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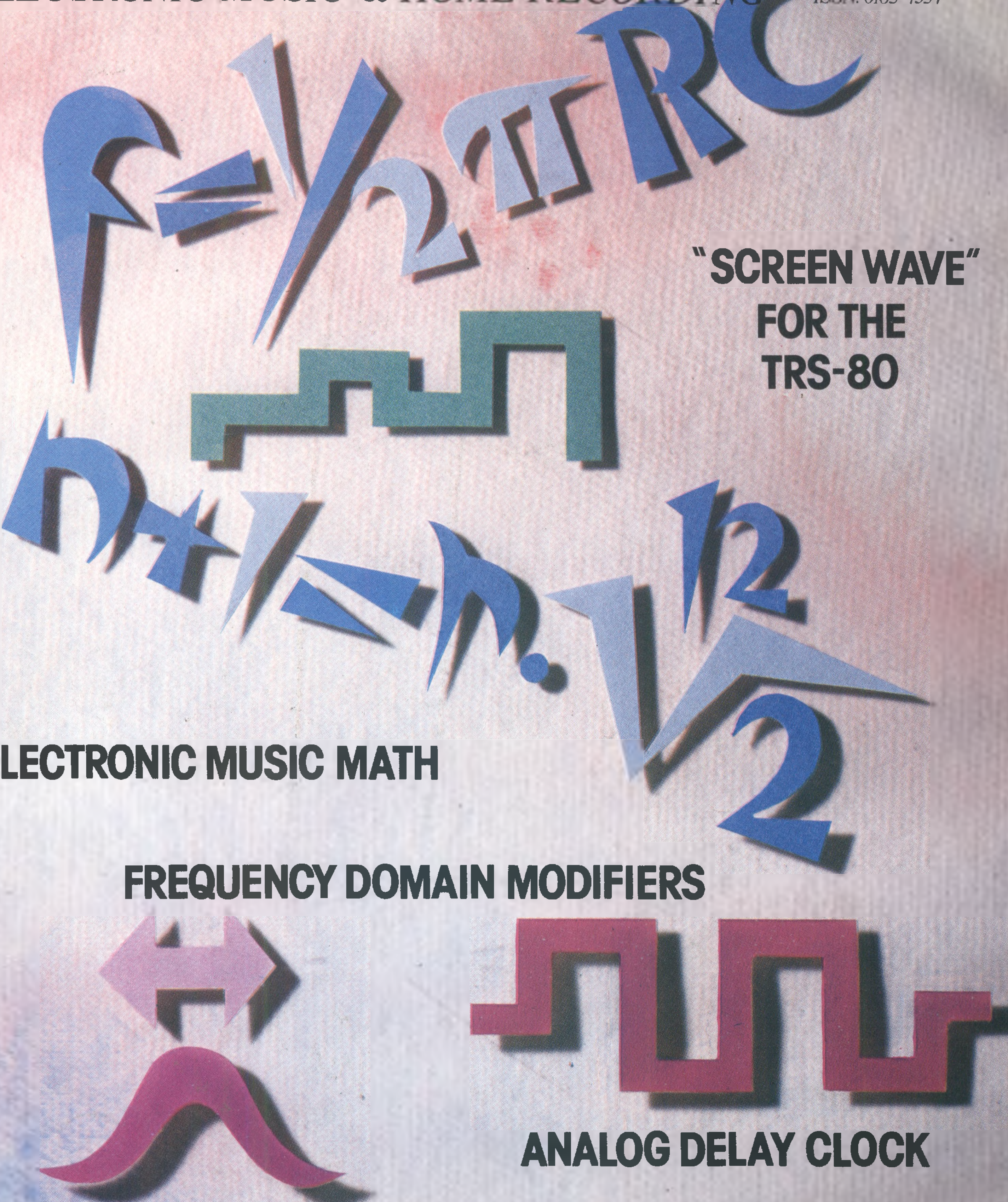
ELECTRONIC MUSIC & HOME RECORDING



"SCREEN WAVE" FOR THE TRS-80

ELECTRONIC MUSIC MATH

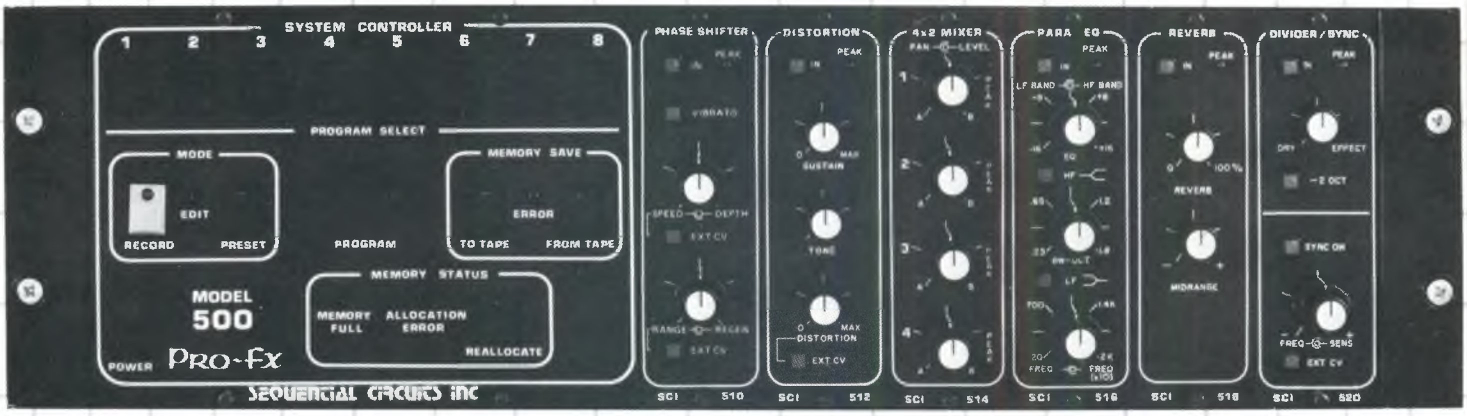
FREQUENCY DOMAIN MODIFIERS

ANALOG DELAY CLOCK

PROGRAMMABLE EFFECTS

The PRO-FX System from Sequential Circuits is the industry's first modular, expandable, rackmounted, completely programmable signal processing system that lets you:

- **Pick the combination of effects you want. Six are now available: Phase Shifter, Distortion/Sustain, 4 x 2 Expandable Mixer, Parametric Equalizer, Reverb, Transpose/Sync, and, coming soon, a Flanger/Doubler. Many more modules will be available in the future.**
- **Program 64 different combinations of control settings.**
- **Step through preset sequences of programs with the footswitch.**
- **Activate program changes with a synchronized tape pulse.**
- **Save sets of programs and program sequences on tape.**



The **Model 500 PRO-FX** offers the flexibility and control of full programmability with the convenience of modular rackmount design. Each module is carefully crafted for studio quality low noise and distortion specs. Every knob and switch on each module is completely programmable, which allows complex control settings to be stored and recalled instantly at the touch of a switch. You no longer have to worry about reproducing a particular effects setting in the studio or on stage. Now you can consolidate all your effects and mixing into one concise package, and have instant control over complete sound changes while playing.

The **Model 500 System Controller** is a micro-computer designed to store in memory 64 knob and switch settings for up to 30 effects modules using up to 3 expansion chassis. The System Controller will also allow you to preset sequences of program changes, actuated by the footswitch (for live performance) or tape track (for playback/mixdown).

For more information, see your dealer, or write Sequential Circuits, 3051 N. First St., San Jose, CA 95134 (408) 946-5240

Pro-FX

Another Industry First

from

SEQUENTIAL CIRCUITS INC

the people you expect it from!

Phase Shifter (Model 510) uses low-noise circuitry and a 4-stage phase shifting network. Besides speed, depth, and range controls, the regeneration knob controls the depth of the effect from very mild to intense. (External speed and range switches allow pedal or voltage control of sweep and range.

Distortion (Model 512) allows independent variable control of sustain and distortion, from completely clean sustain with no distortion to maximum distortion with a minimum of sustain.

4 x 2 Expandable Mixer (Model 514) has four inputs with level and pan controls. Maximum gain can be set from 20 to 40 dB via the internal trimmer on each input channel to accommodate both line and instrument level signals.

Parametric Equalizer (Model 516) is a 2-band, fully parametric, overlapping range equalizer with switchable peaking or shelving operation; any number of these units can be placed in series to obtain as many bands as required.

Reverb (Model 518) is a very smooth reverberation unit using a 6-spring delay line and active limiting to eliminate spring side-effects. This module also has a sophisticated EQ which sets bandwidth and tone of reverb signal.

Transpose/Sync (Model 520) provides many synthesizer effects for other instruments. It will track a note one or two octaves below the original pitch as well as a separate note from an octave below to an octave above. The upper voice can be "hard synced" to the original voice or one octave below. The upper voice can also have its pitch swept up or down by a sweep triggered on each new note; this gives the "sweeping sync" tone used on many keyboard synthesizers.

Coming soon - **Flanger/Doubler (Model 522)** an analog delay in two ranges designed to cover both flanging and doubling effects.

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REGULARS

Cover Photo By: Vesta Copestakes

Letters

Multi-Human Mixdown

I enjoyed Chris Meyer's article on Live Plus Tape Techniques, and thought the following comments might be of interest.

With regards to mixing down a multi-track tape with multiple humans, while automation has certainly taken its place in the contemporary world of electronic recording, nothing will replace 8 hands or 16 hands on a mixing console - especially when there's a score to orchestrate dynamics and such. This takes some practice, but with a little practice you can achieve incredible results.

I've done this on two or three occasions, and the results were excellent. While I've found automation is great for eliminating a bad sound or glitch, when it comes to true articulation of a sound or real dynamic expression, nothing beats the hand. Co-ordinating 16 hands together is not all that difficult; actually, the hardest part is getting them to fit in the space with the mixing console. Once that has been accomplished, the results of multiple human mixdowns are generally quite good.

Jay Lee
Norwell, MA

(Editor's note: Jay's interviews have proven very popular, so rather than have you all write in and ask why there isn't one in this issue, don't worry - he's real involved with finishing up his album, and the interviews will resume shortly).

Questions ....

I would like to see a continuation of the AMS-100 series. I remember you saying you had some plans for a continuation of the series, but I don't recall what they were. Now I know you're going to say, "How can I be an editor and do the AMS too?" Pretty impossible, I should say. What I would suggest is to let the readership know that the electronic guitarists are still out here...if anyone has any modules that they have expanded their AMS with, let us know about it.

Finally, whatever happened to the Gizmo? I am really interested in it, and I'm quite sure that a lot of the other readers are scratching their heads too.

Joseph Mosesso
Quincy, MA

Joseph - I'm glad you brought up the point about having more articles for electronic guitar. Don't think this is exclusively a magazine for synthesists; it's a magazine for people who are into musical electronics. I do have some more AMS-100 modules I'd like to present, but you're correct about time being a big problem. In the meantime, check out Jacques Boileau's AMS-100 compatible flanger clock in this issue; also, you might note that the Chorus I presented in the January 82 issue of **Guitar Player** is AMS-100 compatible and synchro-sonic to boot. There will be more, but like you said, there's only so much one human being can do.

Re the gizmo: As I understand, there were two main problems. First, they couldn't make the little wheels that hit the strings durable enough; second, the guitar's pickups would pick up noise from the DC motor that made the little wheels go around. These problems were apparently insurmountable, and to the best of my knowledge the Gizmo is a dead issue for now.

Spiffy Op Amps

Another source for NE5534 op amps is MCP/Davisound (PO Box 521, Bypass 76, Newberry, SC 29108). Price is $2.85 each for more than 25 pieces. Also, Aphex Systems has an IC, the 1537A VCA, which has good noise, distortion, and attenuation specs. Price (1-99 pcs) is $12.00. The sonic quality is excellent. An info packet is available from Aphex Systems, 7801 Melrose Ave., Los Angeles, CA 90046.

Just thought the readers might be interested in where to get these high quality parts at reasonable prices.

Michael Teed
Wizard Productions
Pewaukee, WI

Soundsheet?

Congratulations on the publication of the Jay Lee interview with Harald Bode. I too have the good fortune of knowing Harald personally, and agree that he is a "special man". He has enough information for electronic music artists and engineers to fill volumes, and the Bode Frequency Shifters, Vocoder, and Barberpole Phaser are all beautifully designed instruments.

When I interviewed Harald for National Public Radio, the audience had an opportunity to hear tapes of his history-making instruments as well as the capabilities of his recent devices. I wish the Polyphony article could have included similar tapes. Your readers would have been treated to some amazing sounds!

Walter J. Gajewski
Burbank, CA

Walter - We've looked into binding occasional soundsheets in the magazine, so the same thought has occurred to us. But for now, all you have to do to hear what Harald's devices sound like is turn on the radio!

A Sweeping Question

Craig has often mentioned cascading phasers for a more dramatic effect. How do you sync up the phase sweep? Also, if I built the flanger from DEVICE 1:9, how could I get it to emphasize even or odd harmonics (like the A/DA unit)? Thanks a lot, you guys are great!

Dana Almasi
Seattle, WA

Dana - With voltage controlled phase shifters, you simply plug the same sweep voltage into both phasers. With phasers that have their own sweep circuitry, you can't easily sync up the sweep; however, often the sound of one sweep working against the other produces a very pleasing effect. As for the flanger, with the phase switch in the (+) position you get even harmonics, and with the phase switch in the (-) position, you get the odd ones. Remember also to glance at the corrections in issue 1:10 before you start building!

More on REMCO FX

In response to Dave Wilson's letter (Polyphony, Nov/Dec 1981), the REMCO FX uses the SN76477's brother (sister?), the SN76488. These two chips are functionally 90% identical, and are available (with extensive applications notes) from Radio Shack. For those investigating the 76488's application notes, let me point out two apparent errors: 1) in the logic entries for the Mixer Select Inputs, the data for columns C and A are reversed; 2) there is no difference in the first two entries for the logic table under Envelope Select Logic - Mixer Only applies to both.

Overall, I'd give the 76488 a very slight applications edge compared to the 76477: the 76488 contains its own power amp stage, and its accessible one-shot output makes the "X2" two-stage sound effect possible on the FX. On the other hand, the 76477 permits varied attack (as well as decay) time on its envelope generator.

Many new effects, and much new mileage, can be gotten from the REMCO FX through modification, and the box does have space to accommodate these additions. The back of the goodsized board is readily accessible, and the main modification problem is soldering on to those miniscule IC legs. Incredibly, my original chip survived; it must be rugged indeed!

Incidentally, during a Christmas "special" I picked up one for $10. Quite a bargain if you go for sound toys. (On the other hand I'm also writing a serious piece of contemporary music for it - to explore its musical potential.)

Dr. Arthur B. Hunkins
University of NC
Greensboro, NC

(Editor's note: Dr. Hunkins informs us that he is preparing an article about these modifications for Polyphony.)

