POLYPHONY

ELECTRONIC MUSIC & HOME RECORDING

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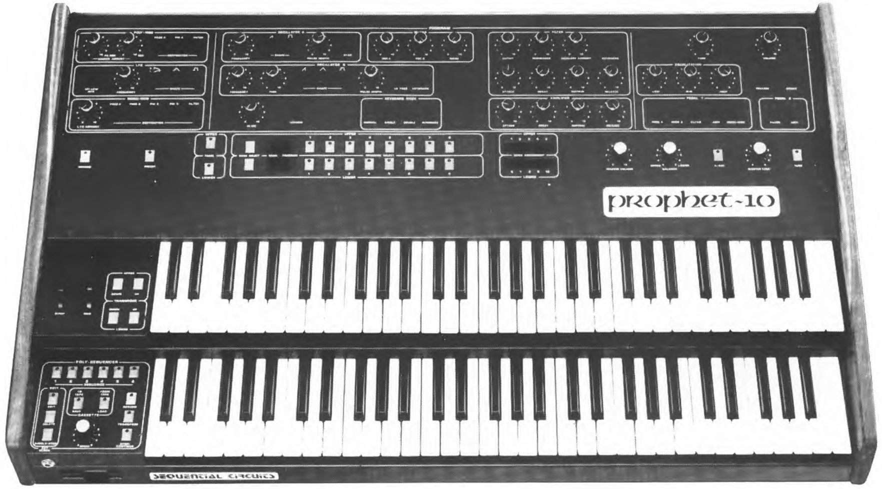
ANATOMY OF A PRIVATE RECORD

AN INTERVIEW WITH DON SLEPIAN

A DIGITAL FILTER

OPTICAL AUDIO CONTROL

THE ULTIMATE KEYBOARD



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REGULARS

This is a special double issue. It spans the four Summer months from a time standpoint but counts as only one issue towards subscriptions.

Cover Photo By: Vesta Copestakes

Editor's Notes

"Unauthorized duplication is a violation of existing laws". Those eight words are creating reams of controversy, what with the matter of video taping currently in court, and the record industry claiming to lose vast sums of money from people who tape friend's records.

Who's right in all this? Who's wrong? Well, neither side is completely without blame.

First things first: If you like a record your friend has, and tape it for your own use, you are violating the law by committing theft - it's like shoplifting, where you enjoy the use of a product but never pay for it. Some people are aware they're doing something incorrect, and justify their actions by saying that "records are so expensive nowadays". That's true - but cars are expensive nowadays as well. Does that mean it's all right to steal a car if you can't afford one?

On the other hand, if you've bought a record and want to make a tape of it for your car, it seems to me (I'm no lawyer, of course) that by buying the record, you've brought the rights to listen to that piece of music as many times as you like. If that involves taping the music for your own private use, I don't see any problem. Besides, many pre-recorded tapes are of dubious quality, so the careful home taper is going to be able to make a higher quality dub.

It would be convenient if the record companies were easily identifiable "good guys" in this matter. But, the fact that a majority of albums (I've heard over 75%) fail to recoup their investment means that the labels are definitely doing something wrong; so far, the answer has been to increase prices (not the best of all possible solutions for a slumping industry). And record companies have certainly been known to rip off the consumer - with warped records, terrible quality vinyl, repackaged greatest hits albums with one previously-unreleased single to snag all the "collectors" in the crowd - and to rip off their own artists as well. Sure, they're crying the blues; home taping is a convenient scapegoat. However, the true root of the problem has more facets than the record industry would care to admit, and involves more than just home taping.

Let me emphasize right now that I don't have any solution to the problem. If all people were idealists who strived for honesty, we wouldn't have any problem...but I don't think the human race is going to change overnight. So, we need some kind of solution which will penalize those who are dishonest, not penalize those who are honest, and make sure that people get paid what's due them for their work. Granted, some rock musician may be making millions and you might think he or she won't miss the extra buck - but what about the 10 years of living on spaghetti and peanut butter sandwiches? Musicians (and yes, record companies too) pay their dues, and deserve to be paid for doing what they do.

Some people think they have a solution, which is to add a surcharge to recording tape and certain types of tape recording equipment. In fact, I recently received a letter from ASCAP urging me to support the bill being proposed in Congress to create this surcharge.

At one point I thought a surcharge would be a good way to make sure that artists didn't get ripped off. But now I'm not so sure.

For example, what about all the people who use cassettes for data storage in computers? Will they have to pay? What about the people who record their own music and send out cassettes to friends - will they have to pay a surcharge on cassettes over which they will record their own music? What about cassettes used in answering or dictation machines...or motivational cassettes, religious cassettes, demo cassettes, and cassette letters?

Admittedly I'm not your average audiophile consumer, but of the last eighty cassettes I've bought, two taped someone else's pre-recorded music, while the other seventy-eight either were used for making copies of my own music, or for making "greatest hits" tapes and car tapes from my record collection. Oh, and before you say "see, he tapes stuff too", let me add that of those two records one was out of print, and the other was impossible to find where I live.

I know there are plenty of people out there who buy cassettes strictly to tape their friend's records, but I also know there are plenty of people out there who have legitimate uses for cassettes which do not involve any kind of dishonesty. The proposed surcharge would penalize these people - the people who have done nothing wrong - as well as those who blithely ignore the copyright laws.

What about a technological fix, such as putting a supersonic tone on the disc which drives the bias oscillator of the tape deck crazy? Possible, perhaps, but then someone will just market a filter to get rid of it. I don't think a technological fix is the answer, because any electronic lock can be picked...just ask the computer software or pay-TV people.

I'm hoping some Polyphony reader can come up with a simple, elegant, workable and above all FAIR solution to the problem of home taping. Something which makes sure that people are justly compensated for the fruits of their work. Something which doesn't penalize the honest user.

The fundamental question goes much deeper, though, and touches on much more than cassettes. We create technologies which can save lives, take lives, make music, or steal music. The machines have no sense of right or wrong - that's up to us. We have the intelligence to make incredible devices; all we need now is the wisdom to use them correctly.

Craig Anderton

Letters

ATTENTION TAPE MAKERS!

In Robert Carlberg's Review column in the Sept/Oct 1981 issue, he restricted his reviews to tapes with national distribution. Because I do not have my material handled by a national distributor, I feel that I have been eliminated from an important phase of our craft: feedback. I shall not put a hex on any of the national distributors, for they provide a necessary function. I, however, am not yet ready for them. I also feel that I am not alone in this respect. For each and every Don Slepian (whom I consider to be a real master) there exists at least a hundred of us who are still trying to "make it". Our talents range from "squeeks and whistles" to some "good complex music". If we must compete with the Slepians of the craft for review space, there is no question who wins.

It may seem at this point that I am suggesting that Mr. Carlberg "clean up his act". This is not true - Mr. Carlberg produces an excellent column. This does, however, bring me to my point. The Pros should be reviewed by the Pro (currently being done by Mr. Carlberg), and the Novices should be reviewed by some other means. With this concept in mind, I'm making the following proposal to Polyphony and to the many Novices who avidly read it.

I would like to provide a review service, with the results published in a simple one or two line format, and including a star rating (similar to many of the movie reviews published in local newspapers). The format would include only the basic information such as name, type of material, address, purchase price, and a one to five star rating. This format would allow the maximum quantity of artists to be covered in the minimum amount of space. This type of format will also make available a large list of names of people who might want to get to know one another.

The review service would also provide direct feedback; if the artist encloses a self-addressed, stamped envelope, I'll send a detailed critique by mail if desired. Also include your phone number so that I have the option to give a phone critique.

Sound interesting? Here's how the service works.

1. We operate on a first in, first out basis.
2. The number of listings published will depend on the space available in Polyphony.
3. Each listing will appear only once.
4. Tapes will only be returned if you include return postage.
5. To receive telephone confirmation and brief critique, you must include your DAYTIME phone number. Calls are restricted to the continental United States.
6. To receive a detailed critique, include a self-addressed stamped envelope.
7. We will not distribute your material. Selling your stuff is your business, not ours.

So...if you're interested, mail your material to:

Marvin Peterson
Dragonbreath Music Systems
34 Winchester Drive
Wakefield, RI 02879

Marvin - This sounds like a good way for those just getting into home recording to hear some valuable feedback on their work. What do you think, readers? If you'd like to see this becomes a regular feature in Polyphony, send your tapes to Marvin. If the response is sufficient, we'll go with it.

SING ALONG WITH...

I was hoping you'd be able to help with my search for a device which would enable me to "play" my solo lines vocally. I am a better singer than instrumentalist, but I'm uncomfortable with "scat" singing. Is there a box which will follow my vocal line, pitch and amplitude wise, but produce the sound of a trumpet or sax (or organ or zither, for that matter)?

My second question concerns the Casio 202 keyboard I own. I've seen articles on modifying Casio's less expensive toys, but nothing on making the 202 more useful. Is there a way to reprogram some of the canned voices? Are they stored on ROMs, which could be replaced with other ROMs, programmed for more interesting, realistic sounds? Although there are 44 different voices, lots of them are either useless or redundant. If we could work out a way to make this inexpensive axe more versatile, we'd have a world beater.

George Western
Clovis, CA

George - What you want is something like the Gentle Electric Pitch to Voltage converter, which provides pitch and amplitude extraction from a signal. Roland also makes a pitch-to-voltage converter as part of their Roland Rack series. While these devices are often used with guitar, you'll find the performance is much less critical with voice, which has a simpler waveform.

In the meantime, try putting your voice through a fuzz to convert it into something with lots of harmonics, follow that with a footpedal so that you can alter the dynamics, and follow the pedal with a flanger, phaser, envelope followed filter, or whatever. You'll probably be limited to kazoo type sounds at first, but eventually, you'll get to the point where you'll have some musically valid effects.

As far as the Casio 202 is concerned...what about it, readers? Any good mods floating around out there? Let us know. ●

Applied Synthesis:

HARPSICHORD ELECTRONIC PIANO PIPE ORGAN BELLS

By: Bill Rhodes

In this installment of applied synthesis, we will approach the electronic keyboard as a pseudo "acoustic" keyboard instrument. As everyone knows, acoustic piano is probably one of the most difficult instruments to fully capture synthetically. Sax, voices, and guitar follow close behind, but piano is a stickler. The various overtone series, harmonics and their anomalies, the shift of waveform according to registration, the velocity and/or tactile sensitivity, and attack (filter and volume) of the instrument are the criteria recognized for synthesis. In a future issue of Polyphony, we will deal with acoustic piano imitation in depth; in this issue, we will deal with the electronic piano (e.g. Fender Rhodes), harpsichord, pipe organ and bell sounds.

Harpsichord. One of the best sounding synthetic imitations is the harpsichord. Some of the various poly-synths out on the market (Korg Polysix B7 program, OBX-SA, Moog Poly, Yamaha electric pianos, Roland electric piano, Prophet 5, and numerous others) have great sounding preset harpsichords. But to duplicate from scratch, let us look at the most important parameters.

- **Registration:** Octave and combination of octave voices is very important to give the depth and punch we need. Combine a 16 foot and 8 foot oscillator in exact (if possible, use sync) tuning to create the proper sound.
- **Waveform:** Harpsichords are rather buzzy and tinny sounding in the upper registration. Select a combination of sawtooth and pulse waveforms in both the 8 foot and 16 foot registrations. Don't bother with triangle waveforms because they don't have the harmonic content of odd footages we need for the harpsichord.
- **Envelope:** Instantaneous attack (VCF and VCA), relatively short decay, full sustain, and slightly reverberant (depending on the virtuosity or velocity of the passage) release is needed for the harpsisound. Punchiness of attack in the VCF is very important; the low end punch should be felt as well as heard if you crank up your amplifier.
- **Filter:** Use a high pass filter and low pass filter in combination if you can. The high pass will give you the sibilance (brightness and "nasality" if you will) and the low pass, the balls. A variety of synthesizers sound too phony because the filter is too reedy or too bassy. Also, since a harpsichord string is plucked with a plectrum (pick) rather than hit with a hammer like a piano, you might want to duplicate the pop and click of the string by adjusting your resonance control.
- **Execution:** Practice your Bach 2 Part Inventions and clavichord pieces (i.e. French Suites and Goldberg Variations). They will help with the articulation you will need in harpsichord performance. Listen to harpsichord players such as Anthony Newman, Rick Wakeman (a different genre), and the masters. Remember, the fingers act like tiny pistons driving down the keys, so the execution of the passages must be articulate and intentional.

Bells. For bell-like keyboard synthesis, try the following:

- **Registration:** Keep oscillators tuned to unison at the same octave.
- **Waveform:** Use the triangle or square with filter settings that are more rounded than harsh. Bells and bell-like instruments have fewer harmonics than brass or string instruments; however, chimes are a complex arrangement of many waveforms, registrations, and ring modulations (the sum and difference of two frequencies).
- **Envelope:** Instant attack, sharp decay, lingering sustain, and long release are basic characteristics of bells. A slight triangle or sine wave modulation at a slow rate is very interesting with this combination - so experiment. Because bells are usually struck by a hard mallet, try to imitate the attack characteristic by adjusting filter and VCA decay to approximate a percussive hit. Be careful not to overuse these sounds (unless you write a bell and orchestra concerto!), but use them effectively in your performance.

Pipe organ. The following approximate the sound of a pipe organ with all stops pulled.

- **Registration:** Combine 2 foot, 4 foot, 8 foot, 16 foot, 32 foot, Big Foot, etc. to obtain the thunderous quality of a Bach Toccata. The footages we use today were derived from the lengths of a pipe organ pipe. The pipe organ used a myriad footage combinations for both manuals as well as the foot pedals.
- **Waveform:** Use square waves in different footage combinations to become more complex "sawtooth" sonorities. Tune each oscillator almost to unison. Remember, the pipes and flues of the organ are never exactly in tune. You may want to demodulate or de-chorus the entire tuning. The old Arp

string unit (Solina) had this feature, as do some Korg instruments (called "ensemble defeat").

● **Envelope:** No attack, no decay, full sustain - but you may want to lengthen the release somewhat for a more reverberant sound.

● **Filter:** Low pass filter set at a very bright, "non-wow" (keep resonance tone low) position. The full pipe sound is very loud and bright.

● **Execution:** Listen to Wakeman's "Six Wives of Henry VIII", Keith Emerson's "The Three Fates" (first ELP album), E. Power Biggs and Virgil Fox's interpretation of Bach's "Toccata and Fugue in D minor", etc. Try to get access to a real pipe organ so that you can experiment with the various stops and footages. Drag your synth into church with you and try to duplicate the sounds (God and preacher willing).

For a mellow, flute like pipe organ sound, use the same envelope as above, change registration to one footage (4' or 8'), lower filter setting to "mellow", change waveform to square or triangle only, lighten up on volume, try to establish a "breathing" type effect (use a slight bit of white noise). Use a sound like a recorder, but put it to polyphonic use.

Rhodes-like sound. We won't attempt to duplicate the pianistic qualities (the mechanical and dynamic nuances) of this instrument exactly, but try a good approximation. Use the envelope of a piano-type instrument, that being instantaneous attack, sharp filter decay, slightly less than full VCA sustain and release depending on the passage you are playing. If you have provisions for a damper control pedal (such as the Korg Trident), use it like a piano's sustain pedal. The waveform for a Rhodes-type sound would be in the bellish square wave category.

A Rhodes has a characteristic bell-like timbre if equalized with a high top end. On some Rhodes set-ups, you can hear the sound of the "hammer" striking the key tine. Vary the filter resonance control on the filter to simulate this pop or click sound.

● **Effects:** That rolling stereo "vibrato" sound we all like and recognize on the Rhodes can be duplicated by the modulation circuit on your synth. With the LFO modulating the VCO with a sine or triangle wave (set at a slow rate and shallow depth) you can to some degree imitate that "tremulated" effect. Please note there is a distinct difference, though, between vibrato and tremolo, so let's clear up any confusion: Vibrato is frequency modulation (FM) or a dynamic change in pitch. With rotating speaker organ cabinets, the full effect created by the spinning rotors and horns results from a change in pitch, not volume. The change in pitch is a Doppler effect created by the sounds being thrown out of the cabinet and moving around the room. A good analogy of the Doppler phenomenon is this: A car approaches as you stand by the side of the road. In order not to hit you and cause bodily injury, the driver blows his horn at you. As he passes you while sustaining his volume with the horn, you will hear a dramatic change in pitch. The ear perceives one pitch or pitches before he passes you and then hears another pitch after he passes you; hence, a modulation of the basic pitch or frequency.

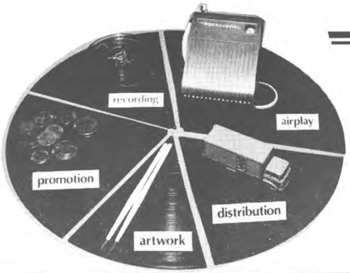
Tremolo is simply an on-off alteration of sound. This is called amplitude (volume) modulation (AM). This can be realized by twisting a volume control back and forth while holding down a note on the keyboard. In the Rhodes, the circuit which creates the shifting of sound between speakers is called the tremolo circuit. Square wave modulation set at the proper depth on the LFO circuit will duplicate the on-off effect.

A neat trick to create an echo effect with varying repeats is using the square wave modulation. Get the desired on-off effect which is in tune with itself (meaning that the LFO is "synched" with the audio oscillator and not changing the pitch) and add some VCA release. The Crumar Orchestrator brass can be echoed by increasing the sustain fader and playing very articulately on the keys. You will experience an interesting echoed brass effect if you mix the strings slightly with the polyphonic instrument.

In this installment, I've touched on the basic keyboard instrument sounds which we all use. According to your synth and sound reinforcement system, certain parameters may have to be varied to a different degree of intensity. Please experiment and realize you have a good foundation to start creating the sounds you need.

Just think, by imitating keyboards with your synth, consider the weight you won't have to carry around: all those pianos, pipes, wind chests, mallets, bars, effects boxes, road cases, candelabra, tuning hammers, pedals, Rhodes piano legs, covers, etc. etc...

Next time around, we cover synthetic choir voices. Stay tuned, everfriends.



Anatomy of a Private Record

by Robert Carlberg

Craig Anderton's comments in the November/December issue regarding independent releases got me thinking. Do all independent releases break even? Is there a network of distributors available? Does the artistic freedom outweigh the financial gamble?

To answer these and other questions, I distributed a questionnaire to a number of artists who had recently released private electronic music albums. Their responses provided verification for most of Craig's comments, with plenty of warm encouragement and a few small surprises.

