Electronic Music Circuits: The Reprints

— Volume I —

A collection of 38 articles from the pages of *Polyphony*, *Electronic Musician*, *Common Mode*, *Device Newsletter* and *Electronotes*

by

Thomas Henry

MIDWEST ANALOG PRODUCTS

Kits, Parts and Plans for the Do-It-Yourself Electronic Musician

Electronic Music Circuits: The Reprints
Volume I

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About this Book

The items in this collection have been reprinted more or less as they originally appeared. Every now and then there will be a reference to a price, a kit or an address. This might occur in the article proper, or possibly in some incidental material appearing elsewhere on a page. Please be aware that with the passage of years since the articles first appeared, most of these references are no longer accurate! Prices might have changed, companies may have moved or gone out of business, and so on. I've left these references in simply for historical interest.

Now remember, these reprints span nearly two decades and represent a bit of a pilgrim's progress. Some of my earlier designs, though still interesting, do show a relative lack of experience; I would probably do them differently today. But what can I say? I was a young pup then!

As far as I can remember, there is only one typo to report. This occurred in the schematic of "Superseque: A Full Featured Analog Sequencer." It's easy to fix; simply exchange the plus and minus signs which designate the inverting and non-inverting inputs of op-amp A1. Notice that the pin numbers (13 and 14) are correct as shown.

The Table of Contents arranges the articles in chronological order by magazine, which are themselves arranged alphabetically. If you're looking for a complete citation, be sure to examine the Bibliography on page 158. This lists all of the articles, again in chronological order, and gives the full publishing details.

If you're new to the electronic music world, I'm willing to bet that some of the magazines mentioned here will really whet your appetite to build up a collection. *Device Newsletter* and *Common Mode* are no longer with us and copies of these periodicals are quite rare now. But occasionally you will run across a person trying to sell off some duplicates. The Internet is a good place to initiate a search for these. To get started, you might want to check out the WWW site of Midwest Analog Products which contains a wealth of helpful information:

Midwest Analog Products http://prairie.lakes.com/~map

As of this writing, PAiA Electronics, Inc., still has some back issues of *Polyphony* and *Electronic Musician* available. Write for a catalog, or explore their WWW site to see what's in stock:

PAiA Electronics, Inc. http://www.paia.com
3200 Teakwood Lane
Edmond, OK 73013

Back issues of *Electronotes* are still available. To obtain a list, send a stamped, self-addressed envelope directly to its publisher at:

Electronotes
1 Pheasant Lane
Ithaca, NY 14850

Sadly, some of the integrated circuits and other parts mentioned in the articles are no longer commonly available. Nonetheless, you'll still be able to get some great ideas from the circuits that call for them. In many instances, new parts may be substituted with only slight changes to the designs. Finally, it's not unheard of to bump into someone on the Internet who is clearing out a small stash of hard-to-find parts. For example, I've seen some great deals on Curtis and SSM synthesizer chips pop up from time to time, so keep your eyes open.

If this collection proves popular, then a second volume might be possible. There's plenty more material to choose from, so it all depends on the public (that's you!) response.

Happy reading...

Acknowledgments

It's hard for me to realize that I've been penning articles on electronic music circuit design for over 17 years now! During that time, I've had the good fortune to work with some of the best publishers and editors in the business. Not only did they help me get started in this fascinating profession, offering all sorts of help, encouragement and suggestions, but they also became good friends. When it came time to put this collection of reprints together, they all generously gave their "okays." Let me acknowledge them here.

I'll start by thanking Craig Anderton for permission to reprint my *Device Newsletter* articles. Craig edited and published this wonderful periodical for guitarists during its one year run. My very first article saw the light of day in *Device Newsletter*.

John Simonton of Polyphony Publishing Division of PAiA Electronics, Inc., graciously granted permission to reprint my *Polyphony* and *Electronic Musician* articles. Let me hasten to clarify any possible confusion here. The original name of this publication was *Polyphony*, but in June 1985 it became *Electronic Musician* to reflect a wider range of coverage. Still later, in January 1986, *Electronic Musician* was sold to Mix Publications. The articles reprinted in this collection are from the pre-Mix days.

One of my favorite persons in electronics has always been Bernie Hutchins. Besides being a top notch researcher, writer and editor, Bernie is also the publisher of the unique *Electronotes* newsletter. I am grateful to him for permitting me to reprint my articles here.

PGS Electronics got me started in the hobbyist kit side of things. Under the head of Greg Schneck, this neat company (now gone, but still missed) published one of my circuits in its magazine *Common Mode*. I thank Greg for letting me share this article with you.

In the Acknowledgments to my *Electronic Drum Cookbook*, I said:

> "Do you get the feeling that the field of electronic music circuit design is populated with some wonderful people, or that I've been blessed with fabulous friends? So do I, and I thank them all!"

It's still true. I thank Craig, John, Bernie and Greg not only for their permission to reprint these articles, but also for their support and wise counsel over the past 17 years.

Thomas Henry
North Mankato, Minnesota
January 1997

A Note Concerning the Format...

Arranging this material turned out to be a much more fiendish task than I had first imagined! The main problems were where to position the new page numbers, and how to physically place the material to leave room for the plastic comb binder. Many of the older articles were originally printed on pages with very small margins, so this required compromises from time to time. (Incidentally, the page numbers appear on the upper outside corners. This is a bit non-standard, but was really the only feasible place to keep them from running into the text.)

Also, some of the magazines are nearly two decades old and are getting a bit tattered and yellow now. Because of this, you might notice wandering margins, faded ink and even an occasional solder splatter here and there. Despite these minor imperfections, I hope you will find this material useful in your own work.

BUILD THE PGS LOWPASS VOLTAGE CONTROLLED FILTER

by Thomas Henry

The PGS Lowpass Voltage Controlled Filter is configured around the SSM2040 integrated circuit. Use of this IC greatly simplifies construction and leads to a very reliable operation. Among the many features of the VCF circuit are the following:

* Wide sweep range, typically 10 octaves.
* Voltage controlled resonance.
* All input impedences are greater than 50K.
* Output impedence is 1K.
* 10v pp signal levels.
* Fully temperature compensated.
* Four pole response.
* Able to oscillate in a pure sine wave.
* Standard power supply voltages of plus and minus 15v.

UNDERSTANDING THE VCF

The heart of this whole circuit is, of course, the SSM 2040 integrated circuit. Essentially, this IC contains four voltage controlled amplifiers, all sharing a common control stage. The response is already exponential; all we have to do is provide some RC tuning elements and the circuit is ready to go!

Refer to the schematic in figure one. C3 through C6 are the tuning capacitors. These capacitors should be polystyrene for maximum temperature stability. Resistors R10 through R18 round out the RC network. These capacitors and resistors then, are the components which set the basic tuning of the circuit. With these values, the circuit is perfect for audio synthesizer work. Resistors R1 through R4 have nothing to do with the tuning. It is their duty to attenuate the incoming signal to meet the operating requirements of the SSM2040. With the value as shown (200 ohms), distortion is minimized.

The audio input is applied to J3. This jack is fed to an attenuator, R39, and this allows the user to reduce the amplitude of any incoming signals. A feature such as this is often needed when applying a mixer output to the VCF. A mixed signal, composed of several signals could easily exceed the filter's nominal 10v pp input. Amplifier A1 and its associated circuitry attenuate the input by a factor of ten. This must be done since the SSM2040 likes to see signal swings of 1v pp.

Since the signal has been attenuated, the output must be boosted to compensate. Amplifier A3 and its associated components see to this function. Note also capacitors C1 and C2. These are selected to roll off the high end response (up around 100 KHz) to avoid any undesired radio frequency interference and, more importantly, spurious oscillation.

Part of the output is fed back to the input via IC4, a 3080 transconductance op-amp. This provides the voltage controlled resonance function. The 3080 is set up as a standard voltage controlled gain stage. R34 and R6 chop the input signal down to allow the amp to work in its linear range. Trimmer R42 is adjusted to minimize any "thumps" caused by a DC offset being fed through the OTA.

A4 is configured as a current to voltage converter. This is needed since the OTA is a current output type device. Trimmer R20 is adjusted so that a maximum voltage generated by R41 just sets the filter oscillating. By setting this trimmer so that the filter justs begins to oscillate, clipping can be avoided and the net result is a very pure sine wave output.

Amplifier A5 forms the control voltage input stage. A5 and Q1 actually perform a voltage to current conversion. Since pin 5 of IC 4 expects to see a control current, this conversion is necessary. D1 is in the circuit simply to protect Q1 from reverse voltage conditions. This insures that the circuit is "goof-proof". The actual voltage control input is at J4, and the voltage thus applied may be attenuated by R40. R41 allows a manual adjustment of the resonance.

Amplifier A2 is set up as a summing amplifier and provides the frequency control for the circuit. J1 is the 1 volt per octive input. This input would normally be used for a tracking keyboard control. The 1 volt per octave response can be tweeked for extreme accuaracy by R22.

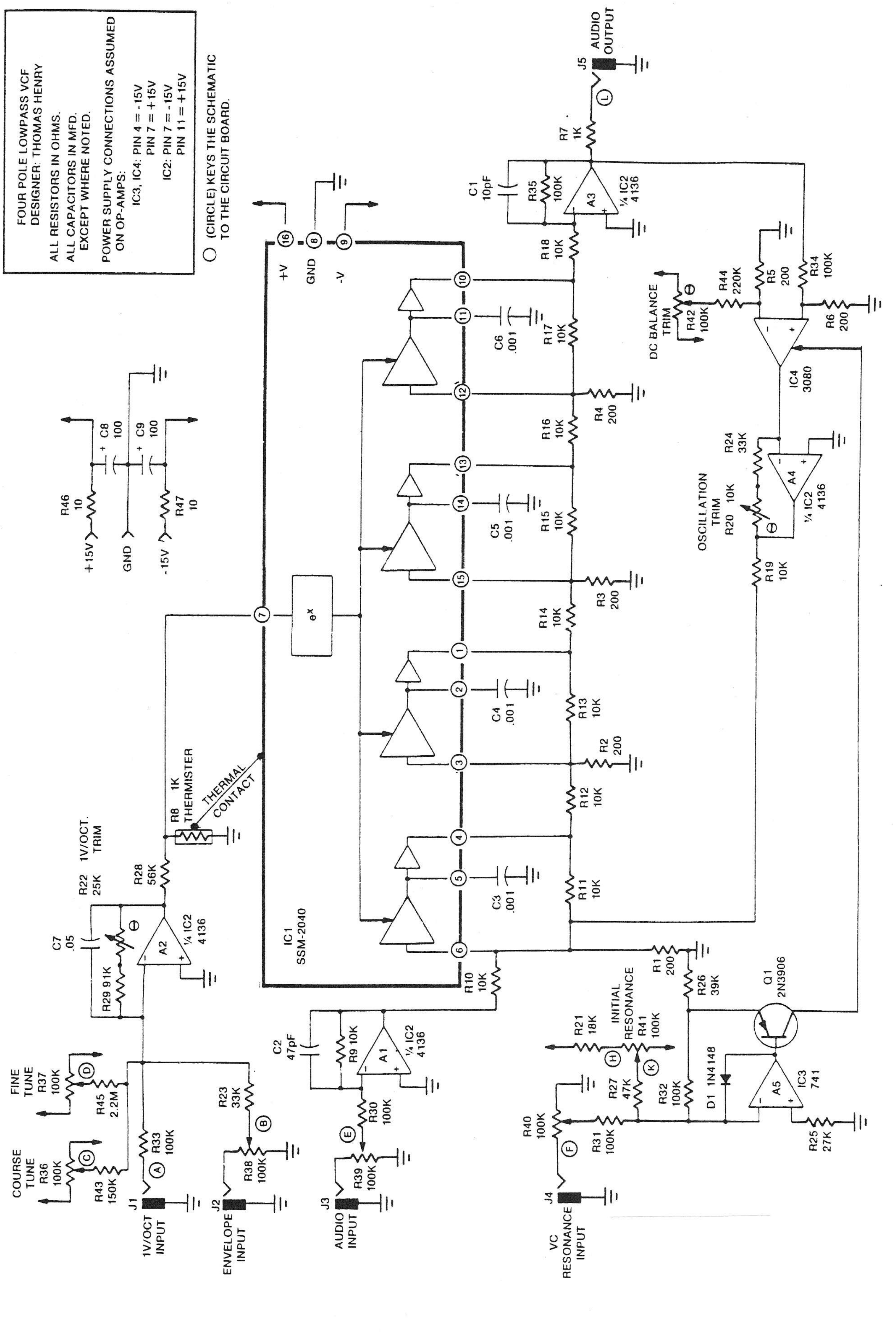
R36 and R37 are the coarse and fine tuning controls. The coarse tuning control covers a range of about twelve octaves, while the fine tuning control covers a range of less than one octave. These controls are especially handy when using the filter in the oscillation mode.

"I have had two copies of this circuit in my system for over two years now and I still marvel at the quality of sound possible with it."

J2 is the envelope input jack. Since envelope signals are usually 5v or less, this input has been given a gain of three. Thus, a positive 5v input can cause the filter to sweep across its entire range. C7 is put into the feedback loop of A2, and in this configuration helps minimize "pops" caused by modulating the filter's cutoff frequency rapidly.

To provide total temperature compensation, R8, which is actually a thermistor, is included in the circuit. This component will automatically compensate for any temperature drift. The result is a filter which will stay in tune, no matter what the playing environment may be like. Since this filter may be used like a VCO, (by setting it oscillating), temperature compensation is definitely a must.

Rounding out the circuit, power is provided by a bipolar (plus and minus) 15 volt supply. The supply is decoupled by R46-R47 and C8-C9. continued.....



FOUR POLE LOWPASS VCF
DESIGNER: THOMAS HENRY
ALL RESISTORS IN OHMS.
ALL CAPACITORS IN MFD.
EXCEPT WHERE NOTED.
POWER SUPPLY CONNECTIONS ASSUMED ON OP-AMPS:
IC3, IC4: PIN 4 = -15V
PIN 7 = +15V
IC2: PIN 7 = -15V
PIN 11 = +15V
(CIRCLE) KEYS THE SCHEMATIC TO THE CIRCUIT BOARD.
+15V
GND
-15V
R46 10
R47 10
C8 100
C9 100
IC1 SSM-2040
R8 1K THERMISTER
THERMAL CONTACT
R22 25K 1V/OCT. TRIM
R28 56K
C7 .05
R29 91K
A2 ¼ IC2 4136
COURSE TUNE R36 100K
FINE TUNE R37 100K
R43 150K
R45 2.2M
R33 100K
R23 33K
R38 100K
J1 1V/OCT INPUT
J2 ENVELOPE INPUT
C2 47pF
R9 10K
A1 ¼ IC2 4136
R10 10K
R30 100K
R39 100K
J3 AUDIO INPUT
C3 .001
C4 .001
C5 .001
C6 .001
R11 10K
R12 10K
R13 10K
R14 10K
R15 10K
R16 10K
R17 10K
R18 10K
R1 200
R2 200
R3 200
R4 200
+V
GND
-V
C1 10pF
R35 100K
A3 ¼ IC2 4136
R7 1K
J5 AUDIO OUTPUT
DC BALANCE TRIM
R42 100K
R44 220K
R5 200
R34 100K
R6 200
IC4 3080
R24 33K
A4 ¼ IC2 4136
OSCILLATION TRIM
R20 10K
R19 10K
R26 39K
Q1 2N3906
INITIAL RESONANCE R41 100K
R21 18K
R27 47K
R32 100K
D1 1N4148
A5
IC3 741
R25 27K
R40 100K
R31 100K
J4 VC RESONANCE INPUT

Figure 1

BUILDING THE LOWPASS VCF

Building the PGS Lowpass VCF is not difficult at all, as long as you follow the instructions carefully. But remember, the SSM2040 is an expensive chip, so utilize extreme care at all points of the assembly.

Start by reading this text over several times. After reading once to get the general idea, read it again, this time for details. (Ed note: Also study and learn the schematic as you read. This will not only help in the building of the circuit, but will also be quite helpful should you experience any difficulties and find that you must do some troubleshooting.) You are now ready to build.

(If you are making your own board you will of course have to do this first. A full size pattern is shown in Fig. 2. While the pattern may not be good enough to use the "lift off" method for pulling it off of the page you can make a good positive by laying a clear sheet of acetate over the pattern and using drafting donuts and tapes to make your own "positive". Once your board is etched and drilled you are ready to go on. -Ed.)

Before loading the circuit board you will want to fabricate a front panel, box, or chassis for the unit. Do this first. Drill any mounting holes needed in the circuit board, but be careful not to damage the board in any way.

At this point you can begin loading the board. Scrub the copper side of the printed circuit board with a fine scouring pad (not soaped) or if you are careful, with a piece of 000 steel wool. Be careful not to remove any excess copper.

( ) Start by installing the four IC sockets. A piece of masking tape can be used to hold a socket in place while soldering. Tape the socket down to the parts side, then flip the board over and proceed to solder. Watch for excessive heat which could lift pads, and beware of solder bridges. Next, install the resistors sequentially. Save the excess clippings for use as jumpers. Do not install the thermistor, R8, just yet.

( ) Install the three trim pots.

( ) Now install the capacitors. Observe the polarity of the electrolytics carefully.

( ) At this point you may install the diode and transistor. Be sure to watch the orientation of both of these. A black band generally marks the cathode on diodes. For the transistor, with the flat side facing you, pins sticking downward, the orientation is usually from left to right: emitter, base, collector. (These two components are especially sensitive to heat. Be careful not to overheat them while soldering. -Ed.)

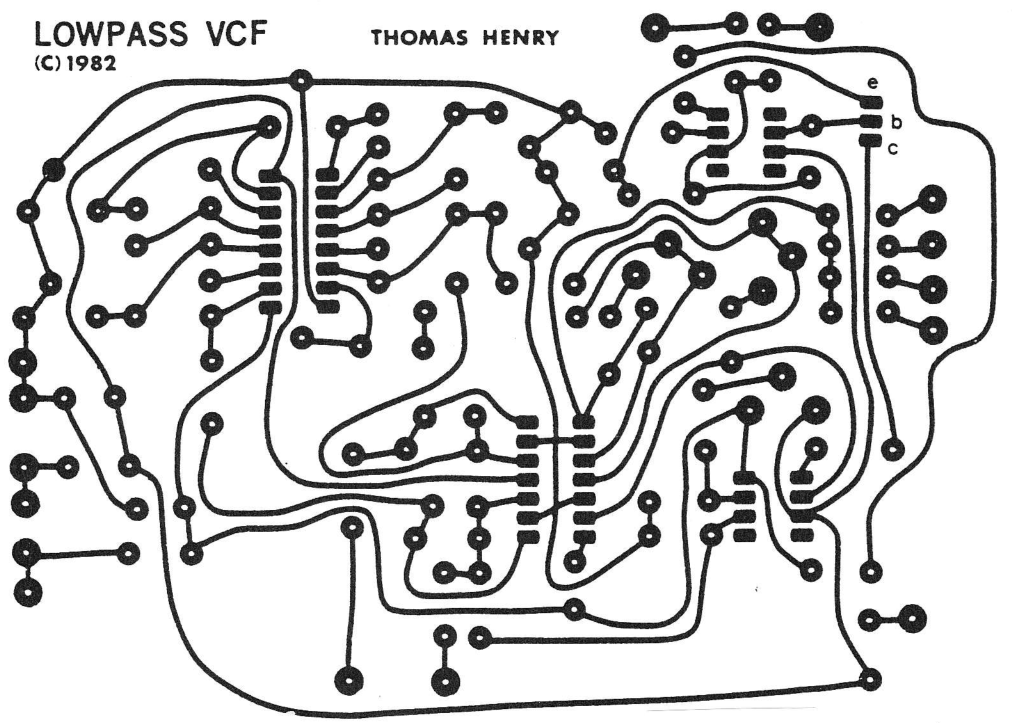
( ) Using the excess resistor clippings, form and install the jumpers. Jumpers are indicated on the parts placement guide by straight lines with the letter "J" in the center.

( ) Now is the time to mount R8, the thermistor. First slip IC1 into its socket. Now look at the circuit board. There is a hole right below IC1 and another one right above R14. These are the two holes which will be used. Now refer to Figure 3. Note how the thermistor is mounted lengthwise along the top of IC1. You may also notice some silicon heatsink grease in the drawing, between the IC and the thermistor. This is optional, but

continued.....

Figure 2

Circuit Pattern

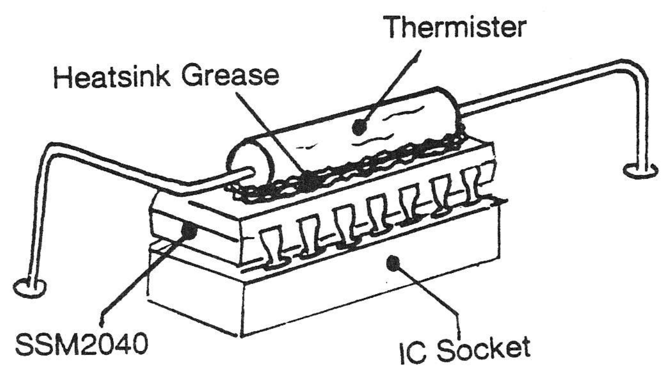


For your convenience, a parts kit is available for this project. For more information see the "parts list" box on page 14.

its use greatly increases the thermal bond between the two parts. At this point pick up R8 and bend the leads to fit squarely in the two holes described above. Apply heatsink grease if desired, and then solder it in its proper location. (Be sure that when you solder the thermistor in place it is held securely against the IC in some mannor (But not with your finger! It will get hot.) so that good contact is made.)

( ) Now set the circuit board aside and return to the front panel. Install the pots and jacks. You may string the common ground lines up at this point if you wish. Next, connect the circuit board to the front panel using small angles, standoff, etc. Wire the circuit board to the panel using the schematic as a guide. The circled letters key the schematic to the circuit board. Letters G, I, J, M and the remaining alphabet are not used

( ) Double check your work now. Examine the circuit board carefully. Remember, the SSM2040 is NOT short circuit proof and the chip could be destroyed very easily by a random solder bridge. So please look everything over. Are there any inferior solder joints, solder bridges or shorted connection wires? Check the wiring to the panel; is everything right?



Thermister orientation.

Figure 3

( ) If your answer is "yes" to the above, then install the rest of the IC's in their respective sockets. Be very careful to note the proper orientation on the parts placement guide. Also be careful not to break or bend the IC pins.

( ) Now install the power supply lines. Do this by soldering a length of wire to the points on the board marked + 15, GND, and -15. Now run each wire to the appropriate termianl on your bi-polar power supply or, if installing the unit in an existing system, to the power supply bus. Do not mix up ground, + 15v, or -15v lines!

( ) You are now ready to tweak up the trim pots.

ADJUSTING THE TRIM POTS

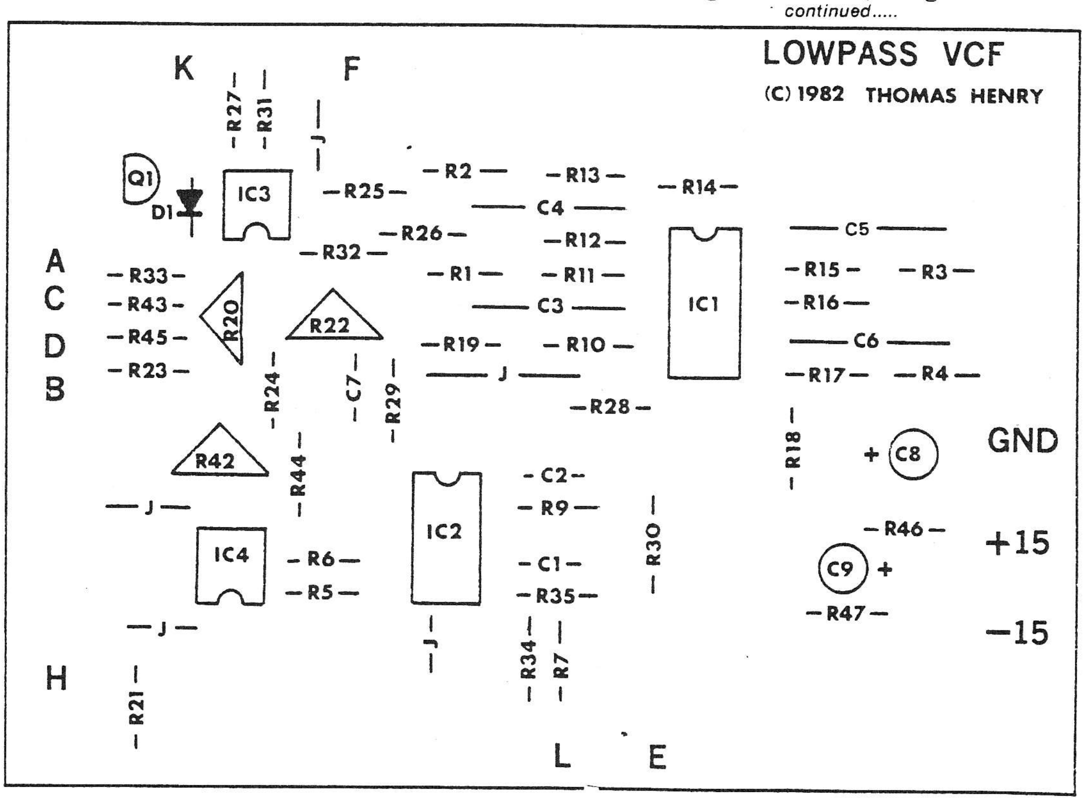
First we will adjust R42, the DC balance trimmer. Follow these instructions carefully.

With a DC voltmeter, measure the voltage on the wiper of R41 and adjust R41 for 0v. Now turn down the audio input control, R39. Apply a 10v pp, 2 or 3 Hz signal to J4 (voltage controlled

continued.....

Figure 4

Parts Placement Diagram



resonance input) and turn up R40 (voltage controlled resonance attenuator) all of the way. You may use a function generator, VCO, or an LFO for this function. The frequency is not critical. We merely want to create some sort of alternating signal at the input to the voltage controlled resonance stage.

Monitor the DC voltage at pin 4 of IC2. Adjust R42, the DC balance trimmer, for the least deflection of the meter.

Now we will adjust R20, the oscillation trim. This control affects at what point the filter will break into oscillation. First, turn down R39, the audio input attenuator. Also turn down R40, the voltage controlled resonance input. Turn R41, the manual resonance control, up full (positive).

Now connect the output of the filter to an audio amp, and set it to a comfortable listening level. Adjust trimmer R20 until oscillation JUST begins to occur. It may be necessary to turn the coarse tune pot, R36, up or down a bit to get the filter oscillating in the human ear's range. If R20 is advanced too far, the unit will still oscillate, however the wave form will not be a "pure" sine wave and will start to lose symmetry. So be sure to adjust R20 to the point of oscillation and no further.

Now we are going to "tune" the filter. Essentially we will start the filter oscillating and then tune it like any VCO. First, turn down the audio input attenuator, R39, and turn up R41, the initial resonance pot. Thiswill start the filter oscillating. Now connect a keyboard to the 1 volt per octive input at J1.

"You will find that the filter is exceptionally easy to use."

Adjust the coarse and fine tuning controls to put the oscillation in a mid-range (500 Hz to 1000 Hz). Now while alternately playing a C and an octive above that C, adjust R22 (the 1 volt per octive trim) until a one octave span is heard. Keep alternating between the two C's and listen for the octive interval.

There are other ways to adjust R22 using test equipment, but the method above is more than accurate enough for any reasonable musical application. However, for the purists in the crowd, refer to a method detailed by Bernie Hutchins in ELECTRONOTES No. 75, page 5.

IN CASE OF DIFFICULTY

Experience has shown that malfunctioning circuits are almost never due to IC problems. Human error is more often than not the culprit. If you are having trouble, follow these steps.

1. Get a handle on the problem. Write down all of the symptoms. You can't fix something unless you REALLY know what's wrong.
2. Apply logic. Starting from the symptoms, narrow down the problem to the logical function. Do not start looking for details right away; look for the general area. Examples: no oscillation at any setting - problem must be in the resonance stage.
3. Now start looking for the dubious component. First check all solder joints. If necessary, remelt the solder and try again. Now look for any suspicious physical damage to components. Finally, check wires going to and from the panel.
4. In a circuit of this nature (lots of pots) be sure something's really wrong before tearing into it. Remember, some combinations of controls may yield odd effects.
5. The circuit is fairly simple, so the only test instruments you really need are your eyes, your ears, an insulated probe (such as a pencil) and possibly a multimeter.
6. Above all, remember that there is no such thing as magic in electronics. Any problem must follow logically from a cause. This is an immutable rule!

USING THE VOLTAGE CONTROLLED FILTER

You will find that the filter is exceptionally easy to use. Since there are quite a few controls and jacks available to the user, many different patches are possible. To get you started, here is how to do a standard "wah-wah" synthesizer patch.

First, consider the audio path. Patch the output of a VCO to the input of the VCF (J3) and then take the output of the VCF (J5) and connect this to the input of your VCA. If your VCO signal is 10v pp or less, you may open the audio input attenuator, R39, all the way.

Now connect the keyboard gate and trigger to an ADSR ans patch the output of the ADSR to the VCA and to

continued.....

Parts List

RESISTORS

R1-R6	200 ohm
R7	1K
R8	1K thermistor
R9-R19	10K
R20	10K trimmer
R21	18K
R22	25K trimmer
R23-R24	33K
R25	27K
R26	39K
R27	47K
R28	56K
R29	91K
R30-R35	100K
R36-R41	100K pot
R42	100K trimmer
R43	150K
R44	220K
R45	2.2M
R46,R47	10 ohm

CAPACITORS

C1	10pF
C2	47pF
C3-C6	.001 polystyrene
C7	.05
C8-C9	100 mfd electrolytic

SEMICONDUCTORS

IC1	SSM2040 VCF
IC2	4136 quad op-amp
IC3	741 op-amp
IC4	3080 OTA
D1	1N4148 or 1N914
Q1	2N3906 pnp trans.

MISCELLANEOUS

jacks, wire, solder, etc.

Watch for these Thomas Henry articles in future issues!

BUILD A VCA WITH NINE PARTS

This design uses the SSM2010 VCA chip so don't let the fact that this circuit only has nine parts fool you! Thomas says "Give this circuit a try; I'm sure you'll be pleased. Out of all of the OTA's and VCA's that I have played with, this one was the easiest to get up and running and results in a very clean and quiet operation.

THE PGS FIVE OCTAVE TONE GENERATOR BOARD

A high quality music circuit capable of producing five complete octaves of equally tempered scale. Nine bit binary division is employed, leading to a minimum deviation from true scale. Thus the circuit is suitable for serious music uses. The board has been designed such that multiple units may be stacked and outputs summed thus yielding a very full flanged sound when each board is triggered by a separate clock. This board can be used for string and brass synthesizers, pop organs, pedal boards, etc. Don't miss it!

THE POT-POURRI MODULE

This unique and inexpensive synthesizer module adds six new functions all at once. They are: A lag processor, a buffer, a control voltage inverter, an audio inverter, a two-in one-out mixer, and a comparator. Uses are too varied to list here, but this module is a real workhorse!

VOLTAGE CONTROLLED FILTER (con't...)

J2, the envelope input of the VCF. If your ADSR has an output of + 5v or less, you may open up attenuator R38 (envelope input) all the way. The keyboard control voltage output should go to both the VCO and also to J1, the 1 volt per octave input.

Now set the resonance to a minimum via R41. Since this pot is strung between the positive asnd negative supplies , the minimum position is actually at about mid-rotation.

Now play some music! If you have accomplished the patch correctly, you should be hearing a traditional synthesizer sound. Now increase the resonance control, R41, and you will hear the " wah-wah" sound become more intense. Play around with the coarse and fine tuning controls and also attenuator R38 to alter the effect.

IN CONCLUSION

I hope you enjoy the PGS Lowpass Voltage Controlled Filter. I have had two copies of this circuit in my system for over two years now and I still marvel at the quality of sound possible with it. Be sure to experiment with the filter ; there are lots of options, and use of the unit is relatively "goof-proof". Good Luck!

—end—

GUITARISTS!

WE NEED YOUR YOUR HELP! WE WANT TO GIVE YOU EQUAL TIME BUT WE DESPARATLY NEED FEATURE ARTICLES, SHORT ARTICLES, NOTES, TIPS, MODIFICATIONS, EFFECTS CIRCUITS, ETC. ALL DEALING WITH GUITAR ELECTRONICS.

WE NEED REGULAR COLUMNISTS WRITING ON GUITARS, AMPS, EFFECTS DEVICES, ETC. IF YOU CAN HELP, CONTACT US IMMEDIATELY!

★☆★ CONTEST ★☆★

WIN $50.00 CASH!

HERE'S ALL YOU DO TO ENTER: Simply examine the drawings on the cover and the back page (page 24) and come up with suitable captions for both. Then write them down and send them to us here at COMMON MODE!

You must submit a caption for both drawings to be eligible, however, each one will be judged separately. So with each entry you have two chances to win! We will pick two grand prize winners (one from each drawing) and two runners-up. The two first prizes are $50.00 each and the runners-up prizes are $20.00 each. Each entry will be judged on uniqueness, originality, and pure entertainment value.

Entry deadline is ~~April~~ MAY 30. Winners will be selected ~~May~~ JUNE 2 and will contacted that week. The winning entries (plus others) will appear in future issues. The decision of the judges is final. Good Luck.

Heres a couple of samples to get you started: For the cover: "How the West was Won" or "So where's all these new-wave western clubs you were talking about?". And for the back cover: "No, No. Not more Slim Whitman commercials, Please!" or "I new that wire should have gone to point D"

Send entries to: COMMON MODE! CONTEST, Route 25 - Box 304,
Terre Haute, IN. 47802

CONTEST ENDS MIDNIGHT ~~APRIL~~ MAY 30!

case histories by Thomas Henry

(Editor's note: many readers wrote in to say they enjoyed Gary Kirkpatrick's article in DEVICE 1:2 on Troubleshooting. Shortly thereafter, Thomas Henry submitted the following article, which seemed like a very good follow-up. Thomas is working on an M.A. in mathematics, and is putting himself through grad school by playing guitar with the "East Side Pharoahs" 1950's Rock and Roll Show. He hates noise with a passion.)

Here are some case histories of various devices I've worked on. There is no better way to learn troubleshooting than to see how someone else successfully solved a problem, so learn from my mistakes. Incidentally, I'm self-taught in this sort of thing. If I can do it, so can you!

CASE #1. I was out on a gig, playing guitar through an Electro-Harmonix Deluxe Electric Mistress Flanger. All of a sudden disaster struck, resulting in a weird oscillating and whistling. I broke down and cried, for this is not a cheap device, and further, it was my favorite. I was the only person in town with one...what would this do to my reputation? I finished the night without it, and brought it home later. I first opened the back and looked. <u>This is the number one secret of trouble-shooting:</u> look before you leap. Well, I looked for quite a while and couldn't see anything awry. So then I applied my 2nd secret of trouble-shooting: <u>Think</u>. So I thought, "what could possibly cause eerie whistling and oscillating?" The guitar sound did get through, but the oscillation was much louder. Did the device have some sort of internal oscillator which could be leaking through into the audio? I didn't think so, but I couldn't be sure. Could it be caused by some interaction between input and output? I looked at the connecting wires and all looked rosy. Then all of a sudden it hit me...what would 60 cycle hum sound like if it were flanged? Ordinary hum is obvious; but what about flanged hum? I listened to the output of the flanger and confirmed <u>something</u> was being flanged. I then grabbed my test capacitor (a 1000 uF, 450V cap on alligator clips; it's indispensable - make one and see) and hooked it across the DC output from the power supply. Good-bye weird noise! I had it! By analyzing the problem, I knew what part to look for. I looked very care- at the big power supply capacitor and found the printed circuit board pad had lifted up and fallen to pieces. I reconstructed this with a small piece of wire and solder, and

(cont. on page 14)

within minutes, had the flanger back on the road. A simple story, but one with an important moral: never believe anything anyone tells you, especially a flanger. Even lowly 60 cycle hum can put on glad rags and somehow become something new.

(Editor's note: Very true. I suspect the "whistling" you noted was high frequency harmonics of the 60 Hz hum beating against the relatively low clock frequency driving the delay lines. This problem is called "aliasing", and can also happen if you feed a signal with lots of harmonics into any analog delay line device that does not have elaborate input filtering -- CA.)

CASE #2. My brand new 10 Band Equalizer by Electro-Harmonix was on the blitz, which could only be called itching problems of the worst sort. If you didn't touch it, it worked just fine; but the minute I touched one of the slider pots, it grew allergic to me and crackled in the loudest manner. Remedy: I opened the EQ up and looked. Everything looked ducky. I then put my 3rd trouble-shooting secret into action: Tap and listen. I hooked the EQ up to my amp, and with the back panel off, proceeded to tap with an eraser end of a pencil (anything insulated will work). I didn't just tap anywhere, rather, I thought functionally and tapped the areas which either were potential loosen-ups, or had something to do with the pots. Now, tapping in an imprecise process, in that you can't narrow the problem to individual pads; but in this case, it was enough to tell me (by the loud pops in the amp) that the solder connections were poor. Upon looking closer with a lens (indispensable for microwork) I discerned a thick layer of rosin between the pads and the lugs of the pots. A touch-up with a soldering iron, feeding in a little bit of new solder, cured all. Once again, there's a moral: Believe some of the things that some of the people tell you. Why? I remember once that someone told me that rosin can foul things up. I ignored this advice and thought, "must be a rare occurance". Not so! This is a common problem. When soldering, the joint must be heated long enough so that the rosin turns to liquid and gets out of the way of the connection.

CASE #3: I had just built the Craig Anderton Ring Modulator, made a beautiful case for it, and had it all tuned up. I called a couple of friends over to witness the wondrous sounds which it produced; they were dying to hear what a real live one sounded like. They hastened their way to the house, I hooked it up and wailed, and nothing came out! Well, that's not quite true. I was able to hear the internal oscillator doing its thing. What a let down. My friends hobbled out of the house all broken hearted, so I had to think and this is how I started. I thought, hmmmm...no guitar, but plenty of oscillator. Not in the output; probably a problem in the input. So, I looked. Sure enough - there was a pad on the board that had broken or lifted up, and it was right on the input capacitor, so my diagnosis was correct. Moral: always think functionally, and the details will follow. Also, don't tell your friends to come over on the same day that you build something.

Well, so much for my first installment of case histories. What I learned the hard way you have now learned the easy way... to sum up:

1. Always think first.
2. When you do think, think in terms of functions or building blocks to help isolate the problem. When you've narrowed things down to a particular function, it's much easier to find individual components.
3. After thinking, look. 90% of my best trouble-shooting was resolved by looking only.
4. Always be prepared for the unexpected. Remember that you may think you know what hum sounds like, but do you really?
5. Make a test capacitor. Power supplies can be a problem.
6. Tap with something insulated to find intermittent problems.
7. Rosin is an insulator (gee, I wish I had known that...)

Please note our new address:

DEVICE 12304 Scribe Drive, Austin, TX 78759

DEVICE the newsletter for the electronic guitarist/musician VOL.1:5-79

MODIFYING THE EH ELECTRIC MISTRESS

by THOMAS HENRY

I would like to start this article out by saying that I am not connected with Electro-Harmonix in any way, nor do they necessarily condemn or condone the modifications I am about to suggest. I am an experimenter and musician, and found these "secrets" out by serendipity! Perhaps they will help you to get more out of your flanger.

I feel the Electro-Harmonix Electric Mistress Deluxe Flanger is one of the best inexpensive devices of it's sort to ever come out. (Most music stores have it for about $90.) I consider it to be one of the two most important devices in my repretoir of sounds (echo being the other) Despite the low price, one can get sounds from the subtle to the dramatic with a few twists of the knobs.

However, I feel that Electro- Harmonix in particular and the whole industry in general has always been secretive about schematics and reluctant to give spec's and suggestions for alterations, so it's to us to come up with valid modifications. The Electric Mistress lends itself well to some minor (but significant) changes that add considerable degree of versatility to the gizmo, but note that the following comments apply only to the EH Electric Mistress Deluxe Flanger, AC powered model.

THE MODS

Here are the modifications we'll be adding:

1. a front panel control for delay time
2. a front panel control for mixing

(cont. p.2)

IN THIS ISSUE: Pg.

YAMAHA E1010 ANALOG DELAY

by PAUL RIVERA

For years, musicians have been using echo in some form or another, and from the beginning they have been putting up with the problems associated with achieving it. And, although studio echo units are highly sophisticated, they are less than suited to the rigourous environment of the road. It has been hoped that new analog and/or digital technology could provide a quiet and dependable electronic delay to replace outdated mechanical delays.

Currently, the echo unit used most by musicians is the Maestro Echoplex. Among other things, the Echoplex is mechanical, noisy, employs poor-quality heads, is prone to wow and flutter, and requires frequent maintenance as well as tape replacement to ensure acceptable sound. Although any one of these problems can be cured, the cost of modifying the unit to get it perfect is prohibitive, and requires the replacement of nearly everything but the chassis. Clearly, there must be a better way.

The first analog delay units, however, were just as noisy as the old Echoplex, and were usually more expensive. Digital units, the much-hearalded wave of the future, were no better. For although digital delays are quieter in traditional noise areas, the distortion they <u>do</u> generate is all "wrong." Our ears are <u>used</u> to certain types of distortion, which we recognise and accept for what it is, when necessary. Digital noise, however, is much different. It comes mostly from quantizing errors, and is inherent in the design of current digital equipment.

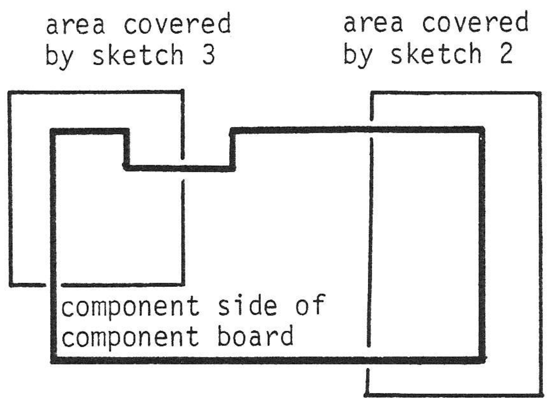
Time marches on, though, and right now, time is working <u>for</u> electronic delays and <u>against</u> mechanical delays. Mechanical delay units have reached (cont. on page 10)

straight
and flanged signals

3. increase regeneration for deeper effects.

4. an on-off switch with a power status LED.

We'll also discuss internal adjustments that can be made if our unit gets out of adjustment.



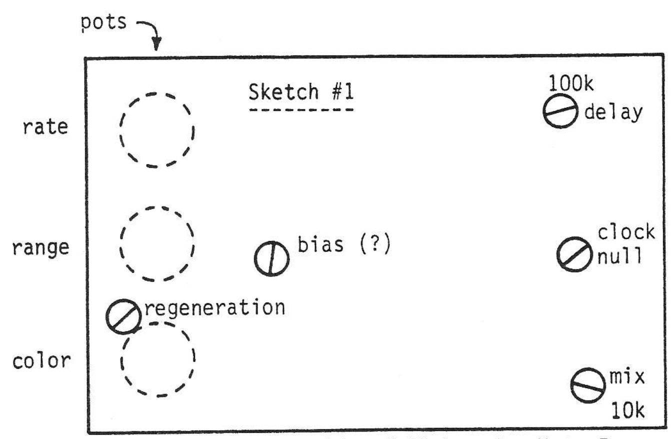
See sketch #1. This is a view of the circuit board from the component side, with the pots oriented to the left side of the board. Note that all the Trim pots are labeled, and we want to concentrate on these. First though, let's see how to get the board out of the chassis without ruining everything. Remember, the Electric Mistress uses a Reticon 1024 BBD, Which is not a cheap IC., so be dainty in your handling of everything. Here are the steps for removing the board.

1. remove the back plate (6 screws)
2. remove the pot nuts (3)
3. remove the jack nuts (3)
4. remove the filter matrix switch screws (2)
5. carefully pull the board out, and fold over. There should be enough slack wire to allow the board to sit comfortably out of the box for test purposes.

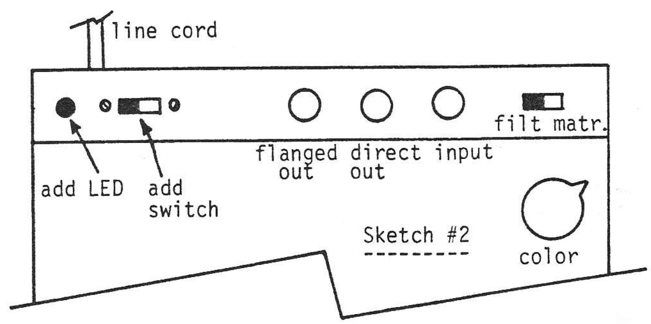
Now, for our first modification locate the delay trim pot. We would like this to be a front panel control. This works out quite nicely, for there is more than enough room in the chassis for several more pots. Simply desolder the trim pots, and run extension wires to a front panel pot. Use a 100K pot for this. This delay control interacts with the range control somewhat, so one must carefully set the two controls in relation with each other: But in return, we gain the ability to use our flanger for a short delay unit. The maximum delay is fixed in this unit, so don't expect more than a slap-back or "garage" effect; however, the sound is nice for special effects. I have found that by easing the delay time up somewhat, and setting the front panel color control to a minimum, I can get a nice doubling effect. On the bass strings the sound is close to that of a 12 string in some ways. All in all, I feel the delay trim pot makes a great front panel control.

The only trouble you may encounter is that at extremely long delay times, the clock signal feeds through to the audio somewhat. If this happens, back off the control untill the clock is less apparent, or add external lowpass filtering.

Another control we would like on the front panel is a mix control. Once again is internal to the unit, but let's put it up front for greater control over the



Component side of PC board. Note 5 white trimpots.

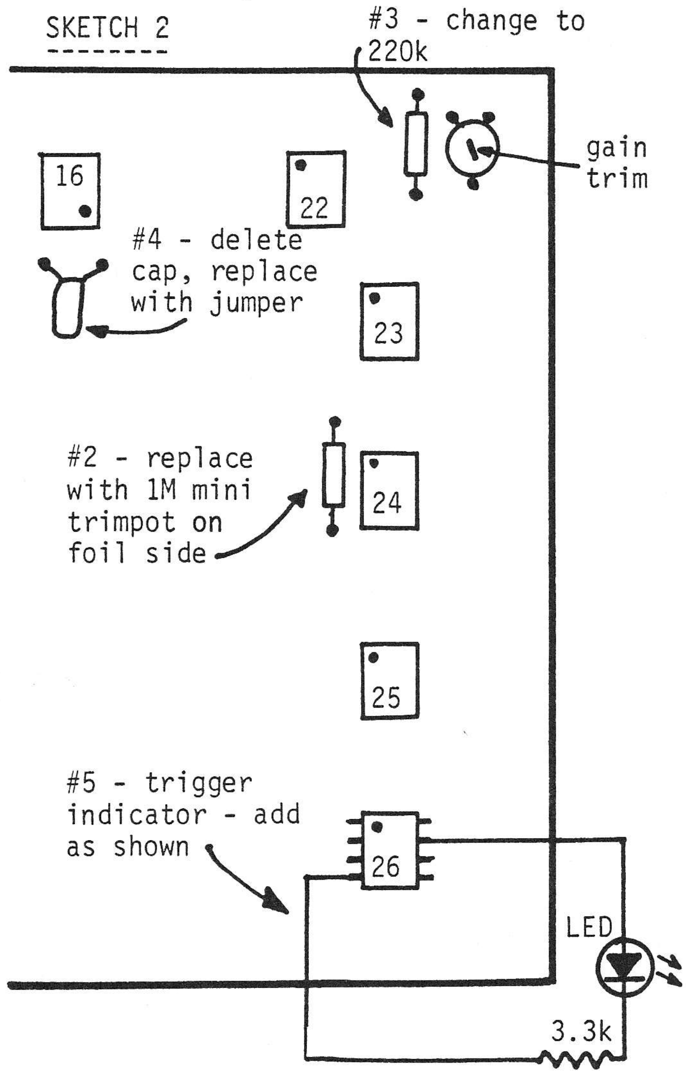


Front panel view of Deluxe Electric Mistress.

intensity of the flanging effect. Again referring to sketch #1, locate the mix trim. Desolder it, and bring extension wires up front to a 10K pot. Be sure to maintain the wiper and side lug orientation. Voila! We can now control the mix.

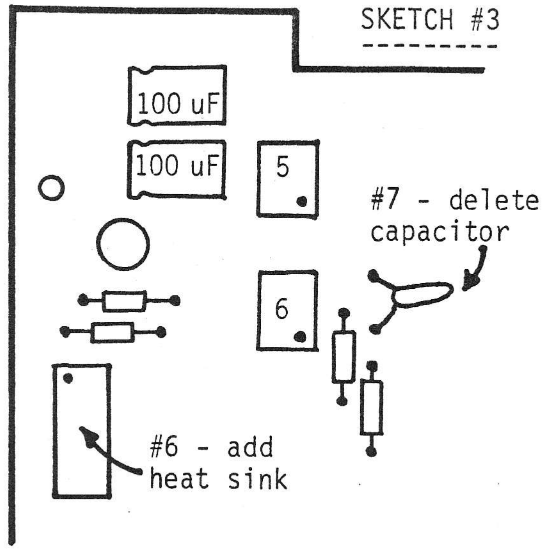
One draw back of the Electric Mistress is that it has no on-off switch, however, there is more than enough room to install one.(See sketch #2 for the placement.) Use a slide switch and two screws to hold it in place. Drill two holes, then form a rectangular hole for the switch with a small flat file. Be sure to use good soldering techniques, and carefully wrap everything with heat shrink tubing. Remember, you're dealing with 110 volt's!

As long as you're at it, drill a hole off to one side (see sketch #2) for a LED. Either use a grommet or a clip-tite LED holder to affix the LED in place. Solder a 4700 ohm limiting resistor in series with the LED, and once again wrap with heat shrink tubing. Locate the plus line and the ground line on the back side of the board and bring out extension wires to the



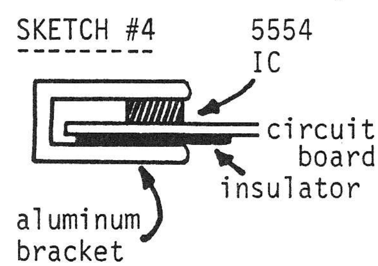
LED. These lines are quite easy to find: Look for the rectifiers and take it away from there. You now have a switch and power status LED. It looks so nice, why wasn't it there in the first place?

Finally, there is an internal control for regeneration that interacts with the front panel control called "color" whose purpose is to keep the unit from



oscillating. Vibration can upset this control however, and I found I was able to increase the intensity of my unit by carefully adjusting it. On sketch #1, locate the regeneration trim pot. With the unit hooked up to an instrument and amp, and with color control set to a max, trim until you get oscillation and then back off just a fraction. This will then give you the maximum amount of regeneration effect, which can be toned down by the color control on the front. Supposedly Electro-Harmonix does this adjustment in the factory, but shipping and gigging have a way of vibrating the control slighly out of synch. You should hear my unit now! Talk about the "jet-plane" effect!

Finally, there is a trim pot for clock-null. You probably won't ever need to use it, but it's nice to know it's there. Should you ever hear a high pitched squeal, adjust the clock-null until it is at a minimum. For short delay times it is



possible to get rid of all the squeal. For longer times you may have to compromise and let a little through.

There is one other trim on the board, but to tell you the truth I don't know what it does. I suspect it has something to do with how the two delay lines are strapped together. But unitl someone comes along and tells us what (cont. p.8)

MISTRESS cont. to do, let's leave it alone. [It probably adjusts the bias on the 1024. Trim for minimum distortion with the loud signals if this is in fact the case...Craig.]

I hope this article will be a help to those electronic music nuts who want maximum vesrsatility, even from a store-bought unit. My attitude has always been to get those controls up front where we can play with them! Just because a unit is store-bought doesn't mean we can't put our electronics savey to work. Let's modify, and not just that, let's tell everyone what were doing! [And write it up for DEVICE...Ed.] We have got to break down this mystique which companies seem to be perpetuating that the guts of a gizmo have no consumer adjustable parts. Bullshit! It's there for the taking.

I should mention however, that tampering with your unit will void the warrantee. However, in my opinion, the warrantee is so weak and worthless that you're not experiencing any real loss. But, you must decide that for your self.

Have fun!

ADDING SERIES-PARALLEL TO THE EH BIG MUFF pi DELUXE

by thomas henry

The ELECTRO-HARMONIX DELUXE BIG MUFF PI, AC powered unit is a terrific device, containing not only the renowned BIG MUFF PI distortion unit, but also the SOUL PREACHER compressor sustainer, all in one case. Front panel controls include a treshold and volume control for the compressor. There is an output jack for the compressor alone, and a jack for a mixed fuzz/compressed output. In addition, there is a slide switch which converts the mixed output to fuzz alone.

Sounds perfect, doesn't it? Wrong. When I first got the unit, the question which was foremost in my mind was: in what configuration are the effects wired, parallel or series? The next question was, which is better? I found out the hard way and pass my results on to you. I wired the unit so at a flip of a switch I can run the compressor in series with the fuzz or, in the alternate position, I can have it in a parallel with the fuzz (the configuration of the stock unit). As it turned out, the series position is the better of the two (for my style of playing, which is distinctly heavy-metal, with lots of sustain). Best of all, there are no holes to be cut for the switch, as we will use the hole provided for the slide switch mentioned above. In the end, we lose the ability to split the compressor and fuzz to separate amps, but gain a whole lot more versatility in altering the sound of the unit. (I only use one amp, anyway, so this was no disadvantage). Let's dig in.

The stock unit is wired as shown in Fig.1. We're going to change it the configuration shown in Fig.2.

Now, before anyone goes bananas let me explain something. Your first impression of this set-up is probably that it won't work.

Note that there is an infinite loop effect in the compressor, when the switch is on series mode. That is, the output of the compressor is being fed not only to the fuzz, but also to the input of the compressor itself. But the fact remains, the modification works! Why? Well, for one thing there are resistor inputs to the compressor and fuzz, and as I will show later, we will put this to good use in avoiding unwanted feedback. Secondly, I must confess that when the compressor volume is way down a slight oscillation can be heard. However, that is at <u>no</u> compressor volume, and I assume that everyone will have at least a little compressor brought up. So under playing conditions no problems will occur. (By the way, the reason for this compromise, is that to completely disconnect the compressor input one would need a triple pole, double throw switch. For those purists who demand the best, and can find said switch, install the third poles at the circled "X" on the drawing, so that the compress tie-in with the fuzz is broken on series mode. For what it's worth, I didn't find that necessary, and have had perfectly good luck with the DPDT

Well, that's the theory, on to the work. Carefully remove the back of your unit, remove the nuts on the jacks, pull the knobs off of the pots, remove the pot nuts, remove the nut on the main in/out push switch, and finally unscrew the "Blend" switch screws.

Carefully pull the board out of the case and fold over; there should be enough slack in the wires to allow for this. Locate the following points on the printed circuit board, following these directions: with the fuzz pots oriented on the right side of the board,

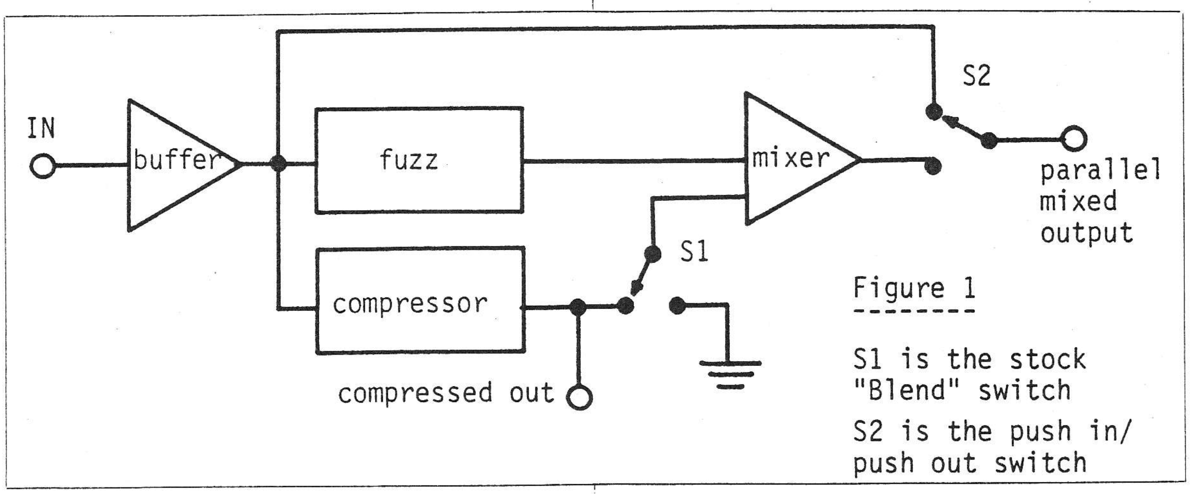


Figure 1

S1 is the stock "Blend" switch

S2 is the push in/ push out switch

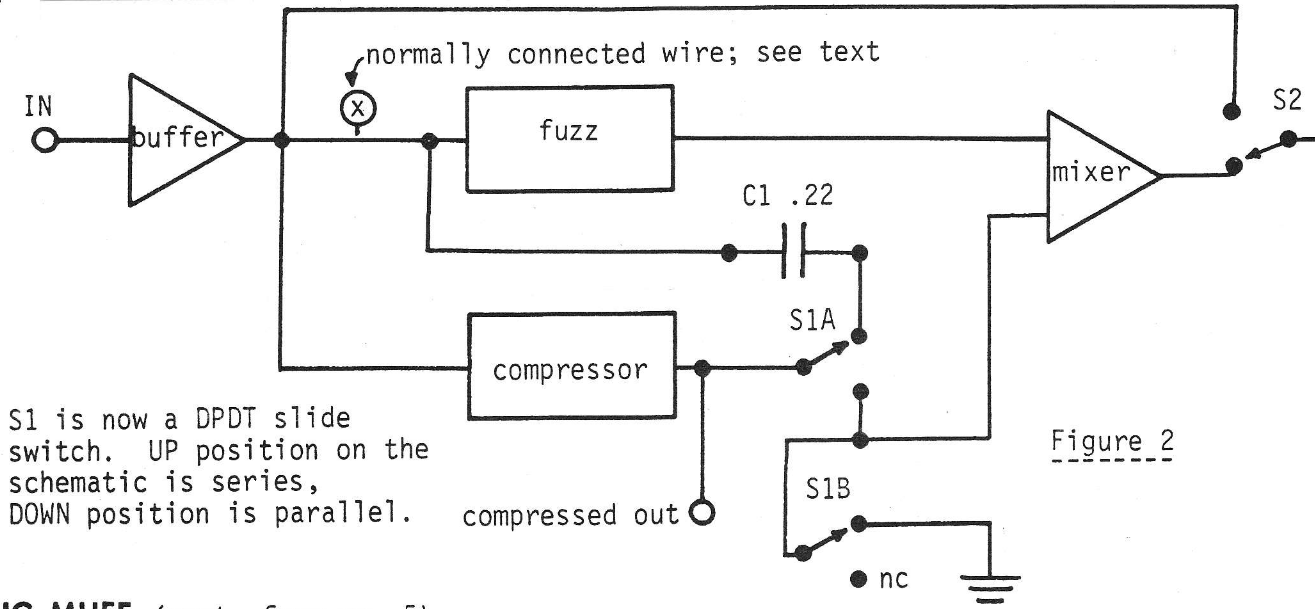


Figure 2

BIG MUFF *(cont. from pg. 5)*

as you face it, look along the middle of the foil side of the board, from right to left. You should see a line of IC's. Starting from the right, there is an 8-pin, and 8-pin and an 8-pin, (all 4558's by the way.) Then there is a gap of two inches until you find another IC, which is a 14-pin, (you should be right next to the power transformer now). About an inch above the 14-pin is another 8-pin (another 4558, which incidently is the output mixer chip). That's your sight-seeing tour of the BIG MUFF PI, presented to you in the hopes of helping you get around in there!

Now, back track, and start from the right again. Count over two chips (i.e., the middle 4558 in the group of three). You should see IC pads similar to Fig.3 on the PC board. The point marked "P" is the point we want. Memorize its position, better yet, mark it for future reference. It is the fuzz input. *

*(Note: Point P is the closest input to the fuzz, and avoids the input resistor, mentioned earlier; hence we can minimize feedback.)

Fortunately the rest of the connections that we want are marked on the circuit board, by the fine people at ELECTRO-HARMONIX. Locate the pad labelled "F" in the upper left quadrant of the board. (We're still oriented as before, pots on the right). This is the point where the compressor enters the mixer. Found it? Then locate pad"E" down on the middle pad of the compress volume pot. This is the compressor output.

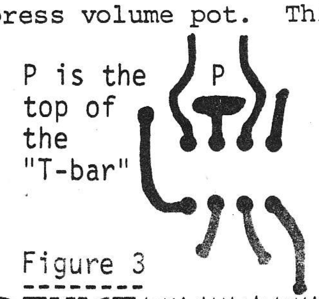


Figure 3

Unsolder the slide switch and replace it with the DPDT mentioned earlier. Follow the wiring scheme shown in Fig.4.

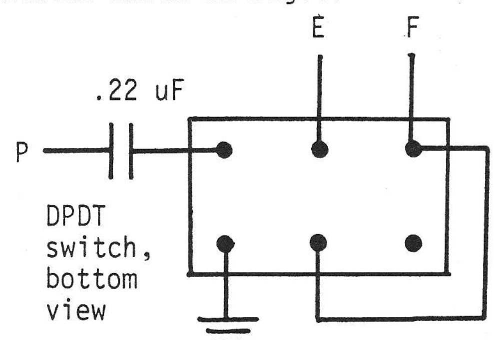


Figure 4

Note that we add a .22 capacitor (I used a mylar) between the compressor and the fuzz. This is a must, since the DC would run bonkers otherwise. There is more than enough room to solder this capacitor directly to pad "P" . Cover the leads with heat shrink tubing, lay the cap down on the board, and run the remaining lead to the switch terminal. The leads are stiff enough to keep the capacitor from rattling around.

You can take your ground connection for the DPDT switch from one of the jacks located next to the switch. Next, mount the switch in the old "Blend" switch hole.

Reassemble your unit, making sure that you watch the dressing of the wires. Keep everything clear of the power supply, and keep any input wires far away from the output wires. Before firing the unit up, double check for any errors. When you are satisfied, replace the back and check the unit.

With the switch in parallel mode, the unit is wired exactly as before. Bring up the compress volume for pure compression. Turn it back down, and bring up the fuzz for distortion. Bring up both fuzz and compression for a paralleled sound. Now, put the slide switch in series mode. Ease up slowly on the compress volume, and then the fuzz volume. The units are now in series (compress first, fuzz second) and so both volume controls must be brought up for sound to get out.

Wow! Talk about sustain! Fiddle with the controls a bit and I think you will agree this modification makes the BIG MUFF PI DELUXE into a super-charged device. I am able to get the longest sustain times I have ever heard in my life (or should I say "felt"). The gain is horrendous in this mode, so keep your amp volume down at first or you might wipe out your hearing. Also, with my Les Paul Standard and Music Man 210 if I get a little too close to the amp I experience some acoustic feedback...so watch out for that too.

For the purest compression possible, with no distortion, switch back to paralled mode and plug your amp into the compress output jack. This gives cleaner compression than taking the output from the mixed output jack, since the signal now bypasses the mixer circuit altogether (that's one fewer 4558 the sound has to go through.)

In conclusion, I wish to make it clear that I don't think the perfect fuzz box or the perfect compressor have been made. So, I am not trying to hard sell anyone on this unit. I do think that Electro-Harmonix has come up with a good inexpensive box (I paid $60.), and this modification has made me like it even more. But I don't think it is the sort of effect that you would want to use all the time (I'm a heavy metalist, myself, but even I grow sick of distortion and sustain <u>all</u> the time). Furthermore, purists might object to the relatively noisy 4558's used inside. But within those limitations I feel the device is useful. Have fun!

P.S. I changed the graphics on the front case to read - "Series/Parallel" instead of "Blend" by doing the following:

Carefully roughen up the word "Blend" with the end of an ink eraser. <u>Do not</u> scrub the words, just roughen lightly. Then rub some acetone on the word with a cotton ball. Careful! Acetone is highly inflammable, so don't smoke while doing this. Better yet, don't smoke at all and do yourself a favor. Anyway, the acetone will disolve the letters. Clean up the spot with rubbing alcohol, then let dry. When dry, apply dry transfer letters to read "Parallel/Series". Finally coat the new letters with clear spray paint. Buff with 0000 steel wool to bring back the original lustre. Voila! ■

Tom Henry's Pages

MODIFICATION TIPS

This time I thought I would pass on some modification tidbits – short and easy things you can do to upgrade your devices. Part of the trouble of modifing devices is that manufacturers are very secretive concerning schematics. My idea is that we form an "underground" army of musician-technicians, and every time we discover something, we spread it around. That's just one of the reasons I'm excited about **DEVICE**. Through it's pages we can inprove our knowlege of devices and spread the good word. So, let's start off divulging some adjustment secrets.

If you open up the EH Zipper Envelope Follower you will find a trimpot staring at you. It is soldered to the foil side of the PC board, so you can't miss it. Will Electro-Harmonix tell you what it is? Nope. Do the instructions that come with it tell you what it is? Nope. Will I tell you what it is? Yep! That trimpot adjusts the tuning of the filter. would you like to tune the filter down into the bass region? Easy, While playing turn the trim down and listen to the change. Turn the other direction to tune the filter up into the very treble region. Now I realize that this isn't very startling knowlege but if you're like me, you like to know what adjustments are there for the taking.

Speaking of adjustments – has your roadworn EXUMA Phaser been sounding dead lately? Then possibly the phase - center trimpot has vibrated free. Open up the unit and you will see a trimpot sitting on some floppy wires. Carefully adjust the trim while playing and should be able to find the "center notch", a position that gives the most phasing sound. If you like, when you have found your favorite position on the trim, seal with fingernail polish to keep the trim from vibrating free again.

While we're talking of phasers, have you built the CFR Associates Audio Phase Shifter Basikit[CFR Associates, Newton, NH, 03858] and don't like the sound? Try these changes and see what you think. I replaced the 741's in the phase shifter circuit with a 4136 quad-op amp and noticed a definite improvement in noise level and distortion. Furthermore, I replaced the two 741's in the buffer and mixer circuit with a 4739 dual op amp with the same good results mentioned above. I think another must is the inclusion of regeneration in the circuit; just tap the output of the last phase shift stage and bring it back to the inverting input of the *second* stage through a 220k resistor, a 1mfd electrolytic and a 25k pot. The resistor limits the regeneration to a point just below oscillation, the capacitor blocks the DC and the pot gives you a variable control. I think the device is positively stupendous now, which is funny, because when I originally built the circuit without these modifications , I thought it was junk. My roommate will attest to the screaming fits I went into when I built it. But now, I love the circuit.

Do you travel with a lot of gizmos? Well, I do and one thing I've always appreciated are those switching jacks on the inputs of devices. Obviously, when your traveling you don't have any plugs plugged into the box, so the battery doesn't get worn down. But then again, if you practice or record a lot, and you have an arrangement and cords that you like, it's a hassle to unplug everything at the end of the day, and then plug-in again at the start of practice. Do what I did. I put SPST switches on all my devices in series with the switching jacks. That way at the end of a session I can leave everything plugged in, and just hit the switches to axe the power. And on the road I still get the security of knowing the worlds!

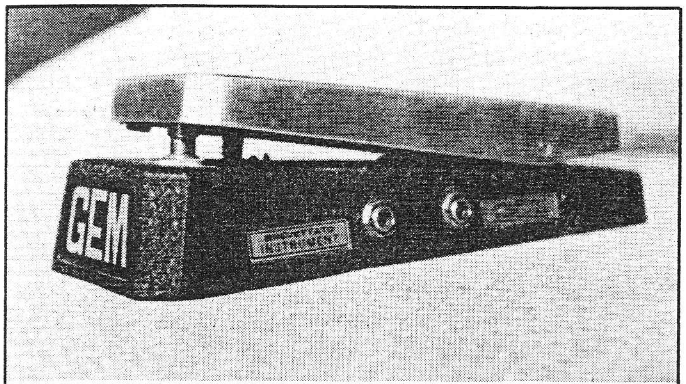
Most gizmos are using power status LED's now, but some don't. I usually put them in all my boxes, since I like to

(continued on next page)

NEW LAMPS FOR OLD

This month's modification really should be called a refurbishing, for we will take that cruddy old wah-wah pedal and convert it into a high class volume pedal.

Several months ago one of my musical friends was moving from town to seek his fortune in Florida. But before he went, he called me over to his house (in fact, minutes before he left) and said he had some old odds and ends he thought I might be interested in. He had a box with old cables, broken microphones, and among all this trash was an old wah-wah, circa 1965. He said he didn't want to carry the junk with him, so would I please do whatever I wanted with it. I obliged him, not really realizing the bargain I had gotten.



Note the pushbutton switch under the toe.

The pedal sat for a month or so, and then one day it ocurred to me that the pedal itself was sound; it was the circuitry that was junk. In fact the pedal was made of cast iron, and was one of those nice heavy types. In addition it had an adjustable tension on the throw, which really was quite nice.

Finally, I decided the time had come for a volume-overdrive pedal to be added to my repetoire of gizmos. Here's what I did.

I started with the preamp circuit from *Electronic Projects For Musicians* by Craig Anderton, (availble for several sources; published by Music Sales Corporation) on page 56. I built up the circuit, using a photo-reproduced circuit board, and stopped there, before doing anything about the feedback loop on the op-amp.

Next, I stripped out the guts of the pedal, and discarded the wah-wah circuit board. I saved the foot switch (under the toe of the pedal), and replaced the two jacks with stereo phone jacks. Next, I removed the pot from its rack and pinion casing and replaced it with a 1 Meg pot. I found that I had to epoxy the new pot in place, as the previous was substandard size. Finally, I re-installed the rack and pinion device, allowing the pedal to control the new pot now.

I mentioned that there was a DPDT push-push switch under the toe. The way this works, is when you throw the pedal all the way to the front and press hard, the switch triggers. I decided to use this switch as a range switch (see figure 1).

Note how the switch in one position allows the pot to be in the feedback loop unaltered, for huge amounts of over drive. Then, in the other position of 56K resistor is shunted across the pot bringing the gain clear down into a non-distortion range. So in one position the pedal goes from zero to about 2 in volume, and in the other position the pedal goes from zero to overdrive. You will note that I have deleted the 18K resistor which is in series with the pot in the book setup.

(continued on next page)

NEW LAMPS FOR OLD *(continued from page 3)*

This allows the pedal to drop clear down to zero gain (silence); we want this to get bowed effects and to stop the signal between songs.

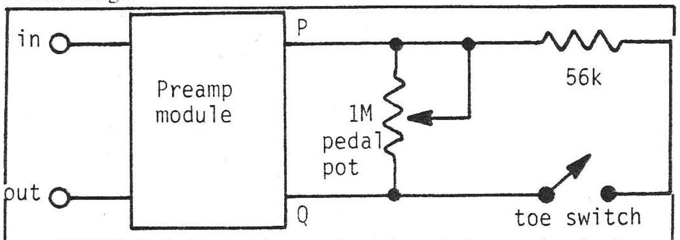
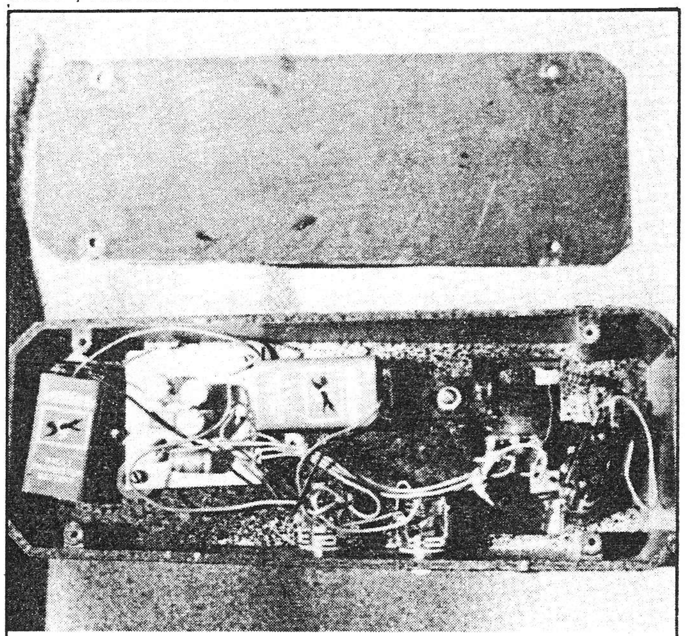


FIGURE 1 (note that points P and Q are the feedback conections for the preamp's op amp)

Construction is exactly the same as given in the book otherwise. Use the stereo jacks as switching jacks on the two batteries. Obey good wiring practice since there is a lot of gain in circuit. Keep the input away from the output to avoid oscillations, and use shielded wire for the input connection.

Now, naturally if you have an old junk pedal, it may differ from the one I have just described, and so your construction will be different. However, just think about what you have to do, and do it carefully, and you too can have a new pedal from old.



The heart of the matter

AFTERTHOUGHT: As I finished writing this article, it dawned on me that another, perhaps better way to wire the pedal would be to use a double section pot and a switching arrangement as shown in Figure 2. The pot should have a 1 Meg section and a 50K section.

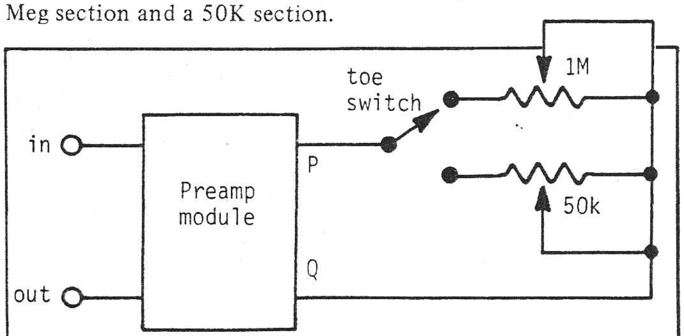
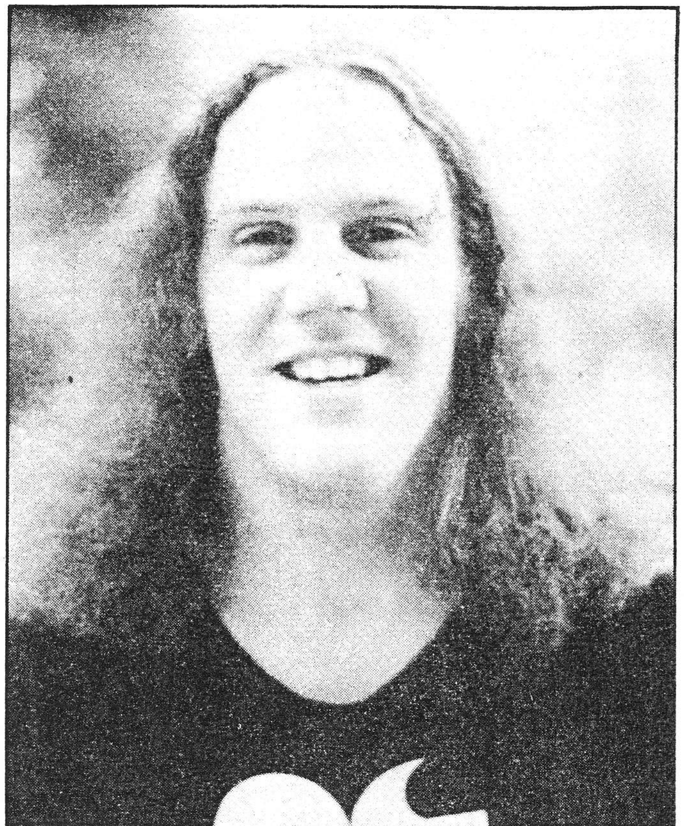


FIGURE 2 (each pot section is part of a dual ganged pot. Moving the pedal changes both pots simultaneously, but only one is selected at a time via the toe switch)

ADDENDUM TO THE VOLUME PEDAL

Note well: The audio is passing through the pot, so choose not just a good pot, but an *excellent* one to avoid noise problems. Allen-Bradley *sealed* pots are perhaps the best, and in this application are worth the extra buck or two.

* *



Here's Tom Henry

MODIFICATION TII *(continued from page 3)*

be able to tell at a glance what's on and off. Here's how I do it. First, estimate the power consumption of the circuit. If it seems fairly low, then you are safe. Simply wire the LED in series with the switching jack and the battery. I use cliptite LED panel mounts for a professional appearance[available at Chaney Electronics, P.O.Box 27038, Denver, Colorado 80227, and from others]. I have put LED's in the EH Octave Multiplexer, the Craig Anderton Pre-Amp and many other circuits from Craig's book. I use red LED's as they are the easiest to see on a darkened stage.

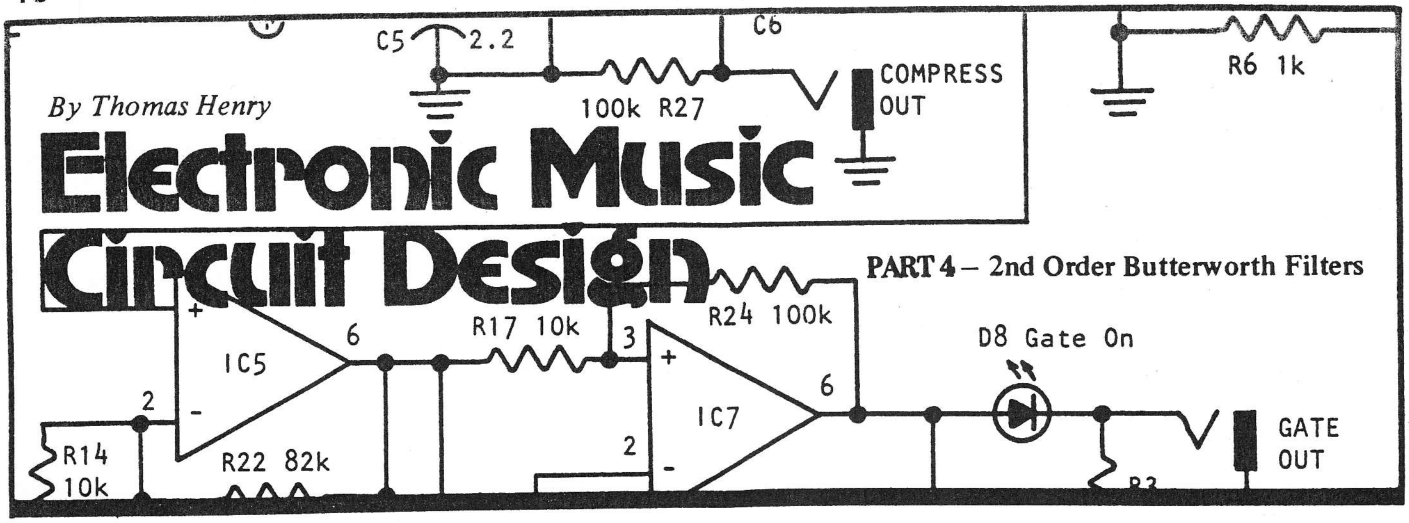
Well, that's all for this time. If you have ideas for modifications, be sure to pass them on. We can't remain silent on such an important subject. By all banding together and pooling our knowlege we can save countless dollars and prove that we can handle the workings of our machines!