Pro-One Interface Credit

This letter is in regard to the article entitled "A Digital Interface For the Pro-One", in

LETTERS Continued on page 32

Electronic Music Math

by: Craig Anderton

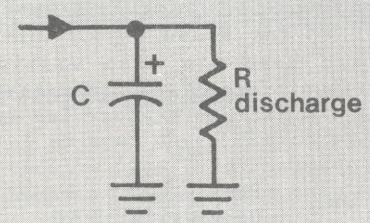


Fig.1

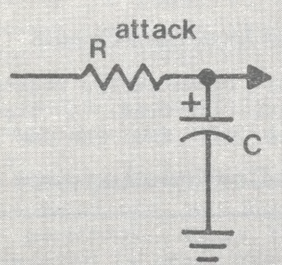


Fig.2

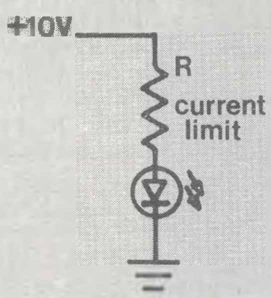


Fig.3

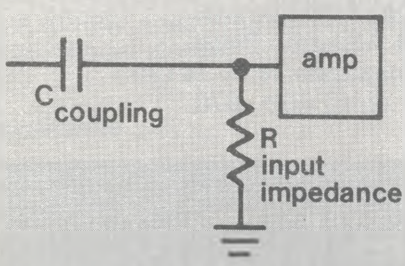


Fig.4

Many people think that electronic design involves sitting down with a computer and solving differential equations. Wrong! All you really need is a knowledge of basic algebra and a $29.95 scientific calculator (I like the kind with reverse Polish entry, but anything that handles algebraic functions and gives logs will do). Scientific calculators are great - for example, if you need to take the log of a number to plug into a formula but you haven't the foggiest idea of what logarithms are all about, no problem; just enter your number and punch the "log" key. Talk about simple.

Following is a list of the most important formulas I use in my design work, with some examples thrown in just to show how easy this all really is.

Time constant equation. Let's say you've got a circuit like figure 1, where a resistor discharges a capacitor (you find this type of circuit in portamentos, slew rate limiters, envelope generators, and so on), and want to figure out different discharge times for different resistor values. Suppose, for example, you want a 1 uF timing capacitor to discharge in 1 second. How do you choose the resistor value that will give this discharge time?

The formula we want is:

T = R times C

Or, time equals resistance (in Ohms) times capacitance (in Farads). Most of the time, it's easier to state the formula in MegOhms and microFarads (since we're multiplying the resistance by 1,000,000, and dividing the capacitance by 1,000,000, according to basic algebra the formula is still valid). So, substituting our real world example, we end up with Time = 1 second, and capacitance = 1 uF; so, R must equal 1 MegOhm. For 1/2 second, R = 500k; for 1/10th of a second, R = 100k.

How about attack time (see figure 2)? Let's say you're designing a limiter with a 10 uF timing capacitor, and want a 10 ms (or .01 second) attack time. First, we need to algebraically transpose our formula so that instead of T = RC, we have T/C = R. Solving for R, we plug in our numbers and end up with .01 seconds divided by 10 uF = .001 MegOhms, or, 1k.

Current drain. One of the most useful formulas in electronics is Ohm's law, which states that voltage (in Volts) = current (in Amps) times resistance (in Ohms), or:

E = IR

Doing some algebraic transpositions gives us two other useful formulas, current = voltage divided by resistance **(I = E/R)**, and resistance = voltage divided by current **(R = E/I)**. Now, suppose you have a circuit like figure 3, and you want to push 10 mA of current through the LED. What would be the correct value for the current limiting resistor? Substituting numbers in our formula, we have resistance = 10 Volts divided by .01 Amps, which gives us 1000 Ohms. Simple, yes?

DeciBel formulas. Here's where your scientific calculator really comes in handy. Suppose you have two peak-to-peak voltages, and you want to know their ratio in dB. The formula is: dB = 20 times the log of (voltage one divided by voltage two), or:

dB = 20 · log (V1/V2)

Suppose we're measuring the

signal-to-noise ratio of an effect, and according to an oscilloscope we have 10V p-p of output signal coming out of the effect, and 2 mV p-p of residual noise with the input signal removed. Getting out our trusty calculator, we find the ratio of V1/V2 is 10/.002, or 5000. We then take the log of that by pushing the log button, and end up with 3.69897. Multiply this by 20 for the answer: 73.9794 dB. That's a pretty decent signal-to-noise ratio.

Or, let's say we have a flanger and want to calculate the depth of the notches. We put a constant signal (like a sine wave) through the flanger, and check its maximum value on either side of the notch - say, 2V p-p. Then, we carefully tune the flanger to notch out the signal, and we find the minimum (notched) signal level is 32 mV p-p. Plugging into our formula, we have 2/.032 for the ratio of V1 to V2, or 62.5. We take the log of 62.5, which is 1.79588, and multiply that by 20, which means our notch depth is 35.917 dB (36 dB is close enough).

Coupling capacitor formulas. Suppose we have a circuit like figure 4, where we need to capacitively couple into a stage with a 100k input impedance. If you use a .1 uF capacitor, at what low end frequency will the response be down by 3 dB? Well, the formula is **Frequency (in Hz) at which the signal is down by -3 dB = 1 divided by (2 pi times Resistance times Capacitance).** Resistances are in Ohms and the capacitances in Farads, but to simplify life, we can express the resistance in MegOhms and the capacitance in microFarads. Also, let's just say 6.28 for 2 times pi. This gives us a more manageable:

F = 1/ (6.28 . R . C)

To solve our problem, we substitute the numbers given above and come up with:

F = 1/ 6.28 (.1 MegOhms)(.1 uF)

This comes out as 15.92 Hz, certainly an adequate low frequency response. Had we coupled in with a .01 uF capacitor, we would end up with a low frequency -3 dB point of 159.2 Hz.

This same formula also works for figuring out high frequency limits. Suppose you have a circuit like figure 5, where you need to limit the high frequency response of an op amp because otherwise, it oscillates like crazy. Plugging numbers into our formula, if we put a .1 uF capacitor in parallel with the feedback resistor, the high frequency response would be down by -3 dB at 15.92 Hz. Putting in a .01 uF cap means the response would be down by -3 dB at 159.2 Hz. A 1000 pF cap would put the high frequency limit at 1.592 kHz, and a 100 pF cap would give us 15.92 kHz. Since that's a little lower than we might like for full audio bandwidth, let's try a 47 pF capacitor instead. That gives us a -3 dB point of 33.863 kHz, which is fine for our applications.

Period and frequency. After all that log stuff, let's get back to a real easy formula. Suppose you're measuring a signal with an oscilloscope, and according to the scope's time base, the period of the signal is 10 milliseconds (see figure 6). What is its frequency in Hertz? The formula we want is:

Frequency (in Hertz) =
1/Period (in seconds)

A close relative of this formula is:

Period = 1/Frequency

So, plugging in numbers from the above example we come up with F = 1/.01, or 100 Hz. Suppose you want to know the period of a 15.962 kHz waveform; now we use the second formula given above, or P = 1/15,962, which equals seconds, or if you like, 62.64 microseconds.

Op amp gain formulas. Here are two more quickies to close out the article. For an inverting amp (figure 7a), the formula for gain is:

Gain = $\frac{\text{R(feedback)}}{\text{R(source)}}$

R can be either Ohms, kOhms, or MegOhms, as long as you express both resistances in the same units. So, a 100k feedback resistor and a 10k source resistor gives us a gain of 10 inverting amp.

A non-inverting amp (figure 7b) is just a little more complex. The formula is:

Gain = $\frac{\text{R(feedback)} + \text{R(source)}}{\text{R(source)}}$

So, again with a 100k feedback resistor and 10k source resistor, we end up with a gain of (100 + 10)/10, or a gain of 11.

Summing up. Believe it or not, the formulas presented above take care of 99% of my design work. I still don't understand transfer functions, and know virtually zero about calculus...but it sure hasn't held me back. So, fellow mathematical idiots, take hope. Get a scientific calculator, practice plugging numbers into the above formulas, and you'll be well on your way to at least <u>appearing</u> that you know what you're doing! ——— #

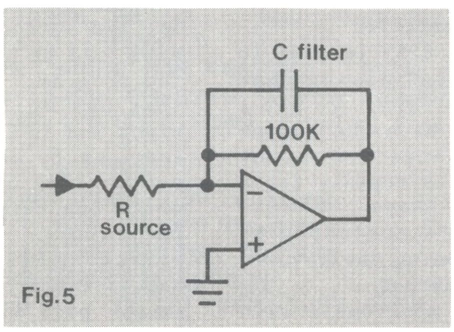


Fig.5

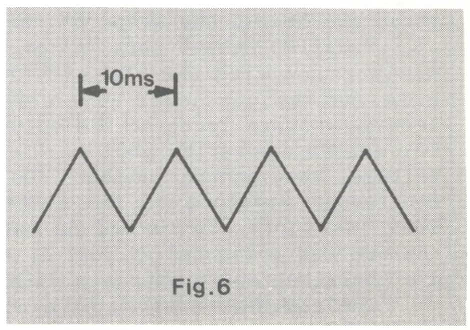


Fig.6

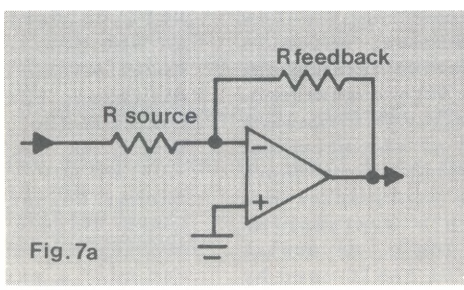


Fig.7a

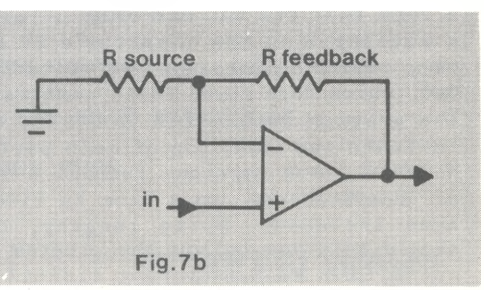
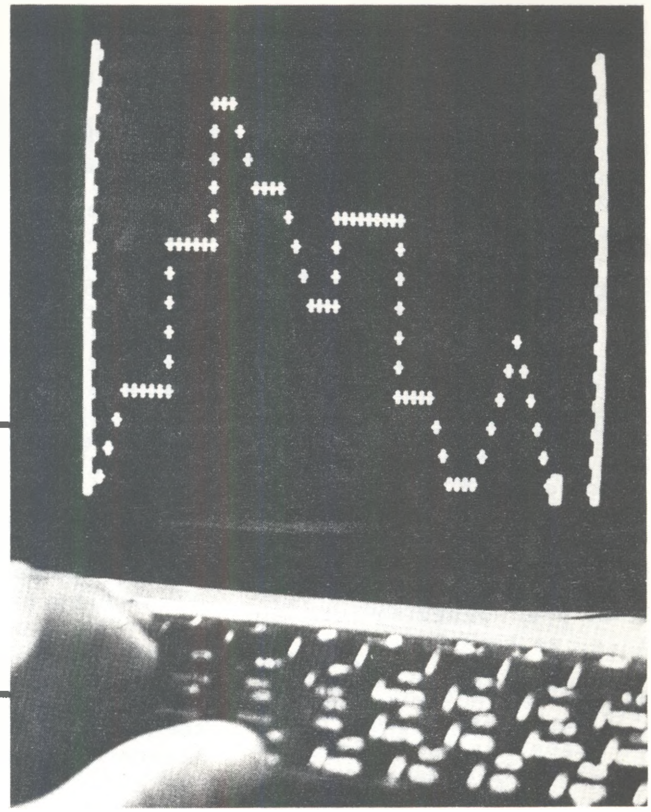


Fig.7b

SCREEN WAVE



This article describes a simple method for using the TRS-80 Model I, Level II (with at least 4K of RAM and a cassette-based mass storage system) as a programmable waveform control voltage or tone source generator. This is a simple, effective approach that can be up and running in an hour's time, and can also serve as a starting point for further experiments.

By: James Lisowski

Overview. In addition to the computer, this system uses a mixture of hardware (electronic circuitry) and software (computer programming). The hardware in figure 1 creates a simple one chip interface to the TRS-80. The 8255 chip is a most useful device, as it contains three input or output ports (channels of information). One of these ports is used a digital to analog converter, abbreviated as D/A or DAC. The remaining ports are available for custom control or sensing functions (such as triggering VCAs or reading a keyboard), although these are beyond the scope of this article.

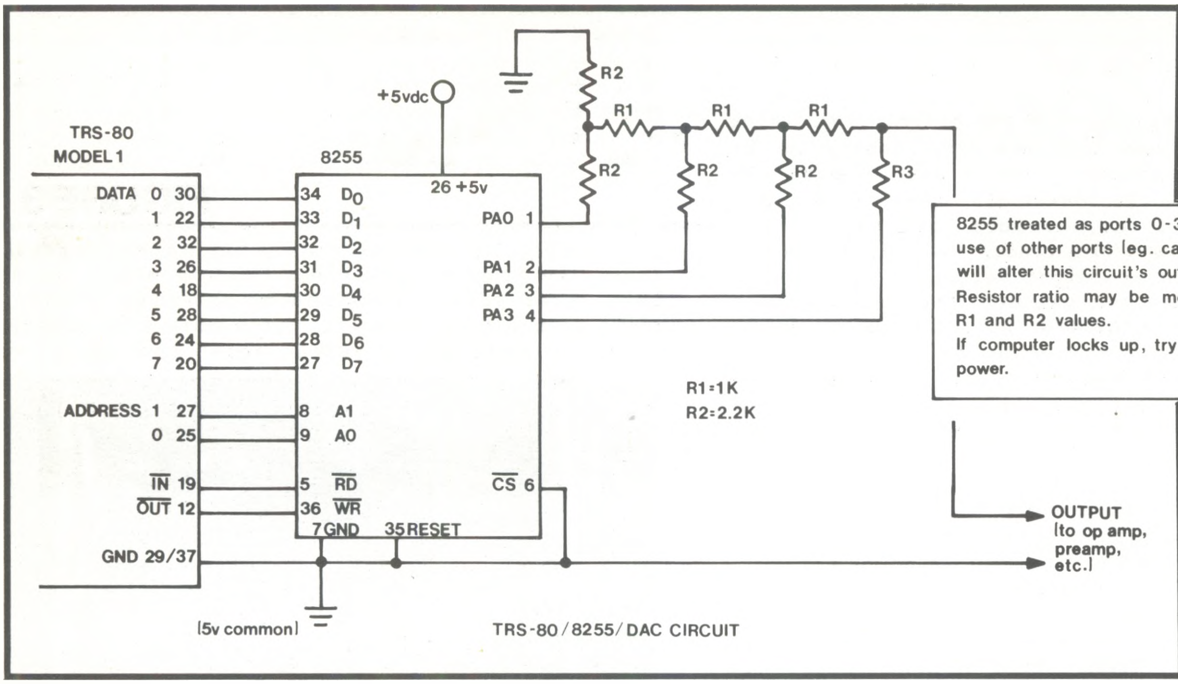
Parallel output port "A" of the 8255 is used in conjunction with the classic "2R" DAC design that PAIA synthesizer users have seen before. To get a certain voltage level at the output, have the computer send a number to port A. To change the output voltage, send another number. Do this rapidly for a continuous waveform of the desired shape. Individual resistors are turned on and off according to the numbers presented at port A; summing their voltage contributions produces a final output.