First and foremost, almost all respondents recommended private releases as a good way to get your name around. Nearly everyone broke even on the deal and a few even made money. However, in the words of Marc Barreca, "If strictly commercial success is looked for, a private release should be mainly for promo". Or again, from Jasun Martz, "The goal of an independent release should be artistic. It seems very few 'indies' make money and thus the project should not be undertaken with (making money) in mind".

The artists who made a profit with their records are the ones who worked hardest to sell them. For independents especially, writing, recording, and making the record is only the beginning. From there on, the artist must promote him or herself tirelessly. The independent market is, according to Gilbert Trythall, "the equivalent of the small specialty press, and the largest problem is reaching people who are interested in what you're doing". Taking out ads in publications which specialize in electronic music, sending promo copies and letters (or better yet, visiting) radio stations who play progressive music, and contacting as many distributors who handle independent releases as possible were all highly recommended. As Leon Lowman put it, "You'd better be a go-getter or you might as well forget it!"

There were some distributors who were frequently mentioned, but the message in the questionnaires was that no record will sell itself. The distributors mainly do that - distribute the record to interested buyers. Getting the potential buyers interested is your job.

The distributors most often mentioned were:

New Music Distribution Service
Division J. C. O. A.
500 Broadway
New York, NY 10012

Aeon Import Records
604 Princeton
Fort Collins, CO 80525

Greenworld Records, Ltd.
23703 Madison St.
Torrance, CA 90505

Square Deal Records
San Luis Obispo, CA

Systematic Record Distribution
Berkeley Industrial Court,
Space 1
729 Heinz Avenue
Berkeley, CA 94710

Most of the problems people had with distributors was cash flow. With the possible exception of the N. M. D. S., all of these distributors are shoestring operations, and they are not in a position to pay in advance for large shipments. But patience and perseverence pay off, and the artists almost always got paid eventually. There were one or two other distributors who reportedly "regularly ripped-off independents", however, so caution is the byword.

The radio stations which play independent releases number in the hundreds. With addresses, the list would take up too much space to print here. Perhaps if there is sufficient interest, Polyphony could devote a couple pages in the near future to listing them. Write Polyphony if you think this information would be useful to you. (Here's my vote! - Editor)

Besides Polyphony, the other publications which have yielded advertising results for private releases are:

Keyboard magazine
20605 Lanzaneo
Cupertino, CA 95014

Eurock magazine
PO Box 4181
Torrance, CA 90510

OP (division of Lost Music Network)
PO Box 2391
Olympia, WA 98507

Advertising outside of the electronic/experimental periodicals is very iffy, according to a couple of respondents, since your chances of reaching the right person do not necessarily go up with larger-circulation magazines - but your advertising costs most certainly do. Cost consciousness is essential to the financial success of an independent release.

As a matter of fact, this is one area where the electronic musician has a huge advantage over almost any other musician. Because a home recording studio is almost a prerequisite for the electronic musician, he or she can afford to spend whatever time is necessary to perfect the recording. The questionnaire answers ranged from 60 hours to over a year in the studio, which would be extremely expensive under commercial studio rates. Many commercial recording studios offer packaged deals where a vocalist can come in and sing over pre-recorded backing tracks for a set fee. Luckily, electronic music has not been limited by such cost restrictions, and the result is that most independent releases rival, if not improve on, the major studio productions.

Certainly, the freedom to "get it right" is important to independents. "The best thing about doing it yourself", says our own Craig Anderton, "is that you make the music you want to make. You never have the situation where you have to compromise, the album flops, and you wonder, '...if only I'd insisted on doing it my way, maybe it would have sold.' Conversely, if the album does well, you can pat yourself on the back without any doubts." Self-expression should be your motivation, says Don Slepian, "and you will allow no person or obstacle to stand between you and your self-expression".

So let's say you've got your master tape all finished - what's next? Don Slepian offers some very sound advice: "Gain access to a high quality cassette recorder and make 20 copies of your prospective master tape, then sell all 20 copies. If you can't sell all 20 copies (and not to dear friends), then you should re-examine your master tape before pressing it into a record". Listen to the advice offered to you by the buyers of your tapes (and everyone's a critic when you ask them). Remember, a total stranger may not appreciate your rendition of Klaus Schulze's "Moondawn" if he or she has never heard "Moondawn". The stranger also might not wish to sit through 22 minutes of intense jamming if you had to be there to appreciate it. Make the record you want to make, but bear in mind that you still have to sell it.

Some advice from respondents about the cover: If you're not an artist (or strangers think you're not), find someone who is to help you out. The cover is the first impression anyone gets of your album, and if you make it as far as the record store, the only impression. Don Slepian, whose record is very professional except for a cover which is a bit silly, admits "Next time I would spend more time and care in packaging. Next time I wouldn't be so naive."

Alright, everyone loves the tape. How do you get it on to vinyl? The first step is to choose a record presser. There are numerous choices, but three names kept coming up in the questionnaires. All three are highly regarded by those who used them, so geographical nearness is probably the determining factor:

QCA Custom Pressing, Inc.
2832 Spring Grove Avenue
Cincinnati, OH 45225

Wakefield Manufacturing, Inc.
PO Box 6037
Phoenix, AZ 85005

Location Recording Service, Inc.
2201 West Burbank Blvd.
Burbank, CA 91506

Beware of "bargain" deals - they usually end up costing more than expected. These pressing plants may charge a bit more initially, but everyone agrees they are still the least expensive overall.

The next step is to have your tape mastered, which means a professional, usually associated with the pressing plant, sets the dynamic range, the timings, and the equalization so that your music will translate on to vinyl in the best possible way. This is no easy feat, and no compromises should be allowed. The masterer will provide you with an acetate, a wax-like pressing for you to evaluate. Three or four tries before you're both satisfied is not unusual, since the mastering technician probably has never heard music like yours before, and you've never mastered a record before. But if you choose carefully, the technician will be a professional with good advice and an honest desire to please your ear.

Next you have a test pressing made. This is your first opportunity to hear yourself on a "real record", a thrill that never seems to wear off. Assuming the final check (changes here cause a new mastering charge!) comes out okay, you're ready to press. You're basically checking the quality of the vinyl, the trackability of the lathing, and the final check of the EQ, stereo balance, and dynamics. Don't be afraid to make changes if you aren't happy - you'll have to live with it for a long time to come.

From there, you simply sign off the test pressings and you're underway. You will shortly have a basement full of assets to get rid of. Since all of this pressing business is cash-in-advance (or half-on-delivery), the return on your investment will be staring you in the face every time you do laundry (or whatever it is you do in the basement). If that isn't motivation enough to get out and sell a few records, nothing is.

Does the artistic freedom outweigh the artistic gamble? I can only report that every questionnaire says yes on this point. Perhaps Leon Lowman summed it up best: "Putting out your own record is not for the lazy, not for the timid - you gotta work. But if you don't try it how will you ever find out where you stand in the world of music? Go for it!"

INTERVIEW:

By: Don Schwartz

DON SLEPIAN

Don Slepian is a familiar name to many Polyphony readers. His self-produced, imaginative cassettes and records have earned excellent reviews from those critics interested in non-mainstream music.

The following interview is excerpted from a lengthier interview conducted by Don Schwartz, and focusses specifically on Don Slepian's compositional techniques. But don't think you have to be a professional musician to get something out of this interview - Don has something to say to the frustrated music maker and amateur as well.

By the way, "D" refers to Don Schwartz, while "DS" refers to Don Slepian.

D: I want to begin the interview by saying that I love your music. Can you talk a bit about your composing process, because I hear a very thoughtful consideration in your work. The many layers and levels sound as if you have spent hours upon hours choosing how to integrate these sounds you make.

DS: One representative piece, the Sea of Bliss, has three levels to it and is made in three layers. Most of my work is done on a four-channel tape recorder, or less. I start out first from a technical aspect. I get a repertoire of sounds which I feel are going to work for me, and once I've found those sounds, then I think of an idea or concept.

Once I have both the technical end and the concept, then I enter an area of improvisation, in which, say, I'll have a clock, and a beginning time, and an ending time, and I'll adjust the lights and outside surroundings to be conducive to a musical state of consciousness, because I really do not sit and plan exactly what is going to happen on tape. Improvisation has been the way I've worked for 20 years...it has always been my path.

I'll entirely improvise the first layer, and do far more than I need. Usually another day or two will pass, and then I will come back and do extensive tape editing and tape manipulation. I'll have, say, an hour of music which has been improvised, and I will take that and cut it down into maybe half an hour of the best parts, in a way that they flow imperceptibly, one into another. The most important tool in my whole craft has been the razor blade and tape splicer. I have put a lot of time and care into developing the art of splicing, and of editing.

That's the editing process for me; next is to learn the improvisation I've done on the first layer to prepare for recording the 2nd and 3rd layers.

These subsequent layers respond to what is already on tape. I'm still free; in other words, I still simply give myself the sonic materials, and create in response to what I hear. However, on the second and third layers I will stop myself, go back, and maintain a certain standard of exactly what I want. It's a different sort of process to do overdubs than it is to create the original source of the music. The first track is the foundation, and that only gets edited and put together - if it's worthy, and a firm foundation, I'll proceed to the next layers.

"Improvization.....it has always been my path."

D: You said earlier that you get a "concept". Can you talk about that?

DS: 360 degrees of freedom is overwhelming in music, and you need not truly begin to find freedom until you put yourself under extremely narrow constraints - particularly in the composition process. You want to look at one small area, and develop within that area.

For example, there are two pieces of mine - "Sea of Bliss" and "Awakening" - in which I look at developing long notes, far longer notes than are common in Western music. I was coming from the eight second melody notes used in gamelan music in the Javanese tradition, since I was a member of the Gamelan for a year and was influenced by that. There, in the slow gamelan, the saron melody can be as long as eight seconds for each note. I was working with a synthetic tone, which can swell up like a violin, and then have a very long sustain, with one note blending into the other. So in that case, the concept was an unusual type of note, an unusual sound.

"The most important tool in my whole craft has been the razor blade and tape splicer."

A concept can be first of all a technical concept, such as a certain type of permutation or different computer algorithm, or it can be an aesthetic concept, such as working with tape-transformed acoustical instruments and having a certain thing to say. No matter what the concept really is, just the idea of narrowing your scope is important, so that what you are doing has some focus.

D: What kinds of music would you like to explore? Where do you see yourself heading?

DS: Well, it's not so much types of music, it's effects or emotions. I'm interested in working with some more challenging types of feelings than simply that of blissfulness and serenity. There are other emotions out there which are equally powerful, such as sadness, tragedy, elation. I think of the baroque composers in the doctrine of the twenty affects, back a few hundred years ago; they had catalogued certain emotions, and the composers were trained by being able to actually produce those emotions in people, to share them, to elicit back reactions from an audience without lyric, with music only. That's what I'm looking for in a direction - to take abstract sound, and find those elements which will touch upon the feelings of people, and in doing that, I would produce an emotional catharsis, or some uplifting of the spirit.

"....I have had to become quite adept at modifying and expanding synthesizers and tape recording equipment."

D: What do you do in addition to recording and releasing tapes?

DS: At this point, I haven't been performing for some time, although I'd like to get back into that in a very specialized way. I certainly do work. My two years in the music business of producing cassettes and the like have yet to bring in a total of one month's rent. As far as income, it actually has been quite a loss. So, I teach one day a week at Fairleigh Dickinson University, which is a private university close to me. I teach electronic music and a lot of private piano instruction...teaching people Bruce Springsteen songs, Tom Petty, whatever they want to learn. I enjoy teaching, learning, and playing the instrument; one of my strongest backgrounds is in classical piano.

D: You're classically trained?

DS: Yes. I also do a great deal of custom electronic work for a clientele that often includes New York studio musicians. I'm mostly involved in modifying commercially devices or building devices which are not available commercially. That's an interesting craft. I've always been involved with electronics and I've always had this duality of electronics and music. I have had both interests, and of course I have had to do a great deal of custom work on my own equipment in order to be able to produce the sounds I want. On an extremely limited budget, I have had to become quite adept at modifying and expanding synthesizers and tape recording equipment.

D: So what you're doing with teaching and working on instruments feeds into your compositional recording work.

DS: I'm not so sure about the teaching. I'm doing it now, but if things pick up, then I will probably take some time away from teaching. Electronics and music go together quite well, and I just have to balance those sides of what I do. I do so much electronics I get frustrated, and I want to be able to put down my soldering iron and do the music.

"Electronics and music go together quite well .... I just have to balance what I do .... I want to be able to put down my soldering iron and do the music."

D: Do you have any interest in video composition?

DS: I'm interested in analog video synthesis, which - at its most mundane level - looks somewhat like electronic wallpaper. Whereas I do music which evokes emotion and yet is without language, I am interested in doing television visual images, on a more abstract level, without the concreteness of consistently recognizable objects, without keying people into saying "this is a chair", or "this is a human face". And yet, visually, I have the same sort of compositional goals as I do with the music. In other words, I am looking for communication to link directly with the feelings, emotions, and thoughts of people. I want to do that with as little cultural definition as possible.

I'm very interested in reaching out to the people of South America, Africa, China...an enormous quantity of people have been completely ignored by, say, certainly the art I'm in, and a great deal of Western music and art. There are huge areas of the planet which have been ignored because it's difficult, if not impossible, to commercially exploit them. One of the keys to reaching these people is to move away from the constrictions of the English language, the straight-jacket of my language, my thinking, of my conceptualization.

I see this idea of music without words, and television without concrete images, as a sort of ideal symbiosis - they can feed upon each other. Well, maybe that's not saying it quite right, let's just say it's an ideal symbiosis of two forms which will support each other.

"I record as much as possible at home"

D: I'm interested in the recording aspect of your music, particularly your recording at home.

DS: I record as much as possible at home. There are times when I will record at home and then take these tracks to a recording studio, transfer them to a multi-track, and overdub additional parts. Or there are times when I will record things on location and then bring them back home and transfer them to what I have at home. For me, a large part of it has been getting the most I can out of four-channel, and doing the best job I can with hi-fi and semi-pro equipment. I consider myself very fortunate to have a four-channel, and to have the tools that I do use, and I see over and over again that you can make a great deal of use out of what is at hand. Before I had a four-channel I had a two-channel, and I was able to make a great deal of music with that.

With the music that I'm doing right now, it's becoming more and more necessary to get more of the tools that are in the mainstream of recording art, such as an 8 or 16 track tape recorder, a synchronizer between that and my video tape recorder, and other such things.

As far as recording, I don't have any unique philosophy or even particularly unique style. I guess the only thing that I do now is use very little equalization. Whenever possible, I put exactly the sound I want on the master tape, and I believe in passing through the minimum amount of electronics in order to get the sound. And I have no regard for standards, other than results - if I hear good musical results from tape saturation or non-linear distortion, I'll use it. I will not fight the equipment in my studio, I work harmoniously with it. It's virtually an unconscious process because when I do create my music, I am not in a normal state of consciousness. I'll often do an hour of taping, in isolation, late at night; then I'll listen back to see what I've done.

A large part of my work is preparing myself and the tools of my craft to be the proper recipient for inspiration. The technical skills that I polish, my work on all my instruments, my electronic work, everything that I do would be useless, would be totally meaningless if, at that moment when the red light is on the tape recorder and the reels are spinning, the inspiration is not there. All that I do is preparation for that moment...in that respect, although it's a cliche, I really do think of myself as more of a channel for the music than anything else.

Of course, you also have to judge that inspiration; a composer has to be his or her most stringent critic. Striking a balance between the critic and composer impulses is the key: years went by without me producing a thing because the critic impulse was too strong - that is, you edit and edit until you end up with about an inch of tape! There has to be a certain letting go, a certain willingness to be wrong. I hear the mistakes and errors in my music...I hear the things that aren't right. And, it's essential, as part of that balance, to look overall at a piece of music and say, "the music in it is here...is worthy...considered as a <u>whole</u>. It is okay with me to let this out".

There's always the tension between excitation and inhibition. For instance, in playing classical piano, much of classical performance is balanced between these two forces - if there is no excitation, the pianist sits there and looks at the keyboard and plays nothing. If there is no inhibition, the pianist starts the piece at double tempo, plays like a maniac, and often ends up on the floor in a tangle! In music composition, you must first have the inspiration and then, from the inspiration, you have the craft, the tools, the technique, the training, the aesthetic maturity, to be able to look over your inspiration and create something focussed.

D: Here's a question...how can somebody find your music?

DS: Well, one way is to write me at PO Box 836, Edison, NJ 08818; Don and Judy Records - our company - will always carry and service our music to at least some small degree. The other sources are Ethan Edgecombe, Fortuna Records, PO Box 1116, Novato, CA 94947, which services the San Francisco Bay Area; Plumeria Productions Hawaii, PO Box 54, Kailua, HI 96734; and Eurock Mailorder, PO Box 4181, Torrance, CA 90510, who sells the music all around the world.

"I see too many people playing records and not enough people playing instruments."

D: Is there anything that you want to say in the way of a conclusion?

DS: This may be contradictory, but I enjoy live music far more than I enjoy anything coming from a loudspeaker. I seek out concerts of live musicians. I'm concerned about recorded music taking over for some people, or for many people, the role that live music has played. I believe that everybody has musical ability, and that we are all musicians. And, I see a tremendous decline in the number of amateur musicians - the number of people who will pick up an instrument - compared to the number of people who will put a cassette in the machine or a record on the record player. I see too many people playing records and not enough people playing instruments. I think if one were to look for music, your first choice would be to pick up an instrument and play it yourself; and, if that weren't right for you, your second choice would be to find a musician next door or down the street, and listen to him or her; the third choice would be loudspeaker music.

D: That's certainly provocative, considering that some of the greatest minds of the century are at work creating the best possible reproduction systems to reproduce live music as accurately as possible.

DS: That's true. The loudspeaker music is excellent, I don't doubt that...I'm just saying there's a spiritual aspect of music that is lost without the presence of the musician, and that quality is lost when music is reproduced rather than created at the moment. Because, in the presence of the creation of music, it is a different experience than to be present at the recreation of music. I hope people wouldn't hear my music and say "I could never do that", and feel discouraged from playing the music themselves.

D: It sounds as if you would like to inspire them to play from your music.

DS: I want them to pick up the instrument. I would like to see a resurgence of amateur musicians. I would like to see more people play piano, guitar, flute, whatever instrument is right for them.

D: I agree totally. I think a lot of the problems of our times come from people being passive recipients of life rather than active participants and creators.

DS: You put your finger on it! Everyone has a need for self-expression, that's part of health. And what I see so little of is people picking up the paint brush or the pencil to write. I want to encourage people to do what I do...I want to say that, in certain ways, who I am and what I do is nothing special, that everybody out there can be a channel if they are open to it. Everyone has a creative force they can use, and everyone can create. And, I'd just like to see people find a medium for creation and express themselves at whatever level. That's all the conclusion I have.

