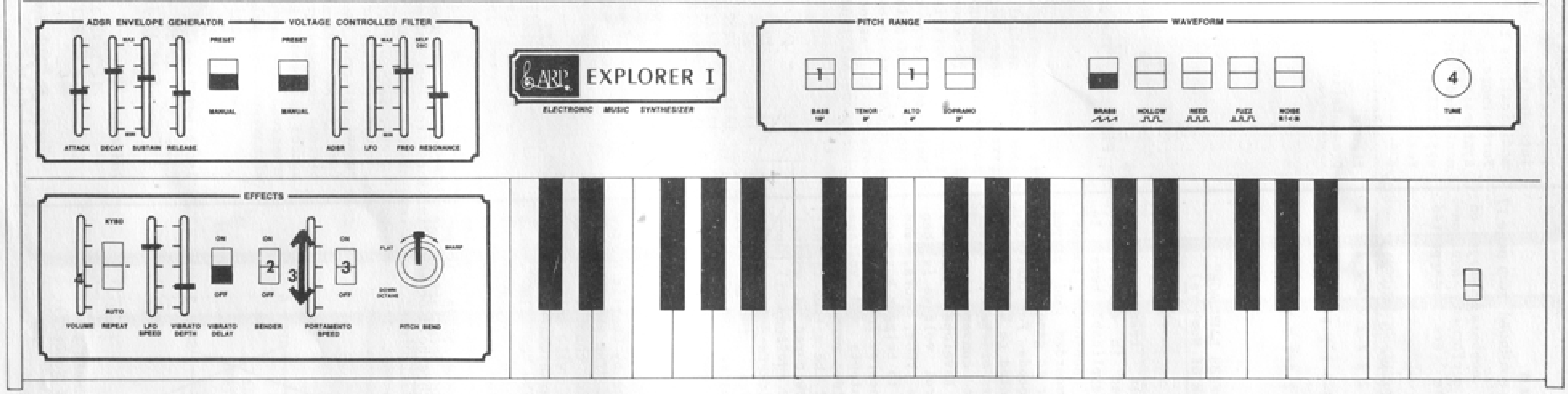
Rich Sulin
Northfield, OH



Stratovarious '75

1. Use 16' for bass, 4' for violin
2. Use bender for added effect
3. Use portamento for added effect
4. Tune & volume as needed

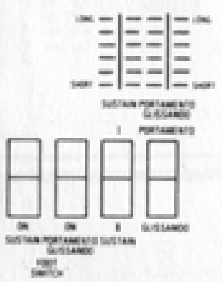
Bill Behrendt
Campbell Hall, NY

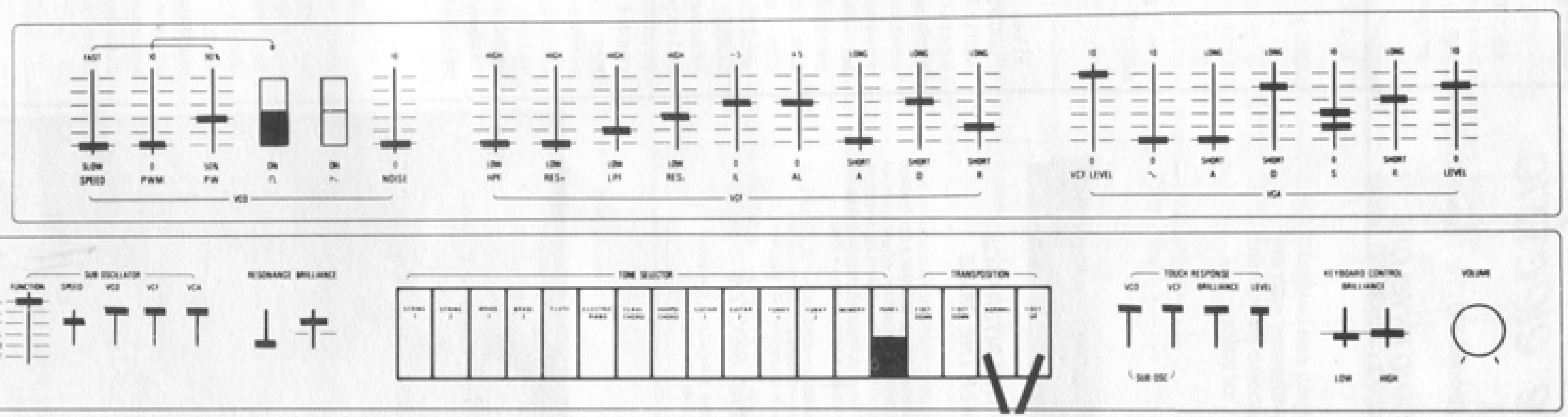


Yamaha CS-60

Steel Drum

Bill Boydstun
Oklahoma City, OK





Size of Drum

Charlie...