The computer program presented here deals with creating the waveform you desire and getting it into a form that the DAC can use. It allows you to draw a waveshape on the computer's TV screen, which is then converted into numbers suitable for feeding the DAC, which then turns these numbers into a waveform. The program is written in BASIC; various comments indicate the function of each part of the program. However, some sections require a little more depth of understanding, so a line by line explanation has been included. While lengthy, this explanation should help you follow the program logic, or assist conversion to a different dialect of BASIC used by other computers. After comparing the program to the comments, you may not know exactly how the process is done, but you will know what steps were required to get from concept to reality. This should enable you to add to, or modify, the program. This program is placed in the public domain, so no fees are required to use or duplicate it. If you do develop something interesting, however, please write me.

Screen Wave: the program. Type the BASIC program in to the computer in the usual manner, RUN it, and correct any spelling or syntax errors. Then save two tape copies for later use. Upon RUNning the program, you will see a clear television screen that has white borders running vertically on the right and left sides. These borders are broken into sections by small rectangular blocks. Also, at the bottom left of the screen, you will see a blinking white box cursor (the <u>cursor</u> is a moveable position indicator). The borders are guides to vertical position, like the scale of a graph. If you press one of the arrow buttons on the TRS-80 keyboard, the cursor will move in that direction (up, down, left, right). If you press the space bar, a plus sign (+) will be deposited on the screen at the current cursor position, thereby marking one dot of the final waveform. You may now move the cursor to a different position. To erase an existing marker, position the cursor directly over the mark and press the CLEAR button.

By now you probably get the idea: moving the cursor to various positions and leaving a marker there defines points on a continuous waveform. By erasing markers and placing new ones elsewhere you can edit a waveform, altering the sound or voltage

FIGURE 1



8255 treated as ports 0-3, incompletely decoded; use of other ports (eg. cassette I/O uses port 255) will alter this circuit's output - Beware.
Resistor ratio may be more uniform with different R1 and R2 values.
If computer locks up, try less than +5 volts for power.

DAC FOR TRS-80

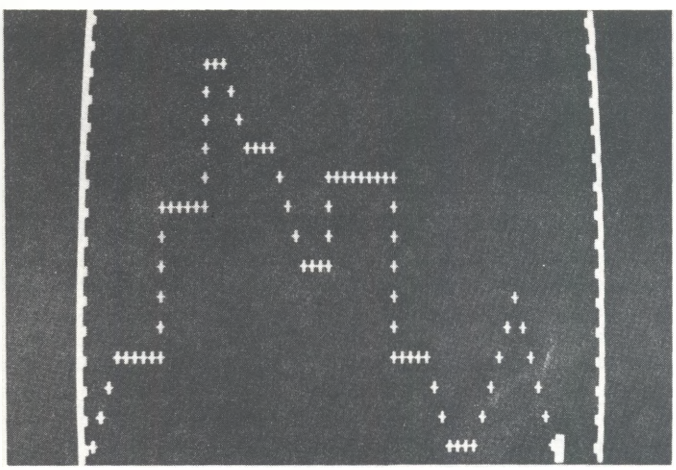
THE PROGRAM

```
5 'CURSOR DIRECTED MUSIC OR CONTROL WAVEFORM INPUT PROGRAM
10 'FOR TRS-80 (TM TANDY) MODEL I LEVEL II 4K AND MORE RAM
11 'AUTHOR: JAMES A. LISOWSKI 902 WILLOW LN., S.MILW.,WI 53172
12 'V1.1  3-18-81
20 DIM W(61):'SAVE WAVEFORM VALUES IN W(0 TO 61)
50 P=16321:M=43:'CURSOR START POSITION, MARKER CHARACTER (+)
55 U=14400:'CONTROL KEYS/ARROWS KEY MEM LOC
80 CLS:'CLEAR SCREEN, DRAW GUIDES
90 FORV=15360TO16320STEP64:POKEV,157:POKEV+63,174:NEXT
100 K=PEEK(U):I=0:'GET CONTROL/ARROW KEY, RESET POSITION INCREMENT
199 'CHANGE INCREMENT ACCORDING TO ARROWS
200 IFKAND8THENI=I-64:'UP
210 IFKAND16THENI=I+64:'DOWN
220 IFKAND32THENI=I-1:'LEFT
230 IFKAND64THENI=I+1:'RIGHT
299 'DON'T MOVE OFF SCREEN
300 IFC+I>16383ORC+I<15360THENC=PELSEC=C+I
399 'BLINK CURSOR/EXISTING CHARACTER
400 T=PEEK(C):POKEC,191:POKEC,32:POKEC,T
500 IFPEEK(U)=128THENPOKEC,M:'SPACE BAR PRESSED, LEAVE MARKER
510 IFPEEK(U)=2THENPOKEC,32:'CLEAR PRESSED, ERASE CHARACTER
520 IFPEEK(U)=1THEN2000:'ENTER PRESSED SCAN DRAWING FOR VALUES
999 'KEEP DOING IT
1000 GOTO100
2000 S=16321:'BOTTOM LINE START POSITION OF SCAN
2010 FORX=0TO61:'SCAN 62 POSITIONS, LEFT TO RIGHT
2100 FORV=0TO15:'SCAN 16 POSITIONS BOTTOM TO TOP
2110 W=(X+S)-(V*64):'CALC VIDEO MEM POSITION
2141 'IF IT IS A MARK, CHANGE IT, IF NO MARK JUST ABOVE IT, STOP SCANING THIS CO
LUMN, SAVE WAVE VALUE IN W()
2150 IFPEEK(W)=MTHENPOKEW,140:W(X)=V:IFPEEK(W-64)<>MANDV<>15THENV=16
2200 NEXTV,X
3000 FORI=61TO0STEP-1:IFW(I)=0THENNEXT:'READ WAVE ARRAY BACKWARDS TILL LAST NON-
ZERO VALUE (END OF WAVE) FOUND
3010 NP=I:'NP IS THE NUMBER OF POINTS (VALUES) MAKING UP THE WAVE MINUS ONE
3020 I=-1:'FINISH I LOOP
3999 '-----CUSTOM USER ROUTINE MAKING USE OF WAVE DATA GOES HERE
4000 '======DUMMY D/A DRIVER======
4005 PRINT"** WAVE BEING OUTPUT **";
4010 FOR I=0TONP:OUT0,W(I):NEXT:GOTO4010
```

SCREEN WAVE

sequence it will produce. Since the creation and editing process is visual, you can easily experiment with waveforms, to "see what it sounds like".

Using the program. RUN the program, and using the up arrow key, move the cursor vertically to the desired height position. There are 16 possible vertical positions, the lowest (0) being the minimum amplitude value of the waveform, and the highest (15) being the maximum amplitude. Each intermediate position represents an output voltage level between the minimum and maximum values. For this hardware configuration, min = 0.25V or height value 0, while max = 2.5V or height value 15; adding an op amp level shifter can change this voltage range. When the cursor is in the right place, press the space bar. Then move the cursor to the next column to the right, using the right arrow key. Move the cursor vertically in this column to the desired height and mark this new point with a press of the space bar. Continue in this fashion until you have defined your waveshape or have reached the maximum of 62 horizontal point positions. If you are satisfied with the waveshape you have drawn, press the ENTER key to complete the process and output the waveform. To alter the waveshape (this must be done before you press ENTER), use the arrow keys to move the cursor to the location of the marker(s) you would like to change and press the CLEAR key to erase that marker. Then, move to a different vertical position and leave a new marker. Repeat as desired, then finally press ENTER when finished.



The above sequence is the input phase of the program. Pressing ENTER indicates to the computer that you are finished defining a waveshape. The program will then automatically scan the screen for markers, determine their height values in sequence, save this information, and start to output height values to the DAC. When a marker is found, it is changed into a white block so that you can follow the scanning process. There will be several seconds of delay as the scan takes place. The program automatically counts the number of points in the waveform for you; see figure 2 for some sample waveforms.

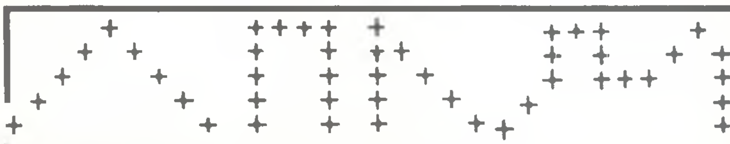


FIGURE 2 Sample Waveforms

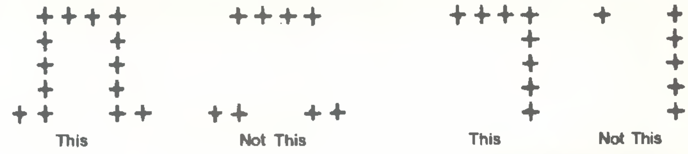
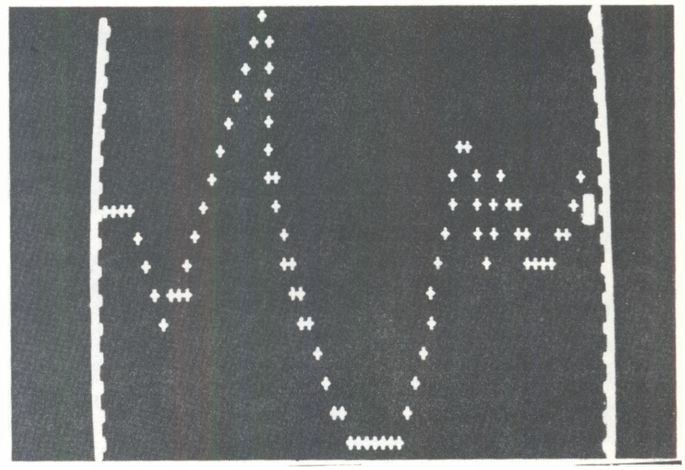


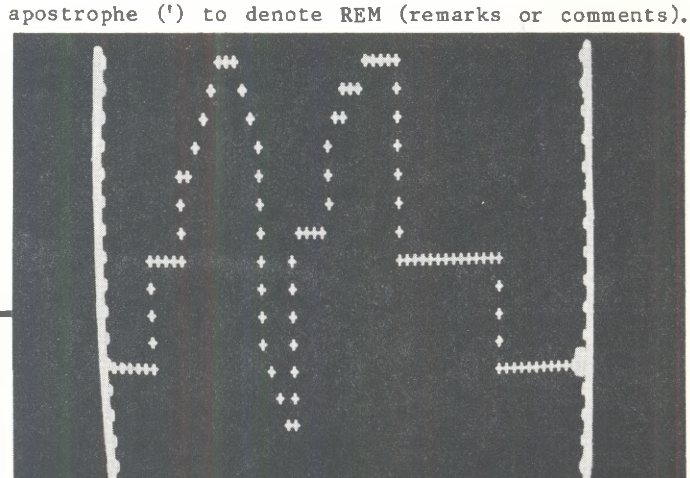
FIGURE 3

Note: In order to reduce the screen scanning time, the program determines the top marker in any particular vertical column by checking to see if there is a marker above it. If a marker is there, a check is made for a marker above that one, etc.



When the top position has been reached or there is no higher marker, the last position is saved and scanning continues with the bottom of the next column to the right. Because of this procedure, all totally vertical wave segments must be a continuous line of markers. You may not define a vertical segment by placing one marker at the bottom and another marker several blank positions above the first marker (see figure 3). Going horizontally, in a leftward direction, each vertical column from the start to the end of the waveshape must have at least one (though possibly more) marked position(s). In other words, the waveshape must be horizontally continuous. Other than these constraints, any other wave configuration is possible.

How the program works. Here are some background notes on the computer system and TRS-80 BASIC. TRS-80 BASIC uses the colon (:) to place more than one BASIC statement on a line, and an apostrophe (') to denote REM (remarks or comments).



The PEEK statement reads a decimal value from a particular memory location and a POKE statement puts a decimal value in to memory. Memory locations in the 15360 to 16383 (decimal) range are the visible screen memory (POKE places characters on the screen). Decimal memory location 14400 represents the arrows and other keys. When PEEKed it returns a decimal value of 1 if the ENTER key is pressed, 2 for CLEAR, 8 for up arrow, 16 for down arrow, 32 for left arrow, 64 for right arrow, and 128 for space bar in a Model I system. This information should help in converting this program for other computing systems. Now, let's move along to the line by line comments.

Lines 5-12 identify the program. Line 20 (L20 for short) sets up a storage area (called "W") that has 62 subparts, labelled 0 through 61. The program can store a number (the individual waveform height value) in each of these subparts. L50 defines the character used for a cursor. In this case, the number representing a plus sign is selected but you could make it any character you like. L50 also defines the cursor's starting place in the screen memory. L55 sets the variable (a temporary storage location) named "U" to the decimal value of the memory location assigned to the arrow/control keys so that this number doesn't have to be repeated later in the program. L80 uses the CLS command to clear (blank out) the video screen. L90 draws the white vertical height guides by POKEing graphics characters into the screen memory. The FOR/TO/STEP-/NEXT statements in this line repeat the POKEing and specify the memory locations to be POKEd. Up to this point, the program has been just setting the stage; the rest of the program from here on does most of the actual work.

L100 - L299 check the TRS-80 keyboard to find out what buttons are being pressed. Variable "I" holds the number of memory positions the cursor should move with respect to the current position; it starts out with a value of zero each time the program executes L100. The values associated with each key are summed into "I" if that key is pressed (determined by IF statements in L200 - L299). L300 checks to see IF the new cursor position (the sum of "C" and "I") might be out of the range of screen memory, placing it off screen. IF this condition is true (off screen), THEN don't alter the cursor position, otherwise (ELSE) allow cursor movements to the indicated position (calculate a new current cursor position, "C"). L400 reads the value in the current cursor video memory position, saves it temporarily in variable "T", POKEs a white block graphics character on to the screen, blanks the block out by POKEing a space, and then POKEs the original character back again. This gives a blinking effect to the cursor while allowing you to see any character "under" it. L500 checks to see IF the space bar is pressed. IF it is, a marker (+) is POKEd into the cursor location (IF not, no action occurs, and execution continues with the next line. This is used to draw the waveform. Similarly, L510 erases a marker IF the CLEAR key is pressed. L520 will direct program execution to the section starting with L2000 IF the ENTER key is pressed (when waveform definition is finished). Otherwise, the next statement the program will GOTO is L100 (which starts the main cycle over again, reading the keys, moving the cursor, etc. until ENTER is pressed).

L2000 - L3020 are the screen scan section. The process, in summary, is this: scan the video screen memory in a bottom to top, left to right sequence, checking for markers. When the top marker in any vertical column is found, calculate and save its height value in the associated subpart of "W" for later use. In more detail, the scan starts at the lower left video memory location and checks the position "X" units over and "V" units up (L2000 - L2110) for a marker. IF on is there, THEN POKE a small white block into that position and also save the height value in the "X"th subpart of "W". If there was a marker here, check the position above it for another marker or check to see if the scan has reached the maximum height position. IF either of these two conditions are true, stop scanning this column and move on to the next. IF false (there is a marker above and column top not reached), continue scanning, POKEing, and saving height values for this column; repeat this scan for the next "V" and "X" positions (L2200). Do this for a possible 62 horizontal positions (L2010), each time checking 16 vertical positions (L2100). When done, continue with L3000.