The Phase Checker A Review

By: Craig Anderton

Every now and then you run across a useful piece of equipment that ingeniously solves a particular problem. The Phase Checker, from Sounder Electronics (21 Madrona St., Mill Valley, CA 94941) is just such a piece of equipment.

Why check phase? When signals are out-of-phase, they cancel (or portions of them cancel). Therefore, when recording it's important to be able to check the phase of microphones, speakers, the elements within a speaker, special effects, connectors, transformers, and the like.

You might think it's simple to make sure speakers are in phase; after all, they all have a little label that specifies which lead is positive and which is negative...but those labels can be wrong. What really turned me on to the Phase Checker was an incident in my own studio, where I couldn't figure out why I had such poor bass response out of my spiffy new speakers, while my headphones sounded just great. I checked and rechecked the wiring, and it all looked OK. I then checked the speakers with the Phase Checker, and wouldn't you know it, the speaker was wired out-of-phase at the factory! I was surprised, but according to the engineers at Sounder, this kind of thing happens all the time.

How it works. The Phase Checker includes two units, the Pulse Generator (transmitter) and Phase Polarity Detector (receiver). The transmitter puts out a positive-going reference pulse, either at mic level (0 to 150 mV) or line level (0 to 17V into 4 Ohms). It also comes with an internal speaker to simplify microphone testing. The transmitter, like the receiver, is powered by a 9V battery and comes with cannon, phone, and phono jacks.

The receiver has an internal electret microphone to test acoustic transducers (speakers etc.), or you can feed a 10 mV to 10V signal into the mic input, or a 1V to 100V signal into the line input.

You use the combination by feeding the transmitter into the device under test, and feeding the output from the device into the receiver. A light on the pulse generator indicates when a pulse is going into the system. When the units are in phase, a bright green light on the phase detector flashes at the same rate as the light on the pulse generator. When they're out of phase, a bright red light on the phase detector flashes instead of the green one. For limited testing purposes, Sounder also makes a pulse generating cassette that replaces the pulse generator when testing cassette deck/hi-fi/speaker combinations.

Construction. The Phase Checker is built extremely well, and is compact and lightweight (3 lbs.). About the only improvement I could think of would be recessing the controls, jacks, and lights. Still, it is clearly built to "rock and roll specs", and would probably survive life on the road better than the humans running it.

Overall Evaluation. The Phase Checker has one large drawback: the two units list for about $450. Still, even at that price it's well worth the expense to sound companies, speaker manufacturers, broadcasters, recording studios, and so on. It is unfortunate, though, that the Phase Checker's price is prohibitive for people with home studios or other small-scale setups. Hopefully, Sounder will be able to make another, low-cost version for the low end market before too long.

In any event, the Phase Checker does what it says it will do and performs an important test routine easily and painlessly; probably the last word is that it ain't cheap, but you certainly get what you pay for. #

Robert Carlberg's

re-view

Mnemonists **Horde** (DYS 03). The Mnemonists, a group of eight audio/visual artists with up to eight guests, tend to cause strong reactions. Heavily cultural yet hardly traditional, their music is a heady goulash of five centuries of musical history with modern tape collaging and electronic music techniques. It's either a new form of music or not music at all, depending on your viewpoint. Modern and Medieval wind instruments, cello, doublebass, electric guitar and tape manipulations interweave in noisy, brash collages that would take a PhD dissertation to unravel. See also the Jan/Feb review of their 2nd album.

Movie Viewers **Rescue 1** (Plexus 218709, EP). Packed with the record are a pair of red/blue 3-D glasses with which to view the red and blue type on the jacket, half in each eye. Next, the four songs on the EP are presented in different mixes on each side, in parallel grooves of two songs each. Get it? The music is rock using synthesizers, rhythm box, guitar and vocals, and is better than we have a right to expect considering the time spent on the album concept.

Mecano **Entitled** (Plexus 709218). A Dutch group which went defunct even before the record was released. They played polysyllabic rock (bass, drums, guitar and vocals) with a vaguely despondent, Teutonic edge.

Jon Hassell **Dream Theory in Malaya** (Editions E.G. 114). Hassell's fourth solo album is similar to the other three: strange, moaning flute-like trumpet over intense African/South American percussion. Some great synthesizers and processed backgrounds also appear.

Clive Robertson **Popular Songs** (Voicepondence 006). Clive would probably like his songs to become "popular", but they are too political, too honest, and too emotionally charged for that. Synthesizers provide the foundation.

Weather Report **Weather Report** (CBS 37616). Weather Report albums (this is their twelfth) always take a while to sink in. Zawinul (not Joe or Josef, perhaps he got tired of people misspelling it) provides the motive force, writing all but one tune and almost all of the evocative musical moments on keyboard synthesizers. I'll probably grow to love this one too.

Charles Brown **Bull in a China Cabinet** (Fracture 101). Electronic-orchestra rock complete with sweeping chords, growling basslines and exploding starbursts. If you like Larry Fast you'll love Charles Brown.

Various **Music from the 21st Century** (GNP Crescendo 2146). Basically an attempt to break some more "normal" electronic music into the charts, this album features outer space themes and son-of-Switched-on Bach voicings.

Mychael Danna **The Electronic Orchestra** (Frederick Harris 803). More classical music performed on synthesizer.

Jasun Martz **The Pillory** (Neoteric 61853). A huge orchestral crescendo that builds and breaks seven times. Japanese-quality second pressing.

Bernard Szajner **Superficial Music** (IRC 006). Half-speed and backwards edits from Szajner's first LP, "Visions of Dune". Manifests an Eno-esque view of previously released material as simply concealed Musique Concrete.

Lee Negin **Two Sides** (Passing Phase A2Z-111, EP) **Wired for Sound/Nothing Goes Right** (Passing Phase A2Z-121, single). Lee Nagin plays synthesizers and sings songs about the failure to communicate. Most of it is pretty straight ahead rock which can be favorably compared to, say, John Foxx. For the new single he cut his hair and a lot of the slack.

CONTINUED ON PAGE 20

PRACTICAL CIRCUITRY

By: THOMAS HENRY

What, another filter? Yes, and there's a good reason for it. When it came time to design a filter for my synthesizer, I was struck by how many exotic designs already existed - phase filters, polygonal filters, multimode filters, and filters which did everything but clean the sink. On the other end of the scale there were noisy filters, trashy filters, worthless filters, and filters which hardly filtered at all! What I wanted was a good quality, middle of the road filter - and this design is the result.

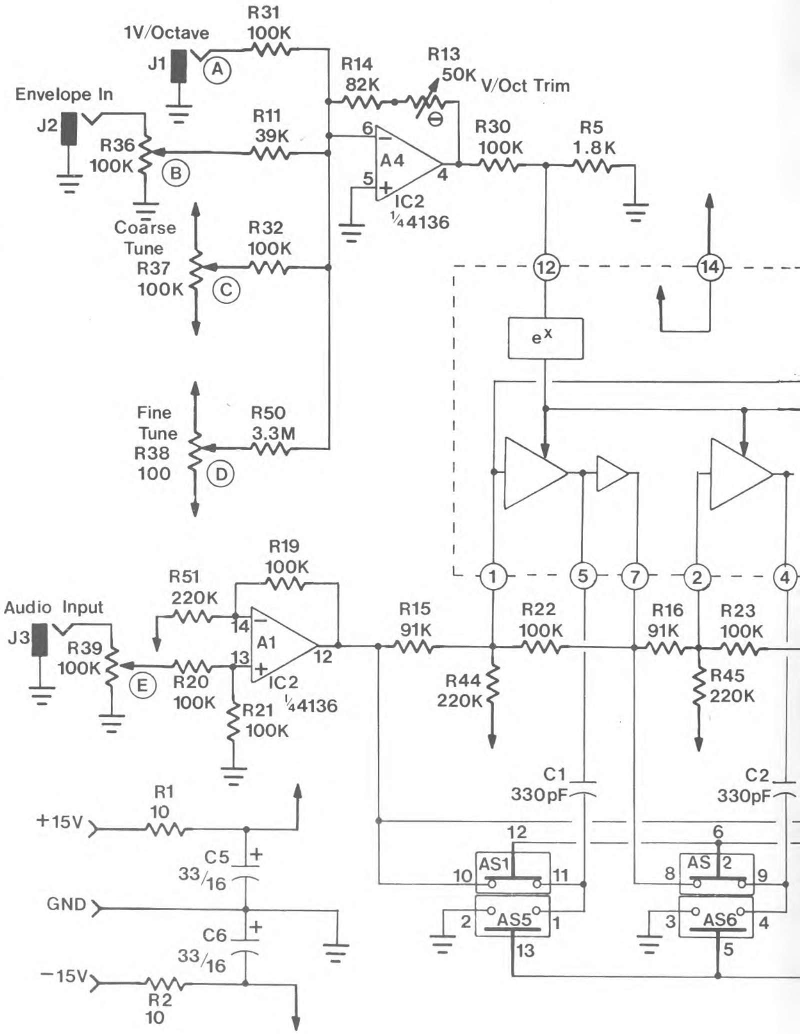
Now, don't get me wrong; I like fancy filters too. But I don't want to play synthesizer in the poor house. The filter I came up with is very clean, but won't cost you an arm and a leg. The good quality at a low price comes from using the CEM-3320 integrated circuit made by Curtis Electromusic. I think you'll find the unit is very versatile and yet easy to understand.

How it works. The name PAL came about because depending on how you adjust the controls, you can have a Phase shifter, All pass filter, or Low pass filter. In addition to the usual voltage controlled options, this unit also features voltage controlled resonance.

Referring to the schematic, the heart of the filter is the CEM-3320, which does all the hard work for us. In fact, all we really have to do is supply it with the correct voltages. For example, look at the resonance controls going to pin 9...two 100k resistors is all it takes! R42 is an initial resonance control, and J5 is the voltage controlled input. R28 and R29 convert the input voltages to currents which are then summed into pin 9 of the CEM-3320. (For the purists in the crowd, pin 9 is not quite at virtual ground, but is close enough to pull the summed current trick. Since resonance doesn't require the precision that frequency does, a two resistor summer works quite well.)

J3 is the audio input. A1 is set up in such a way that the audio signal is shifted up to about half of the positive supply voltage, which is done to interface more easily with the CMOS circuits we'll encounter later. Note that the output of A1 feeds both the filter proper and also R40, the output blend control. And as long as we're talking about the blend control, let's note that the other side of R40 is at a half supply bias as well. This guarantees that there won't be any weird DC level shifts as you pan the pot. At the same time, notice that this lets us preserve DC coupling from the output of the filter.

The wiper of the blend control feeds A2, which is set up to undo the biasing mentioned above. R43 shifts the DC back down to 0V and R35, the output offset trim, lets us trim this precisely. Many of you are probably wondering why I went to all this trouble, first shifting the bias up and then back down again. Well, besides the

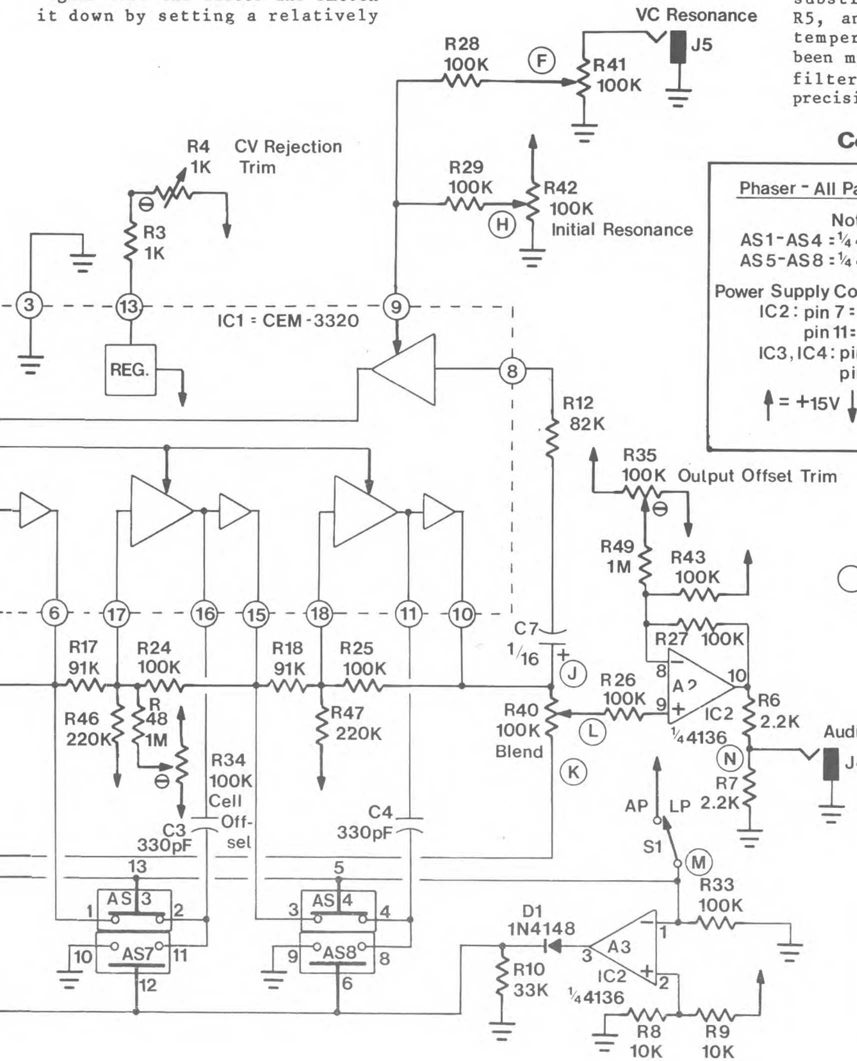


CMOS considerations that we'll be touching on shortly, the net result of all this is that we get to keep the filter DC coupled from input to output. In general, I like all of my synthesizer circuits to be DC coupled if at all possible, since this means I never have to differentiate between audio or control signals. Granted you'll probably use the filter for audio processing most of the time, but there may also be times when you want to put a jagged control signal into the filter and smooth it down by setting a relatively low corner frequency.

A4 and its associated components provides voltage control of the corner frequency. R37 and R38 provide coarse and fine tuning respectively. J2 is an envelope input with attenuator, and provides a gain of about three. The gain is there so that you can sweep the filter over its entire range with a standard 5V output envelope generator. J1 is the one Volt per octave input, and would typically be driven by the keyboard.

R13 is the Volts per octave trimmer. R30 and R5 drop the voltage to an appropriate level before applying it to pin 12 of the CEM-3320. Note that the chip has been temperature compensated for all second order effects, however a first order, or linear, temperature dependence still exists. If you feel that this is going to be a problem, you may substitute a 1.8k thermistor for R5, and this will null out all temperature dependence. It has been my experience, however, that filters generally don't need the precision of something like a VCO.

Continued on page 30



Phaser - All Pass - Low Pass Filter

Notes

AS1-AS4 = ¼ 4066 = IC 3

AS5-AS8 = ¼ 4066 = IC 4

Power Supply Connections Assumed:

IC2: pin 7 = −V
pin 11 = +V

IC3, IC4: pin 7 = GND
pin 14 = +V

↑ = +15V ↓ = −15V

◯ [Circle] Keys the schematic to circuit board

THE PAL FILTER

The SYN-BOW

A REVIEW

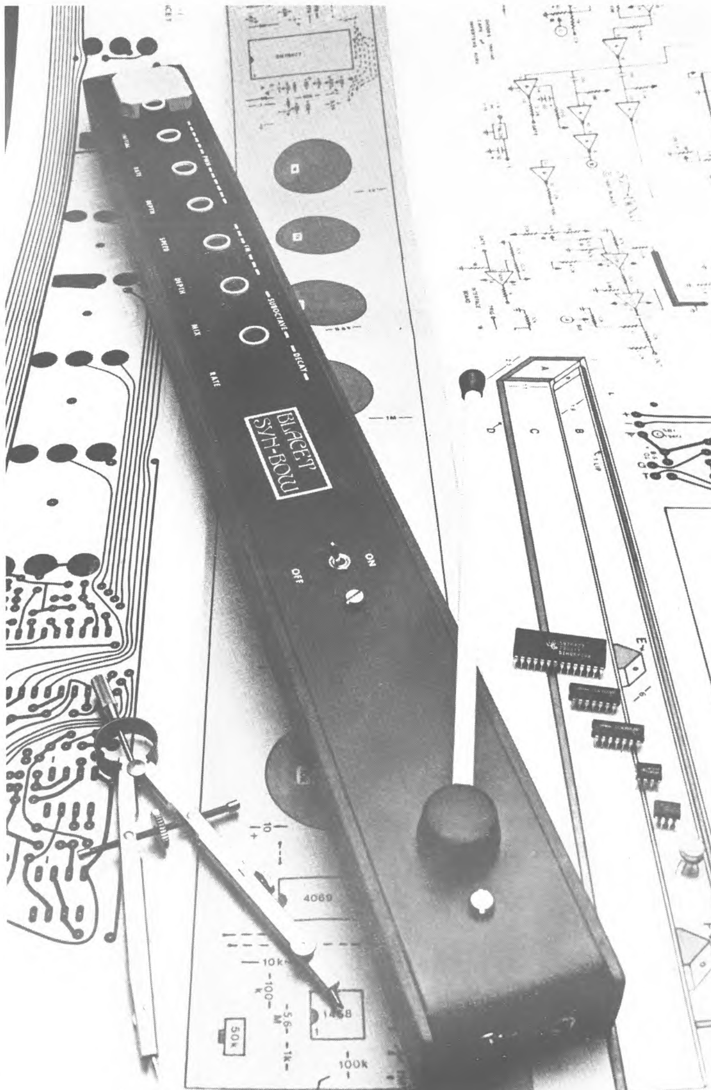
By: Ron Minemier

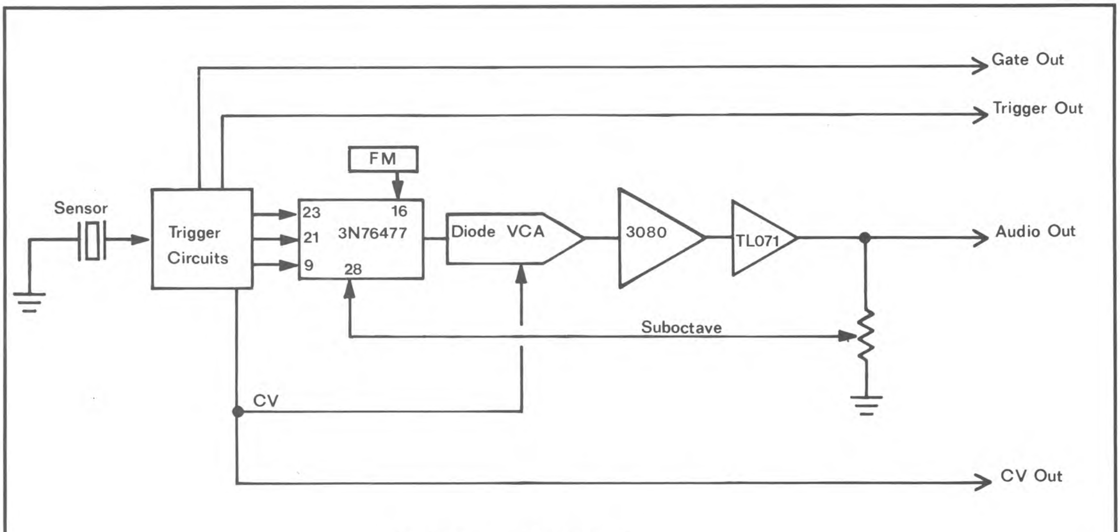
Unlike other portable synths, Blacet Music Research's "Syn-Bow" offers a radical departure in design by eliminating the keyboard. Just carrying the unit around will attract a great deal of attention with the inevitable first question, "what's that"?