.... continued from page 17

in figure 3; the numbers (played on boatwhistles) are performed horizontally, and spaces are interpreted proportionally.

```
1 1 1 1 1 1 1 1 2 2 2 2 2 2 2 2 2 2 2 2 2  2

2  2 2 2 2 2 2 2 2 2 2 2 1 1 1 3 2 2 2 2 2

2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2  2

2 2 2 2 2 etc.
```

Figure 3. "TOADFISH LANGUAGE" (from The Book Of Numbers)

Electro-acoustic transformations like those discussed in relation to "COUNTING" are also applicable here. In addition, authentic non-human communications may be recorded and played through a synthesizer to generate and/or control electronic sounds. For instance, suppose that you wanted to base a piece on the language of crickets. After recording a cricket conversation send the tape through an envelope follower; the resulting control voltage is then available to activate VCOs, VCFs, and VCAs, as well as to trigger envelope generators and sequencers. Inclusion of multiple voltage outputs and voltage inverters can provide rather complex results. The point of the matter is that you will have structured a musical composition (or sound environment) upon the language of crickets-- a perfectly valid language even though not understood by humans. It is the responsibility of the composer to make the piece work.

The final piece by Charlie Morrow that we will discuss, "THE NUMBER SIX", is written for solo chanter and six microphones-- each with its own loudspeaker. The performance layout is shown in figure 4. Like "COUNTING", this work reduces music to the logic of numbers, but each of its six sections progresses gradually toward more complex sonorous relations via echos (repetitions) and numeric permutations of retrograde and elimination.

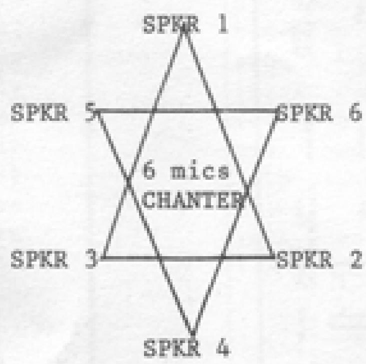


Figure 4. "THE NUMBER SIX" (from The Book Of Numbers)

Spatial effects are of major importance, and they may be enhanced by introducing different electronic modifications to individual microphones. Furthermore, all of the standard tape, synthesizer, and computer techniques mentioned earlier are also applicable, especially if a piece of this nature is realized for solo tape or for chanter and five synchronized tapes.

Part I of "THE NUMBER SIX" (see figure 5) is a straightforward presentation that enables the listener

```
6      6      6      6      6      6
6      6      6      6      6
6      6      6      6
.      .      .
.      .
.      .
6
66     66
666    666    666  etc.
```

Figure 5. "The Number Six", Part I

to percieve the sounds in the aforementioned spatial arrangement, while part II (see figure 6) introduces the numbers one through six. Part III (see figure 7) incorporates a more elaborate form of repetition. When looking at these examples, remember that they are based on equal pulses whereby each space is equivalent to one pulse. Also, each column is associated with its own microphone and loudspeaker.

```
1          2          3
11         22         33
111        222        333
1111       2222       3333
11111      22222      33333
111111     222222     333333    etc.
```

Figure 6. "The Number Six", Part II

```
1
1122          1122
111222333     111222333     111222333  etc.
    .             .             .
    .             .             .
    .             .             .
```

Figure 7. "The Number Six", Part III

The overall triangular shape of the number patterns is continued in Part IV (see figure 8), but increased repetitions (echoes) gradually fill in empty spaces (silence). Part V (see figure 9) preserves the echo principle, combined with systematic deletion of numbers to

```
1st echo: 1
          1 12 12
          1 12 12 123 123  etc.

2nd echo: 11
          11 1122 1122
          11 1122 1122 112233  etc
```

Figure 8. "The Number Six", Part IV

```
1st echo: 1           1
          12         21
          123       321
          1234     4321
          12345   54321
          12345654321      etc.
           234565432
            3456543
             45654
              565
               6
```