When the program was RUN, all the subparts of "W" were set to zero. As the scan section progressed, the zeroes were replaced by height values. If your waveform had markers in all 62 positions, "W" would now be filled with values. If the wave was less than 62 columns long, some of the original zeroes will remain in "W". L3000 scans "W" backwards (STEP -1) from the "left end" until it finds a non-zero value. At this point, the variable "I" contains the position of the last wavepoint (length of the wave, minus 1 - since the count goes from 0

.............CONTINUED ON PAGE 33

FREQUENCY DOMAIN MODIFIERS

By: Mike Beigel

While special effects are commonplace, many musicians still do not understand them well enough to use them to full advantage, or to note the many similarities between seemingly dissimilar devices. In this article (which is a synopsis of a lecture-demonstration given at both the 1981 Midwest Acoustics Conference and the 70th AES Convention), we'll look at devices such as phasers, envelope-controlled filters, flangers, and similar "Frequency Domain Modifiers" . These devices, unlike "static" frequency domain modifiers such as parametric equalizers and graphic equalizers, are interesting from a musical standpoint because their frequency response changes with time, thereby adding a sense of additional motion or "aliveness" to the musical sound input.

General format of frequency domain modifiers. Fortunately, virtually all of these modifiers follow a similar format. Figure 1 shows a generalized system for a frequency domain modifier, which includes two important elements: the signal processing elements, and the control elements. Note that the input signal can go to three places:

- Straight to the output, through an attenuator. This is needed because many audio effects (especially flangers and phasers) combine portions of the original sound along with the modified signal.
- Into the signal processor section. Here, a frequency-response modification, phase shift modification, or time delay alters the signal to provide the main component of the musical effect. These processors will be discussed later.
- Into the control parameter section. This allows the musical effect to respond directly to the music you play into it. Envelope-controlled effects are among the types of processors that use this configuration.

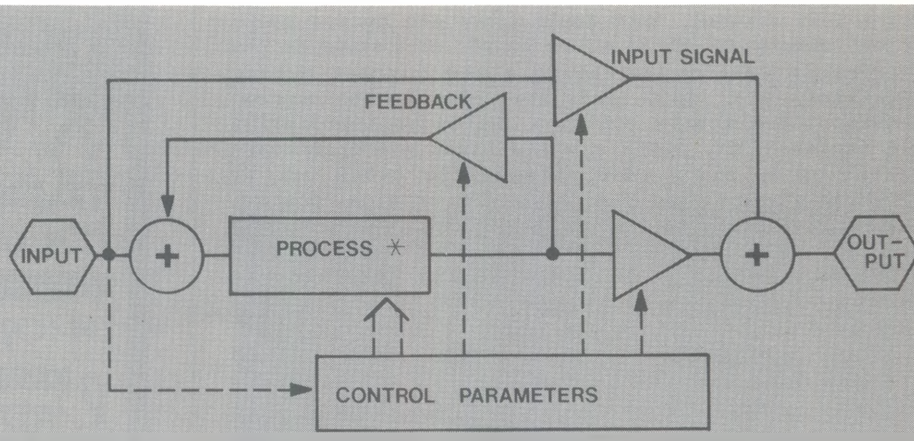
The part of the input signal that goes through the signal processor section has two further destinations:

- To the output, through an attenuator. The attenuators for the unprocessed input signal and the processed signal allow the musician to control the balance between the straight and processed sounds.
- Back to the input of the signal processor section through a "feedback" attenuator. The use of feedback around a frequency, phase, or time process creates a wide variety of frequency response characteristics based on the properties of the process itself.

Notice that the solid lines in the diagram represent audio signal paths, whereas the dotted lines represent control signals.

Types of signal processors. For now, there are three major signal processing elements used in frequency domain modifiers. Hopefully, new processers will be discovered as the technology advances; still, you can create a great many sounds just using these three basic signal modifiers.

- Frequency dependent filters. There are four main filter types: lowpass, bandpass, highpass, and notch filters (see figure 2A - 2D for their frequency responses, with and without feed-



*PROCESS: Filter, Phase Shift Network, Time Delay, Etc.

Fig.1
GENERAL SYSTEM DIAGRAM:
FREQUENCY DOMAIN MODIFIER

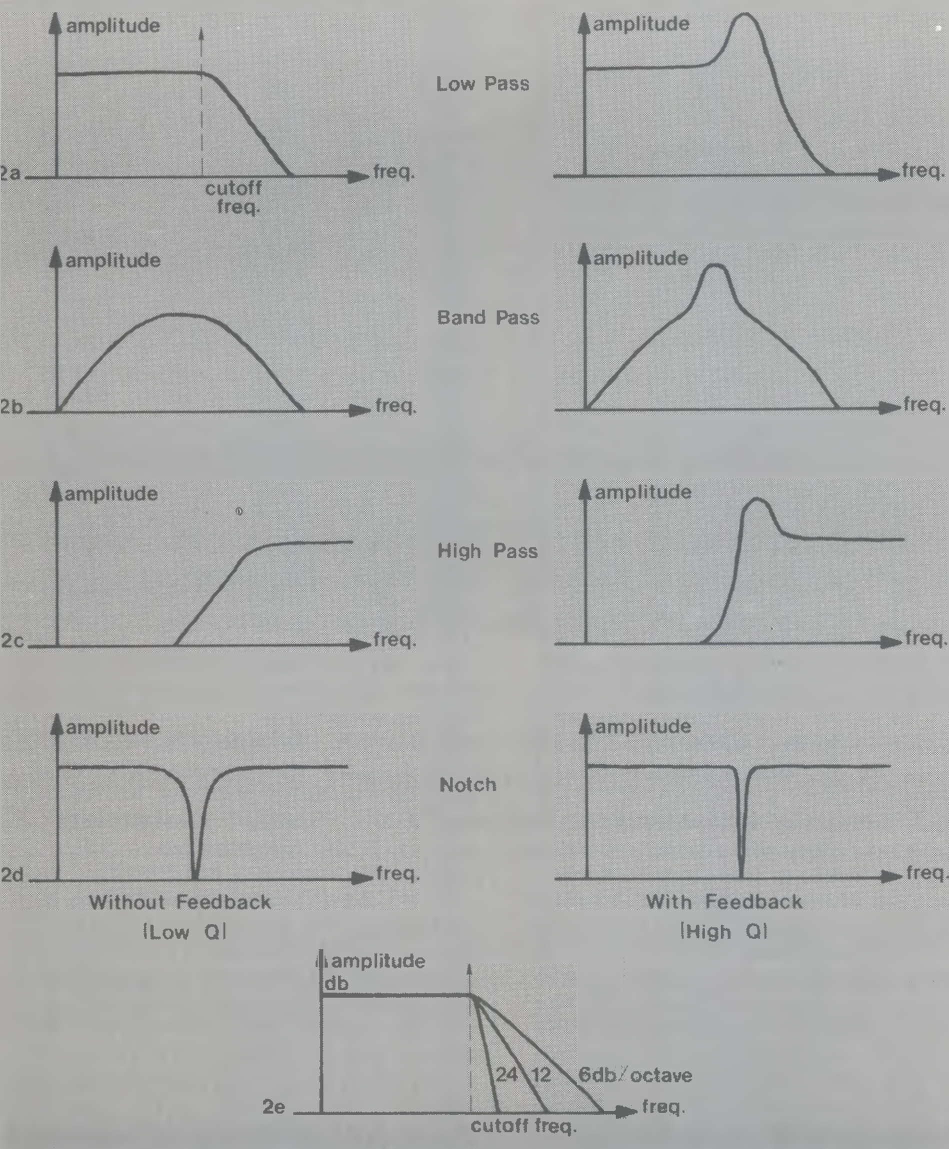


Fig.2
FILTER TYPES: FREQUENCY RESPONSE CURVES

back). You can see that feedback around a filter creates a "peak" at the cutoff frequency of the filter.

Another very important characteristic of these filters is the slope at which the frequency decreases past the cutoff point. This characteristic is easily heard, and figure 2E shows different "rolloff" slopes.

- **Phase-shift networks.** These are filters which have a flat frequency response, which accounts for their other name, "all-pass filters". However, they do <u>delay</u> the input signal, and the amount of delay is a function of the signal frequency. Figure 3A shows this response; the delay is shown in terms of "phase shift" for a single filter stage. Putting more stages in series yields more phase shift delay, but the frequency response still remains constant.

Since the all-pass filter has a flat frequency response, how can it possibly modify the sound of a musical input? The answer lies in adding the phase shifted signal to (or subtracting the phase shifted signal from) the original signal (figure 3B). Due to the time delay, the combined signal includes multiple reinforcements and cancellations in the frequency response. The number of cancellations is proportional to the number of phase shift stages used.

When we add feedback around a phase shift network, we create a multi-pole filter. The basic shape of 3B is exaggerated, and we perceive a distinct difference in sound quality (figure 3C).

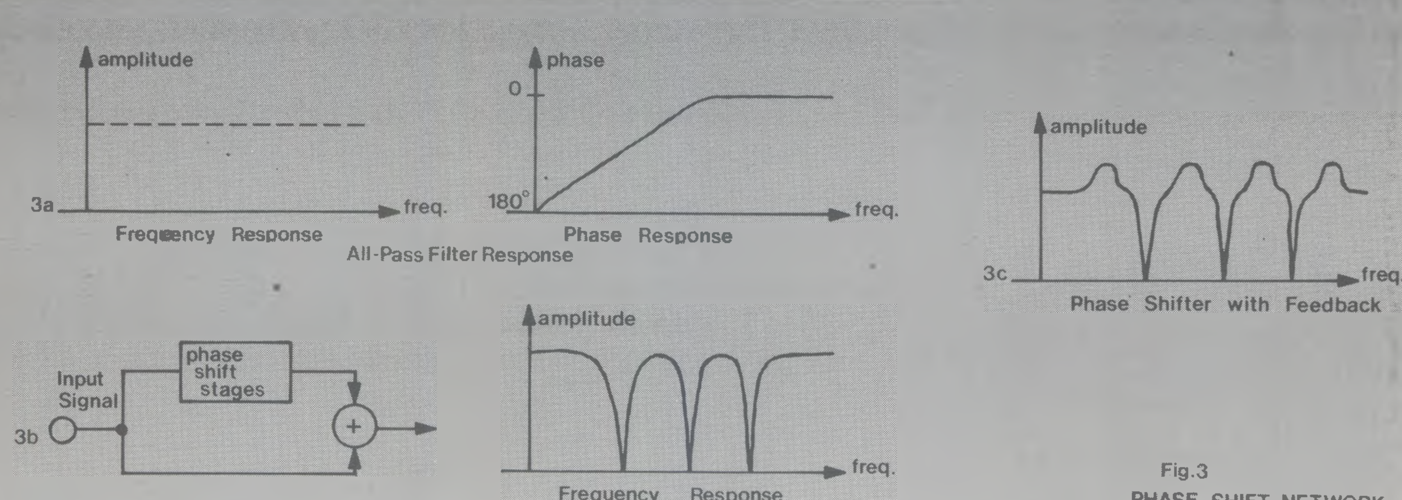


Fig.3
PHASE SHIFT NETWORK

• Time Delays (flangers, chorusers, etc.). Unlike phase shift networks, pure time delays provide the same amount of delay time regardless of the input frequency (figure 4A). Thus, combining the delayed and straight signals produces many cancellations and reinforcements as the delay time represents increasing multiples of the input signal's wavelength. Figures 4B and 4C illustrate the complex frequency response of time delay systems, with and without feedback.

One interesting aspect of time delay systems is that the nature of the audio effect changes at different delay times. Short delay times produce flanging, while the same basic system produces chorusing, "hard reverberation", doubling, and echo functions with progressively longer delays.

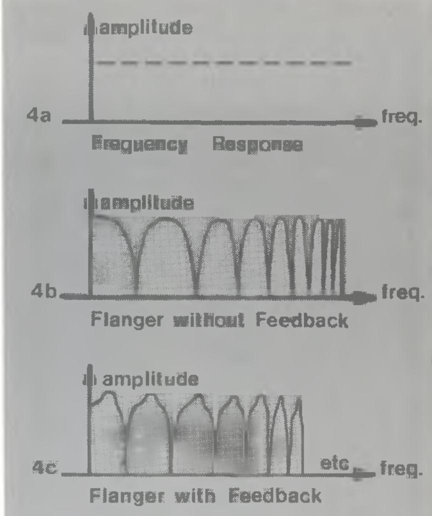


Fig.4
TIME DELAY NETWORK

Control elements. In terms of controlling a frequency domain modifier, we're interested in two main aspects: the nature of the control signal, and what parameters of the audio processor are manipulated by these control signals. Referring back to figure 1, note that every signal path in the general system is controllable. Also, different types of signal processors have different controllable parameters; a filter, for instance, might have its cutoff frequency, mode (highpass, lowpass, etc.), and feedback as controllable parameters.

"Control signals" is a term that encompasses a wide variety of system concepts and circuit elements. I define the term as any method of specifying a parameter of the signal processing system. A controller doesn't have to generate an externally measurable signal as long as it can affect the system parameters. Here's a list of controllers presently used in effects systems:

• **Preset control.** The product designer has decided on a fixed value for a parameter, and the user has no access to it.

• Knob. Controls a parameter from the front panel of the device.

• Foot pedal. The user controls a parameter with his or her foot, allowing control of the system while playing an instrument requiring both hands (example: wa-wa pedal).

• Low frequency oscillator (LFO for short). An automatically varying control function, providing a regular periodic sweep of a process parameter (example: phase shifter sweep).

• Low frequency filtered noise. Another automatically varying control function, but this provides a "random" control voltage with no determinable regularity.

• Sample-and-hold. At a periodic time interval, this devices captures (samples) the value of another signal source (such as an LFO or filtered noise generator), and holds that value until the next sample occurs.

• Envelope follower. This interactive control device generates a control signal proportional to the amplitude envelope of an audio signal present at its input. Hence, you can control parameters of the effect directly by your playing style (example: envelope controlled filter).

• Note follower. This control device generates a control signal proportional to the number of notes being initiated by the musician. Playing more notes in a given period of time gives a higher control voltage than playing fewer notes in the same period of time.

• Triggered envelope generator. Like the envelope follower, this responds to your playing style, but this device triggers a presettable (ADSR or similar) envelope control voltage each time you play a new note. The control signal may also vary according to how loud you play.

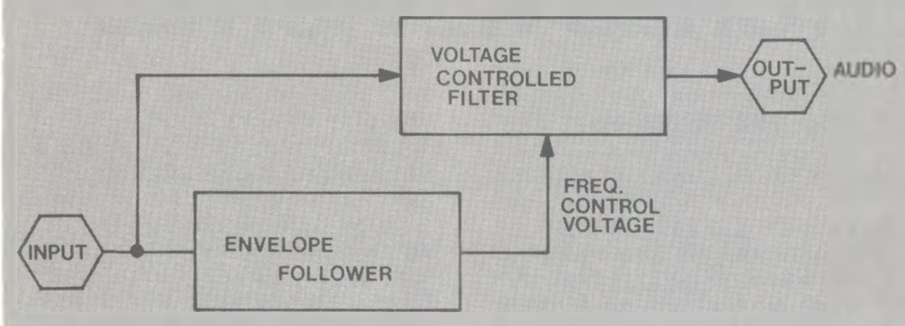


Fig.5
ENVELOPE CONTROLLED FILTER
SIMPLE SYSTEM

• Pitch-to-voltage converter. Provides a control voltage related to the frequency of the note you are playing. This signal can control synthesizer oscillators as well as sound modifying parameters.

Creating a frequency domain modifying system. With so many types of processors and controllers to choose from, it seems unusual that the products now available to musicians only represent a few of the possibilities obtainable with today's signal processing and control methods. As an example of the design options that affect a product's personality, let's look at two ways to make an envelope controlled filter system. Figure 5 shows the simplest possible way: the audio input signal drives the envelope follower, which sweeps the filter, which filters the audio input signal. This gives an automatic "wah" sound with virtually no user controls. On the other hand, you could control every possible aspect of signal processing for this kind of system configuration and end up with the system shown in figure 6. Both kinds of products fill specific needs for musicians, especially with regards to cost-effectiveness.