General description. The Syn-Bow comes with a redwood case, which must be epoxied together. A nicely stenciled top plate fits on to the case. All controls (save the power switch) mount on the PC board, which in turn mounts to the top panel. This eliminates much point to point wiring and gives easy access to the battery compartment and the component side of the circuit board. The total size is 27 long x 2" wide x 1" deep; weight with batteries is a mere 1.5 lbs. Personally, I like the redwood case as it can be finished to taste for a very appealing look, but for heavy-handed types the overall mechanical strength will be intolerant of abuse.

The parts (85 in all) are neatly packaged and of good quality. All pots come with two hex nuts so that you can tighten the pot from the back of the panel, thus preventing scratching or marring of the front panel. The knobs have two Allen type set screws, and the 7 pin Amphenol interface connector jack is keyed to the panel end not only to achieve proper orientation, but also to insure as torque is applied to the locking collar the jack will not rotate in its mounting hole.

The instruction booklet starts out by saying the kit is not difficult to assemble, and that patience is the key. I would underline that section, for while the board and traces are of high quality, the board is not legended and most of the components mount in rather dense areas. Also some of the smaller capacitors are not marked in any conventional sense





Block Diagram of the Syn-Bow

and may confuse the beginner as to what value they really are. Functional assembly should take about two light evenings of work.

What it does. At the right end of the case is the frequency bow. This is a pot whose knob has been tapped into which a 7" plastic bar is glued. Moving the bar in a circular fashion selects pitch. The arc described is about 40", thus lending good control even thought the scale is compressed towards the high end. Total frequency range is 2 to 3 octaves (long time subscribers to Electronotes may remember Craig Anderton's dial controller which describes this basic technique).

Moving to the left down the case you find the power switch, then comes the four major sonic modifier groups. A Decay pot sets the amplitude decay rate, and a Suboctave Mix pot adds a fattening suboctave. Next is the FM section where a balanced triangle wave is applied to the VCO through Rate and Depth controls for smooth vibrato. There is also a pulse width section with Sweep Rate, Depth, and Initial Width controls, which greatly affect the timbre of the instrument. A foam octagon pad at the far left of the unit covers a piezoceramic sensor which allows the performer to initiate the note, as well as control timbre and amplitude. The unit can be played on a flat surface or held somewhat like a guitar. There are no strap pegs but these could be added easily, and even without a strap the unit can be hand-held quite nicely.

Circuit description. The Syn-Bow's sound source is the ubiquitous Texas Instruments SN76477. While the chip was never intended for serious instrumental use, with subtle and clever manipulation of relatively simply outboard chips this design achieves sounds which belie the limitations of the 76477.

The block diagram (fig. 1) is straight forward. Pushing on the sensor drives trigger circuits which provide several different pulses to start the on-chip AD amplitude generator, PW sweep sync (sweep starts at the same point with each new trigger), and set the one-shot on the 76477. Trigger and gate pulses are also brought out so that you can control outboard equipment. The sensor voltage is also available as a control voltage out, although its principle use is to control the box marked VCA. The diode VCA and 3080 are not mislabelled - the VCA is a "diode VCA". With the aid of an RC circuit the ON resistance of the diodes is changed via the sensor voltage, thus varying amounts of signal are presented to a fixed gain 3080. You may recall seeing a similar trick used in Craig's VC distortion module for his AMS-100 system. This idea is a form of timbre enhancement and takes the place of conventional filtering; this scheme accounts for the organic quality of the Syn-Bow and is what Mr. Blacet has trademarked as a "natural filter patch". As in many acoustic instruments, playing harder yields a brighter tone.

I cannot emphasize enough the cleverness with which this sometimes toy sounding chip is being used. For example, the tap off the final buffer feeds a portion of the audio signal back to a nominally digitally programmed pin on the TI chip. This in essence multiplexes the normal square wave output with a suboctave of itself.

The Syn-Bow may be powered from two 9V batteries or a ±15V supply (even if the batteries are still in place). A small external PC board, meant to be housed in your custom ±15V enclosure, brings the gate and trigger pulses up to higher levels (nominal output levels are +5V for the gate, trigger pulses, and CV out). Audio output is approximately 2V p-p max; there is no volume control.

The sound. How does it sound? In one word - good! The range of the timbre produced by the initial pulse width control varies the base wave from thin nasal to thick and gritty. Adding in the suboctave yields a more expansive grit. One limitation of the way the suboctave is derived is that its contribution to the sound will die away before the main wave at longer decay settings. This normally does not detract from the instrument. Also, at narrow pulse width settings the suboctave circuit won't trigger properly and does more harm than good by nearly muting all the signal. Personally I don't see this as a problem as I use the thin pulse settings for their unique voicing qualities, and wouldn't want to muddy it up with a suboctave anyway. Sweeping pulse width patches have always appealed to me and in a circuit without any conventional filtering blocks this feature is necessary and well done. The LFO that controls PW sweep is the TI chip's SLF triangle wave. The sync feature gives predictable effects, but for those who may not want this feature an SPST switch could be added to defeat this function. The instructions mention that there is some interaction between the PW depth and initial controls; this is not obvious, however at long decay times and slow to medium sweep rates you will notice that not only is the pulse width being modulated, but also amplitude. I don't mind this greatly as I like the effect, but if you patch the Syn-Bow into a Pro-One or something like PAIA's envelope trigger + gate extractor this amplitude modulation is going to produce multiple triggers which may or may not be desired.

The FM section is perhaps the weakest link. The modulation speed only hits the vibrato range (7 to 10 Hz or so), and the max depth is so shallow that any PW sweep or suboctave mix at the same time will tend to mask the vibrato completely. (A simple SPST switch connect the SLF to pin 22, thus modulating the VCO directly and giving your typical phaser gun sweep sound. While this sound isn't really in keeping with the Syn-Bow's original character, it is available for the price of a switch.)

Playing the Syn-Bow. It is fairly easy to trigger the sensor consistently; there's a trim adjustment for sensitivity. Changes in the force applied do yield a noticeable change in tone color and amplitude. One thing that's awkward is that you have to accept the Syn-Bow as a triggered source; there is no sustain other than cranking out the Decay control and this is true for the action of the sensor parameters as well.

Playing a melodic line, because of the bow rotation's non-discrete pitch nature and the triggered source effects, makes for an interesting challenge. As in other intuitive instruments (trombone, violin, etc.) pitch acuity will definitely help. But this shouldn't scare anybody off as you can pick up the device and get effects right away - doing more formal numbers comes with experience. I think that the bow idea is basically Jeffersonian in concept, which relates well to Craig's editorial statements about portable electronic music; I mean anybody can have a good go at the Syn-Bow, but if you wish to really sing with the thing you'll have to accept significant responsibility as well.

Overall, I'd have to give a lot of credit to John Blacet for creating a unique, portable instrument which is accessible to anyone wanting to express themselves musically. The price (less shipping) of $124 is pretty accessible as well. The instrument has real character and is fun to play, plus because the SN76477 is being used, there are a number of mods which can give not only the Syn-Bow sound but also some special effects. Add to that the easy interfacing to other synthesis gear and processors, and you get a good value for your money.

re-view

CONTINUED FROM PAGE 15

Port Said **Indian Ocean, Voyage 2/Trail of the Sphinx** (Port Said 001, single). New York duo Stefan Tischler and Keith K. Walsh play Egyptian-seasoned synthesizers somewhat reminiscent of Agitation Free's "Malesch". It's an engrossing blend of large synthesizers with small (Casio), electronic percussion with maracas, and ancient ethnic influences with the latest E.M.isms.

CASSETTE REVIEWS

DEVO **Easy Listening Muzak Cassette.** No one has been more aware of the parodies of DEVO (such as the album "Devotees") than DEVO themselves. It seems only appropriate, therefore, that they should lead the charge with their own corn-ball combo organ versions of their ersatz "hits". Available through the DEVO Fan Club, 9120 Sunset Boulevard, Los Angeles, CA 90069, $4.50 plus postage.

David Byrne **The Complete Score from the Catherine Wheel.** The complete score contains twice the numbers and maintains a much more coherent flow than the LP of selected highlights (reviewed Jan/-Feb issue). The decision to release a cassette rather than one double album probably reflects the skittishness of the industry. $8.98 list at your local record store.

Steven Halpern **Hear to Eternity.** The cassette format allows Halpern to stretch out past the restrictions of his minimalist "Anti-Frantic" series of keyboard LPs. Exotic percussion, more elaborate productions, and total reliance on the Prophet 5 make this the most ambitious and therefore strongest Halpern I've heard. Contact Halpern Sounds, 620 Taylor Way #14, Belmont, CA 94002. $8.98 plus postage.

Gen Ken Montgomery **Gen Ken & Equipment.** A musical dadaist in the fine old tradition of Throbbing Gristle, New Yorker Montgomery recorded these noisy, wall-of-sound pop songs at home on a TEAC Portastudio. Additional copies are available for $6 plus postage from Aeon Import Records, 604 Princeton, Ft. Collins, CO 80525.

Rudiger Lorenz **Queen of Saba.** With a variety of instruments including a large self-build modular synthesizer, Lorenz is able to successfully pull off recording these eight tunes direct to tape, live without overdubs. That's amazing considering how much is going on in them - and no flubs anywhere. Send $10 (includes postage) to Transmitter-Cassetten, c/o Werner Pieper, D-6941 Lohrbach, West Germany.

Jimmy Layton **Stranger in a Strange Land.** Sample-and-hold synthi, usually self-triggered and abstract, with the occasional relief of a tune or tape collage. He's at his best when he's having fun being humorous. Available direct from the artist at ZYX Productions, 494 29th Avenue #1, San Francisco, CA 94121.

ELECTRONIC SOUND PISTOLS

An "Almost" Detective Story

By: Dr. Andrew L. Gelt

I recently took along a couple of Electronic Sound Pistols to a "new wave" gig I was playing, and more people asked about the ray guns and their sounds than anything else concerning the band. These devices emit incredible noises, and the low price reaffirms the importance of technology as it relates to electronic synthesis.

The "Sonic Fazer" (by Kusan, Inc.) has five sounds, is much larger than competing pistols, and has the most attention to cosmetic detail (or should that be "cosmic detail"?). Its red light is in the back, inside the sight and also pulsates with the modulation of certain sounds. Two of the gun's sounds are "Amplify" (rapid LFO sine wave modulation of the VCO and possibly the VCF; the small speakers used in such devices might cause equalization changes), and "Radiate" (sawtooth wave VCO modulated by a sine wave LFO at .11 Hz (!). I especially liked the "Mass Invert" sound (apparently both audio and subaudio modulation of the VCO). The other two sounds are labelled "Anti-Gravity" and "Ion Transport".

Parents might be thankful for the rather soft volume, but it might not be sufficient to drive a microphone in a loud rock band without creating feedback. The "Sonic Fazer" is the most attractive and readily available sound pistol, however.

The "Galaxy Gun 12" has twelve wonderful unnamed sounds, including some sophisticated "patches" which use such tech-

niques as inverted envelope generator modulation and modified square wave width. Unfortunately, its availability is questionable. I acquired mine in a local bar for $7, but neither the Galaxy Gun 12 nor its brightly colored box displays a company name, trademark, patent number, or copyright mark. Although I obtained the name of a local retailer, he would not allow me to list his company name or address since he didn't want to receive inquiries and orders after the product had been discontinued.

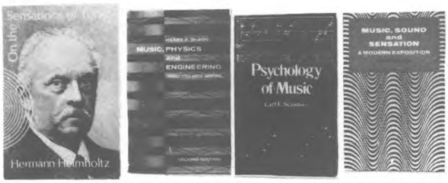
However, with various other pieces of information, some interesting facts came to light. Foreign manufacturers (predominantly located in the Far East) often produce forgeries of existing products (known as "knock-offs"), and then flood the American market. A story was told to me about a ship in the Port of Miami with 36,000 programmable robots from Korea. I got the impression that many of these electronic toys are brought into the country without paying tariffs or perhaps without even going through customs. This gave rise to thoughts such as an international toy black market that actually smuggles toys into the US. One distributor said that most of these products were sold to flea marketers, as well as to people who sell them to cruise ship tourists in island countries such as Jamaica, Haiti, and Curacao.

There even appears to be a "knock-off" of the Galaxy Gun 12, the "2100 Space Gun" which retails for $10.99 and also has twelve sounds. Most of the functions rely on square wave modulation of a VCO but its extended tessitura is an improvement.

At any rate, the Galaxy Gun 12 gets my prize for electronic sound toy of the year. If you ever see one, grab it! You won't believe the sounds.

POLYMART

SCIENCE



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

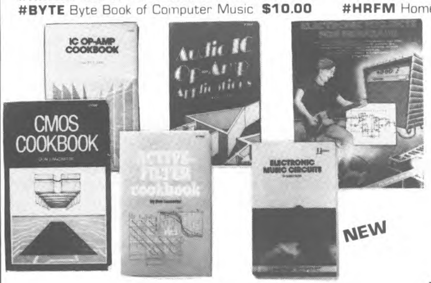
#SENS On The Sensation Of Tone	**$8.95**	**#PSYCH** Psychology of Music	**$6.00**
#MPE Music, Physics & Engineering	**$6.00**	**#MSS** Music, Sound and Sensation	**$3.50**

TECHNIQUE

Synthesists must be well versed in a number of techniques and principles. "How to" and project oriented books are a great way to pick up these skills easily. **How to Make Electronic Music** by Drake, Herder and Modugno is a standard introductory text for music systhesis classes, with chapters on equipment, tape technique, composition projects, and more. **Multitrack Primer** by Teac is a step-by-step guide to building, outfitting, and operating your home studio. The Byte book of **Computer Music** describes computer control of electro-mechanical instruments, Fourier analysis, circuits and loads of software. **Home Recording For Musicians** is the original guide to outfitting and operating a budget studio for maximum results, including mixer and audio processing circuits and a demo recording.

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

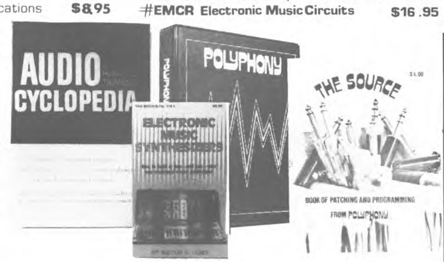
ELECTRONICS



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

#OACB Op-Amp Cookbook	**$ 15.95**	**#CMCB** CMOS Cookbook	**$ 12.95**
#AFCB Active Filter Cookbook	**$14.95**	**#EPFM** Electronic Projects **2nd edition**	**$14.95**
#AUOA Audio Op-Amp Applications	**$8.95**	**#EMCR Electronic Music Circuits**	**$16 .95**

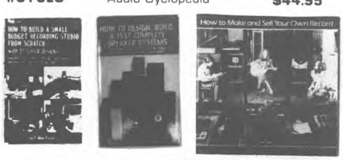
REFERENCE



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

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

HOW TO.......



NEW — How to Build a Small Budget Recording Studio From Scratch by F. Alton Everest – Twelve tested designs How to Design Build & Test Complete Speaker Systems by David Weems – A do-it-yourself guide for the ultimate in sound quality. How to Make & Sell Your Own Record by Diane Sward Rapaport – The complete guide to independent recording.

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#CSS	**Complete Speaker Systems**	**$ 7.95**
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4/8 Track Studio Log Book provides a place to keep all the important information on your tape library. Log in timing, type of tape used, record patches, make notes and use the expanded track sheet to list sequential changes in tape tracks relating to the settings of the index counter. Craig **Anderton's** Contemporary **Keyboard Articles** is a reprint of all the articles from June 1977 through February 1981, covers tips, technique, theory, maintenance, and numerous construction projects. **Device Back Issues** — during the year that this newsletter was published, it featured almost 200 pages of technical information for the guitarist/musician. A wealth of articles on; design, product reviews, and modification and construction projects. Sold in complete set, individual issues not available. Limited number available, order yours now.