(Editor's note: you might run into some problems with the LED indicator, for a couple of reasons. First, the LED will cause a voltage drop which may or may not be a problem. Next, for best results, I'd suggest wiring the LED in series with the positive battery connection – the one that goes to the effect circuit board in most cases – so that the anode goes to the battery + and the cathode goes to positive power terminal on the board. This at least prevents a voltage drop through the ground lead. For best results, use this technique with effects that draw about 10mA or less, and be prepared for the fact that it won't work with all *effects -- but it will work more often than not. ---Craig)*

By Thomas Henry

Electronic Music Circuit Design

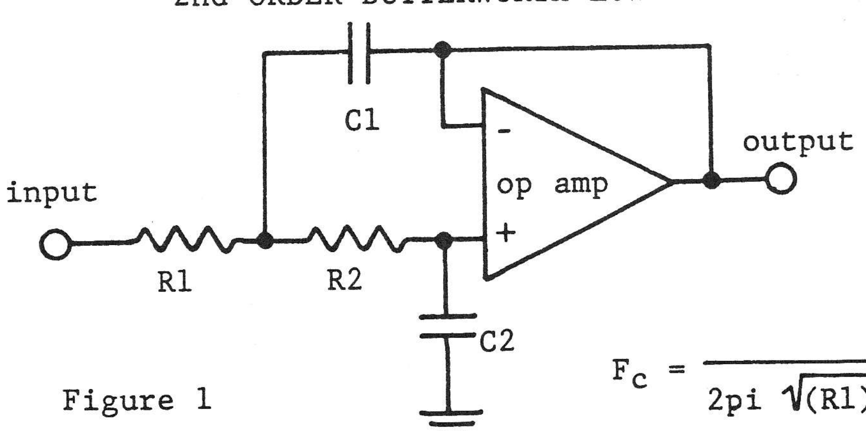
PART 4 – 2nd Order Butterworth Filters



Active filters are the types of building blocks that show up all the time in electronic music circuits, and indeed are handy little things to know something about. For example, several months ago, a friend and I were working on a pitch to voltage converter, and found the need to include both lowpass and highpass filters in the prototype. Filters show up in flangers (see DEVICE 1:9), analog delays, equalizers, even in the "lowly" wa-wa pedal. It therefore behooves the electronic musician to know a little about active filters, and even to be able to brew up a simple circuit at a moment's notice. This little article will concentrate on how to cook up a special type of active filter known as a second order Butterworth response filter. Our approach will be strictly practical.

First, what is a second order Butterworth filter? Well, behind all that jargon is something simple: we have a filter with a very flat passband (that's the Butterworth part) and rolls off the response at about 12 dB per octave (that's the second order part). The flat passband part is quite desirable and in practice, the circuit that you design and build will indeed have a nice, flat response in the passband. If you have a signal generator, a scope, and some semi-log graph paper, it is really instructive to design a filter (say, a lowpass) and then check its response with the instruments and finally graph the results. I did, several months ago, and was amazed at the results!

2nd ORDER BUTTERWORTH LOWPASS



$$F_c = \frac{1}{2\text{pi}\sqrt{(R1)(C1)(R2)(C2)}}$$

Figure 1

SECOND ORDER LOWPASS FILTER. This is simply a filter which passes all frequencies up to a certain critical frequency, fc, then attenuates all frequencies thereafter. The building block circuit in Figure 1 will work for all cases; you just pick the right values for the resistors and the capacitors, and away you go! Let's look at the design equations. The frequency at which this lowpass filter starts to roll off (the -3 dB point) is given by the equation (see note 1):

$$fc = \frac{1}{2\text{pi}\sqrt{(R1)(C1)(R2)(C2)}}$$

Figure 2 shows the calculations for a specific example; in this case, the -3 dB point is 2250 Hz.

Before formulating a design procedure, let's take a closer look at the response of the filter. If you were to graph the results of the circuit above, it might look like Figure 3. Referring to the graph, we see that from f1 to f2 the circuit rolls off the high frequency response at a rate of -6 dB per octave, and then at f2 the response rolls off faster, at a rate of -12 dB per octave (see note 2). This is theoretical, though; in actual practice you will probably only achieve a rolloff of -10 dB per octave due to the interaction between R1, C2 and R2, C1. Further, the critical frequency (fc) may be slightly off, due to tolerance problems in the above components. One can minimize this latter problem by using high quality capacitors and precision resistors, but generally this level of accuracy is not needed. For example, several months ago I whipped up a lowpass filter with a theoretical cutoff of 84 Hz and ended up with 81 Hz in reality...and that was just with ordinary components. Not bad!

A good design procedure would then be:

1) Arbitrarily pick a value for C2. *(continued on page 12)*

CIRCUIT DESIGN
(continued from page 11)

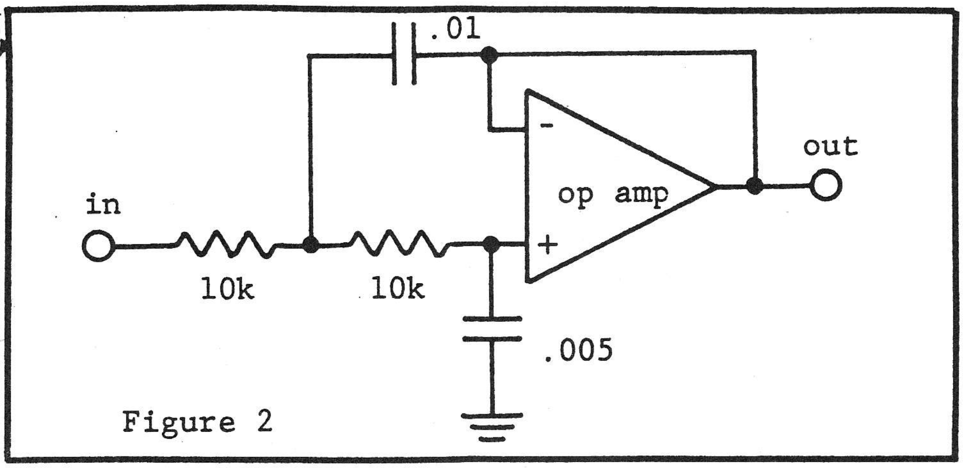


Figure 2

$$F_c = \frac{1}{2pi \sqrt{(10 \times 10^3)(.005 \times 10^{-6})(10 \times 10^3)(.01 \times 10^{-6})}} = 2250 \text{ Hz}$$

2) Let C1 = 2 times C2 (see note 3).

3) Let R1 = R2 (see note 3). Since we know fc, C1, and C2, we can solve for R1 = R2 from the cutoff formula:

$$R1 = R2 = \frac{1}{fc(2pi)\sqrt{(C1)(C2)}}$$

EXAMPLE: Design a second order Butterworth lowpass filter with a cutoff of 650 Hz.

1) Pick C2 = .01 uF (that's a pretty common value).

2) Then C1 = 2 (.01 uF) = .02 uF

3) Compute R1 and R2. Since we know fc, C1, and C2, we can solve for R1 = R2 from the resistor formula:

$$R1 = R2 = \frac{1}{(650)(6.28)\sqrt{(.01 \times 10^{-6})(.02 \times 10^{-6}}} = 17.3k$$

One could then pick a precision resistor, or if accuracy isn't crucial, a close 5% value will do. In this particular case, 18K might be a good choice.

For the values of R1 = 18K, R2 = 18K, C1 = .02 uF, and C2 = .01 uF, the actual cutoff would be 625 Hz, so you can see the effect that rounding off R has. As they say, close enough for government work.

SECOND ORDER HIGHPASS FILTER. We can really buzz through the design of highpass filters, since they are the "dual" of lowpass filters, both electrically and mathematically. All of the same equations hold, and the design procedure is the same.

A graph of the response of a highpass filter might look something like Figure 4. The frequencies fc, f1, and f2 obey the same equations given above. Once again we see a rolloff of 6 dB from f2 to f1, and then it picks up quickly to achieve a 12 dB rolloff from f1 on down.

We said above that the highpass filter is the "dual" of the lowpass. Here's why. Compare the circuit in Figure 5 for the highpass filter with that of the lowpass. Note that wherever a capacitor occurs in the one, a resistor occurs in the other and vice-versa. That's pretty neat! And you'll find this idea of duality occurring all over the place in the design of active filters.

EXAMPLE: Design a highpass filter with a corner frequency of 60 Hz.

1) Pick C2 = .1 uF

2) Then C1 = 2 (.1 uF) = .2 uF

3) Solve for R from the following equation:

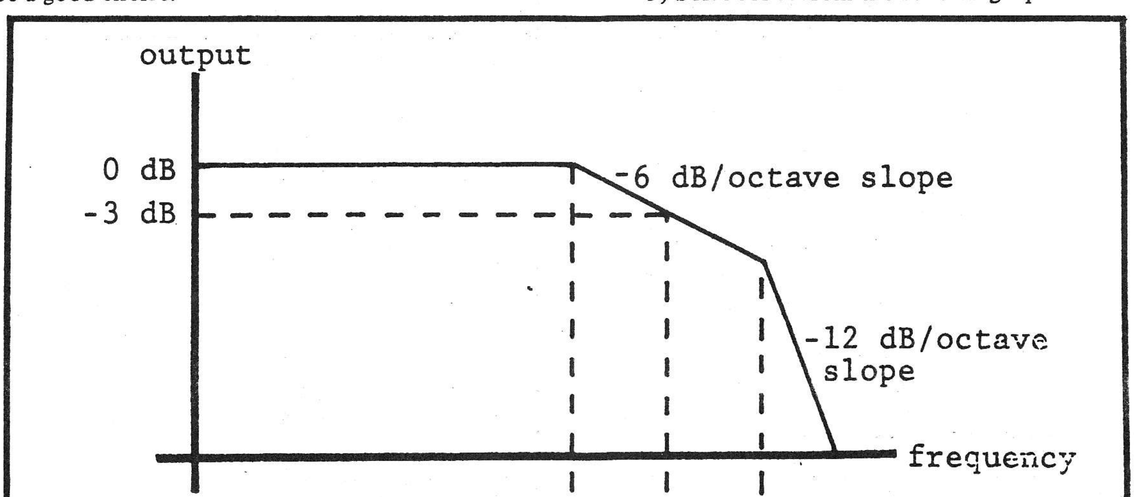


Fig. 3 - Lowpass response

$$R1 = R2 = \frac{1}{(60)(2pi)\sqrt{(.1 \times 10^{-6})(.2 \times 10^{-6})}} = 18.8k$$

CONCLUSION: Of course this isn't everything that there is to second order filters, but unfortunately to really get at the theory of active filters one needs a fair amount of mathematical preparation. However, the above practical methods work quite well, and even if you don't understand the total theory behind them, the fact remains that you now know enough to build them. Try out a few designs of your own (say, with a solderless breadboard), and then compare the theoretical with the practical. If you do just a few practice designs, I'm sure you will see just how easy and convenient the second order Butterworth filters (both the lowpass and the highpass) are to design and build. Here are a few practical notes to help you along:

1) For best results in audio circuits use low noise op amps, say a 4739 or 4136. FET input op amps are also well suited to active filter circuits.

2) It is possible to cascade two lowpass filters or two highpass filters for a steeper rolloff, however you will lose the Butterworth part (flatness of response in passband). For this reason, it is better to go with a fourth order design - see the sources listed below.

3) Cascading a lowpass and a highpass is legal, and by doing so you are really designing a wide bandpass filter. One advantage of doing it this way is that you can independently select the two corner frequencies.

4) Be aware of weird values. If your final design ends up with R1 equal to something like 123 Ohms, you had better select another value for C1 and C2 since such a low value resistor could cause loading problems. Extremely high value resistors could cause hum problems and increased noise. Good values for R1 and R2 would lie between 10K and 100K.

5) Both filters shown in this article have a gain of unity, and are non-inverting with respect to phase. They were purposely designed this way, partly to simplify the arithmetic and also because the Butterworth filter has a tendency to oscillate at high frequencies when used in a gain configuration. If you need gain, add a gain stage before or after the filter (but do avoid clipping).

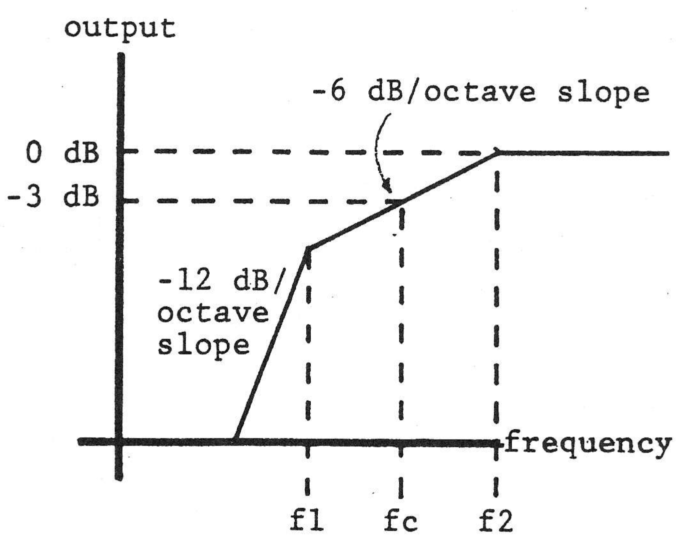


Fig. 4 - Highpass response

6) If at all possible procure a scientific notation calculator (the TI-30 at $17 is my choice). This makes the arithmetic a snap.

7) Be prepared to use a lot of paper and pencils. Your first design may end up with weird values. Play around with different values for the capacitors until you get something nice. Sometimes it takes me five or six attempts until I find a design I like.

NOTES:

(1) A calculator is almost a must here. If yours doesn't have a key for pi, use the approximation 3.1416 or simply 3.14. Note the resistance must be in Ohms and the capacitance in Farads; however, you can also use megOhms and microFarads if you wish to simplify things. Better yet, use scientific notation and save yourself a hassle.

(2) If you are curious, you can predict where f1 and f2 fall from the equations:

$$f1 = 1/\,6.28\,(R1)\,(C1) \qquad f2 = 1/\,6.28\,(R2)\,(C2)$$

(3) The conditions C1 = 2 (C2) and R1 = R2 guarantee that the resulting filter will have a Butterworth response.

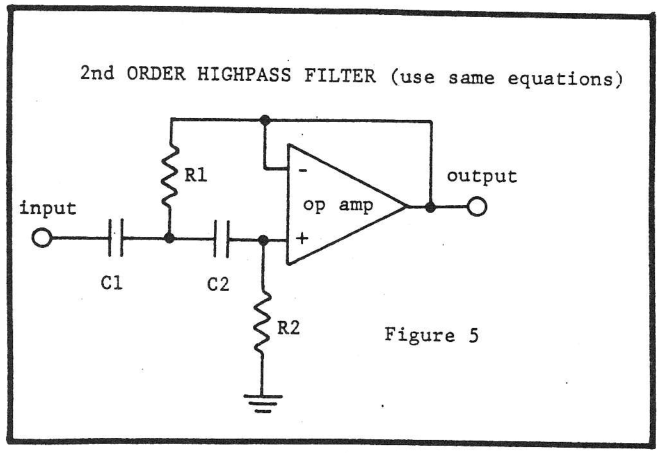


Figure 5

SUGGESTED READING:

Hilburn, John L. and Johnson, David E. Manual of Active Filter Design. New York; McGraw-Hill Book Company, 1973.

Hutchins, Bernie. Musical Engineer's Handbook. Ithaca, New York; Electronotes, 1975.

Lancaster, Don. Active Filter Cookbook. Indianapolis, Indiana; Howard Sams and Co., 1976.

Lancaster, Don. "Understanding Active Filters", Popular Electronics, December, 1976, pages 69-76.

NOTES FROM NAPA *(continued from page 2)*

top everything off some letters that needed replys have yet to be found (oh no, looks like Mister Sluggo's been on the job again). If you feel that you have been slighted or left in the dark, PLEASE, drop us a note. I won't let this happen again!

I have already receivied some subscription renewals and while this is great to see, it is also a little premature. All subscribers will receive 12 (count 'em) issues for their $15. You

(continued on page 16)

DEVICE The Publication for Electronic Musicians

VOLUME ONE:TWELVE:SEVENTY–NINE/ONE DOLLAR&FIFTY CENTS

BY THOMAS HENRY PART ONE

STALKING the SAD-1024

In which Tom takes us through the paces and traces of the popular Reticon BBD chip

This article is a compendium of tips and hints on using the SAD-1024 analog delay chip. In it, I won't concentrate on the theory of the chip, but rather on the practical aspects of getting the chip working in most circuit configurations. There has been a fair amount of literature published on the chip, and in some cases there have been contradictory statements made about the chip's wants and needs. I hope, then, to clear up some of the confusion and also to review some things that you possibly already know.

At the end of the article I have listed my sources. The material has been numbered, and noted in the article by those numbers. For more information on a particular topic refer to the numbered reference. In most cases the original article will give you the theory and design notes and other ideas that can give you further depth on this subject.

I should say a few things about the sources. You can probably find the *Radio-Electronics* and *Popular Electronics* in your local library. However, the other sources aren't quite so easy to locate. I have given some addresses to help you along though. *Polyphony, Electronotes* and *Technotes* are, to my knowledge, all available on a back issue basis. Write to the respective publishers for pricing information. The Reticon information is available to "qualified" persons only. Qualified, in this case, means you must be in a business, school, or so on, and write for the literature on "official" stationery.

Finally, I've listed Marvin Jones' *String Synthesizer* article because it illustrates how the SAD-1024 can be incorporated as an integral unit of and instrument, and further demonstrates the concept of chorusing.

INPUT CONSIDERATIONS

1. The input signal to the SAD-1024 must be appropriately conditioned to avoid distortion. First, the input level *must* be below 2V peak-to-peak. If the signal is lower still, say below 1.4V p-p, the manufacturer claims less than 1% distortion. However, there is some chip-to-chip variation, an a particular chip may be able to handle 1V p-p max. Play it safe then, and limit the input level to 500mV p-p. Between the 500mV and 1V p-p level the peak (not average) signal to noise ratio (S/N) is greater than 70dB. *(See note 8.)*

(continued on page 7)

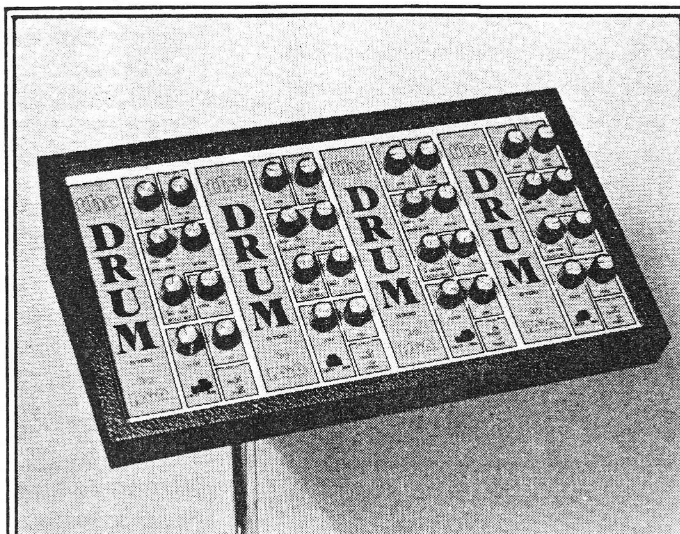
review

DRUMMERS STRIKE BACK!

BY RON MINEMIER

The PAIA DRUM kit is put through a shake-down cruise

Synthesizers have grown; many people still think of synthesizers as keyboard instruments although now some operate under digital control with programmable presets and/or polyphonic voicings. However, the synthesizer's growth has gone far beyond just voicing sophistication, since new inter-



facing electronics allow non-keyboard musicians synthesizer expression. While pitch to voltage converters have garnered a lot of attention since they allow guitarists and other acoustic musicians full synthesizer control, there are a growing number of special purpose interfaces. With this review we will explore some of the capabilities of the percussion/synthesizer interface by examining PAIA Electronics' new kit **The Drum**.

Intended to be a modular block, each Drum kit ($59.95 postage paid) has one complete drum voice. A four channel system with case and power supply sells for $269.75 and assembled units are available.

Many **DEVICE** readers are familiar with PAIA, as it is one of the larger musical electronics firms in the U.S. geared specifically to kit builders. For anyone just starting in electronics, it's difficult for me to think of a better place to start

(continued on page 2)

in this issue

22 2. In order for the bucket brigade stages to pass information, the input must be biased up to approximately 40% of V_{dd}. In case of a +15 supply, this is +6V. The circuit in Figure 1 should be used to accomplish the biasing.

In the end, you will want to adjust the trim while observing the output on a scope. Adjust for no (or lowest) distortion with a full scale input.

3. Limit the input frequency to the SAD-1024 to less than $1/3f_c$, where f_c is the clock frequency applied to Ø1 and Ø2. By doing so you will minimize "foldover" distortion in the output. An active filter stage could easily be cooked up to lowpass everything up to 1/3 f_c, and at the same time increase or reduce the input level of the necessary 500mV p-p, (see above). This bandwidth limiting is *not* a trivial matter, and is vital to getting the output to sound clean.

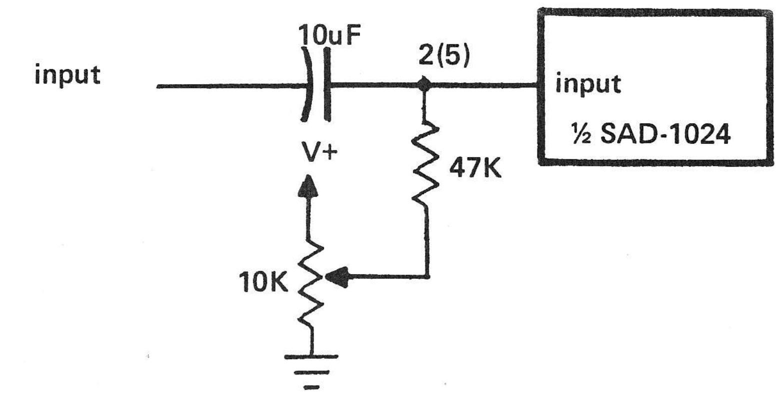


Figure 1

OUTPUT CONSIDERATIONS

4. The outputs of one section of the SAD-1024 can be terminated in either of the following two ways shown in figure 2.

This brings up an important point. There *must* be an output load. Of the above, the transistor output gives better results. The 1K trim is adjusted until the outputs (viewed on a scope) "come together", and have the same amplitude. This minimizes clocking glitches also.

5. The gain of the stage depends directly upon the load resistance, and in general you can expect it to be greater than unity. (a 10K resistance, not a practical load by the way, gives a gain of 1.2). Since there is a gain in general, if you are planning upon serial operation (tying two stages together in series), you must attenuate the output of the first stage somewhat, to attain the input requirements, (mentioned above), for the next stage. Figure 5 (located on page 16) fills the bill.

By the way, this is a good time to remark that all unused *outputs* should be tied to V_{dd}. For more on this read on.

6. The output should also be lowpass filtered, with a cutoff of at least $1/2f_c$, with $1/3f_c$ being preferred. This filter serves two purposes. The first is to remove the residual clock glitches, and the second is to smooth the stairstep output into a close approximation of the input. This smoothing is directly related to the cutoff slope. In the spec sheet, Reticon recommends a cutoff slope of 36dB/octave, but I think in general that a four-pole (24dB/oct.) is adequate. We might note in passing that pre-emphasis and de-emphasis isn't a viable way to reduce noise in view of the aforementioned remarks concerning foldover distortion. The very process of pre-emphasis violates the input filtering requirement. What we need is less high-end response at the input of the SAD-1024, not more.

POWER SUPPLY CONSIDERATIONS

7. The SAD-1024 can function over a wide supply range, with best results when it lies between +9 and +17V. +15V is the preferred supply voltage as all parameters maximize here. Another reason for preferring the +15V supply is that at other voltages the clock amplitude and input biasing must be adjusted considerably. This can be a real hassle, so stick with a standard supply. [*There is also something to be said for running the SAD-1024 at a lower voltage than the clock voltage – see the Flanger in DEVICE 1:9 – CA*]

To keep clock noise from getting into the supply lines, bypass the chip in the following way:

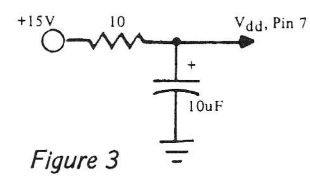


Figure 3

8. V_{bb} (which supplies a voltage to the bias line) is required for some versions of the SAD-1024 and can be provided by the +15V V_{dd} voltage, however the spec sheet claims that best effeciency is obtained when V_{bb} is one volt lower than V_{dd}. The following voltage divider does the trick:

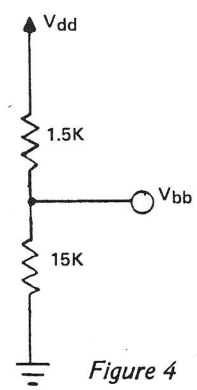


Figure 4

9. V_{dd}, V_{bb}, and ground pins, (7,9, and 1 respectively) are common to both stages of the SAD-1024.

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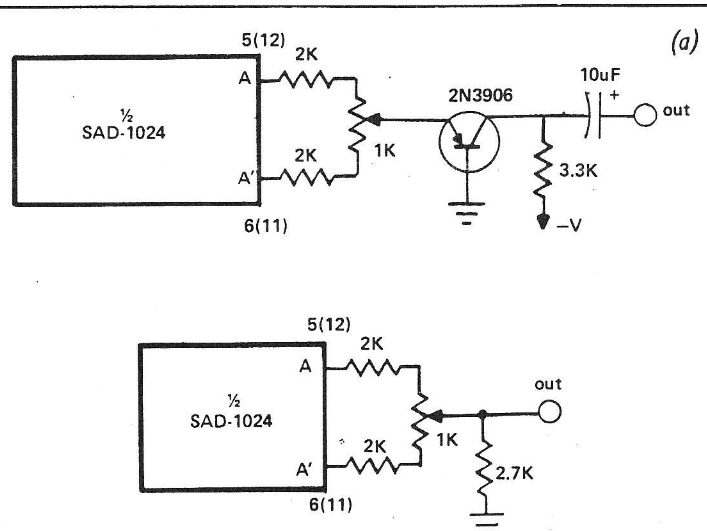


Figure 2

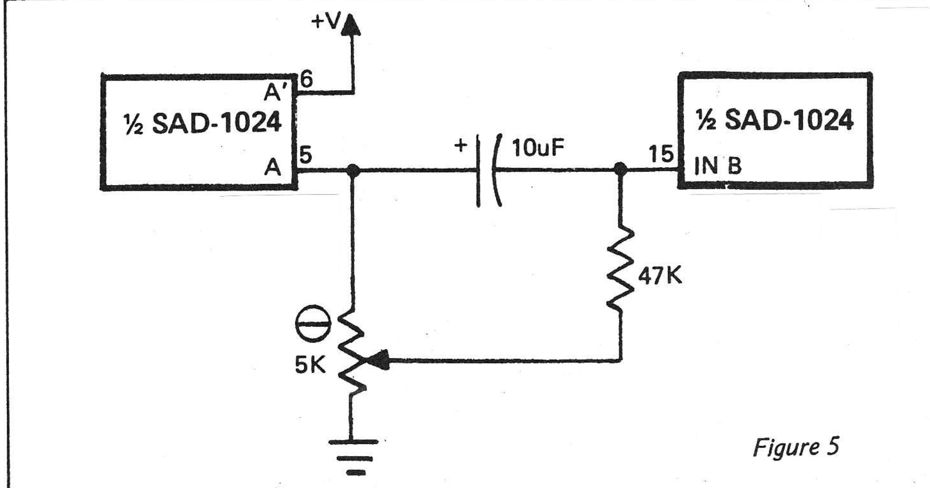


Figure 5

NOTES AND SOURCES FOR *STALKING THE SAD-1024*

1. Gary Bannister, "To Phase or to Flange...", *Polyphony*, April-May, 1978, pp 12 - 15, 29.
2. Bernie Hutchins, "Analog Delay for Musical Engineering", *Electronotes*, Volume 7, Number 56, August 1975.
3. Marvin Jones, "Build the Phlanger for Dramatic Music Effects", *Radio-Electronics*, October 1977, pp 44 - 45, 92 - 93.
4. Marvin Jones, "Build This String Synthesizer", *Radio-Electronics*, February 1979, pp 37 - 41 and March 1979 pp 71 - 75, 104, 108 and 110.
5. Marvin Jones, "Experimenting With Analog Delay", *Polyphony*, July - August 1978, pp 14 - 19.
6. Forrest M. Mims, *Engineer's Notebook: A Handbook of Integrated Circuit Applications*, pp 44 - 45, Radio Shack.
7. *Reticon* Application Note No. 104A, "Making Music With Charge Transfer Devices".
8. *Reticon* Spec sheet, SAD-1024 Dual Analog Delay Line.
9. John H. Roberts, "The Bucket Brigade Audio Delay Line", *Popular Electronics*, June 1976, pp 33 - 38.
10. *Technotes*, Volume 3: Issues 2,3,4, and 5. CFR Associates.
11. Craig Anderton, "The Flanger", *Device* Issue 1:9 (corrections in *Device* 1:10).

ADDRESSES

Electronotes, 1 Pheasant Lane, Ithaca, New York 14850.
Polyphony, P.O.Box 20305, Oklahoma City, OK 73156.
Reticon, 345 Potrero Ave., Sunnyvale, CA 94086.
Technotes, CFR Associates, Newton, NH 03858.

STALKING THE SAD-1024

(continued from page 7)

10. As mentioned above, all unused *outputs* should be tied to V_{dd}. However, all other unused pins, *including those marked NC*, should be tied to *ground*. These requirements are easy to forget, so make a special note of them now.

[This concludes Part One of Tom's article on the SAD-1024. Part Two will be in your mailbox NEXT MONTH! Stay Tuned!!]

A BALANCED MODULATOR APPLICATION OF VCA CHIPS:

-by Thomas Henry
Elect. Music Studio
MB 2055
Univ. of Iowa
Iowa City, IA 52242

Bernie Hutchins' article, "A (Gulp!) Simple Balanced Four Quadrant Multiplier With an OTA", ELECTRONOTES #107, pp 13-15, represents something of a breakthrough in the design of balanced modulators. First the idea is so ingenuous. As he implies in the article, this method has been around for some time, and yet it comes to light relatively late (to the musical engineering crowd). Secondly, though there are several four quadrant multiplier chips available, they are often expensive. There are many times when one wants a "ring modulator sound", and the use of an expensive chip isn't justified, especially if a satisfactory alternative such as the above exists.

So the situation is this. Use a good quality dedicated multiplier chip when cost is no object, and use the CA3080 route when cost is important and one can get by with relaxed standards.

However, there is a middle of the road approach with a cost somewhere between the two methods mentioned above, and that is to apply Bernie Hutchins' method to a better quality VCA. There are several such IC's available now, for example SSM's 2000, 2010, or 2020, or possibly Signetics' 570 or 571 compandor chips. In this article the Curtis CEM3330 Dual VCA chip is pressed into service. Since only one half of the chip is used, further savings can be realized if the other half is needed somewhere else.

Before looking at the calculations the reader should note that he can expect more convenient results with this chip as compared to the CA3080. Both inputs may be AC or DC coupled, the impedences of the driving sources need not be as low as those desired in the original article, and the output is non-inverted. In addition, the noise figure of the CEM3330 is superior to that of the CA3080. Finally, the chip is well suited for music applications and the supply voltages and various impedences fit in with the "standard".

Pin numbers 1 through 9 are used herein, but the reader is reminded that this is a dual unit, and hence the use of pins 10 through 18 is also possible. The two units contained in the chip are identical.

Before getting down to the calculations, some preliminary observations should be made. Perhaps the reader should keep the spec sheet for the CEM3330 handy for these comments. A reference current, I_{ref}, to pin 2 should be established, and selection of this current determines the distortion performance of the chip. A value between 50μA and 200μA is recomended, and 100μA was finally selected for this design. With a ± 15V supply, a 150K resistor from the positive supply to pin 2 does the trick.

Pin 3 which is a second harmonic distortion adjust input is grounded, since it was deemed unnecessary in this application. Pin 6, the exponential input is also grounded. Only the linear gain control is used for balanced modulation.

A resistor from pin 5 to pin 8 sets the operating mode of the amplifier. A 6.8K resistor was chosen, and this sets the amplifier up for class A performance.

Pin 7 is the linear control input, and actually several things happen here. A current through this pin, I_{cl}, has its logarithm taken, and the logarithm is applied to the gain cell, which has an exponential response. The result then is a linear response, (since $e^{\log I} = I$). This pin is at virtual ground, as is the signal input, pin 4. This feature simplifies design considerably later on. Now, on to the mathematics.

The gain equation of the CEM3330 is:

$$I_o = -I_{in}\,(I_{cl}/I_{ref})\,\exp(-V_{ce}/V_T) \tag{1}$$

where:

I_o = output current (pin 1)

I_{in} = signal input current (pin 4)

I_{cl} = linear control current (pin 7)

I_{ref} = reference current (pin 2)

V_{ce} = exponential control voltage (pin 6)

$V_T = \frac{kT}{q}$ (a constant)

As mentioned above, only the linear control pin is used, and the exponential control pin is grounded. Hence V_{ce} = 0V and equation (1) becomes:

$$I_o = -I_{in}(I_{cl}/I_{ref}) \tag{2}$$

Let $I_{in} = (E_{in}/R_{in})$ where:

R_{in} = the resistor to pin 4

E_{in} = the voltage applied to this resistor (since pin 4 is at virtual ground)

Then equation (2) becomes:

$$I_o = \frac{-1}{R_{in}I_{ref}}(E_{in}I_{cl}) \tag{3}$$

See figure one.

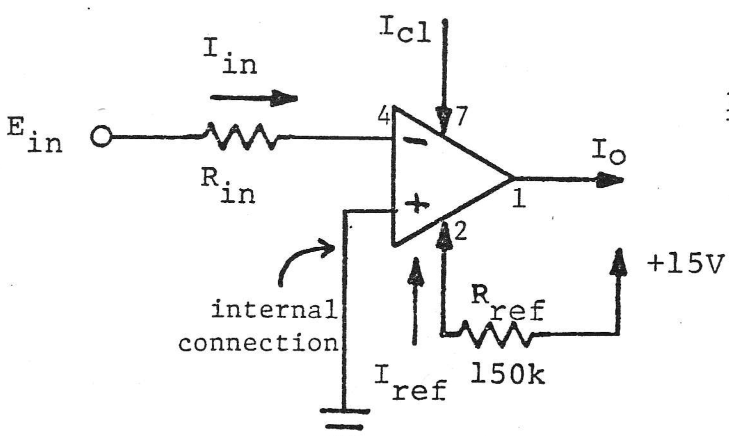


Fig. 1

The signal input will be restricted to the range: $-5V \le E_{in} \le +5V$. In addition for lowest distortion I_{in} must be less than several hundred microamperes, while for best noise figure, I_{in} must be as large as possible. As a compromise, let I_{in} = 50µA maximum. Then from Ohm's law:

$$R_{in} = \frac{E_{in\ max}}{I_{in\ max}} = \frac{+5V}{50\mu A} - 100K$$

I_{ref} has already been selected to be 100μA (see above), so substituting these values for I_{ref} and R_{in} into equation (3) gives:

$$I_o = (-1/10)\ E_{in} I_{cl} \qquad (4)$$

which is the transconductance equation for the CEM3330 in this configuration. Note that R_{in} and I_{ref} are constant for any particular design.

Now consider the schematic in figure two. From Kirchhoff's current law we must have the following equation for node 1:

$$I_2 + I_1 + I_o = (V_x/R_x) + (V_{out}/R_f) + I_o = 0$$

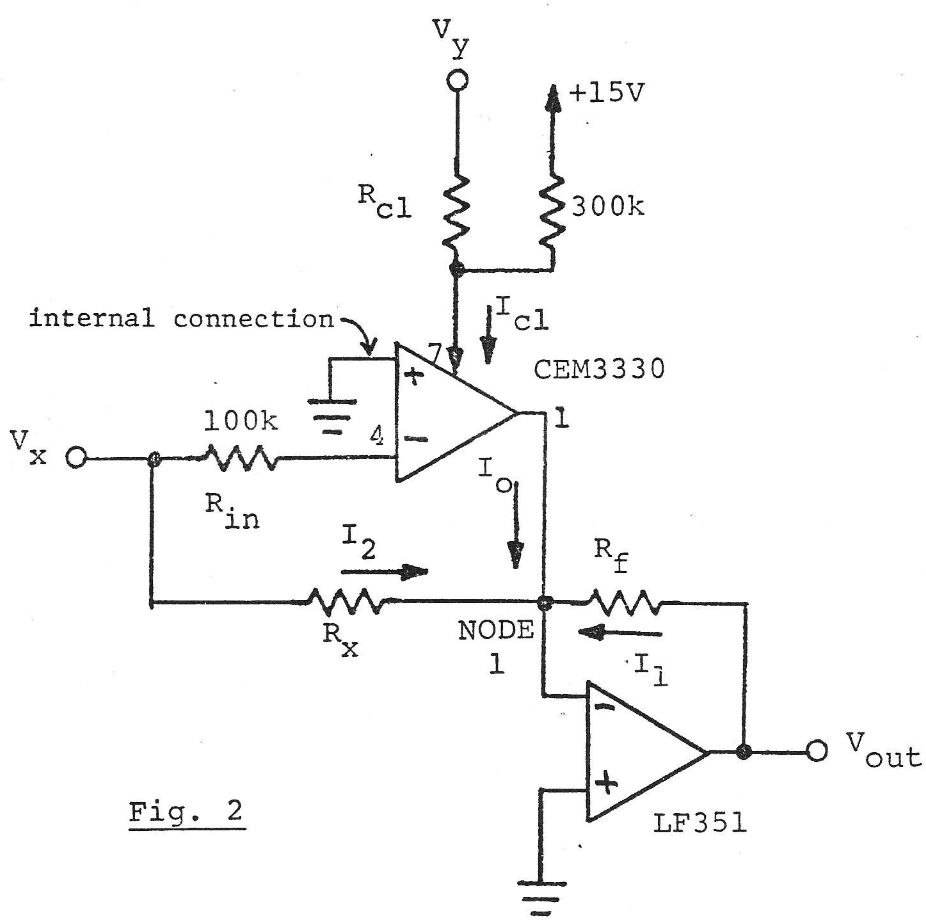


Fig. 2

or:

$$(V_{out}/R_f) = -I_o - (V_x/R_x) \qquad (5)$$

Substitute (4) into (5) noting that $E_{in} = V_x$:

$$(V_{out}/R_f) = (1/10)\ V_x I_{cl} - (V_x/R_x) \qquad (6)$$

Also note, (see figure two):

$$I_{cl} = (V_y/R_{cl}) + (15V/300K)$$

$$= (V_y/R_{cl}) + 50\mu A \qquad (7)$$

To see that equation (7) is valid, remember that pin 7 is at virtual ground. Substitution of (7) into (6), and simplification gives:

$$V_{out} = V_x V_y \ (R_f/10R_{cl}) + V_x \ (R_f) \ (5\mu A) - V_x \ (R_f/R_x) \qquad (8)$$

Let:

$$(R_f)(5\mu A) = (R_f/R_x) \qquad (9)$$

Then R_x = 200K, and substituting (9) into (8) gives:

$$V_{out} = V_x V_y \ (R_f/10R_{cl}) \qquad (10)$$

The scaling factor of the output should be 1/5, hence set:

$$(R_f/10R_{cl}) = 1/5 \qquad (11)$$

To keep I_{cl} within practical limits, (below several hundred microamperes), let R_{cl} = 100K. Then equation (11) implies that R_f = 200K, and hence from equation (10):

$$V_{out} = (V_x V_y)/5$$

as required.

The reader will notice that the output of the gain cell is converted to a voltage though an inverting stage. The CEM3330 has a limited output compliance, and likes to see a virtual ground (the inverting input of the op-amp). This has the nice side effect of making V_{out} non-inverted contrary to the case of the CA3080 in the original article.

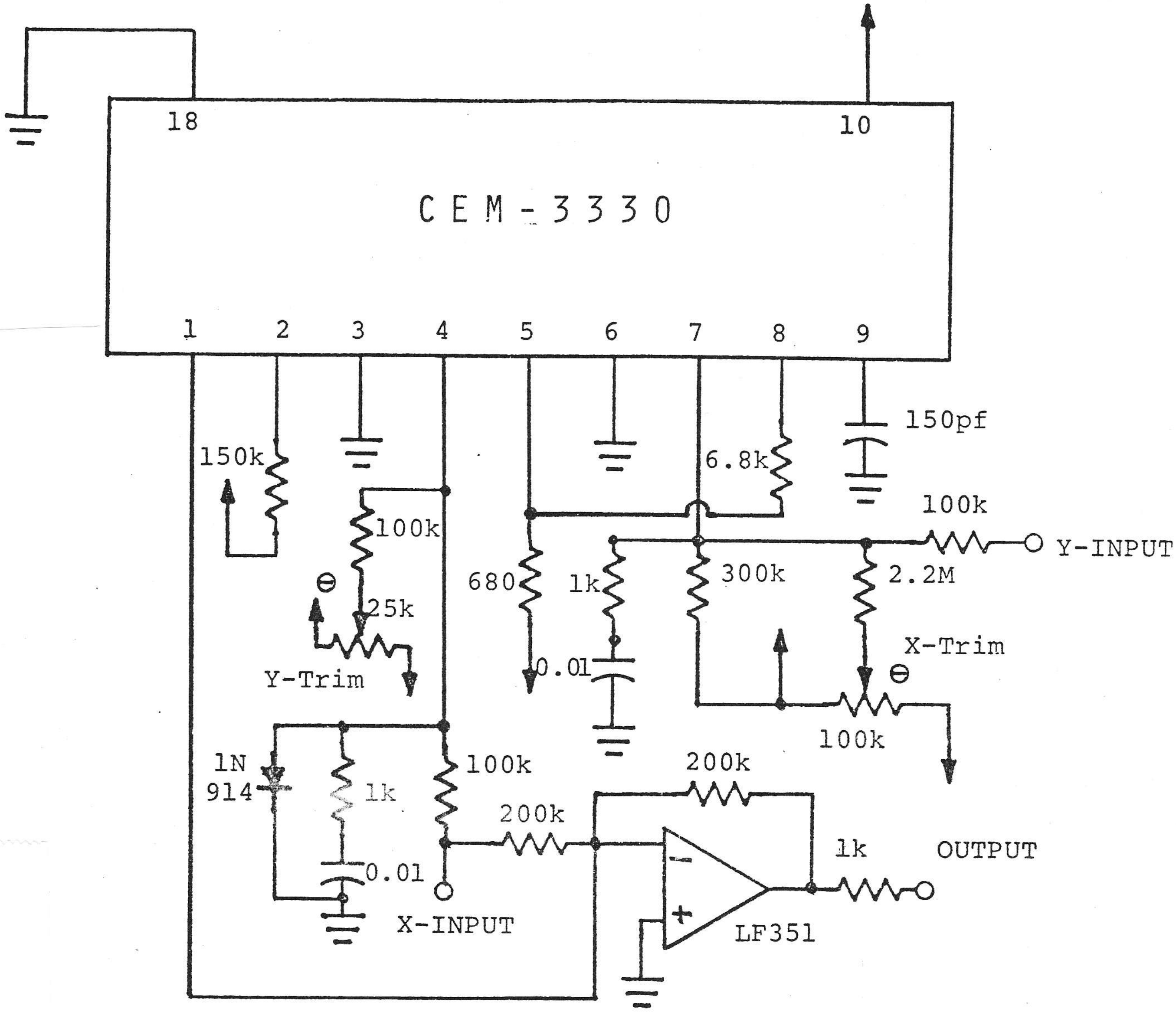
Figure three shows the final circuit, when all of the fine points have been taken care of. Note the improvements over the original design using the CA3080. The Y input has an input impedence of 100K, while the X input has an impedence of 66K, which while not standard, is nonetheless better than the original mentioned above. A further improvement is that either input may be AC coupled.

The trim procedure is fast and easy. Apply a signal to the X input, ground the Y input, and adjust the X trim for minimum leakage. Then apply a signal to the Y input, ground the X input, and adjust the Y trim for best rejection. Repeat the procedure as a final check.

The results were fairly good, and the unit really checked out, both for DC and AC multiplication. However, rejection of the Y signal (measured at 40 dB) still presents something of a problem. The X signal rejection is better at 55 dB. The author is certain that this problem could be improved upon, and would like to correspond with anyone concerning this, or related questions about the circuit.

Note that the CEM3330 doesn't respond to negative currents through pin 7. It simply shuts the gain cell off in this case. In terms of this circuit this means that the voltage to the Y input should never drop below -5V to avoid clipping.

Acknowledgement: The author is grateful to Mr. Stephen Julstrom, Recording Studios, University of Iowa, for his clear explanations of the internal workings of the CEM3330.



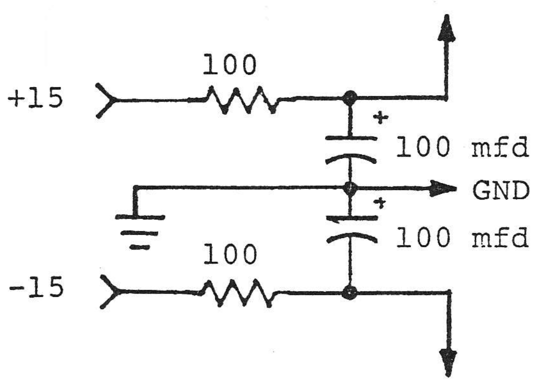


Fig. 3

FOUR-QUADRANT BALANCED MODULATOR

Making Rack Panels:

-by Thomas Henry

Fabricating attractive front panels for a syntheszer can be a real challenge at times, and so I want to describe a system I have evolved over the last couple of years. You will note that I call this a "system", for it is only by some systematic approach that consistent results can be expected. I have built some thirty modules using this scheme, and so have had quite a bit of time to learn from my mistakes. I hope then, that this article will save you from some of the pitfalls I have encountered.

I set fairly high standards for attractiveness when it comes to front panel design. Now, I know there are probably some readers out there who don't feel this strongly about the esthetics of front panels, but nonetheless this article may have a few tips for you too.

What I am going to reveal then, is a method for taking ordinary blank rack panels and turning them into front panels for synthesizer modules. The end result will be quite attractive, and could easily be mistaken for a "professional" job.

Perhaps the first question to consider is where to obtain materials. Well, we will need a blank rack panel, and they are fairly common. Places such as Burstein-Applebee and Allied Electronics are good places to find these. I think the prices might alarm you though! For example, a blank, double width panel goes for around five to eight dollars! (All rack panels are 19" wide; the height comes in multiples of 1 3/4". A single width is 1 3/4", a double width is 3 1/2", a triple width is 5 1/4", and so on.) Well, I don't know about you, but to me that seems kind of a rip-off; I mean, who ever heard of eight dollars for a little, skinny piece of aluminum?

I decided to check the alternatives, since I didn't want to play my home-brew synthesizer in the poor-house. I discovered the following interesting fact. Local sheet metal houses will cut you the right sized panels for anywhere from one-fifth to one-tenth the prices mentioned above. The place where I go will cut me single width panels for 65¢, double width for $1.30, and so on. Quite a savings!

Of course you will have to cut your own mounting holes, but for that kind of price break it's worth it. (It takes me about ten minutes to cut the four notched mounting holes.) When you go to the sheet metal firm, be sure to tell them that you want 1/8" aluminum stock, 19" wide by whatever panel height you have in mind. And you will probably make the man happier if you order more than one panel at a time, since he can then cut a big strip 19" wide, and then sever it down from there to the individual panels.

You may get a choice of quality of stock, in which case go for the least expensive (i.e., the worse looking), since in a later step we will be removing all the surface blemishes anyway.

Now a word about making mounting slots. Figures one and two give the basic ideas. In Figure one, four holes have been drilled or punched in the corners. Their diameter should be 1/4". An easy way to locate their positions is to use a standard store bought panel as a template. Or you can take the measurements off the rack case, or use some of the standard positions indicated in Figure three. (Figure four shows the standard spacing of holes for a rack case. The spacing is such that any two panels of any size may be positioned together.) Now, with a hack saw slots are cut into the holes as shown in Figure two. After this, a flat file and a rat-tailed file are used to smooth up the edges of the cuts.

After following the above steps, you will have a standard rack panel for a fraction of the normal cost. "The only trouble now", you say, "is that it looks ugly, all those nicks and scrapes!" Well, I told you not to worry. Simply load an electric sander with some fine sandpaper and go to it. Don't be concerned about the sandpaper "design" being etched into the panel, since they will be covered later by

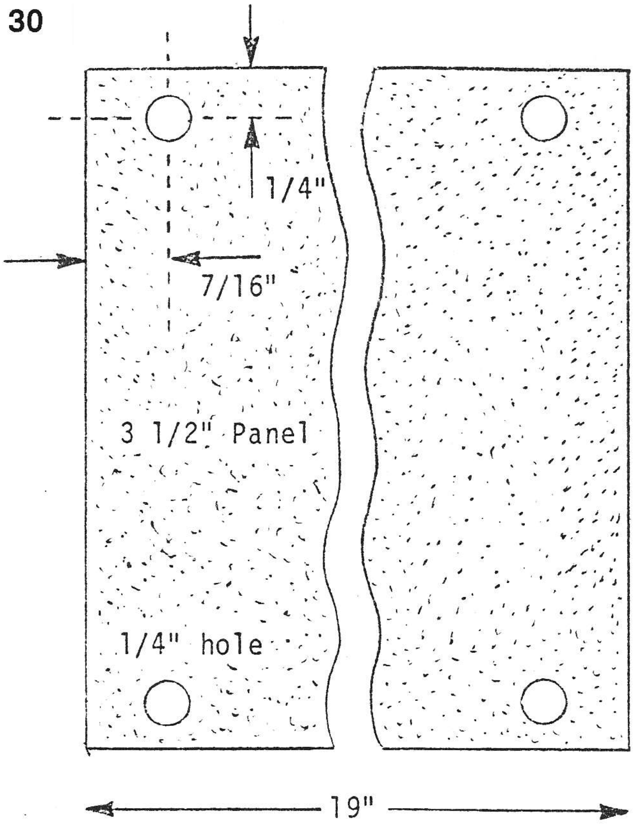


Fig. 1 Panel drilled with 1/4" holes. Spacing of holes is always 1/4" from edge or 1.5" from edge (See Fig. 3)

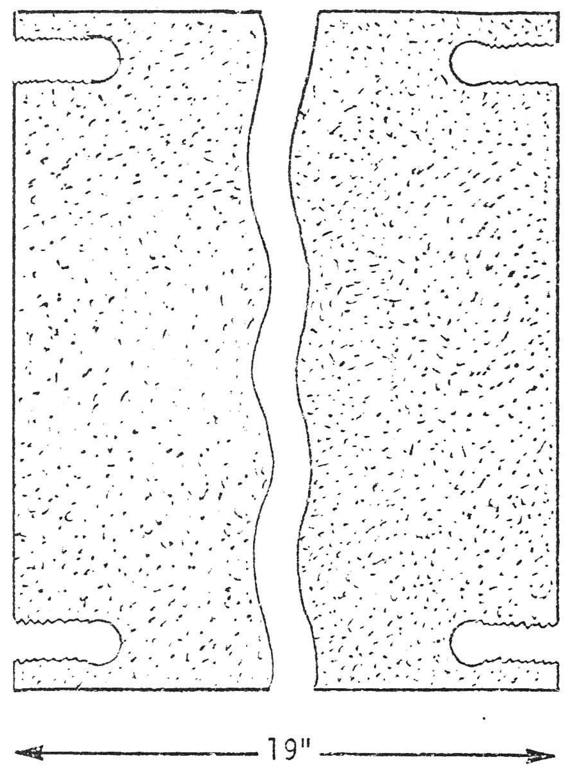


Fig. 2 Slots are rough sawed out. Cut slightly inside and smoothen cut edges with flat file.

some paint primer. But go at those nicks and scrapes, and after several minutes sanding you will have a smooth panel

The next step is to buff the panel with some 00 gauge steel wool. This will do wonders to the surface and will give it a smooth and silky feel. After buffing, wash the whole panel with soap and water to remove any grit and steel wool particles. Wipe dry with a lint free cloth.

Now we are going to put a temporary "cover" on the panel as an aid to the layout process. Using strips of masking tape, cover the entire panel front surface (decide which side you want to use for the front first). This step is easier if you use two inch wide masking tape. The general idea is to cover the front surface with a protective layer which will accept pencil markings.

Working from a life-sized mock up, prepared earlier (more about this later), mark the various spots on the panel (on the tape) where you will need to drill holes. Then with a metal punch and hammer, indent these various centers which you have just marked.

At this point you can drill or punch the holes in the front panel. If you use a drill be sure to make some pilot holes first, and in general be very careful since that metal can really cut if the drill bit should snag and start the whole panel spinning around. I always clamp the panel to my workbench on top of a piece of scrap lumber and this holds the panel firmly in place.

After the holes have been made you can remove the tape. We are almost ready to paint the panel, but first you should clean the surface with a cotton ball and some alcohol. The alcohol serves to clean off any soap film from an earlier step, and will also clean off any adhesive left from the masking tape.

We can now paint the panel with some primer grey. Don't neglect the primer, since it has the property of smoothing out or filling in any small abrasions.

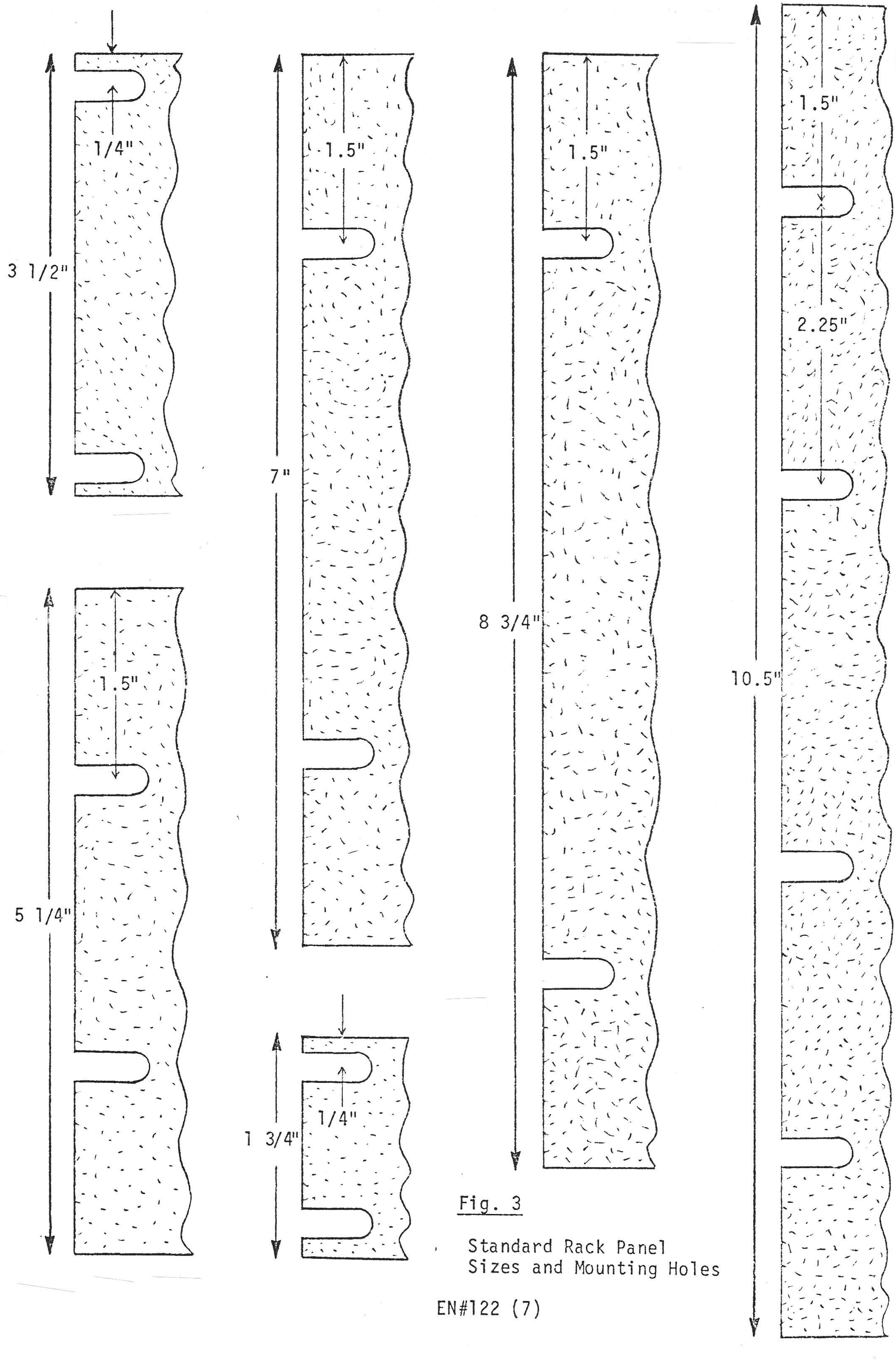


Fig. 3

Standard Rack Panel
Sizes and Mounting Holes

The end result will be very smooth and silky to the feel. When the primer is dry, paint the panel with any color spray paint that you like. Epoxy spray paints give an extra hard surface and are well worth the extra cost. I use a pale yellow color since the dry transfer letters I use show up very nicely on it. Black is also a good color if you happen to have access to white transfer letters.

At this point, you can label the panel. Dry transfer letters give a very nice touch and are really quite easy to use. If you make a mistake, simply lay a piece of masking tape on the error and lift it up, and the letters will lift up on it. In the case of VCO's, you may want to do what I do and actually draw the output waveforms below their respective jacks. I use a special pen for this. The pen is an alcohol base affair, with a narrow tip, and it is available at Radio Shack. The intended use for the pen is resist ink work (for printed circuit boards). In addition to the pen I use a standard schematic template to generate the various shapes. Let me tell you, the use of the template and that special pen has really classed up my modules, and it's really impossible to tell that the various symbols were created by hand. There are all sorts of tricks you can pull to create graphics and you will discover more as you go along. When you are done laying out the panel to your satisfaction, spray paint the surface with clear plastic or lacquer. Allow this to dry and spray again. And one more time, spray again. The purpose, of course, is to create a very hard protective surface on top of those relatively fragile dry transfers. Dispite this protection, you should take extra care when mounting jacks, pots, and the other controls. A slip of a pliers or wrench while tightening a nut can easily mess up your hard work.

At this point you are done with the panel and the results will be durable and attractive. I think you will be surprised at how the panel no longer betrays its heritage; no one will ever know that it was home made and that it started out as a piece of ordinary aluminum.

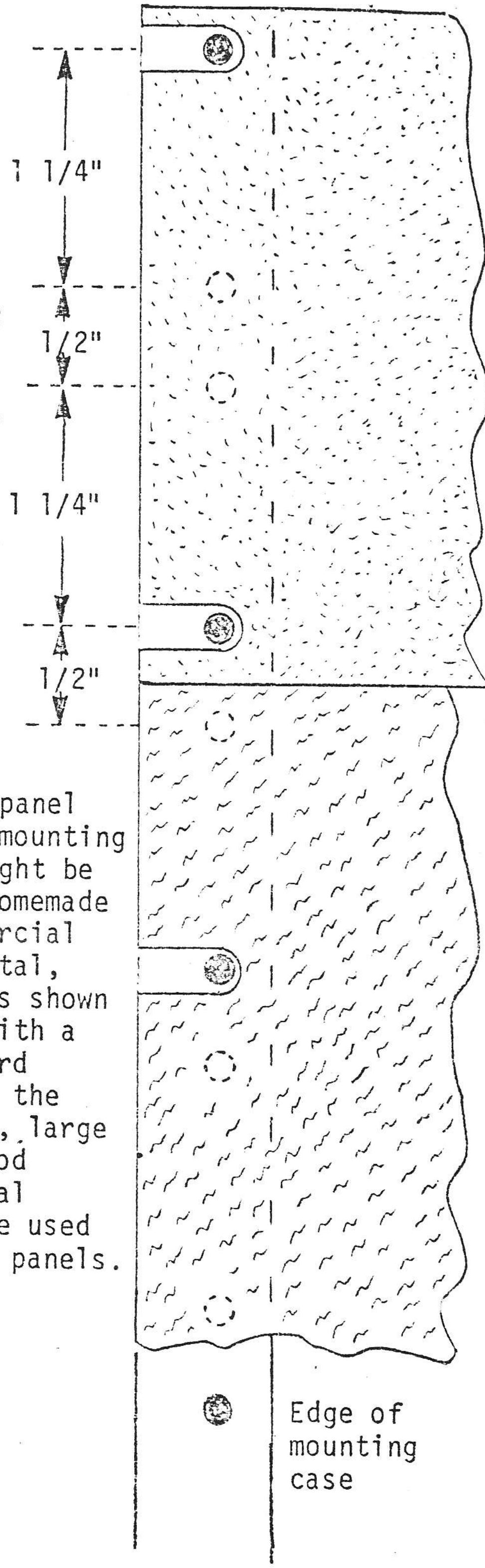


Fig. 4 A portion of the left side of the panel mounting case showing a 3 1/2" panel mounted with a larger panel below. Note how the sequence of holes spaced at alternating 1 1/4" and 1/2" spacings allows for any sequence of panel sizes. The mounting case edge might be wood for a homemade case. Commercial racks are metal, and the holes shown are tapped with a 10-32 standard thread. If the case is wood, large flat head wood or sheet metal screws can be used to mount the panels.

I mentioned above that you should lay out the panel from a mock-up. It is essential to prepare for this by doing a full-scale drawing of how you want the panel to appear. By doing it full scale, you can play around with the actual pots, jacks, and switches themselves to see how they fit together. It has been my experience that this mock-up is the most important part of the whole thing.

There you have it, then, my guaranteed method for making panels. It may seem like a lot of work, but you will find that if you have a clear idea of what's involved, you can actually go through many of the steps without thinking! The night before I ever build anything, as I lay down to sleep, I visualize in my mind just what I will be doing the next morning. I find that this mental preparation makes the actual work go very fast, since I am essentially going through the motions. After all, I've already built the thing (in my mind) so what's so hard about building it again? Who'd have ever thought "Psycho-Cybernetics" would ever come to electronic music?

The point then, is to develop a method. I hope this article will help you to develop your <u>own</u> method, and I hope your front panels will please you as much as mine have pleased <u>me</u>. Happy building!

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A FUNCTION GENERATOR WITH "QUADRATURE" TRIANGLE WAVE OUTPUTS:

-by Thomas Henry

Most musical engineers are aware by now that it is fairly easy to fashion a quadrature oscillator out of a voltage-controlled filter simply by setting the resonance of the filter high enough to cause oscillation, and then tapping the required sine and cosine outputs off the proper points [1, 2, 3]. The chief advantage of this method, (besides its simplicity), is that the resulting waveforms are very pure. However stability can be a problem if very wide sweep ranges are attempted. By stability, it is meant that both amplitude and frequency are fairly sensitive, and in fact, in most quadrature oscillators outputs tend to be a function of frequency. In many applications such small deviations in amplitude may not matter, but for precision applications a function generator approach may be more suitable.

While it is fairly difficult to derive both sine and cosine outputs from a function generator, it is easy to generate two triangle waves 90 degrees out of phase with one another. This is because triangle waves may be represented as a collection of line segments properly connected. The method used in this article then, is to take a triangle wave, disassemble it into its component line segments, then reassemble those segments into a derived triangle wave 90 degrees out of phase with the original.

Before looking at the circuit, an analysis of the method employed is presented. A triangle wave is shown in Fig. 1. Note that it has an amplitude of 10V peak to peak, and that it is drawn on a time scale divided into multiples of five. This will simplify the arithmetic later and in no way leads to a loss of generality.

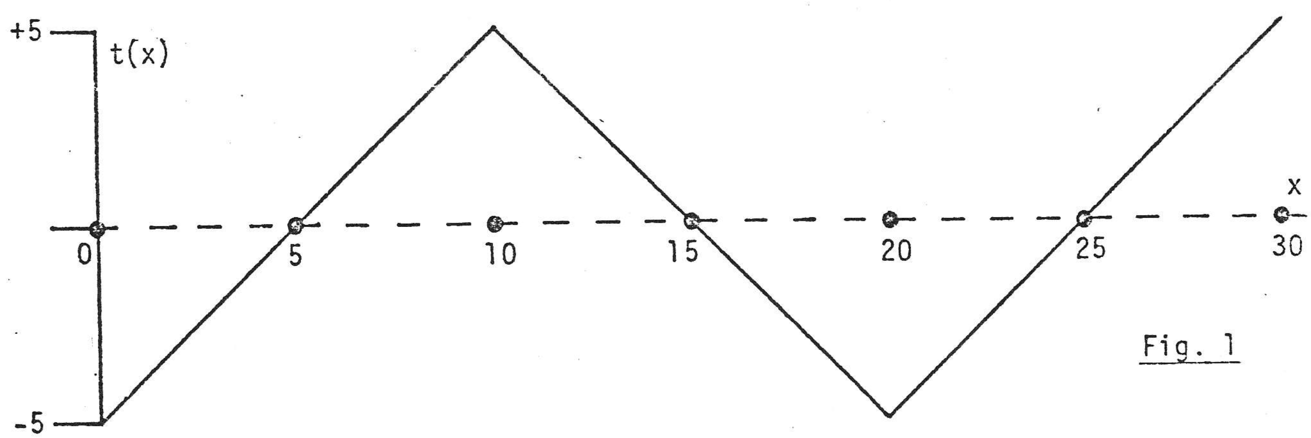


Fig. 1

The triangle wave of Fig. 1 may be represented as the function:

$$t(x) = \begin{cases} x - 5, & 0 \leq x < 5 \\ x - 5, & 5 \leq x < 10 \\ -x + 15, & 10 \leq x < 15 \\ -x + 15, & 15 \leq x < 20 \end{cases}$$

Thus it is seen that the triangle may be represented as a collection of line segments connected at the proper points. That the above expression really represents the triangle wave in question may be proven by some easy arguments from analytic geometry.

Parenthetically it should be remarked that musical engineers usually think of triangle waves in terms of Fourier series. However, since this circuit to be described is non-reactive it makes sense to treat the triangle wave as a collection of line segments and not as a series of sines of different frequency.

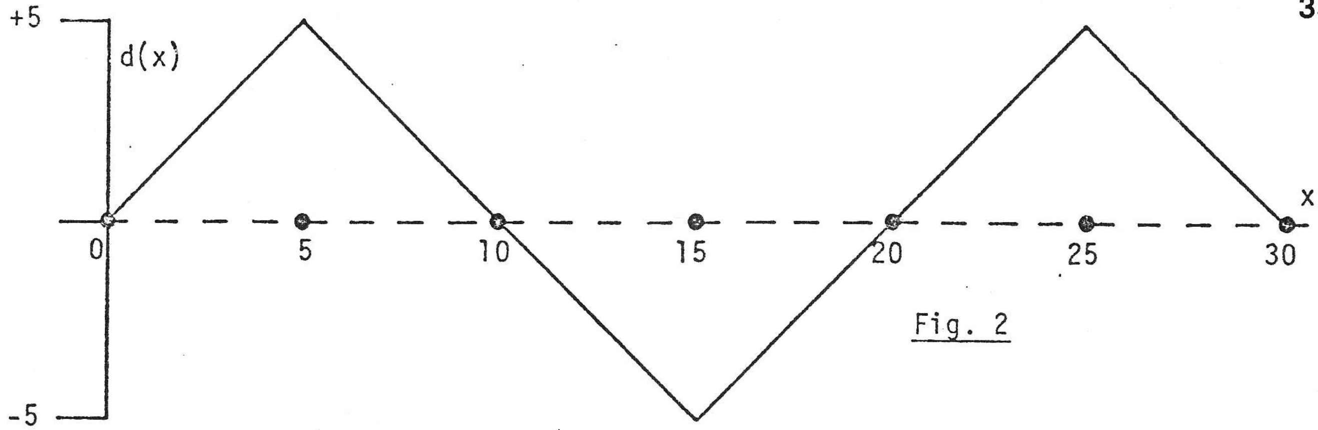


Fig. 2

Using the same argument as above, the phase shifted triangle of Fig. 2 may be represented by the function:

$$d(x) = \begin{cases} x, & 0 \leq x < 5 \\ -x + 10, & 5 \leq x < 10 \\ -x + 10, & 10 \leq x < 15 \\ x - 20, & 15 \leq x < 20 \end{cases}$$

where d(x) stands for "derived triangle,"

Clearly what is wanted, then, is a "black box" that will take t(x) and transform it into d(x). The black box in fact turns out to be another function and the process used is known in mathematics as "composition of functions." That is, if the transformation function is $\delta(t)$, then it is desired that $\delta[t(x)] = d(x)$. The proper function is given by:

$$\delta(t) = \begin{cases} t + 5, & 0 \leq x < 5 \\ -t + 5, & 5 \leq x < 10 \\ t - 5, & 10 \leq x < 15 \\ -t - 5, & 15 \leq x < 20 \end{cases}$$

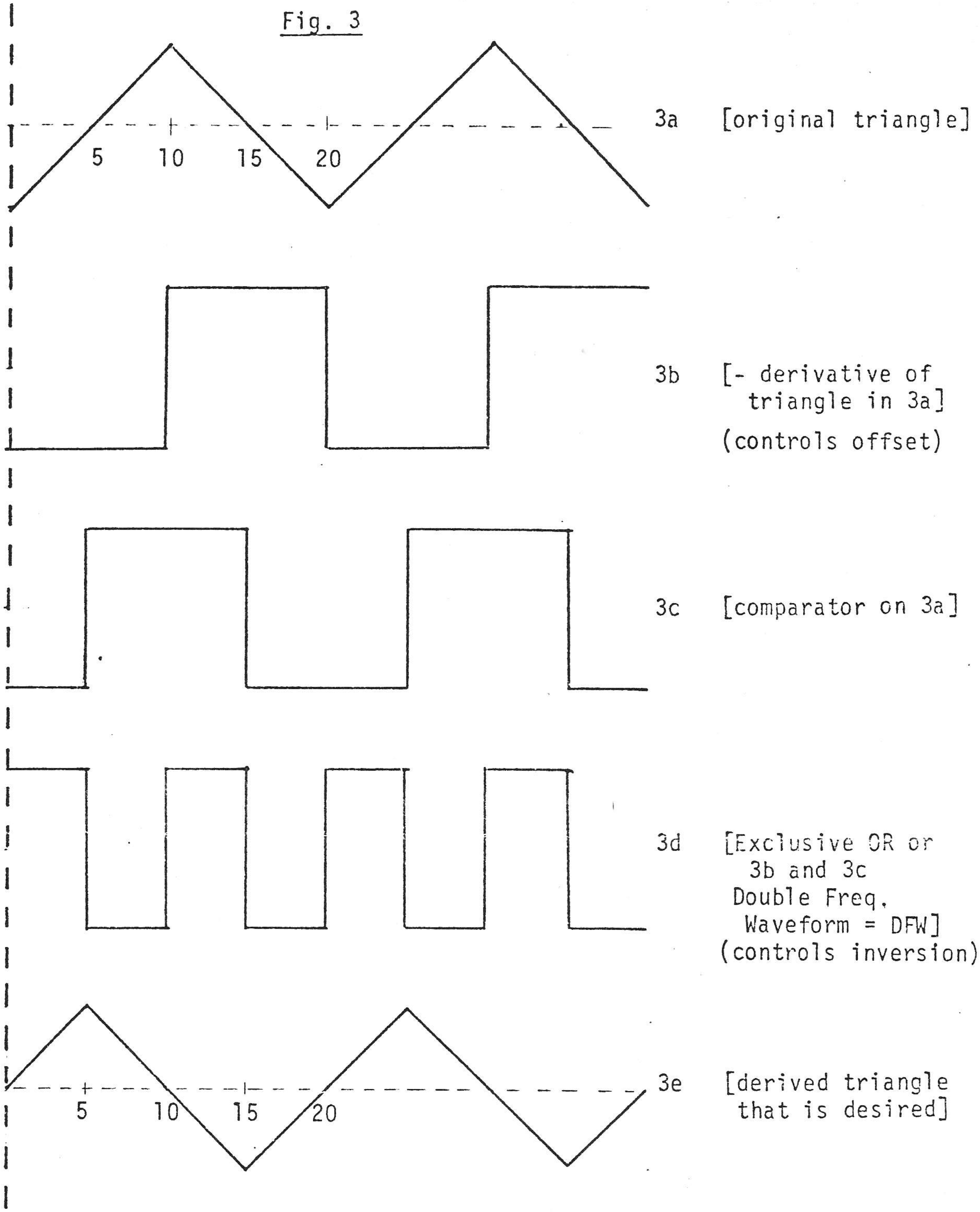
The reader should confirm that $\delta[t(x)] = d(x)$ on all four intervals of the functions. Also, remember that while $\delta(t)$ is a function of t, it is also a function of x, since t is a function of x. It is convenient then to express $\delta(t)$ in terms of x; i.e., $0 \leq x < 5$, $5 \leq x < 10$, etc. Also implicit in the method of this article is the fact that it is necessary only to consider the various functions over the four intervals listed above; the results can obviously be extended to all other intervals by noting that a triangle wave is periodic.

Examining the function $\delta(t)$ it is seen that a function is needed that combines sign changing, adding an offset, and subtracting an offset in various combinations and on the proper intervals. For example, on the interval $0 \leq x < 5$, t(x), the original triangle wave is left unchanged except for adding a 5 volt offset to it. In the next interval, $5 \leq x < 10$, the original triangle wave is inverted and then a 5 volt offset is added. In the next interval the sign of t(x) is left the same but a 5 volt offset is subtracted. And finally in the last interval the triangle wave is inverted once again and then a 5 volt offset is subtracted.

Obviously the only hard part about this is making sure that the inversions and offsets occur at the proper times and in the proper intervals. So now a few words about the timing functions is in order.

Fig. 3 shows the timing diagram for the method. Fig. 3a shows the original triangle, while 3b is the negative of the derivative of this triangle. Fig. 3c shows the square wave resulting when the original triangle is fed to a comparator. Fig. 3d is a pulse wave that is derived by feeding the derivative and the comparator, mentioned above, to an Exclusive-OR gate. Note that this pulse is exactly twice the frequency of either the derivative or the comparator waves. Finally

Fig. 3



3e shows the desired triangle wave. It turns out that the derivative waveform and the double frequency waveform are the only ones needed to time the inversion and offset functions to create $\delta(t)$.

As an example, look back at the expressions for $\delta(t)$. During the first interval the sign of t(x) is left unchanged. This corresponds to the double frequency waveform (DFW) being high. At the same time an offset of 5 volts is added on, and this corresponds to the derivative being low.

In the next interval ($5 \leq x < 10$), t(x) must be inverted and this corresponds to the DFW being low. Once again a 5 volt offset is added, and as before this corresponds to the derivative being low.

In the interval $10 \leq x < 15$, t is non-inverting again, and as in the first interval this corresponds to the DFW being high. In this interval however a 5 volt offset is subtracted, and the derivative switching high indicates this. The reader should confirm that the timing waveforms lead to the proper results in the fourth interval.

It can be seen that to create a "black box" capable of generating the function $\delta(t)$, all that is needed is a sign changer, an offset adder or subtractor, and some means of generating the proper timing waves (derivative, comparator, and DFW). The circuit presented in this article incorporates all of the above.

Before putting it all together and examining the final circuit, it will pay to analyze the sign changer since as op-amp circuits go it is fairly unusual. The circuit is based upon an idea for a manually operated sign changer [4]. In the schematic of Fig. 5, the sign changer is comprised of A5, Q1, and related components. Note that Q1 is set up as a switch and either shorts the (+) input to ground or else presents a large impedance to ground. Fig. 4 illustrates these two possible cases.

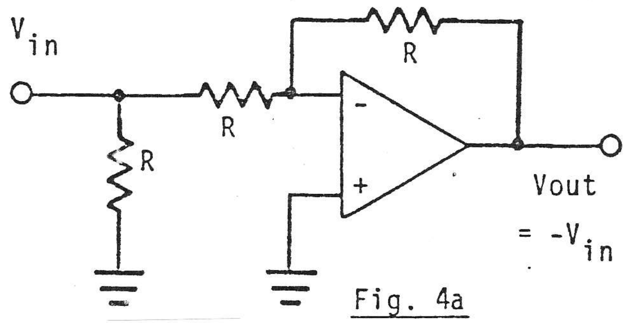


Fig. 4a

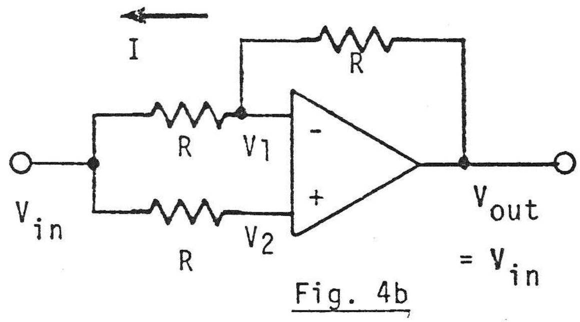


Fig. 4b

In Fig. 4a, the (+) input is grounded and so it is seen that the amplifier is nothing more than a simple inverting amplifier with a gain of -1, and an input impedance of R/2. When the FET, Q1 of Fig. 5 is off, the equivalent circuit of Fig. 4b results. To analyze this circuit start by observing that:

$$V_{in} = V_2 \tag{1}$$

and since negative feedback is present and working:

$$V_2 = V_1 \tag{2}$$

Now a current I is flowing* so:

$$I = (V_1 - V_{in})/R \tag{3}$$

The current can also be expressed:

$$I = (V_{out} - V_{in})/2R \tag{4}$$

Combining (3) and (4) and solving for V_{out} gives:

$$V_{out} = 2V_1 - V_{in} \tag{5}$$

and than substituting from (1) and (2) it is seen that:

$$V_{out} = V_{in} \tag{6}$$

It is seen then that the sign changer really works; it has a gain of either 1 or -1 depending on whether the FET switch is on or off. With this subcircuit out of the way it is possible to examine the complete circuit for a function generator with "quadrature" triangle outputs.

Fig. 5 shows the complete schematic. The heart of the circuit is the Integrator-Schmidt trigger triangle wave generator comprising A1, A2, and associated components. Little will be said about this since this sort of function generator has been treated extensively elsewhere [5, 6]. However several

*note in fact that this current is zero, which suggests an interpretation based on physical considerations.