Frequency domain modifiers are musically useful, not too complex technically, and able to greatly alter the tonal qualities of a musical instrument or tape recorded track. While the purpose of this article was to illustrate the great commonality between this family of devices, remember that the field of musical effects is constantly changing; hopefully future frequency domain modifiers will add on to the existing format described above, resulting in new classes of sound modification devices. #

Michael Beigel currently heads Beigel Sound Labs, manufacturers of a studio-quality Envelope Controlled Filter and distributors of the Dan Armstrong series of effects. He is perhaps best known for designing the Mutron III, Mutron Bi-Phase, and several other sound modifiers, as well as for his articles in Sound Arts magazine and the AES Journal. He divides his time between product design and consulting.

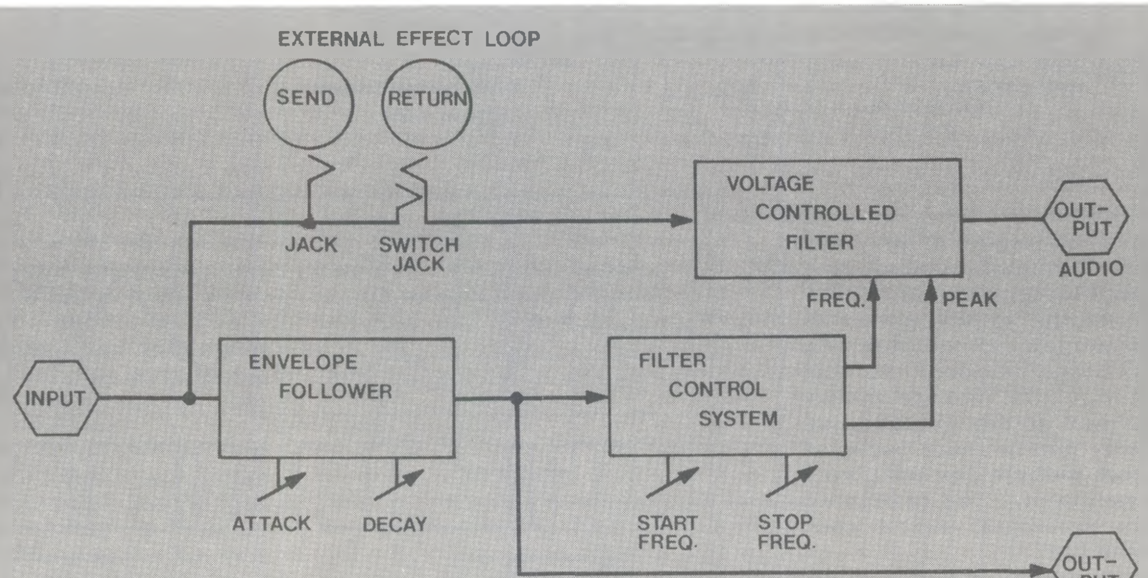


Fig.6
ENVELOPE CONTROLLED FILTER
ADVANCED SYSTEM

TOUCH SWITCHES REVISITED

By: Bill St. Pierre

Not another touch article! But don't worry, these are improved models with hysteresis for extra reliability.

There are two basic varieties of touch switches: latching (touch on/touch off), and non-latching (output active only with switch depressed). We'll describe the non-latching variety, although you can always turn it into a latching type by following the touch switch with a flip-flop.

Touch switch problems. Most non-latching switches suffer from a reliability problem, since the switch will not only sense your finger when you touch the switch, but may also sense your finger prematurely if you're slowly approaching the touch plate. Depending on the sensitivity of the circuit, and the actual proximity of the fingertip, an interesting (annoying) thing happens. Instead of a single, clean voltage change at the output of the circuit, we may suddenly have a short burst of pulses, each pulse a few milliseconds in duration. The effect is similar to mechanical switch contacts "bouncing".

How it works. Figure 1 shows how I solved the problem, which is based on a design by Steve Wood. For a rigorous explanation of how touch switches work electronically, see his article in the Sept/-Oct 1978 issue of <u>Polyphony</u>. The basic idea is that a high frequency (50 kHz) clock passes through D1; the positive half passes through D2, thereby charging R2 to close to supply voltage. This means that the input of IC2 is "high", so its output is "low".

Touching the touch plate introduces enough capacitance so that it becomes charged when the positive half of the clock passes through D1. As the clock goes negative, the finger's "capacitor" tries to discharge through 680k, but can't discharge fully before another positive clock pulse hits through D1. This means that the input to IC1 stays high, so it's output stays low and reverse biases D2. As soon as the charge on C1 bleeds off through R2, IC2 goes high, thus indicating that your finger is touching the plate. Of course, all this happens very rapidly.

Improving the touch switch. Steve Wood's circuit buffered the output of the touch switch with another CMOS gate. The only problem with this is that the voltage present on the finger capacitance at the touch pad varies greatly as your finger approaches (or moves away from) the pad (figure 2). So, if you use a standard CMOS gate that changes state at one-

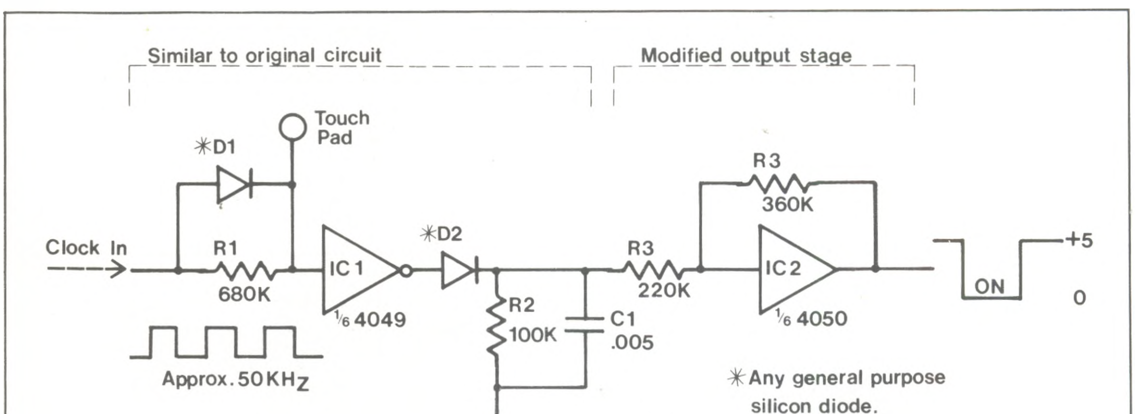


Fig.1

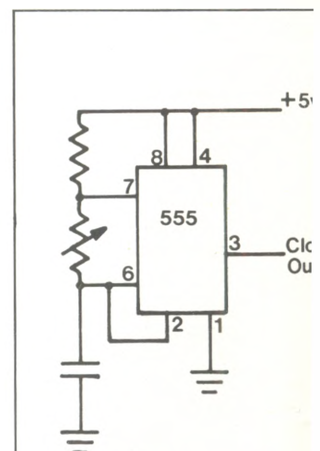


Fig. 5a

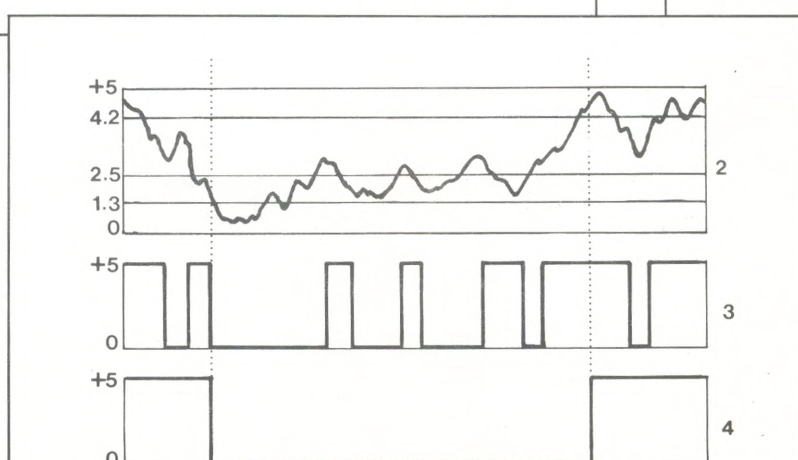


Fig.2,3&4

half the supply voltage, you can get false triggering as shown in figure 3. However, if we apply some hysteresis around a buffer, then the circuit acquires more noise immunity since the buffer now has two trip points. The voltage must drop below about 1.3V to turn the touch switch off, but rise above 4.2V to turn it back on. No matter what happens in between these two points, the switch will remain stable.

Clock sources. Figures 5a and 5b show two sample circuits for clock drivers. 5a is best if you need to feed a lot of switches, because of its large drive capacity (200 mA). 5b is simpler, and only uses 1/4 of a 4093.

Adjusting sensitivity. Varying the clock is one way to create a "master sensitivity" adjust for a group of touch amps, but unfortunately, due to component tolerances each keypad still has a slightly different "feel" or trip point. Giving each key its own clock would solve the problem, but add complexity and cost, not to mention loss of the master sensitivity capability. The simplest way to adjust sensitivity is to replace R1 in figure 1 with a 1 Meg trimpot; now you can tweak each key to match its neighbors, and still use the clock to vary the overall sensitivity.

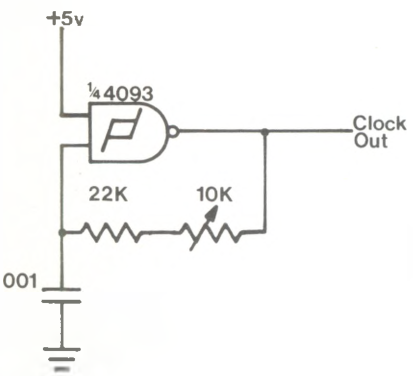


Fig.5b

Construction. Construction of touch pads or touch arrays is pretty easy. Using a resist pen (or a brush and bottle of resist laquer), draw the pad layout on a piece of single-sided circuit board and etch it. Drill a 1/16" hole in each pad, pass through a connecting wire, and then tin the pad with a liberal amount of solder. This holds the wire in place, and the "bump" of the solder also serves as a tactile locator for your fingers. Lengths of stranded wire can then be used to connect each pad to its respective touch amp. The pads should be painted, as they operate on body capacitance and direct electrical contact with fingertips can sometimes cause noise spikes in the circuitry.

Multiplexing the keypads. The touch amp outputs are CMOS and TTL compatible. In the case of a touch activated AGO keyboard, two octaves of notes/keys might be used in conjunction with 5 or 6 octave select pads for a wide pitch range. Interfacing to a computer is fairly straightforward using its input ports; each touch amp output line would go to a particular bit on a particular port. Of course, with maybe 30 lines from the AGO setup and 10 to 20 others devoted to special functions, we'd be dealing with about 50 bits of data. Since the average computer port has 8 bits clumped together into a byte, our 50 bits would require a little over 6 ports of input capacity. This is more ports than most computers have lying about, not to mention the clutter of wires running between it and the controller case.

Figure 6 shows a quick and dirty way of multiplexing this mess down into something manageable. This method only requires a computer with one input and one output port, and a handful of low-cost chips. A maximum of 64 lines (bits) of controller data could be handled and processed.

Basically, the circuit feeds outputs from the touch amps to the inputs of several 74LS373s. These "octal D-type transparent latches" have three-state outputs, which means that when the output is enabled (by taking pin 1 low), "hi" or "lo" data appears at the output, as expected. However, by taking pin 1 high we disable the output, which forces them into a third, high-impedance (open circuit) state. By tying the 8 outputs of the 74LS373 in parallel with several others, and being careful to enable only one of them

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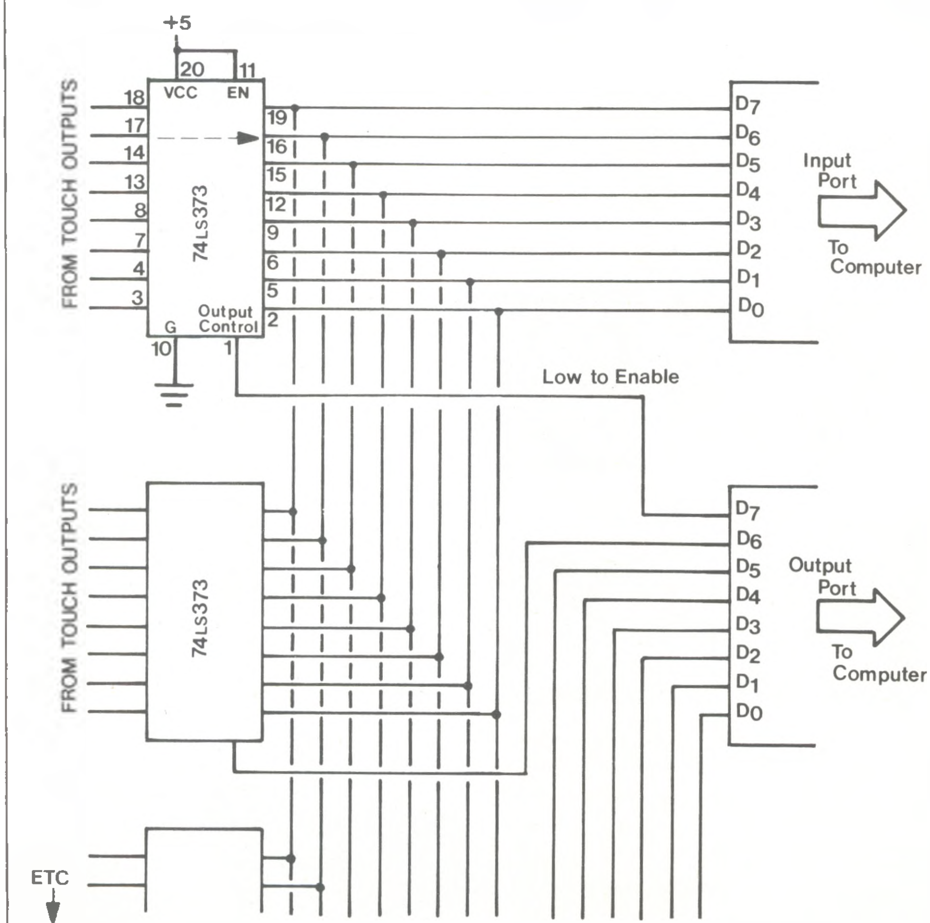
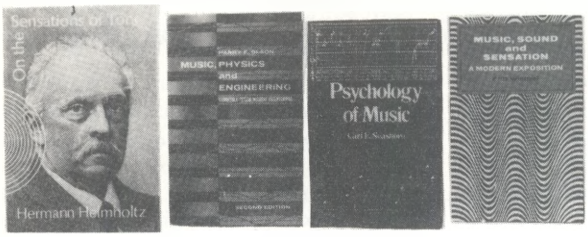


Fig.6

POLYMART

SCIENCE



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

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

TECHNIQUE

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

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

ELECTRONICS

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

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

REFERENCE

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

#SOURCE	The Source	**$4.00**	**#EMS**	Music Synthesizers	**$6.95**
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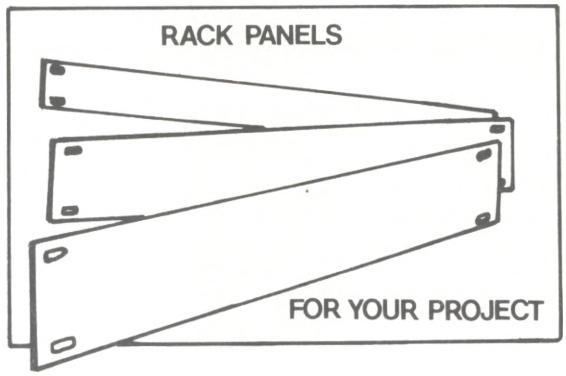

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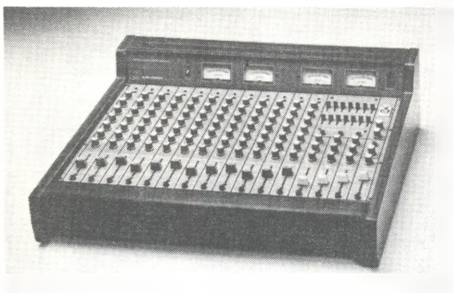
CURRENT EVENTS

'Tell Them You Saw It In Polyphony'

16 bit A/D converter. Burr-Brown (PO Box 11400, Tucson, AZ 85734) has introduced the PCM50KG, a 16 bit A/D converter designed for pulse-code modulation systems. Typical settling time is 5 microseconds, dynamic range is specified as 96 dB, and distortion is under 0.02% distortion at -15 dB. Price is under $50 in hundreds.