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

RECORDINGS

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

BACK ISSUES

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

- ☐ #0204: 4/76: music notation - timing, external inputs for Gnome, Programmable Drums, Equally Tempered D/A, low cost AR project, digitally encoding keyboards, patches, Volume 1 & 2 index.
- ☐ #0301: 7/77: frequency divider project, random tone generator project, normalizing synthesizer controls, eliminating patch cords, computer control of analog modules, Chord Egg modification, adding pitch bending, patches.
- ☐ #0302: 11/77: The Sensuous Envelope Follower, digital gates, LED wall art, build a bionic sax, data to music peripheral project, Apple II as a music controller, using the NE566 as a VCO, patches.
- ☐ #0303: 2/78: computer controled Gnome, using joysticks, build a bionic trumpet, ultra-VCO modifications, voltage control the Mu-Tron Bi-Phase, oral joystick, patches.
- ☐ #0304: April/May 78: Minimoog modifications, non-keyboard module use, phasing and flanging (theory and circuits), memory expansion for programmable drums, digitally addressed transposer project, polyphonic software (with software transient generators), patches, Volume 3 index.
- ☐ #0402: Sept/Oct 78: electronic music notation, notes on the recording of "Cords" by Larry Fast, sequencer software - part one, rhythmic control of analog sequencers, touch switch projects, modular vocoder techniques, PET as a music controller, patches.
- ☐ #0404: January/March 79: add-ons for vocal F and V converter, shorthand patch notation, more on note to frequency conversion, graphic monitor project, George Russell, super VCA circuit, echo software, Vol. 4 index.
- ☐ #0502: July/August 79: hex VCA/mixer project, electronic music schools and studios, modify the Oberheim Expander Module, profile of Ernest Garthwaite, budget microphones, digitizer projects and software, bar graph ICs.
- ☐ #0505: January/February 80: Joseph Byrd, Mort Garson, Larry Fast on 'Games', composing for 'live plus tape', using the CA3280, recording vocals, ADSR circuits.
- ☐ #0506: March/April 80: Computers in Music: real time audio processing hardware, Powell sequencer system, Max Mathews, advanced STG software, PortaStudio, phase modulation, Volume 5 index.
- ☐ #0601: May/June 80: Gary Numan, Microcomputers in Real Time Audio, Build a Digital Audio Delay Line, writing Documentation, Richard Hayman Composer/Performer Home Recording: Applying Harmonizing and Pitch Transposing Techniques by: Craig Anderton.
- ☐ #0602: July/August 80: Peter Gabriel, digital VCO project, dream modules, optimum level settings, dynamic phrasing, patches.
- ☐ #0603: Sept/Oct combined with Nov/Dec 80: alternate controllers, add voices to Casio M-10, voltage controlled quadrature oscillator project, cordless patch bay, recording rules, patches.
- ☐ #0604: January/February 81: Special Construction Edition; Build: Audio Circuit Breaker, Pulse Width Multiplier, Magnetic Harp, 50 Watt/Channel Stereo Power Amp, Quad Sequential Switch, DOD Mods, patches.
- ☐ #0605: March/April 81: Portable Music Issue, reviews of Remco's FX, E-H Mini-synthesizer, Casio's VL-Tone, plus mods for the M-10, GR-500, mini-amp, and the Korg X-911. Introducing; Practical Circuitry and On Location, new columns.
- ☐ #0606: May/June 81: Synthesizer: Hardware Mods and Software. Modular Synthesizer Effects, Environmental music, Keyboard assignment for the 8700, new columns; Details, Practical Circuitry, and On Location. Volume 6 index.
- ☐ #0701: July/August: Guitar Electronics: Modify: Fender Amp, MXR Phase 100, GR-500. Input/Output Structures, $5 Analog Programmer, Sample and Hold technique, Modular Synthesizer Effects, new column: Applied Synthesis, Marketing Your Records.
- ☐ #0702: Sept./Oct.'81: Harald Bode Interview, Live Plus Tape - New Technique, Xenharmonics, Kraftwerk Live - Review, Psycho-Acoustic Experiments, Practical Circuitry - Super Controller, Applied synthesis - Brass, Construction Tips For Beginners.
- ☐ #0703: Nov./Dec.'81: Dave Rossum interview, Applied Synthesis: Strings,Details: Series-parallel/Sum-Difference. The Sound Gizmo and Pro-One Reviews, Practical Circuitry: VCO Deluxe.
- ☐ #0704 Jan./Feb.'82: Bob Moog interview, Chip Power - STK-050/070, Simple Square Wave Shaper, Tape Timer Ruler, Practical Circuitry: VCAs made simple, Details: Gozinda & Gozouta Revisited, Korg Trident & Casiotone 202 Reviews.
- ☐ #0705 Mar./Apr.'82: Electronic Music Math, Analog Delay Clock / Modulation; Frequency Domain Modifiers; Screen-Wave for the TRS-80; Touch Switches Revisited; Practical Circuitry: ADSR the Easy Way; Getting the most out of a Cheapo (Guitar).

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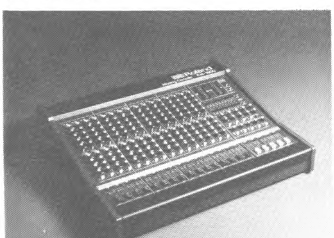
CURRENT EVENTS

'Tell Them You Saw It In Polyphony'

New from Roland. The Juno-6 is a 6 voice poly synth with arpeggiator, digital key transpose (so all tunes can be played in C if you want), chorus generator, and other features; it lists for \$1295. The RM-1200B and RM-1600B are 12 and 16 input mixers respectively designed for sound reinforcement applications and include features such 4 band EQ for each channel, two "master" 9 band graphic EQs, fluorescent bar graph displays for the program-cue-monitor-effects output, and flexible input structures. The RM-1200B retails for \$2195, the RM-1600B for \$2795.



Record News. Gramavision Records (260 W. Broadway, New York, NY 10013) has introduced the "New Directions" series, starting with the work "Rainforest IV" by David Tudor. Future releases will include works by Lukas Foss, Vladimir Ussachevsky, Lou Harrison, and others. For more information, write Gramavision or call (212) 226-7057.

TEAC Model 2A Mods. ARCAS Engineering (RFD 3, Burt Hill Road, Winchester, NH 03470) offers several services for upgrading TEAC's Model 2A. Claimed advantages include improved slew rate, lower noise, better drive capability, etc. ARCAS will perform the complete modification for \$115, or you can do it yourself with a \$65 kit of parts and instructions. Optional, high-performance outboard mic preamps are also available for \$15. A manual describing all modifications is available for \$5 (refundable with parts order).

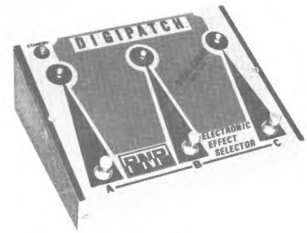
Company Update. Electro-Harmonix, one of the first effects manufacturers, nas filed for bankruptcy. Those of us who were raised on their humorously named, cheap, sometimes frustrating, but musically valid devices are sorry to see them go. But we wouldn't be at all suprised if E-H resurfaces in some form or another in the near future.

Imaginearing Audio, originators of the Alphatone tuners and Echo/Digital Recorder, have ceased production of their devices. However, the company still handles all warranty work and will back up any units requiring service.

And finally, some good news: rumor had it that Heet Sound Products, makers of the "E-Bow", had gone out of business. We contacted Greg Heet to check out those rumors, and we're happy to report that the E-Bow is indeed alive and well. E-Bows may be ordered direct for \$75 postpaid from Greg Heet, 611 Ducommun St., Los Angeles, CA 90012.

New Synth Chip. The MOS Technology 6581, included with some new Commodore computer products, provides three separate voices. Each voice has a 9 octave range, choice of sawtooth, variable pulse, triangle and noise waveforms, and programmable attack, decay, and sustain times. The 6581 output can be sent directly into an amplifier without any further conditioning.

Quiet Op Amp. Analog Systems (Box 35879, Tucson, AZ 85740) have announced the MA-344 single "Qui-FET" op amp and MA-345 dual op amp. The company quotes noise specs of 8 nV/root Hz at 1 kHz (lower than that of standard bi-FET type op amps), slew rate of 10V/microsecond, and unity-gain bandwidth of 2 MHz.



Switcher. PNP Manufacturing (PO Box I, Brooklyn, NY 11204) has announced Digipatch, an electronic effect switching center which allows up to three different effects devices to be switched or series mixed for a total of eight different combinations. Three footswitches optionally bypass the effects, with LEDs indicating status.

Head Games. TEAC's QP-001 head cleaner kit (\$7.95 list) is a "universal" cleaner which takes care of heads, capstan shafts, and pinch rollers.

Power Amp. Who cares about one more power amp? Well, AMP's Model 8000 delivers 200 Watts into 8 Ohms or 350 Watts into 4 Ohms - but without a noise producing fan, making it well suited to recording studio applications. Contact AMP, 8440 Paso Robles Ave., Northridge, CA 91325 (tel. 213-701-0249).

CURRENT EVENTS

New EQ. Loft (91 Elm St., Manchester, CT 06040) has announced the Model 401 parametric EQ with four overlapping frequency bands, continuously adjustable frequency, adjustable bandwidth, and selectable boost/cut. Mounting in a 1.75" rack mount, the 401 claims THD of 0.01% and a noise level of -92 dB ("A" weighted).

New from SSM. The SSM2100 is a complete monolithic subsystem to realize logarithmic and exponential transfer characteristics. It includes two precision op amps, a high conformance transistor pair, and a precision bandgap reference. Applications include log sweep generators, linear to dB conversion, and analog compression/expansion.

The SSM2022 is a low cost, high performance dual VCA. Claimed S/N ratio is 84 dB with 0.2% THD; the chip also features summing node signal inputs, linear and exponential gain control, and low control voltage feedthrough. Contact SSM at 2076B Walsh Ave., Santa Clara, CA 95050 (tel. 408-727-0917).

Korg Trident Update. Korg (89 Frost St., Westbury, NY 11590) has introduced the Trident Mark II, which retains the features of its predecessor but also adds advanced programming capabilities, with double the program storage and full Edit and Tape Interface capabilities. The all-new synthesizer section includes two full ADSR envelope generators per voice, programmable attenuator and auto damping, plus an improved 24 dB/octave filter and better noise specs.

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Understanding Digital Synthesizers: The Digital Filter

By Dave Wilson

A while back I started thinking about building a digital synthesizer, because I have a microcomputer (OSI Challenger-1P) and because my knowledge of hardware is dwarfed by my knowledge of software - the one thing I can always do without guessing (like I sometimes do when I wire op amps). One of the problems with digital synthesizers is the implementation of standard filters (low pass, band pass, etc.) with digital circuits/programs. After playing with a little calculus and the schematic to PAIA's 4730 multimode filter, I came up with these four equations for the four outputs of the filter:

Highpass =

$$\frac{1}{1 + Q} \times \text{BPass} - \text{LPass} - \text{Input}$$

$$\textbf{Bandpass} = -fc \times \int \text{Highpass}\, dt$$

$$\textbf{Lowpass} = -fc \times \int \text{Bandpass}\, dt$$

$$\textbf{Notch} = (\text{Lowpass} + \text{Bandpass})/\ 2$$

These equations seem correct in that they behave exactly like the filters they were designed to replace. If you approximate the integrals (yes, those funny looking squiggly lines are intended to be integral signs - pardon my word processing!) over time by just adding (the other name for integral calculus is SUMatory calculus), then if the sampled rate is high enough to be inaudible, the error introduced by the discrete summations will also be inaudible.

Now picture the digital synthesizer. To make it simple, we'll look at the processing of an external signal:

Guitar---A/D Converter---Digital Synth---D/A Converter---Amp

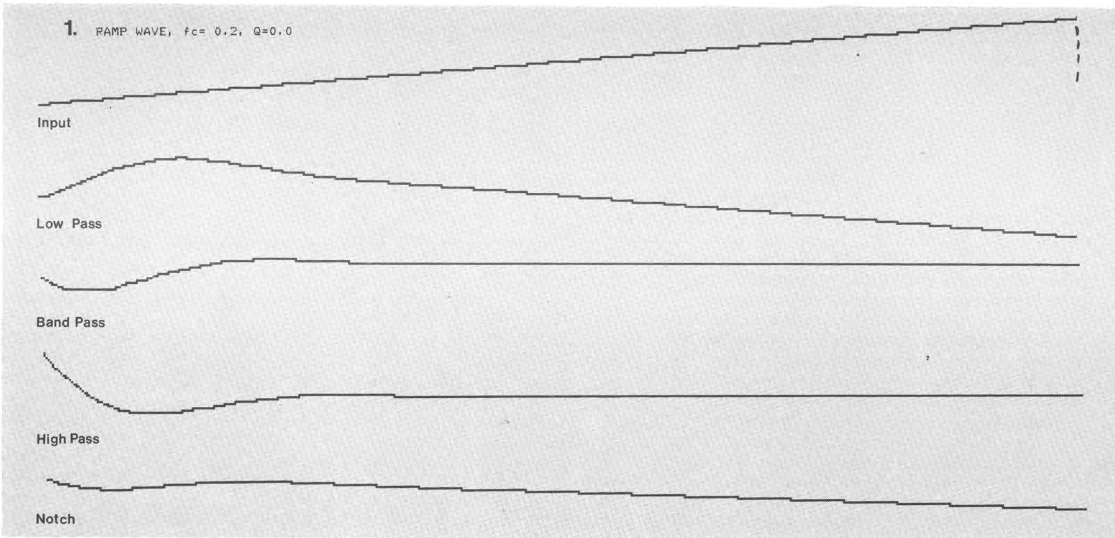
```
10  REM *****************************************************************
20  REM *        ORIGINAL PROGRAM FEB. 82, THIS VERSION 3/3/82.          *
30  REM *  DIGITAL MULTIMODAL 6 db/OCTAVE FILTER (PAIA 4730, EML 101, ETC.) *
40  REM *  PROGRAMMED BY DAVE WILSON OF LLAMA SOFTWARE, A DIVISION OF THE  *
50  REM *  CHEERLEADERS CORPORATION, 1102 MERRITT ST., FAIRFIELD, CT 06432  *
60  REM *                                                                *
70  REM *  LINES 360-440 ARE A DIGITAL OSCILLATOR FOR DRIVING THE FILTER.  *
80  REM *  LINES 460-490 ARE THE FILTER ITSELF (JUST FOUR LINES! SIMPLICITY!)*
90  REM *  LINES 510-630 ARE A PRINTOUT ROUTINE THAT PRINTS H'S FOR THE    *
100 REM *  HIGH PASS CURVE, L'S FOR THE LOW PASS CURVE, B'S FOR THE BAND PASS*
110 REM *  CURVE, N'S FOR THE NOTCH CURVE, AND I'S FOR THE INPUT CURVE.    *
120 REM *  THIS PROGRAM WAS TRANSLATED INTO FORTRAN TO RUN AN ANADEX PRINTER/*
130 REM *  PLOTTER FOR NICER OUTPUT. THE COMPUTER USED IN EITHER CASE IS A *
140 REM *  PDP-11/34 BY DEC WITH THE RSX 11M OPERATING SYSTEM.             *
150 REM *                                                                *
160 REM *****************************************************************
200 P=0
210 L=0
220 B=0
230 PRINT "FREQUENCY (0 TO 1) AND Q (>=0)";
240 INPUT F,Q
250 IF F<0 THEN 230
255 IF F>1 THEN 230
260 IF Q<0 THEN 230
270 PRINT "INPUT WAVEFORM: 1=RAMP, 2=PULSE/SQUARE, 3=TRIANGLE, 4=SINE";
280 INPUT W
282 IF W<1 THEN 270
285 IF W>4 THEN 270
290 IF W<>INT(W) THEN 270
300 IF W<>2 THEN 340
310 PRINT "PULSE WIDTH (0 TO 100%)";
320 INPUT P
330 IF P<0 THEN 310
335 IF P>100 THEN 310
340 REM COMPUTATIONS START HERE *****************************************
350 FOR K=1 TO 100
360 ON W GO TO 370,390,420,440
370 I=K/100-.5
380 GO TO 450
390 I=-.5
400 IF K<=(100-P) THEN 450
405 I=.5
410 GO TO 450
420 I=ABS(50-K)/50-.5
430 GO TO 450
440 I=SIN(K*.06283)
450 REM FILTER STARTS HERE **********************************************
460 H=(1/(1+Q))*B-L-I
470 B=B-F*H
480 L=L-F*B
490 N=(L+H)/2
500 REM THAT'S THE WHOLE FILTER! ****************************************
510 A$="
520 P1=(1.5+L)*40
530 P2=(1.5+B)*40
540 P3=(1.5+H)*40
550 P4=(1.5+N)*40
560 P5=(1.5+I)*40
570 A$=SEG$(A$,1,P1-1)+"L"+SEG$(A$,P1+1,LEN(A$))
580 A$=SEG$(A$,1,P2-1)+"B"+SEG$(A$,P2+1,LEN(A$))
590 A$=SEG$(A$,1,P3-1)+"H"+SEG$(A$,P3+1,LEN(A$))
600 A$=SEG$(A$,1,P4-1)+"N"+SEG$(A$,P4+1,LEN(A$))
610 A$=SEG$(A$,1,P5-1)+"I"+SEG$(A$,P5+1,LEN(A$))
620 REM USE A$=LEFT$(A$,P1-1)+"L"+RIGHT$(A$,P1+1), ETC.
630 PRINT A$
640 NEXT K
650 END
```

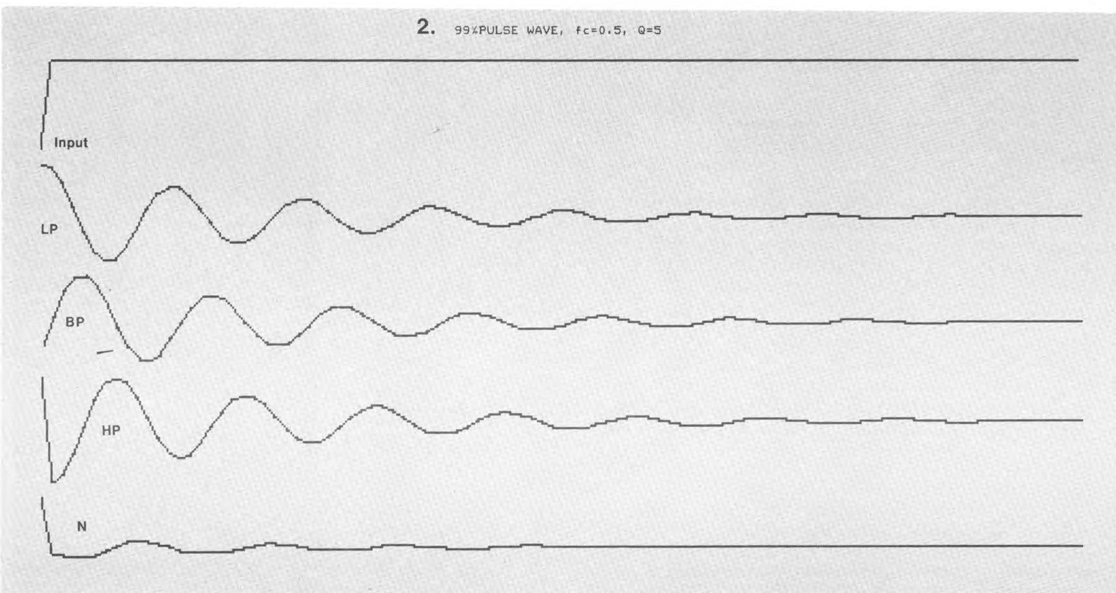
The analog to digital (A/D) and digital to analog (D/A) converters merely allow the signal to exist in a form which the digital synthesizer can use. You play a string on the guitar, and the sound is sampled 50,000 times. Each little sample is fed to the synthesizer where the four equations are applied (in the order given) and one of the output variables - let's say bandpass - is fed out through the DAC to the amp and speakers. If we use a high (greater than 5) value for Q and change fc from 0 to 1 and back to 0 over a few seconds, the result is the classic "wa" sound. The digital synthesizer is nothing more than a computer solving the same four equations over and over again for new values of INPUT, fc, and Q.