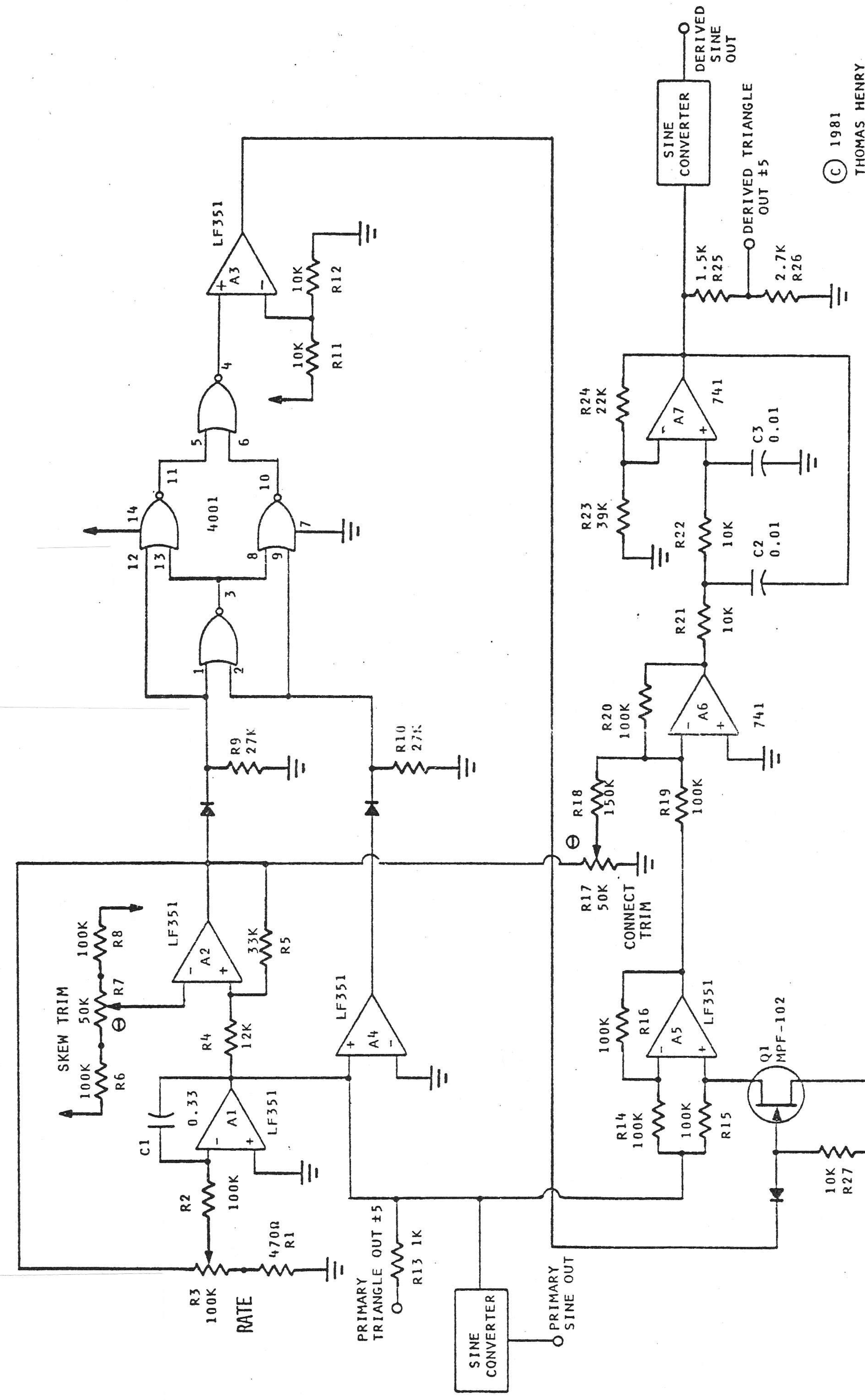


Fig. 5 Complete Schematic of Quadrature Oscillator

special points should be noted. First the output of A2 is a square wave swinging between -15 and +15 volts. This isn't just any square wave, but is in fact the negative derivative of the triangle waveform appearing at the output of A1. As mentioned above, the derivative provides important timing information to the "black box" transformation device. It was also seen that a comparator output was needed as well for timing considerations (see above) and that is the purpose of A4.

The derivative and comparator outputs are suitably restricted for CMOS by some diodes, R9, and R10, and then are sent to an Exclusive-NOR gate formed of the simpler gates in the CMOS 4001.* Note that the 4001 runs off of a power supply of +15 volts and ground, whereas everything else in the circuit needs a bipolar supply. The Exclusive-NOR gate output is high if the two inputs are the same, otherwise it is low. Thus it can be seen that this gate is nothing more than a logical equivalence tester. The output of the gate is a double frequency waveform, and is illustrated in Fig. 3d. The output is sent to comparator A3 which restores the bipolar swing needed later on.

It should be noted at this point that all the required waveforms as indicated in the mathematical analysis above are now present: the original triangle, the negative derivative, the comparator, and the double frequency waveform. One other subtle point is that A6, whose job it is to sum the various segments and offsets together, is an inverting stage. Therefore the offsets are summed in the reverse to the sense mentioned above, and the sign changer is also "backward." After the inversion, everything else is back in order. So as the reader analyzes the circuit with respect to the mathematical analysis given earlier, he is urged to remember that A6 inverts the final waveform, at the very last step, and thus puts everything back in order.

The sign changer composed of A5 and Q1 was treated earlier. The only other thing to note is that the double frequency waveform goes to the FET switch by way of the diode. When the diode is forward biased, the FET is off and the sign changer becomes an amplifier with gain 1. When the FET is on the sign changer becomes an amplifier with gain -1.

The output of the sign changer is sent to summing amplifier A6. Also summed into A6 is the plus and minus 5 volt offset. This offset is sent via R17 and R18. Note that this offset comes from the derivative waveform as required.

The derived waveform, a triangle wave 90 degrees out of phase with the original is then available at the output of A6. However there is one minor problem left to resolve. Obviously if the circuit were built with ideal comparators and op-amps it would work perfectly right away. However, due to the finite rise and fall times of the comparators, small glitches do occur in the derived triangle. Fig. 6 shows where these may occur. This illustration has been greatly exaggerated to display the glitches, and in the author's prototype the glitches were very small, on the order of several microseconds wide. There are several things that may be done to minimize these glitches.

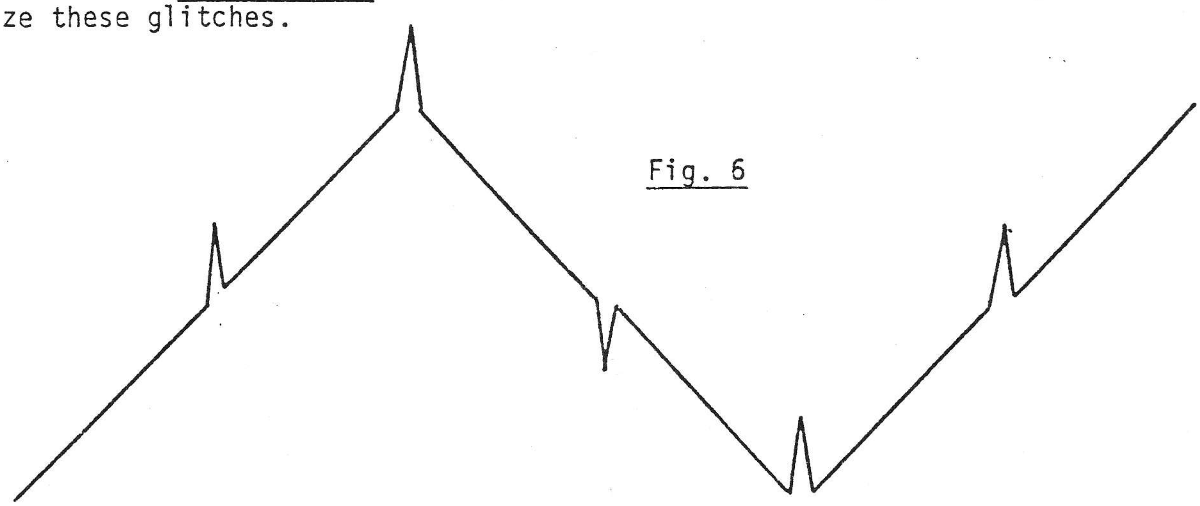


Fig. 6

*a standard Exclusive-OR (e.g. 4070) could be used here if the inputs to A3 are reversed.

First, good op-amps with very high slew rates should be used for A2, A3, A4, and A5. In the author's prototype LF351's were used and these have a slew rate of 13V/microsecond. Next a very slow op-amp should be used for A6. The author used a 741 type with a typical slew rate of 0.5V/microsecond. Thus the relatively poor slew rate of the 741 is put to good use and this does quite a bit to minimize the glitches. In fact, an interesting experiment worth performing is to substitute a LF351 for A6 and then compare the results with a 741. The 741 really smoothes out the glitches.

It was felt that the above two remedies weren't enough and some more cleaning up of the glitches should be attempted. A second order Butterworth lowpass filter was appended and this then cleaned up the glitches completely. The filter is comprised of A7 and its associated components. The filter has been chosen to have a cutoff of about 1600 Hz. Since the function generator as drawn has an upper limit of oscillation of 20 Hz, this means that the cutoff frequency of the filter is up around the 80th harmonic of the triangle wave. The output of the filter is still a very good triangle wave then, and since the glitches are only microseconds long, they are for all intents and purposes completely removed.

As just mentioned, the range of this function generator was made fairly low in frequency. With the values shown in the schematic, the author's prototype oscillated from 0.1 Hz to 20 Hz. Thus the oscillator would primarily be used for control purposes. Different ranges are of course possible with C1 being the chief frequency determining component. However, one must watch the high range, since a frequency whose inverse comes too close to the glitch period will be increasingly more difficult to filter properly without distorting the triangle wave. Of course if a frequency change is made, the filter cutoff should be moved accordingly.

Lest the reader think that the glitch problem invalidates this approach, it should be said that the application the author had in mind for the circuit was creating Lissajous figures on an oscilloscope screen. This application requires much more stringent purity of waveform than some others, and the circuit here works quite well. Obviously then, the circuit will work extremely well for audio applications where such stringent specifications are not needed. It turns out that the eye can see the glitches even when they are very small in a Lissajous figure, but the ear cannot hear them at all if they are reasonably well filtered.

One small point to be mentioned is that the original triangle wave has amplitude ±5 whereas the derived triangle has amplitude ±8, since the Butterworth filter has a gain of 1.6. Resistor divider R25 and R26 chops the triangle down to a ±5V level and also provide an output impedance of about 1k.

Adjusting the trim pots, R7 and R17, is quite easy, although it is difficult to describe in words. First replace C1 temporarily with a 0.05 mfd capacitor. This is to get the frequency up to a range where it is reasonable to watch on an oscilloscope. Next monitor the derived triangle at the output of A6. While watching the waveform, go back and forth between trimmers R7 and R17 until the various line segments connect to form a triangle wave. R7 governs "skew" of the triangle while R17 governs how well the segments connect. Though this may read poorly, the reader will see immediately how to perform the trims while watching the scope.

How well does the circuit work? The only answer is "very well." The practical obeys the mathematical quite well, and the glitches turned out to be very easy to control. The next step from here, naturally, is to come up with a voltage-controlled version. And perhaps for extremely detailed applications more work could be done on controlling the glitches over a wider range. The author would be glad to communicate with anyone who is interested in these aims or with anyone with questions about the circuit in general, and I can be reached at the Electronic Music Studio, MB2055, Univ. of Iowa, Iowa City, IA 52242.

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TRI-COLORED LED INDICATES VOLTAGE PEAKS: -by Thomas Henry

Some modules commonly used in synthesizer systems have sensitive input structures that can be damaged by excessively high input voltages. A common example is the analog delay line, which uses relatively fragile and expensive bucket brigade devices. The input voltages are normally attenuated to a safe level, but there may be times when several inputs are summed together and the voltage peaks may add up to form a dangerously high input. What is needed is some sort of peak level indicator to warn the user of such conditions.

The following circuit presses a tri-colored LED into service as a voltage peak indicator. Under certain conditions this LED can shine green, red, or a yellow-orange hue. Such an LED actually contains two separate LED's inside the same package; one is red and one is green with the respective anodes and cathodes facing in reverse directions For example, when the green LED is lit, the red one is reverse biased, and hence emits no light. And when the red one is lit, the green one remains dark. Apply an anternating current pulse wave to the tri-colored LED and both LED's are on alternately. If the duty cycle of the pulse is varied from 0% to 100%, the color emitted will vary from green to yellow-orange (at 50%) to a solid red. At 50% duty cycle the LED's are actually flashing back and forth from green to red, but if the alternation is of a sufficiently high frequency, "persistence of vision" will blend the two colors into a "solid" yellow-orange.

What is needed then is a circuit that linearly transforms voltage peaks into a variation of duty cycle. Figure 1 gives a block diagram of the circuit needed to accomplish this. -

First the audio signal is sent to a peak detector. The output of the peak detector is a DC voltage representing the highest peak seen so far. Since most synthesizers use a 10V p-p audio signal level, it is clear that the output of the peak detector can be a DC voltage anywhere from 0V to 5V peaks. This voltage is sent to a comparator where it is constantly compared with a triangle wave, hence giving a varying pulse output (pulse width modulation).

Fig. 2 gives the schematic. The peak detector is formed of R_1, R_2, A_1, D_1, and C_2. R_2 protects the diode at start-up and shut-down from dangerously high currents

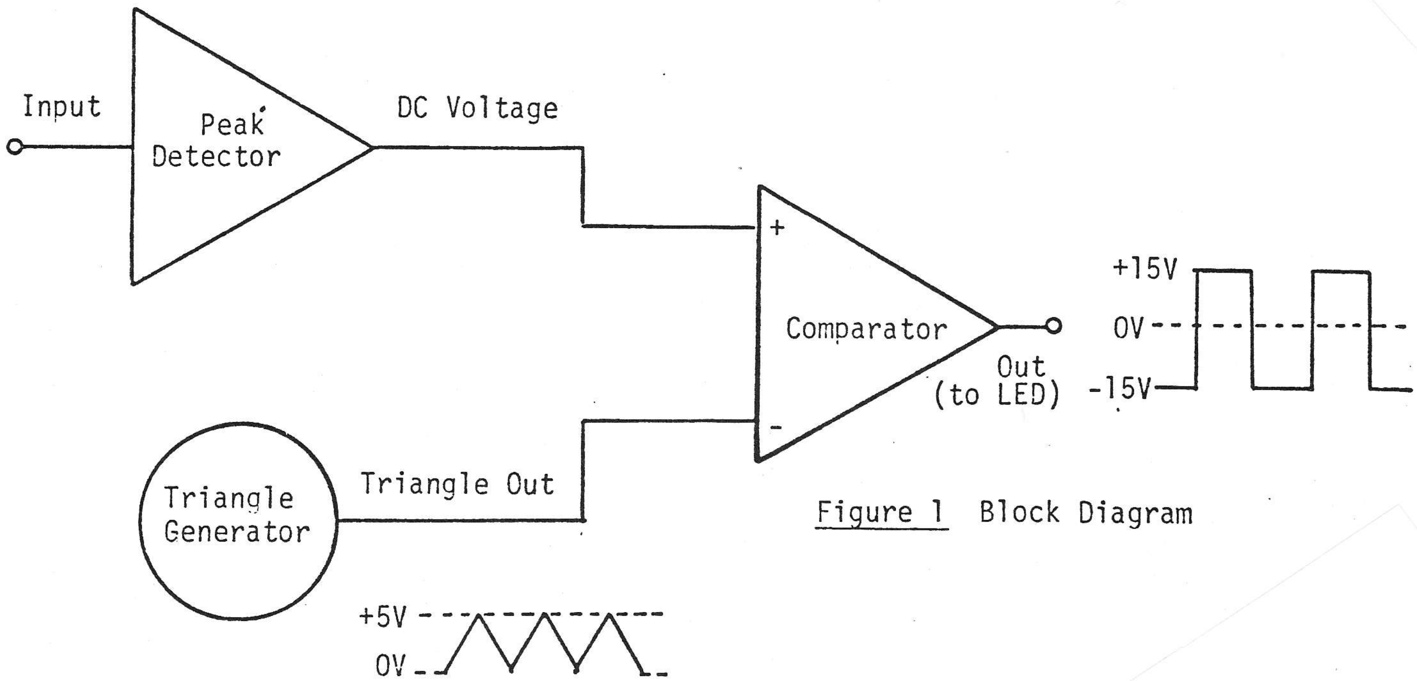

Figure 1 Block Diagram

(caused by C_2 charging and discharging rapidly), and R_1 balances the other input of A_1 with the same valued resistor. The peak charge is held on C_2, (D_1 prevents discharge), and obviously some path must be provided for "resetting" C_2 for another peak reading. R_3 provides this function by damping. A Value of 1M was chosen empirically, to strike

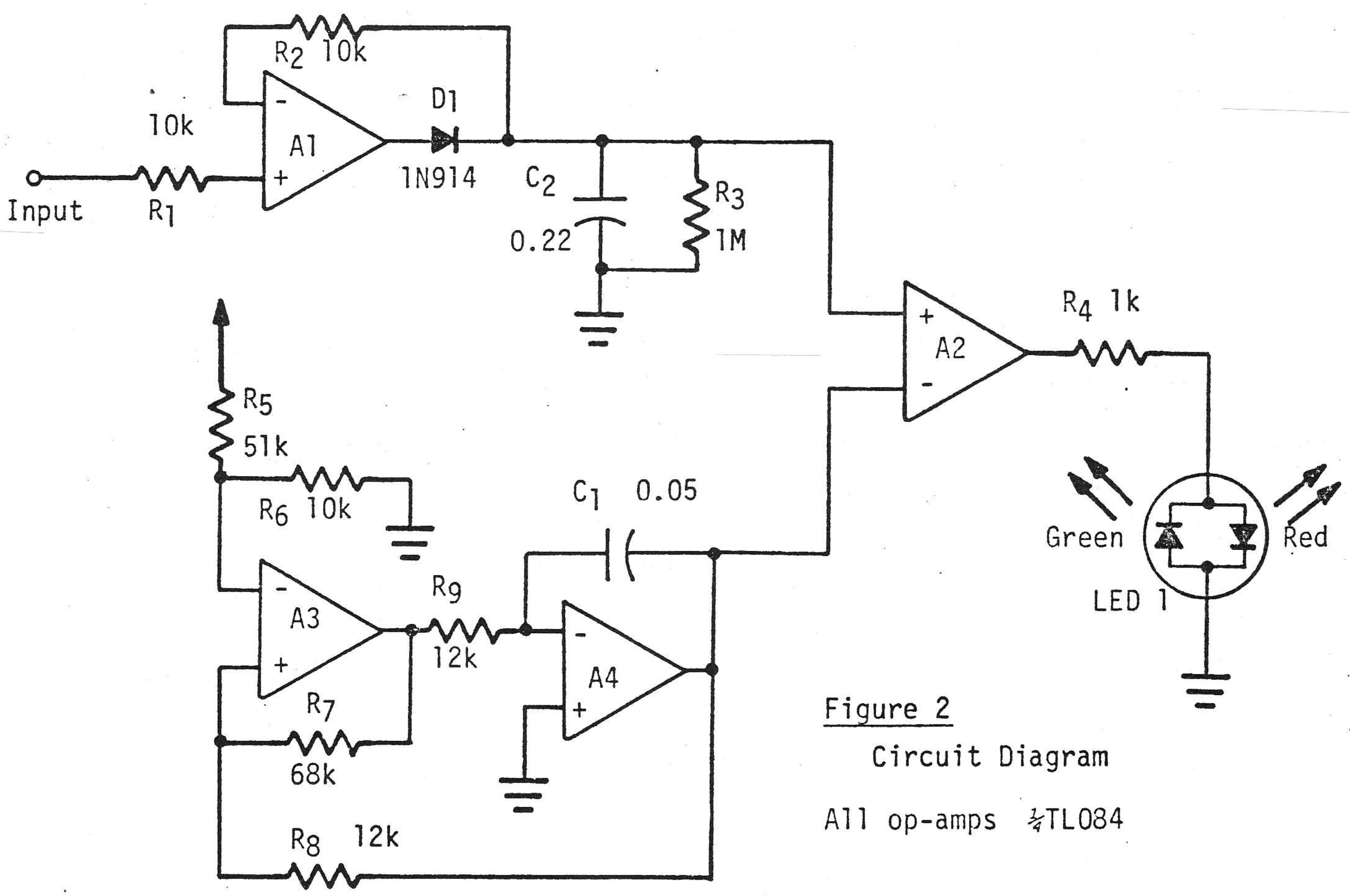

Figure 2

Circuit Diagram

All op-amps ¼TL084

a balance between fast response and a steady readout. You may feel free to experiment with this resistor however, to suit your own esthetic requirements.

The triangle wave is generated by the Schmidt-trigger-integrator function generator composed of A_3 and A_4 and associated components. R_5 and R_6 set the bottom of the wave form at 0V, while R_7 and R_8 set the amplitude at 5V maximum. R_9 and C_1 set the frequency at about 2 kHz. These values may all be adjusted to suit your needs.

The output of comparator A_2 swings between -15V and +15V, and so current limiting must be provided to LED 1. R_4 serves this purpose.

With the component values given, while the input is 5V p-p or less, the LED will glow green. From 5V p-p to 10V p-p the LED will start turning yellow-orange, until at 10V p-p and above the LED will turn bright red, warning of an overload condition.

The use of the tri-colored LED is especially timely now, since it had recently appeared on the surplus market. Bullet Electronics (PO Box 401244-F, Garland, TX 75040) stocks it for 60¢. In addition, Radio Shack stocks the part as well, making it readily available, but for considerably more at $1.39.

While the tri-colored LED is not as accurate a way to measure voltage levels as an LED bar graph, say, still the eye can readily detect color changes quite easily. For simple overload indications, such a scheme is sufficient. And you will note that the circuit is composed only of a quad op-amp and a handful of parts, making the economics of the circuit quite desirable, and worthy of inclusion in the design of any sensitive circuit.

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ELECTRONOTES B

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1 PHEASANT LANE

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GROUP ANNOUNCEMENTS:

CONTENTS OF SPECIAL ISSUE "B", EN#147, EN#148, EN#149, EN#150

MEET THE SN94281:

-by Thomas Henry, 249 Norton St., Mankato, MN 56001

Some time ago in the pages of Polyphony, I discussed the SN76477 complex noise generator [1 - 3]. In those three articles I demonstrated how to use the chip in a variety of non-standard ways. As it turned out, the chip was found to be excellent for quite a few serious synthesizer projects. Well, the news is out; the SN76477 has a baby brother and he's a real powerhouse too! This baby brother, the SN94281, is also a complex noise chip, but unlike the big guy is available in a 16 pin DIP package and hence is considerably easier to use. Since such good luck was obtained with the SN76477, it behooves us to look more carefully at this new chip.

What's the SN94281 and how does it differ from the SN76477? For starters, the chip contains a VCO, LFO, noise source, digital mixer, VCA and regulator all in one package. This is a pretty fair complement of circuits by anyone's standards and what's more the 16 pin package makes it very easy to use. It differs from the big noise chip in that it lacks pulse width modulation, an envelope generator and variable noise clock control. And oh yes, one big difference is that it contains a complete audio power amplifier capable of driving a small loudspeaker!

It should be clear then that the big chip is perhaps best for microprocessor controlled systems, pinball machines, Super Controllers and anywhere a large number of sounds is needed. The SN94281, on the other hand, is at its best in dedicated sound circuits where small size and a built-in amplifier are important. As it turns out, for most of our purposes we will ignore the amplifier since most synthesizer use requires a 10V pp line level signal. Let's dig in and get familiar with the chip.

Fig. 1* presents a complete "application note" for the SN94281. This application note is very special though, for several reasons. First, it contains only the information you need to get the chip running in a synthesizer system. I haven't

*see pg. 33

tried to show every nuance of the chip; anyone who's going to build a pinball machine should go elsewhere! Secondly, unlike many application notes, this one is based upon real measurements, done with real instruments in a real home brew workshop. The various voltages were actually measured, waveforms viewed on an oscilloscope etc. There is almost no theoretical information presented here. Instead I have concentrated on giving you the real dope on using this chip.

Let's get started then. As mentioned above, the chip contains a complete audio power amplifier. Since amplifiers have a way of consuming great quantities of current and since we don't need it anyway, let's figure out a way to defeat it. Refer to Fig. 1 to see how this is done. Normally a speaker is capacitively coupled to pin 7 and that's the end of the matter. What we'll do is leave off the speaker, but put a 10k resistor to ground here. The chip, of course, is used to seeing an indictive load of about 8 ohms, but a resistive load of 10k will do just fine. The output is tapped off of this resistor, and varies in amplitude from 0V to about 2V pp, depending on what the internal VCA is up to.

For some applications this 2V pp-output will be all that is needed. However, for most synthesizer uses it is best to buffer the signal and also to amplify by a factor of five. This will give a low impedance source of 10V pp. Simply follow the output with a standard op-amp configuration having a gain of 5 and you're in business

Let's think about a power supply for the SN94281? This chip, like its big brother, has an internal Zener type regulator available. Thus a fairly wide range of power supply voltages may be used without much hassle. The spec sheet claims that the supply voltage can be anywhere from +7.5 to +10.5V and is applied to pin 6. Well this is a nice range of voltage. Unfortunately, the range doesn't include our standard +15V value, so some diddling must be done.

I took some measurements and found that the chip draws about 10 mA without any speaker/amplifier combination in use. After a little Ohm's law, it's clear that a dropping resistor of 560 ohms is just right. Take a look at this resistor. About 6V is dropped across it, hence 9V is present at pin 6. So with the addition of one inexpensive part, we have adapted the chip to +15V operation.

One very important point should be made here. This trick with the dropping resistor only works if you are defeating the internal audio power amplifier, as we have done above. With a speaker in operation, the current varies from 10 mA to over 20 mA and it's simply impossible to choose a resistor which will drop the right voltage for such a broad range. Or to put it another way, if you pick a resistor that works for a 10 mA drain, then it won't work for 20 mA drain and vice versa. And nothing in between works either! So, once again, do not use the dropping trick for speaker/amplifier type circuits. Instead use a 9V supply.

A 100 mfd. capacitor is used for decoupling and stabilization at pin 6. The need for decoupling should be obvious; who wants white noise running rampant? But the stabilization problem should be mentioned. I notice that the circuit was prone to spurious oscillations and noise problems without the inclusion of this capacitor. This is especially obvious when the VCA (to be discussed later) was very close to being turned off completely. After adding the 100 mfd. capacitor, the problem went away and I've never seen it since. Don't cheat on this; a circuit can't be said to work unless it works right!

By the way, if you need a regulated 5V supply voltage for other parts of the circuit, you can grab it off pin 4. This pin is the regulator output and is good for about 5 mA. That's not very much, so be sure to think in terms of CMOS for support circuitry.

Having taken care of the output conditioning and power supply requirements, we can now consider how to make sounds with the thing. Let's start with the noise source. The noise source is a digital, pseudo-random counter, and is very similar to the generator used in the SN76477. One big difference is that the clock rate is fixed. Therefore there are only two pins associated with the noise source: pins 1 and 2.

continues: pg. 12

Meet the SN94281 (cont. from pg. 3)

The noise characteristic is fixed, but by changing the resistor and capacitors on these two pins, different degrees of filtering may be obtained.

Don't expect wonders with this internal filter; it's very ordinary and offers a mere 6 dB/octave lowpass response. If you use a fixed capacitor at pin 1 and a potentiometer at pin 2, then consider the pot a tone control. In my experiments I didn't think that the filter had as good a range as the one in the SN76477, and it seemed like the response was somewhat warped. Be that as it may, for your initial experiments, try a 470 pF capacitor at pin 1 and a 500k pot at pin 2.

One excellent feature of this chip is that all of the inputs have been current limited. If you cast your mind back to the SN76477, you'll recall that we had to add 10k resistors in series with all the pot inputs. This isn't the case with the SN94281; just hook up one side of the pot to the chip and the wiper to ground and you're all set!

Normally you will want to use the main output at pin 7 (see above), but there are times when you may wish to extract the noise independently of the VCO and LFO. To accomplish this task, you may tap the noise off of pin 1. The signal swing here is 0V to +5V and if you want to interface this as a digital source to CMOS circuitry you can leave off the noise filter capacitor. The output will be a series of digital pulses of "random" width. If you're interfacing to audio circuitry, then you will have to buffer the output at pin 1. Use a Bifet op-amp in a standard voltage follower configuration for this purpose.

(continues on pg. 32)

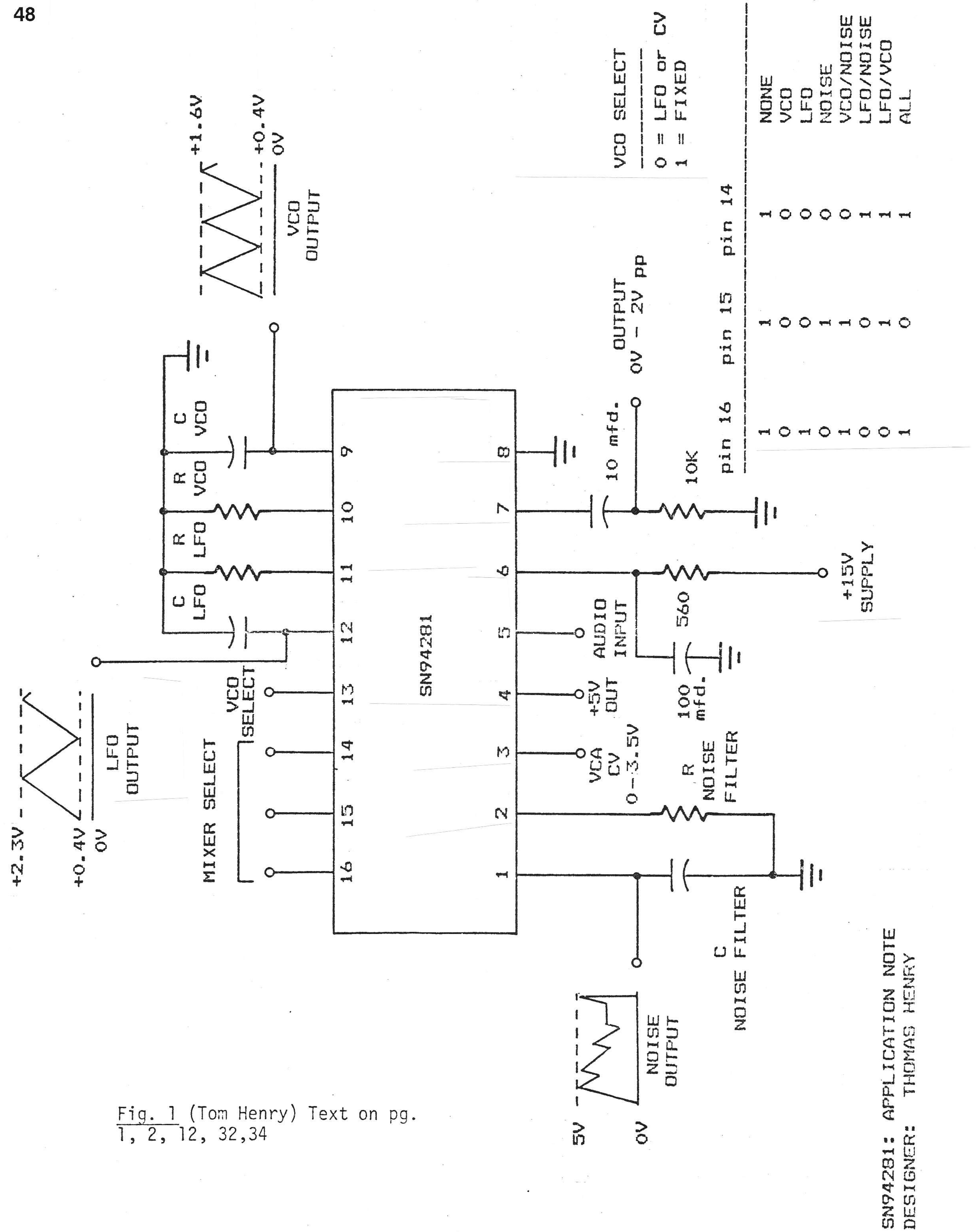
Meet the SN94281 (Cont. from pg. 12)

Consider the VCO next. The basic frequency is set by the VCO capacitor and resistor at pins 9 and 10, respectively. One obvious trick is to make the capacitor switch selectable, thus allowing a number of ranges. Similarly, the resistor could be made into a potentiometer, wired as a rheostat, for variable control of the frequency.

The VCO can be controlled in three distinct ways. First, if pin 13 (the VCO select pin) is high, then a fixed +1V control voltage is applied to the VCO. This happens automatically; you don't have to do anything. Thus a steady tone will be produced. I haven't thought of any good uses for this yet, but it's an interesting feature nonetheless.

If the VCO select pin is brought low, then the +1V reference is disconnected from the VCO. You are free to apply your own control voltage now, and you do this at pin 12. The control voltage versus frequency response isn't especially linear and is temperature dependent as well, so don't even dream of trying accurate keyboard control with it! Another quirk is that the response is inverted. A higher voltage means a lower frequency and vice versa. The control voltage should fall between 0V and +2.3V.

The third way to control the VCO is by letting the LFO frequency modulate it. This is accomplished by pulling the VCO select pin low, but instead of pumping in a control voltage, we let the LFO do the work. Simply hook up an LFO resistor and capacitor at pins 11 and 12, respectively. Note that pin 12 is quite an active pin.



pin 16	pin 15	pin 14	
1	1	1	NONE
0	0	0	VCO
1	0	0	LFO
0	1	0	NOISE
1	1	0	VCO/NOISE
0	0	1	LFO/NOISE
0	1	1	LFO/VCO
1	0	1	ALL

Fig. 1 (Tom Henry) Text on pg. 1, 2, 12, 32,34

It may be used as a control voltage input for the VCO (see above), a capacitor pin for the LFO, or an an output for the LFO. This doesn't mean that you can do all three things at the same time however!

As mentioned, you can tap the LFO triangle wave at pin 12. This will swing from about +0.4V to +2.3V and should be buffered with a Bifet op-amp. And, oh yes, you can also get at the VCO output via pin 9. The triangle wave here swings from +0.4V to +1.6V. It too should be buffered.

We've seen how to get the noise source, VCO and LFO signals independently, but it is also possible to make use of their "digital mix". The mixer, then, is not a true audio mixer, but is instead a three input AND gate. What goes into the mixer is determined by the mixer select inputs, pins 14, 15, and 16. Figure 1 shows a table which specifies what combination of logic input signals to use for the various combinations.

The mixer output is fed to the VCA. The VCA is controlled by a voltage on pin 3. A signal of 0V shuts the VCA off, while increasing this voltage gradually opens it up. +3.5V is the maximum control voltage and any increase beyond this yields no change in volume.

After the VCA, the signal is sent to an internal op-amp, and thence to the audio power amplifier. Pin 5 is the inverting input of the op-amp, and so external audio signals may be summed into the audio amplifier quite easily. Simply sum the signal through a resistor tied to pin 5. Start with a value of 220k for this resistor, and increase it if the signal is too loud. Otherwise refer to the manufacturer's spec sheet for the details on how to select this resistor.

It's an interesting concept, allowing the user to sum in external audio signals, but it's just too bad that the designers of the SN94281 didn't do the summing before the VCA stage. That way, all of the audio signals would have been amplitude modulated. As it stands now, the external audio input will always remain at a fixed amplitude.

I hope that you have enjoyed this "application note" on the SN94281. As mentioned above, this information is enough to get you started experimenting with the chip and is based on my own measurements. Due to the possibility of chip-to-chip variations, you should probably take some of the voltage measurements presented with a grain of salt. However, in general, variations of more than several tenth of a volt are unlikely.

Just recently Radio Shack started to carry the SN94281, so it should be within the reach of just about everyone. In fact, in my town, the Radio Shack had a half-price sale on the chip. Needless to say I bought a dozen of them! So keep your eyes open for deals like this.

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Build A Precision Controller Clock:

-by Thomas Henry, 249 Norton St., Mankato, MN 56001

Described herein is a circuit for a precision controller clock suitable for the most demanding low frequency applications. For realtime use, the controller clock is handy for firing sequencers, sample-and-holds and automatic drum units and in the recording studio it is equally useful for creating precise, repeatable sync tracks. In short, the controller clock may be used whenever a source of precisely timed triggers is needed. The features of this circuit are summarized:

- [] 1 volt per octave response
- [] variety of period control options
- [] periods from 2 minutes to 1 millisecond; seventeen octaves
- [] precision gating for start and stop of output
- [] manual or trigger gating modes
- [] standard gate, trigger and ramp outputs
- [] extremely precise 50% duty cycle
- [] simple design employing the CEM3340 VCO chip

The value of most of these features should be obvious, but several deserve a little more description. Precision gating implies that the clock should begin and end oscillation at well defined states only. For example, a start trigger should start the clock oscillating, with the very first cycle going high and maintaining standard pulse width. A stop trigger should gate the clock off, but only when the output has first gone low (end of cycle). A "memory" remembers that a stop is desired, and the next time the output goes low, the clock is gated off. So, precision gating means that the clock starts by instantaneously initiating a normal cycle and stops only when the output next goes low.

The start and stop signals may be delivered to the unit manually or by means of a trigger. Thus, clock operation can be controlled by either depressing pushbuttons or having a standard synthesizer keyboard send electrical triggers which initiate the gating action.

Before examining the circuit, a few words should be said concerning the accuracy and reliability of the unit. Rather than simply taking the pulse output of the CEM3340 VCO chip (which forms the heart of the controller clock), the signal is first sent to a seven stage binary divider, with switch selectable taps. By dividing the signal down, four major benefits are reaped. First, most VCO's have the best accuracy when they are operating in the middle of their oscillation range. Exponential conversion errors, offsets and leakage currents can be a problem when the VCO is operated toward its lower limit; be forcing the oscillator to operate at a higher frequency and then dividing downward, these problems are avoided. Secondly, it has been shown that binary division acts as a "vibrato killer". [1] Thus, any undesired FM deviations and irregularities in the original VCO signal are averaged out before hitting the final clock output. Next, binary dividers have edge sensitive clock inputs. This implies that the output signal will have a precise 50% duty cycle regardless of any fluctuations of the input duty cycle. Finally, by making the divisor of the binary divider switch selectable, an even greater frequency range can be obtained. In summary, then, by using the simple notion of binary division, a very precise output in both frequency stability and duty cycle is obtained over a broad range of frequencies.

Fig. 1 shows the complete schematic for the precision controller clock: refer to it now. IC1 is the CEM3340 VCO chip which forms the heart of the unit. Pin 15 of this chip is a summing node and may accept a variety of control voltages. A 1V/octave control from a keyboard, say, may be applied via jack J1. In the context of period generation (as opposed to frequency generation), an increase of one volt at this input causes a decrease of period by a factor of two. A control voltage also may be applied to the controller clock via jack J2, and in this case, the signal can be atteunated by R38 as desired. Typically, envelope of LFO control signals would be applied to this input and R38 allows these to be "tamed". Finally, R36 and R37 form the fine and course rate controls. These two controls are strung across the entire bipolar supply and this facilitates compensating for offsets in control signals applied to either J1 or J2.

J4 is the start signal input. A logical signal of about five volts or more (in either gate or trigger form) applied to this input will start the controller clock oscillating. Oscillation begins immediately with a first cycle of normal pulse width. C3 and R32 differentiate the input signal while A2, R23 and R13 shape it up into a rectangular pulse about 1 millisecond wide. D6 blocks negative excursions of comparator A2, thus making the pulse polarity compatible with the remaining CMOS circuitry.

S2 is the manual start pushbutton; since its output is applied to an R-S flip-flop (see below), no switch debounce is necessary. The output of S2 is normally held low by R26. Depressing the switch developes a voltage of +15 volts across this resistor, and D7 conducts the result to later circuitry. Note that in this context, D6 and D7 (which apply voltage across R27) form a simple OR gate. This is an instance of "Mickey Mouse" logic, which despite the disparaging name is a valid logic technique. [2]

The start signal (from either J4 or S2), resets both of the R-S flip-flops via pins 4 and 10. In the former case, the $\overline{Q}$ output (Q-bar) at pin 2 goes high and in the latter case, the Q output at pin 13 goes low. When the Q output goes low, IC1 is enabled via R20 and Zener diode D3 (more about this shortly). Simultaneously, pin 2 of IC2 is brought low, thus enabling the seven stage counter. The VCO starts oscillating and the counter starts counting. Notice that the pulse output of the CEM3340 is taken off pin 4. Voltage divider R16 and R30 set the pulse width of the CEM3340 to about 50% and C4 prevents bounce and chatter of the pulse wave at low frequencies. [3] The output of the counter (which is the complement of the desired waveform) is tapped via switch S3 and sent to gate N3 where it is NORed with the Q output of IC3. Recall that this signal is currently in the low state so that N3 inverts the counter output, thus restoring the desired polarity.

The stop circuitry (A1 and associated components) is identical to the start circuitry. Again, two diodes, D4 and D5, OR the signals, and the result is sent to the SET input of the R-S flip-flop at pin 6. A stop signal causes pin 2, the $\overline{Q}$ output, to go low. This condition is echoed to NOR gate N2, whose output controls the second R-S flip-flop. Notice that the output of N2 will go high only when the counter output

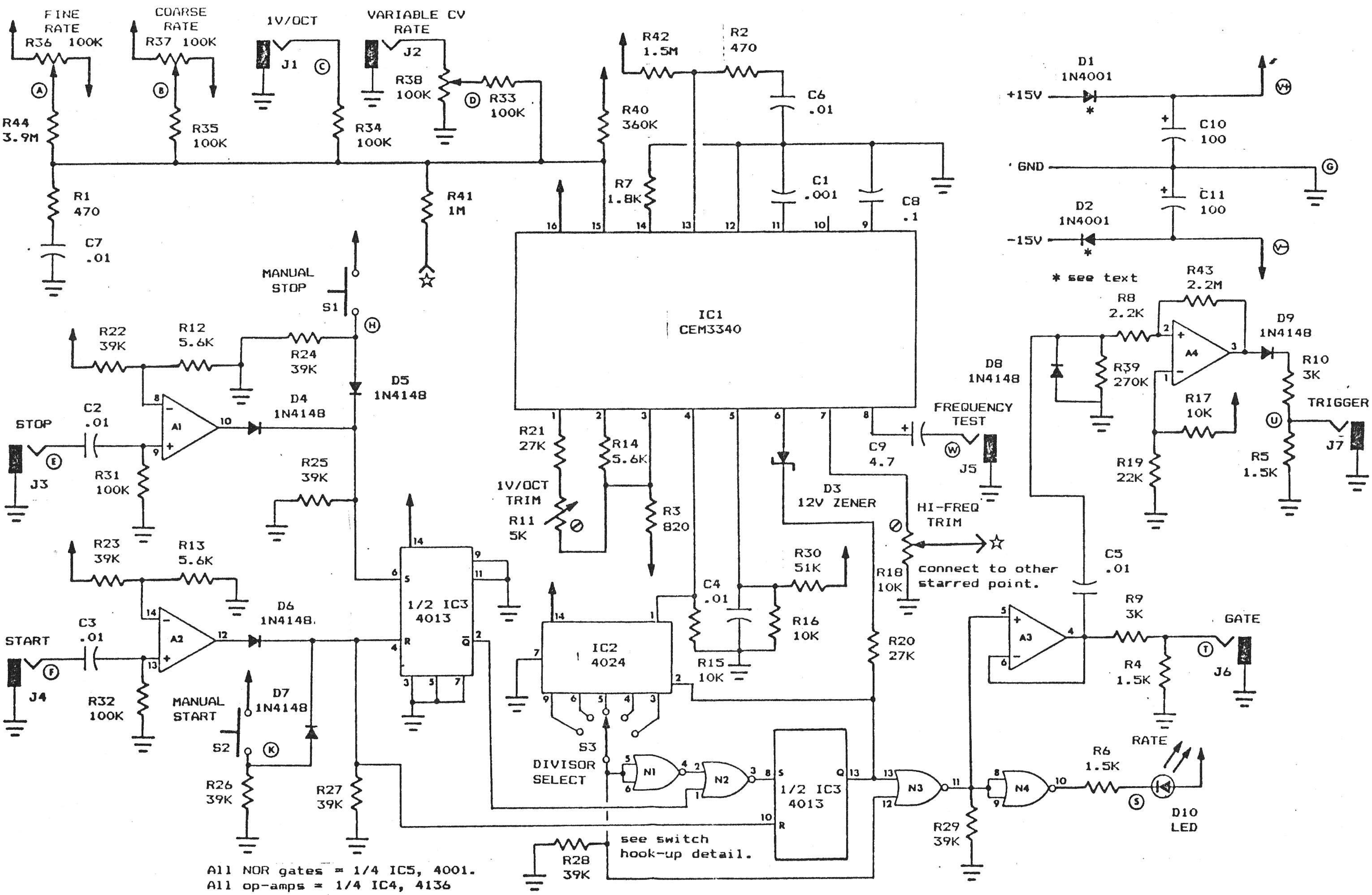


FIG. 1

PRECISION CONTROLLER CLOCK

DESIGNER: Thomas Henry
All resistors in ohms.
All capacitors in mfd.
(circle) keys schematic to printed circuit board.

has gone low. The net effect is that the clock will not be gated off until the output attains its next low state. This is part of the precision gating requirement mentioned above. It is interesting to note that IC3 is really acting like a "memory" here; it acknowledges that a stop signal has been sent, remembering it until the time is right to shut off the oscillator.

When N2 is high, the flip-flop is set thus sending the Q output at pin 13 high. This does two things. First, IC2 is gated off via pin 2. Likewise, IC1 is shut off via R20 and Zener diode D3. This scheme for gating the CEM3340 first appeared in the now defunct (and greatly missed) house journal for Curtis Electromusic Specialties, Synthisource. [4] The reader is referred to this reference for further information on this important topic.

The Q output is NORed with the output of the seven stage counter, IC2. This insures that the final clock output is held in a low state when the oscillator is off and likewise guarantees that the output starts by going high immediately upon detection of a start signal. D10 monitors the output. When the clock is off, so is the LED: when the clock is on, the LED flashes at the selected rate.

A3 buffers the CMOS signal for real world interfacing. The output is chopped down to standard 0V to +5 volt size by voltage divider R9 and R4, which also provides a standard 1K output impedance. This forms the gate output, which will have a 50% duty cycle.

C5 couples the signal to A4 and associated circuitry which creates a nominal 1 millisecond output trigger. D8 dumps the negative swing of differentiator C5, while R39 provides the damping. A4 forms a Schmidt trigger, which gives a sharp output signal. D9 restricts the polarity of the output, while voltage divider resistors R10 and R5 bring the signal down to a standard 0V to +5V swing.

J5 provides a "frequency test" output which is actually nothing more that a 10V pp ramp wave. This output, which runs at the VCO's normal frequency, is provided for several purposes. First, it's a convenient output to monitor when tweaking the scale trimmer for a precise 1V/octave response. Secondly, when using the excellent method of frequency-to-period conversion for playing complex rhythmic patterns by sequencer, the "frequency test" output may be used to set the desired interval. [5] This greatly speeds up the adjustment of sequencers and makes the technique that much more automated. Lastly, the controller clock may be used just like any VCO, and in this context, the ramp output with its even order harmonics is a pleasant sound source.

Finally, trimmer R11 can be used to adjust the scale for a precise 1V/octave response. Trimmer R18 allows for compensating the high frequency response droop (caused by exponential conversion errors and to a lesser extent, to reset lag time).

Building the controller clock is straightforward, but some attention must be paid to circuit layout. This is especially true around pins 15, 13 and 11 of the CEM3340 (which are the exponential input, the reference current and the timing capacitor pins, respectively). Neatness really counts here! The best way to go is with a printed circuit board which eliminates many of the problems of "rat's nest" wiring. To simplify the task of generating a circuit board for this project, Fig. 2 shows the lifesize artwork for a tested design. This copyrighted circuit may be fabricated by individuals for their own use, but commercial users should contact the author at the above address for licensing details.

Fig. 3 shows a parts placement guide for the circuit baord. When loading the board, be sure to observe the polarity of all diodes and electrolytic capacitors, and be certain to orient the IC's correctly. A number of jumpers are needed and these are denoted on the parts placement guide by the letter "J". You may use resistor clippings or bare bus wire for these. Notice that the circled letters in the schematic of Fig. 1 call out the various input and output pads to the circuit board. Thus the schematic may be used as the main wiring guide for the circuit. Fig. 4 shows a detail of switch S3 and makes it clear how to proprely wire up this

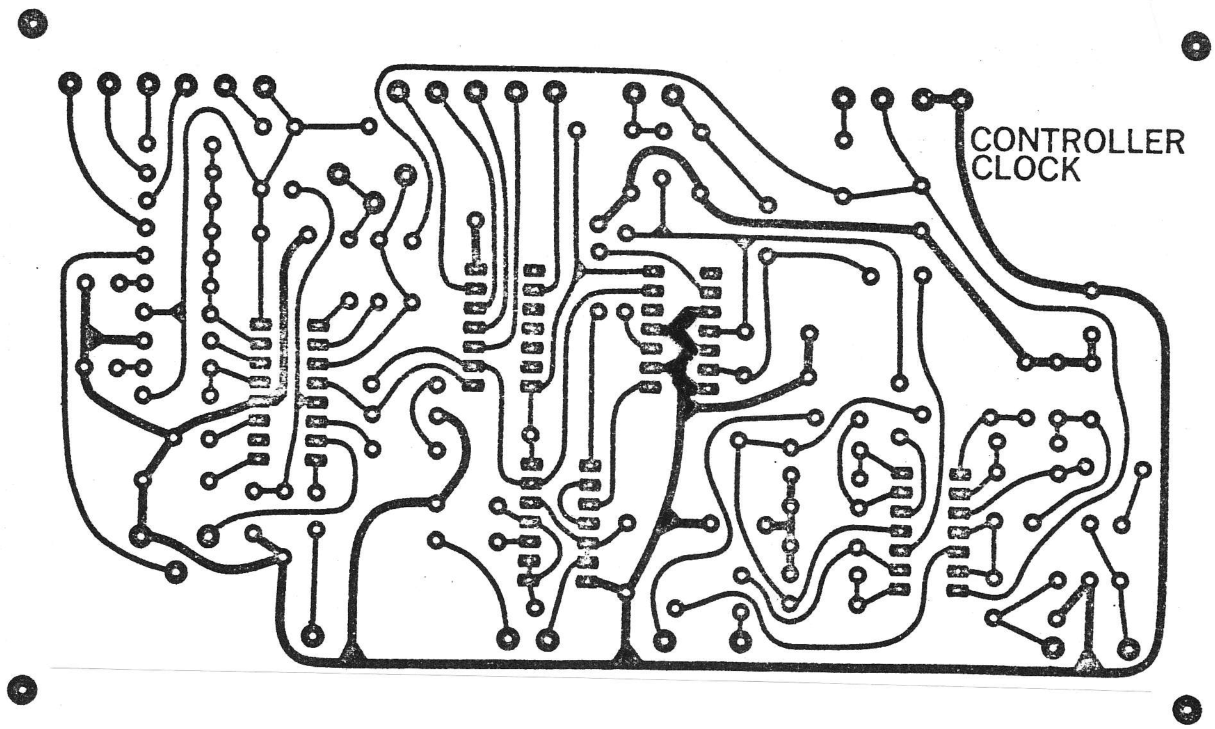


Fig. 2 Printed Circuit Board Layout

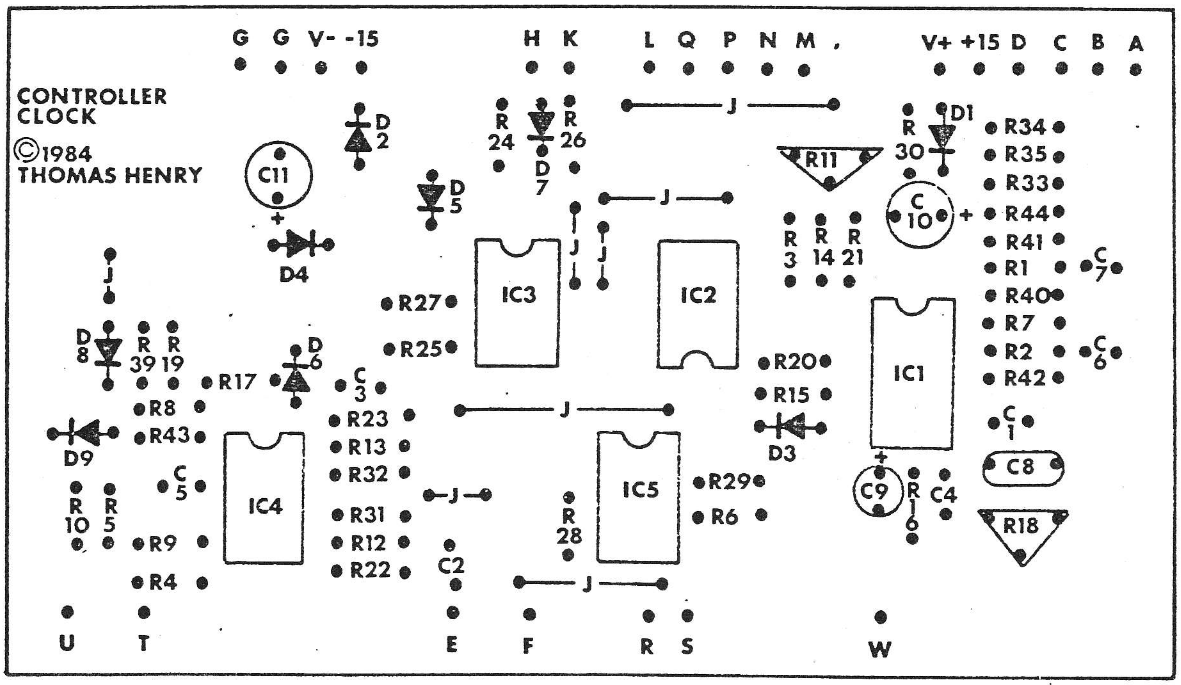


Fig. 3 Part Placement Guide

five position rotary switch. Power is supplied to the board at the points labeled +15, G, and -15. Points V+ and V- may be used to run the supply limits to the front panel controls (like the tuning pots, the pushbuttons and the LED). The pads labeled G, of course, are the ground connections.

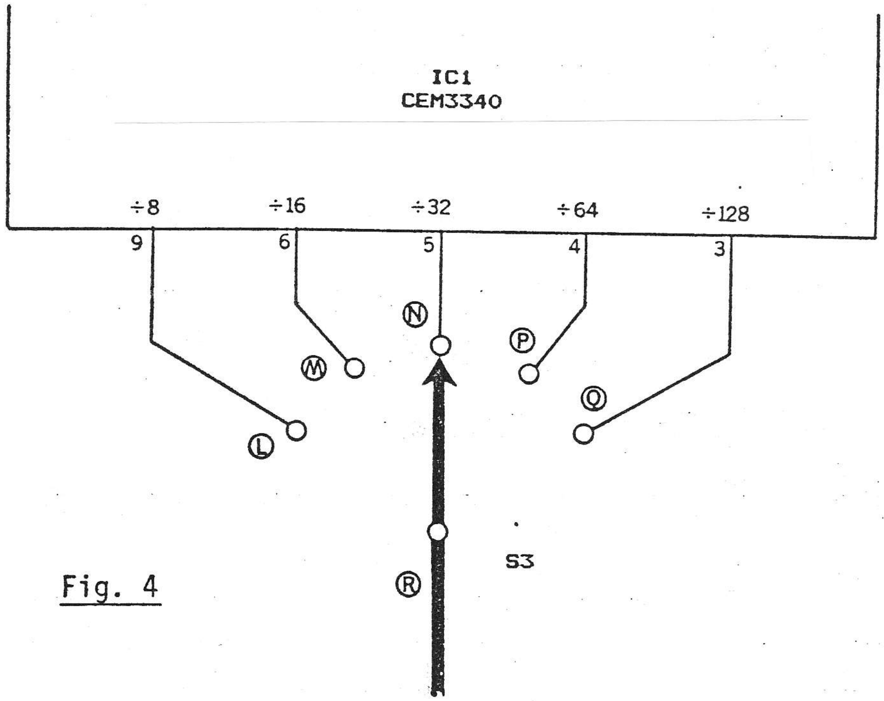


Fig. 4

Detail for switch hook-up.

(circle) keys schematic to printed circuit board.

Rectifiers D1 and D2 and capacitors C10 and C11 provide a measure of decoupling for the controller clock from other circuits running off the same power supply. However, in the original prototype it was found that D1 and D2 raised the impedance of the power supply, as seen by the circuit, sufficiently to inject some undesired FM effects. It was found that the LED flashing caused enough of a differential current flow to actually impose a vibrato on the VCO (probably through the tuning controls or R40). This vibrato effect was very slight, on the order of 10 cents or less at the ramp wave output, and had no perceptible effect on the gate or trigger outputs (for reasons mentioned earlier concerning the benefits of binary division). Nonetheless, its effect on the ramp wave was annoying enough to suggest replacing diodes D1 and D2 with jumper wires. This, in fact, cured the problem in the prototype, so if frequency deviation problems occur, try replacing the rectifiers with jumpers. As mentioned above, D1 and D2 were originally added to aid in the decoupling of this circuit (they act like low valued resistors and form an RC network with C10 and C11). In retrospect however, it should have been clear that the disadvantages of raising the effective output impedance of the power supply far outweigh the benefits of improved decoupling. In general, VCO's need tightly regulated power supplies and nothing should come between the supply and the VCO!

Fig. 5 gives the complete parts list for the circuit. After finishing construction, using Figures 1 and 5 as a guide, apply power to the unit. Confirm that the start and stop functions work correctly and then test the various outputs for proper waveshape and amplitudes. If everything checks out, the controller clock may be tweaked. First, turn trimmer R18 to the ground position, thus injecting no correction voltage into the circuit. Now, while monitoring the ramp wave output, adjust R11 for a 1V/octave response in the range of 500 Hz. A standard synthesizer keyboard may be used to supply the control voltages for this step. Next adjust the high frequency trimmer, R18, for the same response in the region of about 5 KHz. This completes the alignment of the controller clock but after a burn-in time of several hours, the trimmers should be realigned for maximum accuracy.

As mentioned at the start of this article, the uses for a controller clock are

Fig. 5 CONTROLLER CLOCK: PARTS LIST

RESISTORS

R1, R2	470 ohms
R3	820 ohms
R4-R6	1.5K
R7	1.8K
R8	2.2K
R9 - R10	3K
R11	5K trimmer
R12 - R14	5.6K
R15 - R17	10K
R18	10K trimmer
R19	22K
R20, R21	27K
R22 - R29	39K
R30	51K
R31 - R35	100K
R36 - R38	100K linear pot
R39	270K
R40	360K
R41	1M
R42	1.5M
R43	2.2M
R44	3.9M

CAPACITORS

C1	.001 mfd poly
C2 - C7	.01 mfd mylar
C8	.1 disk
C9	4.7 mfd electrolytic
C10 - C11	100 mfd electrolytic

SEMICONDUCTORS

D1, D2	1N4001 (*)
D3	12V Zener
D4 - D9	1N4148
D10	LED
IC1	CEM3340 VCO chip
IC2	4024 ripple counter
IC3	4013 dual flip-flop
IC4	4136 quad op-amp
IC5	4001 quad NOR gate

(*) see text

MISCELLANEOUS

J1 - J7	1/4" phone jacks
S1, S2	SPDT pushbutton switch
S3	SP5T rotary swithc

LED holder, wire, solder, knobs, front panel, hardware, etc.

myriad. But its value is really apparent when the clock is used to fire sequencers. For example, a two bank sequencer may devote the second bank for setting the duration of each of the notes, using the method described elsewhere by Duesenberry. [5] The period-to-frequency conversion method is applicable here since the controller clock has a standard 1V/octave response. And, or course, the gating feature will be handy for percussion and sync effects in the recording studio. All and all, while the controller clock may not be the most glamorous of synthesizer modules, there is no doubt that it makes possible a number of effects which would be difficult or even impossible to accomplish any other way.

REFERENCES:

[1] Bernie Hutchins, "The Digital Divider as a 'Vibrato Killer' " , Electronotes, Vol. 12, No. 119, Nov. 1980, pg 2

[2] Don Lancaster, CMOS Cookbook, (indianapolis: Howard W. Sams and Co. 1977) pp 186-189

[3] Doug Curtis, "Comments on 'Two Hints on Using the CEM3340 VCO IC' ", Electronotes, Vol. 13, No. 122, Feb 1981, pg 2

[4] Anonymous (suggested by an idea of John McColm), "Reader's Circuit Corner", Synthisource, Vol. 1, No. 2, Spring/Summer 1981, pp 13-15

[5] John Duesenberry, "Rhythmic Control of Analog Sequencers", Polyphony, Sept/Oct. 1978, pp 26-29

Build This!

COMPLEMENTARY OUTPUT LFO

BY THOMAS HENRY

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

Specifications

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

How It Works

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

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

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

Construction

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

Power Supply

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

Calibration

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

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

Going Further

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

Conclusion

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

Notes

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

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

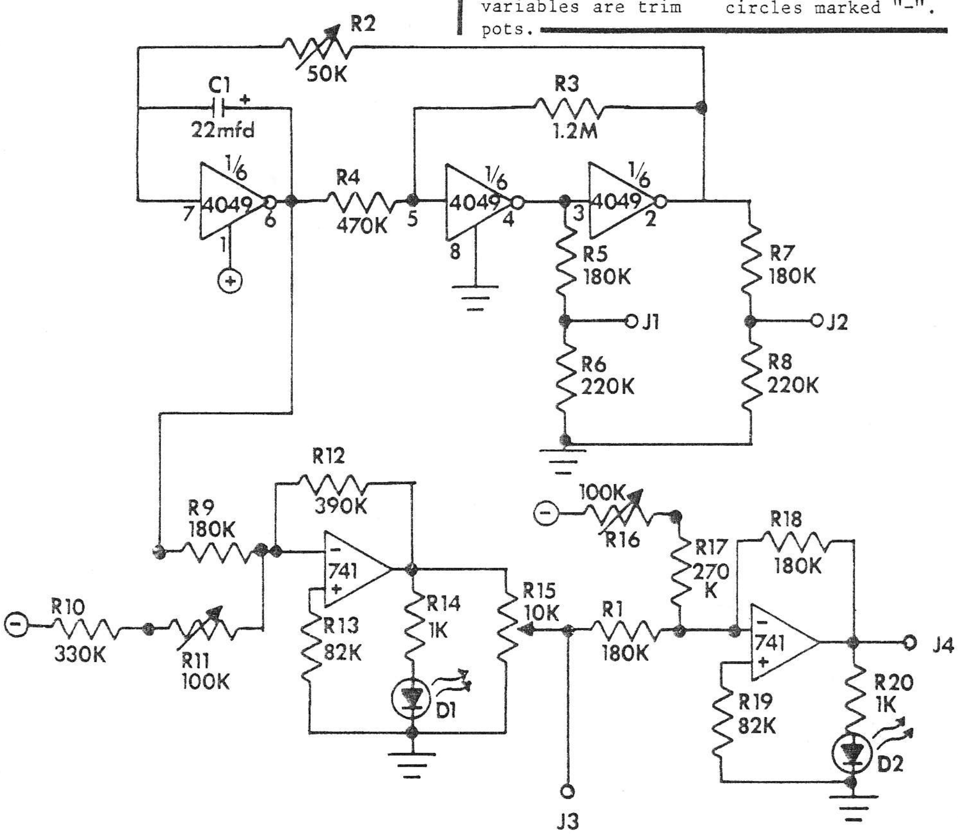
3) Craig Anderton made use of this basic oscillator block in a tremelo circuit in Contemporary Keyboard, August 1979, page 69, "Electronic Projects, Tremelo Part 1".

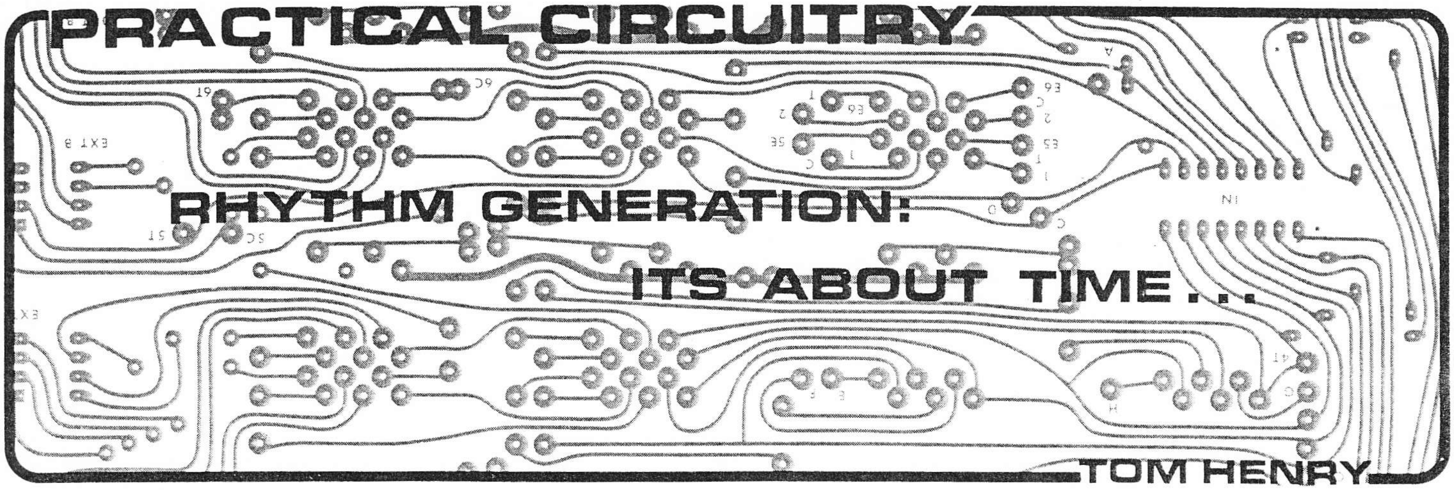
schematic

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

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

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





PRACTICAL CIRCUITRY

RHYTHM GENERATION: ITS ABOUT TIME...

TOM HENRY

Hello and welcome! In "Practical Circuitry" we will discuss many things, but in one way or another they will all relate back to synthesizer design and construction. We'll start off with a detailed look at rhythm generation. One of my reasons for coming up with the unit was that I didn't like to play alone! By integrating the unit into my synthesizer system, I'll always have a nice pattern to play against. I think that you'll find this can be a great aid to composition as well. Often, just hearing a beat starts the creative juices flowing.

I think that every one has, at one time or another, played with automatic rhythm boxes - you know, those neat toys that when you push a button labelled "Rock 1" kicks out a repetitive "book-chik-a-boom" over and over. Such boxes are great to practice with in lieu of a metronome, but for performing are next to worthless due to the constant repetitive nature of the sound. Well, guess what? We're going to build one anyway! But by applying synthesizer techniques and generalizing the unit, we'll get a broad range of sound and rhythms that would be unheard of with a simple rhythm box...we'll even get into tape synching and other neat topics.

THE CIRCUIT. The basic rhythm generator is configured around a special ROM, the AMI S-2566 (see fig. 1). This chip contains eight standard drum patterns, and was originally designed to be used in drum boxes. As we'll see shortly, there are ways to make it do much more. This chip must be used in conjunction with a clock circuit, and the clock's main job is simply

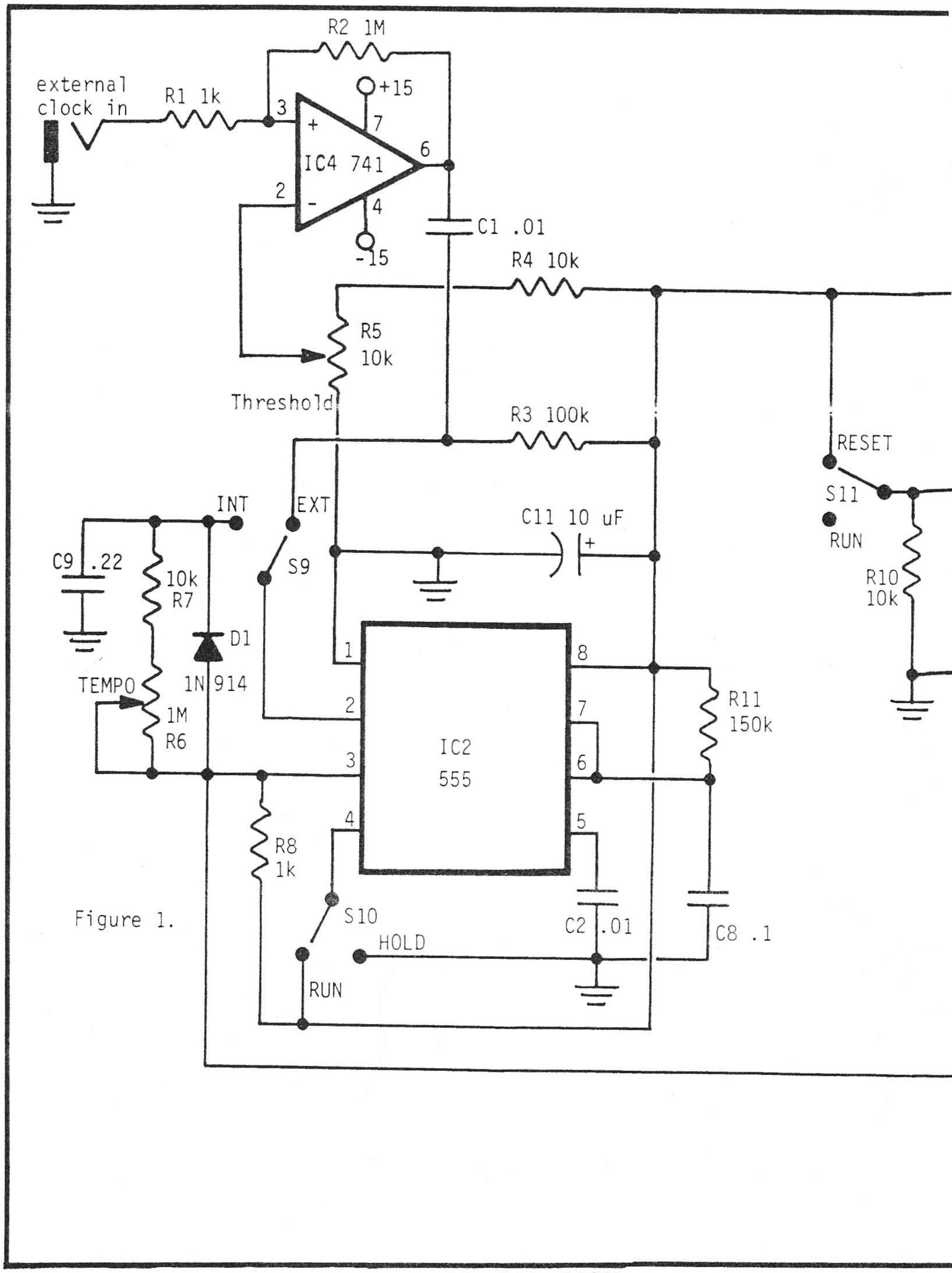


Figure 1.

to walk out the various patterns.

Switches S1 through S8 address the various rhythms, and in most drum boxes, a single pole/eight throw switch is used for selection. However, by using individual switches, one may combine patterns by throwing any number of switches. The results are quite interesting, and the new patterns quite complex (it is rare when I use one of the "standard" patterns for composition). If you are into combinatorial analysis, the total possibilities for switch combinations is now 2 to the 8th power, or 256. I haven't really tried all the possibilities, but I suspect that the real number is somewhat less than this, since some combinations may be duplicates of others. And then of course "no switches on" is mathematically possible, but musically not very useful, since no sound is generated.

There are five outputs, which the manufacturer labels bass drum, snare, brush, congo, and bongo. Now on my unit, I labelled the outputs 1, 2, 3, 4, and 5 and the pattern selectors A through H since I didn't want to be constrained to think about the unit in its normal sense. More about this later.

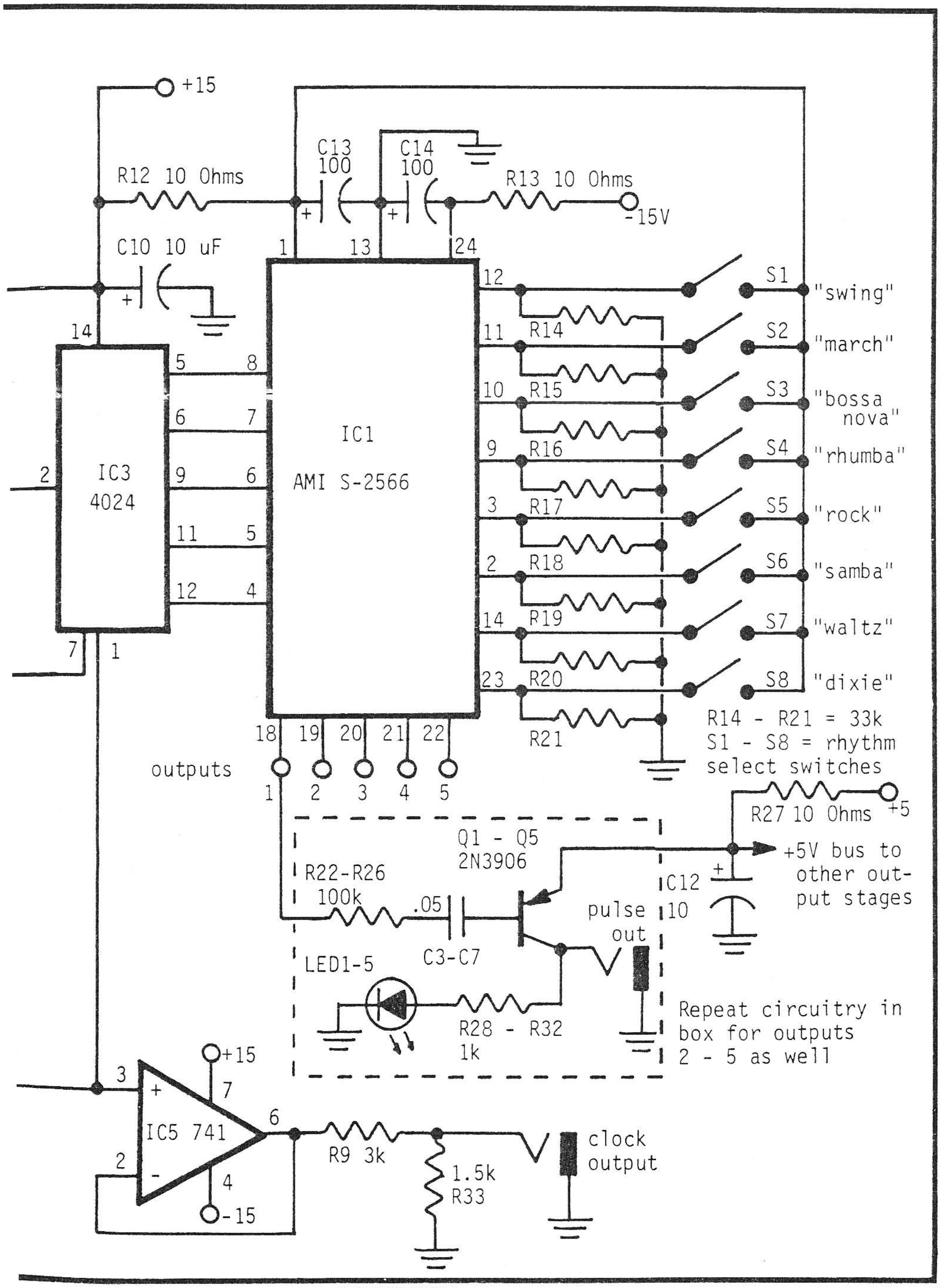
Let's look at a typical output, as shown on the schematic. The outputs give a simple "voltage on" or "voltage off", and must therefore be differentiated by R22 and C3 to give a synthesizer compatible trigger pulse. Q1 performs the level conversion necessary to generate a 5V trigger. LED 1 gives a visual indication of what's going on. The five LEDs are most important, since not every rhythm uses all of the outputs, and with the LED you can see which output to use for what purpose (as well as get a multi-media feel for the patterns). The outputs terminate in standard 1/4" phone jacks.

Now let's look at the clock circuitry. The basic clock is a 555 timer, but note the unusual configuration. With S9 in the INTERNAL mode, the 555 acts as an astable oscillator. With S9 in the EXTERNAL mode, it becmes a monostable (one shot) and must be fired from an external clock. I'll say more about that in a moment; for now, let's leave S9 on INTERNAL. C9, and R7 are the basic frequency determining components, and establish the tempo of the selected pattern. R11 and C8 determine the pulse width of the clock output pulse, and remain fixed. This is the reason for the unusual configuration of the 555; regardless of whether one is using the INTERNAL or EXTERNAL mode, the 4024 will always receive the same width clock pulse - even in the EXTERNAL mode, R11 and C8 still determine the pulse width. This standardization of clock pulses is important for multi-track recording, when we want to pick up clock pulses off of a click track.

S10 disables the 555. This switch is important, since it allows you to start a pattern right on the beat.

The external clock input accepts a possibly nonstandard clock pulse and conditions it via the 741 and differentiator R1, R2, and C1. R2 adds a bit of hysteresis, giving the input clock pulse a little more snap. This sub-circuit is really nothing more than a simple comparator, and R5 sets the threshold at which the comparator trips. The threshold control is useful when recovering pulses from a click track (you generally want to record the click track at a low level to avoid bleed-through on to adjacent tracks, and use a sensitive comparator for recovery).

Another 741 buffers the clock output, and presents it to a 1/4" phone jack, through R9 and R33. This resistor network does two things; first, it divides the output voltage to a standard +5V



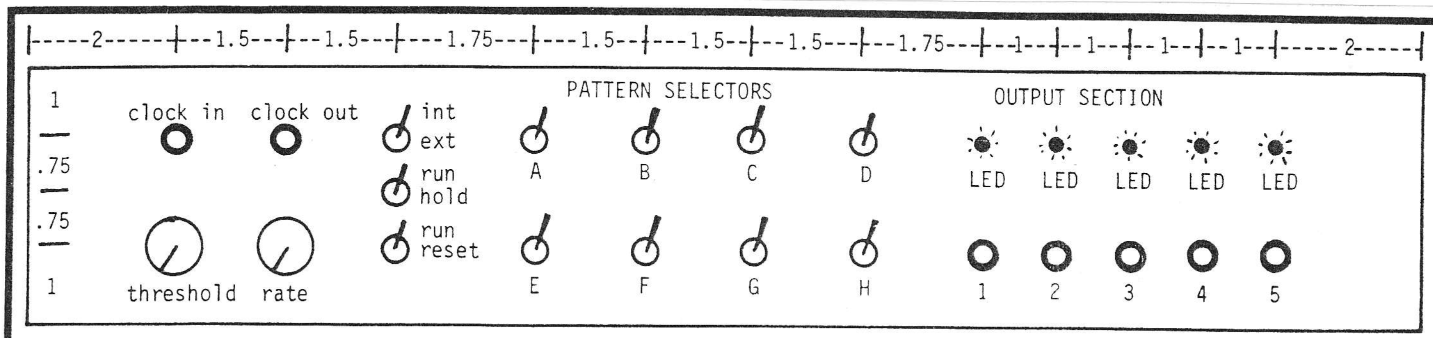


Figure 2. Rack panel: 19" X 3.5". All dimensions in inches.

range, and second, adjusts the output impedance to a standard 1k. This is the output one would use for synching other rhythm units to one clock, driving a sequencer, and the like. In addition, one would use this output to record sync pulses.

The clock also feeds pin 1 of the 4024, whose outputs drive the S-2566. S11 allows for resetting the 4024, and the musical effect of this is to cause the pattern to return to the start of the measure. When it comes time to do some synching effects with the unit, you will appreciate this control.

A few words about the power supply...the 741s and S-2566 require ±15V DC, the 555 and 4024 need +15VDC, and transistors Q1 through Q5 require a +5V supply (this can be dropped from the positive bipolar supply with a 5V regulator).

CONSTRUCTION. I used a standard 3.5" X 19" rack panel for packaging, with the layout shown in figure 2. All jacks are standard 1/4" phone jacks; I used clip-lites for mounting the LEDs, and the results were nice looking.

FINDING THE AMI S-2566. The AMI S-2566 is currently available on the surplus market from Diamondback Electronics (PO Box 12095, Sarasota, FL 33578) for under $2 with data (quite a contrast to what these parts cost when they first came out). While you're at it, you might want to order a couple, since I have plans to do some series/parallel applications in the future.

HOW TO USE IT. Before you can really play with the unit you must have some drum voices that accept 5V triggers. In my system, I use two of the Percussive Noise Sources described by John Blacet in Polyphony Nov/Dec '79, pages 12 and 13 (corrections in Jan/Feb 80 issue, page 5), and two typical twin-tee sinusoidal "bongo" generators set for bass drum and tom-tom frequencies. Circuits for this type of circuit abound; see Polyphony 3/76, pp 37-42. I use my synthesizer proper for the fifth voice. Typically, this voice employs a noise source, VCA, and VCF.