Figure 9. "The Number Six", Part V

```
2nd echo: 1           1 1           1
          1           1 1           1
          12         21 12         21
          12         21 12         21
          123       321 123       321
          123       321 123       321
          1234     4321 1234     4321
          1234     4321 1234     4321
          12345   54321 12345   54321
          12345   54321 12345   54321
          12345654321 12345654321
          12345654321 12345654321
           234565432   234565432
           234565432   234565432
            3456543     3456543
            3456543     3456543
             45654       45654
             45654       45654
              565         565
              565         565
               6           6
               6           6
```

Figure 10. "The Number Six", Part VI

cause more varied spatial movement. Notice also how the modes of retrograde and inversion affect the number patterns. Finally, part VI (see figure 10) intermixes all of the preceding processes to culminate in diverse rhythmic and spatial activity.

The compositional principles of Charlie Morrow reflect many of the tendencies predominant among young composers; theory of communication, linguistics, mathematics and logic, laws of nature and the universe, and technology. His methods, like those of earlier composers and theorists, serve as a point of departure. Through his systematic work, Charlie has enabled artists to explore less common avenues of structural organization.

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_____. "What's Going On?", EAR Magazine IV, 8/9 (Dec. 1978/ Jan. 1979): 7.

EQUIPMENT EXCHANGE

List equipment or services for sale, trade, or wanted, plus job openings, positions wanted, events and so on. Keep listings brief; enclose $1.00 for each 10 words. Prices, zip, and phone numbers count as a single word each. Display classifieds are available for commercial ads; rates are $10 per column inch (minimum, negative supplied). Respond directly to the advertiser. Please don't write to POLYPHONY. Polyphony is not responsible for claims made in ads, or the results of any transactions. We reserve the right to edit or refuse any ads submitted.

TEAC Multitrack 4 ch. reel deck Model TCA-42. Modified for Simul-Sync. Functional equivalent of 2340SX except auto reverse in stereo. Excellent operating shape, but looks like a dated piece of equipment. $600 firm. Paia modules for sale: (2)4720, (2)4730, (2)4740, (4)4770, 4710, 4711, 4712, 2720-1, -5, -11, -12, 8780, (2)4761. Some modifications. DAC and square wave outputs need work. All else fine. System price, $375. Consider module trade for custom PC work. David Mays, 212 Elm St., Hurst, TX 76053, Ph. 817-268-5009.

LIVE TAPES: Jan Hammer Group- Chapel Hill, NC- 3/27/79- excellent quality, 90 minutes. Robert Fripp- The Kitchen, NYC- 2/5/78- good+ quality, 90 minutes. All tapes on cassette, Maxell UD, Dolby. $7.00 each, including postage. Add $1.50 for XLII. Gary Broyhill, Rt. 2, Bx. 198, Morrisville, NC 27560.

PROGRAMMABLE Drum Set, assembled, works fine, $90. Dan Vandre, 2104 Alton Dr., Kokomo, IN 46901, Ph. 317-452-7703.

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360 SYSTEMS: polyphonic guitar synthesizer w/ 6 Oberheim expander modules, 6 seperate pitch to voltage interfaces, Rex Bogue guitar with special synthesizer pickups, custom made pedal board that controls all 6 voices. $15,000. 714-548-6826 or 714-871-1923.

SELL or TRADE (for test equipment): Phlanger modified for lower noise, $70. Two Electro-Voice Musicaster PA speakers, good for monitors, 12" bass reflex and VHF tweeter, 30W each, $175. Small patchable, tunable drum kit, snare/cymbal, bass, and strike pad trigger, $30. VCF LP-BP with internal envelope follower, good add-on for Gnome, $20. All units assembled, calibrated, and include manuals, postage, and insurance. Ron Minemier, Bx. 153, Ripon, WI 54971, Ph. 414-748-2793 after 6PM.

ASSEMBLED 4700S, needs minor calibration, must sell, asking $350 with shipping. Contact Bob Celmer, 305 B Vairo Blvd., State College, PA 16801, Ph. 814-237-3934 after 11PM.

4700 SYNTHESIZER, 4712, 4710, 4740, 4720, 4730, (2)4761, (2)4770, 8780, 8782 (homemade case with Pratt Read), (3)2700 modules. All assembled, needs only adjustments, $500. Anthony Lassiter, 630 S. Hermitage #402, Chicago, IL 60612, Ph. 312-942-4837.

4700 SYNTHESIZER, excellent condition, has bandpass filter, ring modulator, sequencer. Also a custom mixer with reverb; and a portable Oz polytonic synthesizer. $375 for all. Greg Narizny, 4449 Morrow Rd., Modesto, CA 95350, Ph. 209-579-3213.