And there you have it - a look inside an (oversimplified) Digital Synthesizer showing what it does. To provide more detail, I have written a small program in BASIC which will run on most home computers. It doesn't work in real time; that is, you can't actually filter sound with it, but you can plot with it and then compare the output it produces to the output of an analog filter as displayed on a scope. The program contains the following software modules:

1. A software "VCO" with five available waveforms.
2. The filter.
3. A software oscilloscope with five beams - eat your heart out, Tektronix!

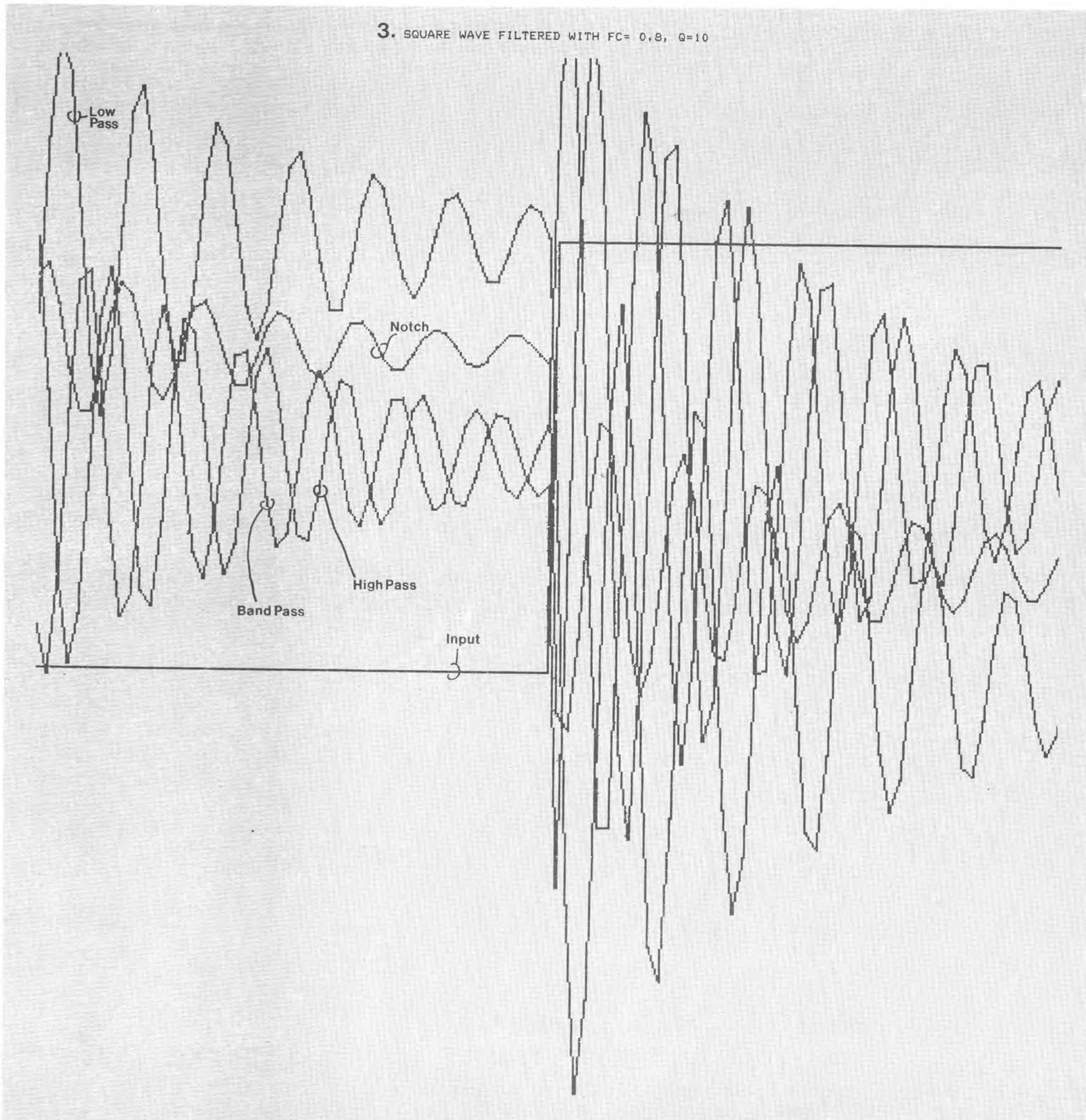
It should be noted that two running totals are kept by the program given in listing 1; the values of L and B (lowpass and





bandpass) from the previous solution of the equations are used in generating the next solution. There are the "running totals" that serve as our integrators. Their analog counterparts are the capacitors in the VCF, and if you've ever wondered why the filter is called "state variable", well, these two variables ARE the state variables, because for the same instantaneous values of INPUT, fc, and Q, the values of the output will depend on the values of these variables, i.e., what state the filter is in when the inputs are received.

In looking at the sample RUNs of the program (figures 1-5), one point should be made: the BASIC program is intended to run on any home computer and plot by printing letters on the paper or screen. For demonstration, however, the outputs shown were made on an Anadex printer/plotter, with a Deanza Image Processor used to store the images before printing. To drive the Deanza, the whole program had to be translated into FORTRAN and modified somewhat - the bottom line is that your output will resemble what's shown here, but it will be cruder (unless you own a graphics program and beef up the program a little). Line 620 shows how to translate the SEG$'s used in lines 570-610 into the more standard LEFT$ and



RIGHT$. Line 510 is simply 132 (or thereabouts), spaces being store in A$, and the output is 132 columns wide. To run on 80, 64, or 40 columns, change the *40s in lines 520-560 to a lower number (like *20 for 80 column). Don't change the 1.5+s, as they seem to work fine just the way they are.

Figure 1 is a Ramp wave filtered at a relatively low fc, with Q = 0.

Figure 2 is a narrow pulse filtered at a higher fc, with a Q of 5. Note the ringing and the relative phase of the LP, BP, and HP outputs.

Figure 3 shows the phase relationships of the outputs by overlapping all five waves (the four outputs plus the input). This is the style of output of the BASIC program.

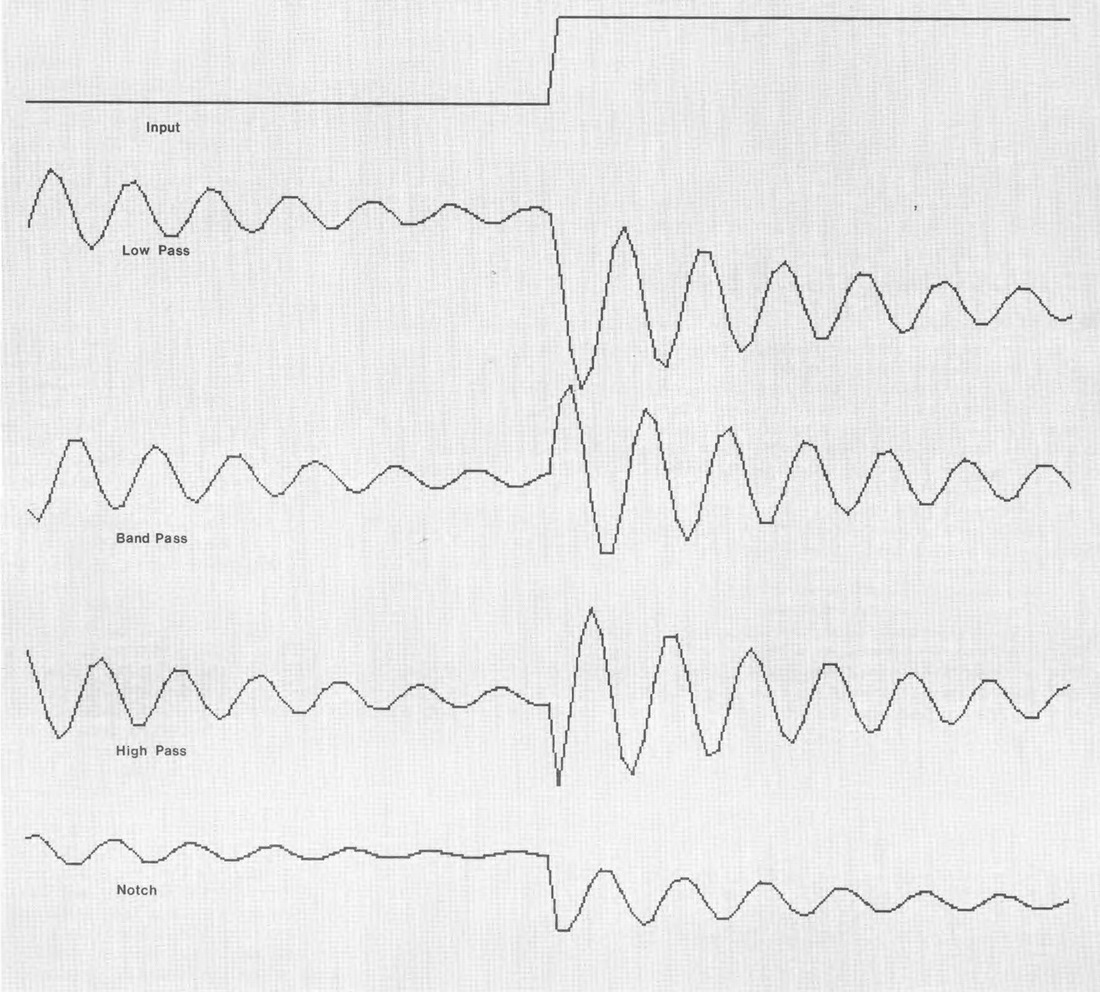
Figure 4 shows the same waveforms as in figure 3 (square wave, fc = 0.8, Q = 10) but separated out so that they are separately visible.

Figure 5 shows the phase relationships of the lowpass and bandpass outputs by plotting one vs. the other. This picture (when drawn on a scope by an analog synthesizer filter) appears in much synthesizer literature - ARP user manuals, older CFR catalogues, Aries data sheets, and PAIA even has a section at the end of their VCF manual entitled 'Scope Art, showing some of the fun pictures analog filters can make.

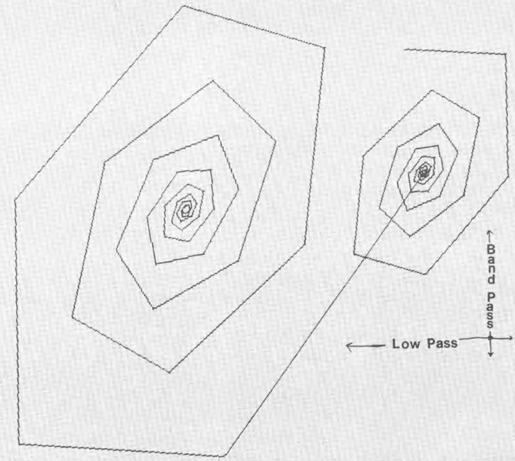
Sine and triangle inputs look rather boring, so there aren't any examples of these. The reason is that there just aren't that many harmonics in a triangle wave (and none in a sine wave) to filter out.

Many things happen inside a digital synthesizer, but filters are probably the most complex. I hope my digital filter has helped others understand digital synthesizers; it has sure helped me a lot.

4. SAME SQUARE WAVE FILTERED WITH FC=0.8, Q=10, NEW PLOT SCHEME.



```
1  REM  THESE ARE THE CHANGES
2  REM  NEEDED TO RUN THE
3  REM  FILTER PROGRAM ON AN
4  REM  APPLE II COMPUTER
5  REM
170  DIM A$(40)
505  REM  A$=40 BLANK CHARACTERS
510 A$ = "
                              "
520 P1 = (1.5 + L) * 13.3
530 P2 = (1.5 + B) * 13.3
540 P3 = (1.5 + H) * 13.3
550 P4 = (1.5 + N) * 13.3
560 P5 = (1.5 + I) * 13.3
570 A$ =  LEFT$ (A$,P1 - 1) + "L"
     +  RIGHT$ (A$, LEN (A$) - P
     1)
580 A$ =  LEFT$ (A$,P2 - 1) + "B"
     +  RIGHT$ (A$, LEN (A$) - P
     2)
590 A$ =  LEFT$ (A$,P3 - 1) + "H"
     +  RIGHT$ (A$, LEN (A$) - P
     3)
600 A$ =  LEFT$ (A$,P4 - 1) + "N"
     +  RIGHT$ (A$, LEN (A$) - P
     4)
610 A$ =  LEFT$ (A$,P5 - 1) + "I"
     +  RIGHT$ (A$, LEN (A$) - P
     5)
```



5. SQUARE WAVE, FC=.9995, Q=5, PLOT IS LP VS. BP.

PRACTICAL CIRCUITRY

Continued from page 17

So far, we have examined the standard features which you might expect to see with any filter; now let's look at the single most unusual feature of this circuit. AS1-AS8 are analog switches (each is 1/4 of a CD4066), and have been set up pairwise to form single pole, double throw switches. Essentially each one of these SPDT affairs will switch the tuning capacitors either between ground or the input resistors of the gain cells. With the capacitors hooked to ground, you have a low pass filter; when they're swung to the gain cell input resistors, you get an all pass filter. I used electronic switching for several reasons: first, who wants to solder all those wires! Besides, a couple of 4066s are less expensive than a ganged, rotary switch (one ordinary SPST switch, S1, does the job now).

A3 is set up as an inverter so that when one side of the 4066 switch is enabled, the other side is disabled. Since I used a quad op amp for this project, and had one op amp left over anyway, it was convenient to do things in this manner.

Calibrating the PAL. Follow this procedure exactly:

1. Turn R39, R41, and R42 down completely. Turn R37 and R38 to the center of their rotations. Monitor the output at J4 with an oscilloscope or multitester (the higher the impedance, the better).

2. Turn the blend control, R40, for full filter output. Adjust R34 until the output reads 0V.

3. Turn the blend control, R40, for full straight signal at the output. Adjust R35 until the output reads 0V.

4. Repeat steps 2 and 3 until the output reads 0V no matter position R40 is in.

5. Now turn R39 up full and apply a 1 Hz signal to J2 (use the LFO from your synthesizer for this if you want). The signal should be about 10 V peak-to-peak in amplitude. Now while monitoring the output, adjust R4 for minimum control voltage feedthrough.

6. Finally adjust R13, the Volts per octave trim. This is best done with your keyboard hooked up to J1. Set the filter into oscillation by turning resonance control R42 full up. Now tune the filter like any VCO by comparing accuracy at different octaves.

Applications. First, set S1 to the low pass mode and turn blend control R40 to full strength filter output. The filter is now acting like a standard low pass filter.

Next, put S1 in the all pass mode and keep the blend control on full strength filter output. Sweep the coarse frequency control, R37, and notice the pitch shift characteristic of all pass filters.

Now keep S1 in the all pass mode, but blend in some straight signal. Sweep the coarse control and notice the typical "swooshing" sound of a phase shifter.

As I mentioned in the opening paragraph, there are lots of filters around and some of them are hard to use. I think you'll find this one is exceptionally easy to use, yet has enough novel features to make its inclusion in any synthesizer system worthwhile. And by the way, in my original prototype I had some front panel space left over, so I included the ADSR circuit I presented here several months ago. Let me tell you, an ADSR and VCF combination makes for a great pair! Happy building! ■

The Realistic MG-1 Synthesizer A Review

by: Chuck Pogan

I doubt there is anyone reading this publication who has never been to Radio Shack; this store has done for electronic hobbyists what McDonald's has done for hamburgers (and there are more Radio Shacks than McDonald's - ed.).

This is not to imply that small semiconductors and related items are their only strong suit - the "Shack" has recently proven stiff competition to all major stereo and computer specialty companies.

In the course of their expansion into other fields, the Shack has gotten together with the Norlin Co./Moog brainiacs and produced a polyphonic synthesizer which they claim is the best buy, feature for feature, of any synth around. Well, let's put their "Concertmate MG-1" through its paces and see for ourselves.

Overview. The unit itself appears to be solidly built. The face panel, at first glance, seems almost over-simplified , with plenty of room between all controls. All control functions are color-coded and laid out in the now expected performance synthesizer arrangement of LFO, OSCs, filter, env. generator, and mix output.

The instruction manual uses the "welcome to the wonderful world of the electronic music synthesizer" type of approach which may ruffle the feathers of a more conventionally schooled synthesist. Terms like oscillator are renamed (for simplicity's sake) "Tone Generator" and the envelope generator is called "Contour". Since the new terms are still self-explanatory, there should be no confusion as to their functions.

The two Tone Generators (TG for short) each have a three-position octave select switch. TG #1 can transpose one or two octaves down while TG #2 transposes up one, down one. Choice of waveforms on #1 is sawtooth or square and on #2, sawtooth or pulse. TG #2 also includes an oscillator "sync" switch and a "detune" slider. Two tuning knobs on the upper left hand of the instrument control the master tune (all oscillators), while the "Polyphony" control determines the frequency span between the two TGs.

The 24 dB/octave low pass filter has three sliders controlling the cutoff frequency, peak emphasis, and the response to the envelope (contour) generator. A three position filter/tracking switch selects full keyboard tracking, half keyboard tracking, and no tracking.

The contour generator is an attack/release type envelope trigger. You can bypass it using the "in-out" sustain switch and have it "repeat" the envelope at the LFO clock rate. The LFO section offers sine, square, and sample-and-hold modulation. Two sliders control the amount of designated modulation sent to the oscillators (both simultaneously) and/or the filter.

The mixer section has separate volume output levels for the two oscillators, a noise level output, and a "Bell-Tone" slider which seems to be an additional tone with a complex waveform. The latter is a much more harmonic sound than its name would imply, and is a nice, if not unusual, addition.

So far, the unit has met all of the basic mono synth requirements. But the inclusion of the "Poly" slider is what interested me the most.

The "Poly" section is most curious. With the mono section silent (mix sliders down), turning up the "Poly" slider generates a fully polyphonic organ-like tone. Add the mono section sliders and bingo, you hear a lush multi-layered orchestral sound, and the mono section takes on a polyphonic playing characteristic.

Pros of the MG-1. The MG-1 can be easily patched for the many musically useful sounds in the mono section. The filter sounds okay, and the oscillator stability seems fine. The "Poly" sound, when mixed with the mono section (set for no sustain), affords a rich, textured tone that is probably the best patch on the unit and sounds as good or better than a lot of more expensive organs. The Shack also had the foresight to include an aux input plug for future computer control and the like.