I did make a small change in both Percussive Noise Sources to make them more useful with my system. The resistor from pin 24 to ground, given in the original schematic as 150k, sets the sustain time of the audio output. Internally this means that the resistor sets the ON time of the one-shot generator, and so has a direct bearing on the sustain time of the envelope imposed on the audio signal. This initial ON time was far too long, in fact, a series of triggers coming to the Noise Source simply held it on. So, I changed the resistor from 150k to 47k...this allows me to now generate such interesting sounds as a dog scratching for fleas, rhythmically! And, all kidding aside, other short-noise-burst type sounds.

By the way, I have found that using all white noise type drum sounds is very distracting and using all bongo type circuits is very boring. It is a combination of the two types that really produce a pleasing sound.

Well, let's suppose you've hooked up some drum voices (see figure 3). If all has gone according to plan, flip S11 to RUN, S10 to RUN, S9 to INTERNAL, select some pattern via S1 - S8, and you should hear a madcap electronic drummer wailing away! R6 adjusts the tempo of the pattern.

MULTI-TRACKING EXPERIMENTS. As much fun as real time percussion is, it is in the area of multi-tracking that this unit shines. For a basic multi-tracking experiment, record the clock out on to track 1, and the mixed drum output to track 3 (see figure 4). Record a minute or so of material; now rewind the tape to a point a few feet before the start of the recording. Switch the patch cords around, so that the output of track 1 goes directly from the tape machine to the external clock input, then, remove the patch cord from track 3 and patch into the input of track

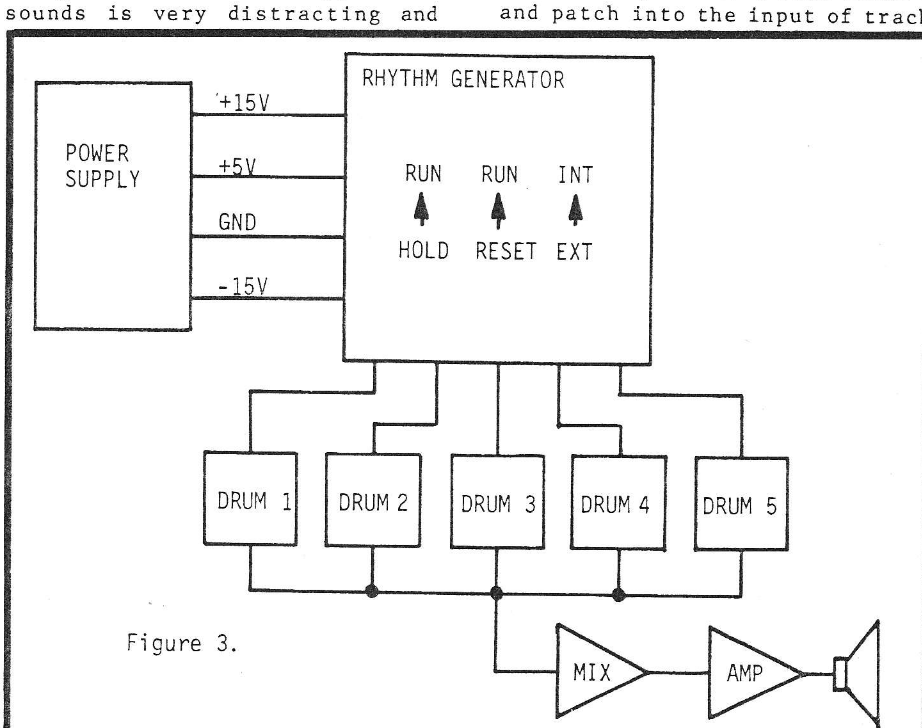


Figure 3.

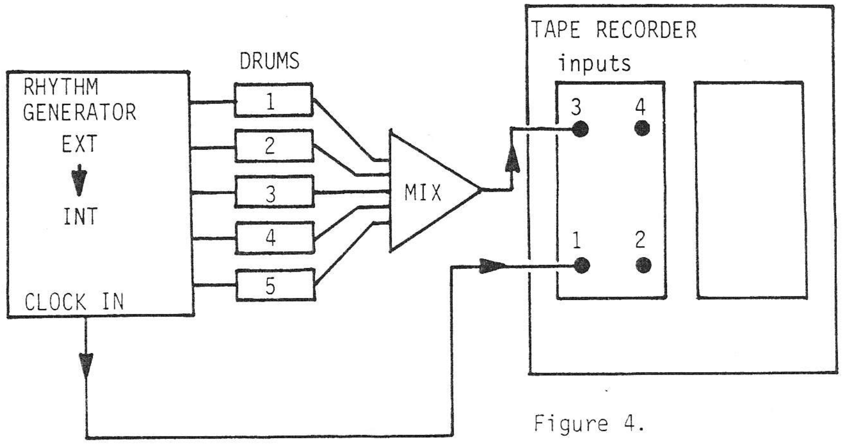


Figure 4.

4. This time we are going to record on track 4, so be sure to leave track 3 alone (see figure 5). Flip S9 to EXTERNAL mode, put S10 in the HOLD mode, and toggle S11 from RUN to RESET back to RUN again.

Start the tape machine rolling, then flip S10 to RUN. The reason for starting the tape machine first is that engaging the tape machine in the record mode may enter a mechanical or electrical click through the system, thus causing the 555 to trigger once and offsetting the tempo from the original. Now, if all has gone well, the rhythm unit will pick up the clock pulses from track 1 and convert them into the drum pattern being recorded on track 4. Rewind the tape and play back tracks 3 and 4; they should sound the same. While this isn't too musically useful, it does demonstrate that one set of synch pulses has perfectly synchronized the two drum parts. The fun starts when you record two different patterns synched to the same pulses, on to different tracks. The resulting sounds are wonderfully complex.

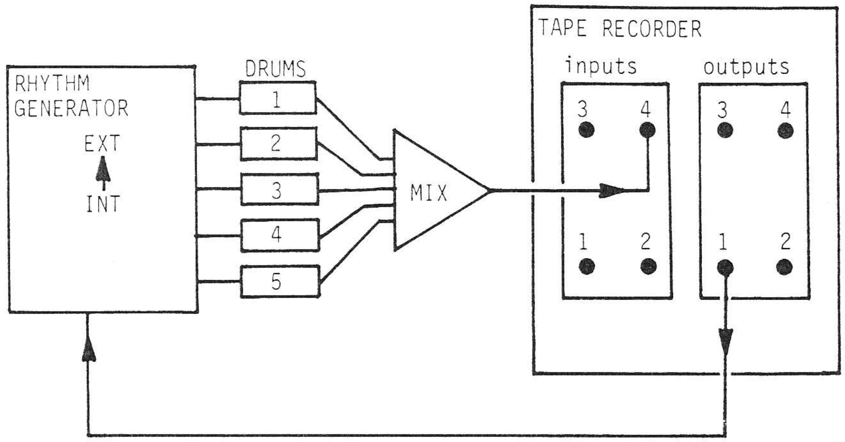


Figure 5.

I should probably say a few things about the use of sel-sync, simul-sync, or whatever your tape machine calls this function. When recording on track 4, monitor tracks 1 and 3 through the record heads, using the above mentioned sync function. Failure to do this would mean that during the recording process tracks 3 and 4 would sound synched, but during playback, the two tracks would be horribly mixed up due to the distance between the record and playback heads. If you're not sure what I'm talking about, refer to Craig Anderton's book, Home Recording for Musicians, pp 24-25 for more information.

RANDOM COMMENTS. You'll note that we didn't use track 2 in the previous example; this serves as a guard track between the click track and the other audio tracks. These clicks represent a fair amount of energy, and you don't want to have any leakage getting over to the other tracks. Of course, after recording all the drums you don't need the clicks anymore, at which point they can be erased, leaving room for additional overdubs.

Here's one idea I came up with on a sleepless night. Recall that the S-2566 has five outputs. Try to visualize six of these chips wired so that one chip was the master chip, and each output would drive the clock inputs of the remaining five chips (see figure 6). The master clock would have to run much faster now, since there would be quite a few divisions taking place. Anyway, the result would be 25 drum outputs (five from each chip), resulting in a very complex pattern. Practical? I don't know. Like I said, this came to be during a bout of insomnia...

Here's another wild and crazy idea. Notice how the 4024's divide by 2, 4, 8, 16, and 32 pins are used for clocking the S-2566;

continued on page 32

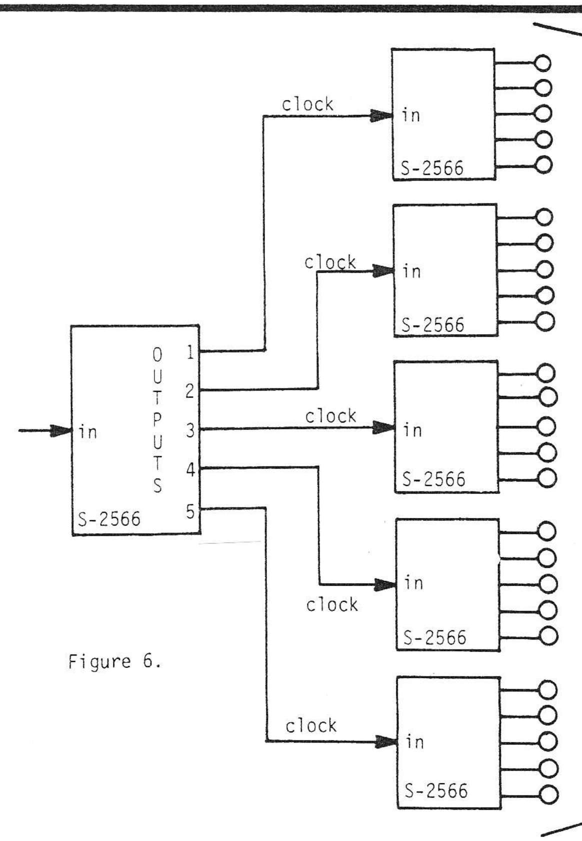


Figure 6.

by Andy Bassford

(This is the start of what we hope will become a semi-regular feature. We encourage readers from other countries to contribute similar articles that describe the musical electronics scene in their country. Tell us about do-it-yourself activity, how well musical electronics is accepted, and the like...just as Andy has done below.)

I moved from Hartford, Connecticut to Kingston, Jamaica about eight months ago. I am a guitarist with WE THE PEOPLE band, one of the island's most popular groups, and do many sessions as well.

The electronic music scene in Jamaica, like everything else in Jamaica, is dramatically affected by the problem of foreign exchange. Without going into a dissertation on international finance, the Jamaican dollar is largely valueless in the outside world, and this, coupled with the fact that there are extensive government restrictions on buying foreign currency and importing in general, means that a 741 op amp costing maybe a dollar at Radio Shack sells for $10.50 Jamaican, if you can find one to buy. A Fender Vibro Champ amp sells for $750 Jamaican - you get the picture.

Because of this problem, there are no do-it-yourselfers among the Jamaican musicians I know. However, they have a tremendous interest in effects, and guitarists especially own at least one or two commercially produced devices (usually bought while on tour out of the country). If you're a musician in Kingston you play reggae, especially if you do sessions. And, as anyone who has heard a "dub" album (remixed rhythm tracks with psychedelic effects) can tell you, echo and reverb effects are vital components of reggae. Phase shifters and Mutron-type devices are common on sessions, but the real electronic star in the Jamaican firmament is the Syndrum sound. It is common to have two or three tracks of Syndrum dubbed on to a record in addition to regular drums and percussion. Keyboard synthesizers are also popular, and Robbie Shakespeare (one of reggae's top bass players - Ed.) has a Gizmo. I am the only musician that I know of with a pedalboard. Session fees are $50 a song, and reggae musicians work very quickly. One day I was on a session where we laid down 14 rhythm tracks in six and a half hours!

Unlike the states, live performances of music in Jamaica are relatively rare, due to the difficulty of obtaining instruments. A few clubs have resident bands, and the large hotels have lounge-style groups, but most dances and parties are serviced by portable discos rather than bands. Stage shows, when they do occur, last for anywhere from three to five hours <u>straight</u> as a variety of singers, vocal groups, and DJ-rappers come on one after another, do their four or five songs, and leave. After my experiences in the backup band at these shows, I'll never complain about four sets a night again! Effects use in these situations is generally confined to the PA system, although our keyboard player often plays Syndrum parts on his synthesizer, and the musicians simulate echo effects manually on their instruments in order to approximate the "dub" style.

In conclusion, I would say that despite the many obstacles facing them, Jamaican musicians have developed an original and sophisticated style of using effects to enhance their music which other musicians can learn from - a style in which effects are used more as an organic part of the music rather than as a gimmick, and are integrated into the rhythm track rather than being piled on top of it. Anyone seriously interested in electronic music owes it to him or herself to check out reggae. Σ

RHYTHM GENERATION: IT'S ABOUT TIME...

continued from page 25

how about scrambling these in some other order? Or what about using the divide by 64 or 128 pins in some combination with the others? Crazy!

As you can imagine, there are infinite variations on the basic rhythm generator circuit. I would be interested to hear your ideas on the subject, no matter how strange or wild they may be; I'm sure that what we've done here is to just scratch the surface. Σ

E-H MINI SYNTH REVIEW

continued from page 13

there's no VCA and that the filter pops when you remove your finger from the key, this is not something that most synthesists would use on stage; however, the M/S does make some truly grandiose bass sounds. If I played bass regularly and could only spend $200 to improve my setup, the M/S would be my first choice. The keyboard is easy to play, and you can get a wealth of great alternative bass sounds. Drop the octave, turn on the "phaser", and wail - the results are most impressive. I've even used the M/S for some recorded bass parts because frankly, no other synthesizer gets quite the same sound. If you want to get trumpets and strings, you had better move on to something more costly. But for bass lines and an occasional screaming solo, the M/S is pretty impressive when heard through a decent sound system.

However, evaluating the M/S solely from a technical standpoint overlooks some of its most attractive features - namely, the social implications of playing something like the M/S (see "Editor's Note"). Non-synthesists seem far more fascinated by the M/S than the M-10 simply because...well, it's a synthesizer, and the M-10 is still basically an organ (of course, if you add the M-10 mods we've covered in Polyphony, then you've got one heck of an organ). But, in the world of portable music both units fulfill a unique place. Give the rhythm player an M-10, and the lead player an M/S; that should make for a pretty good jam! Σ

PRACTICAL CIRCUITRY: TRICKS WITH THE SN76477

THOMAS HENRY

I think that anyone who has ever played with the SN76477, Texas Instruments' complex sound generator, has gone through two phases. The first phase is "Wow, does this thing ever do a lot!" And it does. The chip certainly represents a triumph of large scale integration, incorporating many different sound functions in one 28 pin package.

Well, that euphoria probably gave way to dejection. Yes, in this second phase you probably discovered that the VCO didn't have a very large sweep range, and was temperature dependent. And the noise source was rather static and didn't really have a great variation of possible sounds. Most importantly, you probably found out that despite containing all these neat functions, the chip was basically organized in a fixed manner. That is, it contains lots of sub-circuits, but they are all interconnected, internally, in a certain fashion, and rearrangement of them seems impossible. You probably drew the same conclusion I did: The chip was suitable for pinball machines and kiddie games, but wasn't really adapted for serious synthesizer work.

Now it's time for phase three, wherein we discover how to separate the functions, get a broader range of sounds from the noise source, and do all sorts of other neat things. Most importantly, our applications will be suitable for much more than kiddie toys...like serious synthesis. In this installment we will discuss how to isolate output structures; next time we'll examine the noise source in detail, and then we'll close out this particular series by integrating all these ideas into a "Super Controller Module" suitable for use with the best synthesizers. I'm really fired up on this, and I think you'll be pleased as well!

I'm going to assume in this article that you're already familiar with the basic layout of this chip (if not, refer to Craig Anderton's article on the SN76477 in Synapse magazine's Summer 79 issue, or check out the data sheet Radio Shack provides with their SN76477s). This will save referring to the spec sheet throughout the article.

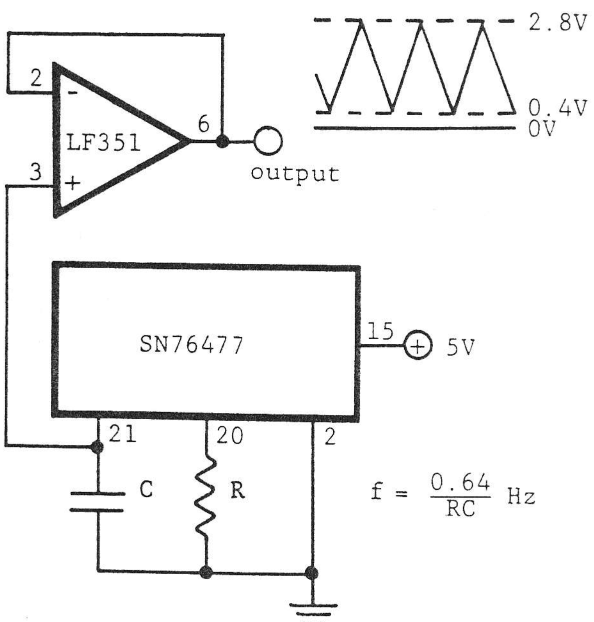
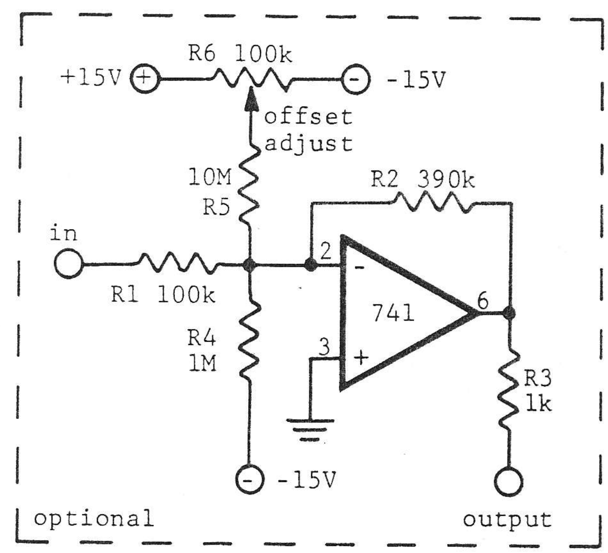


Fig. 1 Power supply connections to op amps assumed (±15V)



Tapping the LFO Triangle Wave. Ordinarily, the only LFO waveform accessible at the chip output is a square wave; however, figure 1 shows how we can obtain a triangle wave from the LFO. We simply tap the waveform off of the timing capacitor, making sure that we buffer it sufficiently to prevent loading down the cap. The LF351 is an ideal buffer, since in the configuration of a voltage follower (unity gain buffer), it has an input impedance of thousands of megohms! Other bi-fet family op amps (LF356, TL071, etc.) are equally suitable.

Note that the output goes from about 0.4V to about 2.8V, for a total swing of 2.4V (these figures will vary somewhat from chip to chip). The optional circuit shown beefs up the signal to a standard 10V peak-to-peak, referenced to ground, with an output impedance of 1k. R4 sets the approximate ground reference, while trimmer R6 allows for precise adjustment. This trimmer may not be necessary in less precise applications.

Since we have not interfered in any way with the internal mixer or output stages, a square wave version of the same frequency is available at the output proper of the chip (pin 13). So we get two waveforms for the price of one!

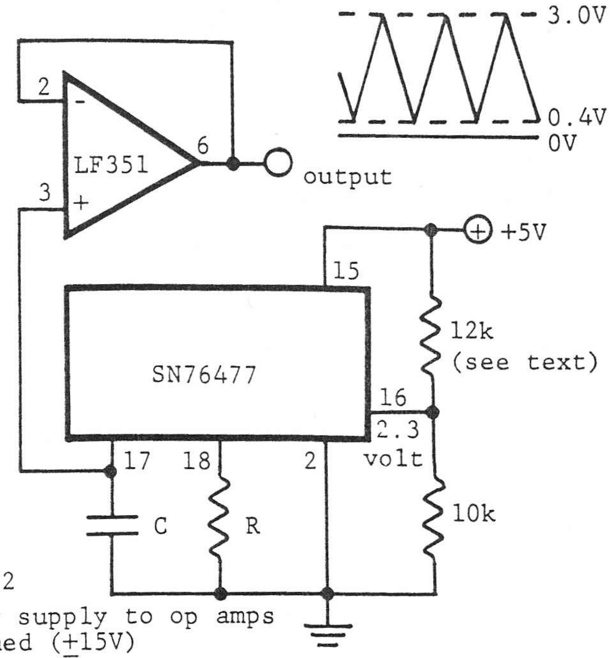


Fig. 2
Power supply to op amps assumed (±15V)

Tapping the VCO Triangle Wave. Figure 2a shows how to pull this same trick on the VCO. This time the buffered output goes from about 0.4V to about 3.0V, which is slightly greater in amplitude than the LFO output. The signal may be amplified and level shifted with the same optional circuit from figure 1, for approximately the same results. You may wish to slightly reduce feedback resistor R2 to compensate for this small difference in amplitude.

One problem with the VCO circuit is that the amplitude decreases as the control voltage to pin 16 decreases, meaning that amplitude decreases for increasing frequency. The internal square wave is unaffected. This being the case, if you plan on using the VCO triangle output it is probably best to apply a fixed control voltage to pin 16 and leave it at that value, thus making the VCO into a manually controlled oscillator. Control the frequency by using a pot at pin 18. Figure 2b shows a voltage divider applying a fixed voltage of 2.3V to pin 16. Experimentation has shown this to be the best value.

Tapping the Noise Source. Why stop now? Figure 3 shows how to tap the noise output. The buffered output goes from 0 to about 5V (with the noise filter wide open), which may be suitable for many applications. However, to make the noise source compatible with standard gear, the optional circuit shown is recommended. With the added stage, the signal is now ground referenced with an amplitude of 10V p-p. Note that the output impedance is a standard 1k. The 600k resistor specified is ideal case: 560k works almost as well.

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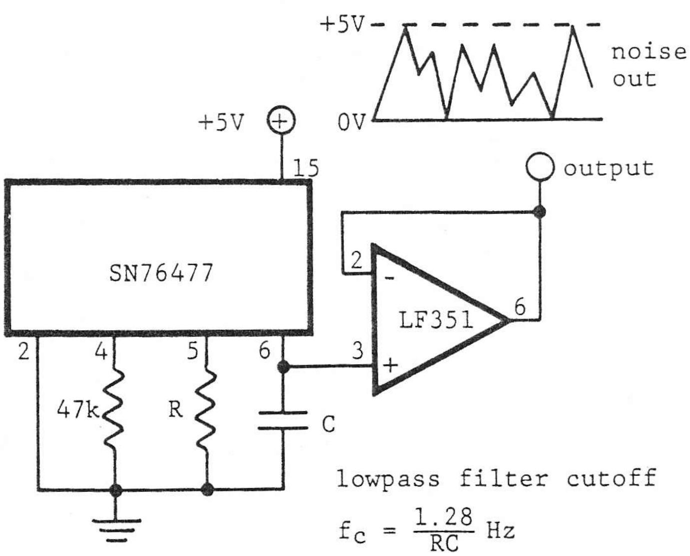


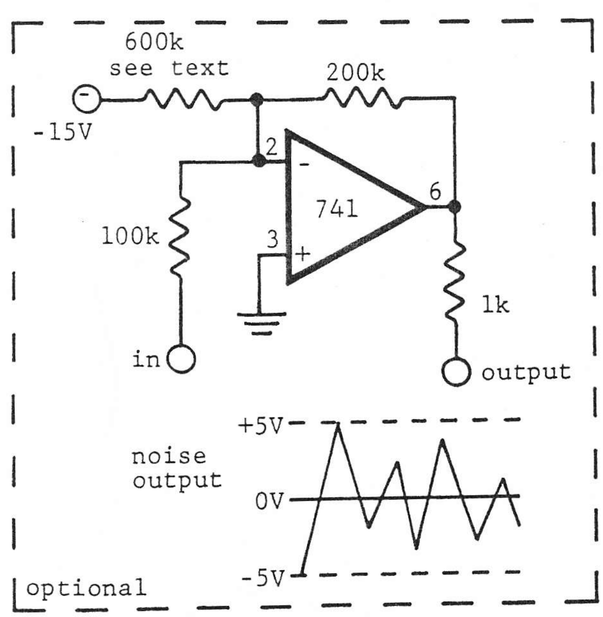


Fig. 3
Power supply connections to op amps assumed (±15V)

Tapping the One-Shot. Not so obvious is how we can tap the one-shot output (see figure 4). A negative going edge at pin 9 initiates the one-shot, while a resistor at pin 24 and a capacitor at pin 23 set the time constant. Note that we can tap the capacitor's charge cycle at pin 23 and get a triangle going from 0V to about 2.5V, or we can take a 5V pulse out at pin 8. Strictly speaking pin 8 is really the envelope generator output, but since we have no capacitor connected to it to charge and

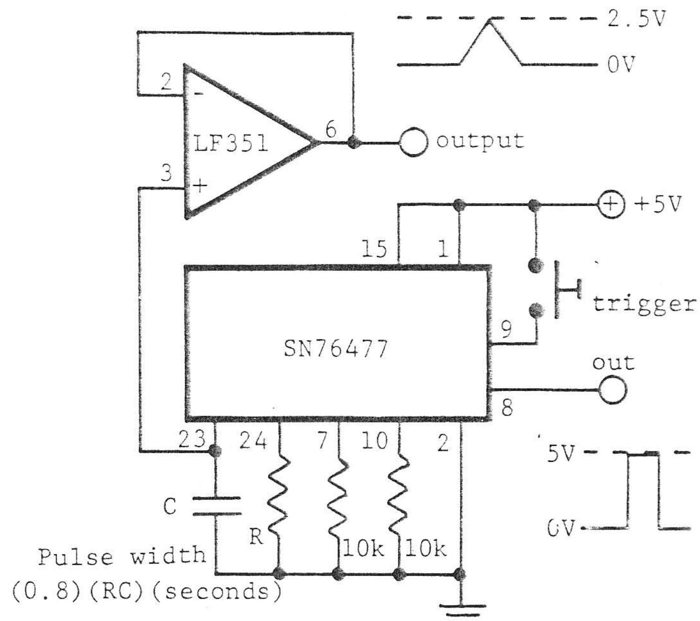


Figure 4 Power supply connections to op amps assumed ($\pm$15V)

discharge, the envelope generator opens and closes quickly. In other words, the envelope happens to be a pulse! The 10k resistors at pins 7 and 10 provide paths for what would normally be the charge and discharge currents of the envelope generator.

Summing Up. So far we've managed to isolate four different functions of the chip: Two oscillators, a noise source, and a one-shot. Now that these various outputs have been tapped, we are free to mix and combine them in any way we want. In other words, we have finally broken free of the internal signal routing. This versatility is exploited in the "Super Controller Module" coming up shortly.

We're running out of space, but not ideas. This article has just scratched the surface, and I'm sure there are countless more ideas just waiting to be found. Do you have any? If so, drop me a line c/o Polyphony. Next issue, we'll move on to the noise source...and if you thought all white noise sounded the same, you're in for quite a surprise!

PRACTICAL CIRCUITRY
TRICKS WITH THE SN76477
PART 2

THOMAS HENRY

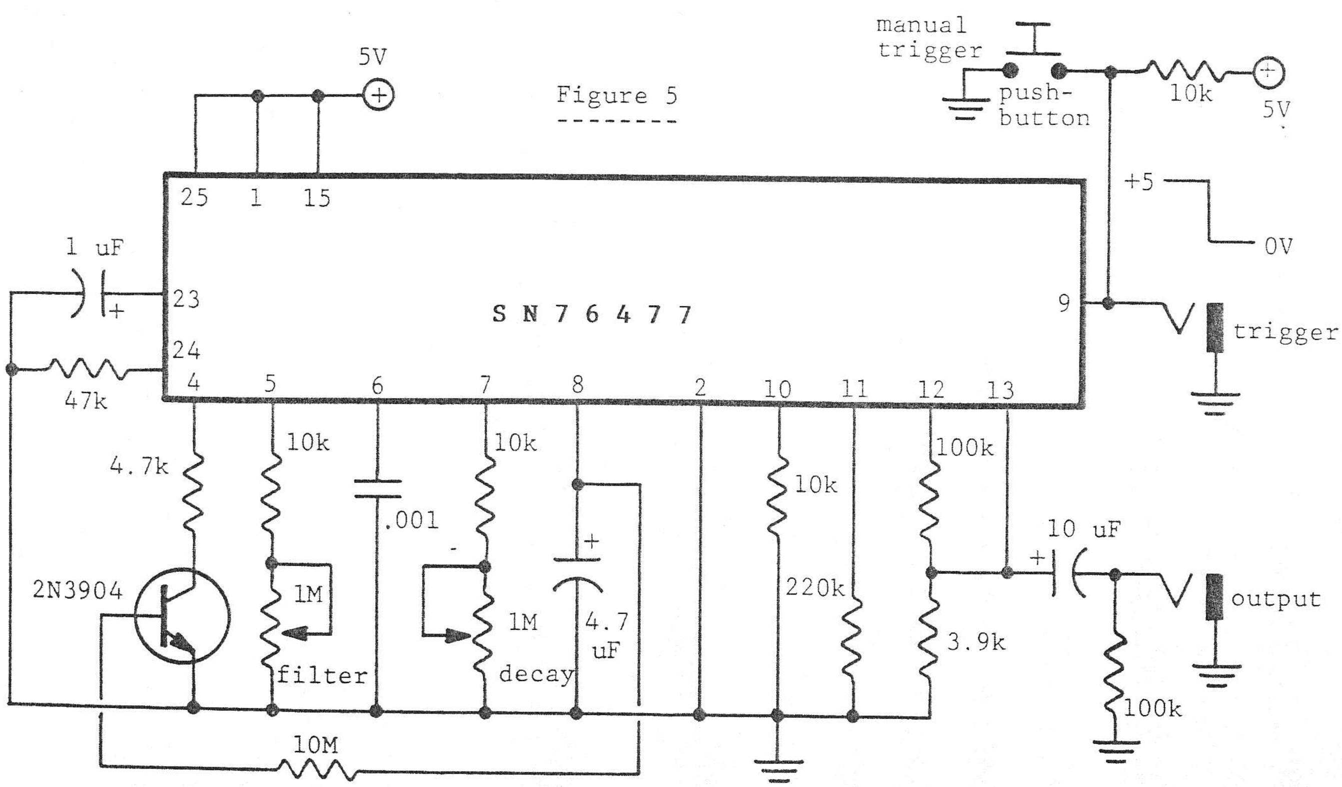
Last time we talked about how to find new outputs on the SN76477. In this installment, let's switch gears and talk a little about how to get more sounds from the noise source.

SN76477 noise source background. The white noise source in this chip is not the back-biased transistor you might be familiar with, but a binary pseudo-random noise generator. I don't want to get too heavily into the theory of this (see Don Lancaster's TTL Cookbook, Howard W. Sams, 1974, pp 277-283 for a good treatment), but all we really need to know is that the noise source is basically a shift register, with various bits moving down the register at a rate determined by a master clock. This clock can be either internal or external to the chip.

Let's consider an internal clock circuit first. Normally the resistor connected from pin 4 to ground sets the clock rate, and that's that. However, we could just as easily replace the resistor with a transistor (see figure 5). In this configuration the transistor acts like a throttle, and determines the current flow through pin 4, hence varying the clock rate.

Before getting too heavily entrenched in details, I suppose that I should say something about why we want to change the clock rate. The best answer I can give is "try it, you'll like it!". The sound is an incredible swooshing noise, and is very similar to phasing or flanging. The noise takes on a new tonality, and sweeping the clock changes the spectra in an eerie and dramatic manner.

White noise percussive voice. The circuit in figure 5 is specially adapted for a percussive voice, with the noise amplitude being controlled by the envelope generator and VCA. In addition, the envelope is tapped off of pin 8 and applied to the base of the transistor. The result is a voice which sweeps as the envelope dies away.



You will also note that the envelope is tapped via a 10 Meg resistor. The reason for such a large value is to avoid loading down the envelope generator capacitor. We could have buffered this voltage first using something like an op amp, but the loading caused by the 10M resistor is negligible, and is certainly cost-effective in this situation. Since the Beta (or DC current gain) of transistors varies from one unit to another, it may be that 10M is too large to allow the particular transistor you pick to turn on sufficiently. Feel free to experiment with other values; any value from 1M to 10M is permissible, but don't drop below 1M or excessive collector current may flow through pin 4 of the SN76477.

Before leaving figure 5, you should be aware that this would make a nice modification to the Percussive Noise Voice (John Blacet, POLYPHONY, Nov/Dec 19779, pp 12-13, corrections in the Jan/Feb 1980 issue, p 5).

Expanded noise sweep. Sweeping noise effects really appeal to me, and ever since discovering how to do them, I have been constantly at work developing new ways to employ this technique. One of the limitations of the circuit in figure 5 is that the internal noise clock is slightly limited in its sweep range; when you need lots of sweep, try the circuit in figure 6.

Here we avoid the internal noise clock completely, and use an external clock instead. Actually, the "external" clock is really internal to the chip, being the VCO. This is a good way to save parts, space, and wiring hassles. However, if you were planning on using the VCO for something else, you could always clock the noise with virtually any other type of square wave oscillator.

Let's analyze the circuit; consider the VCO first. R9 and a capacitor selected by S1 (C2, C3, or C4) set the basic VCO frequency. A control voltage applied to J1 is summed through IC2 and applied to the control voltage input of the VCO, which sweeps the VCO frequency. Since IC2 is an inverting stage, an increasing voltage at J1 yields an increasing frequency. R3 offsets the VCO if desired, or can be used as a manual sweep control. Trimmer R6 sets the zero point of the VCO so that 0V applied to J1 gives the minimum VCO frequency. Diode D1 prevents any inadvertent negative voltages from creeping into the VCO. The VCO's square wave output is taken off of pin 13 and sent to the external noise clock input at pin 3. Pin 4 is tied to the +5V line, which programs the chip to accept an external clock.

The rest of the circuit is the same as that described in the last installment, with the noise taken off of timing capacitor C1. We have to take out the noise this way since the chip's normal output, pin 13, is already committed to the VCO.

Calibration. To adjust trimmer R6, turn down R1 and R3, and advance R6 until the noise just begins to be audible. This sets the lowest noise frequency.

Once the circuit is calibrated, start experimenting. You could apply an ADSR to J1, or an envelope follower, LFO, sequencer, etc. - you get the picture; we're talking about a staggering amount of sounds, and they're all extremely useful. If you thought white noise had to be a static, one-dimensional effect, this circuit will definitely turn your head around.

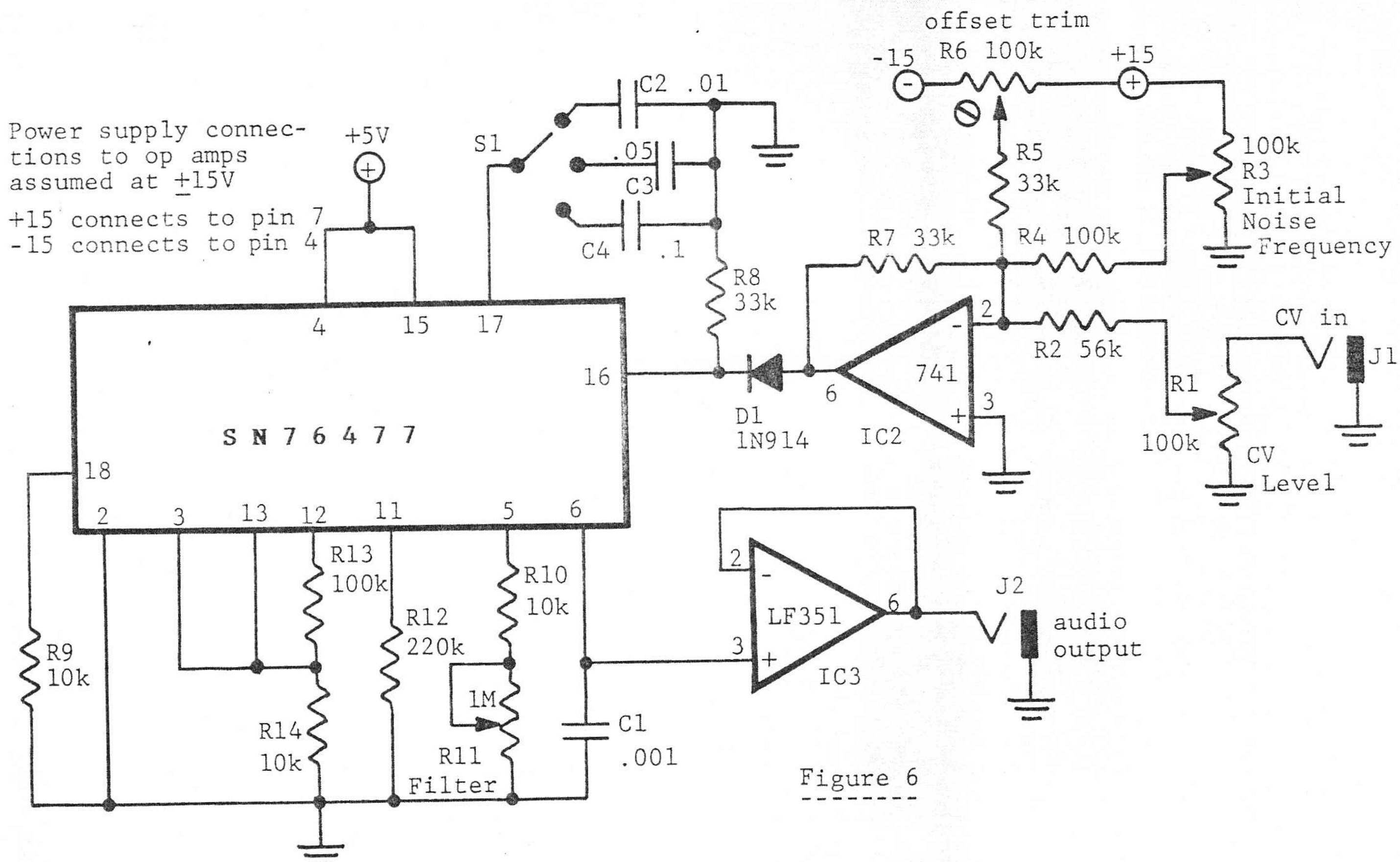


Figure 6

Next month, we'll tie together what we've covered so far, plus more, with a complete project, the "Super Controller Module". I think you're going to like this one!

Acknowledgement. My thanks go to Craig Anderton for turning me on to the use of a transistor to control the internal noise clock.

PRACTICAL CIRCUITRY

MORE ON SN76477-

If the VCO, VCF, and VCA form the heart of a synthesizer, this module is surely the brain! It gives you most standard controller options (LFO, noise source, and sample and hold), but also includes some extras that have no real precedent on commercially available synthesizers.

The Super-Controller Module (SCM) incorporates many of the SN76477 tricks discussed previously in this column. Since this is a BIG circuit, I can't waste much time or space in introductions, so let's get right into the design analysis. (see figure 1)

Power supply requirements. The SCM requires a regulated and well-filtered source of +15, -15, and +5V DC. Carefully note which points on the schematic connect to which power supplies; incorrect connections will keep the SCM from working correctly.

Noise source. This is basically set up in the same way as described in the last installments of "Practical Circuitry". An external control voltage, attenuated by R2, controls the noise clock frequency. The noise output is buffered via FET Q7; R60 helps cancel some of the offset inherent in the FET source follower configuration. The noise out is a standard, 10V p-p voltage centered about ground, with an output impedance of 1k.

LFO. The LFO has lots of options, so let's examine them one by one. To simplify matters, we'll ignore Q1 and its related circuitry for the moment.

As discussed in a previous installment, the triangle output of the SN76477 is tapped off at pin 21 and buffered by bifet op amp A2. The signal then branches off in two directions: the path through R33 is amplified by a factor of about 3.9, and level shifted by R35, which presents a 10V p-p (centered about ground) triangle wave at the output. R35 sets the approximate offset of the output; trimmer R37 centers the output exactly about ground. An LED at the output of A4 monitors the LFO rate.

This same triangle wave feeds comparator A6 and its related components. R41 adds 0.1% of hysteresis, which gives the re-

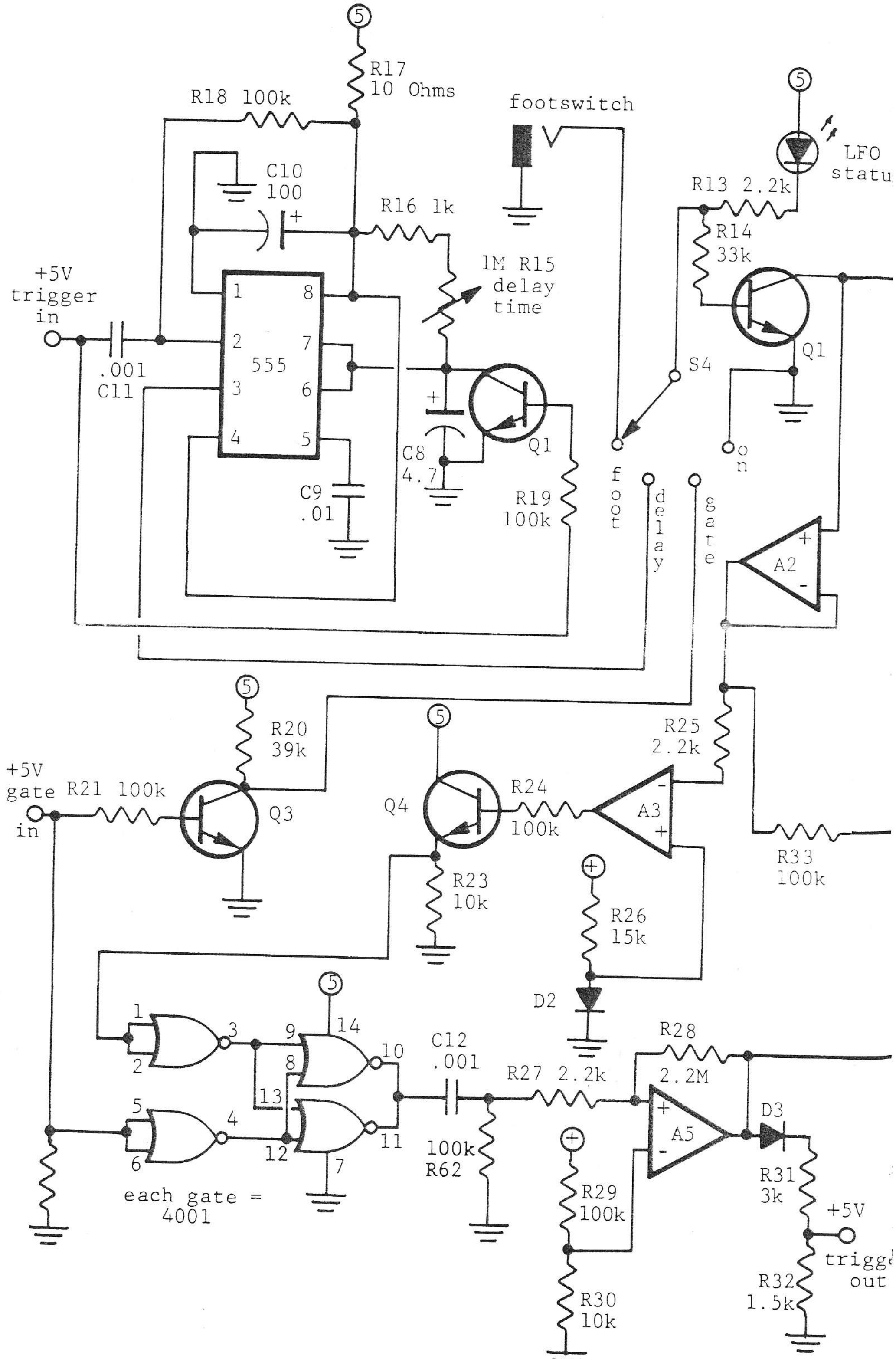
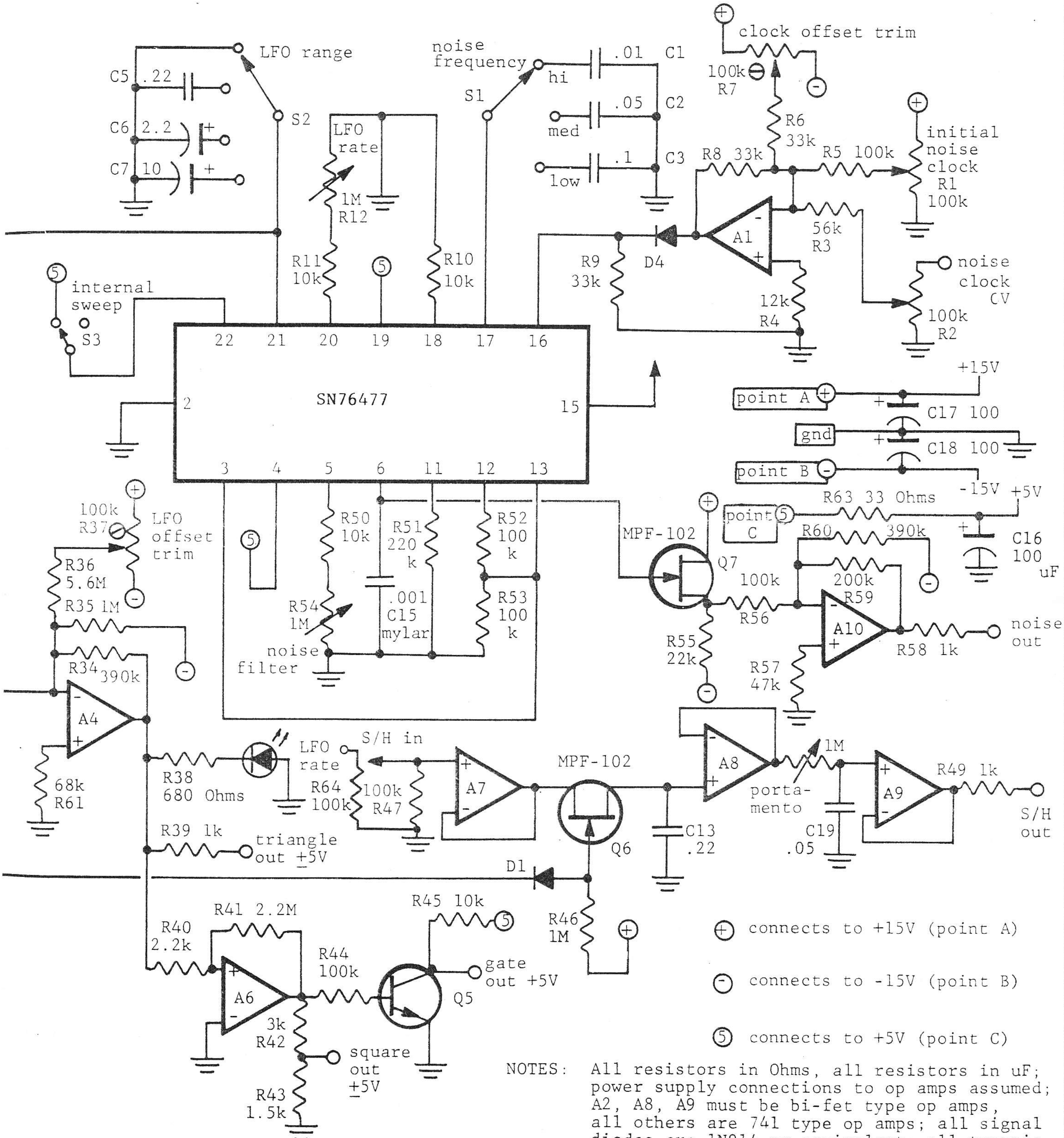


Figure 1: Schematic for the Super Controller

-THE SUPER CONTROLLER



⊕ connects to +15V (point A)

⊖ connects to -15V (point B)

⑤ connects to +5V (point C)

NOTES: All resistors in Ohms, all resistors in uF; power supply connections to op amps assumed; A2, A8, A9 must be bi-fet type op amps, all others are 741 type op amps; all signal diodes are 1N914 or equivalent; all transis tors are 2N5129 except as noted.

THOMAS HENRY

sulting square wave a nice clean edge. R42 and R43 attenuate this signal, giving a standard 10V p-p, centered about ground, square wave output with a 1k output impedance.

Since there are many times in synthesis when a bipolar 5V signal is inappropriate, a gate signal going from 0 to +5V is also derived from this square wave and is available at Q5's collector (the transistor is powered off the +5V supply, hence the 5V swing). This gate would typically be used for driving ADSRs or the like in sync with the LFO.

Now let's shift our attention to the circuitry associated with Q1. This is set up as a switch that can discharge whichever LFO timing capacitor is selected by S2. Not only can Q1 discharge the timing cap, it can also hold the LFO waveform output at 0V.

If S4 is **on**, then the junction of R13 and R14 is pulled to ground. This does two things. First, it shuts off Q1, meaning that the capacitor selected by S2 is free to charge and discharge normally, which lets the LFO run as usual. In addition, a current flows through R13 to ground, turning the **status** LED on. Whenever this LED is on, you know that the LFO is running.

With S4 in the **footswitch** position, and an ordinary SPST push-on/push-off footswitch plugged into the footswitch jack, closing the footswitch ties the junction of R13 and R14 to ground; thus, the LFO runs normally, and the **status** LED is lit. However, opening the footswitch releases R13/14 from ground, which initiates two events. First, the LED is extinguished; second, Q1 is turned on, and this shorts the LFO's timing capacitor to ground. In other words, the LFO is in a "hold" state and the extinguished LED indicates this. Footswitch control of the LFO, while simple to implement, can really add a lot of versatility to your sound.

Now consider another patch. First, set S4 to the **delay** position; then, take a 5V trigger out from your keyboard to the DELAY trigger input at C11. Patch the LFO of the SCM to an FM input of a VCO. Now let's follow the chain of events. Push a key down on the keyboard - a trigger enters C11, thus turning the 555 timer on for a time determined by C8, R15, and R16. This is a one-shot circuit, and so pin 3 goes high. Note that pin 3 is connected via S4 to the R13/R14 junction mentioned above. Since this junction is high, Q1 turns on and therefore turns the LFO off for the delay time set by the one-shot. After the one-shot turns off, the R13/R14 junction is brought to ground and the LFO turns on again. The result is a pleasing delayed vibrato effect, which I have found to be extremely well suited to creating string effects.

Note that C8, the timing capacitor for the one-shot, is bridged by Q2, in the same way that the LFO capacitor is bridged by Q1. This guarantees that each trigger coming in to the delay circuitry shorts out C8 momentarily, so the charging of C8 always starts from 0V. In more technical terms, the one-shot is retriggerable. Thus, as long as your keyboard is putting out triggers, the LFO is off (remember, each new trigger discharges C8 and starts the one-shot all over again). But as soon as the triggers stop, and as soon as the one-shot turns off, the LFO turns on again. And of course, the **status** LED monitors the whole thing. R15 adjusts the delay time.

Now it's time to consider the gating function of the LFO. To give you an idea of where we're going with this, we're going to set the LFO so that when we push a key down on the keyboard, the LFO puts out a series of triggers, suitable for driving an envelope generator. The result is repeating envelopes under keyboard control (pluck-a-pluck-a-pluck banjoes, anyone?).

Put S4 in the **gate** position and connect the gate output of your synth to the GATE INPUT jack near Q3. Suppose that no key is depressed. With no gate present, Q3 is off, and its collector is at +5V. The collector is coupled to the junction of R13/R14 via S4, hence Q1 is on, and the LFO is in a HOLD state. Now, depress a key. This sets up a chain reaction; current flows through R21 into the base of Q3, thus turning it on. This pulls the collector (and the R13/R14 junction) down to ground, which shuts off Q1, and allows the LFO to run. Hence the various LFO outputs are off and running.

Now consider the trigger output. Suppose the gate input is off (no keys down); this holds the LFO triangle wave at pin 21 at 0V. This is buffered by A2, and this 0V output is applied to the inverting input of A3. A3's non-inverting input is held at 0.7V by virtue of the voltage drop across the diode in series with the current limiting resistor, R26. Since the non-inverting input voltage exceeds that of the inverting input, A3's output is high. This "output high" condition is chopped down to a +5V level by Q4 (which is configured in a non-inverting manner) and the output of Q4 is coupled to an AND gate composed of the four sections of the 4001. The AND gate output is differentiated by C12 and squared up to a nice 1 msec pulse by A5, which is then clipped and attenuated by D3/R31/R32, presenting a 5V output trigger.

However, remember that we said the GATE input is off. As a result, the other input of the AND gate is low (pins 5 and 6 of the 4001), so no triggers appear at the output yet. But - push a key down (GATE on), and the AND output goes high, which allows triggers to pass to the output.

This may seem like an elaborate scheme for generating triggers under keyboard control, but it is necessary. Without this logic scheme, there would be a perceptible delay time between pushing a key down and the appearance of the first trigger. Trust me, other ways of attempting this fail miserably (I should know; it took me a half year to get all the bugs out of this project!).

The triggers at A5's output also clock the S/H. A7 is an input buffer for the S/H, and its output is applied to FET Q6 which acts like a switch. With no trigger applied, R46 keeps the FET pinched off, hence no current flows through the FET channel. But when a negative going trigger hits the FET's gate, it turns on, allowing whatever voltage is at the output of A7 to pass to the hold capacitor, C13. Then the FET turns off again and the charge is safely "sealed in" by the FET on one side and A8 on the other. A8 must be a bifet type op amp. The output of A8 reflects the charge on C13, and that charge passes across R48 into C14. R48 is a portamento control that can "glide" or "slur" the sampled voltages together. The output is buffered by bifet op amp A9, and is then presented to the S/H output.

Construction. I built my version on a printed circuit board, and since this is such a large project perhaps that is the best way to go. However, there is absolutely nothing critical about the circuit so there is really no reason why you couldn't build your version with perf board and flea clips. When I made my circuit board, I used photographic techniques since I had a notion that I might want more than one in my synthesizer system. That hunch

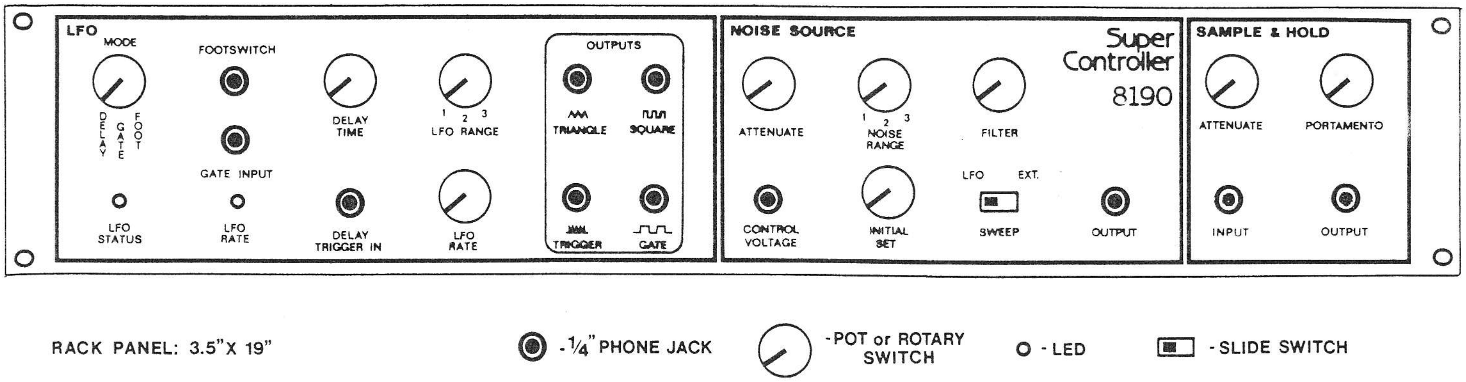


Figure 2: Panel graphics of the Controller

paid off! I think you'll find that with two Super-Controllers you can do both incredibly complex and amazingly subtle sounds.

I built my prototypes behind standard 3.5" by 19" rack panels. Figure 2 shows the layout I used. The circuit board is supported on angles behind the front panel and the whole construction has a nice solid feel to it.

Tuning the SCM. Tweaking the Super-Controller is quite easy. First let's set R37, which is the DC offset trim for the triangle wave output. The fastest way to set this is to simply monitor the triangle wave on an oscilloscope and set the control so that the triangle wave is symmetrically oriented about ground. If you don't have a scope, monitor the triangle wave with a center-zero Voltmeter (most inexpensive VOMs have such a function). Then, set the trimmer so that the needle on the meter swings an equal amount on either side of zero Volts. Or if you're not fussy about zeroed out triangle waves, simply set the trimmer to mid-position and leave it. This will give sufficiently close results.

To adjust the clock offset trimmer, R7, monitor the noise source with an amplifier. Turn R1 and R2 completely down, then spin R7 around a few times to get familiar with its effect. At one extreme you won't hear anything through the monitor amp; at the other extreme you will hear a very shrill white noise sound. Starting from a no-noise position, ease the trimmer up until the noise just starts. This is the optimum position. By setting the trimmer in such a way, the initial set pot, R1, will have a full range effect.

Well, that's it...all built and all tweaked up; we're ready to make some music!

Using the SCM. There are zillions of possible uses for the Super-Controller, and hardly a week goes by that I don't see some new, off-the-beaten-path way of using this machine. But I've had over a year now to play with the thing, and experience is what really counts. To help you get started on collecting your own experiences, I'm going to detail four very simple patches. But let me reiterate, these simple patches are for example only. Once you're familiar with these, I'm sure that you'll find many more.

Delayed vibrato. Figure 3 shows the first patch, a delayed vibrato effect. Follow this patch chart carefully, with your own system and the SCM. Note that triggers come from the keyboard to the delay trigger input. To make this more fun, you might want to set up the VCO, VCF, and VCA to approximate the sound of a violin.

Now push a key down and hold it. At first the note will appear without any vibrato, and after a certain amount of time the vibrato will enter. Experiment with various settings of the delay time pot, R15. You will note that since the delay time

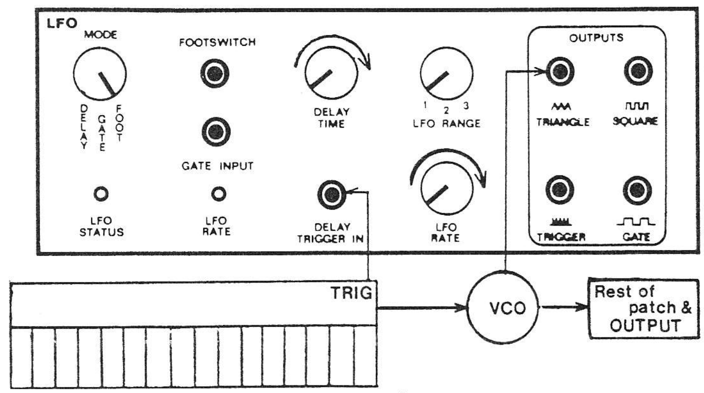


Figure 3: Delayed vibrato

is retriggerable, a series of triggers will keep firing and refiring the timer for as long as you keep playing. The upshot is that as long as you keep your fingers moving on the keyboard no vibrato will occur, but as soon as you stop and hold a key down, the vibrato will appear after the delay time has elapsed. This can be very useful when you want no vibrato until the last note of a passage.

Gated repeating ADSR effect (see figure 4). In this patch the gate from the keyboard determines when the LFO is on, and gate and trigger signals from the LFO fire the ADSR. Push a key down and you get a series of repeating sounds; let up on the key and they stop. As mentioned earlier, you can do some great plucking banjo sounds with this patch if that's your bag.

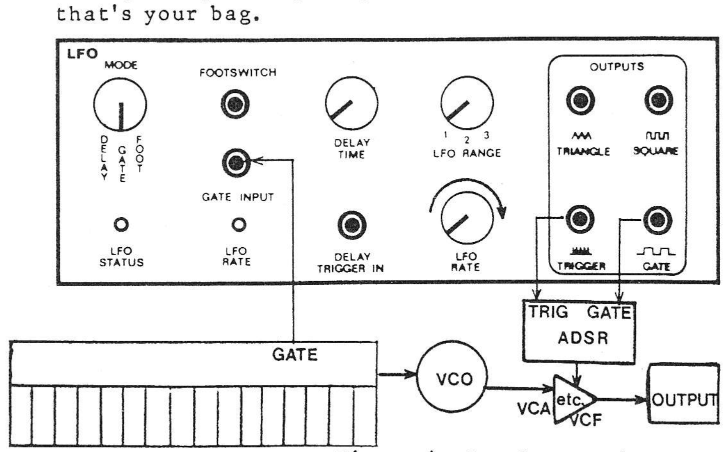


Figure 4: Gated repeating ADSR

Basic noise source patch (see figure 5). The ADSR (besides modulating the VCA) also sweeps the noise source. You'll note that there isn't any filter in this patch, and yet push a key down and what do you hear? A sound very much like a low pass filter being swept! Adding a filter to this patch really intensifies the sound. This basic patch can form the basis of some really far-out percussion effects.

Sample and hold (see figure 6). The noise source is sampled by the Sample and Hold unit, and the output is then sent to the VCO. The result is a series of random pitches, with the SCM's own LFO controlling the rate of these pitches. Add some portamento by turning up R48 and these random pitches become slurred. If you're into movies, you'll find that you can do a great "R2D2" from "Star Wars" with this patch, especially if you add some portamento.

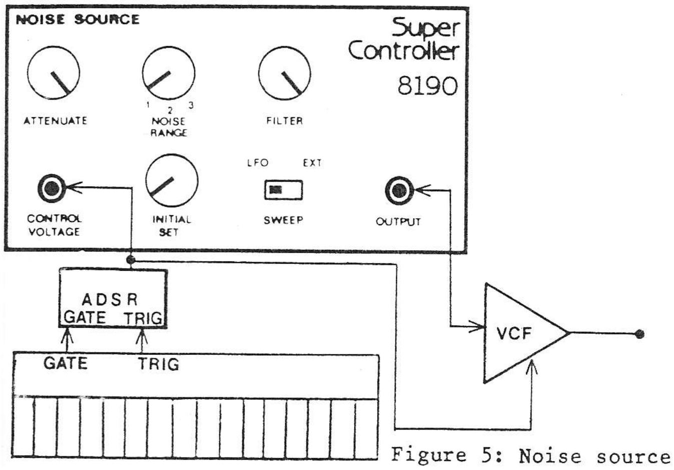


Figure 5: Noise source

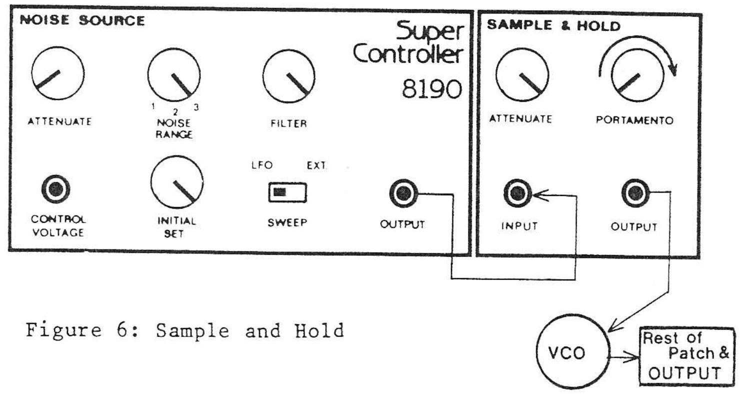
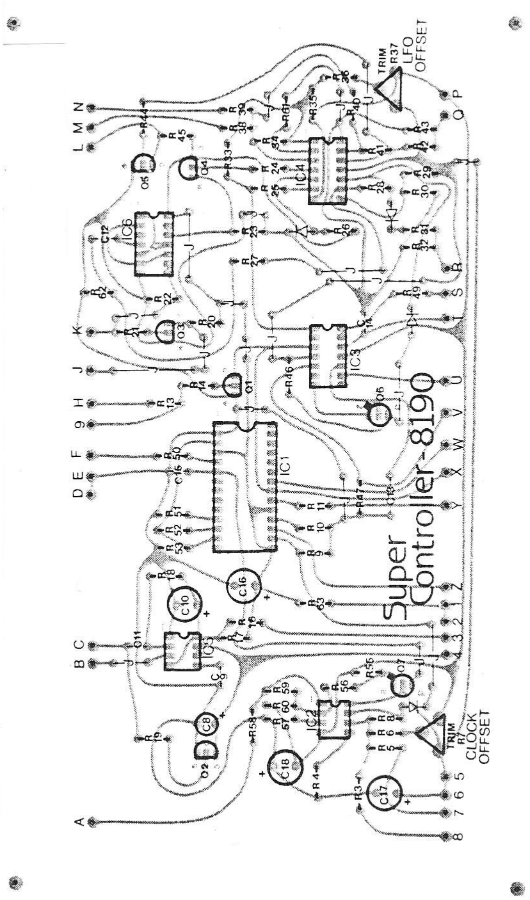


Figure 6: Sample and Hold

Conclusion. Well, sad to say I'm out of time and out of space again. This is such a fun module that I could go on and on telling you about some of the neat patches that I've come up with...but since I can't, you'll have to find them yourself (which is at least half the fun of this kind of project anyway). The best way to really discover new patches with this thing is to understand the circuit thoroughly. Once you know how it works, you'll be well prepared for applying the module intelligently and creatively.

Be sure to stop back next issue, when I'll present another exciting new synthesizer module. Until then, happy Super-Controlling! *f*



SCALE: 1/2" = 1'
SUPER CONTROLLER circuit board and parts placement graphics.
NOTE: the circuit board pattern is reversed to show parts placement.

The following are available from Polymart (use order form page 25):
Etched and drilled circuit board with parts placement graphics...#8190pc.......................$22.50.
Punched, painted, and screen printed Rack mount Front Panel...#8190fp.........................$19.95.
Please add $4.00 postage and handling to your order.

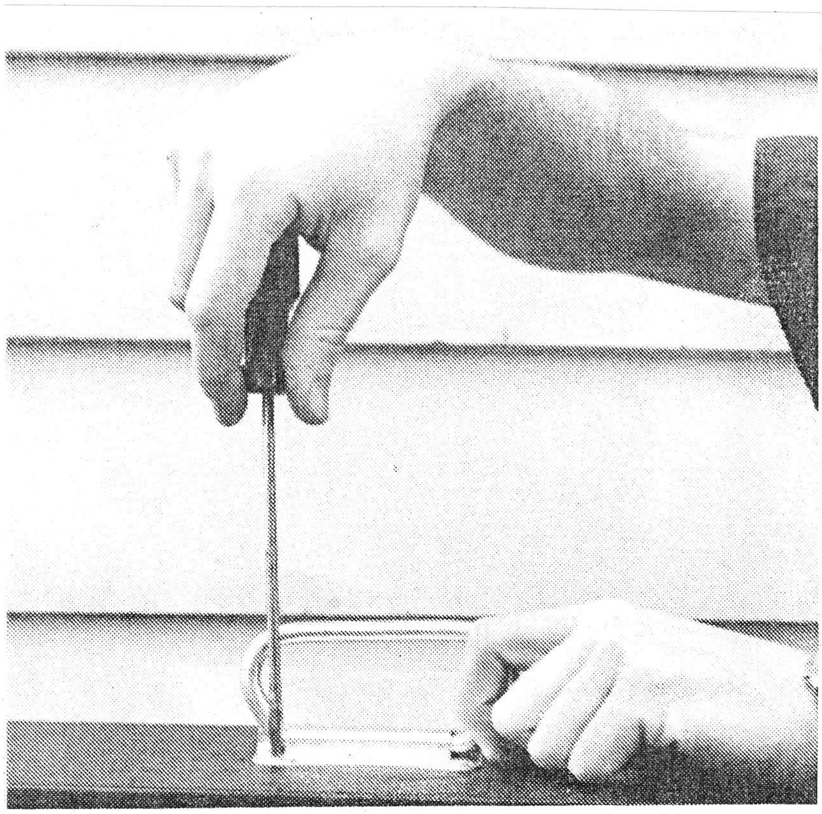
LIVE PLUS TAPE - NEW TECHNIQUE

Continued from page............20 -fying performers (who would hopefully be in some form of contact with the source performers to pick up their cues) could build dense and complicated textures with their loops.

Complications? Particularly with the last two techniques, it may seem that I wandered quite a bit from my original goal of making live performance easier and in particular less expensive. However, consider that a pair of four-track decks would work fine to handle the oft-illustrated four loops. Since live performance does not always demand studio quality, a pair of 7.5 IPS decks (such as two TEAC A2340s) would not only do the job, they could be bought used for less than $1000 total - cheaper than many synthesizers. In those techniques requiring a synthesizer, something as simple as a minimoog or Korg MS-20 would be satisfactory. With techniques requiring modifiers, certain selected modules could be easily configured in a case. All of this is considerably cheaper than buying a couple of programmable poly machines or even some digital behemoth. These techniques also tend to be less demanding of a performer's skills, which would not only give less-experienced performers a chance to play but also make the music a more intuitive and interactive piece that matches the mood of the occasion.

I am sure that there are many other live performance techniques other than the ones that I have mentioned, and that we're dealing with an open-ended field. In the even that you feel demented enough to try any of these ideas, let us know the results and what you've learned. Let's keep those information channels open! *f*

And now for a page from The Scrapbook...



Building electronic music equipment requires many different skills. As this junior high graduation picture demonstrates, I fully mastered advanced tool techniques in 8th grade boy's wood shop.

PRACTICAL CIRCUITRY

VCO

NOTES

A1 - A4 = ¼ 4136, IC2
Q1, Q2 = 2N4124
D1, D2 = 1N4148

All capacitors in uF, except where noted.

For dual unit, repeat all parts except for R31, R29, C10, C7, and IC3. IC3 is a dual flip-flop, and the other half may be used independently.

Power to IC2: V+ = pin 11
V- = pin 7

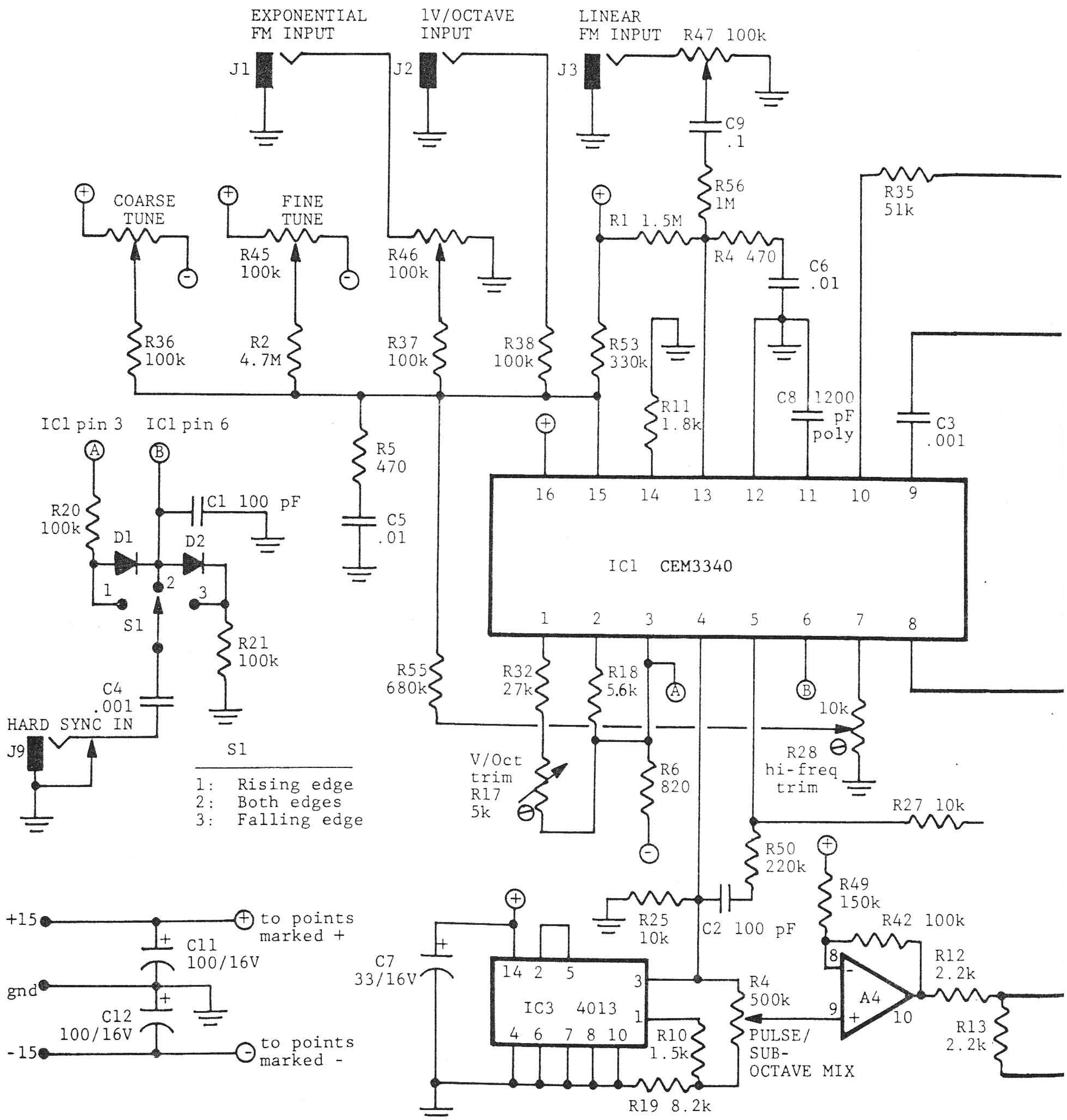
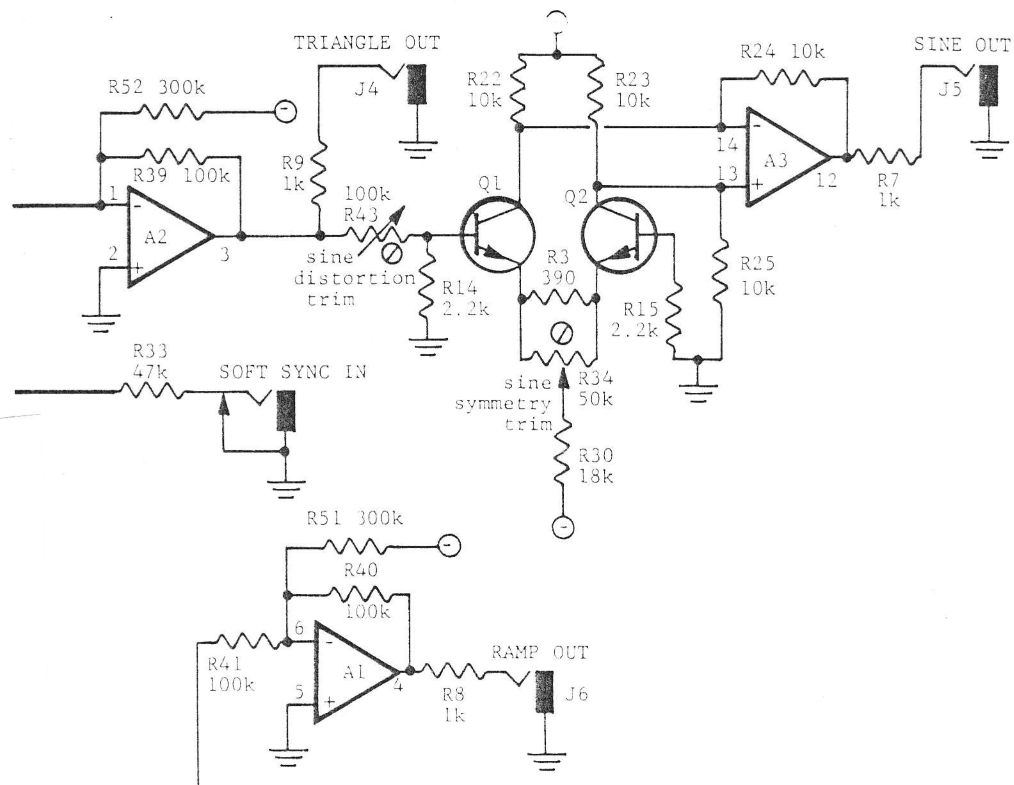


figure1

DELUXE



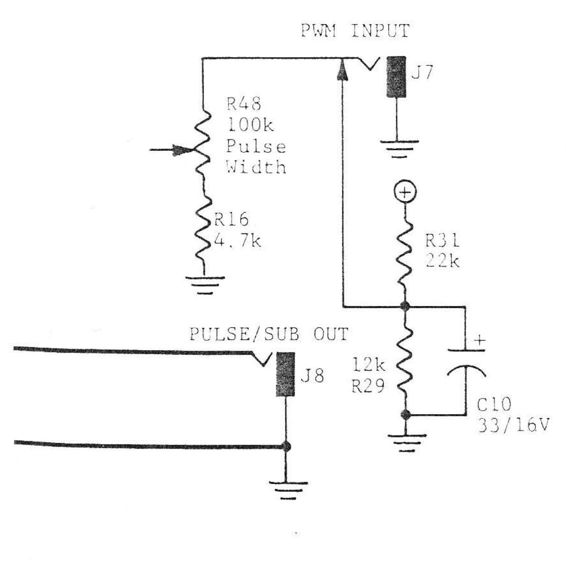


TABLE I: Specs for the VCO Deluxe

OUTPUTS:	Sine, triangle, ramp, pulse and sub-octave pulse All output impedence 1k. Pulse and sub-octave continuously mixable to a single output. All outputs DC coupled.
INPUTS:	All input impedences, except hard and soft sync, are greater than 50k.
CONTROL VOLTAGE INPUTS:	1V/octave. Exponential FM. Linear FM (AC coupled). Coarse and fine tuning.
SYNC INPUTS:	Soft sync (set for 10%). Hard sync (switch selectable for reset on rising edges, falling edges, or both).
PWM INPUT:	Manual or voltage controlled.

It's time to start building again! This issue we're going to work on a VCO with lots of features; in fact, I call it "VCO Deluxe". Since this is a fairly big project I won't be able to dwell on any point in great detail. However, since the circuit is based on the CEM3340 VCO chip, and since that chip is supplied with an excellent data sheet, I don't think you'll have any trouble filling in the details.

There are a number of features in this circuit that pop up again and again, so to save space, Table I summarizes these common features in the form of a "spec sheet". These specs will make many facets of the design easier to understand.

We can breeze through the circuit description now, especially since the CEM3340 does most of the hard work for us. Refer to the schematic in figure 1. J1 and J2 are the two exponential control voltage inputs. The voltage applied to J1 can be attenuated by R46, while the voltage at J2 is left alone and is simply the 1 Volt per octave input.

J3 and its associated attenuator R47 form the input for linear frequency modulation. This would commonly be used for vibrato (FM by an LFO), or for creating gong sounds (FM by another VCO). This input is AC coupled which makes it very easy to use, since you don't have to worry about any DC offsets shifting the fundamental pitch.

R44 and R45 are the coarse and fine tuning controls. R44 covers a very wide range; in the farthest counterclockwise position, oscillation is well below 0.5 Hz. At the other extreme of rotation, the oscillation is around 35 kHz. The fine tuning control has a much more restricted range and covers a musical interval of about a fifth.

We've now overed the entire frequency control input structure comprising two tuning controls, the exponential FM input, the 1V/octave input, and the linear FM input. Now let's take a look at the sync inputs.

The soft sync signal is injected at J10 and should be some sort of rectangular pulse. This

THOMAS HENRY ©

sync input responds only to negative going triggers. Resistor R33 is installed to limit the amount of synching to about 10%. This seems about right to my ear, because more synching causes greater distortion and starts to sound like hard sync, while less sync gives inferior phase locking. However, feel free to experiment with R33.

The hard sync signal is injected at J9. Now to understand S1 and its associated circuitry, we must be aware that the CEM3340 hard sync input responds to either positive or negative going pulses. This being the case, it is easy to see that if the switch is in position 1, only positive sync pulses are transferred to the sync input at pin 6. Negative pulses are blocked by D1. Likewise, when the switch is in position 3, only negative pulses are transmitted to pin 6. When the switch is in position 2, both positive and negative pulses are coupled to the sync input of the chip. The upshot is that we have three types of sync sounds. For more details on this sync-switch structure, see my recent article in Electronotes[2].