Services for Prophet 5. Syn-Comp offers two low-cost services for Rev 3 Prophet 5s. One is combining programs from two or more program cassette dumps on to a third cassette dump; the other is providing printed listings of cassette dumps which include both the switch positions and numeric pot values. For more information, and instructions on what form dumps should take, write Syn-Comp at PO Box 7471-A, Birmingham, AL 35253.



New mixers. Audio-Technica (1221 Commerce Dr., Stow, OH 44224) is now making two mixers, the ATC820 (eight channel) and ATC1220 (twelve channel). Features include hi/mid/lo EQ, master output stereo graphic, p-pop filters, built in headphone amps, and provision for transformer balanced inputs/outputs.

New music publications. OP, a tabloid format magazine, is required reading for anyone interested in independent music. The current issue contains a slew of reviews of independent releases, index of previous reviews, news of local music scenes, and occasional ads. Sample issue is $2; subscriptions are 8 issues/$8 ($16 first class). OP, Lost Music Network, PO Box 2391, Olympia, WA 98507.

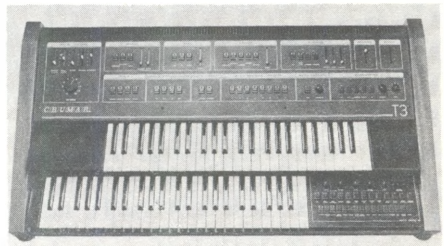
Surface Noise covers progressive, experimental, microtonal, and other modern music through articles, reviews, and artwork. For info on subscriptions and back issues, write John Loffink, 428 Citrus Road, Melbourne, FL 32935. Of particular interest is listings of stations that play independent and new music.

Interface, now in its 9th year of publication, covers "...borderline areas between music, science, and technology". Published quarterly and totalling 260 pages, subscription cost is US$ 63.75. For more information write Swets Publishing Service, PO Box 825, 2160 SZ Lisse, The Netherlands. By the way, most articles are in English. For USA subscriptions, send remittance to Swets North America Inc., PO Box 517, Berwyn, PA 19312.

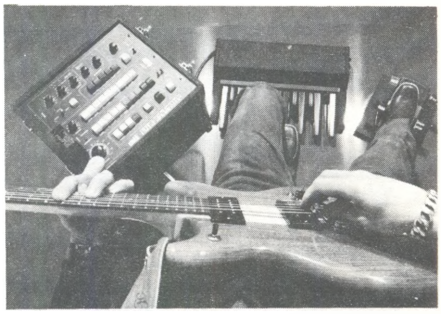
NEW TAPE RECORDERS. TASCAM from TEAC, 7733 Telegraph Road, Montebello, CA 90640, has introduced the Series 30 family of cost effective recorders. Units include the 8 track model 38 ($2,700 list), 4 track model 34 ($1,700 list), 2 track model 32 ($1,300 list). All models accomodate 10.5 inch reels, provide full sync recording and include long life heads.

Airplay in Tasmania. 7CAE-FM is the only public broadcasting station in Tasmania, Australia, and is actively soliciting new music albums for airplay. If requested, Ben Lee, the station's New Music Co-ordinator, will write back with any feedback on your work from listeners. Send records or related new music information to Ben Lee, Hobart FM Inc., Box 1415P G.P.O., Hobart, Tasmania 7001, Australia.

New from MTI. MTI (105 Fifth Avenue, Garden City Park, NY 11040), probably best known for their Crumar keyboards, has introduced several new products including:



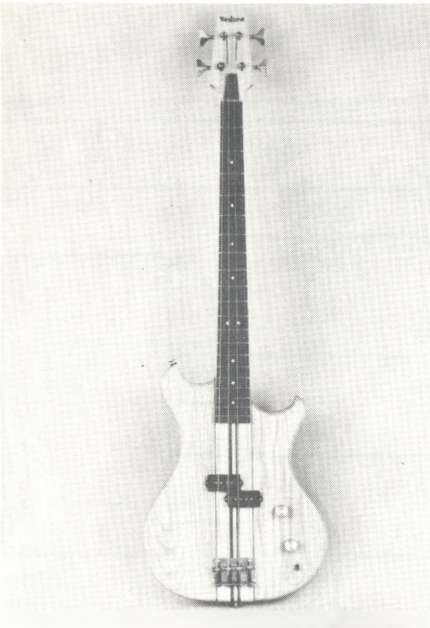
- The T-3 double manual keyboard (organ, polyphonic strings, and piano sounds with multiple outputs). Features include rotating speaker effects, reverberation, automatic accompaniment, and other options. List price is $2350.



- The Auto/Orchestra, designed specifically for solo and duet acts. If you know the chord progression to a song, the A/O backs up your music with drums, bass, piano or organ, and string synthesizer. Over 100 different background orchestrations are available via accessory plug in modules. Two of the modules, Rock and Country Funk, feature drum patterns and solos programmed by Billy Cobham. MTI claims that the quality of each individual sound is so realistic - from bass runs and background strings to cymbal crashes - that the normal mechanical feel typically associated with programmed devices is gone.

MORE........................

CURRENT EVENTS



• The Westone series of electric guitars, designed for maximum cost/performance ratio.

Circuits sourcebook. The "Electronotes Builder's Guide and Preferred Circuits Collection" is a compendium of electronic music circuits. It is available to the public for $15 from Electronotes, 1 Pheasant Lane, Ithaca, NY 14850-6399. Recommended!

Monolithic Filters. Reticon (345 Potrero Avenue, Sunnyvale, CA 94086) has introduced several 8 pin IC filters using switched-capacitor techniques (rather than the standard resistor/capacitor combination) for tuning. This results in filters whose frequency parameters are set by an external clock; however, like analog delay lines, these filters require anti-aliasing filters at the output with low clock frequencies. Chips available include the R5609, a seven pole lowpass filter; the R5610, a programmable array of four state-variable type filters; the R5611, a 30 dB/octave rolloff highpass filter; the R5612 notch filter; the R5613 linear phase lowpass filter; and the R5630, a modem filter that comprises two bandpass filters. Prices are quoted as $7 in large quantities.

Subsonic garbage collector. ACE Audio (532-5th St., East Northport, NY 11731) has announced the 4000b subsonic filter ($98.50 list). It removes the effects of record warps, rumble, and similar subsonic signals by filtering below 20 Hz at 18 dB/octave. IM distortion is 0.002%.

XFRMR. Sescom, Inc. (1111 Las Vegas Blvd. North, Las Vegas, NV 89101-1197) has introduced the TR-125 in-line audio transformer for matching low Z balanced microphones to audio cassette tape recorders or video cassette recorders. List price is $27.75.

MINIATURIZED SOUND EQUIPMENT is being introduced for sale early this summer by Yamaha Combo Products. These compact, portable and battery powereed components are the first in the Producer Series and includes the CS01 Micro Monophonic Synthesizer at $249.95; Mic/Line Stereo Portable Mixer (MM10) at $109.95; open-air Stereo Headphones (MH10), $29.95 and the Ma10 Mic/Line Stereo headphone amplifier at $124.95.

PRACTICAL CIRCUITRY

By: THOMAS HENRY

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 package1.

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

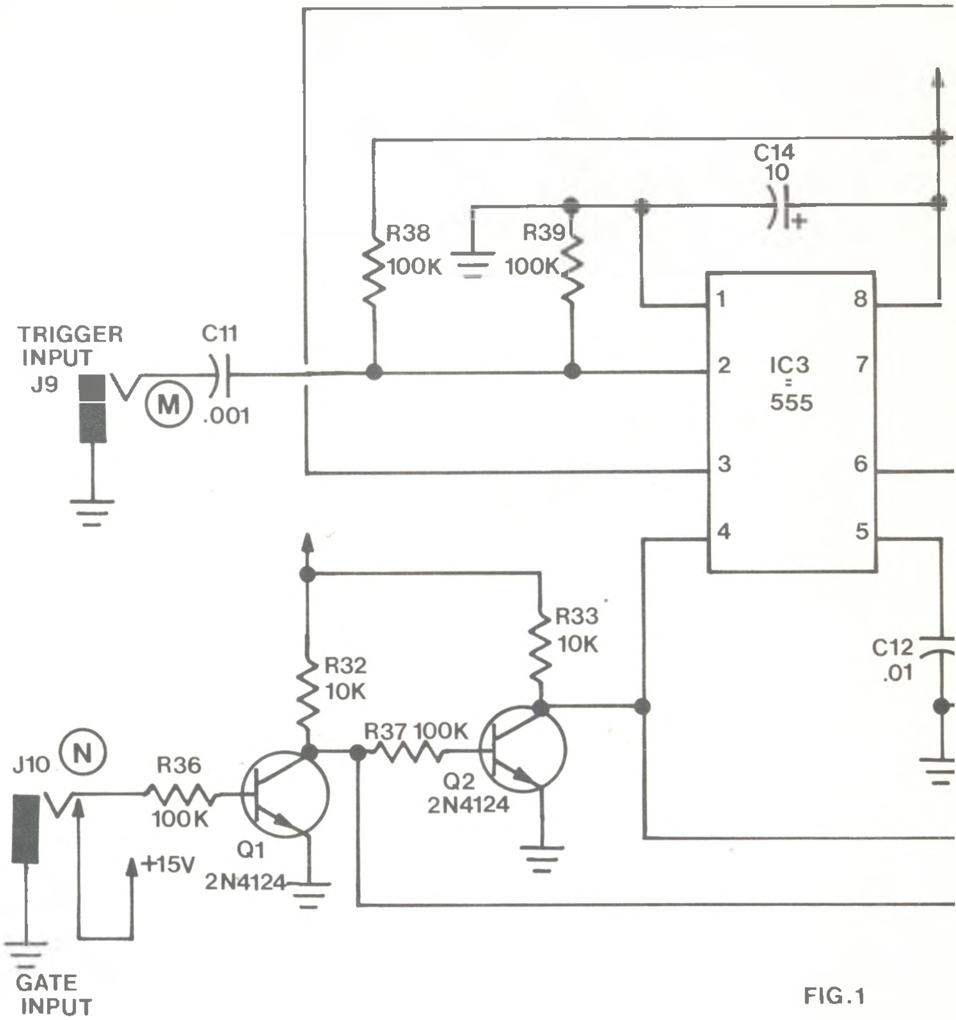
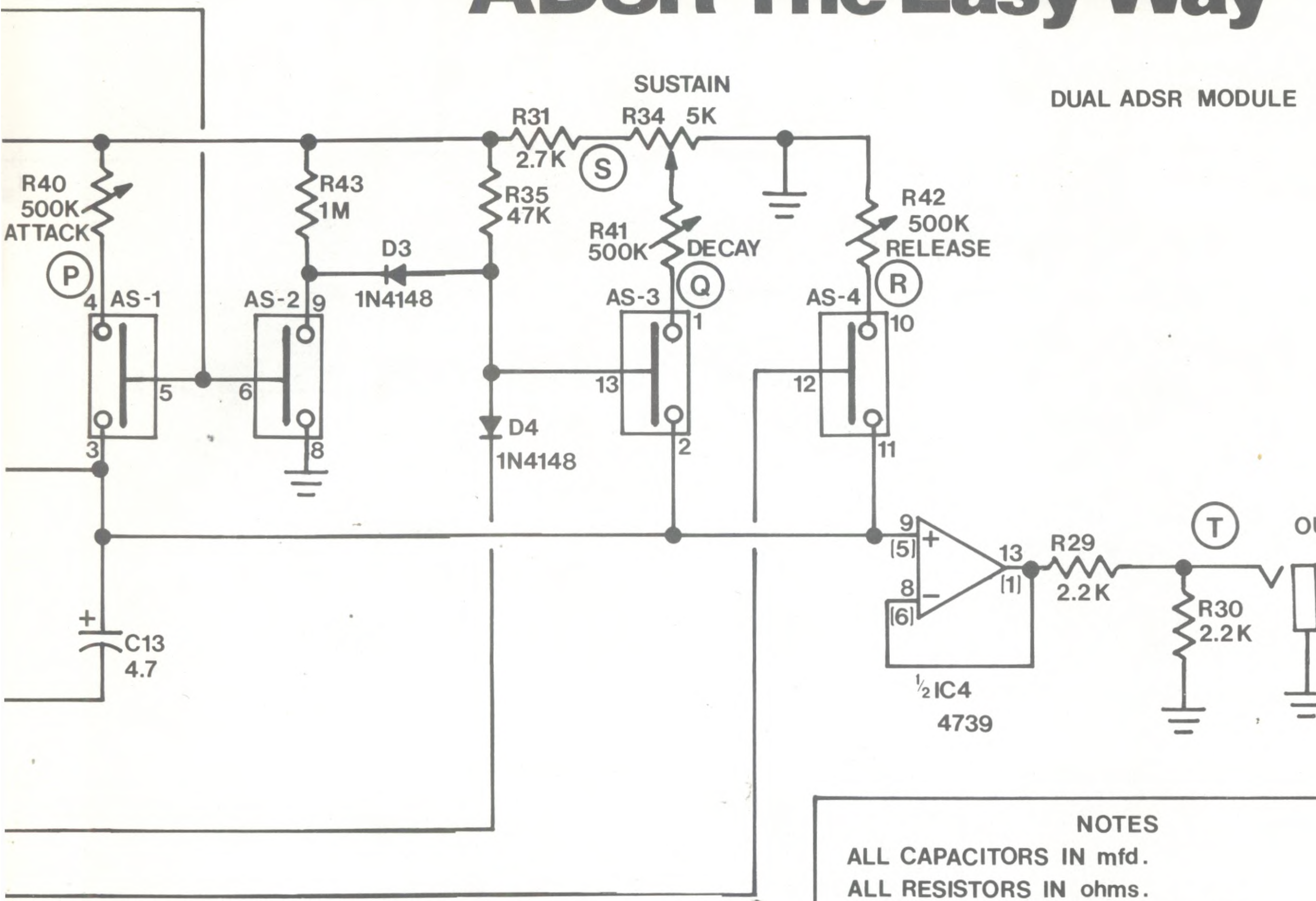


FIG.1

ADSR The Easy Way



DUAL ADSR MODULE

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

Continued on page 30

Getting The Most Out Of A Cheapo

by: Adrian Legg

The neck is the first and most important thing to take care of, as it will help or hinder your development.