One of the major advantages to buying a synth like the MG-1 is the expected speed of service should the need arise. Radio Shack prides itself with respect to service, and if you're a synthesizer owner who has had to have repair work done, I'm sure you will agree that fast service should be a "top-three" priority when buying a synth. Considering the "name-brand" manufacturer and the widespread Radio Shack repair services, you should have minimal problems in this regard.

Cons of the MG-1. The "Poly" feature could have a lot more

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potential if the poly tone could be controlled by the contour generator. As it stands now when using the mono section in conjunction with the poly section, the poly tone will decay the moment your finger leaves the key (like an organ), leaving the mono sound (if any decay is used) to linger on. This also causes a noticeable amount of false triggering and "glitching". The mono source will dart to whatever key is released last. As mentioned before, when the mono section is set for no VCA decay, the poly sound is real good. But try to get cute with the VCA on the mono section and mix it with the poly tone, and you're in for a tough time.

General evaluation. I don't expect Norlin to put the same keyboard triggering system in the MG-1 as the Polymoog, but it would have made for a much more playable instrument if a more reliable multiple trigger were included. And if the poly tone could be treated by all the synth's functions (LFO + VCA) and triggered simultaneously with the mono section, this would create a superior unit.

This is the Shack's first contribution to the synthesizer market, and they are approaching this area gingerly. But perhaps because of the market they are trying to appeal to, they have kept the unit overly simple and overlooked some minor modifications which would have make the unit instantly recommendable.

Whether or not the MG-1 is for you, I really can't say. It has enough appealing features to at least warrant a "test drive" (as the Shack would say). You may even find the unit's strong points are worth the suggested price.

But in the meantime, I'm hoping they will have Norlin modify this unit to compete head-to-head with the big-name brands. What I'd like to see is a unit with the extras I mentioned before, even if it costs a bit more. Then Radio Shack would be able to compete with other synthesizer manufacturers as equally as they do with the stereo and computer companies.

Still, try it out for yourself. Knowing the type of people who read this magazine, I'm sure you'll find yourself near a Radio Shack in the near future anyway

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FEATURES:

PROJECTS

COLUMNS

Conventional audio synthesizers are really special purpose analog computers; dedicated circuitry (filters, level shifters, and so on) process a basic audio waveform to produce some desired (musical) result. Traditionally, most musical instruments (and computers) have been mechanical and electrical in nature. However, optical analogs of these devices are possible, which opens up the possibility of optical synthesis devices.

Optical devices are not unknown to audio. You've probably seen optics used in existing audio equipment, such as wa-wa pedals and end-of-tape sensors. Also, opto-isolators have been applied to audio signal control and processing since the 50s. But, these are examples of rather low level optical methods and are primarily electrical in effect.

An optical keyboard. Figure 1 shows something that is more purely optical, namely, an optical keyboard. It consists of a light source and an array of beam splitter prisms, each of which divert a portion of the source light beam in an outward direction. Adding matching light detectors creates a set of simple "switches", since a finger touching the prism surface reflects light back on to the detector, thus closing the "switch". Besides its novelty, there are several real benefits:

- "Solid state" operation means no moving parts.
- With suitable depth of field, an optical keyboard can sense velocity and attack.
- The keyboard is optically isolated, thereby minimizing any possible shock hazard.

A guitar could also use light beams instead of strings, in fact, such a device was described at the 1981 AES show in New York. Interrupting or reflecting the beam as it travels to various sensors could allow for chords, notes, and different styles of "picking or plucking".

Optical cable. Fiber optic cable can also replace the usual metal shielded wire cords that transfer audio signals from instruments to mixing boards or amplifiers. Not only are fiber optic "wires" smaller and lighter, unlike standard cords they will not act as antennas capable of picking up 60 Hz hum or power switching spikes. Off-the-shelf fiber optic "Y" and multiple connectors can provide patching and splicing. Of course, it is also possible to completely do without cords and transmit light beams through the air; there would be a "transmitter" at the instrument which would modulate an infra-red LED, and a photo-sensitive "receiver" at the amplifier. However instead of infrared beams, perhaps beams in the visible range could be used to give a combination of light show and audio event. As long as the ambient light isn't a problem, the modulation scheme could even shift the frequency (color) of the beams, thus giving a "color organ" effect.

OPTICAL AUDIO

By: James Lisowski

Creating a keyboard out of thin air. Imagine this: instead of a physical keyboard, consider a hologram of a keyboard (or other instrument) projected into the air. Anyone looking at the scene sees a keyboard, where none actually exists; there

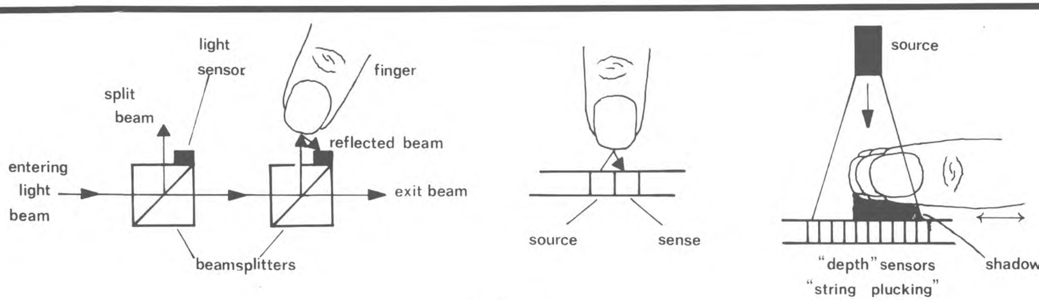


Fig. 1
Keyboard Elements

is only an image in the air. But interaction with this optical image can be sensed by noting the interference the player's fingers make within the "space" of the image (this could be realized with optical/electronic computers or video pattern recognition equipment). Thus, you could "play" the note associated with a key on the holographic keyboard - and could even move the image of the key down in space, making the key apparently respond as if it were a "real" key.

This idea opens up a whole realm of possibilities...for starters, how about multiple keyboards in small regions of space at any angle or three dimensional location arrangement, instant instrument redefinition, patch call-up, or change in microtonicity? Keyboard images could be switched in and out from under your hands faster than it would take to move physically from one "solid" keyboard to another. Key and control layouts would not be limited by physical construction design limitations, and as a bonus, setup time would be reduced (there would be fewer bulky boxes to stack and lug around). I'm sure you can think of other new tricks to be done with a "virtual keyboard".

The optical synthesizer. Up to this point, I've been discussing optical input and output devices that are not that far removed from conventional end-of-tape switching. Let us now make a leap to the optical computer/processor/synthesizer.

As you may remember from high school physics, the electromagnetic spectrum covers a long, continuous range of frequencies. The audio signals we normally encounter in synthesis are in the lower regions; as we move progressively higher in the scale of energy events, we find radio waves, IR/-visible/UV light, X rays, and Gamma rays. The main point is what we can do to waves in one part of the EM spectrum, we can do to waves in other parts of the spectrum.

For example, we could construct an optical oscillator with the use of lasers, lenses, optical filters, mirrors, and other optical components. Although the output is far removed from audio frequencies, we can translate it downward with familiar techniques such as heterodyning. When you mix two very close frequencies together (even optical frequencies), a beat or sum and difference frequency develops; once you've done that, you've changed optical into audio. Because of the very high frequency ratios involved, the range or bandwidth of the optically produced audio far exceeds that of common electronic synthesizers. Not only that, the materials used in optical audio are easy to manufacture. For instance, a solid crystal block can dynamically alter the frequency or phase of light passing through it by a property known as bifurgence. Doing such things as "pitch bending through touch", that is, stressing the optics to change the frequency, could give the conventional type of human control over an optical oscillator. Devices such as pulse mode-locked lasers (devices which give off pulses at equally spaced intervals) could act as frequency sources or synchronization references (metronomes or click tracks). Again, using reflective or other optical techniques, it's possible to make frequency multipliers/dividers (just like the electronic ones) by shifting the light frequencies up or down. You could also alter the amplitude (brightness) of the light to create low frequency oscillator effects. Naturally, like conventional synthesizer components, these devices can be used for control purposes as well as being sound sources.

An optical bandpass filter. Ordinary colored optical filters can be used in the same manner as fixed electronic filters (high/low cut or bandpass filtering). The components shown in figure 2 might be found in a photospectrometer. In this example, a beam of white light (many different wavelengths) is refracted (spread into its component wavelengths) by a rotatable prism. The refracted rays pass through a narrow slit in an otherwise opaque material, which allows you to choose various frequency ranges at the output end. Known as a monochromator, this is an adjustable optical bandpass filter. Like the electronic audio bandpass filter, this one can be swept, held fixed, or have its pass bandwidth changed since you can pick and choose which frequencies to transmit by rotating the prism. These signals could be simple audio encoded frequencies, or they might be entire channels of information (several audio or control signals modulating a "carrier" frequency). This gives you the equivalent of a channel selector-/mixer in one device. Several could be cascaded for more complex filtering or routing. Now, start adding lenses, mirrors, modulators, inverters, amplifiers, splitters, multiplexers, delay lines, mixers, and you get an idea of how similar optical synthesis processing is to conventional electronic synthesis processing. Remember too that these are mostly analog methods - you can have digital optical control/processing as well. The world is wide, and not all electronic...so see the light!

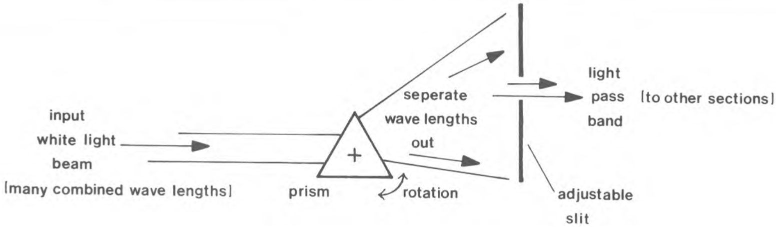


Fig. 2
Optical Adjustable
Bandpass Filter

Component Profile: SSM 2033 and SSM 2044

by: Tom Helfrich

While several manufacturers make ICs suitable for electronic music, there are only two companies specializing in the field, Solid State Music and Curtis Electromusic Specialties. This article will examine two relatively new chips from SSM, the SSM2044 dedicated low pass filter and SSM2033 temperature compensated VCO.

The SSM2033. The SSM2033 is a sawtooth based oscillator with hard and soft sync capabilities, triangle converter, pulse width comparator, and full linear/exponential converter. Requiring only passive components, all inputs accept industry standard voltages and present standard impedance levels. Outputs are short circuit protected.

All active components for the linear/exponential controls are located on chip. Discrete components may offer improved accuracy, but the 0.05% typical, 0.2% maximum exponential error is more than adequate for reliable polyphonic performance. A solid-state heater/sensor combination for temperature compensation keeps drift to a maximum of ± 100 ppm/°Centigrade.

Referring to fig. 1, resistor ratios (R2 + T1)/R3 and R22/(R23 + R24) establish one Volt per octave tuning. The ratio of R23/R24 sets high frequency tracking, where R23 + R24 must equal 1k. Thus, a 1k trimpot, T3, may substitute for R23 + R24. If frequencies above 5 kHz are not needed (which is often the case), you may replace R24 with a jumper and make R23 equal to 1k. The remaining trimpot, T2, trims out frequency offsets of multiple oscillators when using several SSM2033s in polyphonic applications.

Since the SSM2033 derives its negative voltage from an on chip Zener regulating diode, its out-

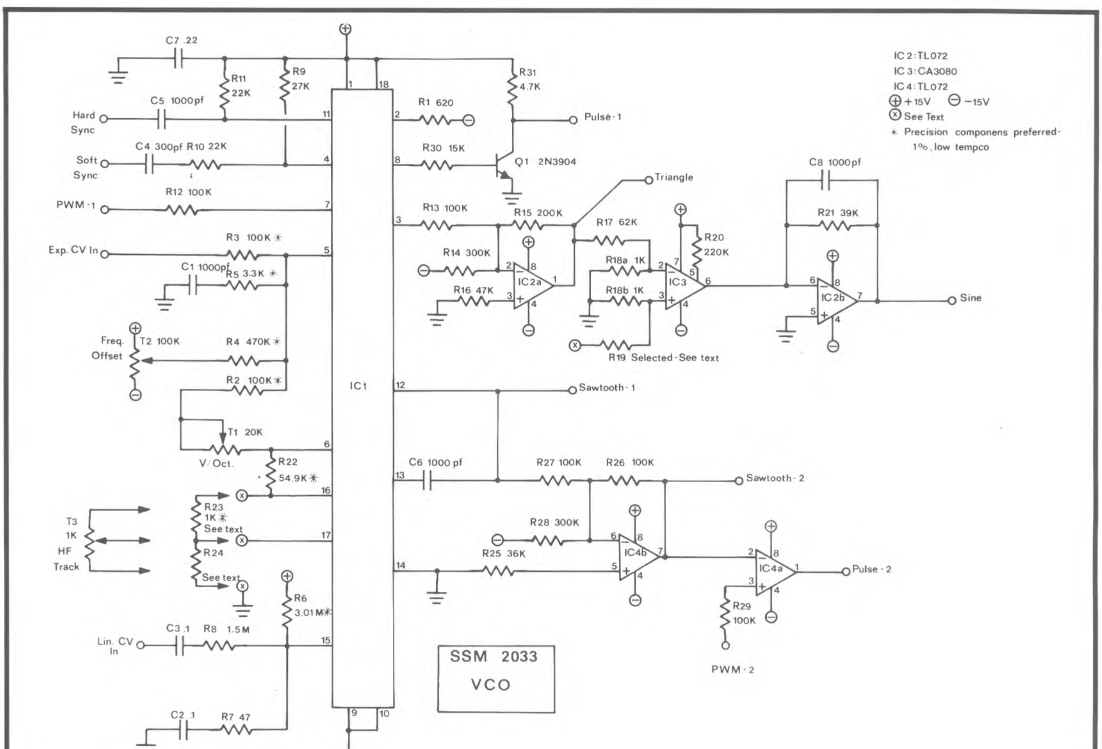


figure 1

puts are clamped to about zero Volts on negative transitions. External circuitry has been provided to level shift and amplify these signals to ± 5V standards. Thus, IC2a outputs a ± 5V triangle.

IC3 smoothes this triangle into a sine wave, buffered by IC2b. To optimize sine wave characteristics, observe the sine symmetry at IC2b's output while testing the assembled module. If the positive peak is more rounded, connect R19 to -15V; if the negative peak is more rounded, connect R19 to +15V. Use a potentiometer to determine the optimum value for R19. After establishing correct symmetry, adjust for lowest distortion by selecting R17 in the range of 50k to 100k.

Sawtooth-1 is a 0 to 10V output; IC4b converts this to a ± 5V standard. IC4a's output is a ±12.5V pulse whose width may be varied from 0% to 100% by voltages applied to R29 (PWM-2).

The SSM2033's main pulse output has been purposely inverted by Q1 for the following reasons. Note that the hard and soft sync inputs trigger on falling edges. The falling pulse edge observed at pin 8 is fixed to the discharge of the sawtooth, while the leading pulse edge varies with the PWM-1 control. If the pulse is inverted and applied to the sync input of another VCO, then the oscillators will be frequency locked at a phase angle determined by the PWM-1 control voltage. Thus, PWM-1 may be considered to be a voltage controlled phase shifter in a multi-oscillator voice. This might also be useful in an LFO system employed in location modulators.

The SSM2044. Unlike the SSM2040 and CEM3320 (filter ICs which allow a variety of complex responses), the 2044 is a dedicated 24 dB/octave low pass filter. Its accuracy, low parts count, and ease of application should attract many musicians interested in developing their own performance systems. Filter input signals are accepted and processed differentially, canceling some offset and linearity errors. The on-chip resonance circuitry also uses differential techniques. Output signals are quiet, "pop-free", and stable within the audio spectrum. Applications include electronic music low pass filters, tracking filters for analog delay lines, stable sine wave generation, and the like.

Referring to fig. 2, scaling resistors R1/R3a and R2/R3b allow the differential signal inputs to accept input levels up to the power supply limits. If using two oscillators in a voice, the output of one voice should go to one input, and the output of the second voice should go to the opposite input with a 3 dB level difference (values shown assume this type of application). Filter tuning capacitors C1 - C4 should be 1% tolerance, low temperature coefficient parts for best results and precise response. IC2b along with R10 and C5 convert the output current to a voltage.

On the evaluation boards manufactured by Global, op amp IC2a provides for differential summing. There are provisions for up to eight CV input resistors, which terminate in one PC board trace that bridges pins 2 and 3 of IC2. This trace may be broken anywhere along its length to split the eight resistors between the inverting and non-inverting inputs of IC2a. The circled x between IC2a pins 2 and 3 represents where you would break the trace.

To configure this stage so that a positive voltage increases the resonant frequency of the filter, feed the control voltage into VC-1 and ground R9c or R9d. Since pin 2 of IC2a is a summing junction, adding more 100k resistors to this point allows multiple control voltage mixing. For a response where negative control voltages decrease the resonant frequency of the filter, feed the control voltage into VC-2. This stage can also accommodate both positive and negative going control voltages by feeding them into VC-1 and VC-2 respectively.

Scaling resistors R9/(R7 + T2) and R5/R6 give a 1V/octave response, where R6 may be selected for a +3600 ppm/°C response if extreme temperature stability is required.

Since the on-chip resonance control has a reverse audio response, a CA3080 was selected to sum the resonance control voltages and approximate the proper taper for electronic music applications. Maximum attainable resonance is set by R11, and break-point for the non-linear function is set by R14/R13. Note also that IC3 is set up in a manner similar to IC2a, and can accept both positive and negative going control volt-

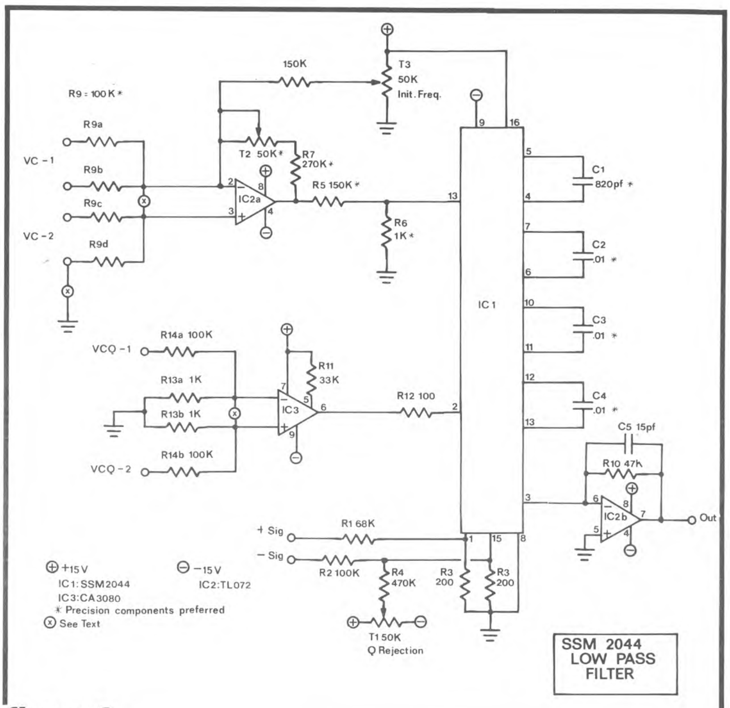


figure 2

ages; again, the input resistors connect to one trace which may be broken to assign different resistors to IC3's two inputs.