P4700J, CUSTOM front panel and Arp 2600 style case, 4.5 octave keyboard in custom case (all cases covered in leatherette), with 8700 computer, MUS 1.0, and proto of forthcoming Paia video/expander board (2K RAM and interrupt driven video), $750 or best offer. Rick Andrewski, 6015 N. Ross, Okla. City, OK 73112, Ph. 405-843-7209.

GNOME & PROGRAMMABLE Drum set, both factory calibrated & checked, Gnome $65, Drums $100, or $150 for both. David Christensen, 308 Thacher St., Attleboro, MA 02703, Ph. 617-222-2411.

4700J, ASSEMBLED (QuASH and D/A need final assy.), keyboard and computer checked by factory, never used. Also includes 3 VCOs, VCF, 2 Env Gen which need some work. $750 shipped freight collect (certified or cashiers check). Frank Rogala, Box 264, Mackinaw City, MI 49701 Ph. 616-436-5141.

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POLYPHONY

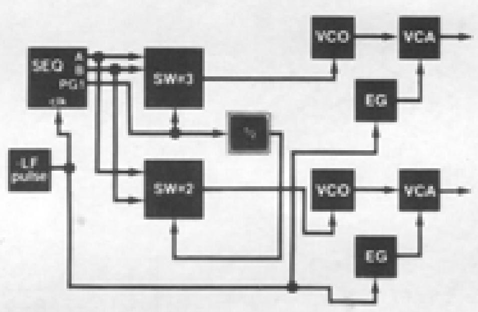
Sequencer & Switches Contest

We at Aries Music are happy to announce our first contest, the subject being discovery of interesting uses for our new AR-334 Sequencer and AR-335 Switches modules. The AR-334 is a potentiometer-memory, 8-step by 2 row sequencer with position gate outputs along with reset and run, enable & step inputs. The AR-335 is a unique set of 4 bidirectional analog switches: 2 single pole, double throw (pulse controlled); 1 single pole, four throw (pulse controlled); and 1 single throw, four throw (voltage controlled).

Contestants will be asked to submit a block diagram for a patch which takes advantage of as many features of these modules as possible, while producing a musically interesting and useful result.

The twenty best patches and their descriptions will be published and sent to all of the entrants in the contest.

The five finalists will have their patches and descriptions published in *Polyphony*, whose readers will be invited to vote for the best patch. The winner will be able to select $600 worth of Aries Music modules. (Which, coincidently enough, is the value of one each, Sequencer and Switches, assembled, or three modules in kit form.)



This patch produces a cyclical pattern of 32 notes.

The finalists will be selected by Bob Snowdale, president of Aries Music; Ron Rivera, designer of the modules; and Mark Styles and Ken Perrin, noted Boston area composers of electronic music.

So send today for the contest details, which will include the 47-page Aries Music *Owner's Manual Supplement* on the AR-334 and AR-335 modules. The contest entry deadline is January 31, 1980.

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PAiA P-4700/J POLYPHONIC...Of Course!

