DEVICE The Publication for Electronic Musicians

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the return of Steven St. Croix

A DEVICE INTERVIEW

(In issue 1:3, we ran the first part of an interview with Steven St. Croix, vice president in charge of research and development for Marshall Electronics. Marshall Electronics should not be confused with the amp manufacturer: the company is primarily known for manufacturing the Marshall Time Modulator, a sophisticated delay line for studio applications. In part 1, we covered Steve's background and talked a bit about delay lines in general. In part 2, he gets a lot more specific. Some of the technical concepts in this part are pretty involved, but for anyone conversant with delay lines, the ideas are most interesting. The interview was conducted by Pete Johnson, our East Coast correspondent.)

Pete: How do you format the delay lines in the Time Modulator?

Steven: There are 8 delay lines total, that are grouped together into 2 main delay lines; each main delay comprises 4 delay sub-lines. A signal feeds four parallel delay lines, and those 4 lines are divided into two families. The two lines of each family are clocked out of phase; they see the same audio input, and the outputs are positively summed. What that does is basically double the sample frequency, in fact, your Nyquist frequency (explained later) can almost reach the clock frequency. In most charge transfer systems you get two copies of the sample to fill up the "gaps" caused by clocking, which means the data is not as valid as you would like it to be. With the Time Modulator clocking scheme, every piece of data is valid; any gaps are filled with the real signal, not a copy of something that happened previously. This way we can clock at lower frequencies, but you have the advantages of clocking at higher frequencies...you need fewer transfers, thus reducing distortion, noise, and other problems.

Now remember that we still have a second family of delay lines. This family is clocked the same way as the first family (parallel multiplex mode), but the audio signal going through one of the families is inverted. Essentially, you're driving the two families differentially; and when you add them together, you cancel out even harmonic distortion. So, you end up with a long, clean delay without having to go through a great number of transfer stages. With this system you can clock at 100 KHz and keep a bandwidth of 90 KHz - that's the something

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DOD Analog Delay

BY THOMAS HENRY

Reviews of new products can sometimes be meaningless unless comparisons are made to standard, well-known devices. Therefore, in this review we will first look at the new DOD 680 Analog Delay, and then we'll compare a few of its performance characteristics to the Electro-Harmonix Deluxe Memory Man to observe any similarities or differences. I picked the MM for comparison because I happen to own one (note that this product is slightly different from the Memory Man Deluxe with Chorus that Craig Anderton reviewed in DEVICE 1:8). Bear in mind, however, that the E-H unit lists for $279, whereas the DOD lists for $189 - making it one of the least expensive analog delay units available on the market. The question that immediately comes to mind is what has been sacrificed in terms of performance in order to achieve this low price, and whether the potential user would find those sacrifices to be objectionable or not. Hopefully this review will allow the musician interested in buying an analog delay to make a better-informed decision.

GENERAL: The DOD 680 is a standard analog delay device, useful for generating echoes and spatial effects. Let's look at the controls and see what they do. The first control sets the delay time. This gives continuous variation of the delay time, and the manufacturer claims the limits of this to be from 20 to 330 milliseconds. Next we encounter the repeat control. This control, (often called a regeneration or feedback control) determines how much of the output signal is returned to the input for reprocessing, hence determining the number of repeats. In the far counterclockwise position this is good for one repeat, and in the far clockwise position one can cause the signal to go into multiple repeats to the point of "runaway" echo.

Now comes an interesting feature, especially for live performance use. Instead of the usual mix or blend control (for mixing th straight signal with the delayed), DOD has provided two mix controls. The mix control entitled "remote mix" remains fixed (at whatever setting you decide upon) regardless of the position of the in/out foot switch. The output of this mix circuitry is fed to a separate "remote out" jack. Remember, the output presented to this jack remains fixed (that is, the proportion of dry to delayed signal is determined solely by the setting of the remote mix control).

The other mix control (the "local mix") operates in

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in this issue

notes from napa

DEVICE IS BACK ON THE AIR!

Well I warned you last issue that there might be a delay and, sorry to say, there sure was. But have no fear, ardent supporters, we are back and we are commited to bringing this publication up to date.