Optional "Q rejection" and initial frequency trims are available as T1 and T3 respectively.

Evaluation boards available. Global offers experimenter's/-evaluation boards for the SSM2044 and SSM2033 for $6 each ($5.25 each in quantities of 10-up). The SSM2033 IC itself is available for $10, the SSM2044 for $5.75. Terms are $2 shipping/handling on all orders, and 6% sales tax for Pennsylvania residents. Address all inquiries and orders to Global, 541 East Market St., Marietta, PA 17547.

Parts List

SSM2033 Oscillator

R1	620 Ohms
R2, R3	100k*
R4	470k*
R5	3.3k*
R6	3.01M*
R7	47 Ohms
R8	1.5M
R9	27k
R10, R11	22k
R12, R13, R26, R27, R29	100k
R14, R28	300k
R15	200k
R16	47k
R17	62k
R18a, b	1k
R19	see text
R20	220k
R21	39k
R22	54.9k*
R23	1k; see text
R24	see text
R25	36k
R30	15k
R31	4.7k
C1, C5, C6	1000 pF
C2, C3	0.1 uF
C4	300 pF
C7	0.22 uF
T1	20k trimpot
T2	100k trimpot
T3	1k trimpot
IC1	SSM2033
IC2, IC4	TL072
IC3	CA3080
Q1	2N3904

SSM2044 Low Pass Filter

R1	68k
R2, R14	100k
R3a, b	200 Ohms
R4	470k
R5	150k*
R6	1k*
R7	270k*
R8	150k*
R9	100k*
R10	47k
R11	56k
R12	22k
R13	1k
C1	820 pF*
C2-C4	0.01 uF*
C5	15 pF
T1	50k trimpot
T2	50k trimpot*
T3	50k trimpot
IC1	SSM2044
IC2	TL072
IC3	CA3080

*1%, low tempco for best stability

DETAILS: by Dennis Bohn

BASIC TRAINING

Back to basics.

The gain of the inverting amp shown in fig. 1A is -R1/R2. Big deal! Okay, I never promised a lot of excitement, but, what is the gain from point 1 to the output? And what is the input resistance at point 1? And the output resistance of the amplifier?

The answers that will get you into trouble are "one", "zero", and "zero", respectively.

How about figure 1B? What is the input resistance at point 1? And the output resistance?

To answer these questions, we need to back up a bit and review the concept of "loop gain". Loop gain is the difference between the closed loop (feedback in place) gain of a circuit and the open loop (no feedback) gain expressed in dB. For example, a circuit with a closed loop gain of 20 dB and an open loop gain of 80 dB has a loop gain of 60 dB. Loop gain is commonly given the symbol T, and it is the magic stuff which makes feedback amplifiers a wonderment.

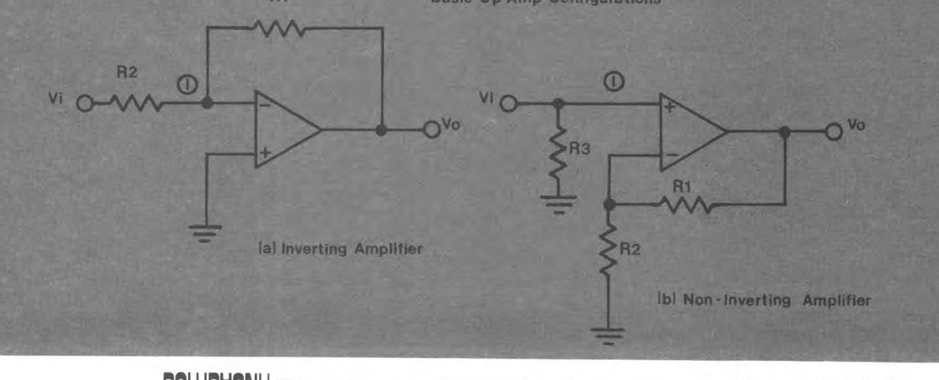
It is loop gain that reduces distortion and output impedance and either increases or decreases input impedance. Loop gain is the glue which keeps feedback amplifiers from coming apart when hit by complex audio waveforms. And it is loop gain, or rather, lack of loop gain, that will sneak up and bite you on the ankle just when you think that your circuit is working perfectly.

The problem comes about from the frequency dependent nature of loop gain, i.e., it is not constant over frequency; it falls off at a 6 dB/octave (20 dB/decade) rate for well designed, stable op amps. Each type of op amp has a different open loop characteristic and the data sheet must be consulted in order to know how much loop gain is available for a particular frequency. If a data sheet is not available to you and you happen to know, or can find out, the unity gain frequency of the device, then you can use this information to calculate the open loop gain at any frequency. This is the concept of Gain Bandwidth Product (GBW) and is most useful. The unity gain frequency of any op amp is the frequency where the open loop frequency response is unity (how's that for double talk?). The GBW is therefore this same number, since it is the produce of one times this frequency. For example, an LF351 has a unity gain frequency response equal to 4 MHz, hence, GBW = 4 MHz. Now, to find out how much open loop gain there is at, say, 10 kHz, all you do is ivide the GBW by 10 kHz. The answer is 400 V/V or 52 dB; therefore, an LF351 has an open loop gain of 52 dB at 10 kHz. And operated at a closed loop gain of, say, 20 dB (times 10) then it has a loop gain, T, equal to 32 dB (52 dB - 20 dB = 32 dB). This, then, is how much "stuff" is available to fix distortion, impedances, etc.

Now, back to the original questions: The gain from point 1 to the output for the inverting configuration of fig. 1A is the available open loop gain at the frequency of interest. If not totally understood, this can come as a big shock in using inverting amplifiers. Even though the closed loop gain from Vi to Vo may be unity, the gain from the input pins is always whatever open loop gain is available, times the differential voltage present between the plus and minus inputs.

A practical example of where this can be a real problem is in mixer designs. Point 1 becomes the main summing node for all of the mixer inputs and because of the real world truths of mixers, it is generally a long piece of bus wire strung between each input module, sometimes reaching a length of several feet! Now, anything unwanted induced into

Fig.1
Basic Op Amp Configurations



this node (hum, spikes, noise, RFI, etc.) gets amplified by the open loop gain...which is why many designers pull their (already gray) hair out in trying to produce really quiet mixing nodes.

So, how can you ever live with this characteristic? This brings us to the second question of what the input impedance is at point 1. It is very low, but it is not zero, and it is frequency dependent. For practical purposes, it is the feedback resistor, R1, divided by the open loop gain, T. So at DC, it can be very low, but it rises at the same 6 dB/octave that the open loop response falls off. Back to our example of the LF351 operating with a closed loop gain of 20 dB: Assume R1 equals 100k Ohms, then the input impedance at 100 kHz is 100k Ohms divided by T. T for 100 Hz equals a factor of 4000 (72 dB), so the input impedance is 25 Ohms - not a big problem. But, at 10 kHz this impedance rises to 2500 Ohms, since T is now only 40 (32 dB) - a much bigger problem. And if R1 was large, say 1 Meg, then the impedance at 10 kHz would equal 25k Ohms - definitely a problem! Of course, going the other way can save your bacon. Let R1 = 10k Ohms, then the impedance at 10 kHz is only 250 Ohms.

What about output impedance? For both configurations of fig. 1, it is the open loop output impedance divided by 1 + T. So again, it is very low, but has a rising frequency dependent characteristic. For the LF351 it starts out at around 20 milliOhms for 100 Hz and rises to about 2 Ohms at 10 kHz. (Output impedance of op amps turns out to be a very complex subject, and is not well behaved at all. Rarely does it follow a nice 6 dB/octave curve, but is usually stays low enough at all gains and frequencies so as not to be a problem. If output impedance is of critical interest to your circuit's behavior, then the exact data sheet must be consulted very carefully.)

Our last unanswered question concerns the input impedance of point 1 of the non-inverting configuration of fig. 1B: It is resistor R3 in parallel with the input impedance of the op amp. The input impedance of the op amp is the open loop input impedance times 1 + T. So, it starts out very large and drops at the 6 dB/octave rate. For Bi-Fet op amps such as the LF351, it never drops to a level of real concern unless you are using extremely large values for R3 (like 10 MegOhms, for instance). However, for bipolar input op amps, like the super quiet NE5534, you must keep an eye on things, or surprises may be in order - particularly at higher gains. The open loop input impedance of an NE5534 is only 100k Ohms (typically, guaranteed is only 30k!) compared to 10^{12} Ohms for Bi-Fets. So, for closed loop gains of 40 dB, the input impedance at 100 Hz is 100 Meg, but drops to 1 Meg at 10 kHz.

Figure 2 gives a summary of feedback amplifier parameters and how they affect certain operating characteristics. Figure 3 compares three popular op amps for your reference.

There, you ate your spinach. Now you may have dessert!

Fig.2
Summary of Feedback Amplifier Parameters

Amplifier Type	Input Impedance	Output Impedance	Harmonic Distortion	Noise Gain	Bandwidth (closed-loop)
Non-Inverting	$(1+T)\,R_{in}$	$\frac{R_o}{1+T}$	$\frac{THD}{1+T}$	A_{VCL}	$\frac{GBW}{A_{VCL}}$
Inverting	$\frac{R_f}{T}$	$\frac{R_o}{1+T}$	$\frac{THD}{1+T}$	$A_{VCL}+1$	$\frac{GBW}{A_{VCL}+1}$

A_{VCL} : closed-loop gain
GBW : gain bandwidth product : unity-gain frequency
R_f : feedback resistor
R_{in} : open-loop differential input impedance
R_o : open-loop output impedance
T : loop gain
THD : open-loop total harmonic distortion (%)

Fig.3
Op Amp Comparison

Device	Voltage DC	Gain (dB) 10kHz	Input Res (ohms)	Output (4) Res (ohms)	Loop Gain (5) (1+T, V/V)	R_{in} (5,6) (closed)	R_{out} (5) (closed)
5534 (3)	100	67	100	22	72	7.2	0.3
LF351	100	52	10^{12}	75	14	14×10^{12}	5.4
TL071	106	50	10^{12}	200	11	11×10^{12}	18

Notes:
1. Commercial grade, plastic pkg, TA = 25°
2. All data is open loop and typical values unless otherwise specified.
3. Compensated for unity gain (22pF)
4. See text.
5. Closed loop gain = 30dB; Frequency = 10kHz
6. Op Amp input resistance; in parallel with network resistor effects.

Remote Switching for DOD 500 Series Processors

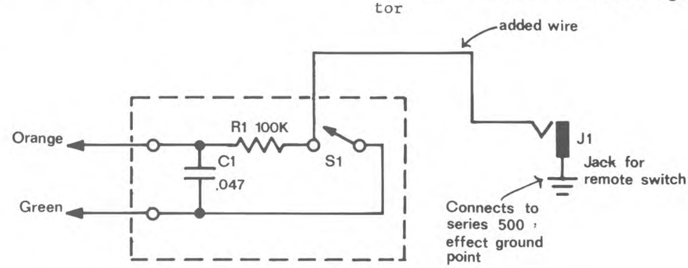
By: David DiFrancesco, DOD ELECTRONICS CO.

Remote switching is useful in several applications. One is pedalboard systems, where you might want to have the effect at arm level (for ease of adjusting the controls) while having the status footswitch located on the floor. Another is for electronic switching systems, where you might want to control an effect's status with logic signals.

It's easy to add remote switching to all DOD 500 series products, since each one uses a momentary, normally open switch to change the status of the effect. Simply putting another momentary switch in parallel with this original switch gives remote operation. Because the footswitch only switches a DC control line, there are no hum problems with adding the remote switch.

Figure 1 shows the circuitry located on the switch board, which is underneath the main circuit board. You can add J1 by drilling a hole opposite the LED position on the side of the chassis, and run a wire from the tip of the jack to the junction of S1 and R1 on the switch circuit board.

That's all there is to adding remote operation. But while we're at it, here's one more tip for 500 Series devices: if you already have a DOD 200 power supply, you can use it to power several 500 Series effects by simply the replacing the DOD 200's LM78L10 regulator with an LM78L15 regulator



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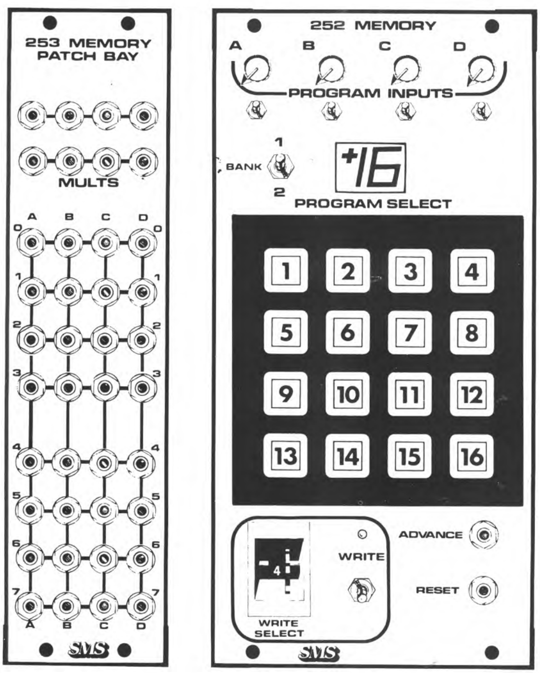

½ ACTUAL SIZE

Instantly set your pitches, filters or any other voltage controlled functions.

The 252 Memory System is designed to provide instant presetting of music synthesizers. It will dramatically increase the usefulness of your modular (or custom) music system by turning it into a programmable system with performance capabilities.

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Many systems may not have all voltage-controlled functions so some auxiliary modules may be required. For example, to implement programming of an LFO mod depth a VCA would be required between the LFO output and the input to be modulated (VCA cv in would come from the programmer and thus control the depth of modulation). The 215 Quad VCA and 256 VC Dual ADSR's are well suited for use with the 252 Memory System.

SPECIFICATIONS

Outputs: 32 control voltage outputs, fully buffered for systems use, output range is ±5.3V (0 - 10 available at higher cost, but is usually not necessary), quantitization level is .041V or ¼ step resolution over 10 octaves at 1V/oct.

Advance: Counts 1-16 only, will not count through bank 1/2.

Power Requirements: Either ±18 or ±15V (specify if you are supplying power, ±18 is to power optional on-module regulators) AND ±12 (nominal) unregulated.

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VelocI-Touch™ Controller

WHAT DOES IT DO

The new Veloci-Touch controller from PAIA adds what may be the two most important parameters that any electronic keyboard can have — velocity and second touch pressure sensitivity.

Velocity is a control voltage proportional to how hard you play. Pianissimo and the voltage is low. Fortissimo and it's high. Use this parameter with a VCA and presto . . . output level changes that follow your playing.

But, since "loud" and "soft" mean more in human terms than just level changes, the Veloci-Touch controller also provides a velocity sensitive transient generator with variable decay time and an output control that's continuously variable from normal to inverted transients. Use as a filter parameter control for the most natural timbral changes ever.

Even that's not all. The Veloci-Touch controller also provides an output proportional to the pressure you apply to the key after it's down (second touch). Imagine — tremolo the natural way.

The best part of this minor miracle from PAIA is that it's non-denominational. You can retro-fit it to essentially any synthesizer from any manufacturer.

HOW DOES IT WORK

The secret of the Veloci-Touch controller is a thin piezo-electric transducer (called a bender) which slips under the keyboard and in effect "listens" to how hard you play and how much second touch pressure you apply. The electronic processing of the controller uses the synthesizer's gate signal and other cues to separate the velocity transient from the second touch pressure and provides independent outputs for both.

IS IT POLYPHONIC

The Veloci-Touch controller does not provide an individual output for each key in a polyphonic system, but is still useful in this application. Even a single output for velocity and pressure is a major improvement over no touch sensitivity at all.

HOW DO I INSTALL IT

Installation of the transducer is surprisingly straight forward. In most cases it is simply a matter of removing or loosening the screws that hold down the front edge of the keyboard, slipping the bender under the frame and re-tightening the screws.

Both stand-alone and modular packaging is available for the electronics. The 8786 package illustrated includes a line-operated power supply, trim contemporary case and provides standard ¼″ jacks for control voltage outputs.

Also available is the 4786 configuration which is mechanically and electrically consistent with PAIA 4700 series modular equipment.

In some instances, the front panel supplied with the 4786 may be used as a bezel or discarded and the 4″ x 4½″ circuit board containing the Veloci-Touch processing electronics mounted inside the instrument with the ±9v. to ±15v. power required by the V-T tapped from the keyboard's own supply.

CAN I USE IT WITH MY (your axe here)

One of the most over-worked lines in advertising is "limited only by your imagination". It should come as no surprise that it is most often used when the person writing the copy has no idea how you use it. In this case, though, it may be a valid observation because while the V-T's uses with synthesizer ar fairly obvious, many people will find creative ways to use it with combo organs, electric pianos and other instruments. Even when an instrument already has velocity sensitivity capabilities, second touch for tremolo or vibrato can be very handy.

ARE THERE ANY "YEAH....BUT's"

Not really, but remember that you are installing the V-T equipment to produce a more expressive instrument and this implies the development of a technique that realizes that potential. While you can get really extraordinary "effects" the first time play a V-T'd keyboard, the technique required for real control only comes with practice.

For example, it is more difficult than you would first guess using the second touch pressure for pitch bends that precisely hold a note, partially because the controller circuitry does not hold a constant output for constant pressure but rather decays away with time. But with a little practice you quickly develop a feel for how much you must increase pressure to hold a bend. For most, a little constructive play will yield extremely gratifying results.

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