The pulse width modulation input is at J7. In the absence of any plug being inserted in J7, pot R48 manually controls the pulse width via a fixed voltage supplied by voltage divider R31 and R29. However, inserting a plug into J7 removes the fixed bias voltage and allows for voltage controlled pulse width modulation.

Well, we've covered all the input considerations; now it's time to talk about the outputs. Remember, our "standard" is to have 10V p-p outputs, centered about ground, with a 1k output impedance. With that in mind, let's look at the triangle wave output first.

The triangle is available at pin 10 of the CEM3340, however it is a non-standard voltage. A2 level shifts and amplifies this signal by a factor of 2, making it standard.

The triangle wave also feeds the sine converter formed by Q1 and Q2. If you're used to the old 3080 type sine converters, this one will give you quite a surprise. Distortion is incredibly low, even with garden variety transistors. What makes this converter superior to the 3080 type is the inclusion of feedback via R3, which practically nulls out the "pip" on the extreme end points of the sine wave[3]. Trimmer R43 minimizes the odd harmonic distortion, while trimmer R34 adjusts the even order harmonic distortion.

The ramp wave makes its exit via J6. It should be obvious how A1 level shifts the ramp output at pin 8 of the CEM3340.

The pulse wave is available at pin 4 of the CEM3340. R25 is a pull-down resistor for the emitter of the internal transistor. The pulse developed across this resistor then splits off in two directions. First it feeds pot R54, which is the Pulse/Sub-octave blend control. Then the pulse goes to the clock input of a CMOS 4013, configured as a binary divider. The output of the binary divider goes to a voltage divider comprising R10 and R19. This resistor string chops the sub-octave wave down to about a 13V level, which is the level of the original pulse. This means that both sides of R54, the blend control, see signals of the same amplitude, allowing smooth blend transitions. The wiper of R54 is buffered, amplified, and level shifted by A4, which is configured in a rather unusual way. If you imagine that R49 goes to ground, it is easy to see that A4 becomes just an ordinary non-inverting amplifier. However, pulling R49 to the positive supply instead of ground level shifts the input signal and at the same time retains the non-inverting amplifier characteristic[4]. The result is an output that is 20V p-p centered about ground, which is then chopped down to 10V p-p by divider R12 and R13. This combination of 2.2k resistors gives approximately a 1k output impedance at the same time.

So, by utilizing this somewhat strange configuration, we can see that rotating R54 in one direction gives full pulse wave, with no sub-octave. Rotating it in the other direction adds an increasing amount of sub-octave wave, until the pot has been rotated all the way, at which point the output is full sub-octave. And we've kept a 10V p-p signal the whole way around the rotation!

Calibration. Tweaking the module isn't all that hard, but it does take patience. The spec sheet gives the details on how to tune up trimmers R17 (the V/octave trim) and R28 (the high frequency error trim), so I won't say anything about them! The sine trimmers, R43 and R34, are best adjusted with an oscilloscope, with your ear serving as the final arbiter. While watching the waveform, tweak R43 until a nicely rounded sinusoid is formed. Then adjust R34 for a symmetric waveform. While performing both of these trims, listen to the sine wave on an amplifier and speaker (at around 500 Hz), and note the relationship between waveshape and timbre. Believe it or not, your ear will allow you to tweak up the sine wave for very low distortion. However, if you have a distortion analyzer...

Well, it's too bad that there's not enough space to discuss every point of this VCO, for I think it's a really good one. However, if you take the time to study the CEM3340 data sheet, I'm certain that this will clear up many of the fine points. In the meantime, enjoy the VCO Deluxe and we'll discuss something new in this space next issue.

References

1. The CEM3340 and spec sheet are both available from PAIA Electronics.
2. T. Henry, "Two Hints on Using the CEM3340 VCO IC", ELECTRONOTES, Volume 13, Number 121, January 1981, pp. 13-16.
3. For more information on discrete triangle to sine converters, refer to W. Jung, "Triangle to Sine Wave Converter", IC Op Amp Cookbook, Howard W. Sams and Co., Indianapolis, 1979, pp. 386-388.
4. This method presupposes that you are using a well regulated power supply.

And now for a page from The Scrapbook...

I first became interested in designing electronic music equipment back in 1974 when I joined the band *Spiff Cool and the Keen-O-Jets*. In the back row, from left to right, are Riff Andrews, Spiff Cool and Howie Buttz. Crouching in front you'll find me on the left and Angelo Spimoni on the right.

PRACTICAL CIRCUITRY

We're starting to accumulate a pretty good collection of synthesizer circuits here in "Practical Circuitry". In fact, I hope to present enough plans over the next year to allow you to build a complete synthesizer. So far, we've covered the noise source, LFO, and sample-and-hold (all part of the SuperController module), and a fairly elaborate dual VCO module. Now it's time to start thinking about amplitude modulation, hence this month's dual VCA (and a companion dual ADSR, which we'll cover next month). The dual VCA and dual ADSR fit conveniently behind a 19" by 3.5" rack panel; if you wish to use the same construction technique, you may want to wait until next issue before you start building. I think you'll find that a dual VCA/dual ADSR module makes for a pretty sharp combination.

Now, about that title. The name of the game is "think simple", since VCAs can get pretty wild in a hurry. For example, the VCAs used in a dbx unit, or for automated mixdown, can get very complicated. This is because such applications demand very wide range, low noise, and accurate tracking. However, for electronic music it isn't necessary to go that far. You know the old saw about the human ear being relatively insensitive to amplitude variations as opposed to frequency changes? Well, it's true. So we can cast out accurate tracking as being of rather minor importance to us. However, we would like a fairly wide dynamic range, and this also implies that we would like low noise as well. Finally, we want something that won't cost us an arm and a leg (remember, we're going to build dual units).

I think there's a good solution to the above requirements: the CEM-3330. If you're not familiar with this chip, let me tell you about it. The CEM-3330 is a dual VCA designed specifically for electronic music. It has "standard" input and output structures, is easy to apply, offers linear or exponential response and a choice of class A, B, or AB operation, and best of all, it's relatively inexpensive - about $8. If all this sounds good to you, be sure to check out the spec sheet to get a real idea of the power of this chip.

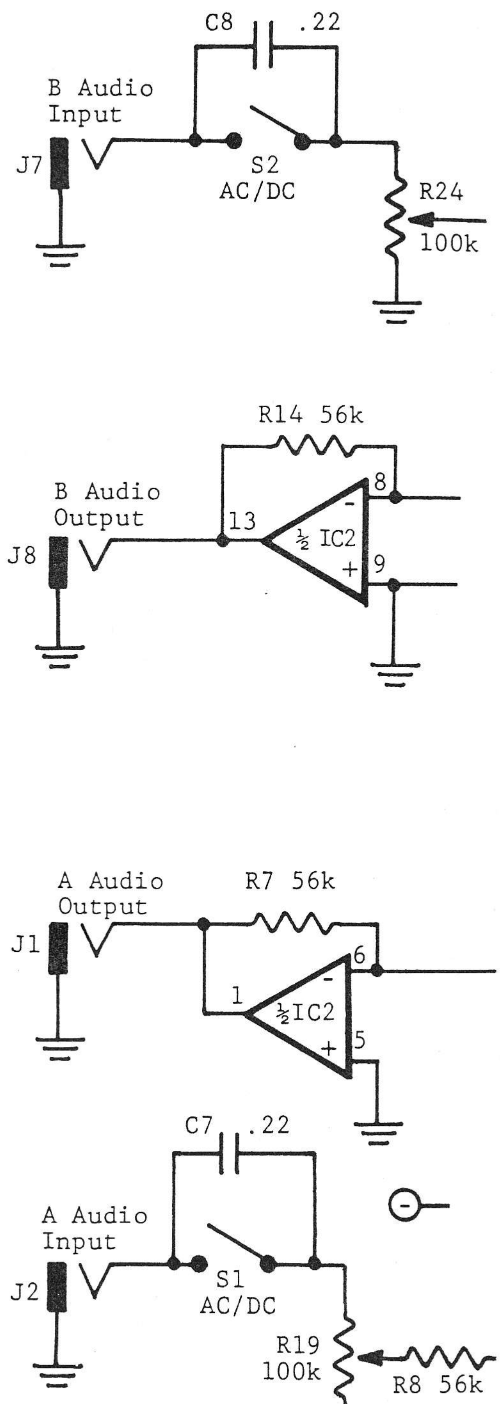
"All right, I'm sold", you say - "Let's get cooking!". I'm with you - but wait a second while I justify still further the title of this article. As I said, this chip has a lot of features; but that doesn't mean we have to use them all. So in this design, I'm going to strip down, streamline, and sometimes ignore various features of the CEM-3330. What's left is a terrific, low noise, and simple to build VCA.

The audio signal path. Figure 1 shows the schematic. Since this is a dual unit, I'll only describe one half (the other half is identical, except that the power supply connections are shared by both halves). The audio input enters through J2; S1 chooses either AC coupling (for audio processing) or DC coupling (for processing control voltages). If the audio signal were not centered about ground (i.e. if it had a DC offset), you would hear a terrific "thump" every time you opened the VCA quickly. We can certainly do without that, and that's the purpose of putting C7 into the circuit. To further reduce thumping, trimpot R17 allows you to trim out any residual offset in the chip itself. To set this trimmer, repeatedly hit some fast envelopes and adjust R17 for minimum thumping in the audio output.

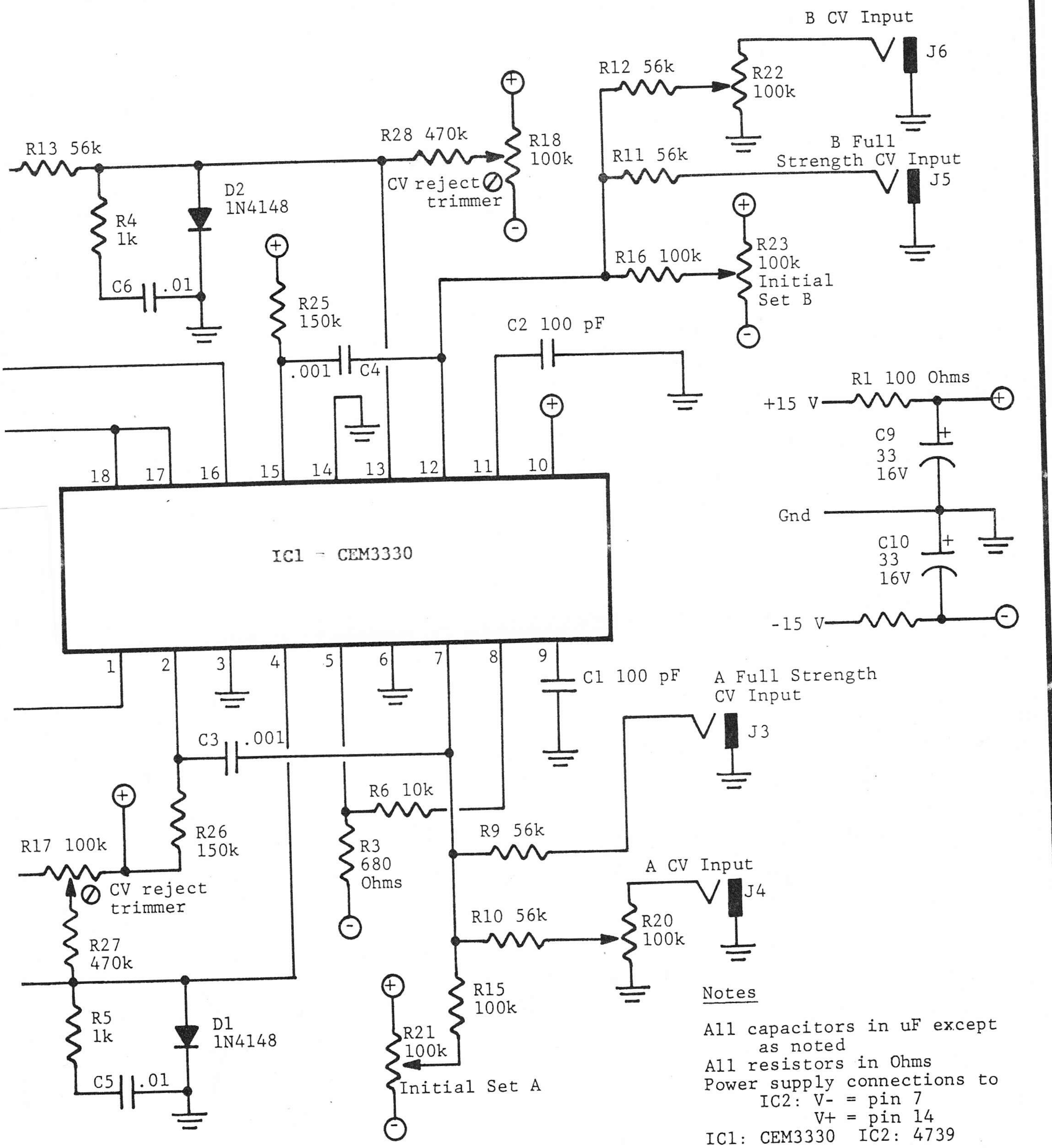
R19 is an attenuator that pares the signal down to size as needed. This design follows a standard that I have been using for some time now, namely ±5V audio signal levels and 0V to +5V control signals. With R19 wide open, the VCA accepts ±5V audio signals.

Now for a few words about D1, R5, and C5. Ordinarily pin 4 should remain at a voltage somewhat below 0.7V. If that voltage were to rise above this level, latch-up and possible damage to the chip would ensue. D1 makes sure this never happens by clamping pin 4 to a maximum of 0.7V (a diode drop). R5 and C5 form a compensation network. Their job is to help C1 (the actual compensation capacitor) keep the amplifier from breaking into supersonic oscillation.

For you "theory" buffs, this VCA is a current in, current out, current controlled amplifier. R8



VCAs: THINK SIMPLE



.... Continued on page 27

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PRACTICAL CIRCUITRY

........continued from page 17

converts the input signal to a current, while the output current is converted to a voltage via R7 and A1. The output is then presented to J1. Once again, this circuit is set up to give a $\pm$5V output signal under normal conditions.

The control signal structure. The exponential control, at pin 6, is not used in this circuit so we simply ground it. My reason for going with a linear mode VCA is simple. Almost all ADSRs (including the one I'll be presenting next issue) already put out an exponential control voltage. This being the case, it makes sense to follow the ADSR with a linear VCA since the end result will be an exponential envelope anyway. This is perhaps the most "natural" envelope. However, suppose that we were to follow the exponential ADSR with an exponential responding VCA. The result would be "exponentiating the exponential", an unusual envelope. So let's play it simple and ground pin 6.

Pin 7 is the linear control input, which, unlike pin 6, is at virtual ground (which in plain language means that we can sum as many inputs into this node as we want). The full strength CV input feeds J3 and then R9. The value of this resistor is such that a 0V to 5V input yields full off to unity gain. This is the input that you would normally use with an ADSR. J4 is another control voltage input with an associated attenuator. You would use this input most commonly for tremolo or amplitude modulation applications. R21 (linear tape) is an initial set pot, with wiper centered giving full off. This pot can be used to manually open up the VCA, or to offset any possible bias present in a control signal applied to J4.

C3 is a rather new development. If you have the spec sheet for the CEM-3330, you won't find C3 mentioned anywhere. But if you are a subscriber to Synthesource (Curtis Electromusic, 110 Highland Avenue, Los Gatos, CA 95030), you may remember a little note that was presented in the Winter 1981 issue, page 11, on how to stabilize the linear control input. This capacitor helps prevent spurious oscillations at low control currents.

Well, that's it: VCAs the simple way! I have four of these VCAs in my system and love them. I confess that VCAs are not exactly the most exciting modules in a synthesizer, but if you have suffered through differential pairs, FETs, and CA3080s like I have in the past you will really appreciate the simplicity and reliable operation of this chip.

Looking ahead. For a dual VCA/dual ADSR circuit, a good printed circuit board layout is important, since stray capacitance can be troublesome. If you want to see how I implemented this dual module, stop back next issue when I present the ADSR and say a few things about construction. ♯

Specifications

Audio Input:	AC/DC selectable $\pm$5 signal level 50k input impedance, with attenuator
Control voltage:	Full strength input, (0 to 5V), 50k impedance Initial set, from 0 to greater than unity gain

PARTS LIST

All resistors in Ohms; 1/4W; 5% preferred.

R1, R2	100	R15, R16	100k
R3	680	R17, R18	100k trimmer
R4, R5	1k	R19 - R24	100k pot
R6	10k	R25, R26	150k
R7 - R14	56k	R27, R28	470k

All capacitors rated 15 VDC or higher.

C1, C2	100 pF
C3, C4	.001 uF
C5, C6	.01 uF
C7, C8	.22 uF
C9, C10	33 uF, electrolytic

Semiconductors

D1, D2	1N4148 or equivalent
IC1	CEM-3330
IC2	4739 dual op amp

Mechanical parts

J1 - J8	Open circuit 1/4" phone jack
S1, S2	SPST slide switch
Misc.	Knobs, wire, solder, hardware, etc. ♯

And now for a page from The Scrapbook...

I helped form the *East Side Pharaohs,* and played with them for nearly 18 years before moving on to browner pastures. This is an early incarnation of the band, which has grown and changed considerably. That's Clancy on the left, me in the middle and Riff on the right. The *Pharaohs* are still going strong and are considered to be the Midwest's zaniest (and most educational) band.

PRACTICAL CIRCUITRY

By: THOMAS HENRY

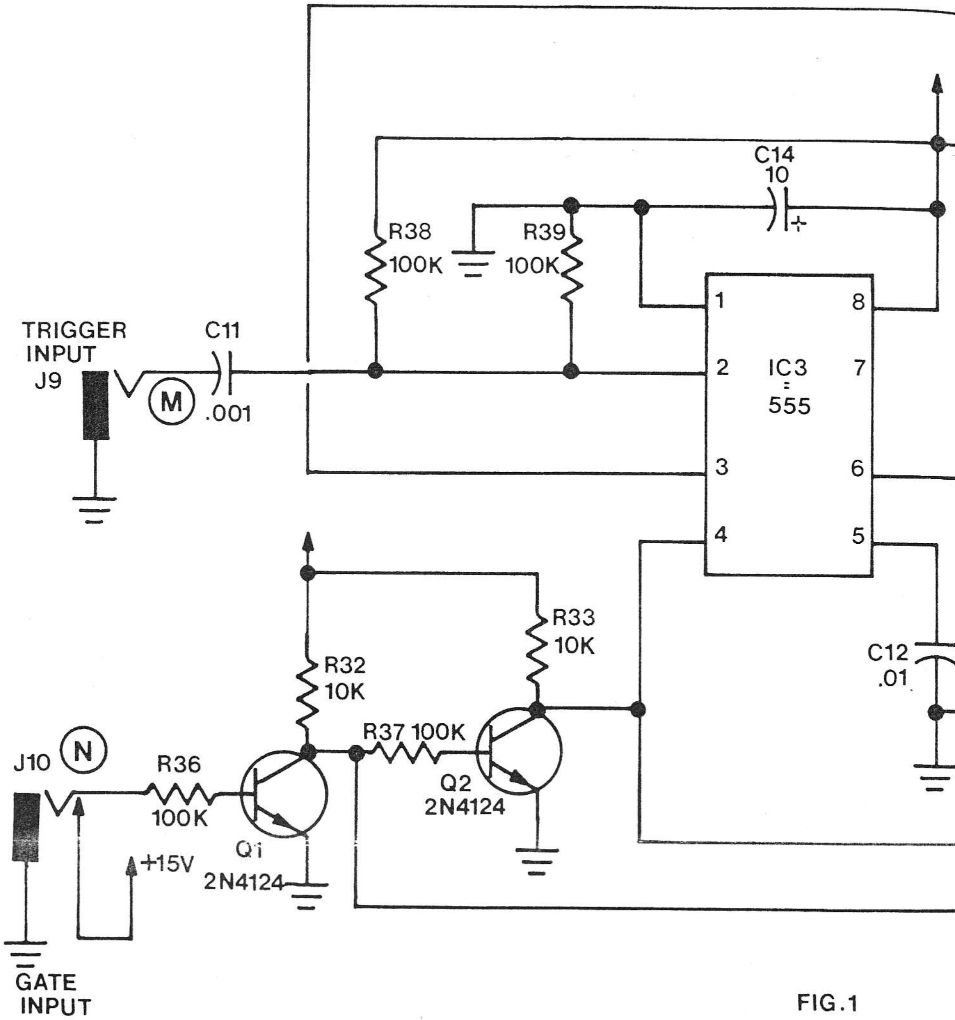


FIG.1

In last issue's "Practical Circuitry" we used new integrated circuits (developed for synthesizers) to make some high quality, low cost VCAs. This time we're going to shoot for the same results, but instead of relying on the technology of new integrated circuits we're going to dip back into our bag of "old standbys" and pull out the trusty 555 timer IC. However, let me hasten to add that this ADSR uses the 555 in a somewhat nonstandard fashion, and therefore achieves more sophistication than you might expect from such a "simple" circuit.

The 555. The 555 is an industry standard chip that's easy to use; the intended applications are monostable (one-shot) and astable (oscillator) circuits. But instead of thinking of the 555 as a timer or an oscillator, let's get microcosmic and think of it as a collection of circuits in an 8 pin DIP package[1].

The 555 contains (among other things) some comparators, a flip-flop, a buffer, and an enable pin. If you're like me, you probably started using the 555 right off the bat and never really worried about the finer points; now it's time to really see what's in the thing (you can find a suitable spec sheet in the IC Timer Cookbook[2]). You might be surprised to find that most of the internal devices are accessible from the pins.

Thinking about ADSRs for a bit, the logic for a true ADSR with retriggering ability has been known for some time[3]; and of course, for any given logic problem there are scores of possible circuit solutions. But note that the standard ADSR logic scheme requires a flip-flop, a comparator, and an enable circuit...and the 555 has all of these!

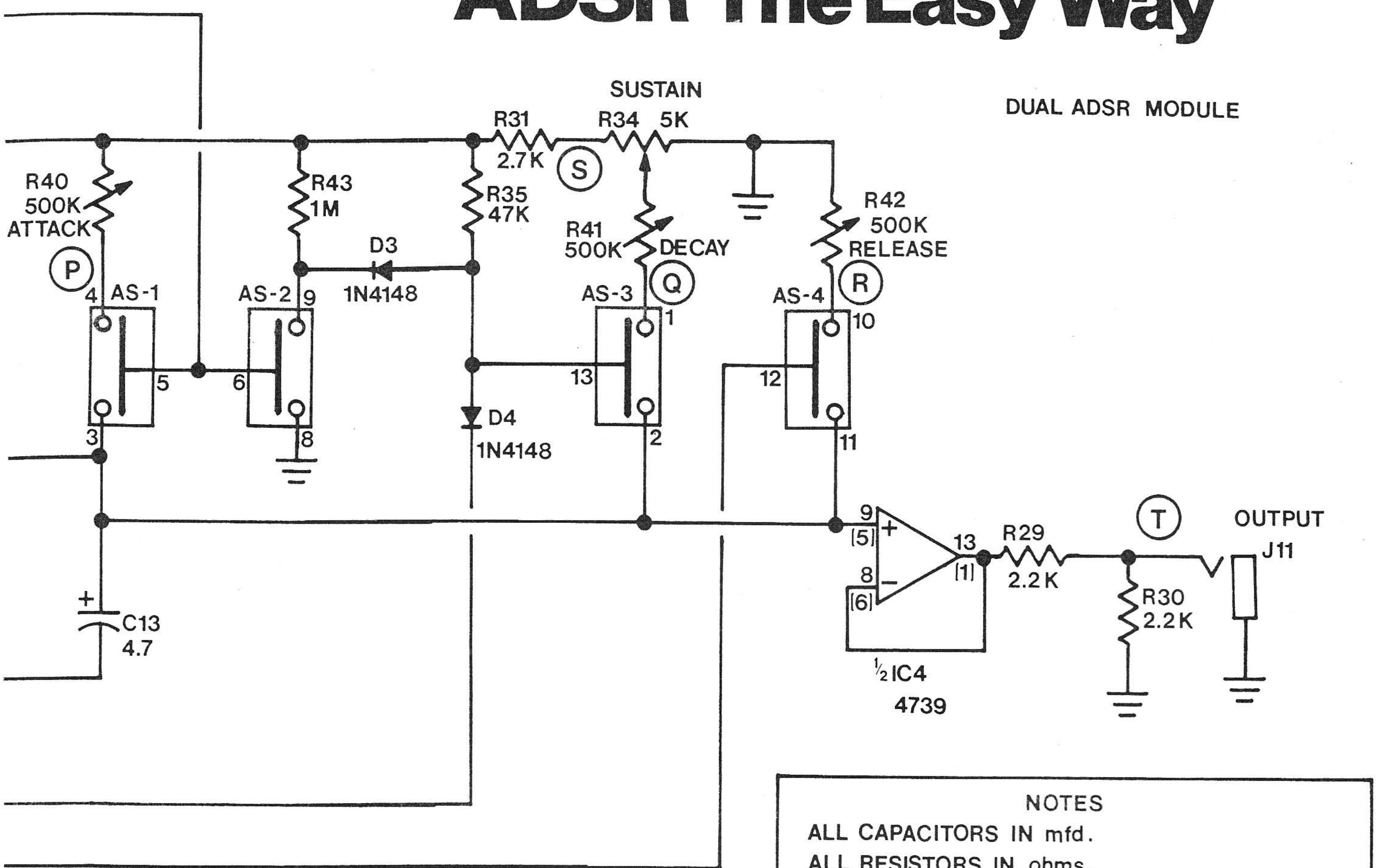
How it works. Referring to figure 1, note that since I built the ADSR as part of a Dual ADSR/-Dual VCA module (the VCA was covered last issue), the circuit designations start up where they left off last time. The resistors start at R29, the capacitors at C11, and ICs at IC3 and so on.

A true ADSR needs both gate and trigger input signals. J9 is the trigger input jack and J10 is the gate input. The usual standard for gate and trigger signals is that 0V represents OFF and +5V represents ON (in addition, the trigger signal should have a pulse width of about 1 ms). However, when running the 555 with a +15V supply, the trigger would have to swing at least 2/3 of the supply voltage, or 10V. We solve this problem with resistors R38 and R39, which form a voltage divider that pulls the quiescent voltage at pin 2 down to 7.5V. When a +5V trigger is applied and differentiated via C11, the combination of the voltages is enough to fire the 555.

The +5V gate signal must also be conditioned. Since the logic of the ADSR requires a NOT-GATE signal as well as a GATE signal, it is easy enough to implement the signal conditioning and inversion all at once with transistors. Q1 inverts the gate signal, giving a NOT-GATE output with a 0V to +15V swing. This output couples to Q2 which inverts it again, yielding a GATE signal with a 0 to +15V swing.

Having taken care of the input signal conditioning, let's now analyze the rest of the signal in earnest. To fire the ADSR both gate and trigger signals must arrive at their respective input jacks. The gate signal taken from the collector of Q2 then enables the 555 via pin 4. The trigger signal simultaneously couples into pin 2, which starts the ADSR on its attack cycle. When the 555 is triggered, pin 3 goes high, turning on analog switch AS-1. This allows C13 to charge through R40, thereby generating the attack phase. Pin 6 of the 555 monitors

ADSR The Easy Way



NOTES
ALL CAPACITORS IN mfd.
ALL RESISTORS IN ohms.
AS-1, AS-2, AS-3, AS-4 = ¼ 4016 [IC5]
FOR DUAL UNIT, REPEAT ALL PARTS
EXCEPT FOR IC4 WHICH IS A DUAL OP-AMP
POWER SUPPLY FOR IC4: −V = PIN 7
+V = PIN 14
POWER SUPPLY = ± 15V
+V = ↑

○ [CIRCLE] KEYS THE SCHEMATIC TO CIRCUIT BOARD. FOR DUAL UNIT, LABEL M AS M′, N AS N′, AND SO ON.

the voltage present on C13; when this voltage reaches +10V, the flip-flop in the 555 shuts off, which brings pin 3 low and opens AS-1. This ends the attack portion of the cycle.

D3, D4, and R35 form an AND gate. When the voltages on the cathodes of both diodes go high, pin 13 of AS-2 goes high; otherwise, pin 13 is low. At this point of the ADSR cycle pin 3 of the 555 is low, so that AS-2 (which is configured as an inverter) is on, biasing D3's cathode high. And since the gate signal is still present, Q2's collector is still high, meaning that D4's cathode is high as well. Since both conditions have been met, the AND gate is on, and therefore AS-3 is on. This initiates the decay cycle.

C13 then discharges at a rate determined by R41 (the decay control) to a level set by R34 (the sustain control). The sustain control has a range of 0 to +10V, since R31 drops about 5V. The output will remain at the sustain voltage as long as the gate signal is present at J10.

However, if the gate is removed Q1 will turn off, causing the collector to go high. This turns on AS-4, which then allows C13 to discharge to ground at a rate set by R42, the release control. This completes the ADSR cycling of the circuit.

The charge on C13 must be relatively unloaded, so it's buffered by one half of IC4, configured as a standard voltage follower. Since the voltage on the capacitor achieves a peak of +10V, and since our synthesizer system would like a +5V level, R29 and R30 chop the signal down by a factor of two and at the same time present a 1k output impedance (another of our standards).

We've examined the entire ADSR cycle based on the assumption that the gate and trigger arrived simultaneously. What happens if a

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PRACTICAL CIRCUITRY

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gate and trigger arrive simultaneously, but after the waveform has died down to its sustain voltage another trigger arrives? It's fairly easy to see that the 555 will be triggered again, hence another attack and decay cycle will be initiated. But unlike the first case the attack will start from the sustain level; the decay will occur normally.

One final case to consider is when the gate and trigger arrive simultaneously, but the gate is removed before the attack phase has had a chance to finish. Well, the attack phase will start as usual, however, when the gate signal is removed AS-4 is turned on and then the waveform goes immediately into the release portion.

Applying the ADSR. I hope the above shows that all eventualities are covered, and the ADSR does work as expected. Figure 2 shows a timing diagram that should make this all clear. However, there is one option that I wanted to add to the circuit, and as it turned out this was fairly easy to do. There are times when all you really need is an AD waveform, not the full ADSR (such as percussive waveforms). Normally AD patterns are initiated by a trigger signal only, say from a rhythm generator[4]. (Editor's note: the AMS-100 system, and its synchro-sonic offshoots, are also good examples of trigger-oriented, rather than gate-oriented, systems). No gate is available in these cases. Since a gate is needed by the above circuit to start the attack cycle, some provision must be made to fool the unit into thinking a gate signal is present. This is the purpose of closed circuit jack J10. Since the switching lug connects to +15V, when a plug is not inserted into the gate input an "imitation" gate signal is still there. This enables pin 4 of the 555, thus allowing it to be fired. For true AD patterns, the sustain and release controls should be turned down to a minimum.

For an AR type pattern, apply gate and trigger signals as usual but turn down the decay to minimum and turn up the sustain to maximum.

Since this circuit is so

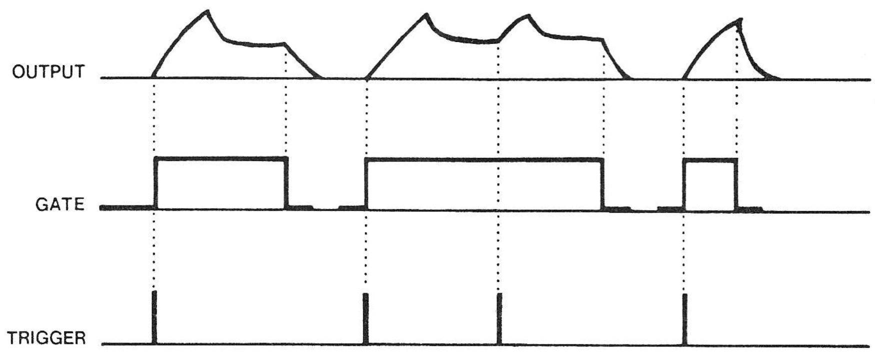


FIG.2 TIMING DIAGRAM / DUAL ADSR

small and inexpensive it is a snap to build a dual unit. All of the parts in the schematic should be repeated for a dual unit, except for IC4 (which is a dual op amp). If you want to build it the way I did, then you should incorporate the dual VCA (from the last installment of Practical Circuitry) on to the same circuit board. The result is a dual VCA/dual ADSR module that fits nicely behind a 19" X 3.5" rack panel. The circuit board mounts behind the panel on small angle brackets.

That's it! We've now implemented VCAs and ADSRs the easy way, and while they work great, they are inexpensive enough so that we can afford to build several of each. I sure hope you will enjoy building these two circuits as much as I have enjoyed designing them. If you come up with some neat modifications, be sure to write me c/o Polyphony. I'd love to hear what you've cooked up!

(1) For a good treatment of this see B. Hutchins, "The 555 as a Collection of Devices", Electronotes, Application Note #32.

(2) W. G. Jung, the IC Timer Cookbook, (Indianapolis: Howard W. Sams), 1978.

(3) D. Rossum, "On Transient Generators", Electronotes, #22, pp. 6-8.

(4) For example, see my circuit, "Rhythm Generation: It's About Time", Polyphony, March/April 1981, pp. 22-25, 32.

#

Specifications

Power	+15V @ 40 mA, -15V @ 20 mA (ADSR + VCA)
Input Modes	Gate and trigger for full ADSR Trigger only for AD mode
Output	0 to +5V 1k output impedance
Attack Time	2 msec to 5 sec
Decay Time	2 msec to 8 sec
Release Time	2 msec to 8 sec
Sustain Level	0 to +5V

Parts List

R29, R30	2.2k
R31	2.7k
R32, R33	10k
R34	5k pot
R35	47k
R36-R39	100k
R40-R42	500k pot
R43	1M
C11	0.001 uF
C12	0.01 uF
C13	4.7 uF, electrolytic
C14	10 uF, electrolytic
Q1, Q2	2N4124 (or any other NPN)
IC3	555 timer
IC4	4739 dual op amp
IC5	4016 CMOS quad switch
J9, J11	Open circuit 1/4" phone jack
J10	Closed circuit 1/4" phone jack

For dual unit, repeat all parts except for IC4, which is a dual op amp.

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.

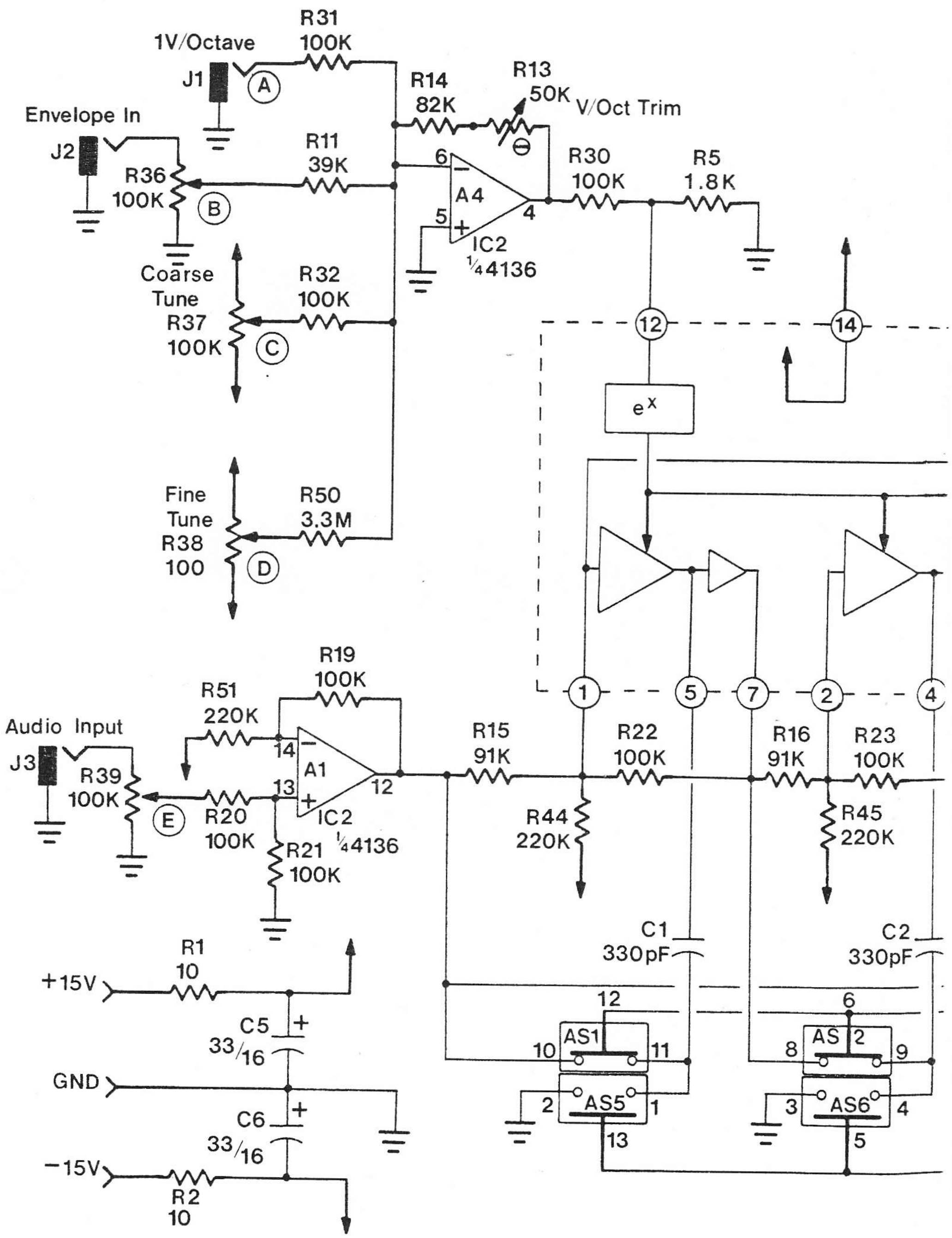
Now, don't get me wrong; I like fancy filters too. But I don't want to play synthesizer in the poor house. The filter I came up with is very clean, but won't cost you an arm and a leg. The good quality at a low price comes from using the CEM-3320 integrated circuit made by Curtis Electromusic. I think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

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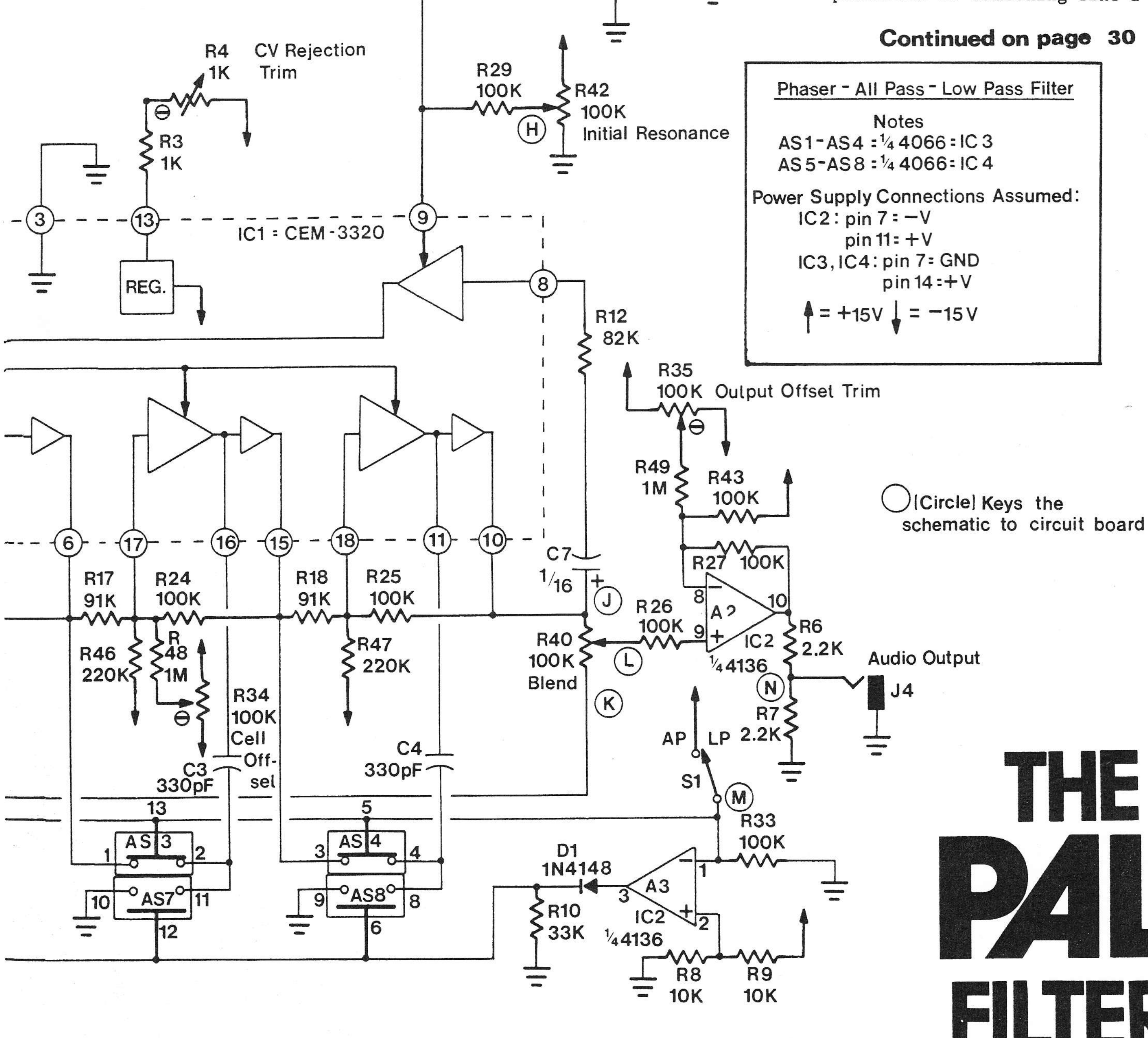