Now don't go expecting miracles (like three issues in three weeks) but rest assured that we will be working steadily to get DEVICE on schedule in as short a period of time as possible. I, as publisher (and the wandering member of the partnership), wish to extend my apologies for the tardiness and confusion over the last couple of months. I hope that we won't have to talk about this much more in the future.

As you may have already determined from this column's heading we have established our latest location in California's beautiful Napa Valley (most noted for its wonderful wines). This puts Craig and I about fourty minutes away from each other and makes putting DEVICE together very convenient. The both of us are quite pleased with this new arrangement.

So, for all the moving around, boxing and unboxing, pulling of hair, and exhaustion, we find DEVICE in good shape. The staff is healthy and happy, people are sending in articles, and (due in large part to our intrepid editor) DEVICE is gaining respect as an innovative and sane voice in the musical instrument biz.

Good changes are in the wind and we have a lot in store for you this year, so,
STAY TUNED!

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If you have experienced any problems with getting your issues that have not been cleared up to your satisfaction, please, take the time and drop us a card with the nature of your problem and the last issue you received. You should be up to DEVICE 1:9:79 prior to this issue. There were some mailing errors due to a number of last minute hassles associated with moving and to

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STEVEN ST. CROIX (continued from page 1)

which the other units just cannot do at present.

P: What's the Nyquist frequency?

S: The Nyquist theorem involves optimum sample rates. If you wish to pass a certain audio signal, you have to take enough samples of that signal to give a valid reconstruction of that signal. Any errors show up as distortion; for example, if you have a 10 KHz signal and you're only sampling it at a 5 KHz rate, there's no way that the sampled signal will resemble the original 10 KHz signal. Remember the way that traditional bucket brigade devices work: There are long chains of switches and capacitors, actually, the capacitors are the capacitances occurring at the junctions of a series of transistors. An input signal gets clocked into the first capacitor, but it's just there for a fraction of a second - so the capacitor remembers this charge for a tiny fraction of a second. On the next clock transition, this charge moves down the chain to the next capacitor, and the first capacitor is charged to whatever the input signal happens to be at that moment. This shows one of the advantages of analog delay, namely, there is no quantizing error; a 1.05V signal appears as 1.05V, it is not rounded off to something like 1.0V or 1.1V as can happen in digital units. However, unlike digital systems the analog types are not regenerative; they don't self-correct the value of the charge as it gets passed down from one delay stage to the next. Unfortunately there are side effects and byproducts that degrade the sound...

P: Like the high frequency clock slipping in between the audio signals?

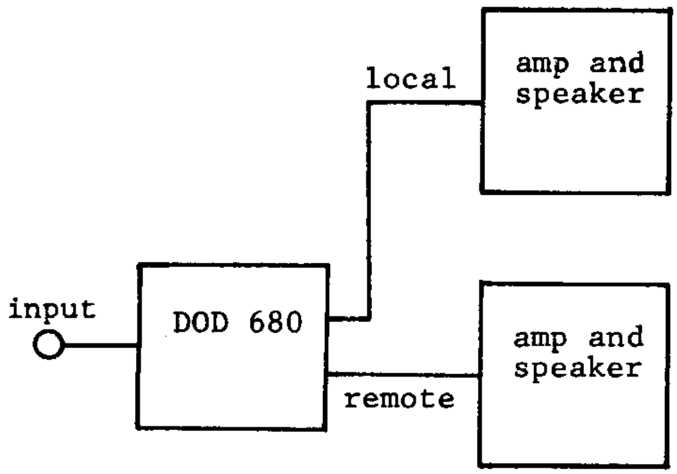
S: Yes, and more. Every time that little switch changes to move the charge along, it makes a little switching noise and when you go through several thousand of these switches, the noise starts to accumulate. In addition the effective capacitance is about 0.7 pF, living in a rather crowded environment to keep the chip somewhat smaller than the island of Manhattan, so the capacitor can only remember its voltage for a couple of microseconds at most. With all this going on, it's amazing an analog delay system works at all. Each time you go through a transfer stage you get some switching noise, and a little of the voltage leaks off...and you also get electron noise that exists whenever you're in an active transistor state. You can try to cut down the noise, but the damn capacitor is still going to leak; this is called transfer inefficiency. Remember that bucket brigade lines are just like the bucket brigades that firemen use to put out a fire, and like a bucket brigade, some of the water (charge) gets lost along the way. Now, that doesn't change the accuracy of the signal, because all of the amplitudes are lessened by an equal amount...so you just add some gain at the end of a line, and you end up with the same amplitude spread that you had originally. Unfortunately, you're bringing up the noise as well as the signal when you add that gain. The cumulative effect is so bad that you just can't get real long delays out of bucket brigade devices.