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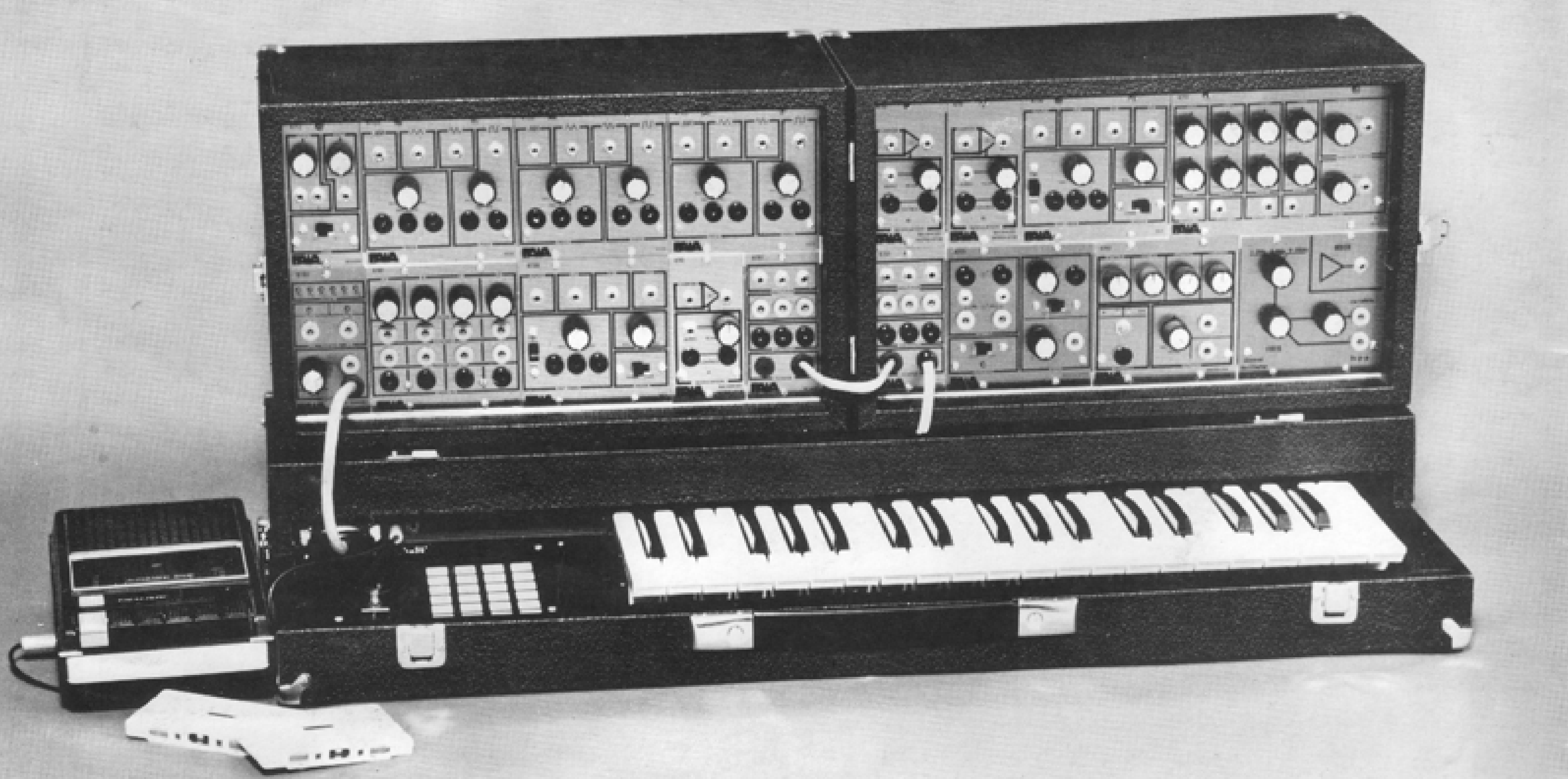

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( ) Please also send complete instruction manual set for the P-4700/J ....$10.00 (refundable upon purchase of P-4700/J Kit)

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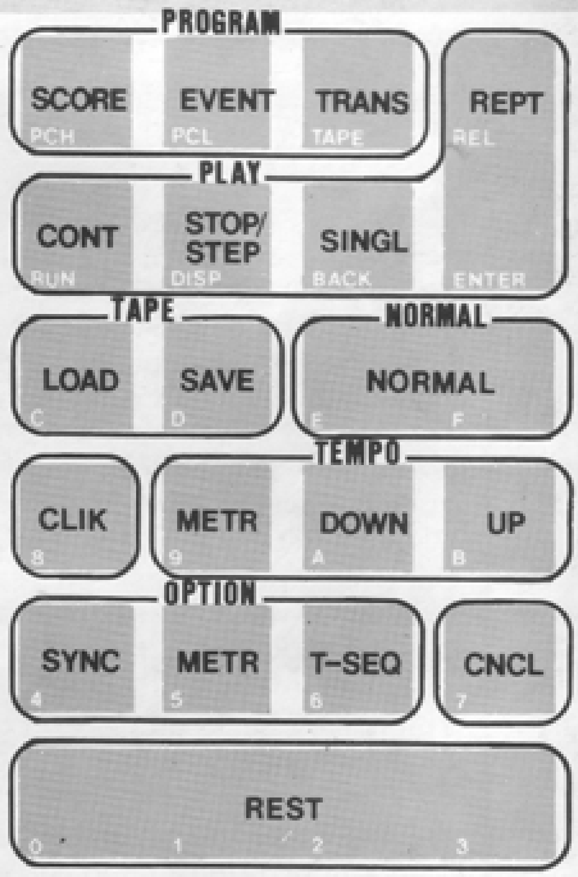

Typical control panel configuration

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