THE PAL FILTER

```
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)
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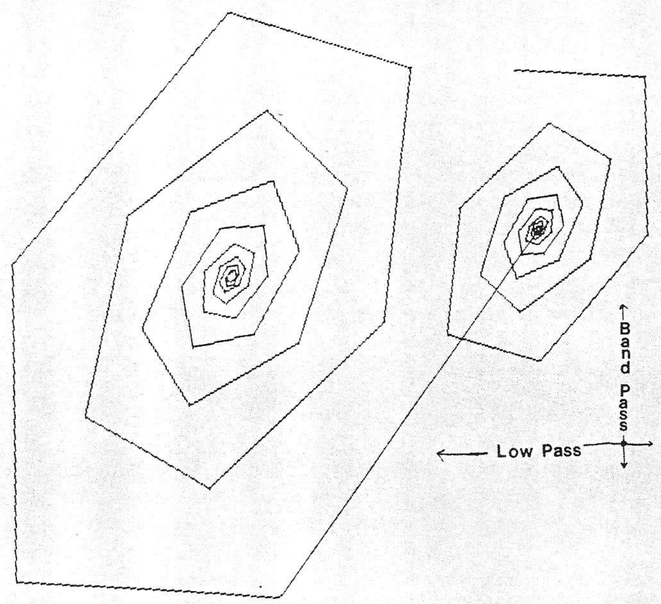


5. SQUARE WAVE, FC=.9995, Q=5, PLOT IS LP VS. BP.

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.

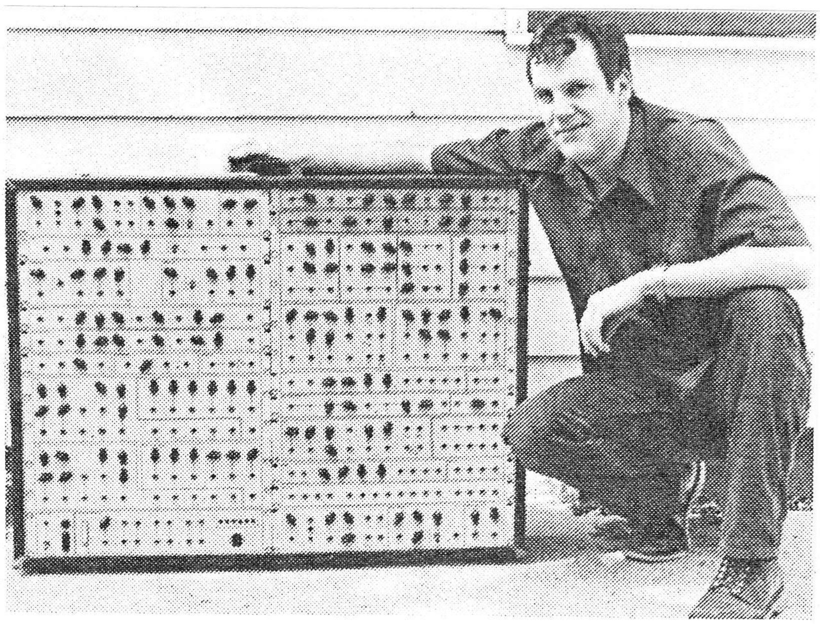
Applications. First, set S1 to the low pass mode and turn blend control R40 to full strength filter output. The filter is now acting like a standard low pass filter.

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! ▪

And now for a page from The Scrapbook...



Here's a picture of me with the first major synthesizer I built, from about 1983 or so. I've always found my best musical inspiration comes from leaning on gear by the side of a house.

PRACTICAL CIRCUITRY

The Snare + Drum Voice

By: Thomas Henry

I used to think synthesizing drum sounds was easy. That was before I tried it! As it turns out, percussive sounds are very difficult to synthesize in a useful way; and when it comes to home recording, you will find that not only are realistic sounds hard to generate, but they are also somewhat touchy to record...things like background noise, tape saturation, and bleed-through become critical issues. I won't be able to answer your recording questions here, but I can help you generate some realistic drum sounds.

I call this circuit the "Snare+" drum voice, because it does much more than just create snare drum sounds - it can generate sounds from tom-toms to space collisions. I'm not going to claim that this circuit duplicates a snare drum, bass drum, or tom-tom exactly; if you want "real" drums, you would be well advised to hire a real drummer with a real drum set. However, this circuit certainly suggests drum voicings more closely than any other percussive voice with which I've played.

Putting things into an historical context, early snare voices usually imposed an amplitude envelope on some white noise. The kindest thing that can be said about this type of circuit is that it indeed sounded more like a snare drum than, say, a bass drum. Later on, designers added in a simultaneous damped sinusoid with the white noise. This simulated the shell resonance common to all drums, and did wonders in improving the overall effect; it almost sounded like a drum. However, the sound was still "splatty". Circuits such as this never quite gave the impression of a drum stick striking a stretched drum skin. And that brings us to the first of the Snare+'s two magic ingredients: the strike tone. In addition to the two sounds mentioned above (the snares and the shell resonance), one must add in a strike tone to gain a convincing drum sound. The first time I saw this in print was in an article by Roger Powell ("Practical Synthesis: Percussion Sounds", Contemporary Keyboard, April 1981, p. 51). After reading the article, I ran into the other room and tried out a patch derived from the article. I was amazed! For the first time in my life I could almost visualize a stick cracking against a drum head!

How is it done? The answer will probably surprise you. A very sharp envelope frequency modulates a pulse wave. The envelope has such a fast attack and decay time that you won't hear a sweeping tone; instead, you'll hear a "thud" if the VCO is tuned low or a "crack" if the VCO is tuned up higher.

As neat as this patch is, the major drawback is that it consumes two VCOs, one noise source, two VCAs, a mixer, two ADSRs and a VCF just to do one quasi-snare sound. That doesn't leave much left over to synthesize the rest of the drum kit! However, the Snare+ has been pre-patched to approximate the arrangement mentioned in Roger Powell's original article. One module does the job of many, which leaves the rest of the synthesizer free.

The second magic ingredient is the shell resonance modulation circuit. When you first strike a drum, the head stretches tightly, raising the resonant frequency. Upon removing the stick, the head goes back to its relaxed state, returning to its normal frequency. The resonance modulation circuit creates this effect by raising the shell resonance frequency slightly when the drum is first triggered, and decaying over time back its normal frequency. With short decay times (the best choice for a "realistic" sound), this frequency modulation happens too fast to sound gimmicky, although with maximum modulation, you can even get those "dooo-doooo-dooooo" electronic drum sounds which were so popular a few years back. In any event, when the strike tone and modulation are properly adjusted, the sound is uncannily like that of an acoustic drum.

How it works. Figure 1 shows a block diagram of the circuit. All of the "modules" marked with asterisks are internal to the SN76477, the main chip used in the circuit. As you can see, this chip just about does it all! First, the noise source creates the sound of the snares. Since this is a digital noise source, we can modulate the clock rate (see "Tricks With the SN76477, Part 2", Polyphony, July/August 1981, pp. 16-17, for more information on this). The sonic effect is very similar to a VCF closing down, so in the diagram this function is labelled "VCF". The "VCF" is modulated by an envelope produced by envelope generator #1. This creates the illusion of snares losing energy. The noise output can be further altered by the manual filter.

The SN76477's LFO, configured as a manually adjustable audio oscillator with optional modulation, creates the shell resonance sound. The output of this oscillator is a triangle wave, which is nothing more than a sine wave with 20% total harmonic distortion.

The impact noise is created by using the SN76477's internal envelope generator (set for a fixed, short attack and decay time) to sweep the VCO, which is also internal. The initial frequency can be adjusted by the "Impact Pitch" control.

Envelope generator #1 and VCA #1 are outboard to the SN76477. This pair has the job of imposing a final envelope on the mix of snares, shell resonance, and impact noise. The master envelope has a fixed attack time, but the decay is adjustable.

Looking at the block diagram, you'll see seven controls. By manipulating these in various ways, countless other sounds are possible. I think you will find this to be one of the most versatile percussive voices around. And unlike many such units, this one, with its impact tone and

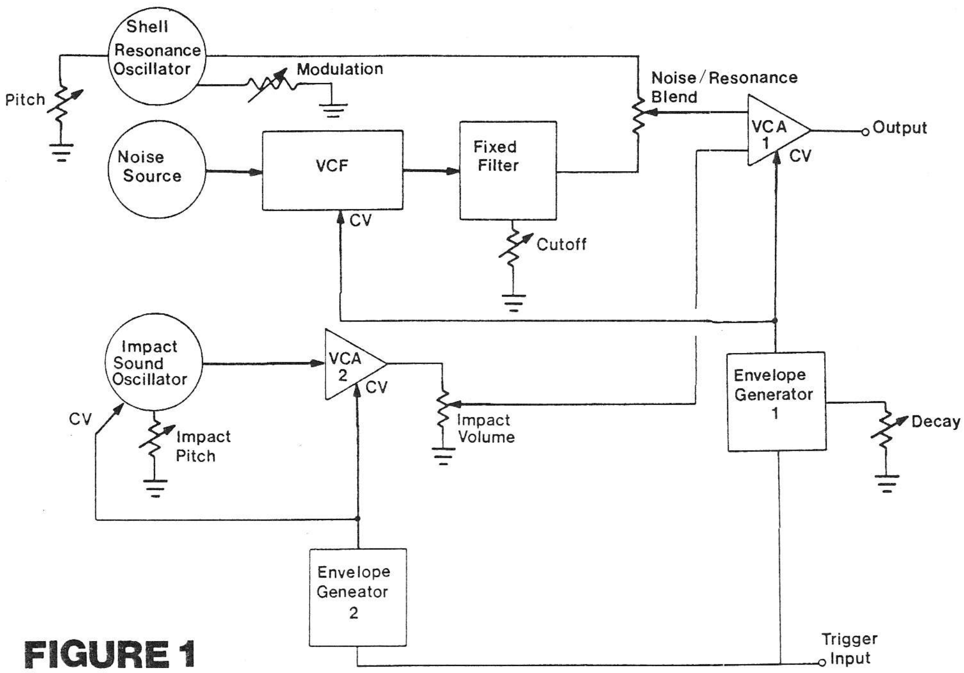


FIGURE 1

shell resonance modulation, really gives the illusion of someone drumming!

Nuts and bolts. Since the basic setup has been described above, we can breeze through the actual circuit operation. Look at figure 2. The heart of the whole circuit is the SN76477, which is powered by a dropping resistor from the +15V supply to pin 14. The SN76477's internal regulator takes this and produces +5V at pin 15 to power the sound generating circuitry. The other chips are powered by a bipolar ±15V supply.

Taking a tour of the complex noise chip, Q1 forms a voltage control input for the noise clock. As mentioned above, this sounds like a VCF closing down when modulated. Q2 forms a similar voltage control input, but this time it is the impact tone VCO which is being modulated. Q4 acts in a similar manner to Q1 and Q2 and modulates the shell resonance frequency. Pin 8 is the output of the internal envelope generator, and this is the voltage which sweeps the VCO. The impact tone output is available at pin 13.

The shell resonance sound taps off C6/C2; a triangle wave appears at the output of A2. Since R33 blends the triangle wave and noise source together, this pot should have a linear taper to insure a smooth transition between the two extremes. The noise/-resonance blend and the impact tone are then summed by IC3, which is the master VCA for the system.

To form the master envelope generator, when the unit is triggered we simply dump some charge on capacitor C9 via diode D2. R5 sets the minimum decay time, while R34 gives a variable decay time. Q3 buffers the voltage present on C9; the voltage output of this buffer taps off the tie point of R7 and R10, which not only modulates Q4 via R41 and Q1 via R40, but is also converted to a control current by Q5. This control current then modulates IC3, the master VCA.

Op amp A1 conditions the input trigger, so that just about anything may be used to trigger the unit. C3 and R29 differentiate the trigger, thus narrowing the pulse width. D1, R6, and R11 chop the signal down to a 0 to +5V range, suitable for firing the SN76477.

Like other circuits shown in this column, the unit features a trigger level of +5V and an audio level of 10V p-p. If this is a bit hefty for your system, add a voltage divider to the output of op amp A4.

Construction details. There is only one critical area in the project. White noise can be diabolical in the way in which it infiltrates other parts of a synthesizer. Be sure to include sufficient power supply decoupling to keep that noise from getting back to other parts of your system (this is provided by R1/R2 and C12/C13 in the circuit). In addition, if you have a long way to go from the circuit board to the front panel, be sure to use some shielded cable on any of the noise source related controls. Other than this, construction of the circuit is non-critical.

Construction can be carried out via any means you prefer. However, due to the relative complexity (caused mainly by a 28 pin chip), printed circuit construction is preferred.

All of the non-electrolytic capacitors should be mylar. For best results, use 5% tolerance resistors.

One quirk that you should be aware of is the unusual action of the impact pitch control, R21. Since the VCO internal to the SN76477 increases in frequency with a decreasing voltage input, the sense of R21 should be reversed; in other words, reverse the hot wire and the ground wire on R21 from the normal way of wiring audio pots. You want it wired so that turning the pot counter-clockwise moves the wiper closer to the hot side.

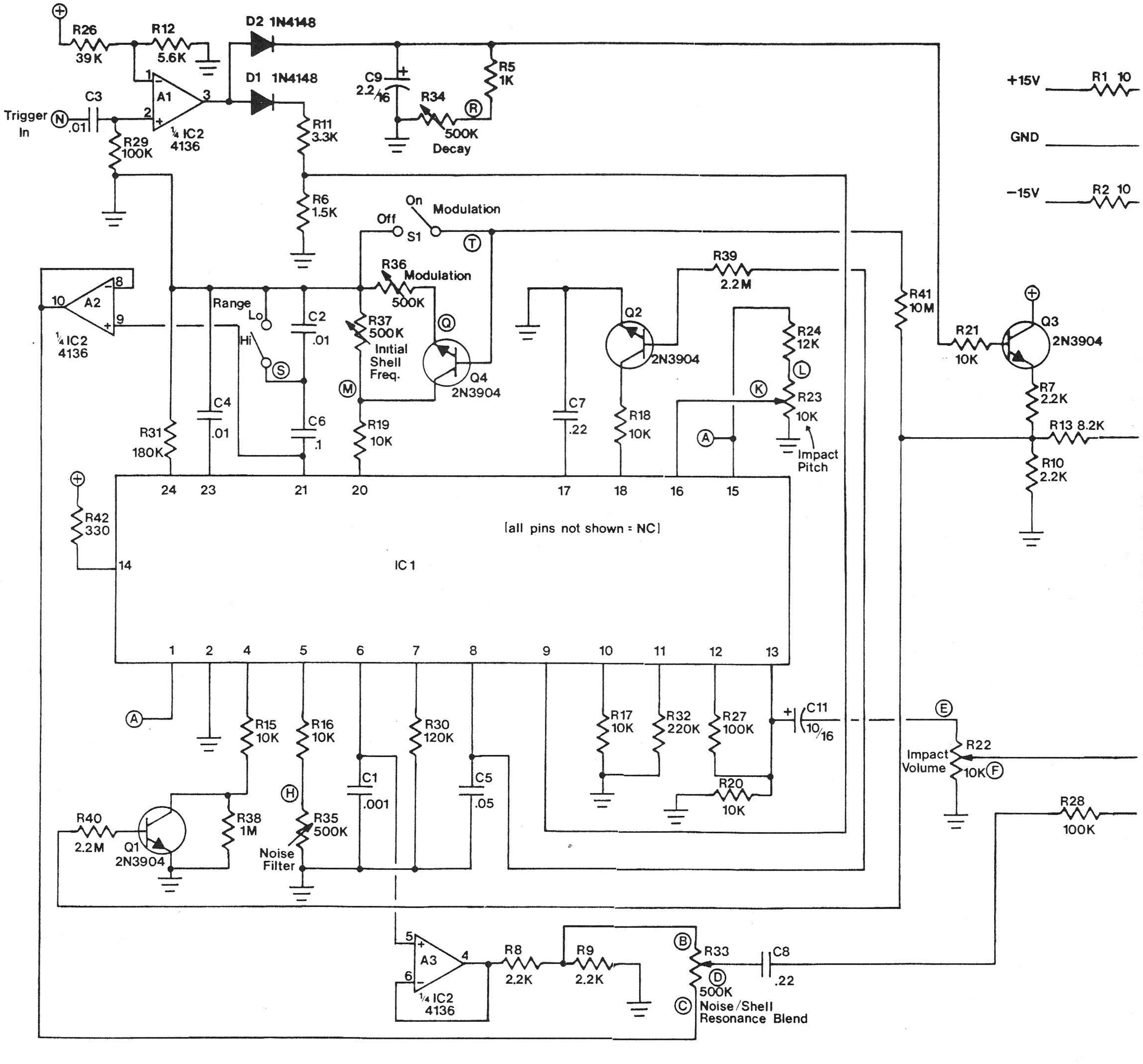
There are several ways to mount the Snare+, but a 1.75" high, 19" wide rack panel is probably your best option since all controls and switches fit easily in the available space (see Polymart for a source of these panels).

Using the Snare+. Of course there are zillions of sounds this thing can make, but to get you started here's how to create a quasi-snare sound. Patch in an LFO to the trigger input to fire the unit continuously (not too fast) for experimentation purposes. Hook up the output to a good amplifier and speaker. Now let's concentrate on the shell resonance. Start off by closing S1 and S2, then make sure that R22 and R36 are turned all the way down and that R33, the noise/resonance blend control, is set for resonance sound only. Next fiddle with R37, the resonance pitch control, until you hear a tone which approximates a snare drum with its snares turned off. Now add some more modulation by turning up R36. This may raise the perceived pitch of the drum, so might want to readjust R37 downwards a little bit. (Incidentally, note that the modulation has less effect as you raise the shell resonance pitch via R37.)

At this point, ease up on R22 to add in some impact sound. Adjust R23 to create an impact sound which sounds good with the chosen resonance pitch. The low end gives a dull "thud" while the high end gives a bright "crack"...somewhere in the middle is probably about right.

Finally, turn up R33 to add in some snare sound. Not too much; it's easy to overdo this control! Finally, adjust R34 to create the most realistic decay. At this point you should be hearing some sort of snare drum sound. Now go back and fine tune everything until it's "just right".

It's also important to note what some equalization can do for the realism of this sound. The preferred tool would be a parametric equalizer (see related story in this issue), although other bandpass-type equalizers (state variable filters et al) will also do the job. Generally, boosting in the 1 kHz region gives a "fat" sound, while a boost around 2 to 3 kHz gives a real "sharp" sound which, incidentally, can also sound like handclaps with a little delay and short decay time.



Of course, only an acoustic drum can sound exactly like an acoustic drum. Nonetheless, with proper adjustment (some of these controls work in a fairly subtle way) at some point, everything will fall into place and if you close your eyes, you'll swear you're hearing a tight, well-tuned snare drum.

As you experiment, try for a bass drum sound (similar to the above, but no noise source and deeper in tone). You can get a good tom sound by adding a bit more modulation, and increasing decay time for lower-tuned drums and decreasing decay time for higher-tuned drums. By flicking S1 to the "hi" position, you can get a variety of bell/woodblock sounds. Then try some high hat sounds (white noise only, and vary R34 manually), some outer space drum sounds, and then try for...the sky's the limit!

DON'T MISS POLYTEST Page 23 FOR EDITORIAL COMMENTS ON THIS PROJECT

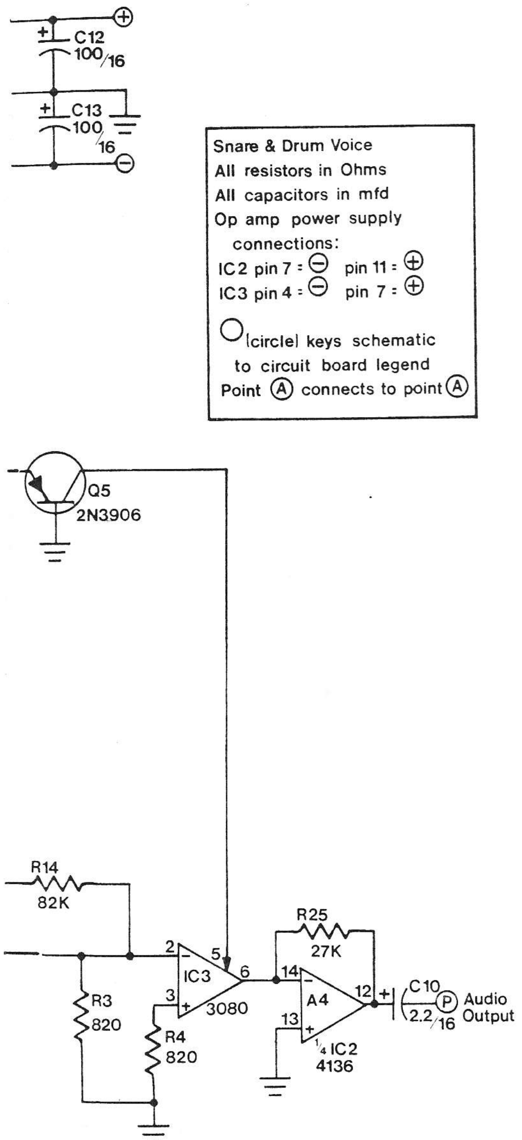


FIGURE 2

PARTS LIST

Resistors (1/4 Watt, 10% except as noted)

R1, R2	10 Ohms
R3, R4	820 Ohms
R5	1k
R6	1.5k (1k5)
R7-R10	2.2k (2k2)
R11	3.3k (3k3)
R12	5.6k (5k6)
R13	8.2k (8k2)
R15-R21	10k
R22	10k pot (impact volume)
R23	10k pot (impact pitch)
R24	12k
R25	27k
R26	39k
R14	82k
R27-R29	100k
R30	120k
R31	180k
R32	220k
R33	500k pot (noise/shell mix)
R34	500k pot (decay)
R35	500k pot (noise filter)
R36	500k pot (shell tone mod)
R37	500k pot (initial shell res)
R38	1M
R39, R40	2.2M
R41	10M
R42	330 Ohms

Capacitors (15 WVDC or greater)

C1	.001 uF (1 nF)
C2-C4	.01 uF (10 nF)
C5	.05 uF (50 nF)
C6	.1 uF (100 nF)
C7, C8	.22 uF (220 nF)
C9, C10	2.2 uF (2u2)
C11	10 uF
C12, C13	100 uF

Note: C1 is polystyrene or mylar, C2-C8 are mylar, C9-C13 are electrolytic.

Semiconductors

IC1	SN76477 noise chip
IC2	4136 quad op amp
IC3	3080 amp
Q1-Q4	2N3904 (NPN)
Q5	2N3906 (PNP)
D1-D2	1N914 or equivalent

Miscellaneous

S1, S2	SPST switches
J1, J2	Input and output jacks
Misc.	Panel, knobs, wire, solder, etc.

PARTS AVAILABILITY

A kit of parts for the "Snare+ Drum Voice" including all parts listed above (except for "misc."), and an etched/drilled circuit board, is available from PGS Electronics for **$29.95**; orders received from Polyphony readers before 12/15/82 will be shipped postpaid. Order by phone (VISA, Mastercard, or COD; COD add $1.50), or by mail. Please - all inquiries should be by mail.

PGS Electronics, PO Box 749-C, Terre Haute, IN 47808 (order desk tel. 812-894-2839).

PRACTICAL CIRCUITRY

PATCH-OVER SCHEME FOR A SMALL SYNTHESIZER SYSTEM

By: Thomas Henry

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

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

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

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

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

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

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

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

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

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

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

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

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

Without any patch cords inserted, the mixer allows you to

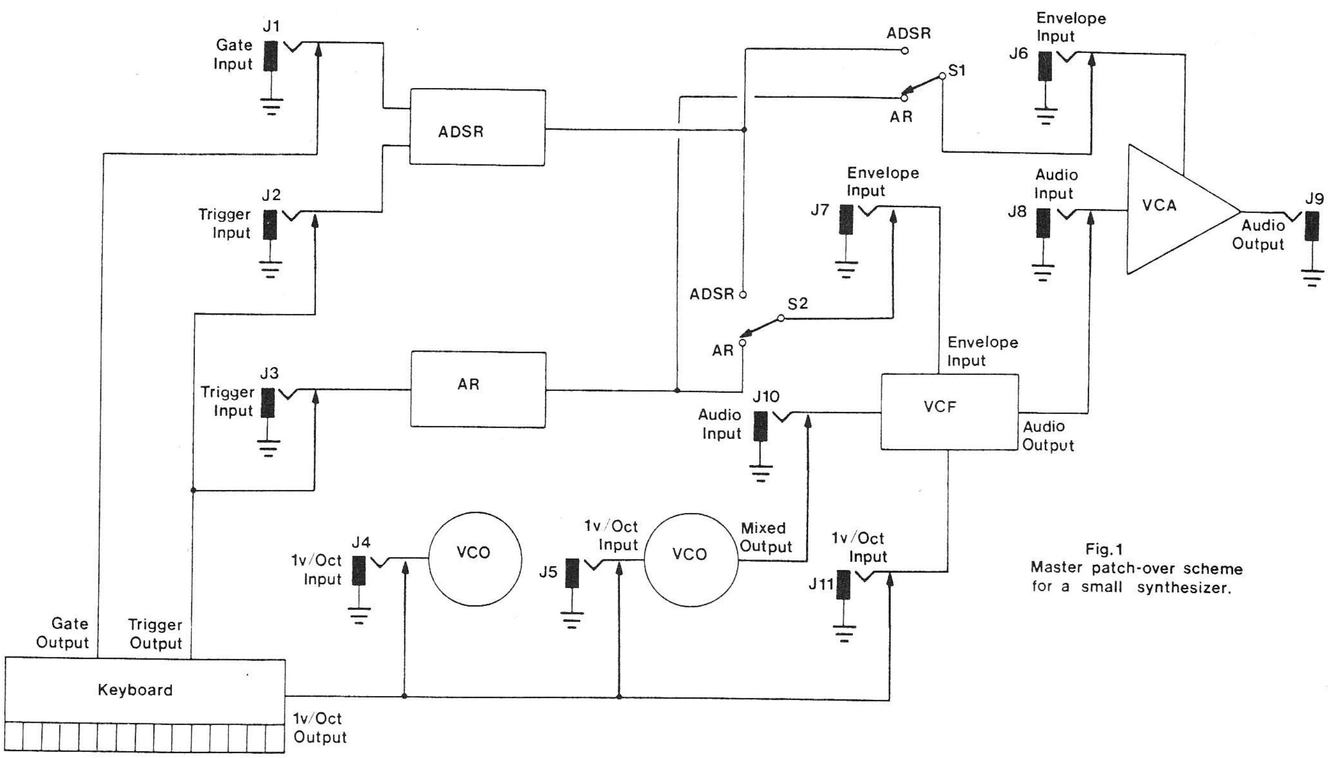


Fig. 1
Master patch-over scheme for a small synthesizer.

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

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

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

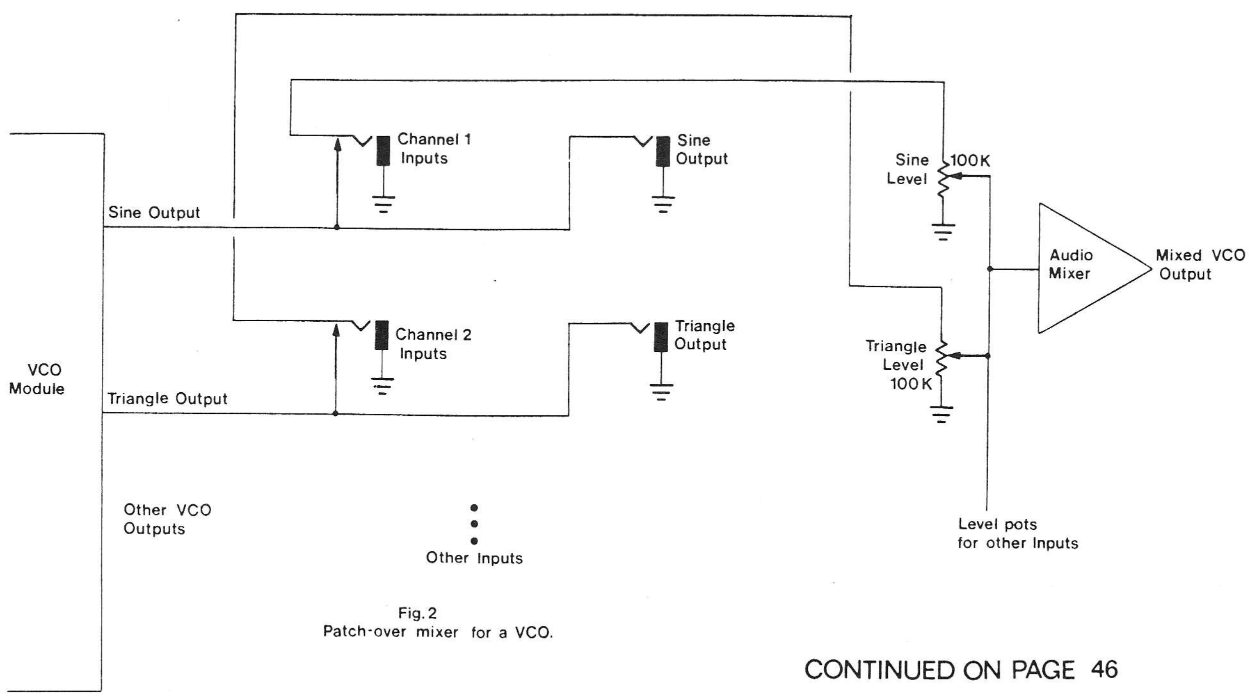


Fig. 2
Patch-over mixer for a VCO.

CONTINUED ON PAGE 46

a way to generate monophonic, noncyclic illusions by always using all 16 output functions to control 16 corresponding processing elements. But that's where we're getting of the track: You don't have to use all the outputs all the time, and the results don't have to be monophonic.

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

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

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

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

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

No new effects indeed!

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

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

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

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

PRACTICAL CIRCUITRY

CONTINUED FROM PAGE 27

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

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

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

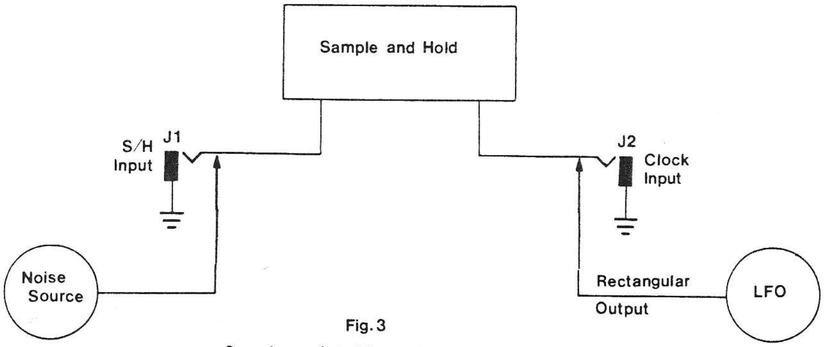


Fig. 3
Sample and hold patch-over scheme.

Practical Circuitry

My Favorite Transistor Circuits

By: Thomas Henry

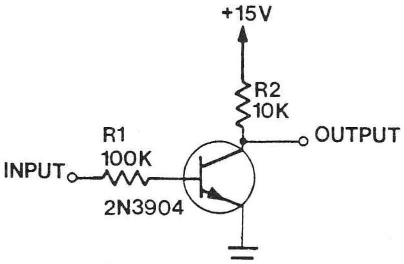


Fig.1

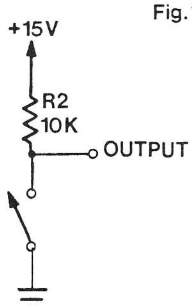


Fig.2

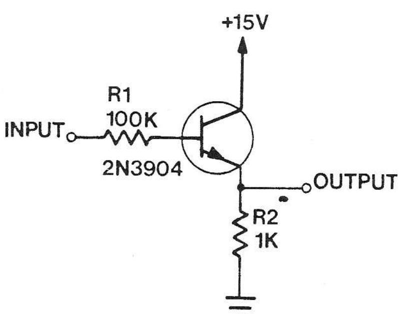


Fig.3

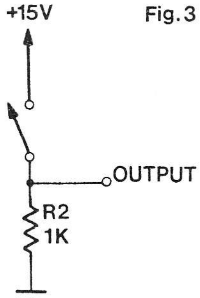


Fig.4

It used to be that there were lots of oldtimers who could do the most marvelous things with vacuum tubes, but were totally out to sea when it came to transistors. This was simply a case of not keeping up with the advances in technology. But how about the reverse? Lots of people today seem to be able to design all sorts of neat things with solid state parts, but are lost when it comes to vacuum tubes. And finally, let's carry this to the extreme. What about the person who grew up with integrated circuits but can't design with either tubes or transistors? I'm that way! I learned electronics by building things with ICs and somehow seemed to miss all of the basics of semiconductor theory.

I suspect that there are lots of people who are in the same boat, especially among the younger Polyphony readers. A couple of years ago I decided to do something about this gap in my knowledge, so I bought a number of self-study books on the subject. At the end of the article I mention some of these books if you want to do likewise. But for right now let's look at four of my favorite transistor circuits that I have found along the way.

I should probably mention that in your own design work, most of the time you will want to use integrated circuits. They are usually very easy to use, make circuit board design simple, and will save you money. However, there are times when a transistor circuit will fill the bill more closely. Cases of this might be where you have only a single polarity supply (most linear ICs require a bipolar supply), need a heftier drive current than the IC can supply, or need to use some unusual characteristic of the transistor (perhaps for exponential or logarithmic conversion). With this in mind, let's look at four circuits where transistors are ideally suited.

Figure 1 shows about the simplest circuit you can design with a transistor: an inverting switch. It's hard to beat this for simplicity and versatility. Figure 2 shows that when the switch is open the output is at +15V, and no current flow through the switch contacts. However, close the switch and current flows through the switch to ground. The output is thus at ground as well.

Looking back at figure 1, with no voltage at the input of R1 the transistor is off and no current flows. However, apply a voltage to R1 and current will flow through the base-emitter junction. By transistor action this will cause current to flow from the power supply through the collector straight through to the emitter and ground. The output is pulled to ground. Thus you can see that the switch is inverting.

You may wonder how such a simple circuit can be used. Well, one of the most important uses is level conversion. Suppose the input voltage is TTL level (0 to +5V) and you want to interface to another circuit using a +15V power supply. This circuit will do the trick, since for a transistor with a fairly high DC current gain (say of 100 or so), the +5V signal will be enough to turn the switch on. Hence, a +5V swing can trigger a +15V swing.

Another typical use is as a simple inverter. Suppose in your design you've used up all of the logic gates you had available, but you need one more inverter. You don't want to open up a new hex inverter package and waste the five unused units. This is a perfect time to use a transistor! And by the way, you can use just about any NPN transistor. I like to keep a big stock of 2N3904s on hand for this sort of application.

Figure 3 shows a non-inverting switch and figure 4 shows its mechanical equivalent. Consider figure 4 first. When the switch is open, no current flows through the switch and therefore the output is at ground. However, close the switch and current flows through the contacts. The output then swings to full supply.

By analogy, in figure 3 if no voltage is applied to R1 then the transistor is not conducting and the output is at ground. However, apply a voltage to R1 and current flowing through the base circuit will cause collector current to flow. A voltage drop will be developed across R2 and this is the output. As you can see, this is a non-inverting switch. A positive voltage on the input causes a voltage to appear at the output.

What's the use of such a circuit? Once again, level conversion comes to mind. But perhaps more importantly is the fact that the circuit is really a buffer. The input impedance of the circuit, as shown, is about 100k so that this shouldn't load down the driving circuit appreciably. However, the output impedance is about 1k which can drive another, say, 100k source, with only 1% loading. So you can see that the output drive current is quite good. And this is the purpose of a buffer!

Figure 5 shows how we can take two of the inverting switches described above and cross-connect them to form an R-S flip-flop or latch. Can you see how Q1, R1, and R2 form one of the switches we talked about above? And likewise, Q2, R3, and R4 form another switch. The output of Q1 couples to the input of Q2 and the output of Q2 couples to the input of Q1. (Symmetry like this always gives me goosebumps!) Let's analyze the operation.

Suppose that output Q is low. This implies that transistor Q1 is off and therefore Q bar ($\bar{Q}$) is high. Q bar is in turn coupled to Q2, turning it on. Therefore Q is low, confirming our original assumption. Now apply a brief logic one to the set input. This turns transistor Q1 on and pulls Q bar to ground. This in turn switches Q2 off and causes the Q output to go high. Hence the outputs have switched states. Note also that they will remain in this condition until a pulse is applied to the reset input. You can prove this to yourself by going through an analysis similar to the one given above for the set input.

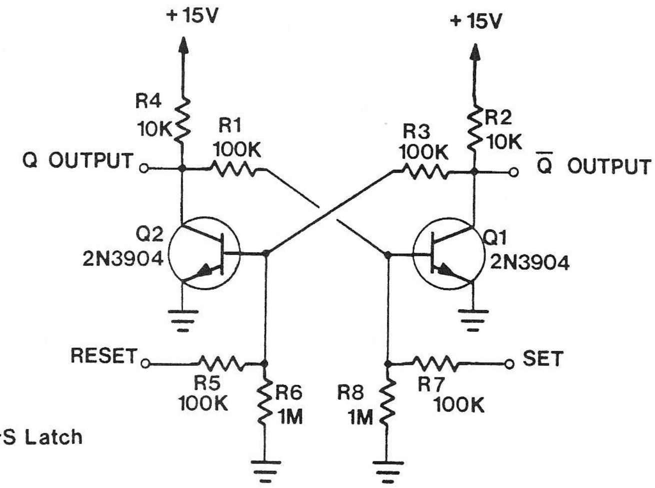
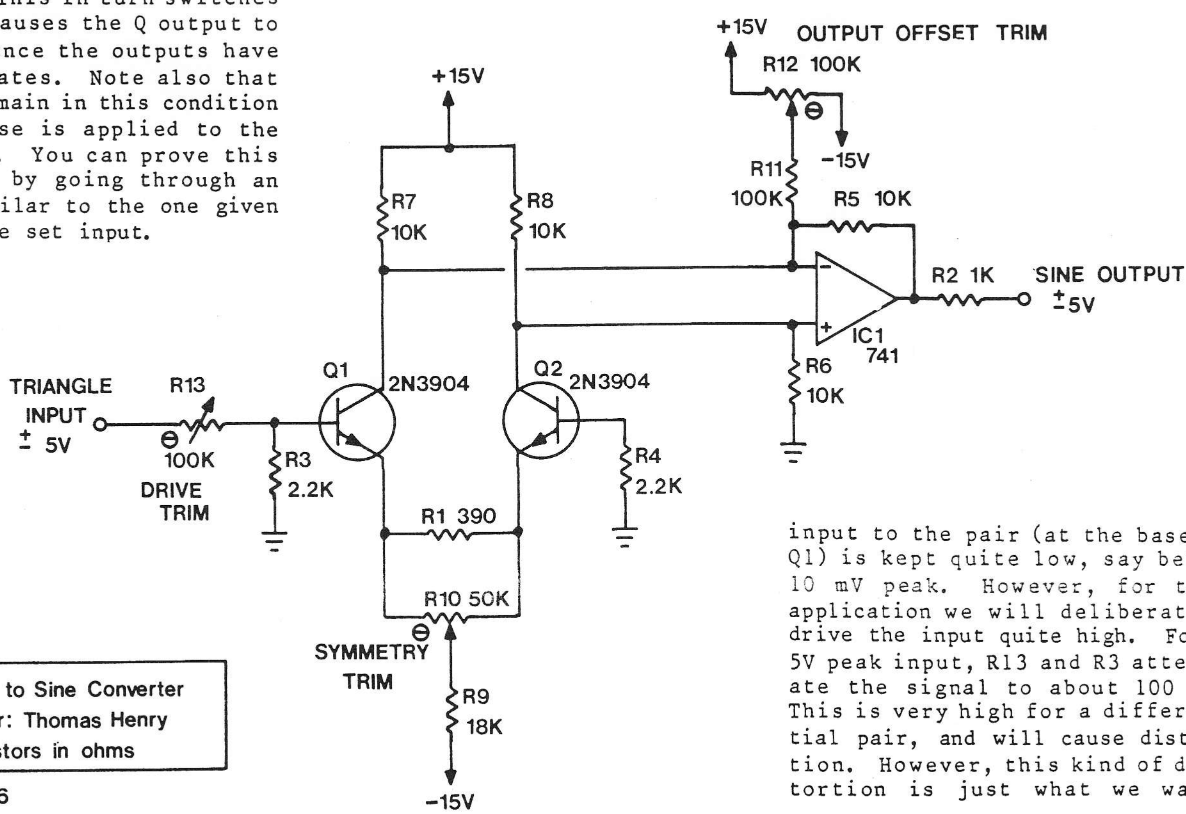


Fig.5 : R-S Latch

Figure 5 also shows the truth table for this circuit and you can see that this is the standard table for an R-S flip-flop. Most of the time you will want to use standard cross-coupled NOR gates for this circuit. But as mentioned above, occasionally you just don't want to crack open a new set of gates to only use a couple of them. Or perhaps the power supply is +24V. CMOS or TTL won't work here, but it's easy to find transistors that will work at this supply voltage. Conclusion: sometimes it pays to think discrete!

Let's leave the digital world and move on to the analog. Figure 6 shows one of my all time favorite transistor circuits: the triangle-to-sine converter. Readers of Practical Circuitry will remember this from the VCO Deluxe circuit presented some months back. I'm going to get technical for a moment so you can skip ahead to the next paragraph if you want; but if you've had a smattering of calculus, the theory of operation is quite simple. Q1 and Q2 form a differential pair. Normally for approximate linear operation the input to the pair (at the base of Q1) is kept quite low, say below 10 mV peak. However, for this application we will deliberately drive the input quite high. For a 5V peak input, R13 and R3 attenuate the signal to about 100 mV. This is very high for a differential pair, and will cause distortion. However, this kind of distortion is just what we want!



Triangle to Sine Converter
Designer: Thomas Henry
All resistors in ohms

Fig. 6

Practical Circuitry.....

Consider the circuit to be a transformation device. For a triangle wave input (over a certain range) it can be demonstrated that the output will be a modified hyperbolic tangent (abbreviated tanh). You study these sorts of functions in the calculus, but what's important to us here is that the hyperbolic tangent is a very close approximation to our ordinary sine wave (over specific ranges). This is all more technical than I want to get here, but if you want to see the full mathematical derivation of this, refer to the excellent article by Bernie Hutchins ("Mathematical Analysis of Differential Amplifier: Triangle-to-Sine Converters", Electronotes, volume 9, number 82, October, 1977, pp. 5-17).

If you skipped the last paragraph let me summarize what was said there in one sentence: the input to the device is a triangle wave and the output is a curve which resembles the ordinary sine wave very closely. Let's look at some of the circuit details.

The input should be a triangle wave with a 10V p-p amplitude. R13 adjusts the drive voltage. R10 soaks up some of the inaccuracies caused by mismatching of the transistors. The output is a differential type and so must be converted back to a single-ended operation for standard audio work. The differential output is taken off of the two collectors and is sent to op amp IC1. The op amp converts the signal to a single-ended output with an amplitude of 10V p-p. R12 adjusts the offset so that the signal is symmetric with respect to ground.

You can trim this circuit up quite easily with an oscilloscope, but don't neglect your ears! The human ear is an excellent distortion measuring instrument, so be sure to listen to the signal while you tweak the circuit, even if you are using an oscilloscope. First adjust R13 while watching the waveform for smooth transitions at the peaks. Next trim R10 for basic symmetry. You want the positive excursions to look essentially like the negative excursions. Finally, tweak up R12 for zero center. That's all there is to it!

Writing this article was a lot of fun for me, and I hope you enjoyed this trip into the realm of transistors too. More importantly, if you're like me and feel uneasy working with them, I hope this will inspire you to learn more. When I decided to correct my "deficient" background in transistors I dug up two books and conducted a self-study course of my own. These are Electronic Circuits by D. Casasent (New York: Quantum Publishers, 1973) and Electronics: A Self-Teaching Guide by H. Kybett (New York: John Wiley and Sons, 1979). I have never regretted looking into these books; they're both recommended, but Kybett is especially good, since it is written for transistor tyros who know a little bit about electronics. And it's self-paced as well. The other book is an excellent "one-stop" reference for anything transistor but is more technical. Do yourself a favor, read a good book this month!

NEXT ISSUE:
A one chip ADSR
Poly-61 review
& more........

Practical Circuitry

A One Chip ADSR

By: Tom Henry

The technology of electronic music is growing so quickly that it is often difficult to keep abreast of all the developments. For example, I thought I was pretty much up to date on most of the important LSI (large scale integration) chips available for music applications, and yet just recently I discovered the SSM2056 ADSR chip. I was amazed when I found out that this IC incorporates all of the design features that my system employs. The supply voltages and input and output levels are 100% compatible with the circuits discussed thus far in this column. And best of all, the complete ADSR circuit can be implemented with one chip! Where have I been all this time?

Before describing the complete circuit, let's stop to philosophize a bit. We've already discussed one ADSR in "Practical Circuitry", so why do we need another? Well, as you continue to gain experience in electronic music, you will find that no two modules are exactly alike. Even two "identical" circuits will have subtle differences that make one more suitable for a particular application than the other. This is most apparent in the case of filters (everybody seems to have a favorite filter), and to a lesser extent with ADSRs. As I found out after building it, this ADSR has quite a different "feel" from my other units. In particular, it seems to offer quite a bit more control over short attack and decay times, thus making it eminently useful for percussion sounds. In general, this ADSR gives me a good feeling; it was easy to build, worked right off the bat, and gives some new effects too.

How it works. Refer to the schematic. The heart of the whole circuit is, of course, the SSM2056. This chip was designed to operate from our standard bipolar ±15V supply, with pin 14 being at +15V and pin 8 at -15V. Since an ADSR has quite a bit of switching going on inside it, fairly hefty capacitors (C3 and C4) are strung across the power pins. While 100 uF may be more than is actually needed, it never hurts to be extra careful when it comes to decoupling (you might also want to solder some 0.1 uF ceramic caps across the supply lines in parallel with these electrolytics for even more effective bypassing of sharp transients-- Ed.).

J1 sends a trigger signal to the chip through R13 and C1. The input signal should be our standard +5V, 1 millisecond wide pulse. The gate signal couples to the IC through closed circuit jack J3. This jack provides a constant +5V gate signal to the chip if no plug is inserted into the jack. (Voltage divider R14 and R12 drop +5V from the power supply.) With the addition of this switching jack, it is possible to fire the ADSR with a trigger only, thus giving AD type envelopes. For most purposes we will want a full ADSR response, but sometimes an AD envelope is more suited to the application at hand. This is especially true for percussion or AMS-100 effects.

If a plug is inserted into jack J3, the constant gate feature is disabled. The plug now presents the normal ON/OFF type keyboard gate to the chip. A little later, we'll see how to make the most of the various types of envelope patterns available with this circuit.

The time parameters are set via potentiometers R7 through R9. Thus the ATTACK, DECAY, and RELEASE times may be easily set with the turn of a knob. Since the hot sides of these pots are hooked up to the +15V power supply line, the wiper voltages must be attenuated. R4 and R1 form a typical voltage divider. The maximum voltage of +15V is dropped to about 260 mV, a level the SSM2056 likes to see.

The SUSTAIN control is handled a little differently. A fixed 20k resistor (R11) is added to potentiometer R10 to drop the +15V supply line to about +5V. Thus the hottest setting of this pot is at +5V. This in turn means that the SUSTAIN voltage is variable from 0V to +5V, as we would expect it to be. By the way, 20k is a standard 5% value and can be easily obtained from a number of dealers (see below). Do not substitute, say, a 22k resistor for this critical value. (Note that potentiometer resistances are not all that accurate, so you may want to use a meter to make sure that the pot you choose is as close to 10k as possible -- Ed.)

C2 is the timing capacitor for the whole ADSR. Use a good quality capacitor here; mylar is perhaps the best choice, being both fairly stable and not too expensive.

The output appears at J2. I've only shown one jack here, but in my version of the circuit, I actually tied four jacks in parallel for the output structure. You'll probably find, like I have, that you'll often use one ADSR to drive several circuits. So, by making a number of output jacks available, you will get around using up some multiples elsewhere in your system. Four jacks should be the limit though, since this is about the maximum that the SSM2056's internal buffer can handle.

How to build it. In the past, obtaining single unit quantities of the more exotic integrated circuits was rather difficult. However, things are easing up now, and you should have no trouble at all. PGS Electronics (PO Box 749C, Terre Haute, IN

47808) is one source; the price is under $6, although there may be a shipping and handling charge.

The two rather important resistors, R11 and R14 (20k and 200k, respectively) are standard 5% values and can be obtained from a number of places. My favorite source for resistors is Jameco Electronics (1355 Shoreway Road, Belmont, CA 94002). The price is around six cents each for resistors, but you'll have to make sure you meet the minimum order requirements. Write to both PGS and Jameco for catalogs and ordering information.

Since this was such a simple circuit, I built it on an "Experimenter Printed Circuit Board" available from Radio Shack (stock #276-170; about $3). This is one of those generalized breadboard rigs that has a number of rows of pads and traces suitable for building up IC type circuits. Circuit construction is not critical, since there are no high frequencies present in the circuit. Along with the circuit board mentioned above, I used hookup wire and flea clips to finish the construction.

The one chip ADSR mounts easily behind a standard 1.75" by 19" rack panel. Use some small angles and #4 hardware to fasten the circuit board to the front panel. Since you'll probably have some space left over, you can use this for some one by four multiples; in other words, four phone jacks wired in parallel. Multiples are always handy to have around, so when a chance presents itself like this, seize the moment and throw one in! By the way, if you need some help in preparing a front panel, see my article "Making Rack Panels" (Electronotes, Volume 13, Number 122, February 1981, pp. 5-9).

Using the one chip ADSR. Since ADSRs may be new to some readers, here are a few settings that you can play around with. For a full ADSR response, apply both a gate and trigger from the keyboard. Now press a key and hold it. The instant you hit the key, the envelope will launch into its ATTACK portion. When the signal reaches +5V, the DECAY portion will start up. The signal will decay to the level set by the SUSTAIN control and will hold there for as long as the key is held down. Now release the key, and the RELEASE portion kicks in.

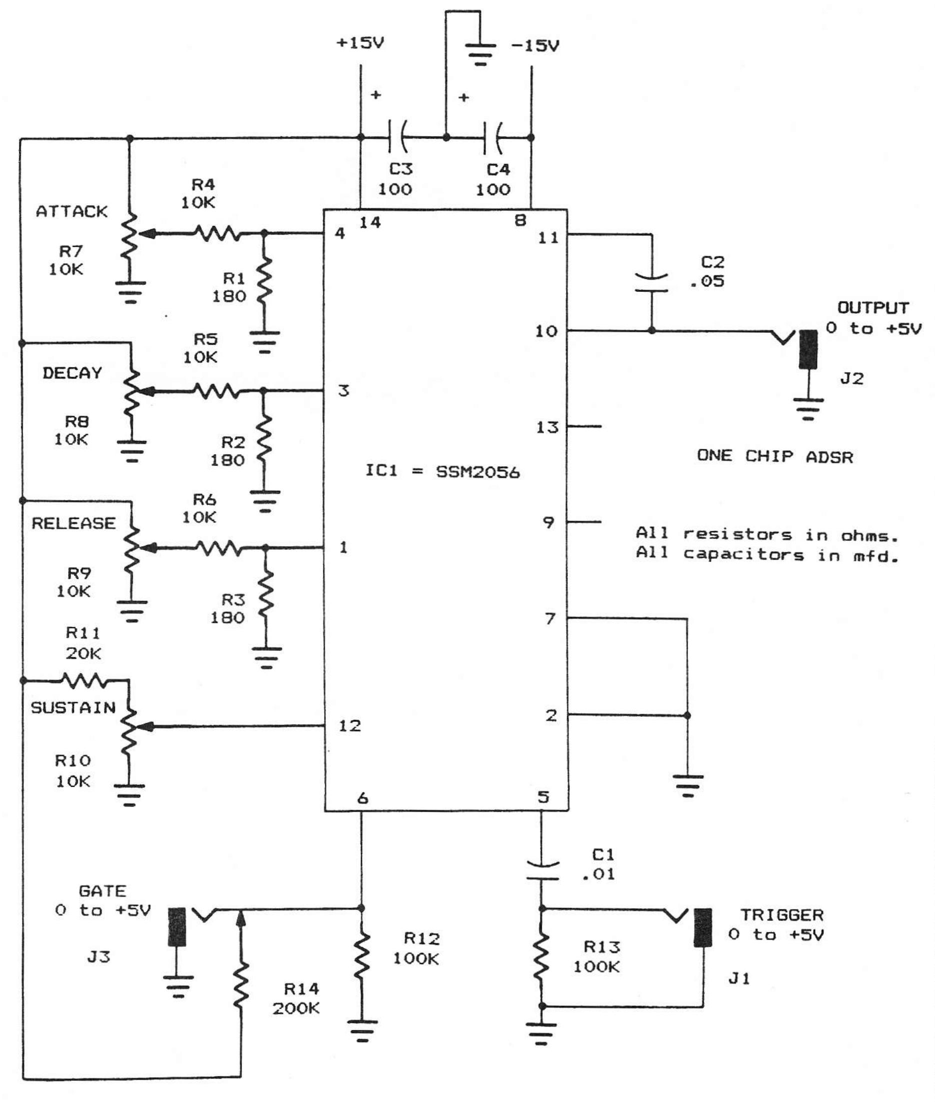
For an AR type response, once again apply both a gate and trigger. Now turn the DECAY control down all the way, and the SUSTAIN control up all the way. When a key is pressed, you will get an ATTACK/SUSTAIN/RELEASE pattern, typical of an AR unit.

For AD effects, you need only apply a trigger signal. Turn the SUSTAIN and RELEASE controls down all the way. Now whenever you touch a key, the unit will go into an ATTACK/DECAY cycle automatically. As mentioned earlier, this is usually the appropriate waveform for percussive effects.

Of course, these three arrangements just described are just the start. There are countless other settings of the controls, hence countless other sounds. Let your ear be the judge of which are most appealing to you! And remember, this circuit allows for full retriggering, so the versatility is even greater.

There you have it, a very complete one chip ADSR! Alert readers will note a similarity to Craig Anderton's "Voltage Controlled Envelope Generator" (Contemporary Keyboard, May 1982, pp. 20-23). Craig's circuit used the SSM2055, an earlier generation of the SSM2056 presented herein, and also offers voltage control of the various parameters. If you're looking for even greater versatility (at the expense of greater complexity, though), check out Craig's circuit. But for most common applications, I think you'll find the simple manually controlled version presented here will really fill the bill.

If you're just learning how to build synthesizer gear, you'll find this ADSR is a great project with which to start. Pay attention to the supply voltages, capacitor polarities, watch your soldering techniques, and before you



know it you'll have a very professional quality ADSR up and running.

PARTS LIST

Resistors	
R1-R3	180 Ohms
R4-R6	10k
R7-R10	10k linear pot
R11	20k
R12, R13	100k
R14	200k
Capacitors	
C1	0.01 uF
C2	0.05 uF
C3-C4	100 uF electrolytic
Other Parts	
IC1	SSM2056 ADSR chip
J1, J2	Open circuit 1/4" phone jack
J3	Closed circuit 1/4" phone jack
Misc.	Circuit board, front panel, knobs, wire, etc.

Letters

continued from page 5

George's second question concerned modifying the Casio 202, which was reviewed in your Jan/Feb '82 issue. I was rather surprised you hadn't looked into a modification for this, as I'm sure quite a few of your readers own some sort of Casio that is not of the "mini" type. Casios are quite often owned by those of us who make less than $12,000 a year. All I can say is if Polyphony can't help, then try J. L. Cooper Electronics, as they are poly-synth modification experts. The address is:

J. L. Cooper Electronics
2800 S. Washington Blvd.
Marina Del Rey, CA 90291

In closing, I'd like to say that it would be of enormous help to amateurs like myself if you would write electronic projects with digital-unfamiliar people in mind. I can't see this as being an annoyance with people who are familiar with electronics...give it some thought.

Kenneth Amaris
Culver City, CA

Kenneth -- You're not the only one who has written recently

continued on page 40

BOOK REVIEW

The Complete Synthesizer, a Comprehensive Guide by David Crombie, Omnibus Press, 1982.

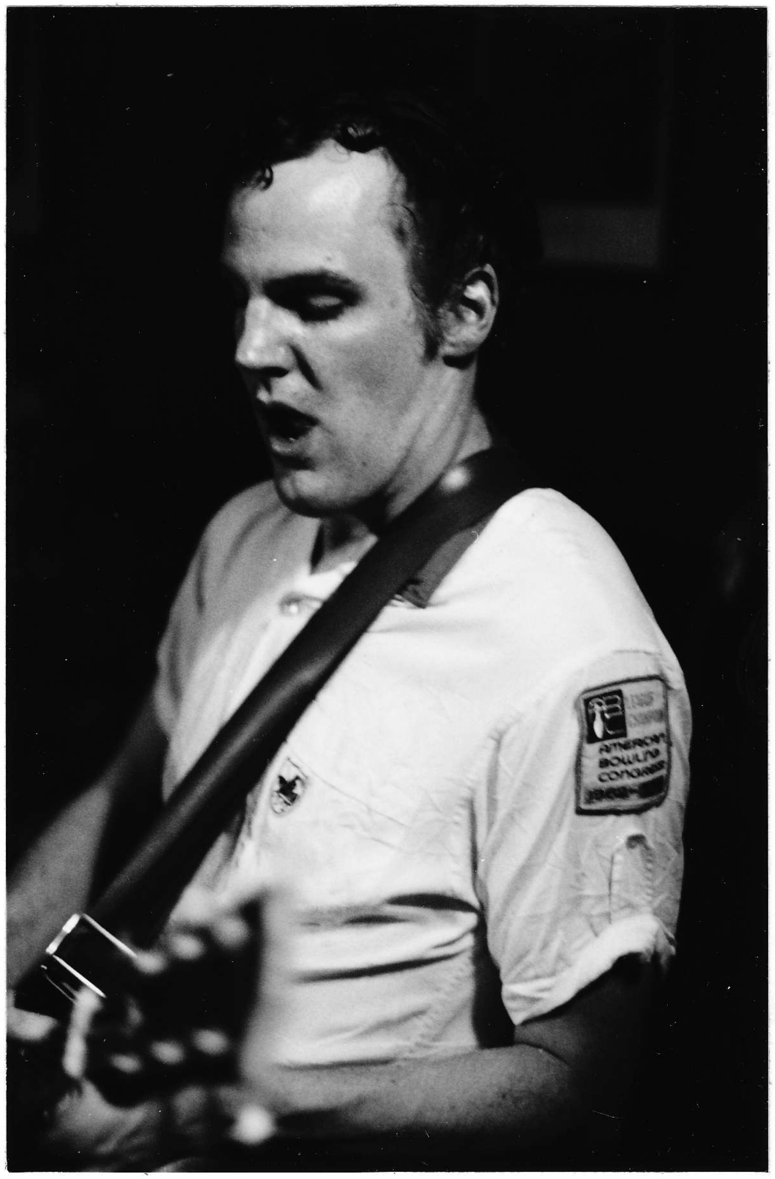
By: David Doty

Despite the growing popularity of synthesizers, instruction books suitable for the beginner are still rather rare. This is the role I believe The Complete Synthesizer was intended to fill; but, although it offers a fair amount of valuable information in a reasonably palatable form, this books is far less than its title claims. Virtually all of the functions common to the contemporary analog synthesizer are presented here; and the information offered is, so far as it goes, correct; but something important is missing: the idea that synthesizers are, at best, extremely powerful musical tools whose potential is yet to be fully explored.

The book begins with a chapter devoted to the basic parameters of musical sound. This is as it should be, and the text here is simple and straightforward enough that a reader with no background in acoustics should be readily able to grasp these important concepts. The second and by far the longest chapter is entitled "The Synthesizer Voice Module", a title which is indicative of the book's viewpoint. The synthesizer is presented as a normalled, keyboard controlled instrument with the signal path that has been prevalent since the introduction of the minimoog. The various modules are described in terms of the roles they play in this scheme, and alternative possibilities are largely ignored. As a result of this approach, the type of synthesis described here is primarily the imitative sort as practiced in recent rock and pop music. Techniques which fall outside this genre are treated with a certain condescension, when they are mentioned at all. For instance, regarding the LFO, we are told that sine and triangle waves are for vibrato, square waves are for trills, and that "Sawtooth frequency modulation is generally limited to special effects, such as sirens...". Similarly, the sample and hold is represented exclusively as a generator of random control voltages which, we learn, "are seldom used except for bizarre, spacey effects". While these attitudes are typical of a certain style of synthesis that is prevalent today, their presentation in a book such as this, to the exclusion of others, is apt to lead the beginner down a path of well-worn cliches, rather than encouraging creativity and exploration.

The remaining chapters reflect the same bias. Under "Types of Synthesizers" the modular synthesizer receives less space than the "pseudo-polyphonic" top octave divider type instrument, which is not properly a synthesizer at all. "Using the Synthesizer", which might properly have been the longest chapter in a book such as this, receives a mere five pages, with four of these being devoted to imitative voicings. A few recipes are given, including the mandatory string, brass, and organ patches, but nothing is said about achieving the subtle note-to-note and register-to-register differences typical of acoustic instruments. The "Synthesizer Accessories" →

And now for a page from The Scrapbook...



Okay, back to the *East Side Pharaohs*—here I am stifling a yawn as I perform the song *Louie, Louie* for the third time in a row at the Kasota Legion Club. Those wedding dances are hell!

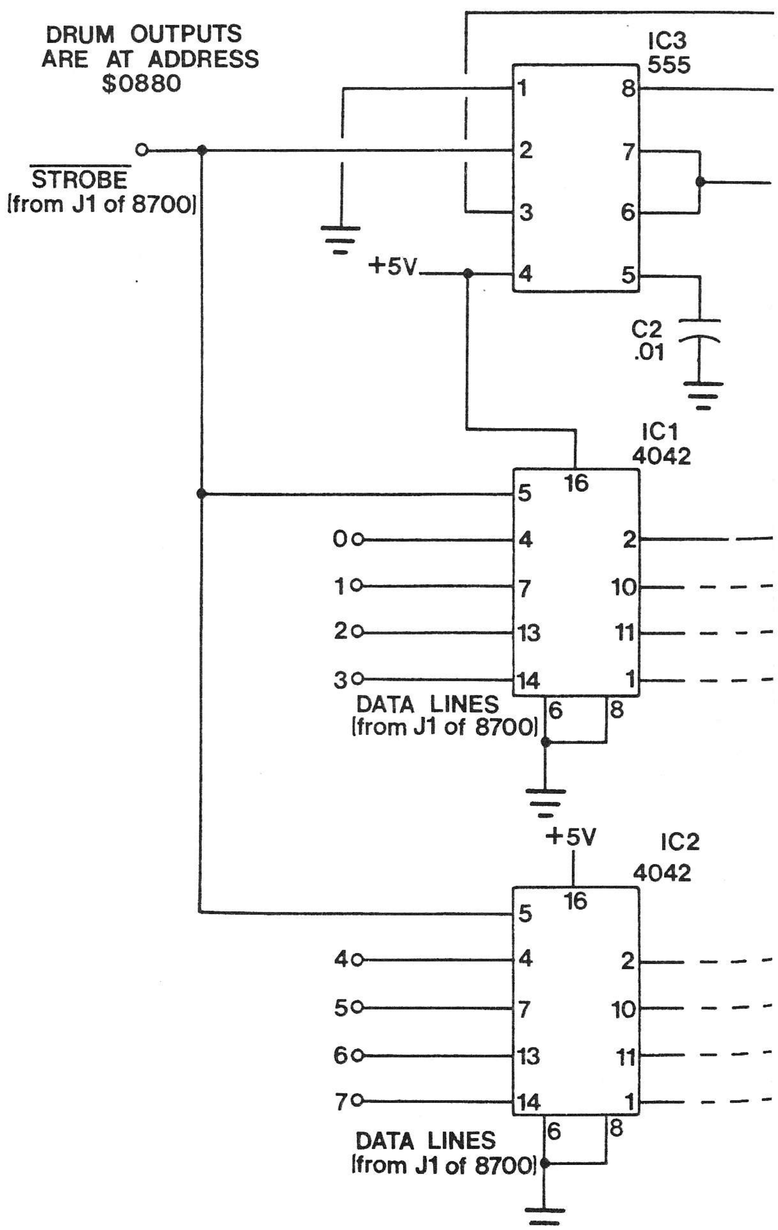
Practical Circuitry

MICRO DRUMS

PART I

figure 2

By: Tom Henry



Analog design has always fascinated me, and as a consequence I've never really gotten into digital or computer type circuits. However, just recently I dipped into this field and was amazed at the things that even a novice (like myself) can get a computer to do. As my first venture into this area, I came up with a computer controlled drum unit, called "Micro-Drums", which has absolutely revolutionized the manner in which I approach composition. Not being a drummer, my music has always been hampered by a lack of rhythmic expression, but this new circuit has changed all that.

What is Micro-Drums? Quite simply, it is a hardware/software combination which causes the PAIA 8700 microcomputer to think that it's a drummer. Up to eight drums can be polytonically controlled, over the range of an entire song. Nuances, bridges, breaks, lead-ins, even mistakes can be programmed into it so that the unit really drums as if it were a person. Depending on various factors, three to ten minute songs can be programmed with a great amount of depth and variation.

Editing a drum score is quite easy since an hierarchical approach is taken which closely approximates the manner in which you would write a song anyway. Once a score has been worked out, it can be saved to cassette and re-loaded at a later date. Thus, it is possible to create a library of drum scores and this eases the task of creating new songs later on.

But here's the real kicker. By using a special technique with the computer (the $\overline{IRQ}$, to be discussed later), we can sync the drum unit to click tracks on tape, other rhythm generators, sequencers -- or we can go the other way around! That is, Micro-Drums can be either a master or slave with equal ease. And best of all, this special technique drastically reduces both the hardware and software needed.

As mentioned, this project is based on the PAIA 8700 microcomputer. However, the same methods will work with just about any other computer using a member of the 6500 family, including the VIC-20, Commodore 64, and PET. As long as you can find one uncommitted address and can get access to the $\overline{IRQ}$ pin of the CPU chip, you can make it work. By the way, if you do configure this around the 8700 you'll be glad to know that the new hardware in no way interferes with the rest of the computer. You can still use it for your other applications.

Like most computer projects, the basic principle is simple although the explanations get quite long-winded. To keep things orderly, this installment of "Practical Circuitry" details the hardware needed for Micro-Drums, while next time the necessary software and programming instructions are described.

Refer to the schematic in figure 1. Essentially, we set up one address in the 8700's operating system to act as a drum output port. This port is memory mapped, and each bit in it controls a separate drum. Writing a byte into this port triggers the drums corresponding to the various bits. Address $0880 ("$" means that the number is in hexadecimal notation) is chosen for the drum output port, since this leaves the one at $0840 free for other synthesizer applications.

IC1 and IC2 are quad latches; their duty is to store the byte which is written to the drum port. Since a typical "write" operation on the data bus is only several microseconds long, we need these two chips to grab the desired byte and hold on to it.

When a "write" occurs, the $\overline{STROBE}$ line of the 8700 goes low for a microsecond or two. This line goes to pin 5 of each of the latches, hence causing them to latch the bytes currently on the data bus.

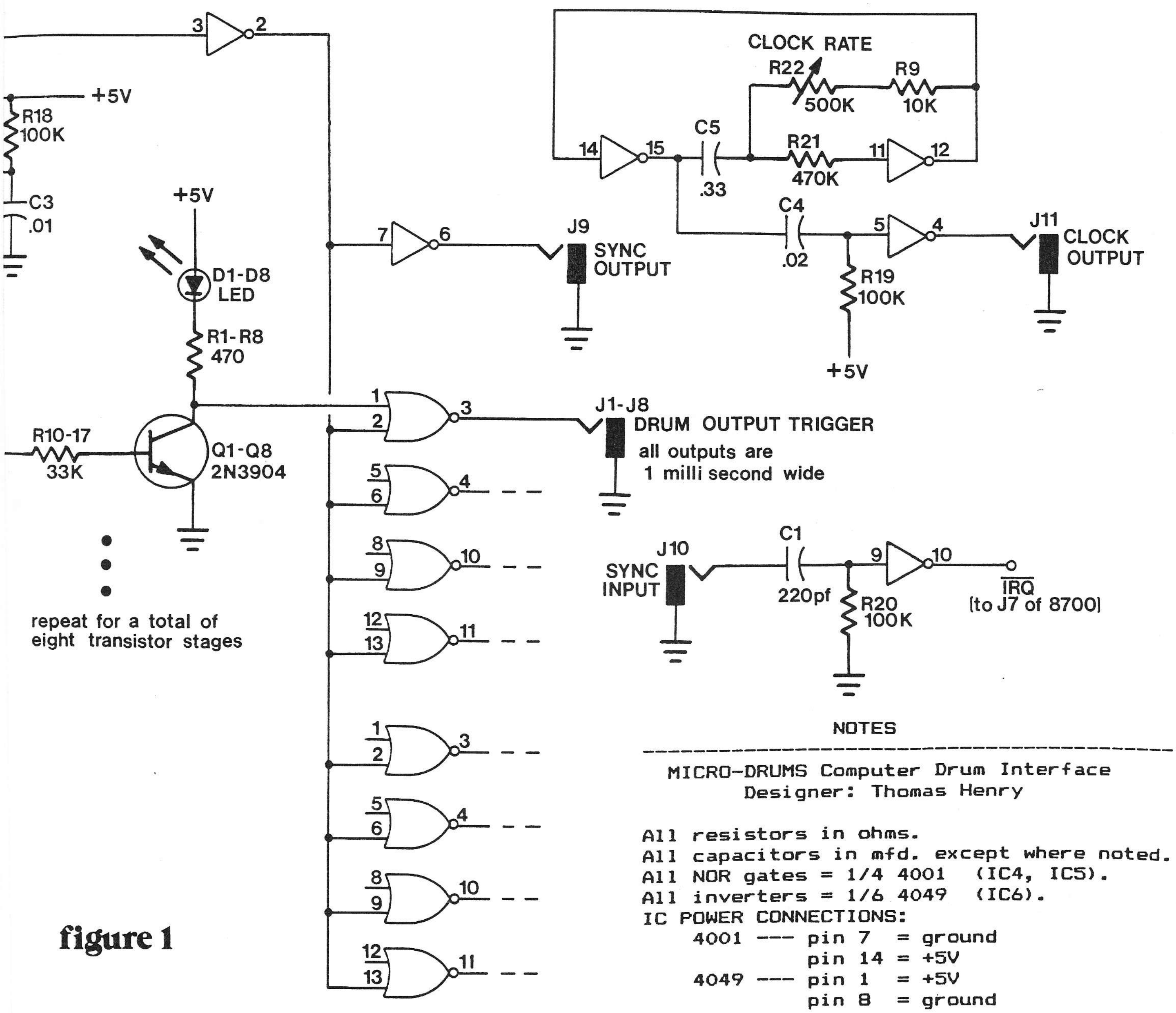


figure 1

Practical Circuitry......

But the STROBE signal also goes to IC3, which is a 555 timer set up as a one-shot. The output of this one-shot, upon being fired, goes high for a period of 1 millisecond. In effect, we have stretched out the STROBE signal (which is several microseconds long) to a 1 millisecond pulse. You will recognize this figure as being one of the standards we've talked about previously -- it's just the right length of time to fire envelope generators and drum circuits. In point of fact, every output in Micro-Drums generates a +5V, 1 millisecond wide pulse, and thus the circuit is compatible with any of the projects described in "Practical Circuitry".

By the way, notice how everything works out conveniently for the 555 input at pin 2. Normally, you have to provide some input conditioning to this pin, but as it turns out the STROBE signal from the 8700 has the correct polarity and pulse width so that all you have to do is just hook it up directly to pin 2. That ought to refute Murphy's Law!

Now backtrack just a bit to the latch outputs. Each output is buffered by a transistor (Q1 through Q8, one for each bit). Note that the transistor inverts the bit; however, the NOR gate following the transistor inverts the bit again. The net effect is that the bit passes to jacks J1 through J8 exactly as it was written to the drum port. To put it another way, write the number $FF to the drum port and all of the drums will be fired; write a number $00 to the port, and none of them will be fired.

Note that LEDs D1 through D8 turn on according to the number written to the port. These provide an excellent indication of what's going on while you're composing a song.

So, the latched data is sent to the transistors and then to the NOR gates. Now one input of each NOR gate is tied to the stretched out STROBE signal (from IC3 and the inverter). This causes the selected NOR gate to go high for a period of 1 millisecond, and as mentioned above, this is just what envelope generators and drum circuits like to see.

Notice that the stretched out STROBE signal is also available at J9. This signal may be used as a SYNC output and can lock sequencers, rhythm generators, and other computers to the main timing logic. Its pulse width is also 1 millisecond.

This takes care of the drum output support hardware. As you can tell, there really isn't very much to it, and in fact it is quite similar to the other output port at $0840 on the 8700. The rest of the circuitry in figure 1 has nothing to do with the drum output, but instead provides the necessary housekeeping circuitry to round out the complete system.

Let's look at the peripheral circuitry. Since we will be syncing the computer through its IRQ line (interrupt request), we must condition this input somewhat. J10 gives us access to this line. C1, R20, and the inverter form a half-monostable and as such take an input pulse of variable length and transform it into a precise 10 microsecond trigger. This trigger couples to the computer via IRQ, found at J7 on the 8700.

Why should this be a 10 microsecond pulse? The answer to this lies in the nature of the IRQ line in general. When the IRQ line of the 6503 CPU is brought low, the computer will cease whatever it is doing and jump to the service routine pointed at by the vector in location $0FFE and $0FFF. Control is then sent to this service routine, and the instructions found there will be executed until an RTI (return) command is encountered. Control is then sent back to the main program.

Now suppose that the IRQ signal which caused all of this to happen is still low. (In other words, the execution time of the service routine was shorter than the pulse width of the IRQ signal.) What will happen? Just what you would expect; the routine is called again! The upshot is that one IRQ signal caused the service routine to be called twice. We clearly don't need that, so the IRQ pulse is shaved down to 10 microseconds. With the 8700, 10 microseconds corresponds to about 5 program instructions, so as long as the service routine is more than 5 instructions long, all will work well. Incidentally, it should be clear that the IRQ pin responds to "levels", not "edges".

As you will see next time, when we discuss the software aspects of Micro-Drums, this IRQ business is the key to the entire system. Not only does it make master/slave relationships possible, but it also allows use to achieve a remarkable analog to digital conversion for the price of a patch cord! And as mentioned before, both the hardware and software can be drastically simplified.

The remaining three inverters of the 4049 package are put to use as a variable clock. There's nothing clever here since this circuit has been around for years. But one interesting aspect is that C4, R19 and an inverter are set up as another half-monostable. This time the pulse width is made to be millisecond wide (our old standard). R23 sets the clock rate, and with this control the output frequency can be continuously adjusted. Even though the frequency can be changed the pulse width will remain fixed at 1 millisecond.

Just to give you a sneak preview, a patch cord will be used to connect the CLOCK OUTPUT (J11) to the SYNC INPUT (J10), and thus interrupts can be made to occur at an adjustable frequency. This method will be employed to set the tempo of the song.

This just about covers the hardware aspects of Micro-Drums, but perhaps a few words about construction methods should be said. I built this circuit on a prototype circuit board (from Radio Shack), using ordinary hookup wire. If you employ this method, be sure to ponder the layout so that you won't run out of room at a crucial moment! Watch your power supply connections, but since the circuit only uses a +5V supply, this shouldn't provide any great problem. Also (need I say it?), use sockets, since this project employs CMOS integrated circuits which can be damaged by static electricity. Figure two shows the complete parts list for Micro-Drums.

After constructing the circuit, give a great deal of thought on how you will interface it with the computer. I used ribbon cable and headers to complete the connections to J1 and J7 on the 8700. After making the connections I mounted the board to the back of the 8700 computer itself. If you already have this computer, then you will know that it is a double-decked circuit board arrangement. By adding the Micro-Drums card, you will be left with a triple-decked affair.

And now is as good a time as any to mention a modification to the 8700 that you really ought to think about. I found that with just the bare-bones computer (no Micro-Drums added on), the power supply ran extremely hot. I took some measurements and discovered that the unit was drawing almost 900 mA! This is clearly way too much for the 7805 regulator to handle with such a small heatsink. The culprit, of course, is the RAM -- each chip consumes almost 70 mA. Multiply that by 8 (the number of 2112s in the 8700), and you've got quite a load for the regulator to handle.

In general I like to have at least a 2:1 margin of safety, so I decided to modify the computer accordingly. I simply built another +5V power supply and put the RAM on their own circuit. It's a crazy scheme, I know, but it does work and both power supplies now run considerably cooler. What's more, in the future I will be able to add on extra circuitry since I have a little more juice to play with now.

If what I've said doesn't make any sense to you, then <u>don't perform the modification!!!</u> Your 8700 is a valuable instrument and you won't want to wreck it. On the other hand, if you understand about power supplies, decoupling, and computers -- and if you have a steady hand for cutting PC board traces -- you might want to give this a try. Remember, the RAMs must be completely on their own circuit; it's no good just wiring two power supplies in parallel (unless you get off on rampant destruction of valuable equipment and enjoy fireworks).

After building the Micro-Drums card and performing the modification (if this applies to you), you can then put the thing in a suitable house. figure 2 shows how I did it. This is a homemade wooden box with two sheets of steel for a top and bottom. The 8700 is bolted to the top panel, and the keyboard shines through a square hole cut in the metal. I put some foam rubber around the hole and this keeps dust and moisture out. By the way, the fuse, switch, and line cord are on their own small panel mounted on the back surface of the box.

If you are using the PAIA 8700 power supply, then bring out the 60 Hz signal output to a front panel jack as well. (This is a logic level signal, NOT the line voltage!!!) Since this is a reliable frequency source, it might come in handy for future use.

As you examine the photo fig. 2 you will probably notice some features not described in the article (knobs, connectors, etc.). These have nothing to do with the Micro-Drums interface. For example, there are two D-25 jacks on the right side; one of these is a dummy (for future expansion) and the other is an interface to my keyboard synthesizer. When you build your unit, you might like to plan for the future, too, and leave some extra room for more connectors and whatnot.

Well, that wraps up the Micro-Drums hardware and it's a good thing too, since we're out of space. But come back next time for the concluding episode and see how to implement the software for a complete drum computer. Until then, here's a challenge for you to ponder: how would you create a real time clock using the SYNC input, the 60 Hz output, a patch cord, and a bit of software?

re-view

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front-to-rear as well as left-to-right. This 3-dimensionality makes his heavily-produced electronic pop tunes a listening experience which goes beyond their significance as pop tunes. Other influences might be Zappa and Godley/Creme.

Berlin **Pleasure Victim** (Geffen 2036). I didn't want to like this group -- their music is too trendy and their videos have been a little pretentious. But after having played the record numerous times looking for a weakness to attack, I have to admit it grew on me; Terri Nunn's voice has a cloying innocence and John Crawford's and David Diamond's synthi backing is very professional. It's well defended from sharks like me.

Men at Work **Business as Usual** (Columbia FC37978); **Cargo** (Columbia QC38660). MAW has been called "the Velveeta of pop". However, it's precisely because their strong, well-crafted tunes appeal to even a jaded old reviewer that they're selling so many records. Despite the cheese.

Deckard/Cardwell/Vosh **Sound** (cassette). "Sound" is what it's all about, as this synthesizer trio spins long, introspective pieces full of original synthesizer sounds. Sounds that soothe, sounds that startle, sounds that bounce off the wall and refuse to leave -- always the devotion for The Sound. $4 postpaid from David Vosh, 6300 Goldenrod Court, Upper Marlboro, MD 20772.

Everfriend **Key Essentials** (cassette). Keyboard artist Bill Rhodes displays his skills, from Keith Emerson-like classical rock to piano fusion jazz to the dramatic "Life and Death of a Star" (reviewed May/June '81 as an EP). There's a couple vocal numbers too -- too bad the vocalist he chose sounds a little tentative. Never mind; the rest is top drawer. Contact Bill at 1 Windemere Rd., Piscataway, NJ 08854.

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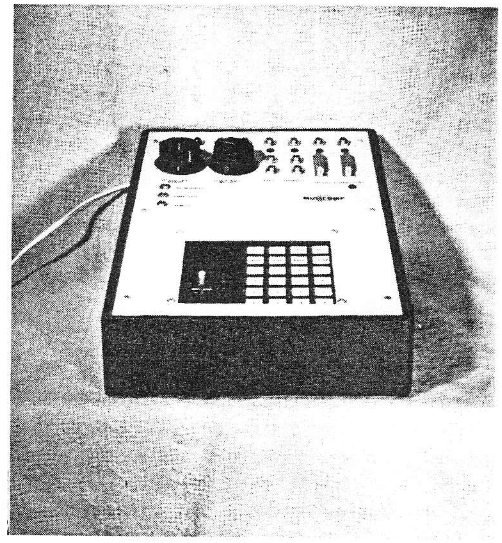
Practical Circuitry

MICRO DRUMS

PART II



By: Tom Henry

Last time in "Practical Circuitry" we discussed the hardware needed to implement Micro-Drums, a computer controlled drum unit. In this installment we will consider the software side of things. Since a lot needs to be said, let's jump right in and see how to punch up the program required to get Micro-Drums off and drumming!

Entering the program. Figure 1 shows the complete listing for Micro-Drums software. Since I used the PAIA 8700 computer, the code is written in 6502 assembly language. Those of you who plan to use Micro-Drums with some computer other than the PAIA unit will need to change the appropriate equates (at the start of the program) and may also need to alter the starting address. The source code in Figure 1 is heavily annotated, so you should be able to figure out how it works quite easily.

8700 users can ignore the line numbers, labels, mnemonics and comments if desired, since all that is needed to enter the program is the start address and the required code (under the heading "CODE", in the listing). Refer to your 8700 Computer/Controller manual for help in deciphering an assembler listing if you experience any difficulty.

Follow these steps to enter the program:

(1) Turn on the computer and hit the reset button.

(2) Load location $ED with the byte $1F. This sets the stack to a known condition needed by Micro-Drums.

(3) Get ready to start loading data at $0120, by punching in this address and hitting the DISP key.

(4) Using Figure 1 as a guide, start entering the data. The first few bytes are $20, $34, $0F, A5, etc.

(5) Keep entering data until you hit the last byte in the listing. This is $02 (at location $028B). Use the PCH and PCL keys to confirm that you're at the right place.

(6) You're now ready to save the program to tape. Follow the normal 8700 Cassette Interface protocol for saving a program. The start address in $1020, the end address is $028B, and you can use $01 for an identifier.

If all has gone well, you now have a working copy of the Micro-Drums program. After debugging your work (if needed), make a few backup copies as well. Now, let's see how to use the complete Micro-Drums system.

Using Micro-Drums. To use Micro-Drums, follow these instructions:

(1) Reset the 8700 computer.

(2) Load location $ED with the byte $1F. This sets the stack to a known condition needed by Micro-Drums. Do not forget this step; the program will not load or run correctly if it is left out!

(3) Using the standard loading procedure, load the Micro-Drums software. The start address is $0120, the end address is $028B and the identifier is $01.

(4) Run the program by typing $025F and hitting the RUN key. If everything has gone well, you will hear a long beep. This long beep indicates that you are in the main loop, and the computer is awaiting your instructions.

When in the main loop (as you now are), you have a choice of four commands. They are COARSE EDIT, FINE EDIT, TAPE and PLAY. After any of these commands have been executed, you will always be ushered back to the main loop. Even though the 8700 has limited display capabilities, you can always tell when you are back in the main loop by the long beep. Also, if you hit an invalid key when in the main loop, a long beep will occur. By listening for this beep, you can always tell what's happening at any moment.

Here follows a description of the four main commands. Note that when within the four main commands, there are other minor subcommands possible.

FINE EDIT. The fine edit command defines measures of patterns which will be used as the basis for the entire song. You may define up to eight different patterns, each one up to 32 beats long. Or you may partition this in other ways; for example, four patterns of 64 beats each.

Each pattern is given a number-name. The numbers 0 through 7 are used for this purpose. To start editing a given pattern, when in the main loop type the number-name and then hit the PCL key. (Mnemonic: think of PCL as "low", for lowest level of editing, the pattern.) So to start editing pattern zero, type $00 and then hit the PCL key.

You will now be sent from the main loop to the FINE EDIT routine. The display will show the current beat number, and the drum LEDs will show the drums to be played during that beat. To turn a drum on for the beat, touch one of the keys from 0 to 7. The corresponding LED will light up, and all of the currently selected drums will be played as a test. You can turn a drum off by touching the corresponding drum key again. Thus, the drum keys are like toggle switches and may be used to turn drums either on or off. There is one limitation, though: you may not have a particular beat play all eight drums at once. You can have up to seven drums playing at once, but not all of them. (The software uses the all-eight condition as an end-of-the-pattern marker.)

When you have achieved the selection of drums desired, touch the ENTER key. This will store the selection, then increment the beat pointer. The display will show the next beat number, and you are all set to enter the next selection. Also, when the display increments, the drums currently selected for the new beat will be triggered once. Thus you can single step through a pattern, for trial purposes, just by touching the ENTER key repeatedly. You can back step with the BACK key just as easily. And if you want to hear the current beat several times, touch the DISP key. This will sound the current drum selection.

As mentioned, keys 0 to 7 stand for drums 0 to 7. Key 8 stands for a rest. This clears out the drums selected for that beat. When you are done editing a pattern, touch the 9 key and you will be ushered back to the main loop. Note that even though the pattern may be up to 32 beats long, you are not obligated to use all of the beats. For example, with 5/4 time you might want to only use 10 beats.

COARSE EDIT. After creating some patterns (see above), you

```
0000                ;****************************************
0000                ;*                                      *
0000                ;*            MICRO-DRUMS               *
0000                ;*  MICROCOMPUTER CONTROLLED DRUM UNIT  *
0000                ;*                                      *
0000                ;*      (C) 1983  THOMAS HENRY          *
0000                ;*   VERSION 2.1  FEBRUARY 5, 1983      *
0000                ;*                                      *
0000                ;****************************************
0000                ;
0000                ;
0000                IRQVEC = $00           ;IRQ VECTOR.
0000                PATTER = $03           ;PATTERN POINTER BASE.
0000                SELECT = $05           ;DRUM SELECT BIT PATTERNS.
0000                PARAMS = $0D           ;TAPE PARAMETERS.
0000                BEAT   = $14           ;CURRENT BEAT POINTER.
0000                SPOINT = $15           ;CURRENT EVENT SELECTED.
0000                REPEAT = $16           ;CURRENT REPEAT COUNTER.
0000                BUFFER = $F0           ;KEYBOARD BUFFER.
0000                SCORE  = $0280         ;DRUM SCORE AREA.
0000                DISPLA = $0820         ;DISPLAY ADDRESS.
0000                DRUMS  = $0880         ;DRUM OUTPUT ADDRESS.
0000                RELAYS = $0E25         ;TURN ON TAPE RELAYS.
0000                CASS   = $0EAA         ;PERFORM CASSETTE OPERATION.
0000                DECODE = $0F00         ;INPUT A BYTE.
0000                GETKEY = $0F1F         ;GET A BYTE.
0000                BEEP   = $0F22         ;BEEP THE BEEPER.
0000                LBEEP  = $0F24         ;WITH SETUP, GIVES LONG BEEP.
0000                SHIFT  = $0F34         ;SHIFT BUFFER BY ONE DIGIT.
0000                ;
0000                ;
0000                ;*** MAIN LOOP ***
0000                ;
0000                ;
0000                       * = $0120
0120                ;
0120  20 34 0F      NUMBER JSR SHIFT       ;SHIFT IN NEW DIGIT.
0123  A5 F0                LDA BUFFER      ;FETCH PACKED ENTRY.
0125  8D 20 08             STA DISPLA      ;THEN UPDATE THE DISPLAY.
0128  4C 31 01             JMP INPUT       ;GO GET NEXT INPUT.
012B  A2 FF         MAIN   LDX #$FF        ;GET READY FOR LONG BEEP.
012D  18                   CLC
012E  20 24 0F             JSR LBEEP       ;DO LONG BEEP.
0131  20 1F 0F      INPUT  JSR GETKEY      ;WAIT FOR KEYSTROKE.
0134  C9 10                CMP #$10        ;IS IT A NUMBER?
0136  90 E8                BCC NUMBER      ;YES, BRANCH BACK AND GET
0138  C9 10         FIND   CMP #$10        ;IS IT 'PLAY'?
013A  F0 0F                BEQ PCMD
013C  C9 14                CMP #$14        ;IS IT 'COARSE'?
013E  F0 52                BEQ COARSE
0140  C9 15                CMP #$15        ;IS IT 'FINE'?
0142  F0 0A                BEQ FINE
0144  C9 16                CMP #$16        ;IS IT 'TAPE'?
0146  D0 E3                BNE MAIN        ;RAN OUT OF COMMANDS.
0148  4C 37 02             JMP TAPE
014B  4C E3 01      PCMD   JMP PLAY
014E                ;
014E                ;
014E                ;*** FINE EDIT COMMAND ***
014E                ;
014E                ;
014E  A5 F0         FINE   LDA BUFFER      ;GET PATTERN NUMBER.
0150  20 57 02             JSR OFFSET      ;GET PATTERN OFFSET.
0153  A0 00                LDY #$00        ;ZERO OUT THE BEAT POINTER.
0155  C8            SHOWIT INY
0156  88            BACKUP DEY
0157  8C 20 08             STY DISPLA      ;DISPLAY IT.
015A  B1 03                LDA (PATTER),Y  ;GET SELECTED BEAT,
015C  8D 80 08             STA DRUMS       ;AND PLAY IT.
015F  20 4F 02      FEDIT  JSR FETCH       ;GET EDIT KEYSTROKE.
0162  C9 12                CMP #$12        ;IS IT A 'BACK'?
0164  F0 F0                BEQ BACKUP      ;YES, BACKSPACE ONCE.
0166  C9 11                CMP #$11        ;IS IT A 'DISP'?
0168  F0 EB                BEQ SHOWIT      ;YES, PLAY CURRENT BEAT.
016A  C9 0A                CMP #$0A        ;IS IT A DRUM NUMBER (0-9)?
016C  90 08                BCC DENTER      ;YES, GO ENTER DRUM BEAT.
016E  C9 13                CMP #$13        ;IS IT AN 'ENTER'?
0170  D0 ED                BNE FEDIT       ;NO, RAN OUT OF COMMANDS.
0172  C8                   INY             ;YES, ADVANCE TO NEXT BEAT.
0173  4C 55 01             JMP SHOWIT
0176  C9 09         DENTER CMP #$09        ;#$09 MEANS END OF PATTERN.
0178  D0 06                BNE NEXT1
017A  A9 FF                LDA #$FF        ;END OF PATTERN MARKER.
017C  91 03                STA (PATTER),Y
017E  D0 AB                BNE MAIN        ;BRANCH ALWAYS.
0180  C9 08         NEXT1  CMP #$08        ;#$08 MEANS 'REST'
0182  D0 04                BNE NEXT2
0184  A9 00                LDA #$00
0186  F0 05                BEQ STORE       ;BRANCH ALWAYS.
0188  AA            NEXT2  TAX             ;INDEX INTO BIT PATTERN.
0189  B5 05                LDA SELECT,X    ;GET PROPER BIT PATTERN.
018B  51 03                EOR (PATTER),Y  ;ADD IN NEW BEAT.
018D  91 03         STORE  STA (PATTER),Y  ;AND SAVE IT.
018F  4C 55 01             JMP SHOWIT      ;SOUND THE DRUM BEAT.
```

```
0192                 ;
0192                 ;
0192                 ;*** COARSE EDIT COMMAND ***
0192                 ;
0192                 ;
0192  A6 F0          COARSE LDX BUFFER      ;GET DESIRED EVENT NUMBER.
0194  86 15                 STX SPOINT      ;STORE AT CURRENT EVENT.
0196  A6 15          REVEAL LDX SPOINT
0198  BD 80 02              LDA SCORE,X     ;GET CONTENTS OF EVENT.
019B  85 F0                 STA BUFFER      ;PUT IN BUFFER AND
019D  8D 20 08       VIEW   STA DISPLA      ;SHOW IT TOO.
01A0  20 1F 0F       LOOP   JSR GETKEY      ;GET KEYSTROKE.
01A3  C9 10                 CMP #$10        ;CHECK FOR NUMBER.
01A5  B0 08                 BCS NONUM       ;NOT A NUMBER, BRANCH.
01A7  20 34 0F              JSR SHIFT       ;SHIFT IN NEW DIGIT.
01AA  A5 F0                 LDA BUFFER      ;FETCH PACKED ENTRY.
01AC  4C 9D 01              JMP VIEW        ;AND UPDATE DISPLAY.
01AF  C9 13          NONUM  CMP #$13        ;IS IT AN 'ENTER'?
01B1  D0 0C                 BNE NEXT3       ;NO, GO ON.
01B3  A6 15                 LDX SPOINT      ;RE-GET EVENT NUMBER.
01B5  A5 F0                 LDA BUFFER      ;FETCH INPUT NUMBER.
01B7  9D 80 02              STA SCORE,X     ;STORE IN SCORE.
01BA  E6 15                 INC SPOINT      ;UPDATE EVENT NUMBER.
01BC  4C 96 01              JMP REVEAL      ;UPDATE DISPLAY.
01BF  C9 12          NEXT3  CMP #$12        ;IS IT A BACKSPACE?
01C1  D0 05                 BNE NEXT4       ;NO, BRANCH ON.
01C3  C6 15                 DEC SPOINT      ;DECREMENT EVENT COUNTER.
01C5  4C 96 01              JMP REVEAL      ;SHOW CONTENTS OF EVENT.
01C8  C9 14          NEXT4  CMP #$14        ;IS IT A 'PCH'?
01CA  D0 05                 BNE NEXT5       ;NO, BRANCH ON.
01CC  A5 15                 LDA SPOINT      ;GET CURRENT EVENT NUMBER.
01CE  4C 9D 01              JMP VIEW        ;AND SHOW IT.
01D1  C9 11          NEXT5  CMP #$11        ;IS IT A 'DISP'?
01D3  F0 C1                 BEQ REVEAL      ;IF SO, SHOW CONTENTS.
01D5  C9 17                 CMP #$17        ;'REL' STANDS FOR ALL DONE.
01D7  D0 C7                 BNE LOOP        ;RAN OUT OF COMMANDS.
01D9  A6 15                 LDX SPOINT      ;RE-GET EVENT NUMBER.
01DB  A9 00                 LDA #$00        ;END OF SCORE MARKER.
01DD  9D 80 02              STA SCORE,X
01E0  4C 2B 01              JMP MAIN        ;RETURN TO MAIN LOOP.
01E3                 ;
01E3                 ;
01E3                 ;*** 'PLAY' COMMAND ENTRY ***
01E3                 ;
01E3                 ;
01E3  A9 00          PLAY   LDA #$00        ;ZERO OUT REPEAT AND
01E5  85 16                 STA REPEAT      ;SCORE POINTER.
01E7  A9 FF                 LDA #$FF
01E9  85 15                 STA SPOINT
01EB  58                    CLI             ;PREPARE FOR IRQ.
01EC  C9 FF          TIGHT  CMP #$FF        ;#$FF MEANS KEEP PLAYING.
01EE  F0 FC                 BEQ TIGHT       ;STAY IN TIGHT LOOP.
01F0  4C 2B 01              JMP MAIN        ;ABORT 'PLAY' NOW.
01F3                 ;
01F3                 ;
01F3  20 00 0F       IRQRTN JSR DECODE      ;SEE IF ZERO KEY IS PUSHED.
01F6  C9 00                 CMP #$00
01F8  D0 04                 BNE PLAMOR      ;IT ISN'T, SO PLAY MORE.
01FA  28             FINISH PLP             ;SET INTERRUPT FLAG
01FB  78                    SEI             ;SO NO MORE OCCUR.
01FC  08                    PHP
01FD  40             RETURN RTI
01FE                 ;
01FE                 ;
01FE  A5 16          PLAMOR LDA REPEAT      ;REPEATED OLD PATTERN ENOUGH?
0200  D0 19                 BNE MORE        ;NO, KEEP GOING WITH OLD ONE.
0202  E6 15                 INC SPOINT      ;YES, UPDATE SCORE POINTER.
0204  A6 15                 LDX SPOINT
0206  BD 80 02              LDA SCORE,X     ;GET REPEAT TIME DATA.
0209  F0 EF                 BEQ FINISH      ;DONE PLAYING WHOLE SCORE.
020B  85 16                 STA REPEAT      ;CONTAINS NUMBER OF REPEATS.
020D  E6 15                 INC SPOINT      ;UPDATE SCORE POINTER.
020F  A6 15                 LDX SPOINT
0211  BD 80 02              LDA SCORE,X     ;GET PATTERN NAME DATA.
0214  20 57 02              JSR OFFSET      ;GET PATTERN ADDRESS OFFSET.
0217  A9 00                 LDA #$00
0219  85 14                 STA BEAT        ;ZERO OUT BEAT POINTER.
021B  A4 14          MORE   LDY BEAT        ;Y INDEXES TO PROPER BEAT.
021D  B1 03                 LDA (PATTER),Y  ;GET OUTPUT DATA.
021F  C9 FF                 CMP #$FF        ;END OF PATTERN?
0221  D0 08                 BNE OKAY        ;NO, GO PLAY THE BEAT.
0223  C6 16                 DEC REPEAT      ;DECREMENT REPEAT TIME.
0225  A9 00                 LDA #$00        ;YES, RESET BEAT COUNTER.
0227  85 14                 STA BEAT        ;THEN TRY AGAIN.
0229  F0 C8                 BEQ IRQRTN      ;BRANCH ALWAYS.
022B  8D 80 08       OKAY   STA DRUMS
022E  8C 20 08              STY DISPLA
0231  E6 14                 INC BEAT        ;UPDATE BEAT POINTER.
0233  A9 FF                 LDA #$FF
0235  D0 C6                 BNE RETURN      ;BRANCH ALWAYS.
0237                 ;
0237                 ;
0237                 ; *** 'LOAD' AND 'SAVE' COMMAND ***
0237                 ;
```

will then string them together in various arrangements to form the complete song. This is COARSE editing. You will create a score by entering some events; each event consists of two entries. The first entry is the number of times you wish a pattern to repeat, and the second entry is the number-name of the pattern which is to be repeated. There is room for 64 events total. This will allow songs up to fifteen minutes long to be programmed! To get into the COARSE EDIT mode from the main loop, type the number of the event you wish to start at (usually a $00) and then the PCH key. (Mnemonic: think of PCH as "high", the highest level of editing.)

The display will now show the contents of the current event. To enter a new event, type the desired number and hit the ENTER key. The event will be recorded, and the score pointer is incremented once. For example, starting at event zero, to get sixteen repeats of one, type $10, ENTER, $01, ENTER. Note that all numbers are in hexadecimal and that each entry must be followed by an ENTER.

You can backspace through a score with the BACK key. Also, to see the current event number, touch the PCH key at any time. To see the contents of the event, type DISP. Using the keys just mentioned, you can step through an entire score in a matter of minutes and change or update it as needed.

To finish off a score, touch the REL key. This puts in an end of score marker and returns you to the main loop. A long beep will occur.

PLAY. Playing a score is easy. First make sure that the SYNC INPUT jack has some source of triggers. You may sync the drum score off of the internal variable clock, an external clock, keyboard triggers, sequencer triggers, or click tracks from a tape deck. The input pulses should be +5V in magnitude. Note that Micro-Drums' internal variable clock meets this need and is perhaps the easiest to use. In addition it allows easy adjustment of the tempo: just dial in the desired speed. This may not seem like much, but consider what we've just done: a potentiometer controls the tempo, continuously, without the intervention of an analog to digital converter. How's that for saving money and keeping things simple!

After providing some source of sync pulses, you may start playing the score simply by touching the RUN key. When the song is finished, you will be sent back to the main loop and a long beep will occur. You can also abort a song while it is playing by touching the 0 key. Once again, you will return to the main loop.

As you can see, the SYNC INPUT (alias the IRQ) is the key to the power of Micro-Drums. Any circuit which can put out a series of pulses can cause Micro-Drums to step through the song, beat after beat. You are not constrained to meet this or that condition, and the circuitry is perfectly general. Simply send the computer some pulses and the song commences! And don't forget the SYNC OUTPUT jack either. You can cause some other circuit (like a sequencer) to follow Micro-Drums just as easily, so Micro-Drums can thus play the role of master or slave with equal ease.

TAPE. You can save or load scores using this command. To save a score, start the recorder going in the record mode, type $DD and touch the TAPE key. The computer will do the rest; there is no need to enter any addresses, etc.

Loading a score is just as easy. Start the recorder going in the play mode, type $11 and hit the TAPE key. The score will be loaded.

At the end of any tape operation you will be sent back to the main loop, and a long beep will indicate this fact. Note that the load and save options affect the entire score and pattern memory, so don't be alarmed if the operation takes up to a minute or so. If you experience any trouble, refer to the 8700 Cassette Interface manual and review how to set volume levels and so on.

The future. Well, that just about wraps up how to use Micro-Drums. Of course, all we have done here is talk about the mechanics of using the unit; it's up to you to think about the musical side of things. For example, if you know that a particular pattern is to contain both eighth notes and triplets, then you will need to divide the pattern into groups of twenty-four (three times eight). This is just one example of one of the musical considerations that must be taken into account with Micro-Drums. However, you will find that the more you play with Micro-Drums, the better you will become at visualizing what needs to be done.

Well, we've run out of room and need to start planning other projects. But in the meanwhile, as you play with the unit, think about how you would implement sequencer interfaces with Micro-Drums. The procedure is actually quite simple due to the "magical" way in which the SYNC INPUT works. Then consider synchro-sonic recording; how would you do this with Micro-Drums? Once again, the basic principle is quite simple. Think about these things and perhaps later on in the pages of "Practical Circuitry" we can compare notes.

```
0237                   ;
0237  A2 07      TAPE    LDX #$07          ;PREPARE TAPE PARAMETERS.
0239  B5 0C      SETFIL  LDA PARAMS-1,X    ;GET PARAMETERS.
023B  95 F0              STA BUFFER,X      ;AND STUFF IN PLACE.
023D  CA                 DEX
023E  D0 F9              BNE SETFIL        ;KEEP STUFFING IF NEEDED.
0240  A5 F0              LDA BUFFER        ;GET LOAD/SAVE TOKEN.
0242  20 25 0E           JSR RELAYS        ;TURN ON RELAYS.
0245  20 AA 0E           JSR CASS          ;PERFORM LOAD OR SAVE.
0248  18                 CLC
0249  20 22 0F           JSR BEEP          ;TURN OFF RELAYS AND BEEP.
024C  4C 2B 01           JMP MAIN          ;ALL DONE!
024F                   ;
024F                   ;
024F  84 14      FETCH   STY BEAT          ;GET A KEY, BUT SAVE
0251  20 1F 0F           JSR GETKEY        ;CURRENT Y-REGISTER.
0254  A4 14              LDY BEAT
0256  60                 RTS
0257                   ;
0257                   ;
0257  0A         OFFSET  ASL A             ;FIND OFFSET BY
0258  0A                 ASL A             ;MULTIPLYING ACCUMULATOR
0259  0A                 ASL A             ;BY SIXTEEN.
025A  0A                 ASL A
025B  0A                 ASL A
025C  85 03              STA PATTER        ;OFFSET ADDRESS IS HERE.
025E  60                 RTS
025F                   ;
025F                   ;
025F                   ;*** INITIALIZATION ROUTINE ***
025F                   ;
025F                   ;
025F  78                 SEI
0260  A2 00              LDX #$00
0262  BD 78 02   MOVE    LDA DATA,X        ;GET DATA BYTE.
0265  95 00              STA IRQVEC,X      ;STUFF IT INTO 0-PAGE.
0267  E8                 INX
0268  E0 14              CPX #$14          ;NUMBER OF BYTES+1.
026A  D0 F6              BNE MOVE
026C  A0 00              LDY #$00          ;CLEAR PATTERN AREA.
026E  A9 00              LDA #$00
0270  91 03      CLEAR   STA (PATTER),Y
0272  88                 DEY
0273  D0 FB              BNE CLEAR
0275  4C 2B 01           JMP MAIN          ;GO START UP MICRO-DRUMS.
0278                   ;
0278                   ;
0278                   ;*** DATA AND ADDRESS TABLES ***
0278                   ;
0278                   ;
0278  4C         DATA    .BYTE $4C         ;OPCODE FOR 'JMP'.
0279  F3 01              .WORD IRQRTN      ;START OF IRQ ROUTINE.
027B  00 03              .WORD $0300       ;PATTERN BASE ADDRESS.
027D  01                 .BYTE $01, $02    ;DRUM SELECT BIT PATTERNS.
027E  02
027F  04                 .BYTE $04, $08
0280  08
0281  10                 .BYTE $10, $20
0282  20
0283  40                 .BYTE $40, $80
0284  80
0285  00                 .BYTE $00         ;FILE PARAMETER.
0286  FF 03              .WORD $03FF       ;TAPE END ADDRESS.
0288  80 02              .WORD $0280       ;TAPE START ADDRESS.
028A  80 02              .WORD $0280       ;TAPE POINTER.
028C                     .END