It should have a slight forward curve, commonly called relief, and this can be measured using a straight edge or feeler gauges. Relief will show up as a gap between the straight edge laid along the center of the fingerboard and the fifth to eighth fret tops, of between five and fifteen thousandths of an inch, decreasing gradually either side of this point towards the nut and the higher frets. On longer neck guitars, the maximum gap may occur a little higher up the fingerboard. The theory, and practical result, is that this curve allows room for the curves formed by the vibrating string. If your neck were dead straight, you would not be able to get a comfortable action without bad rattles in the lower positions. If you think about it, it follows that a harder player needs more relief than a light picker, and it also follows that there should be slightly more relief on the bass side of the fingerboard than the treble. The latter is a bit too fussy for mass production methods, and if it doesn't happen naturally because of the harder string pull on the bass, then don't worry about it very much - a couple thousandths of an inch here and there isn't going to be critical, so long as you've got a good overall relief figure.

Adjusting the truss rod provides the necessary relief. If you have too little, the truss rod must be slackened off, that is, the nut screwed up. Your neck is not such a delicate creature that it will wilt if you say "bugger" in front of it, but on the other hand, it will react very badly to your treating truss rod adjustment as a trial of strength. Until you understand your neck, carry out any necessary adjustment an eighth of a turn or less, with your normal gauge of strings tuned up to concert pitch, with the correct truss rod key, and measure and observe carefully at each little turn. Remember that the neck may take a while to respond to increased or decreased tension, and so if you need as much as half a turn on the nut, spread the operation over a week. Necks vary considerably in the way and speed with which they react, so take your time, and careful observation and measurement will keep you out of trouble.

Relief can also vary slightly according to weather conditions, and regular observation will teach you where to set to allow for this.

If the neck has too little relief, and slackening the truss rod seems to have no effect, leave the truss rod nut loosened for up to a month (with maybe a blob of

Continued on page 32

(Editor's note: There are many inexpensive and second-hand guitars now available, which makes them likely candidates for experimenters - after all, no one's going to get their guitar fix-it chops together by ripping apart vintage Stratocasters. For those do-it-yourselfers who want to get a bit more out of these cheapos, the following article has been reprinted with permission from the British publication "Musicians Only", which ceased publication in late 1980.)

ANALOG DELAY CLOCK/MODULATION

BY: JACQUES BOILEAU

Here is a modulation circuit designed for the flanger in DEVICE 1:9, however, it is also excellent for other flanger and chorus devices. With the DEVICE flanger, this circuit replaces the standard clock and connects to points C and D of figure 2 in DEVICE 1:9 (page 3; figure 5 on page 6 of the same issue should be completely deleted).

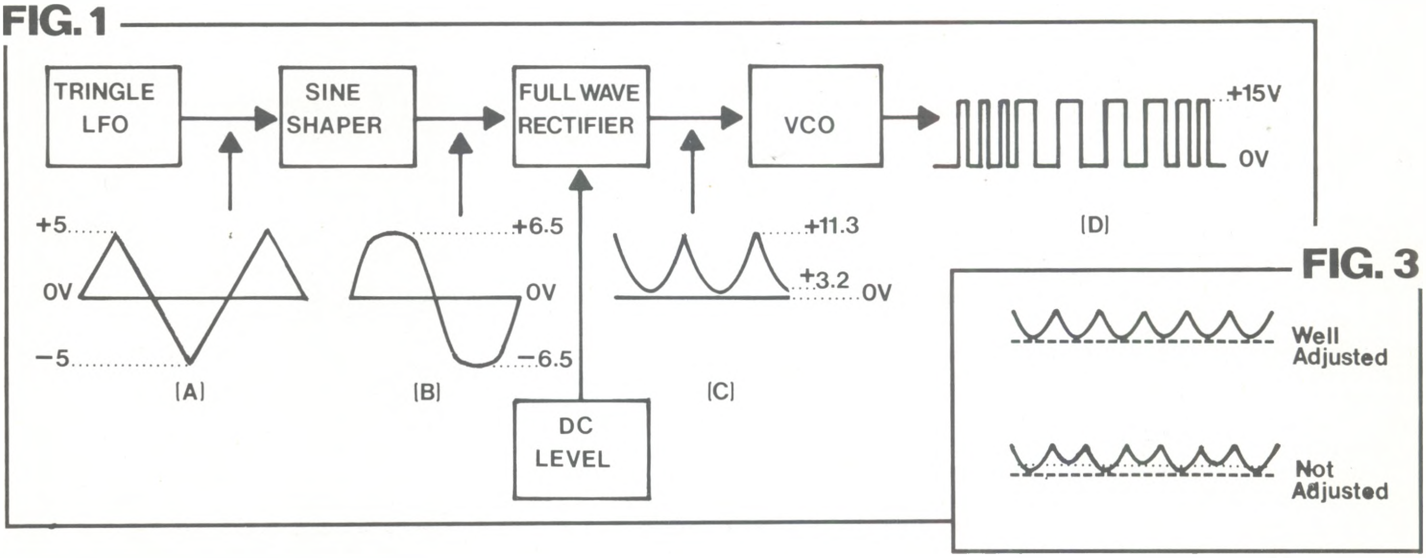
The premise of this clock is that music is a logarithmic, not linear, function. When you sweep a delay in a linear fashion, you have problems such as pitch detuning and a sweep that covers the low end of the flanging range too fast and the high end of the flanging range too slowly. This is the way most commercial delays work.

Rather than create a circuit with log converters, I instead analyzed what the log of a sine wave looks like. It is very close to the shape of an inverted full wave rectified sine, and that's what this circuit produces.

that follows, and may need to be trimmed for critical applications. IC5 is a phase locked loop, but only the VCO is used in this circuit. With R21 (modulation depth) set to minimum, the VCO is driven by +3.5V DC. R24 sets the shortest delay time and R24 + R25 set the longest delay time; again these values can be changed to suit your taste, but trying to get longer delays would lower the clock into the audible range. IC6 is a flip-flop which gives us the necessary non-overlapping, complementary clocks for the SAD-1024. It also divides the clock frequency by two, so care was taken to ensure that the clock frequency reaching IC6 is twice the frequency needed by the SAD-1024. (Editor's note: If you need to drive delays which require more current from the clock drive circuitry, you may parallel both halves of the 4013. If you need to generate a higher frequency clock signal, and avoid dividing by two, you can drive all four gates of a 4041 in

FIG. 1

FIG. 3



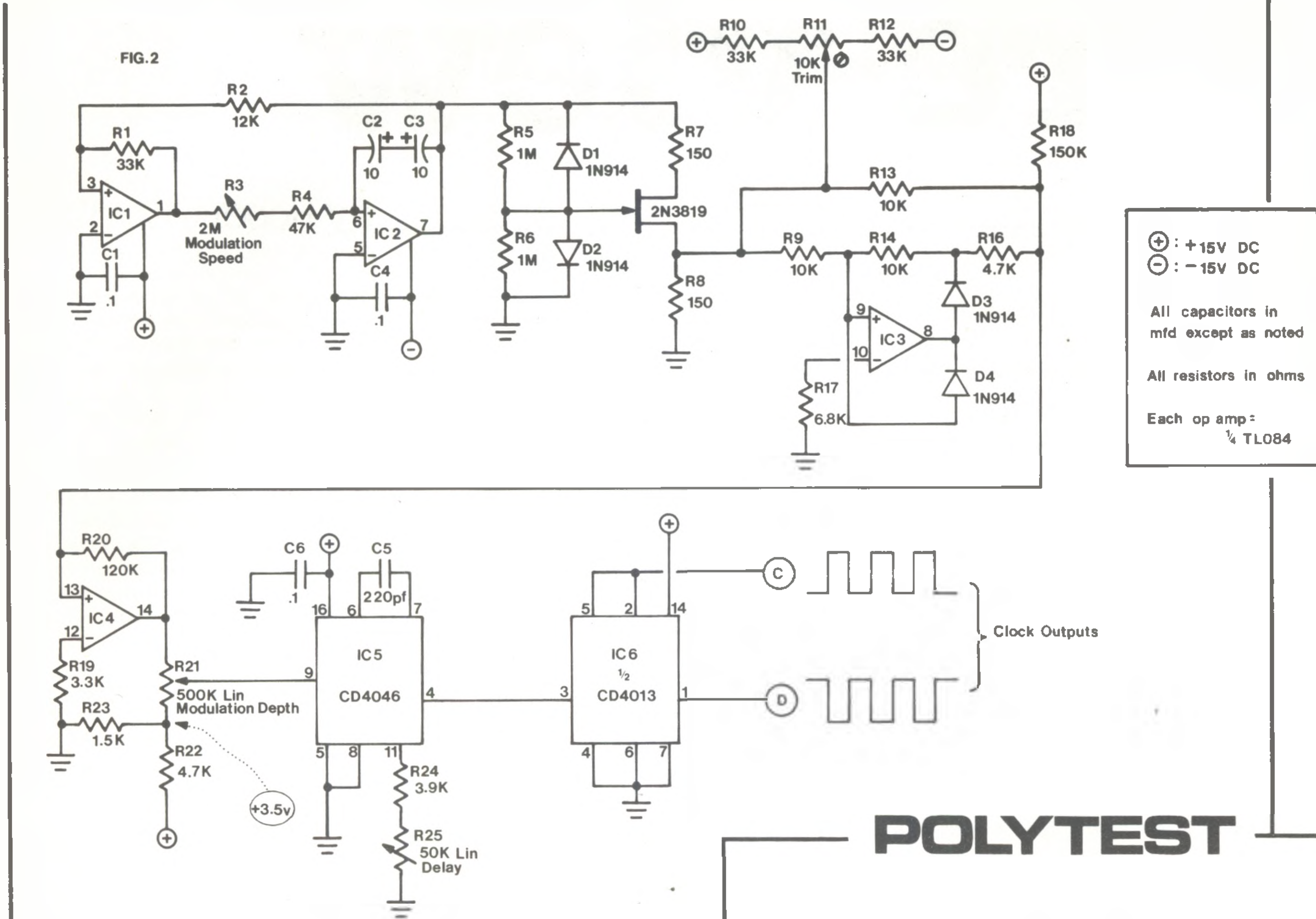
How it works. Referring to the block diagram in figure 1 and the schematic in figure 2, IC1 and IC2 form a triangle LFO; R4 sets the highest frequency and R3 + R4 set the lowest frequency. These values may be adjusted to suit different tastes. R1 and R2 set the triangle wave level for best results with the FET sine shaper. This sine wave is full wave rectified and inverted by IC3 and IC4. IC4 also adds some gain (hence the 120k rather than 10k resistor for R20), both to recover what was lost in the sine shaper and to properly drive the next stage. You may increase R20's value slightly for a wider sweep, but remember that we are already close to the positive supply limit. R18 sets up a DC offset to level shift the rectified sine wave for a voltage range compatible with the CMOS circuitry

parallel to produce non-overlapping clock signals at the same frequency as the 4046.)

Although you could use a standard bipolar op amp for the triangle wave LFO, at slow settings the input bias current becomes critical. Thus, a FET input op amp allows slower LFO sweep rates.

Calibration. The only adjustment needed is to balance the bottom limit of the full wave rectified sine (see figure 3). The easiest way to do this is with an oscilloscope: adjust the modulation speed for maximum, and preferably, set the scope so you only see one period on the screen (connect the scope probe to pin 14 of IC4). You will see the bottom of the waveform go up and down; adjust R11 for minimum shift. Note that only the bottom part of the wave-

FIG. 2



form is of interest, so enlarge it as much as possible - even if you lose the upper portion of the waveshape.

To calibrate with a DC Voltmeter instead of a scope, set R3 for the slowest speed and connect the Voltmeter to pin 14 of IC4. You will see the needle move slowly up and down the scale; adjust R11 so that each time the needle comes down to the same DC level. You can also adjust by ear by adjusting R11 so that each cycle sweeps the same delay range.

Final comments. Using the clock is simple. Setting the modulation depth control (R21) at minimum gives only straight delay. This can thicken your sound, or with regeneration, give pseudo-reverb. Advancing the modulation depth control lets you hear that familiar flanging sound, but may require re-adjustment of R21 for optimum results.

Incidentally, note that the voltages in figures 1 and 2 were taken from the breadboarded version and are guides, not absolutes. Your circuit voltages will probably differ a bit, but should be in the same general range.

Acknowledgements: I'd like to acknowledge Electronotes for giving me the ideas for the LFO and sine wave shaper, Walter Jung's Op Amp Cookbook for the full wave rectifier, and Popular Electronics for turning me on to using the 4046 as a VCO. #

POLYTEST

The full wave rectified sine is a far better choice for modulating delay lines than the standard triangle wave or sine wave LFO. The sound is much closer to "real" tape flanging, and an excellent side effect is that the flanger spends more time at the lower end of the flanging range, where the most interesting effects occur. For vibrato, this circuit also seems more musically useful.

For chorusing, where you are only adding a little bit of modulation anyway, there isn't as much difference between this circuit and standard triangle wave modulation sources. But for flanging, the full wave rectified sine is an excellent control voltage source. In fact, I was so impressed with the idea that I have devised a way to produce a similar waveform by feeding the triangle wave output of a CEM3340 back to the control voltage input. I call this a "Hypertriangular" clock, and it will be featured in a flanger currently being submitted for publication in Modern Recording magazine. - Craig #

re-view

by Robert Carlberg

(Editor's note: Whenever there aren't enough review records to fill the space allocated for "Re-View", I've asked Robert to write some articles that relate to reviewing, records, independent recording, and so on. In this issue, Robert explores "The Enjoyment Factor in Music".)

Ever notice how some records you like right away, play a lot, then suddenly get tired of them? Other records you don't think much of, but you find yourself playing them quite often when nothing else seems appropriate. Both records probably have given you the same overall enjoyment - one hard and quick, the other slowly and in smaller doses.

What distinguishes one from the other? Well, for one thing, the type of music can have a lot to do with it. Most popular music - rock and roll in particular - is a music of instant gratification, tailored specifically for commercial radio formats. Much of it is designed to catch your attention long enough to sell you some jeans or pimple cream, and maybe a record or two. The requirements of Top-40 fare are not too different from those of commercials - grab the attention, make a quick sale, then get the Hell out of there. Pillage and plunder, in other words.

Not all rock and roll, of course, is this simplistic; some of it comes with a message - "Deadman's Curve" or "Don't Eat Yellow Snow" - and some of it has been raised to a positive art form. The arty stuff usually doesn't provide instant gratification, though, since it takes time to unravel the artist's intentions. Consequently, you don't hear much of that on the radio.

Other styles of music, notably jazz and classical, reside almost exclusively in the slow-and-steady enjoyment. Adults are always telling children, "when you get older you'll learn to appreciate this". To an extent, that's true. As we grow out of our egocentric youth we learn to plan ahead, to prepare for the future. Who wants to be stuck with a bunch of singles you can't listen to? Instead you go for the smaller victories, the records you could live without but enjoy having. The "instant gratification" stuff seems shallow and childish, or else enticing but ultimately unfulfilling.

These sorts of divisions, mostly pretty grey and foggy, occur in almost all styles of music. Some very popular classical music - such as Ravel's "Bolero" - rely more on initial reaction and hummability than lasting artistic value. Some rock and roll (if that's the opposite end of the spectrum) takes years to appreciate and remains vital indefinitely.