ERRORS = 00000

END OF ASSEMBLY
```

Practical Circuitry

Modifying the PAIA EKx-40 VCO

By: Tom Henry

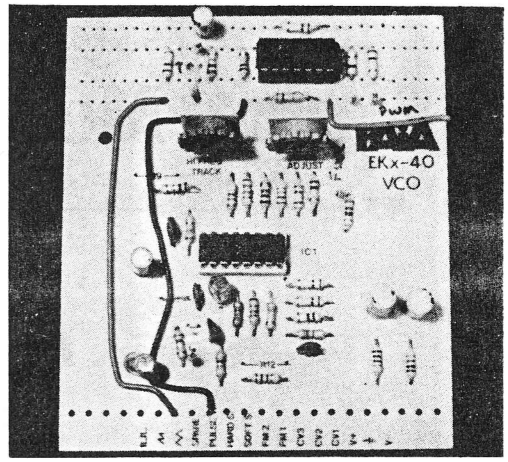


figure 2

The PAIA EKx-40 is a VCO kit configured around the CEM-3340 integrated circuit. It was not intended to be a complete VCO module, but rather a starting point upon which experimenters can expand. In this installment of "Practical Circuitry", we will modify the EKx-40 to give it standard output voltages and impedances. We will also beef up the input controls to include more options, and jazz up the pulse width circuitry.

Fortunately, the EKx-40 circuit board has some extra "kluge" room, thus simplifying the process of adding additional circuitry. In a few instances you will need to tack-solder some components to the board or cut some traces with a razor knife; but, all in all, the work is fairly straightforward.

Before starting this project, be sure to review the EKx-40 owner's manual and also look over the CEM-3340 application note and spec sheet. Once you've got this all out of the way, let's get started!

Modifying the control inputs. We will begin by adding some extra features to the exponential frequency control inputs (labelled CV1, CV2 and CV3 on the circuit board edge connector). CV1 should go to a front panel jack; this jack becomes the 1V/octave input, and is a "wild" input since it has no attenuator associated with it. Next, send CV2 to the wiper of a 100K pot, whose hot side and ground should go to the hot terminal and ground, respectively, of a second jack. Mount both the pot and jack on the front panel; this pair forms an exponential FM input with attenuator. The attenuator is needed since it allows you to dial in the amount of FM required by the application at hand.

CV3 should go directly to the wiper of a 100k pot, where one of the remaining terminals connects to +15V and the other terminal connects to -15V. This pot becomes the coarse tuning control and sweeps the VCO over a range of about 1 Hz to 35 KHz. However, we need a fine tuning control as well. To implement this, solder one end of a 2.7M resistor to the summing node terminated by R17, R18, and R19. (Essentially, you are just adding another control input; call it CV4 if you wish.) The other end of this resistor goes to the wiper of another 100K pot, where again, one remaining terminal connects to +15V and the other to -15V. Since 2.7M is such a large value, the pot will have a much more restricted range, which is what you would expect from a fine tuning control.

This takes care of the four exponential type inputs. Let's examine the linear input next. Start by eliminating R11, since we really only need one linear FM input (labelled FM1). This input should then go the the wiper of yet another 100K potentiometer. The hot side of this pot goes to a jack on the front panel, and the remaining terminal goes to ground. This jack and pot form the linear FM input control duo, and allow you to dial in the required amount of FM.

What about the sync inputs? Well, these are quite easy to fix up as well; consider the hard sync input first. Modify this input by adding a 27K resistor in series with C1 (which leads to pin 6 of the CEM-3340). This resistor restricts the sync input voltage swing, so that a full 10V p-p pulse may be used as the sync signal (10V p-p is our standard signal level). The 27K resistor should terminate at the hot terminal of a closed circuit jack; the switching contact of this jack should be wired to ground so that the sync input is grounded until a plug is inserted into the jack. This guards the sensitive sync input from outside interference.

The soft sync input is modified in almost the same way. Put a 47K resistor in series with C2 (which leads to pin 9 of the CEM-3340). Once again, the other end of this resistor should terminate in a closed circuit jack which has been wired so that the input is grounded if no plug is inserted. (By the way, remove the jumper labelled "*" on the EKx-40 circuit board if it had been installed prior to making these changes.)

So far, our modifications have involved nothing more than adding a resistor here or there, and slapping in a few pots and jacks. To jazz up the pulse width control, we will need to use some slightly more sophisticated circuitry. In particular, we will have to add an op amp and several more resistors. Luckily, the "kluge" area on the circuit board is more than adequate to accommodate our needs.

Refer to figure 1, which shows a schematic for the remaining modifications to be made to the EKx-40. One half of a 4739 dual op amp (or equivalent) beefs up the pulse width circuitry. There are two controls associated with this structure. One is the initial pulse width control, which can manually adjust the pulse width from about 0% to 100%. The

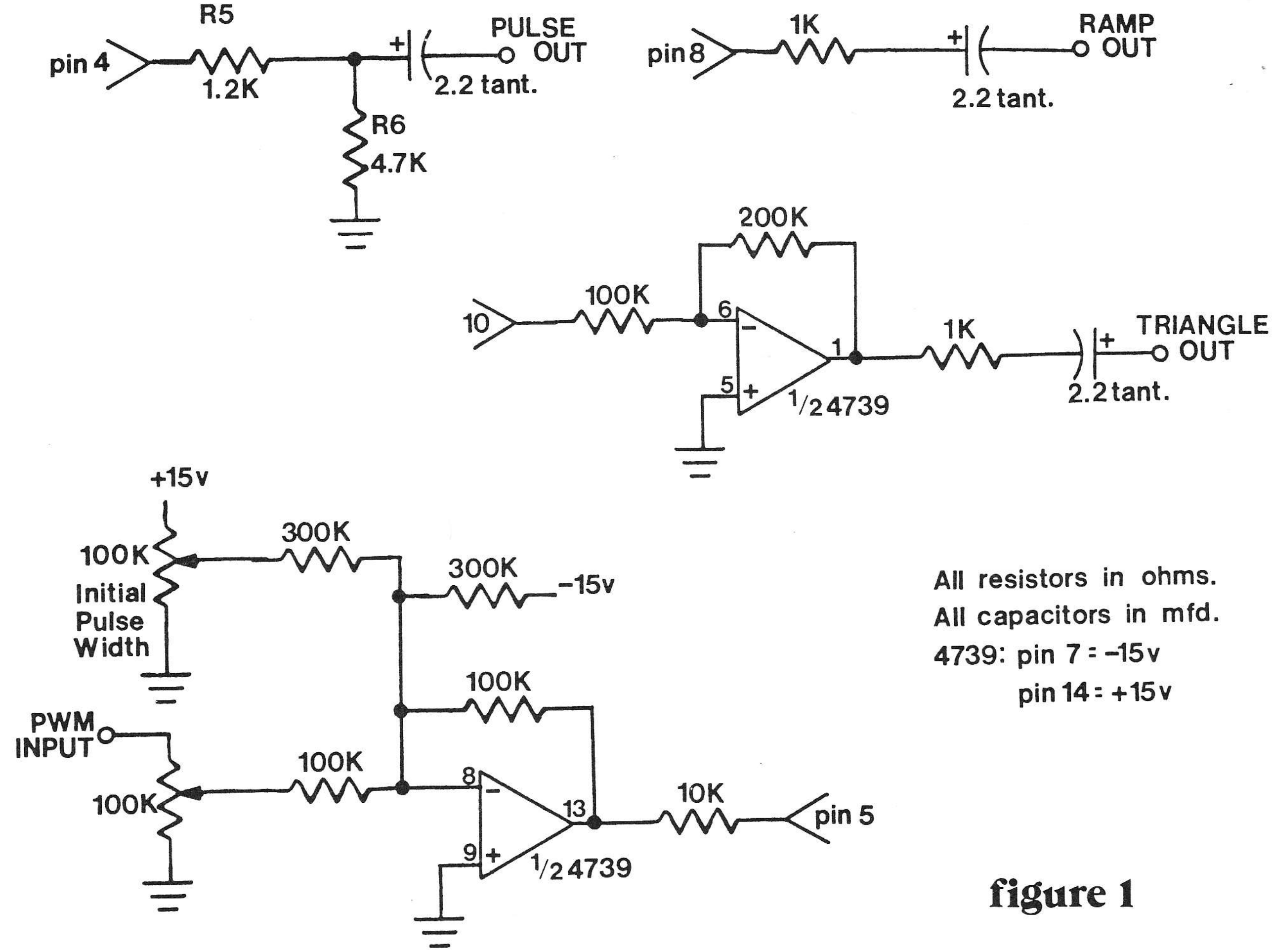


figure 1

other control is an attenuator for the PWM input. This lets you reduce the level of modulating signals.

Use some insulated wire to carry the output of this structure to pin 5 of the CEM-3340. After adding these modifications, I think you will find the pulse width to be much more manageable. Pulse width modulation yields such a neat sound that the extra work needed to give you more control over this parameter is well worth the effort!

Modifying the outputs. Now that we've taken care of the EKx-40 inputs, we want to fix up the outputs so that they meet the standards established in this column over the past couple of years (10V p-p signals with output impedances of 1K).

One standard that we will have to forego is that of direct coupling. To add direct coupling (meaning no capacitors) to the outputs would require quite a bit of extra circuitry, and is hardly worth the effort. This implies that the modified EKx-40 will not be suitable for use as a low frequency control oscillator. However, this in no way detracts from its value as a good tone source. If you really need direct coupled outputs, then try the "VCO Deluxe" previously described in this column (Polyphony, November/December 1981, pp. 28-29, 31).

Consider the pulse wave output first. To bring the signal level in line with our standard, replace R5 with a 1.2K resistor and R6 with a 4.7K resistor. Besides adjusting the signal level down to 10V p-p, this also provides an output impedance of about 1K. To correct for any DC offset, add the 2.2 uF tantalum capacitor shown in figure 1. A good way to install this cap is to find a suitable place on the PC board trace and cut it with a razor knife. You can then drill holes for the capacitor, and solder it in place (or tack-solder it across the cut trace).

The ramp wave is brought up to spec by eliminating R9 altogether and replacing R8 with a 1K resistor. Another 2.2 uF tantalum capacitor blocks DC. Once again, the level will be 10V p-p.

Upgrading the triangle wave output from 5V to 10V requires the other half of the 4739 dual op amp, which amplifies the triangle wave by a factor of two (see figure 1). You can pick up the input triangle wave at pin 10 of the CEM-3340. The output will have a negative DC bias on it; use a 2.2 uF tantalum capacitor to block this. Please note that the polarity of this capacitor is the opposite of the other two used thus far.

Building the modified EKx-40 VCO. Finishing up the project is quite simple, although with the addition of all of these new fea-

Practical Circuitry

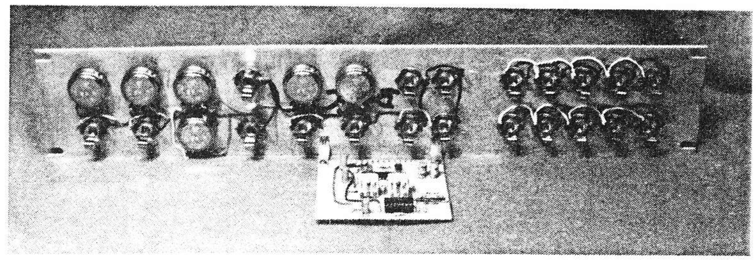


figure 3

tures, a standard single width rack panel is just a bit too small. So, I suggest using a 3-1/2" by 19" panel. You can devote any extra space to a two-by-five multiple. (You can never have enough multiples!)

Figure 2 shows the modified circuit board. Note the jumper wires, and the use of the "kluge" area to accommodate the new components. Figure 3 shows the back side of the front panel just prior to completing the final wiring. The circuit board is supported by two small angle brackets and some #4 hardware.

While this VCO is very easy to build, it doesn't have quite the same versatility as the "VCO Deluxe" (see above). Nonetheless, as VCOs go, it is still quite a bit better than most units -- even compared to those found in commercial synthesizers. It stays in tune, tracks accurately over a very wide range and generally has a very clean sound. And the price is right -- who would have thought, just five years ago, that it would be possible to build a quality VCO with superb features in your own workshop for under $30! How times change!

AD INDEX

Practical Circuitry

Build A Quadrature Function Generator

By: Tom Henry

What exactly is a quadrature function generator? To answer that, we must first understand the concept of a quadrature oscillator. An oscillator is said to have quadrature outputs if it produces simultaneous sine, cosine, negative sine, and negative cosine outputs. Such an oscillator can be created quite easily by setting a four pole lowpass filter into oscillation, and tapping the required outputs from consecutive stages[1]. If trigonometry alarms you, just consider the outputs to be four sine waves, each one ninety degrees out of phase with the previous output.

Despite the interesting patches possible with a quadrature oscillator, it does have one main drawback: Sine wave oscillators are fairly touchy, and you will often find that the amplitude will change as the frequency is swept over a wide range. In addition, under some conditions the oscillator may fail to oscillate; or, at the other extreme, hard clipping may bring about some undesirable distortion.

That's where the quadrature function generator comes in, since we will throw out the oscillator completely and replace it with a function generator. Oscillators are reactive; they depend on a resonating RC network. Function generators are non-reactive (in the sense that they don't resonate); their timing depends solely on the charging and discharging of a capacitor. With this circuit, the outputs are very stable in amplitude and purity over a very wide range. Finally, one more distinction between oscillators and function generators is that the former generates sinusoidal outputs, while the latter generates triangle waves (or sometimes ramp waves).

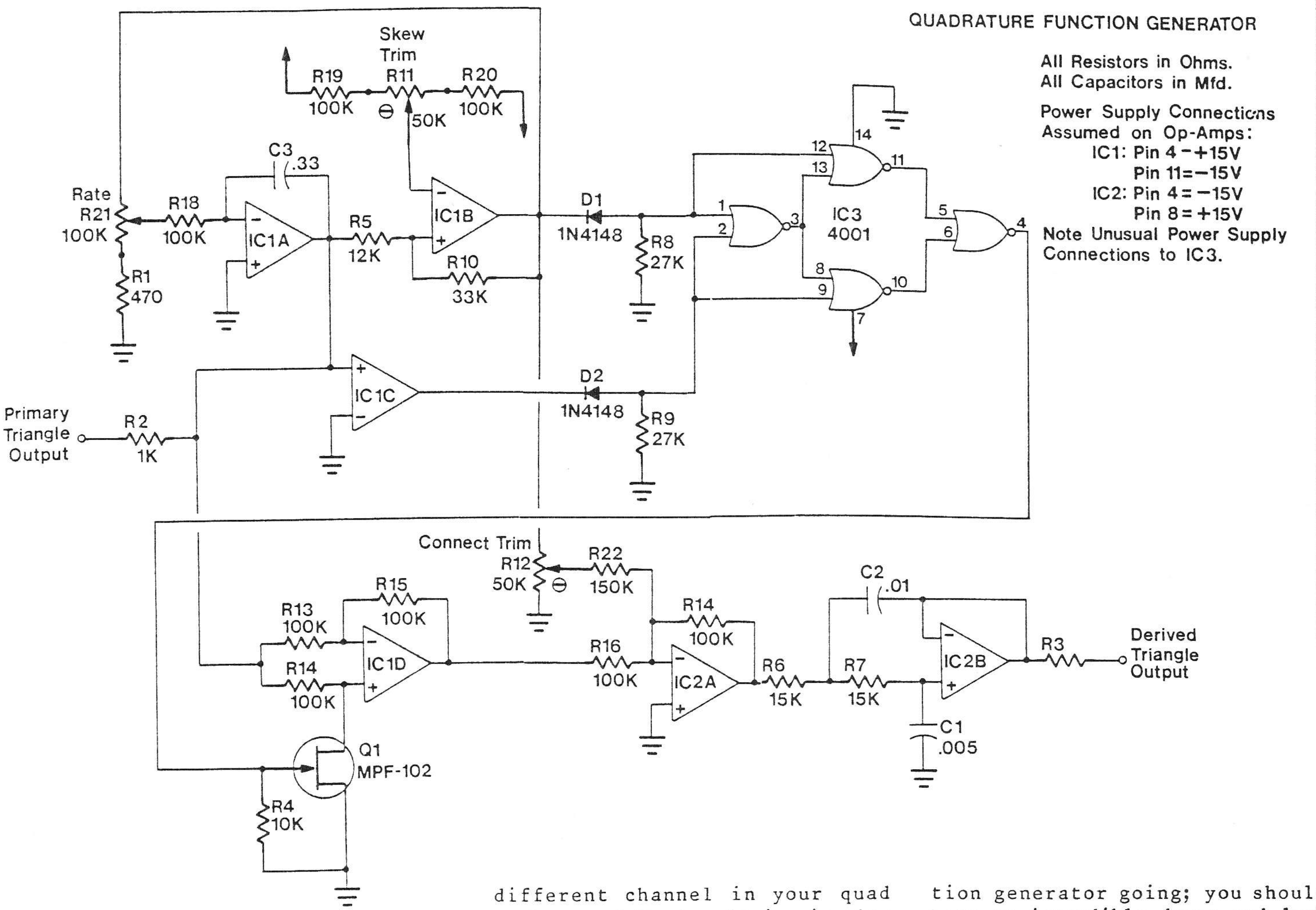
How it works. To fully understand the workings of a quadrature function generator, we must resort to some mathematics. There's hardly room to do that here, so if you're interested in the math behind the circuit, please refer to another article of mine which gives the complete analysis of a quadrature function generator[2]. (Actually, the circuit presented here is more compact and uses fewer parts than the earlier version, but the circuit action is very similar.) But even though we can't go through the mathematics here, we can still get an intuitive feel for how the circuit works. Referring to the schematic, op amps IC1A and IC1B form a Schmitt trigger/integrator function generator, an old friend from way back. C3 sets the basic frequency range, with R21 allowing for an adjustable rate. It is important to note that the output of IC1A is a triangle wave, and the output of IC2B is a square wave. The triangle wave goes directly to the "Primary Triangle Output" via R2. In addition, various line segments of the triangle wave are used in conjunction with Quadrature Function Generator's other circuitry to construct a new triangle wave ninety degrees out of phase with the first -- and that's where the square wave output from IC1B comes in. This output tells the circuit when to grab the various segments needed in the construction of the new triangle wave.

IC3 is configured as an EXCLUSIVE-OR gate; it seemed more cost-effective to use the dirt cheap 4001 quad gate for this, rather than using one gate out of a 4070 EX-OR gate. Note the unusual power supply hookup for this chip. Since this part of the circuit drives the gate of an N-channel FET, we need a swing from negative to ground.

IC1D, Q1, and their associated circuitry comprise a sign changer. This circuit will invert or not invert the input, depending upon the control voltage at the gate of Q1. Once again, see (2) for more details on the function of this sub-circuit.

Note that, so far, all of the op amps have each been 1/4 of a TL084 quad bi-fet op amp package. This particular chip must be used (instead of 741s, for example), since this circuit requires an extremely high slew rate -- we want all of the switching to be as clean as possible. But now we're going to reverse this philosophy and specify as slow an op amp as possible for IC2A and IC2B! IC2A sums together the various line segments to form a new triangle wave which is ninety degrees out of phase with the original. By specifying a low slew rate for this amp, any of the discontinuities in the derived triangle will be masked by the op amp's inability to slew fast enough. Pretty sneaky! In addition, IC2B is set up as a lowpass filter with a cutoff frequency of 1.5 kHz, which also helps smooth out the new triangle output.

Adjusting the trimpots is fairly easy. To simplify the process, temporarily replace C3 with a 0.05 uF capacitor. This will increase the frequency to a more easily observable range. Now monitor the "Derived Triangle Output" on an oscilloscope. While watching the waveform, go back and



forth between trimmers R11 and R12 until the waveform "comes together" and connects to form a smooth triangle wave. This process is quite magical; I think the sight on the scope will really amaze you! If you have a dual channel scope, compare the two waveforms ("Primary" versus "Derived") and confirm that they are indeed ninety degrees out of phase with respect to each other.

To round out the circuit you will probably want to provide inverted versions of the "Primary" and "Derived" triangle outputs. This will give you a total of four outputs, each one ninety degrees out of phase with the previous.

Applications. Well, what shall we use it for? For a start, how about automatic quadraphonic panning: Gang the inputs of four VCAs together, and control each one by a different triangle wave. Send each output of the VCAs to a different channel in your quad system, and the result is circular location modulation. What?!? You don't have a quad system? Then you can still use the quadrature function generator for some neat stereo effects. For a really bizarre sound, try this patch: Gang the inputs of VCAs 1 and 3 and apply a dry signal to these VCAs. VCA 1 should feed the left channel and VCA 3 the right channel. Now gang the inputs of VCA 2 and VCA 4, and apply an echoed signal to these inputs. VCA 2 should mix into the left channel, while VCA 4 mixes into the right. Now really crank up the delay time and feedback and hit some staccato notes -- but don't get seasick!

What?! You say you don't have a stereo rig either? That makes it harder to think up patches, but here's a good one to try. Apply an audio signal to four different flangers, with each one set for a slightly different initial delay time. Then send each flanger output to a VCA, and finally sum the VCAs together into a monaural output. Set the function generator going; you should hear an incredibly dense and lush sound. This is especially good for full-bodied instruments, such as rhythm guitar.

You may think I'm getting ridiculous, but what do you do if you don't have any synthesizer at all? (I'm serious now!) Well, you can create some great Lissajous figures on an oscilloscope screen, or better yet, hook the unit up to your laser art show for a far out display. (What?! No laser? Well...) There's quite a lot this little black box can do.

I've had a real blast designing and building the quadrature function generator. The circuit has a real "that's neat!" aspect to it, and is lots of fun to play with. If you come up with some interesting applications be sure to jot me a line c/o Polyphony.

(Editor's note: Splitting a signal into four filters, whose outputs feed four VCAs controlled by the Quadrature Function Genera-

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end first, over the cable. The wire carrying DB7 should be soldered to the "hot", or tip, lug of the plug. The remaining wire, ground, should be soldered to the remaining, or ground, lug of the plug. After the connection has cooled down, bend the tabs at the end of the plug ground around the cable, and screw the cover on to the plug.

The interface between the 6770 and the P8782 can now be tested. Patch the 1/4" phone plug end of the new cable into FROM TAPE on the 6770, plug the DB-25 interface connector into the 8700, and apply power to both units. Since the encoder clock on the P8782 is running constantly, it is not necessary to load a program for testing. On the 6770, set the SENSITIVITY control to minimum, the +5/+10 switch to +5, the INT.-/EXT. switch to EXT., and RUN/STOP to RUN. Now, advance the SENSITIVITY control and you should notice that the LEDs will begin to flash, which indicates the synchronizer is accepting the 8700's data bit as a clock signal. If the LEDs do not flash, the trouble could be in the patch cord, the encoder circuit, or the 6770. The encoder can be checked by loading and running the 8700's K-test program. If this program works correctly, then the encoder is probably functioning properly. If not, unplug the patch cord to the 6700 and try again. This will isolate the trouble to either the encoder or the cable. If the program runs with the cable disconnected, the trouble could be a miswired cable, or perhaps a problem at the input of the 6770 such as a miswired jack, improper solder connection, etc.

And now the rewards... Finally, everything can be patched together and tested as a system. With all the gear running in time and synchronized, you can easily create a multi-layered recording using only one track. (Incidentally, I often record that one track onto a cheapo portable tape recorder and save the tape for future inspiration.) You could also record Patchmod's sounds for an introduction or "fill" in a song to be recorded at a later date. But aside from recording, Patchmod is just plain fun -- and synchronizing a bunch of units together in a rhythmic fashion makes for a more complete sound when jamming or coming up with new ideas. ●

Practical Circuitry

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tor, can make for fascinating timbral changes. Also, for those of you who are into synchro-sonic recording techniques, adding a CMOS switch in parallel with C3, and feeding the switch control terminal with an appropriate trigger pulse, will reset the oscillator at the rate of the trigger. All in all, for those experimenters who don't quite need the sophistication or versatility of John Simonton's "Shepard Function Generator" presented in the February 1982 issue of Polyphony, the Quadrature Function Generator provides a low-cost way to experiment with voltage controlled panning, cross-fading, channel-splitting, and the like.)

NOTES

(1) For example, see J. Patchell, "Build a Voltage-Controlled Quadrature Oscillator", Polyphony, Nov/Dec 1980, pp. 26-27.

(2) T. Henry, "A Function Generator With 'Quadrature' Triangle Wave Outputs", Electronotes #122, pp. 13-20.

PARTS LIST, QUADRATURE FUNCTION GENERATOR

Resistors (1/4 Watt, 5% tolerance preferred)

R1	470
R2,R3	1K
R4	10K
R5	12k
R6,R7	15K
R8,R9	27K
R10	33K
R11,R12	50K trim pot
R13-R20	100K
R21	100K potentiometer
R22	150K

Capacitors (15 or greater working Volts)

C1	0.005 uF, mylar preferred
C2	0.01 uF, mylar preferred
C3	0.33 uF, mylar preferred

Semiconductors

IC1	TL084 quad bi-fet op amp
IC2	1458 dual op amp
IC3	4001 CMOS quad NOR gate
Q1	MPF-102 N-channel FET
D1,D2	1N4148 or equivalent switching diode

Practical Circuitry

EASY FIRING ADSR

by: Thomas Henry

You'll probably think that I have a one track mind, but we're going to look at ADSRs again! Recall that back in the March/-April 1982 issue and again in the June 1983 issue of Polyphony we looked at two distinct approaches to building ADSRs for your home-brew synthesizer. By now you might think that the topic has been exhausted, but I hope that after you see this new circuit you'll agree with me that electronic music design is nowhere near being a finished subject. There are always new circuits out there just waiting to be discovered!

Since the publication of the two aforementioned articles, I've come to discover more and more uses for ADSRs (especially in the area of electronic percussion). Due to this new emphasis in my music, I spend about equal time with my keyboard synthesizer and drum unit, so it's important to me that an ADSR be useable with both systems. Thus, this issue's circuit is called the "Easy Firing ADSR" since anything from synthesizers to drum boxes to computers can fire it. Best of all, besides being a most versatile unit, it is also extremely easy to build.

Figure 1 shows the complete schematic for the circuit. Simple, isn't it? Like many of the circuits described in "Practical Circuitry", the simplicity comes about because we exploit some of the new integrated circuits developed especially for electronic music. The heart of this circuit is the Curtis CEM3310 ADSR chip (see below for availability). This IC contains all of the logic needed for a complete ADSR and requires a minimum of external circuitry. A few pots, a handful of resistors and capacitors, and a few hours of construction time will reward you with a very professional envelope generator.

How it works. Scanning the schematic, note that pins 15, 12 and 13 are the inputs for the Attack, Decay and Release voltages, respectively. By varying the voltages at these pins from 0 Volts to about -240 mV, we can modulate these three parameters over a 2 millisecond to 20 second range. Three identical potentiometer and voltage divider sections provide this control voltage. For example, consider the Attack section consisting of R8, R5 and R1. R8 is a potentiometer which allows you to pick off a voltage between 0 and 15V. Then, divider R5 and R1 attenuate this voltage to an absolute maximum of -240 mV. The sections for Decay and Release work similarly.

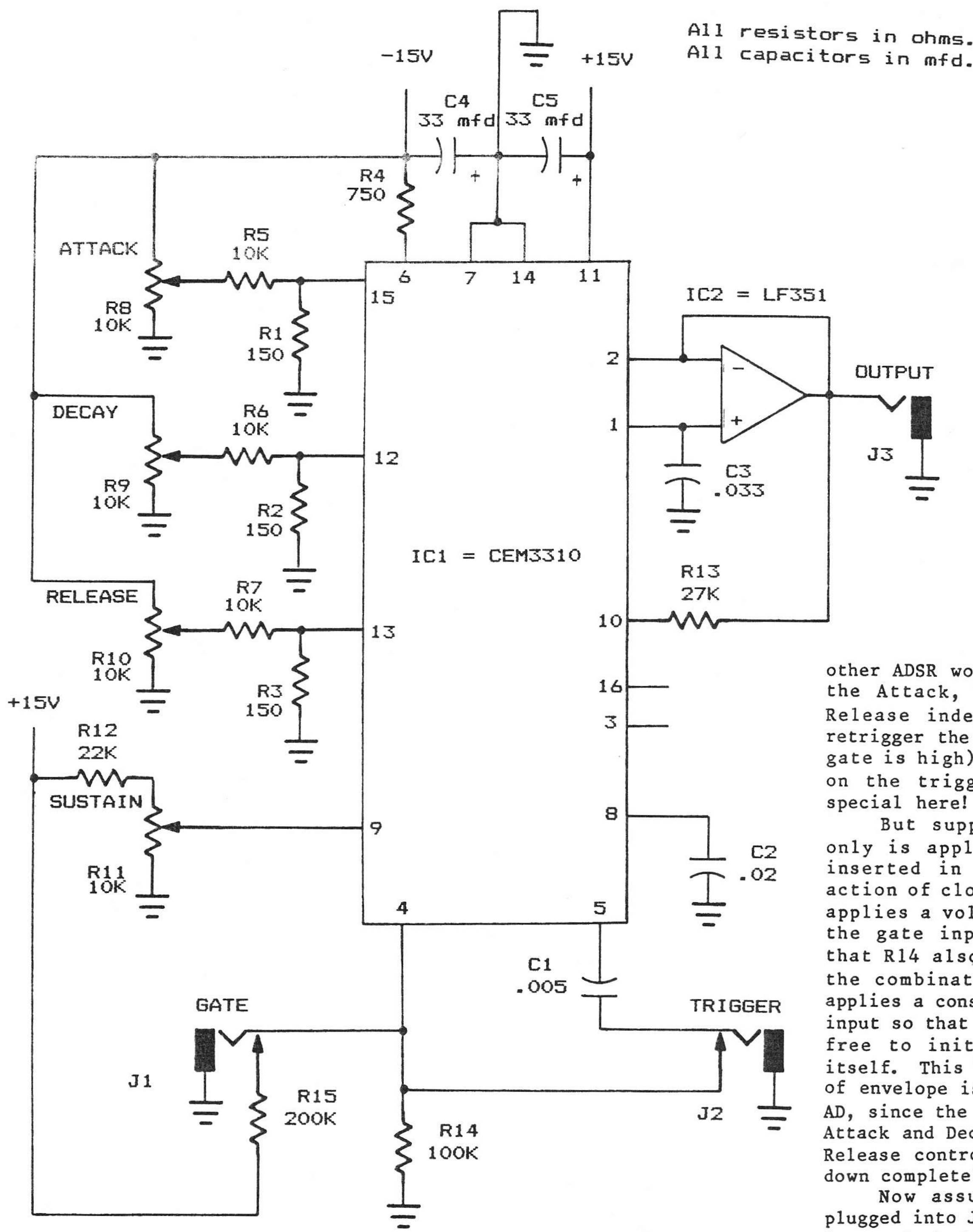
The Sustain control operates in a slightly different fashion. Pin 9 expects to see a positive voltage ranging from 0V to +5V; this voltage is mirrored to the output and sets the output sustain level. R12 drops about 10V from the positive supply, leaving approximately +5V across potentiometer R11.

However, one subtle point that may be of some concern to purists in the crowd is that if the Sustain voltage is greater than the Attack peak voltage, the output waveform will exhibit a "pip" (i.e. the waveform rises to the Attack peak level and then jumps suddenly to the higher Sustain level). Although this "pip" can be quite small (100mV or so), in some applications the resulting distortion might be unacceptable. The spec sheet and application note for the CEM3310* show two ways to correct this problem, but for our purposes a simpler solution is to select R12 so that the maximum voltage across R11 is less than the Attack peak level. According to the spec sheet, the CEM3310 is guaranteed to have an Attack peak voltage between 4.7V and 5.3V. By selecting R12 as 22K, R11 creates a voltage drop of about 4.7V, which is low enough to meet worst-case conditions. This is a quick-and-dirty way to solve the problem, but works quite well and only costs the price of a resistor. Of course this implies that the envelope will only go from 0V to +4.7V (instead of +5V), but for most applications this hardly matters.

Having covered the four input parameters (Attack, Decay, Sustain, and Release), let's take a look at the output. Since the internal buffer of the CEM3310 has a fairly high output impedance, I felt it would be necessary to provide additional buffering through an external op amp. Using the method suggested in the application note, IC2 provides this extra buffering. Notice the unusual configuration in the schematic. The timing capacitor, C3, connects to pin 1 of the CEM3310 and the non-inverting input of IC2. Also, part of the output is tapped off via R13 and heads for pin 10 of IC1. (C3 and R13 are the two principal time constant devices for this circuit and may be experimented with if desired).

Although the main output at J3 is shown as a single jack, I wired five jacks in parallel. Thanks to IC2, we have plenty of drive current available now and it makes sense to provide additional output jacks.

And now is a good time to mention that IC2 must be a FET type op amp. According to the spec sheet, whatever op amp is used here should have a positive current flowing into the input pin and op amps like the LF351 or TL071 meet this condition. Just for the sake of experiment, I tried a 741 op amp and observed that the ADSR locked up on long time settings. So, stick with an LF351 or something similar if you want reliable operation for any setting of the controls.



C2 is not a timing capacitor, but instead compensates an amplifier internal to the CEM3310. Likewise, resistor R4 has nothing to do with the time constant, but is instead a dropping resistor for the internal Zener diode. A value of 750 ohms is recommended for safe operation with a ±15V supply.

How "easy firing" works. Finally, let's examine the input structure and see just what makes this an "easy firing ADSR." J1 and J2 are the gate and trigger inputs, respectively. When both gate and triggers are used, the circuit performs just like any other ADSR would: you can adjust the Attack, Decay, Sustain and Release independently and also retrigger the unit (as long as the gate is high) with another burst on the trigger input. Nothing special here!

But suppose that a trigger only is applied and no jack is inserted in J1. The switching action of closed-circuit jack J1 applies a voltage through R15 to the gate input at pin 4. Note that R14 also ties to pin 4, and the combination of R14 and R15 applies a constant +5V to the gate input so that the trigger input is free to initiate events all by itself. This "trigger-only" type of envelope is commonly called an AD, since the trigger initiates an Attack and Decay. The Sustain and Release controls should be turned down completely.

Now assume that nothing is plugged into J2, and a gate signal is patched through to J1. R15 is now disconnected from the circuit and R14 simply acts as a tie-down resistor; therefore, the input gate signal coupled directly through to pin 4 exactly as it comes from the keyboard or drum box (or whatever!).

Since no jack is inserted, the gate signal is allowed to pass through J2, where it is then differentiated by C1 before hitting

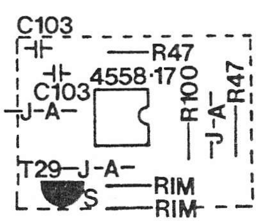
..........continued on page 18

quencies. An additional 27k resistor prevents the clock input from floating when the switch is changing state. At first I was hesitant about installing both octave mods, but I find them both very useful because they produce very different results.

Tapped filter outputs. Another simple yet useful mod is to tap the filter outputs before they are mixed to a mono output (see Figure 5). Sometimes I prefer to take the unfiltered sound and process it through some of my other synthesizers using an envelope follower to create new effects. Therefore, I take the signal directly after the DACs before it reaches the filter block.

The DAC outputs are located on the main circuit board at test point 2 (TP2) and test point 4 (TP4) which are located lower center and lower right, respectively on the pc board. The filter outputs are present at IC 4558-17. To the immediate right of the 4558 is a 100k resistor; the Consonant filter output is present at the end of this resistor closest to you (the "front" of the resistor) if the phone jacks are at the top of the board. The next resistor to the right is a 47k resistor, and the "front" of this resistor is the output of the Vowel Filter block. By attaching wires here and mounting extra jacks on the back panel (there's plenty of room), you're all set. Right next to my jacks I installed two additional single pole switches. One side is wired to the DAC output, the other to the filter output so I can switch between the filter block output and raw DAC output in stereo.

Figure 5



In his bulletin, Robin also presents a coupler circuit which allows you to play two Casios through one keyboard or have your computer trigger the notes. Although I have not implemented it, this would enhance the capabilities of the instrument even more. I hope you find these ideas useful. Happy synthesizing!

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the trigger input at pin 5. As a result, a single gate input supplies both the gate and trigger signals needed by the CEM3310. Although it is possible to generate a full ADSR response this way, the ability to retrigger is lost. For most applications you will want to turn the Decay control down all of the way and the Sustain control up full. The resulting response is known as an AR since it is the Attack and Release which determine the envelope shape.

Summarizing the input structure, then, we've seen how to get a full ADSR response with both a gate and trigger, an AD response with just a trigger, and an AR response with just a gate. With these three options at your disposal you should be able to interface this envelope generator with just about anything including synthesizer keyboards, the Super-Controller, Micro-Drums and many types of computers.

As a quick review of voltage levels, let's note that the power supply is a bipolar 15V, the gate and trigger inputs should be 0V to +5V level and the output is a nominal +5V maximum. Thus, the "Easy Firing ADSR" is compatible with all of the other circuits described in "Practical Circuitry."

Finding parts for this project is easy. PAIA electronics (1020 W. Wilshire, Oklahoma City, OK 73116) stocks the CEM3310 and they'll throw in a spec sheet/application note when you buy one as well. The LF351 can be obtained from DIGI-Key Corporation (P.O. Box 677, Thief River Falls, MN 56701) as well as from several other dealers. All of the other parts listed in the parts list are common and may be obtained from a number of sources.

Since this was such a simple circuit, I built it by hand on a small piece of copper clad prototype board (from Radio Shack) using pieces of ordinary hook-up wire and flea clips. You will want to use a good quality capacitor for C3 since this is the main timing element in the circuit, but the other passive components may be garden variety. Finally, remember to use an LF351 for IC2, and by all means use sockets for both ICs.

I mounted my ADSR behind a standard 1-3/4" by 19" rack panel and used epoxy paint and dry transfer letters to complete the project. Since I had a little extra panel space left over, I added an uncommitted 100K pot with an input and output jack. You might want to add this too, since a spare attenuator can come in handy from time to time.

And that's it! While a very easy project to build, there is no doubt that this is a fine ADSR. Due to advances in integrated circuit technology, small circuits often yield very professional results and this circuit is no exception. Give it a try and see if you don't agree that the Easy Firing ADSR is a great add-on to any system!

Parts List

Resistors

R1-R3	150
R4	750
R5-R7	10K
R8-R11	10K potentiometer
R12	22K
R13	27K
R14	100K
R15	200K

Capacitors

C1	0.005 uF
C2	0.02 uF
C3	0.033 uF poly or mylar
C4,C5	33 uF 16V electrolytic

Semiconductors

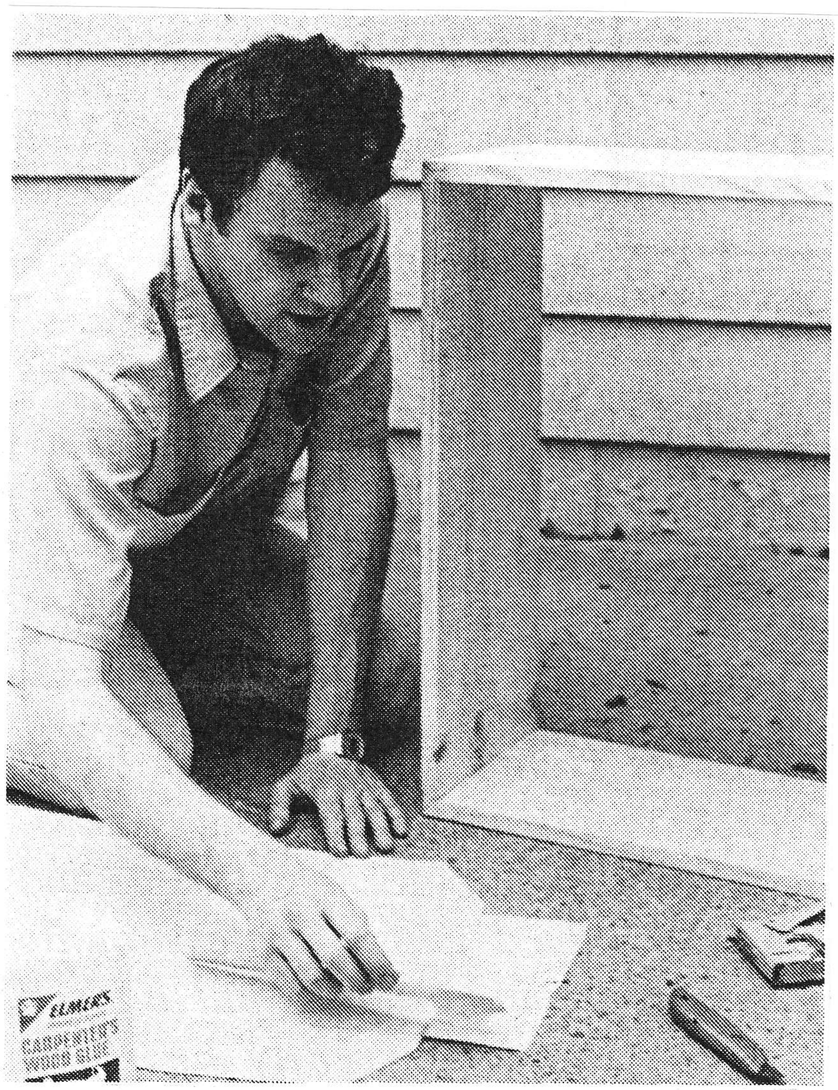
IC1	CEM3310 ADSR IC
IC2	LF351 Bifet op amp

Mechanical parts

J1,J2	Closed circuit 1/4" phone jack
J3	Open circuit 1/4" phone jack
Misc.	Wire, knobs, sockets, etc.

* See **Databank** page 23.

And now for a page from The Scrapbook...



Back in my college days I was too poor to afford a workshop, so I did all of my synth construction on the driveway. Using the equipment outside was a bit dicey, since this was before the advent of ground fault outlets, and my neighbors tended to prefer Tex Ritter music. Note the geek haircut.

Practical Circuitry

BUILD A SYNTHESIZER DELAY LINE

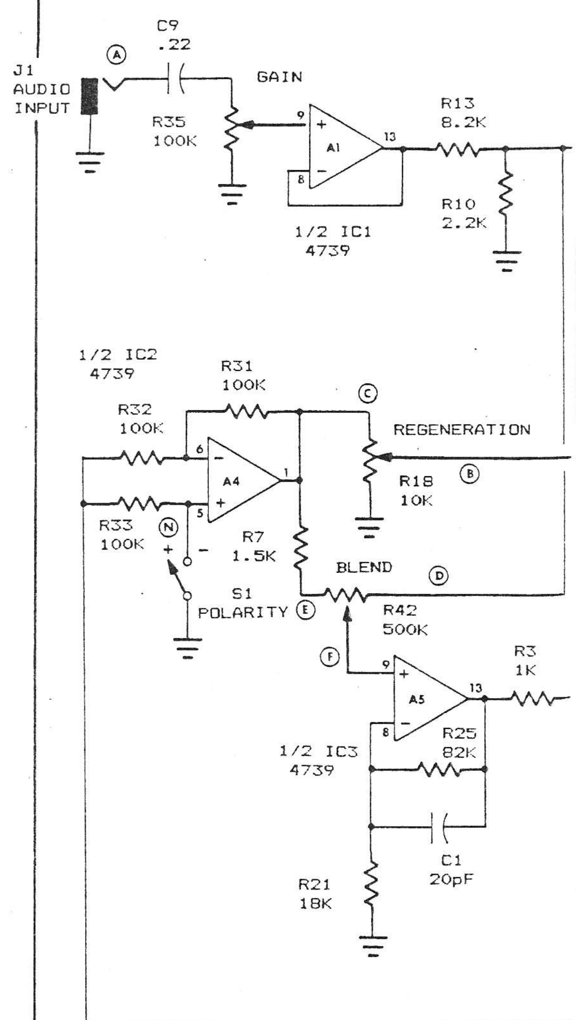
by:Thomas Henry

A lot of people tend to think of flangers and chorus units as being strictly suited to guitars, and not of any real value for synthesizers. After all, a synthesizer has many more control options available and scores of VCFs and VCAs to alter the sound; what more could a flanger possibly add to a synthesizer's already large repertoire of tonal colors? Well, lots! A delay line capable of flanging and chorusing opens up a whole new world of sonic capabilities. Considered as a "timbre modulator", a delay line can transform traditional VCO waveforms into a spectacular spectrum of harmonics that would be impossible to achieve with standard methods. I used to be skeptical about combining flangers and synthesizers, but changed my mind when I heard dozens of new sounds coming forth from this simple homebrew unit.

A delay line for synthesizers. This issue's "Practical Circuitry" describes a delay line which has been especially designed to work with synthesizers. While not offering long enough delays for echo and reverberation, it is suitable for a whole range of flanging, chorusing and vibrato effects. Best of all, it's easy to build and align, and uses commonly available parts to boot. If you've been thinking about getting into analog delay projects, this might be a good place to start since the design is straightforward and relatively goof-proof.

There are basically three differences between this synthesizer delay line and the typical guitar flanger. First, a synthesizer delay line must allow for a variety of generalized control options (as opposed to the single triangle wave generator control found in most guitar-oriented units). Secondly, the assumed signal levels are considerably different (synthesizer oscillators put out far stronger signals than guitar pickups). Finally, noise considerations (all analog delay line chips suffer from some noise problems) may be relaxed somewhat. This is because with a guitar unit, the output always connects to the amplifier. Any noise is always present, so most circuits use compansion or noise gating to reduce noise. With a synthesizer, however, matters are simpler since the last module in the signal chain (before going to the amplifier) is usually a VCA. When no keys are held down, the VCA is cut off (closed) and this gates out the noise. Pressing a key down opens the VCA, but remember that the signal going through the VCA is much greater in amplitude than the background noise. Therefore, in either case background noise is simply not apparent.

Now I'm not going to claim that this delay line is the quietest unit around, but if (as described above) it's inserted into the signal chain before the final VCA in a patch, then it will perform quite admirably and noise will not be a problem. There are



always tradeoffs involved in electronic design, and the factors I tried to balance here were noise and cost. By assuming that the delay line would be followed by a VCA, I was able to trim the cost and complexity of the circuit by a large factor. The result is a delay line which is more than adequate for most synthesis and doesn't cost an arm and a leg. One of the goals I had in mind was making banks of two to four delay lines economically feasible, and the present design satisfies this goal. Using multiple delay lines opens up many possibilities for Shepard function and quadrature function generators.[1,2]

Delay line features. Let's quickly sketch out the salient features of the synthesizer delay line. As with previous projects presented in this column, the flanger follows certain standards. The audio signal is presumed to be 10V peak-to-peak, and the control signal 0V to +5V. The input impedances are 100k and the output impedance is 1k. One standard that isn't obeyed is DC coupling; the extra complexity involved in maintaining DC response throughout didn't seem worthwhile since flanging and chorus effects are generally applied to audio signals only.

The synthesizer delay line incorporates a number of useful options. For example, a continuously variable blend control tailors the mix between straight and delayed sounds. I like this "pan-pot" type of control much more than the individual level mixer controls found on most delay lines; a fringe benefit is that you don't need a separate vibrato mode switch (for vibrato, simply turn the blend control to the full delay position). Another option is the polarity switch. In the positive position, the delayed signal is added to the dry signal; in the negative position, it's subtracted. The two sounds are quite different, with the positive having a more metallic scrape to it and the negative generating a smooth bassy sound. Best of all, unlike many other designs that require a DPDT switch, the polarity switching requires only an ordinary SPST switch. Finally, two LED indicators keep a tab on any possible signal conditions which could overload the delay line.

All in all, while this is a simple delay line circuit, it nonetheless incorporates the kind of features that make using it a breeze. Now let's take a look at the schematic and see how it works.

How it works. Refer to

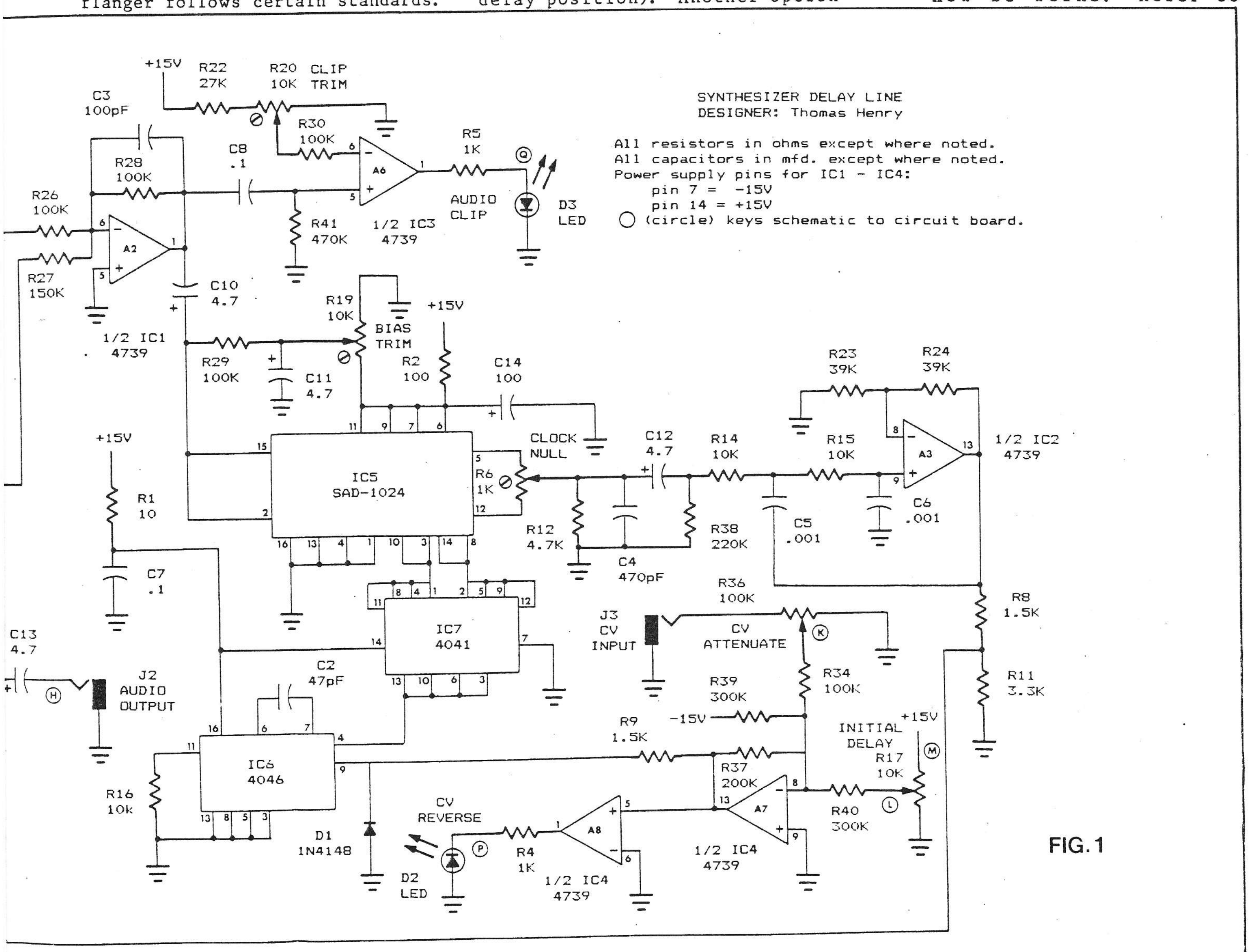


FIG. 1

Figure 1. The audio input enters the circuit via J1, then AC-couples through C9 to attenuator R35. A1 buffers the signal, which is then attenuated by a factor of ten by divider R13 - R10. (The SAD-1024 distortion increases rapidly with signal levels above 1V peak-to-peak.) The signal next goes to regeneration mixer A2, where C3 limits the high end to about 16 kHz. Restricting the response in this way keeps high frequency energy out of the SAD-1024, thereby minimizing foldover distortion.

At this point the signal splits into two paths. One path goes to the clip detector formed around comparator A6 and associated components. Adjust trimmer R20 so that the LED comes on when the SAD-1024's input signal exceeds 1V peak-to-peak. This not only lets you monitor potentially harmful conditions, but also, gives some indication of the onset of clipping. While this is a quick-and-dirty design for the clip detector, it does work!

The other path couples to the SAD-1024 via electrolytic capacitor C10. Since the delay chip requires a DC bias to properly pass a signal, R19, C11 and R29 provide an adjustable bias voltage. By the way, the delay chip is wired in the parallel-multiplex configuration, which tends to give the cleanest results when only short delays are needed. R2 and C14 serve to decouple the SAD-1024 from the rest of the circuitry.

After passing through the delay stages, the signal is applied differentially to trimmer R6, which can then be used to null out any clock glitches. The reconstructed delayed signal develops a voltage across R12; C12 then couples the signal to a second order Chebyshev low pass filter with a cutoff of about 14 kHz (the filter comprises A3 and its associated components). This type of filter has a steep initial rolloff at the expense of a slight peak in the pass band. In this application, a fast rolloff is crucial for good fidelity while any small response peaks really don't matter -- in fact, this can add some more complexity to the timbre. Bernie Hutchins' excellent manual, "Laboratory Problems and Examples in Active, Voltage-Controlled And Delay Line Networks" made the design of this filter a snap.[3]

The low pass filter serves to smooth the delayed signal from a stairstep to a more linear form, and at the same time, removes any residual clock garbage. Since the filter has a gain of about 2, resistor divider R8-R11 chops the signal back down to size, whereupon it goes to the polarity switcher (composed of A4 and related components). With S1 open, the signal is left unchanged; with S1 closed, the signal is inverted. Using this particular signal inversion circuit makes it possible to get by with a single SPST switch and at the same time keep the audio signal close to the circuit board. Note that the signal is inverted or not inverted to both the output blend control and to the regeneration loop; this is essential in order to obtain the full benefits of phase switching. (Incidentally, you may recognize this polarity changer since it appeared in the "Quadrature Function Generator" circuit.2)

The audio signal then heads off to one side of blend control R42, which allows you to dial in the desired amount of delayed signal. One extreme of the pot's throw gives full dry signal, the other extreme gives full delayed signal, and in-between settings blend the two signals to varying degrees. This versatile structure is described in "The Musical Engineer's Handbook."[4] Note that the signal coming from A4 can also be fed back into the audio path (thereby producing a more intense effect) via regeneration control R18.

The final mixed signal is brought up to full synthesizer level (10V peak-to-peak) by amplifier A5. C1 stabilizes the output by rolling off the response above 16 kHz. Finally, R3 and C13 present a nominal 1K output impedance.

We've hit all of the major points concerning the audio path; now let's switch gears and examine the control voltage path. R17, the initial delay control, lets you dial in a specific fixed amount of delay. However, you can also control the delay by injecting a variable control signal from other synthesizer modules (such as LFOs and ADSRs) into J3 and R36. A7 sums the initial delay and the variable control signal and applies the result to voltage controlled clock IC6. Note that IC6, which is a CMOS phase locked loop chip, operates from the +15V supply only and can be damaged if negative voltages are applied to its control pin. D1 shunts any reverse current to ground, and R9 limits the current flow to a safe level for D1.

In addition to safety considerations, it's useful to know if the control pin "sees" a reverse voltage since the control voltage will have no effect under these conditions. Comparator A8 turns on the "CV reverse" LED when this happens; adjust initial delay control R17, or CV attenuator R36, to get into a more appropriate region of the clock's operating range.

As mentioned earlier IC6 is a phase locked loop, but we are only using the VCO section of the chip. Pin 9 is the control voltage input, while pin 4 is the logic level output. R16 and C2 set the basic operating range. With respect to driving the SAD-1024 clock inputs, there are two major requirements: First, the SAD-1024 inputs want complementary waveforms so that when one clock pulse is high, the other is low. Second, the clock inputs can absorb lots of current since they have a rather stiff input capacitance (typically 100 pF per input). To drive this capacitive load, we use TTL buffer IC7 to provide a biphase clock output with plenty of "oomph". If you've never seen this chip before, you might want to look it up in the "CMOS Cookbook"[5] since it is a strange one -- it looks like CMOS on the input and TTL on the output! Of course, a TTL output has plenty of drive, and so keeps the clock signals going to IC5 nice and clean. (However, note that unlike standard TTL you can parallel the 4041's outputs together for even more current drive.)

Building the Synthesizer Delay Line. Building the synthesizer delay line is quite straightforward, but only if you employ good construction techniques. Since this circuit has not only audio but some hefty radio frequency signals floating around, it is essential that you pay close attention to the circuit board layout. In the final analysis, a printed circuit board is really the only way to go, since this minimizes the interaction between the various critical sections. **Figure 2** shows a tested circuit board layout for the synthesizer delay line, while **Figure 3** gives the parts placement guide.

As anyone who has ever designed an analog delay line will

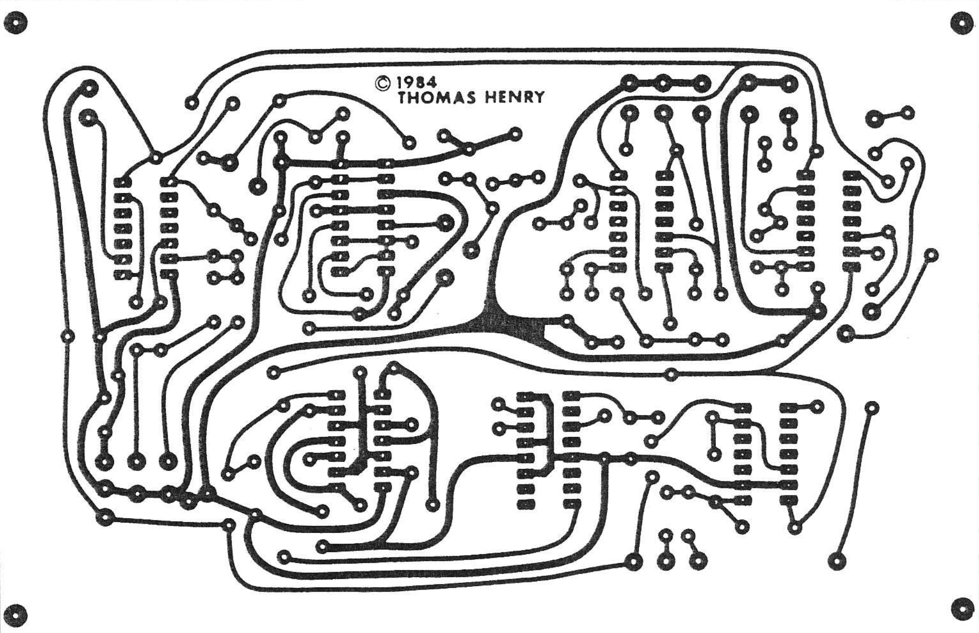


FIG. 2 PO BOARD FOIL PATTERN (not actual size)

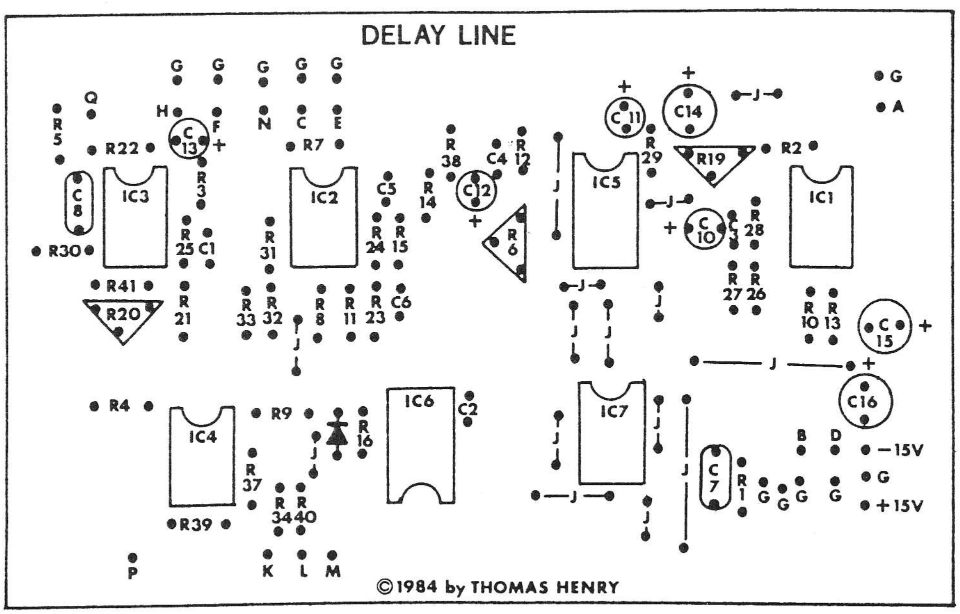
attest, the layout of the circuit board is no trivial matter. This board employs several exotic techniques to minimize hum and clock noise feedthrough. For example, each major sub-section of the circuit must have its own path back to ground (preferably through a large trace). Similarly, the clock outputs from the 4041 must go through hefty traces to the delay chip, since this minimizes the effects of inductance and resistance. Finally, attention must be paid to the actual parts placement and op amp configuration. This design uses dual op-amps and situates them well away from any clock circuitry. Because of these unusual layout considerations, the circuit board may look a bit strange with its redundant ground lines and different size traces, but believe me, they're all necessary for good operation!

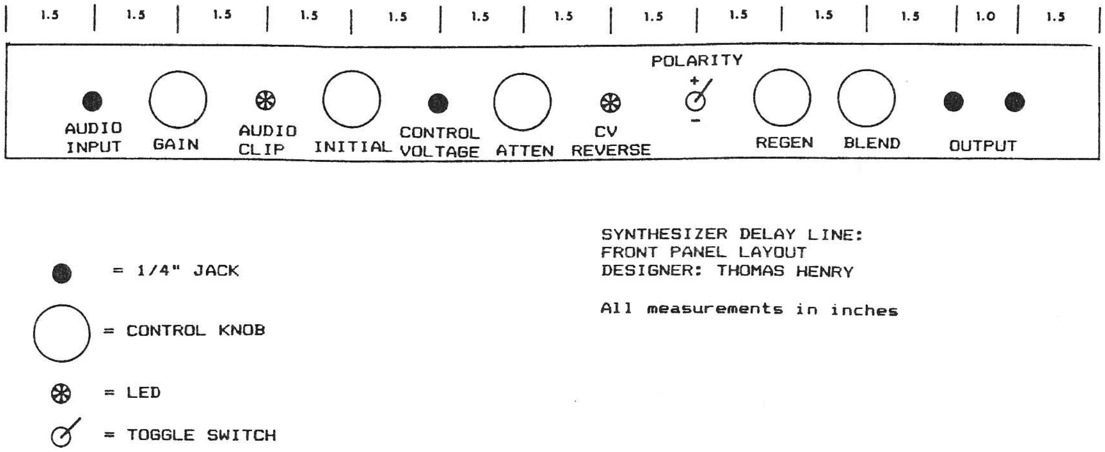
We're all set to stuff the circuit board now, so refer to the parts list at the end of this article. Note the special requirements on capacitors and pots; if a mylar cap is specified, then use one! If no special mention is made, then you may use whatever you have handy. Load the board, using **Figures 1** and **3** as a guide. Notice that C9, the input capacitor, doesn't mount on the board, but is wired between J1 and R35 right on the front panel. Also, 100 uF decoupling capacitors C15 and C16 aren't shown in the schematic for clarity, but should nonetheless be installed on the circuit board. Be certain to observe the polarity of these and other electrolytic capacitors. Finally, use sockets for all the ICs, since the SAD-1024 and CMOS chips can be damaged by static electricity. Install the ICs in their sockets only when all soldering is completed and do not handle them unnecessarily.

After loading the board, prepare a suitable front panel. **Figure 4** shows the layout I used in my prototype. Everything fits comfortably behind a single width rack panel (1-3/4" by 19") and there was enough room left over to mount two paralleled output jacks. The circuit board mounts on little angle brackets behind the front panel, and #4 hardware secures everything in place.

At this point you can complete the final wiring to the panel. Again, using the parts placement guide in **Figure 3**, connect the various wires as needed. Since delay circuits are susceptible to hum pickup, shielded wire must be used for the connections to points A, B, C, D, E, F, H and N. There is a ground contact for the braid next to each of these lettered locations on the board; here's how to use it. Strip one end of a shielded cable, exposing the braid and the hot wire. Tin both of these, then solder the hot wire into hole A and the shield into the ground hole (lettered G) next to this point. Run this cable to input pot R35 and connect the hot wire to the appropriate lug on the pot. The shield wire is left floating at the front panel. Complete all of the other hookups in the same way. Again, the eight connections mentioned above <u>must</u> use shielded cable for hookup to the front panel. The braided shield wire connects to the circuit board only; the panel side of the shield is left floating. This prevents any ground

FIG. 3 COMPONENT LAYOUT





SYNTHESIZER DELAY LINE:
FRONT PANEL LAYOUT
DESIGNER: THOMAS HENRY

All measurements in inches

FIG. 4

loops, which can be a nasty source of hum. One exception here is the wire coming from point N; the shield hooks up to one terminal of the polarity switch while the hot wire hooks up to the other. Incidentally, Radio Shack makes a good shielded cable which is perfect for front panel work since it is barely larger than normal hookup wire and very flexible as well.

The wiring to the other controls is non-critical, so use ordinary stranded hookup wire. As a final step, run a single ground wire from the circuit board to the front panel. I ran mine to the ground lug of the output jack, as all of the ground connections to the other controls and jacks ended at this point. Thus, the front panel has a single ground connection in common with the circuit board, which minimizes ground loop and instability problems. To complete the construction, use some ties to bunch the cables and wires together into a neat harness.

We're now ready to power up the circuit. Your supply must be well-regulated and operating within its power limits (the circuit draws about -15 mA and +50 mA). It's also best if the wires running from the supply to the circuit board are short and direct. On my system, this wasn't possible; the result was hum problems, which could only be eliminated by sticking a 100 Ohm resistor in series with the +15V line just as it entered the circuit board. Short, hefty wires should prevent any problems, but if you do experience hum, try the extra resistor trick just mentioned.

Calibrating the synthesizer delay line. Tweaking the synthesizer delay line is fairly easy. Begin by double checking your work and when you are satisfied, apply power to the unit. Confirm that it doesn't belch smoke or flames! Next, turn gain control R35 up fully, regeneration control R18 down fully, blend control R42 to full delay signal and the initial delay control R17 to about a 1/3 position. Inject a 10V peak-to-peak 1 kHz sine wave into J1 and monitor J2 with an oscilloscope. Adjust the bias trim, R19, for minimum sine wave distortion. Adjusting R19 takes a little patience, since if you trim too far, the waveform will go flat on one side and not going far enough flattens it on the other side. Also, C11 adds a time constant, so turn the control slowly so that the waveform "catches up" with the trim adjustment. If you can't obtain a clean waveform for any setting of R19, decrease the input signal amplitude a bit and try again. After a few minutes of tweaking, you should have a fairly clean replica of the input. Note that this adjustment requires somewhat of a compromise as the SAD-1024's bias requirements change with clock frequency. If you wish, trim R19 to give the best possible waveform for various settings of R17.

Next clip the scope probe to the wiper of clock null trim R6, and set the timebase of the scope up fairly high (enough to see a cycle or so of the sine wave). As you vary R6 from its midpoint, you will see two waveforms. Adjust R6 so that the two signals converge.

Finally adjust the clip trim so that the audio clip indicator comes on as you slightly increase the 10V peak-to-peak signal to, say, 12V peak-to-peak. This provides visual indication that you are exceeding the nominal 10V peak-to-peak input requirements and may be getting into clipping.

If needed, you can also calibrate the unit without a scope. Instead of monitoring the output with an oscilloscope, patch in an audio amplifier. Adjust the bias trim for the least amount of distortion, as perceived by your ears. R6 isn't all that critical and may be just left at mid-rotation. And of course, you can adjust R20 simply by viewing the LED. So don't let the lack of an oscilloscope stop you from building this circuit!

After calibration, play with the delay line for a week or so to allow the components to break themselves in, then give the unit one final tweak. This should be the last adjustment you'll have to give it for quite a while.

Using the synthesizer delay line. If all has gone well, you now have a synthesizer delay line up and running. Since there are a lot of controls, be sure to really get familiar with them before assuming anything is wrong. For example, the control voltage input and the initial delay time give much more range than you'll be able to use in most situations, so be sure to tone these down if you're getting weird results. Also, the initial delay pot has a lot of "oomph" in case you're modulating the clock with a bipolar signal and need to offset the negative portions of the waveform. Similarly, the control voltage input has a certain amount of gain, so pull back on R36 if the clock is coming down into the audio range. Remember, this ability to handle a wide variety of control signals means that you can create some ugly sounds if you're not careful, so experiment.

Don't forget to play with the blend, regeneration and polarity controls as well. With a little practice you should be able to get some very nice flanging, chorusing and vibrato effects. If the audio clip indicator comes on at some settings of the regeneration control, reduce either input attenuator R35 or regeneration control R18.

Are you interested in experiencing "hypertriangular" flanging? Nothing could be simpler, providing that you have a wide-range, voltage-controlled, triangle wave generator. Modulate the delay line with a low frequency triangle wave from a VCO (like my VCO Deluxe circuit previously published in **Polyphony**), taking care to adjust R36 and R17 so that the clock is in the proper range.[6] Now feed some of the triangle wave back into the attenuated exponential control input of the VCO, which transforms the triangle wave into a "hypertriangular" waveshape. Adjust the exponential voltage attenuator on the VCO as needed to bring in the hypertriangular effect.

For other exotic sounds, put the synthesizer delay line under control of an envelope generator. This is especially neat when used with the "Snare+ Drum Voice".[7] Or how about using the Shepard function generator with multiple delay lines for some truly spacey effects?[1] Don't delay using delays any more!

Acknowledgment: Many thanks go out to Craig Anderton whose several excellent analog delay line circuits provided ideas which simplified the design of this one. Craig is truly a delay line guru, so refer to his articles on this subject for the real dope on getting these sometimes unruly chips to do remarkable things! (8), (9), (10), (11)

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(1) John S. Simonton, Jr., "**Lab Notes:** Shepard Functions", **Polyphony**, February 1983, pp. 42-46.

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(4) ______________, "Musical Engineer's Handbook", pp. 3a-4 through 3a-5. (see address above).

(5) Don Lancaster, "CMOS Cookbook", (Indianapolis: Howard W. Sams and Co., 1977), p. 99.

(6) Thomas Henry, "VCO Deluxe", **Polyphony**, Nov./Dec. 1981, pp. 28, 29, 30.

(7) ______________, "Snare+ Drum Voice", **Polyphony**, Sept./Oct. 1982, pp. 28-31.

(8) Craig Anderton, "Pedal Flanger", **Guitar Player**, February 1980, p. 123; March 1980, p. 124; April 1980, p. 116.

(9) ______________,"The Flanger", **DEVICE**, Volume 1, Number 9, pp. 1-6.

(10) ______________,"Building the Hyperflange + Chorus", **Modern Recording**, July 1983, pp. 36-40 and September 1983, 56-60.

(11) ______________, "Build a Chorus Delay", **Guitar Player**, January 1982, pp. 26-30, 33, 34 and February 1982, p. 116.

PARTS LIST

Resistors

R1	10 Ohms
R2	100 Ohms
R3 - R5	1k
R6	1k trimpot
R7 - R9	1.5k
R10	2.2k
R11	3.3k
R12	4.7k
R13	8.2k
R14 - R16	10k
R17, R18	10k audio pot
R19, R20	10k trimpot
R21	18k
R22	27k
R23, R24	39k
R25	82k
R26, R28-R34	100k
R27	150k
R35, R36	100k audio pot
R37	200k
R38	220k
R39, R40	300k
R41	470k
R42	500k linear pot

Capacitors

C1	20 pF
C2	47 pF polystyrene
C3	100 pF polystyrene
C4	470 pF
C5, C6	0.001 uF mylar
C7, C8	0.1 uF
C9	0.22 uF mylar
C10 - C13	4.7 uF electrolytic
C14 - C16	100 uF electrolytic

Semiconductors

D1	1N4148 (or equiv.)
D2, D3	LED
IC1 - IC4	4739 dual op amp
IC5	SAD-1024 delay chip
IC6	4046 CMOS PLL
IC7	4041 TTL buffer

Mechanical parts

J1 - J3	1/4" phone jack
S1	SPST mini-toggle
Misc.	Wire, knobs, sockets, solder, LED clips, etc.

Note: C9 mounts on the front panel between the hot side of R35 and J1. C15 and C16 decouple the +15V and -15V supply respectively, where these lines first enter the board. Although they aren't shown on the schematic, provision has been made for them on the circuit board.

Practical Circuitry

The HI-HAT & PERCUSSIVE VOICE

by:Thomas Henry

I hope your soldering iron is hot and you're all set to start building, because this time in "Practical Circuitry" we're going to develop a new circuit which is sure to appeal to lots of users! You will recall that here in the pages of **Polyphony** we've already seen how to synthesize fairly nice snare drum and bass drum sounds (see my "Snare Plus Drum Voice," September/October 1982, pp. 28-31 and Craig Anderton's "Build the Hip Bass Drum," October 1983, for full details). This time we're going to wrap up the drum kit with the addition of a hi-hat synthesizing circuit. In fact, this circuit does quite a bit more than just hi-hat type sounds, so I have dubbed it the "Hi-Hat Plus"! If you're looking for a new drum sound, check this one out; it creates a number of sounds unattainable with standard drum voices.

Now I'm not going to claim that the Hi-Hat Plus exactly duplicates the sound of a standard hi-hat, but it does suggest the sound more closely than any other analog circuit that I've heard. It does this by making available an "open" and a "closed" sound, tunable "clank" and several other exotic features. It can be triggered by a computer output but is equally usable with a manual triggering system. A foot pedal can "open" and "close" the "cymbals" just like a real hi-hat. If I haven't enticed you sufficiently, read on and see what else this unusual drum circuit has to offer.

To fully understand how the Hi-Hat Plus works, we first need to get some terminology out of the way. Not being a drummer, I just made up the words but they should be descriptive enough to convey the ideas. A true hi-hat generates at least three distinct sounds. The first is the sound of the stick hitting the surface of the cymbal; I call this the "impact" tone. The metallic chime of the cymbals (as opposed to the sound of a drum body, for example) follows immediately. We'll call this parameter "clank". Finally there is the sound of the two cymbals beating against each other and this will be called the "clatter." Obviously the clatter will sound different depending on whether the cymbals are open or shut tightly against each other. To distinguish between these two cases, we'll refer to the clatter sounds as "open" and "closed." With these notions under control, set's see what it takes to generate a hi-hat sound electronically.

Figure 1 shows a block diagram for the Hi-Hat Plus circuit. In general terms, note that there are three sound sources; one creates the impact, another the clank and the last generates the clatter sound. The three sounds feed the master VCA, which is modulated by one of two possible envelopes. The envelope select logic determines whether the envelope should be that of the open sound or closed sound. It selects the proper envelope generator by means of an open trigger, closed trigger or by detecting what the foot pedal is doing at present. If you're playing the circuit in realtime, for instance, you would feed a series of triggers to the open trigger input, and then depress the pedal (which is nothing more than an SPST footswitch) to select the closed envelope generator. Alternatively, a computer could send either open or closed triggers to the unit and these would automatically select the proper envelope.

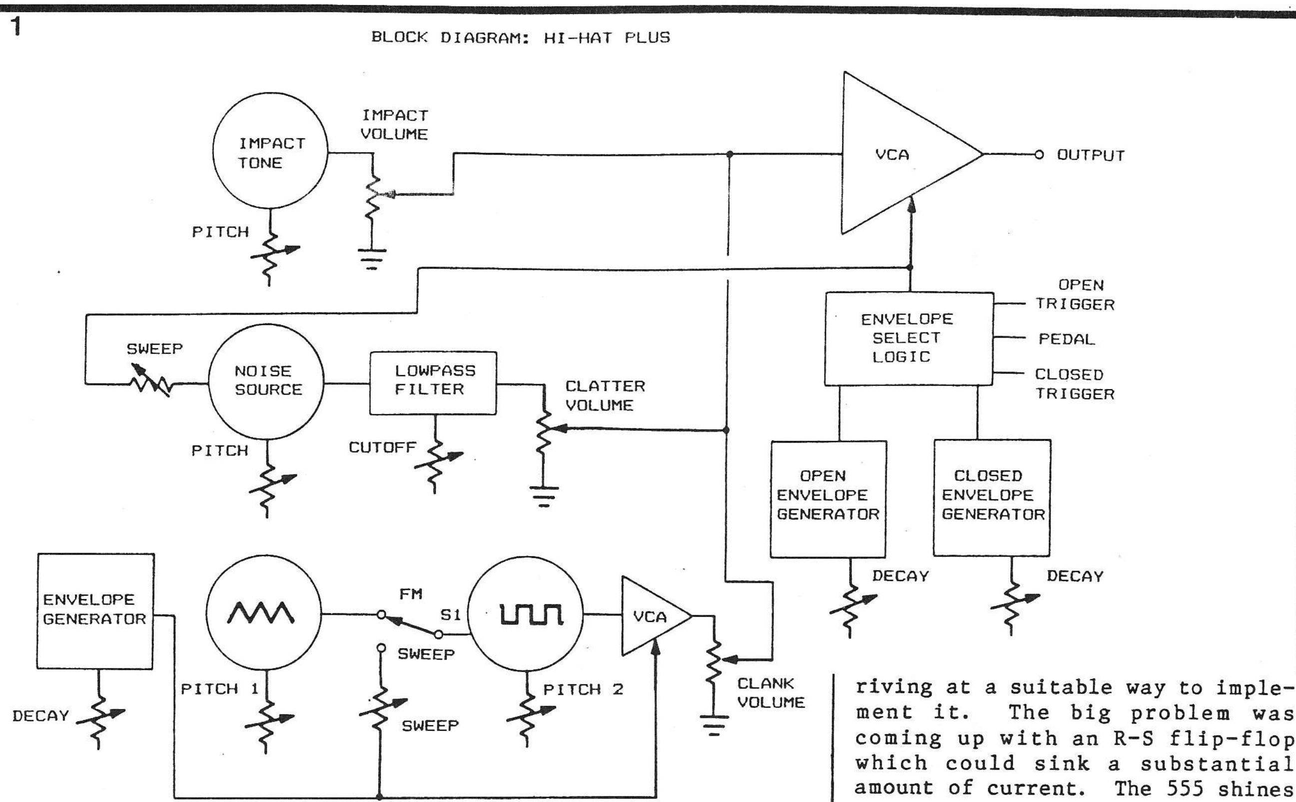
Let's back up a bit and see how the three sounds are generated. The impact tone generator is identical to the one used in the "Snare Plus Drum Voice" (see reference above). Two parameters are available for twiddling, the pitch and the volume. By adjusting the impact pitch control it is possible to go from dull thuds to bright snaps.

The clatter is synthesized with white noise generated by a pseudo-random noise generator. The pitch control sets the clock rate and hence the basic center frequency of the noise, while the sweep control adjusts the depth of envelope modulation. The effect is similar to a lowpass filter closing down, and in terms of a real hi-hat, the sound is not unlike that created by shutting the two cymbals against one another. The clatter generator is followed by a manually adjustable lowpass filter and volume control.

The clank is created by means of two VCOs, with one frequency modulating the other (FM). The sound thus generated is indeed quite metallic in nature. The pitch 1 control adjusts the frequency of the triangle wave generator, and the output of this device modulates the square wave generator. The pitch 2 control sets the center frequency of the square wave. By experimenting with the pitch 1 and pitch 2 controls it is possible to create a variety of clangorous sounds, from the tinkle of thin shards to the dull roar of boinging sheets of metal. As a bonus, S1 lets you switch the modulation so that the square wave is controlled by an envelope generator, thus creating unearthly upward sweeping sounds. The sweep control sets the strength of this effect. Finally, notice how the clank sound has its own envelope generator. This allows for the more realistic effect of the clank dying away before the clatter (as in a real hi-hat).

If the block diagram makes sense to you, then it's time to move on to the actual schematic. Since this is a big circuit, there simply isn't space to discuss every little detail. However, by referring back to the block diagram for the "big picture," you

FIGURE 1



should be able to keep the details in their proper place. Also, you might want to check out "The Snare Plus Drum Voice" article mentioned earlier, as this sheds light on a number of the ideas employed here.

Refer to the schematic in **Figure 2.** IC5, which is our old friend the 3080 transconductance op-amp, is pressed into service as the master VCA. Note how three lines feed into this chip via R41, R42 and R43; these lines come from the clank, clatter and impact tone generators, respectively. Three pots, R27, R28 and R29, allow for setting the desired ratios of these sounds before the mix is amplitude modulated by the master VCA.

The VCA is controlled by one of two envelopes chosen by the envelope select logic. Let's see how this works. Jacks J1 and J2 send open and close triggers to the unit. Op-amps A1 and A2 shape these up into standard 1 millisecond pulses which are then fed to the logic circuitry consisting of the NOR gates and IC3, the 555 timer. Now before you say, "I've seen this all before," you might want to note that IC3 is not being used as a timer! In this configuration, it is set up as high power R-S flip-flop. Pin 2 is the set input while pin 6 provides the reset function. Pin 3 is the output. Depending on the state of the output, either D3 or D4 (but not both) is pulled to ground, thus providing a discharge path for C11 through either R50 (the open decay control) or R51 (the closed decay control). Pulling this all together then, an open or closed trigger either sets or resets the R-S flip-flop, and its output selects one of two possible discharge paths. Hence, we now have the means to create the open and closed sounds. By the way, it should be obvious that the attack is fixed, and is created by dumping a charge onto the timing cap, C11, via diode D5. Notice that D5 passes current when either the open or closed triggers occur. This gives an instantaneous attack; it is the decay which is adjustable by R50 or R51.

Pin 4 of the 555 acts as a master reset control. It will override whatever the chip is currently doing and pull the output to ground, thus closing the hi-hat. An ordinary SPST footswitch can be plugged into jack J3 for pedal control.

By the way, two LEDs give an indication of which decay control is currently selected. D9 lights when the open decay control is in effect, while D10 indicates that the closed decay control is in operation.

I really puzzled over this envelope select scheme before arriving at a suitable way to implement it. The big problem was coming up with an R-S flip-flop which could sink a substantial amount of current. The 555 shines in this respect and at the same time provided all of the niceties like a master reset (pin 4) and an auxiliary output (pin 7) for the LED.

As mentioned above, the currently selected envelope is developed across C11, and the emitter follower made up of Q5 and associated components buffers the signal. This envelope voltage is converted to a current by Q7 which then modulates the master VCA.

Let's now look at the sound sources. The impact tone is generated by the VCO and envelope generator within IC6, the SN76477. The method in which this is done is identical to that employed by "The Snare Plus Drum Voice," so not much more need be said about it (see above). R30 provides control of the impact pitch.

The clatter is created by the pseudo-random noise source within IC6. Normally a resistor from pin 4 to ground sets the basic operating range and hence the "color" of the sound, but to provide dynamic control over this parameter, Q6 is set up as a variable resistor. Notice that the base of Q6 is modulated by the envelope generator via R59. This, then, creates the sweeping sound characteristic of cymbals dying away. R55 sets the center pitch of the noise while R56 lets you dial in varying amounts of sweep. The output of the noise generator then goes to a simple one pole filter within IC6, and R57 sets the cut-off frequency. The clatter sound is finally tapped off of C1, at pin 6, and buffered by A4 before being sent to the clatter volume control, R28.

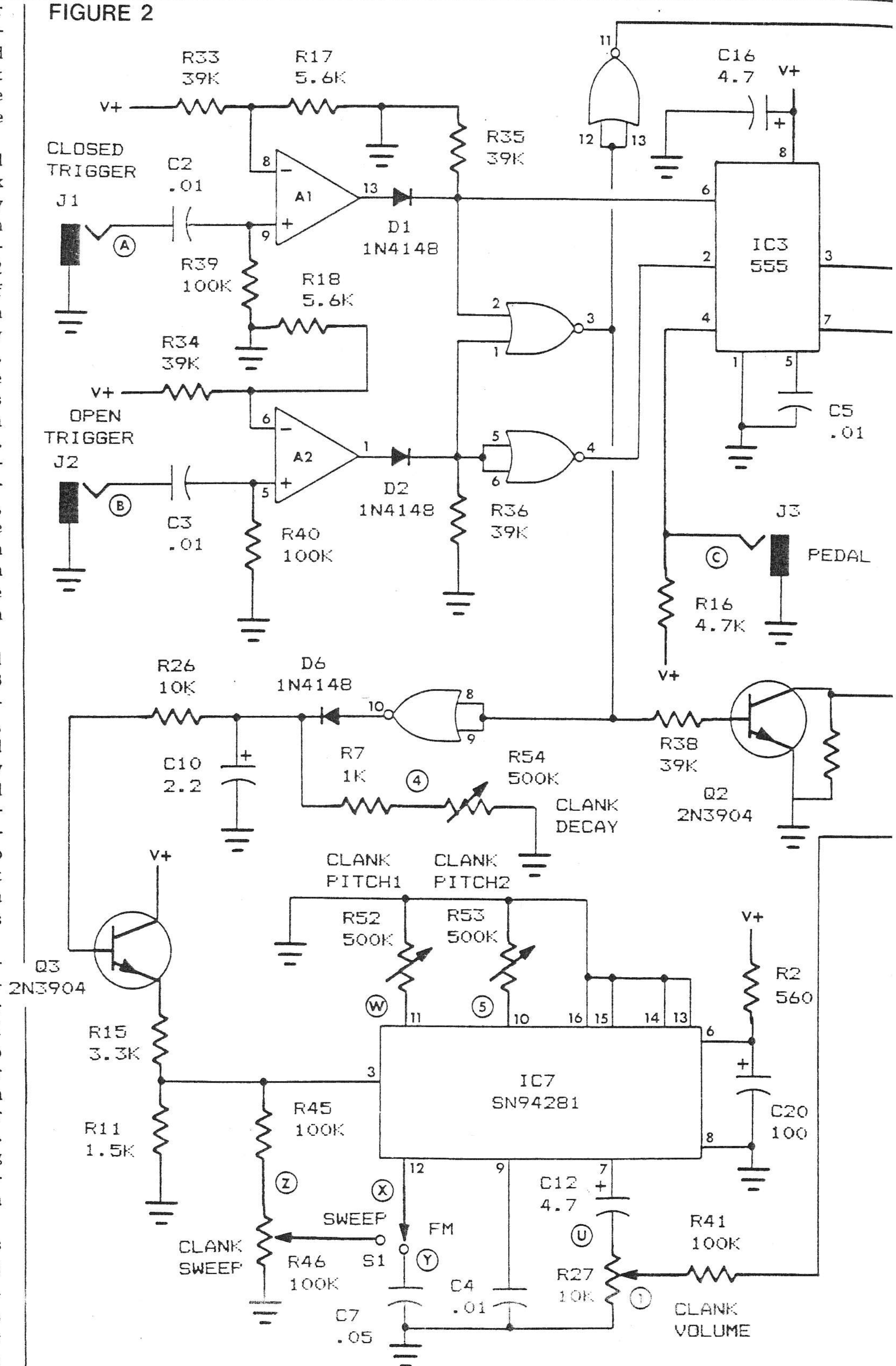
The clank sound is generated by IC7, a simpler type of complex noise chip. This unit has many characteristics in common with IC6, but is available in an easier-to-use 16 pin package. R52 allows for frequency control of the triangle wave generator within IC7, while R53 sets the frequency of the square wave generator. With S1 in the FM position, the triangle wave frequency modulates the square wave, thus generating a very convincing metallic clank. If S1 is thrown to the sweep position, however, the incoming envelope modulates the VCO frequency, creating an upward sweep. The depth of the sweep can be set with R46. Notice that in this position of S1, R52 has no effect since the triangle wave generator has been disabled.

IC7 contains its own VCA, and a voltage of 0V to +3.5V at pin 3 controls the gain. A simple envelope generator comprising D6, R7, R54 and C10 creates the desired envelope and this is buffered by Q3. The final signal is chopped down by R15 and R11 to the required range and this then modulates the VCA within IC6 (and also creates the clank sweep effect mentioned above). All in all, a very simple affair -- but it does work quite well.

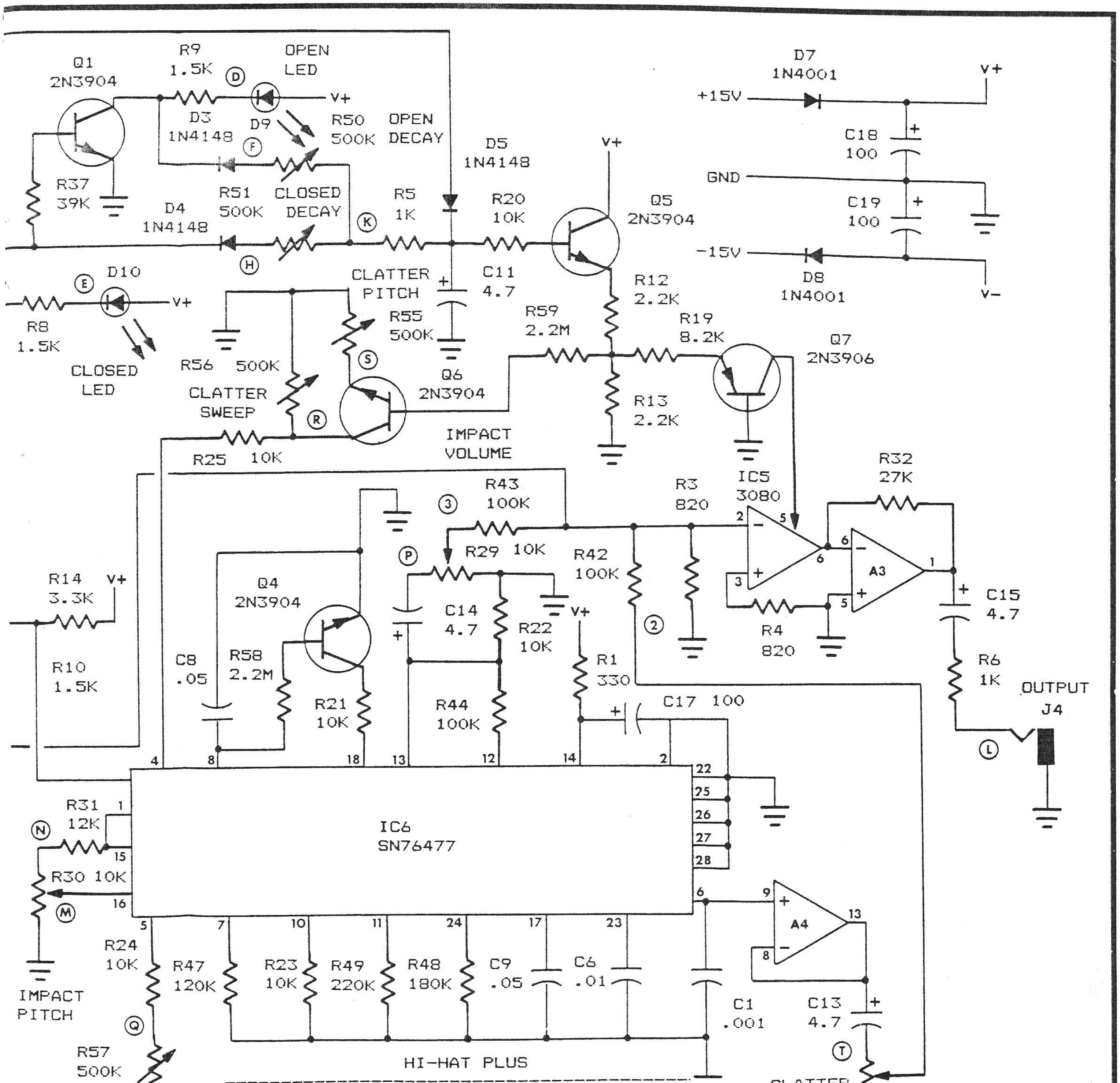
Both IC6 and IC7 require non-standard supply voltages, but fortunately these chips contain their own internal Zener diodes. R1 drops the supply accordingly to IC6, while R2 performs a similar role for IC7. Hence, we have been able to retain our normal bipolar power supply voltage of ±15 Volts. Since there is a lot of switching and noise going on in this circuit, C17 through C20 provide a hefty amount of decoupling.

Well, this pretty much covers the operation of the Hi-Hat Plus; I'll leave you to ponder the details. Now, however, we need to quickly cover how to actually build the thing. With a circuit this big, the easiest way to go is with a printed circuit board. To simplify the task of whipping a board up, **Figure 3** shows a tested circuit board design, while **Figure 4** presents the parts placement guide. **Figure 5** shows the transistor orientation assumed by the circuit board; be sure to check that your transistors obey this configuration.

FIGURE 2



Here are a few tips to guide you in the task of building the Hi-Hat Plus. First, when loading the circuit board, be sure to note the polarity of all of those diodes and likewise, watch the electrolytic capacitors. Use sockets for the ICs, being sure to



HI-HAT PLUS

All resistors in ohms except where noted.
All capacitors in mfd except where noted.
All NOR gates are 1/4 IC4, 4001.
All op-amps are 1/2 4739.
◯ (circle) keys schematic to circuit board.

note the proper orientation of pin 1 in all cases. The circuit board requires some jumpers (denoted by J); use excess resistor clippings for these. Finally, be certain that you have correctly installed the transistors, and have noted that Q1 through Q6 are all NPNs while only Q7 is a PNP.

After loading the circuit board, prepare a suitable front panel and secure the board to it with small angles and #4 hardware. **Figure 6** shows a suggested panel layout using a standard 3-1/2" by 19" rack panel. Complete the wiring to the panel and note that the circled letters on the schematic key the circuit board for this operation. Notice that a pad labeled V+ is available for running to the two LEDs. Also, one subtle point is that R30, the impact pitch control, should be wired in "reverse". This is because the VCO within the SN76477 gives decreasing frequency output for increasing control voltage input. Therefore, while looking at the back of the pot with the terminals facing downward, the leftmost terminal is grounded while the rightmost is hot.

Concerning availability of the SN76477 and the SN94281, you'll be glad to know that both of these are available from Radio Shack, thus putting them within reach of just about everyone. All of the other parts are common and may be obtained from a variety of mail order houses.

For the final hookup, apply +15V, -15V and ground to the appropriate pads on the circuit board (ground is denoted by "G"). After checking for wiring errors, power the circuit up and feed it some triggers. If all has gone well, you're on the air!

I'll turn you loose to play with the unit, but before doing so, let me give you a bit of warning. This circuit has a large number of parameters (translate: knobs), and as a consequence will take some practice to master. You'll find that it is very easy to create some hideous effects by improperly adjusting the controls. Your goal, then, is to find the good effects and keep a log of your results. Like any musical instrument, practice is the key! After you have determined the basic setting which gives a convincing hi-hat sound, let your mind and ears wander through the fourteen (count them!) controls and look for new and exciting percussive effects. This is one of the biggest drum circuits I've ever seen, with the most controls, so be prepared to spend some time with it. But I think you'll find that your time has been well spent!

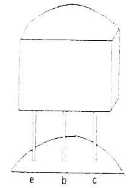


FIG.5

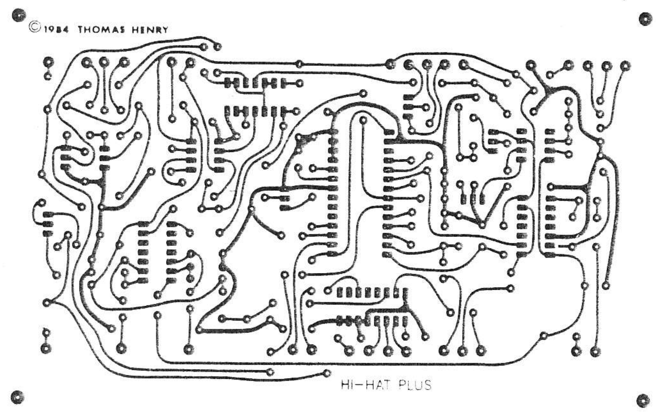


FIG.3 CIRCUIT BOARD FOIL PATTERN

HALF-SIZE DRAWINGS

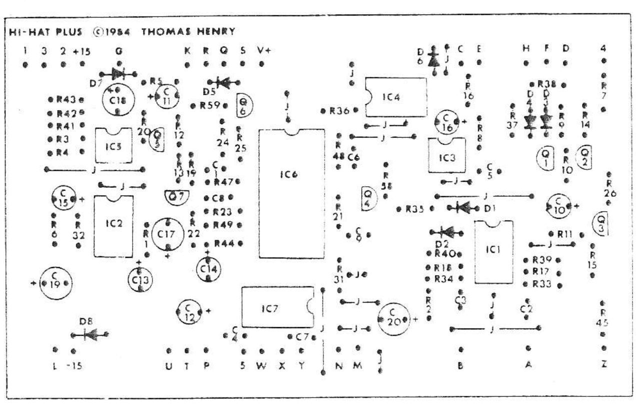
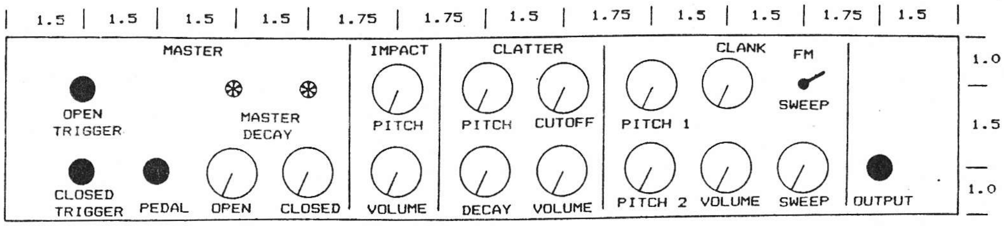


FIG. 4 CIRCUIT BOARD COMPONENT LAYOUT

FIGURE 6



= POTENTIOMETER

= JACK

= SWITCH

= LED

FRONT PANEL DESIGN: HI-HAT PLUS

All dimensions in inches.

Parts List

Resistors (5% tolerance, 1/4 Watt)

R1	330
R2	560
R3, R4	820
R5-R7	1k
R8-R11	1.5k
R12, R13	2.2k
R14, R15	3.3k
R16	4.7k
R17, R18	5.6k
R19	8.2k
R20-R26	10k
R27-R30	10k audio pot
R31	12k
R32	27k
R33-R38	39k
R39-R45	100k
R46	100k linear pot
R47	120k
R48	180k
R49	220k
R50-R57	500k linear pot
R58, R59	2.2M

Capacitors (15 or more Volts)

C1	0.001 uF mylar
C2-C6	0.01 uF mylar
C7-C9	0.05 uF mylar
C10	2.2 uF electrolytic
C11-C16	4.7 uF electrolytic
C17-C20	100 uF electrolytic

Semiconductors

D1-D6	1N4148 or equivalent
D7, D8	1N4001
D9, D10	LED
Q1-Q6	2N3904 NPN
Q7	2N3906 PNP
IC1, IC2	4739 dual op-amp
IC3	555 timer
IC4	4001 quad NOR gate
IC5	3080 OTA
IC6	SN76477 sound chip
IC7	SN94281 sound chip

Mechanical parts

J1-J4	1/4" phone jacks
S1	SPDT switch
Misc.	LED holders, wire, solder, knobs, front panel, hardware, etc.

Practical Circuitry

SYNTHESIZER PHASE SHIFTER

by: Thomas Henry

It's easy -- often too easy -- to stereotype musical devices. For example, some people (and this used to include me) tend to think that flangers and phase shifters are primarily guitar effects. Well, even though guitarists may have been the first musicians to popularize these exotic devices, flangers and phase shifters also make perfect additions to any synthesizer system. Of course you already know this if you built the Synthesizer Delay Line presented here in the June '84 issue; that circuit made possible a whole new family of weird timbre modulation effects. This time, we'll open up more possibilities by examining a high-quality, low-noise phase shifter you can build for your synthesizer.

In terms of sound, flangers and phase shifters have a lot in common. For example, with high resonance, both create the so-called "jet sound." This being the case, why would a system need both circuits? Well, we are entering a highly subjective area, but I feel that the flanger imparts a mood of tension to music. There's something almost brittle, metallic or mechanical about the timbre changes it renders. The phase shifter, on the other hand, always makes me feel like I'm sitting in front of a warm fireplace on a winter night. It has a smooth, even sound about it. My metaphors may not jibe exactly with the sensations you'll experience when you hear both units, but that's not the point. The important thing is that the flanger and the phase shifter, while creating similar effects in a vague sort of way, each have something new to add to a sound. In short, having both units available in a system makes a lot of sense!

This circuit has been optimized for use with a synthesizer system. As such, it assumes most of our usual standards. For example, input impedances are 100K, output impedances 1K, signal levels 10V p-p, and the control signal response is 1V/octave. If you're looking for a phase shifter more suitable for use with a guitar or other low level signal sources, get out your back issues of DEVICE and check Craig Anderton's AMS-100 design ("AMS-100, Part 4: Voltage Controlled Phase Shifter Module", DEVICE, Volume 1, Number 4-79, pp. 7-10.) You will notice a lot of similarities between that design and the circuit we're going to look at right now, since both circuits are based upon Solid State Micro Technology for Music's application note for the SSM2040 chip.

We won't get into the theory of phase shifters here beyond noting passing a signal through an allpass filter, then mixing it with the dry signal, creates the "swoosh" sound. An allpass filter, as the name implies, passes signals of all frequencies. However, even though the amplitudes of these signals are left unmolested, the phase responses are greatly modified. In general, then, an allpass filter changes phase but not amplitude.

How it works. Refer to **Fig. 1**, the Synthesizer Phase Shifter schematic. Now before you go and assume that I fibbed about the simplicity of the circuit, notice that the entire design utilizes only two chips. The four op amps labeled A1 through A4 are each one-fourth of the 4136 quad op amp package, and all of the other amps (toward the middle of the schematic) are contained in the SSM2040 chip. The four transconductance amplifiers in this IC form the allpass filter network mentioned above. As an extra benefit, the SSM2040 also contains all of the necessary exponential converter circuitry. So, while the schematic is somewhat drawn-out, the actual circuit is a snap to build.

In general, circuits employing the SSM2040 need to see a 1V p-p maximum input signal. Thus, after the audio input enters jack J1, R24 and gain control R17 attenuate the signal by a factor of 10. This reduces a 10V p-p audio signal to 1V p-p, which is then buffered by amplifier A1 and sent to the rest of the circuitry. You may wonder why the input is attenuated in this fashion as opposed to, say, using a standard inverting amplifier with a gain of less than unity. As it turns out, the output amplifier (A3) of this circuit must be a non-inverting stage to make the blend control work properly, so to maintain phase integrity of the dry signal, the input must be non-inverting as well.

Let's return to the input signal. The buffered signal first goes to the input of the SSM2040 via voltage divider R8/R1. This pair of resistors reduces the signal level to the transconductance op amp in the SSM2040, which keeps the amp working in its most linear range. C3 is the timing capacitor of the allpass network and R9 forms the feedback loop. The audio signal then passes down three more stages identical to the first stage; modulating the effective resistance of all four transconductance amplifiers phase-shifts a particular harmonic of the input signal from 0 to 720 degrees. The sum of the shifted signal and the dry signal then create the cancellations which characterize the "swoosh" sound of a phase shifter.

Next, the output of the allpass network goes to a polarity changer formed around amplifier A2

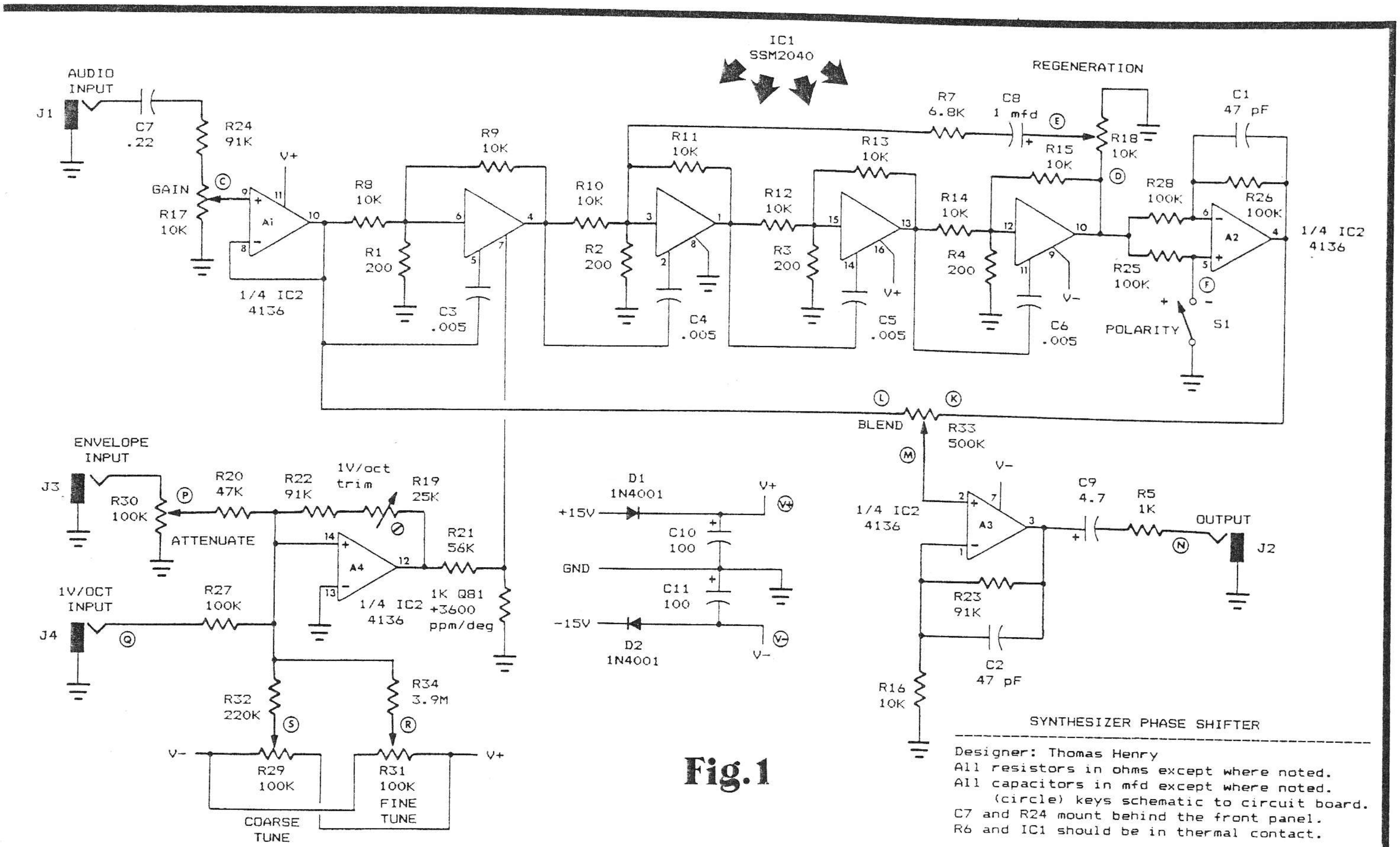


Fig.1

and associated components. We have seen this idea before in previous columns, ("Quadrature Function Generator" and the "Synthesizer Delay Line"), so not much need be said about it. With switch S1 open, the allpass signal is left uninverted. However, when S1 is closed, the signal is inverted before being sent to the final blend. The sound created by the two positions of this switch are quite different, with positive phasing giving a more biting edge and negative phasing yielding a more mellow, bassy response. Note that by using the sign changer idea, we get around the need for a DPDT switch (usually required for this type of function) and as a bonus we keep the audio signal close to the circuit board.

In the design of lowpass and band pass filters, undesired supersonic oscillations are seldom a problem since the amplitudes of the frequencies involved are naturally attenuated by the filtering nature of the circuit. Highpass and allpass filters are altogether a different matter, though. In particular, allpass filters can start oscillating very easily at supersonic frequencies due to the unpredictable nature of the phase response. To get around this problem, C1 rolls off the response of amplifier A2 at about 35 kHz, which is low enough to stop unwanted supersonic activity without affecting the general tone color of the audio input signal.

After the polarity processing, the signal finally goes to the output blend control, R33. Note that one side of this pot receives the allpass signal while the other gets the dry signal from the output of the input buffer, A1. R33's wiper thus picks off a blend of the two. This can be continuously varied from full dry signal to full allpass, with any ratio of the two available at in-between settings of the pot. By the way, many phase shifter circuits include a switch which can be flipped to generate a pleasant vibrato effect. The Synthesizer Phase Shifter eliminates the need for this switch, since the vibrato effect can be attained simply by dialing R33 to its full allpass position.

The output mix, as created by R33, is buffered and amplified by A3. Since the audio signal was attenuated way back at the input of this circuit, A3 brings the level back up to spec by introducing a gain of 10. Again, the frequency response is rolled off a bit by C2 to reduce the possibility of supersonic nastiness.

To keep annoying "thumps" to a minimum and also to avoid hassling around with output trimpots, I decided to AC couple the Synthesizer Phase Shifter. C7 accomplishes the input coupling, while C9 handles the output. Of course, this conflicts with our standard specifying DC coupling throughout, but since it's unlikely you will want to shift the phase of a DC signal, the tradeoff seems justified. By allowing this deviation in our standard, we are able to maintain simplicity of execution without losing versatility.

To increase the effect of phase shifting by accentuating the hills and valleys of the response, C8 and R7 provide a feedback path back to the second stage of the allpass filter. R18 is the regeneration control, and can add a real bite to the sound. Select R7 so that the filter will start oscillating if this control is increased to its maximum setting. This bonus allows you to employ the unit as an auxiliary sine wave oscillator as well! When using the circuit in this way, turn the blend control to the full allpass position and then raise the regeneration control until oscillation just begins reliably. This will yield the lowest distortion sine wave.

Controlling the Synthesizer Phase Shifter. A4 forms the control voltage summer. R29 and R31 are the coarse and fine tuning controls, respectively. Since these controls are strung across the bipolar supply, they have enough "oomph" to compensate for any offsets in the other control signals. The coarse control will sweep the phase shifter over a range of a dozen octaves or so, while the fine control sweeps less than an octave.

Envelopes, which usually swing from 0V to +5V, may be applied to J3, the envelope input. Resistor R20, with respect to R22 and R19, sets this input's gain to about two, thus letting an envelope signal modulate the device over its entire range. Attenuator R30 tames the sweep as needed.

J4 is the 1V/octave input. You may wonder why an input of this nature is needed in a circuit like a phase shifter; consider a situation where your keyboard is controlling a VCO whose output is fed to the phase shifter. Now imagine that you hit upon a pleasant timbre while adjusting the coarse and fine tuning controls. By connecting the keyboard's control voltage output to the phase shifter's 1V/octave input, the circuit will track the VCO wherever it goes. Therefore, the timbre remains constant regardless of frequency. Of course, there is some evidence which shows that acoustic instruments themselves don't obey this relationship, but all the same it's probably better to have this feature and not use it (you might use it someday) then not have the feature at all.

After A4 mixes all of the control signals, the result is attenuated by divider R21/R6 and sent to pin 7 of the SSM2040. Note that R6 is actually a thermistor and performs temperature compensation for the exponential converter within the SSM2040. In order for this to work correctly, R6 must be in close thermal contact with the chip. If this is done, the Synthesizer Phase Shifter will perform reliably under any temperature conditions. Of course, if precise tuning doesn't matter all that much to you, R6 could be replaced with a regular 1K resistor. This would make the sine wave oscillator application unpredictable, among other things, so stinting on the thermistor really doesn't seem like such a good idea. After all, we've come this far, so why not cinch the circuit with the addition of a fairly inexpensive part!

Building the Synthesizer Phase Shifter. **Fig. 2** shows the complete parts list. Most of the components are easy to find; the SSM2040 and the thermistor are the only unusual parts, but fortunately PGS Electronics (Route 25, Box 304, Terre Haute, IN 47802), a regular advertiser in **Polyphony**, has these parts available thus putting them within reach of readers of this column.

FIG.2

SYNTHESIZER PHASE SHIFTER: PARTS LIST

RESISTORS

R1 - R4	200 ohms
R5	1K
R6	1K Q81 thermister
R7	6.8K
R8 - R16	10K
R17, R18	10K potentiometer
R19	25K trimmer
R20	47K
R21	56K
R22 - R24	91K
R25 - R28	100K
R29 - R31	100K potentiometer
R32	220K
R33	500K linear potentiometer
R34	3.9M

CAPACITORS

C1, C2	47 pF disk
C3 - C6	.005 mfd mylar
C7	.22 mfd mylar
C8	1 mfd electrolytic
C9	4.7 mfd electrolytic
C10, C11	100 mfd electrolytic

SEMICONDUCTORS

D1, D2	1N4001
IC1	SSM2040 filter IC
IC2	4136 quad op-amp

MISCELLANEOUS

J1 - J4	1/4" phone jacks
S1	SPST switch

sockets, wire, solder, knobs, front panel, heatsink grease, hardware, etc.

NOTE: C7 and R24 mount behind the front panel.

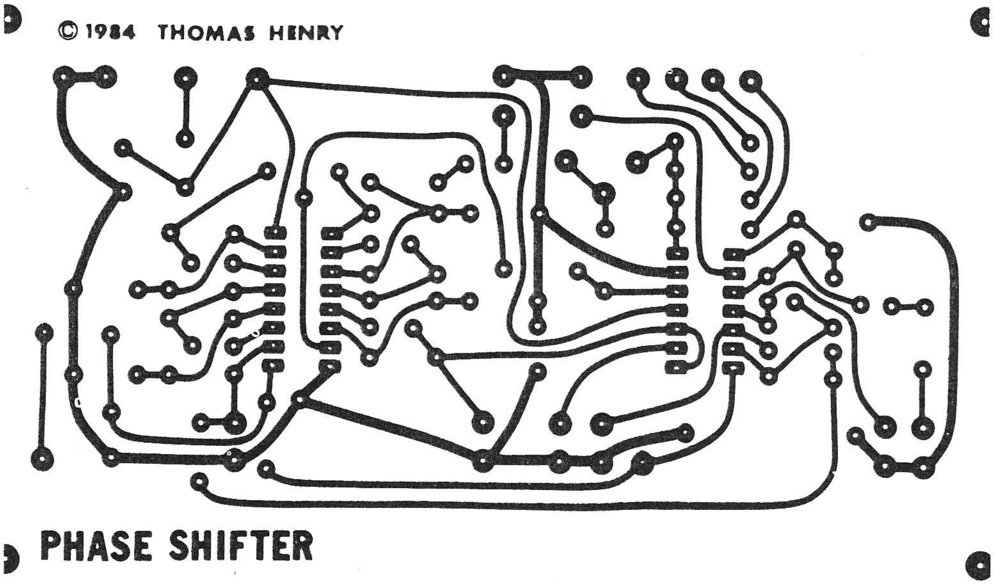
To simplify building this device, **Fig. 3** shows the artwork for a printed circuit board. Other modes of construction are possible, but a circuit board tends to give the best and cleanest results, while making things that much easier for the final front panel hookup. **Fig. 4** gives the circuit board parts placement guide.

Using **Figures 1, 3 and 4**, load the board, saving the thermistor, R6, for the last step. **Fig. 5** shows how this part mounts right on top of the SSM2040 ship. Notice that special solder holes on the board at either end of the chip have been provided to facilitate this operation. Before soldering the thermistor in place, spread some heat sink grease over the top of the chip and then press the thermistor down onto it. This increases the thermal tracking of the two parts.

After preparing a suitable front panel, you may complete the final wiring. Since this circuit is sensitive to hum, notice that shielded wire is used for many of the connections. To ease the task of running shielded wire from the circuit board to the front panel, note that a number of extra ground pads (denoted by the letter "G") are provided on the board. In all cases, the shield is soldered to the circuit board, but not the front panel! Let's run through a sample hookup to point "C" to see how this works. Note that a ground pad, labelled "G", is available next to it. Strip the insulation from a shielded wire, exposing both the hot wire and the shield. Connect the hot wire to point "C" and the shield to point "G". Now, at the panel end of things, strip the wire and cut off the shield entirely. Connect the hot wire to the wiper of R17. Continue this process for all of the other connections which require shields (these are "C", "D", "E", "F", "K", "L", "M" and "N"). In the case of point "F", the shield hooks up to one side of switch S1, but in all other cases the shield is ignored at the panel end of things.

Of course, the front panel eventually will need a ground path, so take one of the uncommitted "G" pads and form an electrical connection between this and

FIG.3



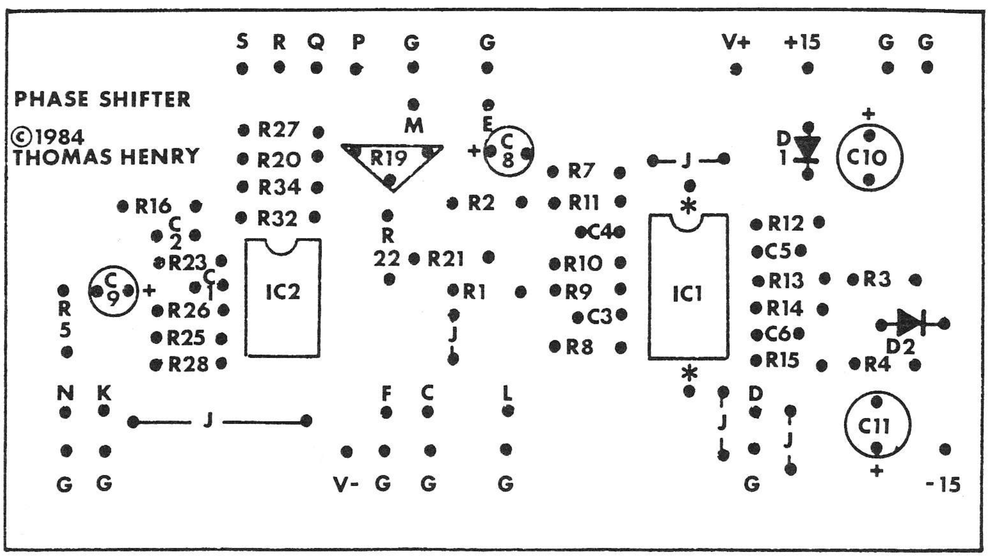
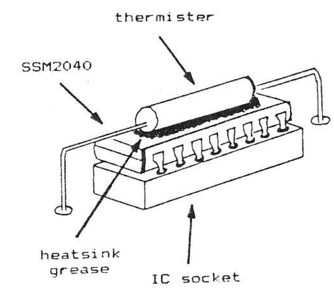


FIG. 5

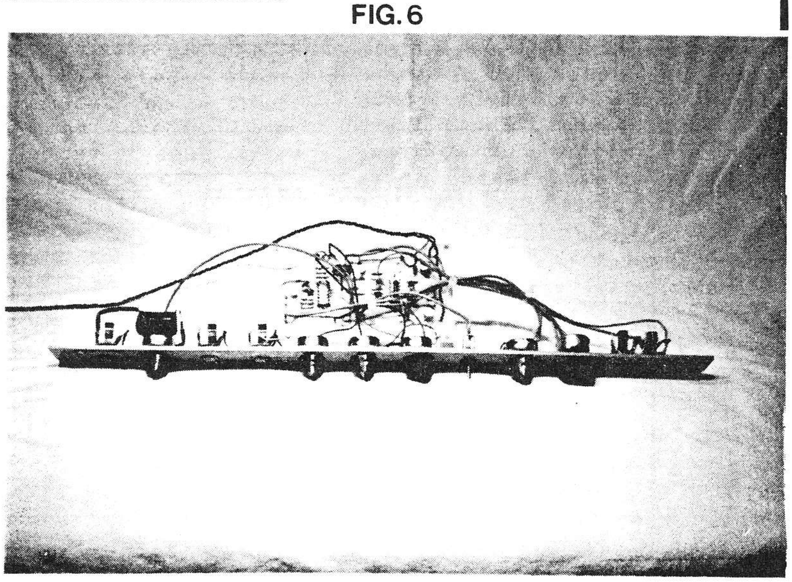


Detail showing how the thermister is mounted in thermal contact with the SSM2040. Note that solder holes are provided on the circuit board, at either end of the socket, for installing the thermister in this fashion. The two holes are marked on the parts placement guide with asterisks.

Tweaking the scale trimmer, R19, is fast and easy. Simply start the device oscillating (as described above) and tune it just like you would any VCO. You'll find this easiest to do if you work in the 200 Hz to 1 kHz range.

And there you have it, a high quality Synthesizer Phase Shifter. I think you'll find this to be an exciting addition to your system, since it really opens up the door to more animated sounds. So what are you waiting for? Start phasing today!

FIG. 6



the panel. What we have done here with all of this monkey business is shielded all of the sensitive wires while maintaining a single ground path to the front panel. This reduces the possibility of hum caused by ground loops.

Fig. 6 shows the prototype of this circuit. C7 and R24 mount behind the front panel and if you look carefully at the picture you will notice C7 directly behind potentiometer R24. I used some five minute epoxy cement to secure the capacitor to the pot and then completed the hookup accordingly.

Practical Circuitry

Build An Electric Drum Pad

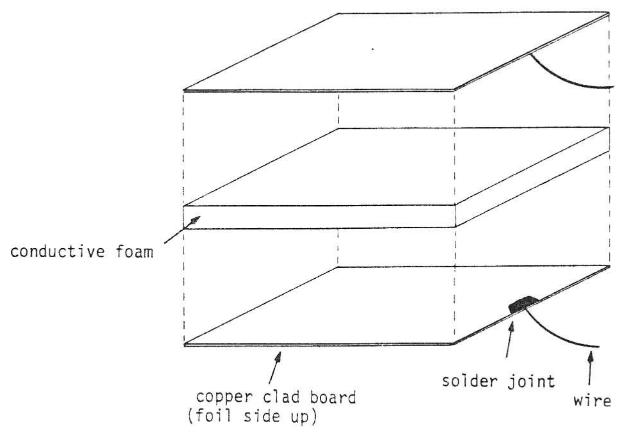
by: Thomas Henry

Computer drums are great, but there may come a time when you'll want to arrange for a "hand-played" percussive part. For example, suppose you've written a song which requires a special bridge or intro in a strange meter. Getting the computer to play something like this might be a drag, so it might be easier to switch over to a hand-played passage. And if you do stage work with a band, the value -- both musical and visual -- of a real-time drum unit will be appreciated. Well, here's some good news! Adding an electric drum pad to your setup is not only easy but inexpensive as well, due to the miracle of "conductive foam." As a matter of fact, in this installment of **Practical Circuitry** we'll see exactly how to build just such a unit, with quality rivalling that of commercial drum pads.

An Inexpensive Pressure Transucer. The electric drum pad consists of two main components: a pressure sensor and the supporting circuitry. Let's handle the mechanical aspects of this project first by examining how to construct a pressure sensor, then we'll cover the supporting circuitry.

Fig. 1 gives the basic idea. A piece of conductive foam is sandwiched between two copper clad circuit boards, each of which has a wire attached to establish the electrical connection. This creates a transducer whose resistance changes with pressure. Compress the pad and the resistance of the unit drops to around several hundred Ohms; release the pad and the resistance shoots back up to fifty kilohms or more. As we'll see in just a bit, the exact resistance values aren't important since the supporting circuitry responds to resistance **changes**, not absolute values.

fig.1



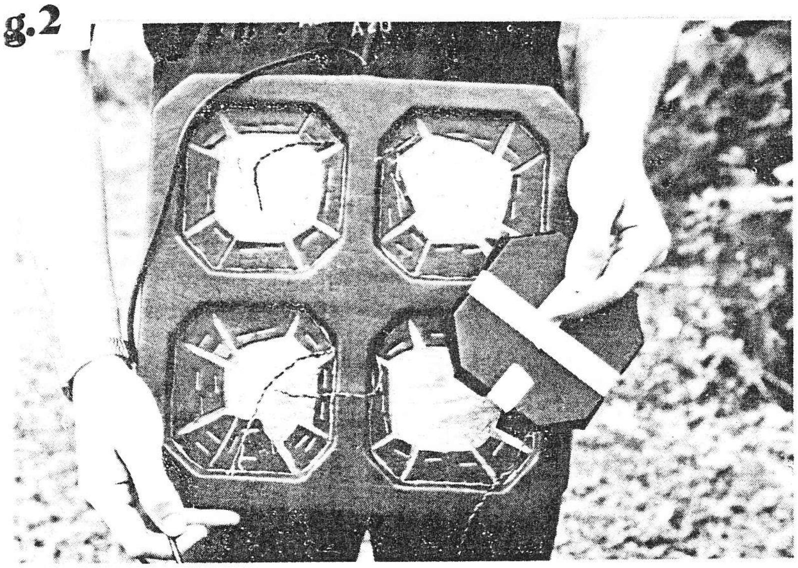
INEXPENSIVE PRESSURE TRANSDUCER MADE FROM COPPER CLAD BOARDS AND CONDUCTIVE FOAM: EXPLODED DIAGRAM

You may be wondering where you can find conductive foam which has this magical property of changing resistance under pressure. Surprise -- you see it every day! The foam employed by this project is the sort commonly used to pack CMOS integrated circuits because of its conductive properties. You know the type; it's a rather coarse, jet black substance. By keeping all of the IC pins at roughly the same potential, static electricity (which is the natural enemy of CMOS) doesn't have a chance to do its dirty work.

A pressure transducer, like that detailed in **Fig.** 1, can generate a variety of signals. For example, see my short design idea in **Electronics**, "Conductive Foam Forms Reliable Pressure Sensor", May 19, 1982, pp. 161, 163, which shows how to use this pad to generate simultaneous control voltage, gate and trigger signals, suitable for use with a synthesizer. Since we don't need all of these features for an electric drum pad, we'll concentrate more on getting a suitable trigger from the unit whenever it is struck by a drum stick. If this sounds like an exciting project to you, then let's get cracking and build the pressure sensor, since it forms the heart of the electric drum pad.

Making an Electric Drum Pad. Any musical instrument should be attractive and rugged, so think carefully about how you will fabricate the sensor. A number of implementations are possible; here's how I did it. I decided to mount all four sensors in a single unit to make the instrument tight, sturdy, and easy to carry in case I decided to use it on stage. Another "must" was that it should be easy to connect to the supporting circuitry, thus avoiding myriad plugs and wires hanging all over the place. As it turns out, these features were simple to achieve.

fig.2



Refer to **Fig. 2**, which shows the construction of the quad sensor package in progress. Start by cutting out two pieces of plywood sufficient in size to house four pressure sensors in a two-by-two array. One piece of the plywood should be left intact, but cut four holes in the other to accommodate the four pressure sensors. You will note from **Fig. 2** that I used a hexagon shape both for the plywood pieces and for the individual pads. There's no particular reason for this other than it adds a bit to the aesthetics of the project.

After fabricating the two pieces of plywood route out four channels, each about 1/4" wide, in the wood. These channels allow the connecting wires of the pressure sensors to be brought out conveniently to a single termination. The wires run through the wood in much the same way that electric guitar pickup wires run to their associated volume and tone controls. After creating channels for the wires, string four twisted pairs through their respective openings, and terminate them at the top of the unit. Examine **Fig. 2** carefully; note that a single cable emanates from the top of the drum pad. We'll have more to say about this a little later.

When all of the wires are in place, apply a liberal dose of carpenter's glue to the two pieces of plywood and secure them together with small brads or tacks. At this point, the drum pad package will be a solid single piece, with the wires already running through the wood to the appropriate openings.

The unit can now be painted or covered in a variety or ways, but covering the package with vinyl fabric gives very attractive results. This sort of material is available at any fabric or department store. To apply vinyl, spread liberal amounts of carpenter's glue to the back surface of the fabric, then stretch it over the wood. Flip the drum pad onto its face and then secure the fabric to the back side with a staple gun. (This back surface will be covered in a later step, so don't worry about the ugly appearance of the staples just now.) You will have to cut and tuck the corners at various times, but this will be obvious to you when you start to tackle the project. After stapling the back flaps in place, turn the drum pad over so that you're looking at the front again, then cut, staple and tuck the access holes as required, making sure to avoid the wires you installed earlier. Again, these staples and flaps will be covered later on, so don't be alarmed at the seedy appearance.

At this point, we've caught up with the picture in **Fig. 2**. Careful examination of this photograph reveals the vinyl covering, various flaps and staples, and the wires emanating from the access holes in anticipation of loading the four individual pressure sensors.

On the left hand side of **Fig. 2** you will see one of the pressure sensors in a partially assembled state. Essentially, the foam has been sandwiched between the two pieces of copper clad board and the connecting wires soldered on to their respective plates. To keep the unit from slipping and sliding around, strap two pieces of masking tape around the affair. When strapping the sensor together, make sure that the plates are snug against the foam without compressing it too much.

By the way, Radio Shack is a good source for conductive foam. They stock a 5" by 5" piece (#276-2400) which is the perfect size for a single pressure sensor. The price is under a buck, so you can tell that this is a very economical project!

To add to the reliability of the electric drum pad, scrub the copper clad boards thoroughly with some 000 steel wool before assembly. You might even want to consider tinning the copper surfaces as an extra precaution against corrosion. I was able to get our local university to do this for me, but you'll be glad to know that a number of mail order houses now stock a chemical which will perform this step at home -- just be sure to follow the directions and obey all safety warnings.

Okay, we have the main holder board done and the individual pressure sensors partially assembled. Let's try to tie up the loose ends now. The sensors need to be protected somehow and at the same time given a surface conducive to being struck by a drum stick. I cut out pieces of vinyl, with appropriate flaps, and actually wrapped each sensor in its entirely. If you go this route, use industrial strength rubber cement (like Weldwood) to secure the vinyl to the pad; this type of glue adheres to just about any surface. Again, each drum pad is covered in vinyl, with all of the various flaps wrapped under the sensor as required. Don't worry about the ugly appearance of the side with the flaps; we'll be sure to glue it into its receptacle with this side down.

After covering the pressure sensor, solder the two leads to the pair of wires emanating from one of the receptacles. Cover any exposed surfaces of the wires with electrician's tape. Now, coat the inside of the receptacle with a thick layer of silicone bathtub caulking, and press the pressure sensor down into it, dressing the wires underneath. Fill any cracks between the sensor and the board with additional caulking. Repeat this for the other three pads. **Fig. 3** shows the final result.

pressure sensors. The result is a decent looking instrument!

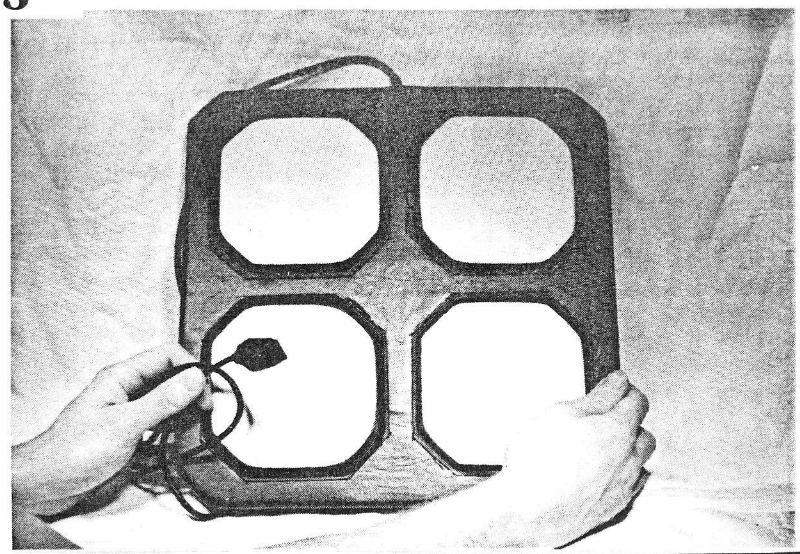
One final problem needs to be rectified. Recall that the back side of the drum pad is a mass of flaps and staples. We can't let that go unattended! Here's how to fix it up. Check the automotive section of your local department store and look for one of those rubber mats intended to sit on the floor of a car. Most stores carry several kinds, and they range from under a dollar to more that twenty dollars in price. Get the cheap one! (The fancy ones have all sorts of exotic carpet materials on them; we don't need that. All

hard part. If you have followed all of these steps, then you are now the proud possessor of a complete, four-in-one drum pad. At this point we need to switch gears and come up with some sort of interface circuitry, but as you'll see, this is a snap.

How the Support Circuitry Works. Fig. 4 shows the schematic for the electric drum pad support circuitry. If you are building a quad drum pad, then you will need to repeat this circuit four times. However, all parts are inexpensive and this shouldn't impose too great a financial burden.

Refer to **Fig. 4** now. The pressure transducer is hooked up in series with R6 and the entire voltage divider is strung between the -15V supply and ground. At rest, the pressure transducer will have a resistance many times greater than R6, so the voltage at the tie point will be very close to -15V. When the transducer is depressed, however, this resistance drops considerably and forces the tie point towards ground. Thus, striking the pad with a drum stick creates a positive going trigger. Notice that C1 is wired in parallel with the transducer; this dumps any scratchiness and noise to ground, thus improving the reliability of the unit considerably.

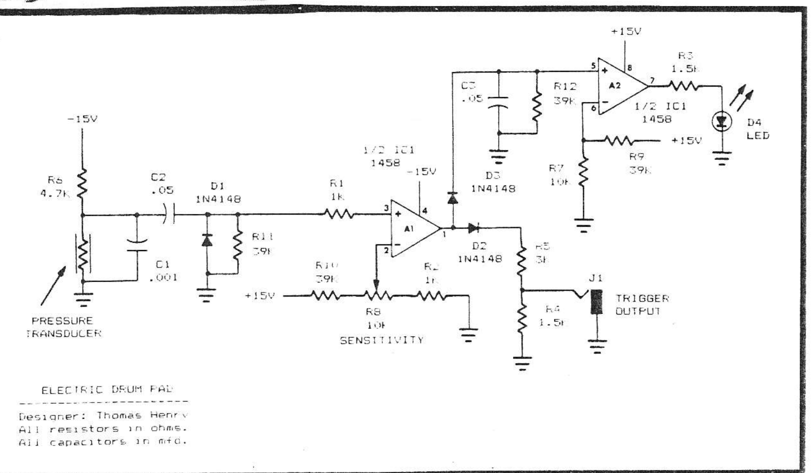
fig.3



After giving the silicone sufficient time to dry (several day, since the lack of air underneath increases the drying time somewhat), outline each pressure sensor with a strip of rubber tape. Rubber tape is available at most department stores. Sometimes the glue on this is rather inferior, so if you need to, spread a little more of that industrial strength rubber cement on the tape strips. Use a razor knife to trim up any unsightly seams or edges.

This may sound like a lot of work, but if you've successfully completed these steps, then you will have a unit which is not only strong but also attractive. You will notice that all of these steps have insured that each pressure sensor is securely fastened in its receptacle, in such a way that the pad actually seems to be part of the surface. Again, refer to **Fig. 3** and examine it carefully, noting that the final product is very homogeneous in nature. By the way, I used blue vinyl for the main surface and yellow for the

fig. 4



we want is a plain piece of rubber with a slightly pebbled appearance.) Now cut this piece of rubber matting to size and affix it firmly to the back surface of the drum unit, again using some industrial strength rubber cement. Besides covering up the unsightly staple and flap mess, the rubber provides a nice non-slip surface for the musical instrument.

Okay, we're done with the

The positive pulse generated by the transducer is AC coupled to the rest of the circuitry by C2. This serves to eliminate the negative bias, among other things. Diode D1 eliminates negative excursions of the pulse, while R11 acts as a load resistor for the voltage passed by C2.

Comparator A1 senses the pulse generated by the transducer and swings positive. Since the quiescent output of A1 is normally -15V, diodes D2 and D3 restrict the travel so that the output actually swings from 0V to +15 V. Notice that the threshold of A1 is set by potentiometer R8 (along with R10 and R2). This allows you to set the sensitivity of the electric drum pad to accommodate a variety of playing styles. The values of R8, R10 and R2 were selected under the assumption that the drum pad would be struck with a drum stick; change these if you have something else in mind. Although the circuit is quite tolerant of different types of foam and playing styles, you can increase the circuit sensitivity by lowering R2's value (or lower the circuit sensitivity by increasing R2's value). My prototype develops a pulse voltage of about +2V as measured at the junction of C2 and R11.

A1's output splits off in two directions. D2 sends the output to voltage divider R5 and R4. The output of this divider has an amplitude of +5V and an impedance of 1K. These are our old standards again. In addition, the pulse width will be somewhere in the neighborhood of 1 millisecond, although this may vary slightly depending on the setting of R8 and the force with which the pad is struck.

D3 couples the output to the peak detector composed of C3 and R12. In this situation, the peak detector serves as a pulse stretcher so that LED D4 turns on for a reasonably long period of time, thus giving positive indication that the pad has been struck.

Finishing Up the Project. And that's all there is to it! As mentioned, if you are building a quad drum pad (as illustrated in **Figs. 2 and 3**), then you will want to build four interface circuits. I used a D-9 plug to patch the pads to the support circuitry; this is the same type of plug used with joysticks on many personal computers. In fact, when my old VIC-20 joystick bit the dust, I clipped the plug and accompanying cable off and saved it just for this purpose. You can see this cable if you examine **Fig. 3** closely.

The D-9 plug can accept up to nine wires (hence the name), but we need only five here: four hot wires and one ground. When I wired up my unit I employed the same pinout as used by Commodore for their VIC-20 and C-64 computer joysticks. This gives the bonus of being able to plug a joystick into the circuit as well as the drum pad, thus adding the capability of joystick generated triggers to the synthesizer! If you want to go this route, check the owner's manual for your own computer and follow the pinout detailed there.

You will need a female D-9 connector to mount on the front panel. Fortunately, these are very easy to find (Radio Shack part number #276-1538).

After building the support circuitry, mount the entire unit behind a front panel. I found that a standard 1-3/4" by 19" rack panel was sufficient to comfortably house the D-9 connector, the four jacks, four pots and four LEDs, while giving a very professional appearance. Mount the circuit board behind the panel with some small angles and #4 hardware, and then complete the final wiring. After hooking up the power supply, plug in the drum pad and confirm that it works. Strike each of the pads and rotate the pots while noting the effect on the sensitivity. You will probably find, as I did, that each pad dictates an optimal setting for its associated pot. In general, you will simply set the sensitivity and leave it for the duration of a session.

As mentioned above, although various types of foam will have different resistances, and a hundred other gremlins might try to invade the project, the electric drum pad gives reliable operation since the support circuitry simply doesn't care about these parameters. It responds to changes in resistance and nothing more. Thus, in spite of the "quick and dirty" nature of the pressure transducer, the unit has proved extremely reliable and very easy for creating music. Best of all, it's a do-it-yourself project costing under $30...so there is an alternative to all those expensive drum pads!

Practical Circuitry

Build a Voltage-Controlled State Variable Filter

By: Thomas Henry

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

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

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

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

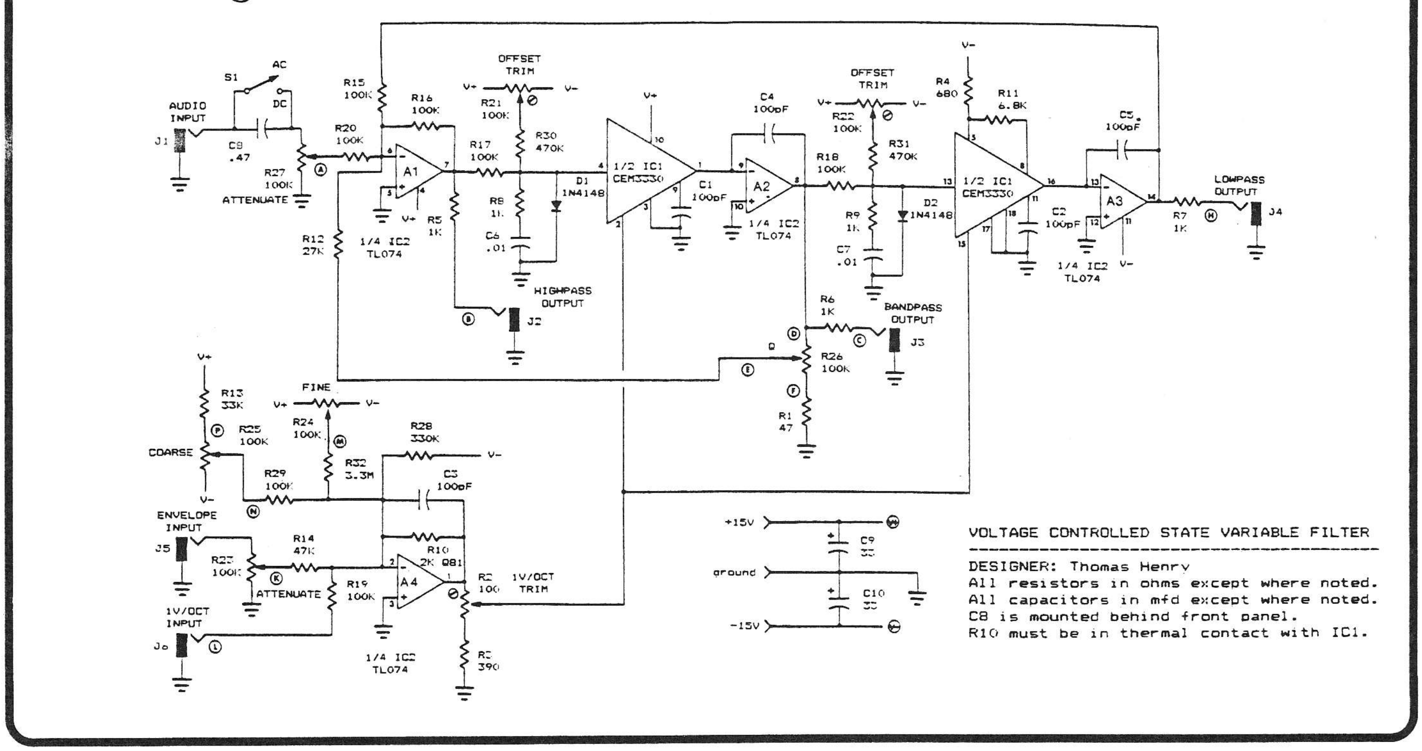
These comments should give you an idea of what a state variable filter is all about and what a good implementation for one should include. Let's look at the schematic for a tested design and check out some of these features in greater detail.

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

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

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

Fig.1



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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

Fig. 2

Parts List

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

Resistors (all values in Ohms)

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

Capacitors

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

Semiconductors

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

Hardware

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

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

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

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

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

Fig.3

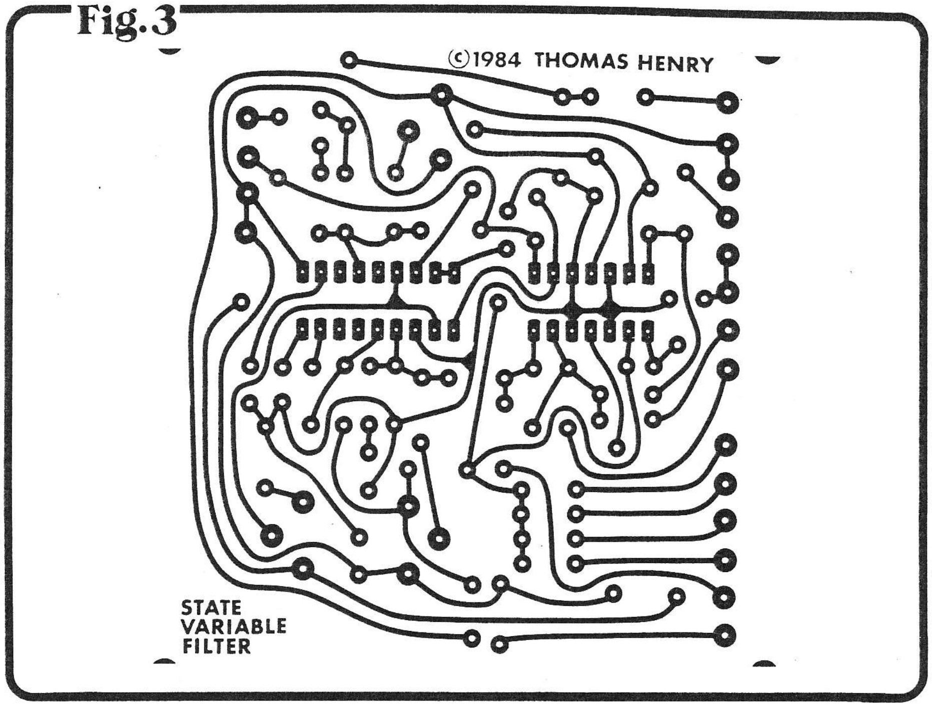
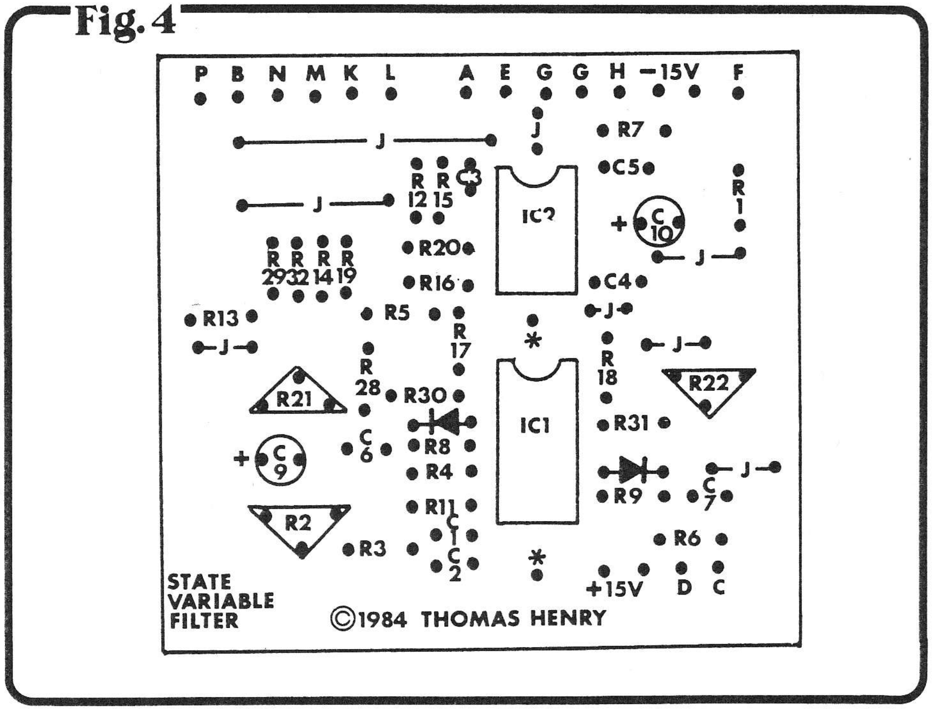


Fig.4



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

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

(continued on page 13)

Jupiter 6 by Roland

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

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

Juno 60 by Roland

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

Memory Moog by Moog

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

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

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Practical Circuitry...

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

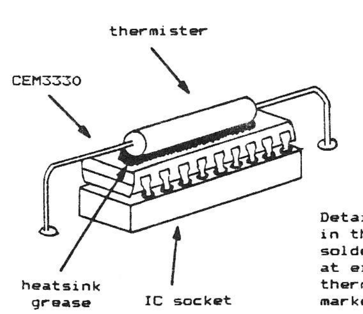
If anyone comes up with a better method for adjusting the DC feedthrough, or can explain why the first method gives weird results, please write to me in care of Polyphony so that I can pass the information along. And by the way, the DC feedthrough is very small anyway. In fact, for most non-critical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

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

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

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

Fig.5



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

PRACTICAL CIRCUITRY

Superseque: A Full Featured Analog Sequencer

by: Thomas Henry

Thomas Henry is a contributor to several magazines including Radio-Electronics, Compute!, Keyboard, *and* Run. *He played guitar professionally for ten years in order to put himself through school; currently he is an Assistant Professor of Computer Science at Mankato State University in Minnesota, and is wrapping up an implementation of the FORTH language that has extensive editing, graphics, and error-detection routines. Aside from computers and electronics, Thomas has a passion for Victorian literature, Sherlock Holmes, and the works of Oscar Wilde.*

Sequencers have been with us from the earliest days of voltage-controlled synthesizers and have always held an important position in the studio. Sequencers are especially useful nowadays since much modern music depends on intricate or rapidly changing rhythmic events.

With the advent of inexpensive personal computers, digital sequencers are now plentiful and cost less than the old analog types. So why build an analog sequencer? There are several reasons. First, Superseque is low-cost and easy to build; you don't have to write any software for it . . . just throw together a bunch of common parts. Second, due to the influx of MIDI equipment there are a lot of analog synthesizers being "dumped" for very low prices, and Superseque will work with any of these devices that follow a 1V/octave control voltage standard. Finally, if you currently use a non-computer controlled synthesizer and don't want to adapt it to computer operation, then an analog sequencer is the best way to go. (Of course, if your system already includes some "computer-ready" circuitry such as scanning keyboards, digital-to-analog converters, and the like, then you would best be advised to pursue a computer generated sequencer that interfaces with these devices.)

Introducing Superseque. This project is called Superseque because it includes just about every generalized feature you could possibly want in an analog sequencer. Superseque is a sixteen stage, dual channel sequencer with onboard control voltage summing. It also provides individual gate outputs for each stage and a master precision trigger output. Additional features include a programmable length input, manual and trigger reset inputs, manual and trigger clock inputs, and the capability to generate one-shot, "circulate," or automatic reversing patterns. Just about every feature is brought out to a jack, control or switch so that it is possible to completely reconfigure Superseque simply by arranging patch cords.

Older sequencer circuits were often designed around available circuitry, not the needs of musicians. For example, how many sequencers have you seen that used the 4017 CMOS decade counter? While this is certainly an easy to find chip and is perfect for some applications, it usually leads to a dreary sequencer. To create Superseque, I wrote down all of the features that I wanted as a musician and then designed a circuit to implement these features, without any preconceptions as to what components would be needed. As it turns out, the result isn't all that much more complex than a 4017 sequencer, nor is the cost that much greater. After all, let's face it—the biggest share of your money will be tied up in the pots and knobs, not the supporting circuitry!

If this sounds appealing to you, then let's get cooking and see not only how Superseque works, but how you can build it for your own system.

How Superseque works. Since this is a large circuit, the schematic has been divided into two sections, the *logic circuitry* and the *output circuitry*. You will probably want to leave the logic circuitry alone, since not a whole lot more can be added to it. On the other hand, the output circuitry may be modified and customized as you see fit to include more or less features. We'll talk about this in just a bit, but for now let's examine the logic circuitry which forms the heart of Superseque.

Like all sequencers, something has to count input pulses and provide appropriately related output signals; in Superseque, IC5 (see **Fig. 1**) does the counting. This IC is a CMOS 4516, "divide-by-16 synchronous binary up-down counter." Don't let the words fool you; this is just a fancy way of saying that the chip can count up to sixteen events in either direction.

Notice that there are four output lines on the chip (pins 2, 14, 11, and 6). These lines output a number from 0000 to 1111 in binary format. Since a binary number is difficult to use directly for sequencer control, IC6 takes on the job of transforming the binary number into something more convenient. IC6 is a CMOS 4514 one-of-16 decoder; input a four-bit number and one of the sixteen output lines goes high while all the others stay low. The 4516 and the 4514 work well in tandem since the one chip needs what the other has to offer!

Let's now look at some of the details. J1 is a clock input. Typically, a rectangular waveform is fed into this

". . . the sequencer can step through the various stages until the last one, where it stops . . . this is a great way to generate 'instant arpeggios' which play at superhuman speed!"

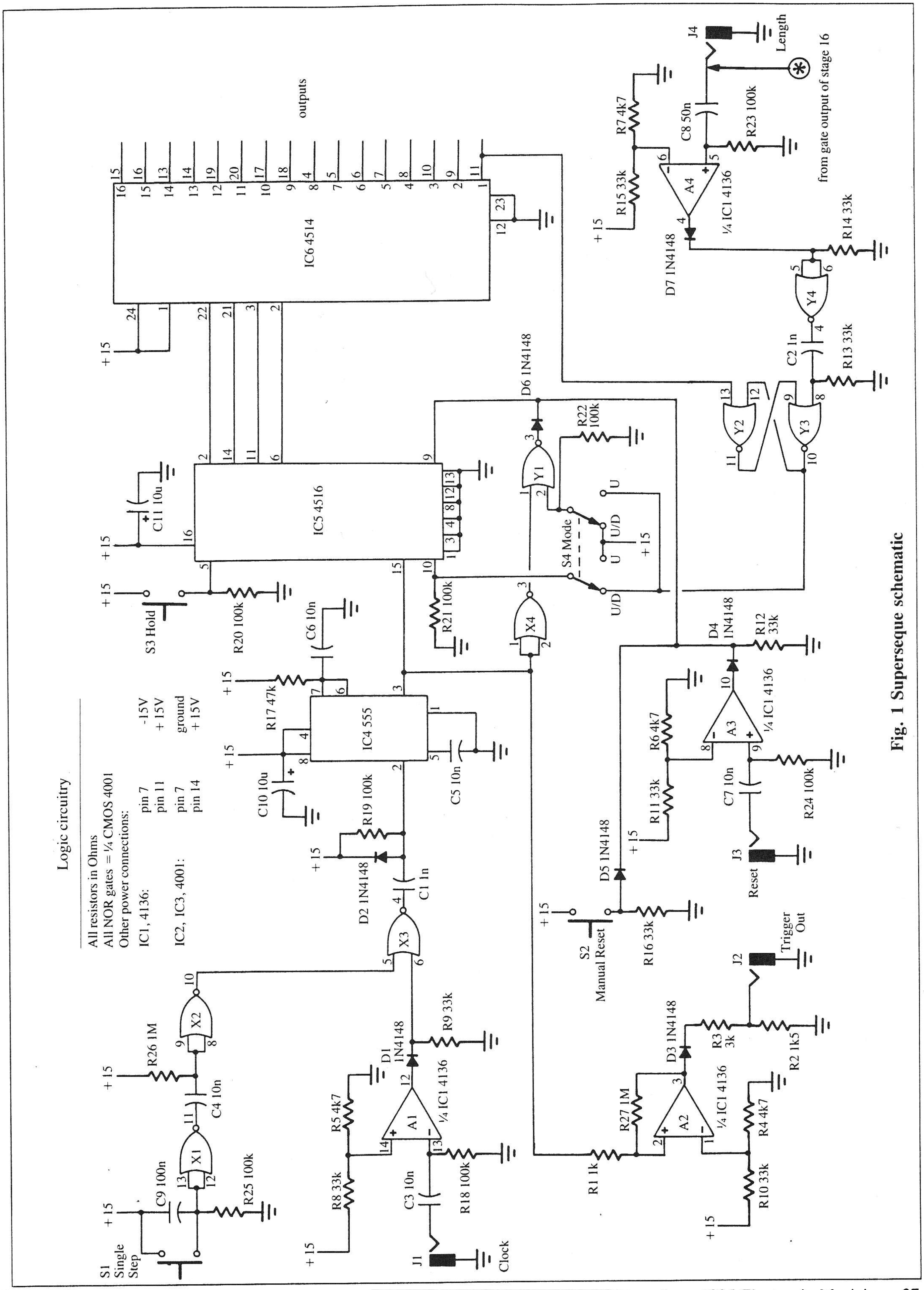


Fig. 1 Superseque schematic

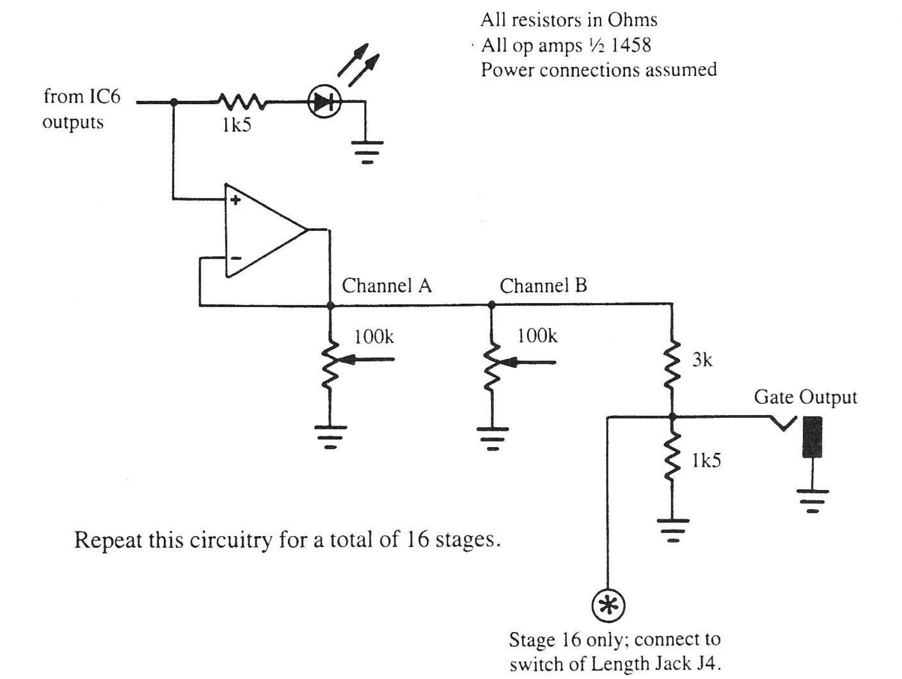


Fig. 2 Output interface circuitry

jack, perhaps from a VCO or LFO. The duty cycle of the input signal isn't important, since the differentiator composed of C3 and R18 trims the pulse width to about 1 millisecond. Likewise, R5 and R8 set a threshold for comparator A1, so that any input signal with an amplitude of two Volts or greater will fire the circuit. D1 restricts A1's output swing to positive excursions only, thus keeping negative going signals out of the subsequent CMOS circuitry.

S1 lets you single step through Superseque's 16 stages. Notice that this switch shorts out the capacitor in the RC combination of C9 and R25. This helps to eliminate the problem of contact bounce in a very simple, yet effective fashion. When S1 is pressed, the output of inverter X1 goes low and this fires the "half-monostable" comprising C4, R26 and inverter X2. The output of X2 goes high for about 10 milliseconds. A fairly long time constant is chosen here, again to minimize the effects of contact bounce back in S1.

The outputs of both the clock and single step circuitry are coupled to NOR gate X3. If either signal goes high, then the output of X3 will go low, firing the monostable composed of IC4 and associated components. IC4, which is a 555 timer, is set up to provide a 1.2 millisecond output pulse at pin 3, and this becomes the main clock signal for the rest of Superseque. During the breadboard stage of Superseque, trial and error demonstrated that the 4516 counter chip is very fussy about what sort of waveform clocks it. The purpose of the 555 is to provide a clean clock signal, whether the origin was a single step pulse or a clock input at J1.

As mentioned, pin 3 of the 555 is the master clock output. It feeds pin 15 of IC5 and thus steps the counter along. The clock also goes to comparator A2 and associated circuitry. A2 trims the voltage swing of the clock to a 0V to +5V range and this signal is then presented to J2 as a trigger output. This output can be used to fire envelope generators, synchronize drum units, or tie in other sequencers for tandem operation.

So far, all we have seen is the drab part of the design which any circuit must have to be called a sequencer. Let's skip ahead now to the reset section, for this is where the fun begins. NOR gates Y2 and Y3 are cross-coupled to form an R-S flip-flop. The first output stage of the decoder (at pin 11 of the 4514) is sent to the set input of the flip-flop, pin 13 of Y2. When stage 1 goes high, at the start of a sequence, the flip-flop is set and so the Q output, at pin 10 of Y3, goes high. Hang on to that notion for a minute, while we backtrack a bit and see what happens at the reset input of the flip-flop.

J4 is Superseque's "sequence length" input. Patching a cord from one of the sequencer outputs to this input programs the sequence length. Notice that J4 is a closed circuit type jack; with no cord plugged in, the circuit defaults to the maximum length of sixteen steps. As usual, A4 and associated circuitry shape the input signal up into something more usable by the CMOS chips (compare this to the clock input circuitry surrounding A1). The output of comparator A4 is inverted by Y4, and then the signal is capacitively coupled via C2 to the flip-flop's reset input. Capacitive coupling is used here since various timing considerations dictate that the flip-flop respond to an edge, not a level.

Okay, stage one of the sequencer sets the flip-flop, while the last stage (determined by the length input, J4) resets it. Now suppose that S4, the mode switch, is in the up/down position (as it is on the schematic). In this case, pin 2 of Y1 is at +15 Volts and this guarantees that the output, at pin 3, is low. This in turn holds the reset input of the 4516, pin 9, at ground throughout. The net effect is that the reset pin of the 4516 plays no part in the operation when in the up/down mode.

But look at the other half of DPDT switch S4. The output of the flip-flop is now directly coupled to pin 10 of the 4516. Pin 10 is the up/down selector for this chip. When this input is high, the counter counts in an upward direction; when it is low, the counter counts downward. We now have all of the pieces necessary to put the puzzle together. When the sequencer is turned on, the counter will count upward and when the last stage is hit, the flip-flop is reset. This pulls pin 10 of the 4516 low and the counter now reverses direction and counts downward. Then stage one is hit, the flip-flop changes state again and the counter counts back up. Thus we have arrived at means for generating an up/down mode of operation.

With S4 in the up mode position, pin 10 of the 4516 is pulled high, and this guarantees that the counter will count only in the up direction. Notice too that the flip-flop is now coupled to the reset pin of IC5, at pin 9. You shouldn't have too much trouble convincing yourself that the counter will now count up until it hits the highest stage, then resets to stage one and starts over again. Incidentally, by using the flip-flop composed of Y2 and Y3, and by making the reset input edge sensitive, any unusual timing anomalies are completely avoided. Every stage is on for an equal amount of time since the reset signal is recognized only at the time of the *next* clock pulse. This is a small, but important, point and took over half of the design time to work out. The net effect is that the sequencer works like you would expect it to; there are no strange effects to hamper your musical feelings.

The entire sequencer can be reset by an external pulse via jack J3. Likewise a reset can be forced by pressing switch S2. Notice that diodes D4, D5 and D6, in conjunction with R12, form an OR gate so that any one of these three signals can reset the counter, IC5, at pin 9. This is an instance of "Mickey-Mouse Logic" as detailed by Don Lancaster in the *CMOS Cookbook* (Indianapolis: Howard W. Sams, 1977). Despite the weird name, Mickey-Mouse Logic works very well in this situation and keeps the IC package count down.

S3 is a hold switch; press it and the counter stops counting, release it and the count resumes. This comes in handy when adjusting the sequencer, since it gives you a way to stop the action momentarily.

Well, that pretty much covers the logic circuitry of Superseque. Although most of the functions should be clear from the schematic, it probably isn't at all obvious how to best use them in a musical setting. We'll cover that in just a bit, but let's first take a quick look at the output circuitry.

Interfacing Superseque to the real world. Since CMOS chips are fairly sensitive to outside interference, I decided to completely isolate them with op amps. Besides beefing up the current drive, using op amps means that you won't have to worry about shorting outputs or hooking two outputs together accidentally. Again, this is just part of good musical engineering, ensuring that the performer can concentrate on music rather than worrying about things at a circuit or component level.

Refer to **Fig. 2**, which shows the interface circuitry for one of IC6's sixteen outputs. As shown in the schematic, the output is buffered by an op amp then drives two pots and a gate jack; but feel free to alter this arrangement as you see fit. Two channels (and hence two pots per stage) seem about right, for then you can use one potentiometer to set the pitch of musical notes, while the second can be used to set the duration of the notes. The gate output, of course, can be used to fire envelope generators or any other type of time related circuitry. And by the way, to help you keep track of things, notice that an LED is hooked up to the output of each stage as well. The blinking light patterns are very inspiring!

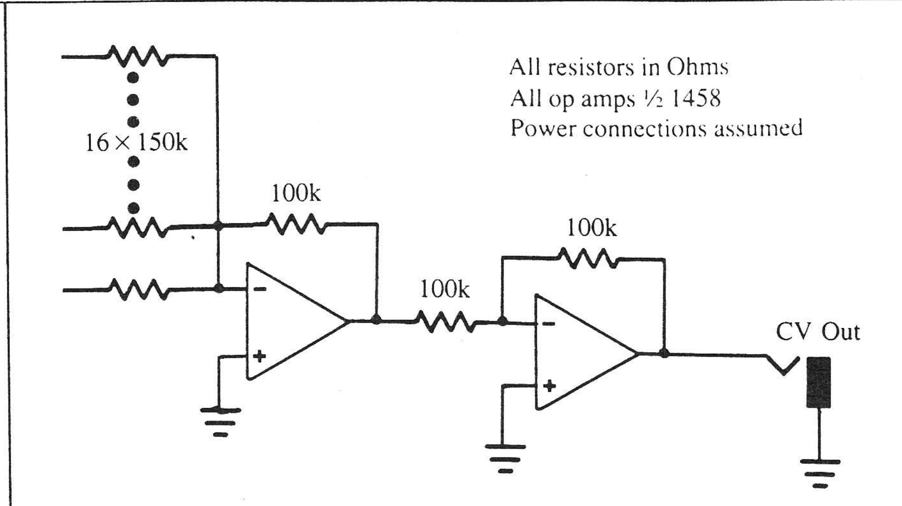


Fig. 3 Control voltage summer circuitry

After isolating each stage and hooking up the various LEDs, pots and jacks, you should have quite an array of controls. But we're still not done yet. Refer to **Fig. 3**, which illustrates the control voltage summing network. The wiper of each pot from **Fig. 2** is sent to this summer via a 150K resistor. If you used two pots for each stage (as recommended), then you will need two of these summer circuits, one for Channel A and another for Channel B. Once the voltages from each stage are summed into the summer, the result is inverted by one additional op amp so that the output swings in the positive quadrant (from 0V to +10V). Notice that by using 150K input resistors and a feedback resistor of 100K, the signal swing of each stage is reduced from +15V peak to +10V peak. This keeps the controls from feeling too "touchy," yet still lets you cover ten octaves if you're driving standard 1V/octave systems.

By the way, it might have occurred to you that an attenuator could be placed between the two op amps in **Fig. 3**, thus allowing you to scale the output as desired. I didn't add this feature since my synthesizer makes provisions for scaling elsewhere, but feel free to add it in if you think you need it. Likewise, you might want to add a pot which feeds in a variable offset voltage to the second op amp; this would allow for setting the range of the output.

Incidentally, like all other circuits presented in "Practical Circuitry" the inputs of Superseque have 100K impedances while the outputs have 1K impedances. Gate and trigger voltages swing from 0V to +5V, and the control voltage output covers a 0V to +10V range. These are all standard values for most synthesizers.

Finally, as a small point of interest, I tried using an inverted power supply for the logic circuitry (−15V and ground) so that the extra op amp required to invert the control voltages could be eliminated. Things didn't work out pleasantly at all! While I did get the circuit working with an inverted supply, other ephemeral problems popped up which made it not worth the effort. Anyway, op amps are cheap so all of the extra work simply didn't justify the savings. By the way, you can use ordinary 1458 dual op amps for the control voltage circuitry since you don't have to worry about bandwidth, noise, slew rate, or other critical parameters.

Building Superseque. It sure took a lot of fast talking to explain how Superseque works, even though the basic circuit action is fairly straightforward. Let's switch gears now and see how to build it.

Superseque's complete parts list is at the end of this article; note that the parts are all easy to find and not too expensive. I picked up the CMOS chips from JDR Microdevices (1224 S. Bascom Ave., San Jose, CA 95128) and the pots and jacks from PGS Elec-

PRACTICAL CIRCUITRY:
Do-It-Yourself Rack Enclosure

by: Thomas Henry

Thomas Henry is a contributor to several magazines including Radio-Electronics, Compute!, Keyboard, and Run. *He played guitar professionally for ten years in order to put himself through school; currently he is an Assistant Professor of Computer Science at Mankato State University in Minnesota, and is wrapping up an implementation of the FORTH language that has extensive editing, graphics, and error-detection routines. Aside from computers and electronics, Thomas has a passion for Victorian literature, Sherlock Holmes, and the works of Oscar Wilde.*

If you've been following my writing over the last several years, then you will have noted that I've often recommended packaging circuits in rack panel format (recall that a standard single rack panel height is 1¾″, a double panel 3½″, and so on). I like rack panels for three reasons. First, the size is an industry standard and this makes it easy to swap various modules back and forth between different enclosures. Second, rack panels are spacious; they provide oodles of room for jacks, pots and switches. Third (and perhaps most importantly), they are quite attractive and yet inexpensive to implement. Of course, if you buy your rack panels ready-made from a manufacturer you might not think the price is cheap! To get around this hitch, be sure to see my article in *Electronotes* on making professional looking panels for under a buck or two ("Making Rack Panels," *Electronotes*, Volume 13, Number 122, February 1981, pp. 5–9).

Okay, if I've sold you on the applicability of rack panels, then the next problem is where to find a good looking rack enclosure. Well, you can always buy a commercial unit, but the price of a new rack enclosure might surprise you—they typically run from $50 to over $500 depending on size. We'll scratch that! Another good bet is to look around at electronics surplus stores where used ones show up for much less. However, the trouble with most rack enclosures, whether new or used, is that they tend to be rather bulky and heavy.

Now I wouldn't tell you about these problems if I didn't have a solution, so let's investigate how to build a professional looking rack enclosure that doesn't cost an arm and a leg. Best of all, no unusual tools and no advanced shop techniques are required. (That's a good thing; I still have dismal memories of junior high shop class!)

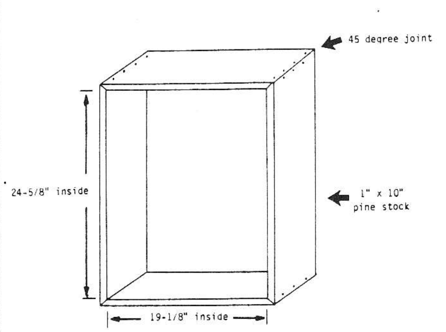


Fig. 1: Dimensions of a wood frame

Let's get started. Refer to **Fig. 1**, which gives the general idea of where we're headed. **Fig. 1** shows a simple frame consisting of only four pieces of lumber. We'll cover it with vinyl, attach some corners, slap on a handle and voilà . . . instant enclosure. Before getting too far ahead of ourselves, though, let's first consider some practical aspects of making the frame. To keep things as simple as possible, we'll use ordinary 1″ × 10″ pine planks. This is a standard type of lumber and is consequently inexpensive. If you've never purchased lumber before, then you might like to know that 1″ × 10″ planks are actually ¾″ × 9¼″ in size. This is probably one of the mysteries of carpentry I missed while sleeping in shop class! Anyway, when in Rome do as the Romans do, so we'll continue to call it a 1″ × 10″ plank.

The 10″ dimension becomes the depth of the box, and this is more than adequate to handle just about any type of circuit mounted behind a rack panel. The width of the box, of course, will be 19″ (on the inside), since this is the size of a standard rack panel. But what about the height? Well, this is up to you, but I suggest a height of 24½″ for two reasons. First, 24½″ is a multiple of 1¾″ and will allow you to mount the equivalent of fourteen single height panels. The other reason is more practical. As it turns out, lumber yards supply 1″ × 10″ planks in standard lengths of eight feet. By sticking with a box which is 19″ × 24½″, we can cut all four boards out of a single plank, thus minimizing the cost and number of saw cuts needed.

Okay, enough philosophy; let's get down to brass tacks now. **Fig. 1** shows the details for making the basic frame. Notice that inside dimensions of 19⅛″ and 24⅝″ are used. Adding on the extra ⅛″ to each dimension makes allowances for the width of the saw blade, and leaves a little slack to play with later on when we cover the box with vinyl.

Note that the joints of the four pieces which comprise the frame are cut with 45 degree angles. This isn't strictly necessary, but does make for a more professional looking box, since no unsightly seams are visible (to quote a brassiere ad). Most circular, skill and radial arm saws allow this type of angled cut without too much hassle.

After cutting the four pieces to size, smear carpenter's white glue on the joints and slap the box together. Use brads or nails at each corner to secure the box. There's really no need to go hardcore with screws here, since the white glue in combination with the nails cinches things up amazingly well. Of course, once you've mounted some panels in the unit and have put a back on, the enclosure becomes practi-

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