So what else defines the difference between instant and long-term gratification? Another factor is the degree of musicianship. Not too many musicians are satisfied doing simple, catchy tunes which very often don't require much skill. You will see many of them move into more difficult playing, more complicated styles, more challenging material. The audience that follows them are moving too into delayed gratification, where perhaps greater attention over a longer time span is needed to appreciate the artist's output. The listener is also challenging him or herself, following the same goal of a deeper understanding of music. The artists who refuse to move backwards in time, from the Schoenbergs who renounced their "Gurrelieder" to the Beatles who refused to get back together, are only asserting the notion that they must continue moving or die, artistically. Few artists can face doing the same material, year after year. Pity the Beach Boys, trapped in 1965. Is our desire for nostalgia so strong that we demand the sacrifice of the artist?

Which brings up a third factor. During our impressionable youths everything we experience is magic and brightly colored. There are no half-measures, no partial successes, no partial failures. This is why the music you listened to as a child was so much better than what your children listen to today. This is why Tommy Dorsey was real music, while Simon & Garfunkel are only pleasant diversions, and why Led Zeppelin was the real thing and Molly Hatchet is not. There's no real difference, of course - it's only when in life you heard it, and how well developed your powers of discrimination were.

As a record reviewer, this change in attitude has to be taken into account, as much as possible. I must be suspicious of catchy, foot-tapping music which I hum for two days straight. On the third day I might memorize it and never need to hear it again. I must also be suspicious of unremarkable music that seems to have no strong character - it can sneak up on me later.

As a listener, I try to "program" different types and styles of music into my life. One day I'll listen to all-Spanish music from the local college radio station. Another day I'll play classical records and go to a symphony performance. When I'm preparing a review column for Polyphony I'll listen to a week solid of electronic music. After that I usually don't want to hear any electronic music for a couple of weeks.

It's all part of the enjoyment factor of music. Every music has a certain amount of joy to give you, and you have to be prepared to receive it. If you close the door on it, nobody loses but you. #

PRACTICAL CIRCUITRY

Continued from page 23

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

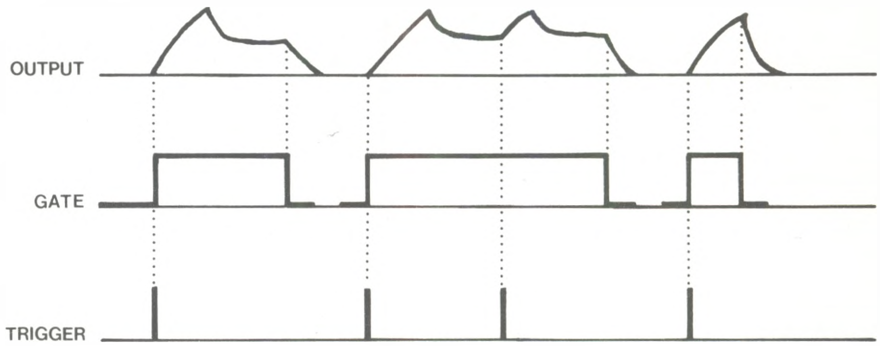


FIG.2 TIMING DIAGRAM / DUAL ADSR

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.

Letters

Continued from page 5

the November/December issue. I would like to clarify the authorship of this article. The instructions and technical information were derived from the Pro-One Operation Manual written by Stanley Jungleib. The beginning of the article was part of a cover letter sent to Craig Anderton by myself in response to his request for information on the Pro-One. This letter and Stanley Jungleib's instructions from the Pro-One Manual were combined by Craig Anderton to create the article published under my name. I cannot, in good conscience, take any but a small portion of the credit for the scholarship of this article. My real contribution has been to bring Stanley Jungleib and Craig Anderton together for the benefit of Polyphony's readership. I would ask that this letter be reprinted in the next issue in order to clear any misunderstanding between Stanley, Craig, the readers of Polyphony, and myself.

Sincerely,
Greg Armbruster
Advertising Coordinator
Sequential Circuits, Inc.

#

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Getting The Most Out Of A Cheapo

Continued from page 25

non-epoxy glue like Elmer's glue to keep it from rattling), and watch what happens. If the neck still won't settle forward, consult a reputable repairman or the manufacturer - a little heat treatment and clamping may be needed just to remind it of what it's supposed to do. This obviously doesn't apply to push/pull truss rods such as those fitted to Shergold and similar guitars, where undoing the truss rod will force the neck forward. Do bear in mind that very many new guitars will have a low relief at the start of their life. This is quite normal, and the neck will usually settle forward quite happily over the first month or two. On some, there can be a very slight backward lean which will gradually cure itself in the same way once the truss rod is slackened and the guitar used at concert pitch for a while. A badly bent neck may well be incurable if it has been like that for a long while.

A brief look through a current crop of pretty average guitars at a local shop showed one with no relief, two with low relief, and a couple more with high relief that were settling forward nicely and about ready for a bit more tension. A little patience, common sense, and the correct truss rod key, and they could all have been set perfectly to suit anybody.

Once you've got relief sorted out, you can worry about getting the action nice, and in the lower price bracket, aiming for 1.5 to 1.7 mm treble and 1.9 to 2 mm bass before fret dressing is reasonable. Remember that the more cambered the fingerboard is from the bass side to the treble side, the harder it will be to get a low action that will bend nicely. Personally, I prefer a flat fingerboard on electric guitars. Where the guitar has plenty of meat on the fret bead, light dressing down the center of the frets can help solve this problem, but that needs more advice than there is room for here.

Rough and scratchy frets can be smoothed off nicely with buffing soap. A good tool shop or hardware store should know what you're after.

Rub it thoroughly lengthwise over the frets, leaving plenty on, and then rub hard with a piece of leather evenly over the frets, and clean off what's left. Where the frets are really scruffy, I tape up the fingerboard and go over with fine garnet paper first, but this is rarely needed these days.

While the strings are off, and where the fingerboard is not lacquered, rub some refined linseed oil into the wood. This stuff is obtainable for about a dollar per small bottle from art supply shops, and will help prevent your fingerboard from drying out and splitting. Used regularly, it will also save tearing when the guitar comes up for its refret. Don't get it on your strings, it will deaden them immediately.

Most of the cheaper electrics I've seen could have done with a slightly lower action at the nut. The slots should be cut deeper with an appropriately sized needle file - three different ones should give you enough variation, and careful inspection of the file edge will show you the sort of cut it will make. Cut slowly, till the clearance of the first sting over the first fret (open) is the same as the clearance over the second fret when the string is stopped at the first fret. I repeat, cut slowly, and check these clearances every few strokes. If you fail to do this carefully, you will inevitably cut too deep and have to replace the nut. Some of the cheaper guitars have very soft nuts, and ultimately, replacing them with bone or brass will lift the general response. If you find that first string response is particularly lifeless, then a soft nut is probably to blame.

Once you've got it feeling nice, you can sort out the electrics. You can go for hotter pickups and brass hardware if you're ready to spend the cash, and in my suck-it-and-see experience a brass nut and bridge will do a lot for your sound. But before you get to that stage, there is a simpler mod that I think needs doing.

Most of the cheaper guitars I've seen suffer from pretty bad treble loss as the volume is turned down. The savior comes in the form of a 0.001 uF capacitor - I am an ardent fan of this little blob, and I suspect many Tele owners would be if they realized what it was doing for them. If you take the electrics cover plate off the guitar, and look at the volume pot(s), you'll see three lugs. One will connect to ground; ignore it. The other two will have various wires attached. Take the 0.001 uF capacitor, and without removing any of the other wires, solder one connection to each of these two lugs. This will allow treble frequencies to bypass the volume pot altogether, although they will cut off when the volume is turned all the way down. If you notice any tone change now as volume is reduced, it will be sharper rather than more muffled. The change will be audibly less on a humbucker than a single coil pickup. A side effect will be that your tone control will have more treble to cut at lower volumes, and will consequently seem more effective.

While on the tone control, if you find that it is cutting too much volume as you turn it down, check the tone control capacitor. If it is 0.047 uF, swap it for a 0.022 uF cap. Some people claim you can't hear the difference - I think in many circumstances you can, and personally, I compromise on 0.033 uF on my regular electric. It is also quite logical to use 0.047 on the neck pickup and 0.022 on the bridge where you have separate tone controls.

Once you've done a few basic jobs like this, you'll probably find the customizing bug starting to bite. If you customize carefully, you can have a lot of fun, learn a lot about your sound, and move towards a sound of your own. On a budget guitar there's not a lot you can do to worsen the already minimal second-hand value. The experimentation carried out now will help later decisions about more expensive guitars. You may well find that you can save yourself a lot of money in wrong purchases.

For interest's sake, and before you start work on the guitar, I'd recommend that you read Hideo Kamimoto's book, Complete Guitar Repair (Oak Publications). It covers the groundwork very thoroughly and though the wiring diagrams are unadventurous, they do cover many basic models. #

TOUCH SWITCHES

Continued from page 17

at a time, we have created a simple form of data bus not too dissimilar from the ones found in microcomputers. The 8 bit data bus feeds the computer's input port, while the output port controls which group of 8 touch status lines gets enabled on to the bus. Eight 74LS74373s could handle a maximum of 64 touch amp outputs. (Take care to only let one bit go "lo" at the output port at any one time, otherwise more than one group of status data bits will be enabled. This situation is called "bus contention" and usually results in total confusion.)

Of course, once your custom touch controller is attached to a computer/synthesizer combination, then the usual galaxy of options becomes available. This includes things like playing in and recalling note sequences, punching up preset voice settings, enabling special effects, and so on. Through the magic of the touch switch, they're yours to control with the tap of a finger. #

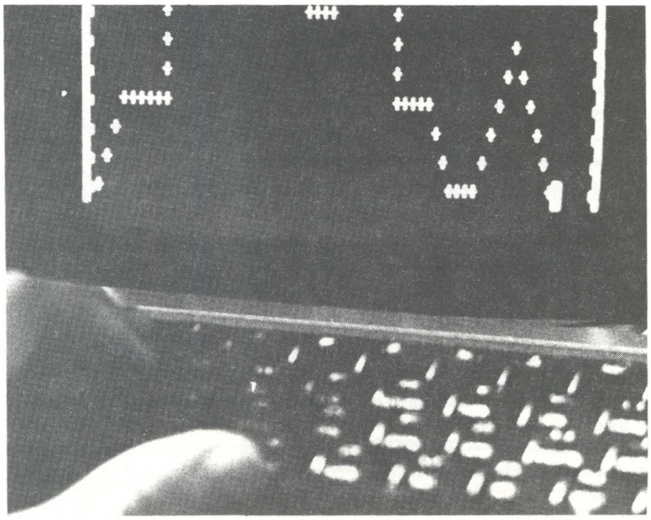
SCREEN WAVE

Continued from page 11

to 61, not 1 to 62). In L3010 this value is placed into variable "NP" (Number of Points) for use in your waveform DAC routines. Each FOR-NEXT program loop must end by exceeding its maximum number of cycles, otherwise an error will result. Since the program jumped out of the loop in L3000 before it was finished, L3020 fixes the potential error source by making the loop counter "I" exceed the loop's ending value. L2150 also used this technique to end the "V" loop. (Don't worry if you don't understand this "fix", if you always let your FOR-NEXT loops finish themselves, there will be no problem.)

Finally, everything is ready, the waveform has been created, the values have been determined and stored. It's time to output the wave! L3999 to L4010 are a sample DAC output routine. Actually, only one line does the work, L4010. L4005 just prints a message on the screen so that you know that the previous routines are finished (they take a few seconds) and the wave is now being produced. The DAC output routine is a very simple example, it just pulls a height value (voltage level) out of "W" and sends it (OUT 0, for example, means send a value to output port 0 = port A of the 8255 chip) to the DAC, gets the next value, OUTputs it, etc., for "NP" (Number of Points on the wave) values, then GOTO the start of L4010 and do it again, endlessly. As you can see, this output routine is not very complex. That's why it's called a dummy routine, it just serves as an example until you replace it with your own custom routine, tailored to your needs.

This is just a start. You will likely want to build on this program, add titles, cassette save and load waveform data routines, store more than one waveform, write machine language waveform output routines for high frequency output, trigger devices, use precision resistor values or a greater number of DAC bits, and so forth...I hope you do! #



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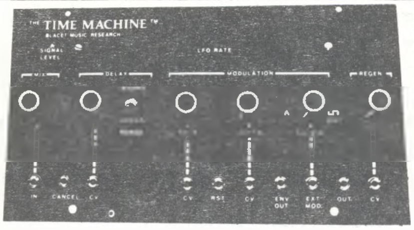

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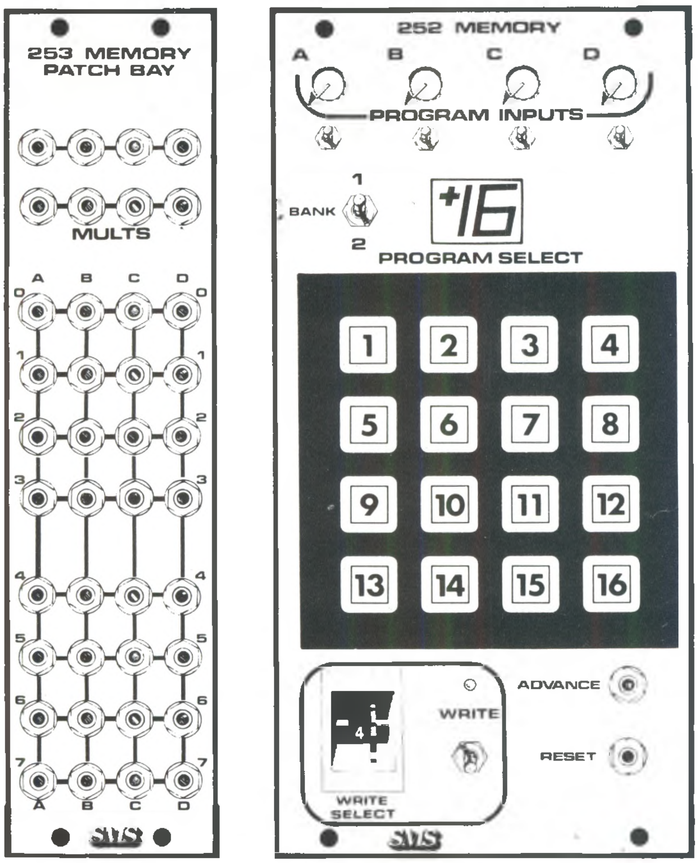
Ad Index

252 Memory System

Give Your Analog Synthesizer New Life With Precise Digital Control



½ ACTUAL SIZE

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

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

There are 32 independent control voltage outputs and 32 presets of these 32 voltages (expandable to 64 presets). These outputs are user programmable and accurately repeatable for pitch grade applications.

The user may write his programs in groups of 4 voltages at a time or singly address any one parameter. Memory is retained without power being applied by a self-contained user-replacable battery. Advance and reset jacks are provided to enable counting through the resets with an external clock source.

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

SPECIFICATIONS

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

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

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

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Veloci-Touch™ Controller

WHAT DOES IT DO

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

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

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

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

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

HOW DOES IT WORK

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

IS IT POLYPHONIC

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

HOW DO I INSTALL IT

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

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

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

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

CAN I USE IT WITH MY (your axe here)

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

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

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

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

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