The CCD (charge coupled devices) we use work on a different principle entirely. Simply speaking, we start off with all our buckets full and drain off whatever charge is necessary to pass a signal along. Due to this process, you do not have cumulative loss since none of the stages can "spill" charge; they begin with all the charge they need, and just bleed off what they don't need. Additionally, a great deal of research has gone into making quieter switches that operate at higher voltage levels. CCDs today can characteristically handle 2 to 3 times the voltage swing that buckets do, so right there you've got more headroom. It costs more be-

DOD ANALOG DELAY *(continued from page 1)*

the same way as the remote mix as far as blending straight sound with delayed sound is concerned, but is dominated by the foot switch. The effect from this channel can then be cut in and out at will. It also has its own output jack, "local out".

A typical setup for this local and remote scheme might be:



One would have the remote set up for a subtle mix of delay and straight signal, whereas the local could be set for huge amounts of delay and very little straight signal. Then, for the verse and chorus of the song one would be playing with the local switched out. This would give a normal, slight echo effect. But then, for that special solo, the musician would engage the footswitch, thus bringing in a bizarre out-of-sync echo effect. Note that with the two amplifiers (and since the local and remote mixes are unequal), one would get a very strange spatial illusion. This of course would be intensified if the amplifiers were in different parts of the room, far apart from one another.

The last control on the device is the "PA/normal" switch. In the normal position, the input is matched to guitars and other low level output instruments. In the PA mode, the input is modified to accept high level signals, making it compatible with effects loops on PA gear.

SPECIFICATIONS: The following specs were provided in the owner's manual:

Input impedance: 500K
Output impedance: 1K
Delay time: 20 to 330 ms
Gain: Unity
Input level: 1.5V peak to peak, normal mode
6V peak to peak, PA mode

MECHANICAL CONSIDERATIONS: The DOD 680 is very well built, and excellent shop practices were employed. The case is very heavy molded aluminum, and when the back is screwed on, the whole thing is as sturdy as a brick. The pots are all of the sealed variety, which is excellent for keeping out dust. The printed circuit board is fiberglass, and is very well laid out.

A definite plus for the 680 is the fact that it incorporates a three prong grounded AC plug, so that the chassis is at ground. It's always nice to be extra safe with electronic gear, and this provides that extra margin of safety (and hum rejection).

My one complaint from a mechanical point of view is that the input level switch (PA/normal) is a very delicate mini-toggle. Normally these are good switches, but since this box was designed to be on the floor, and since the toggle is not recessed into the case, a clumsy misplaced foot could flatten it very easily. An internal switch, or else a very heavy duty external switch (a recessed slide switch, old-fashioned toggle, etc.) would be much more appropriate.

ELECTRONIC PARTS: The integrated circuits used in the 680 are a Reticon R5101 2000 sample audio delay line chip (that's a $30 chip in lots of 100), a 4013 CMOS dual D flip-flop, and two 4136 quad op amps. These are all good semiconductors and I'm pleased to note the use of the 4136s in particular. These, of course, are the musician's favorite op amps, being very low in noise and also well matched internally. The Reticon R5101 is an improvement over the standard SAD-1024, in that longer delays are possible with a reasonably low clock frequency.

THE COMPARISON. Now that we know something about the 680, let's do a little comparing. If you look carefully at the specs provided you will note that two crucial ones (as far as delay lines are concerned) are missing. These are the bandwidth and noise measurements. We decided to conduct these measurements ourselves, and compare them with the E-H Memory Man Deluxe (see note 1). Here is what we found. With both units set for maximum delay, minimum repeat, and full delay (no straight mixed in):

DOD 680, no input: -78.0 dB S/N ratio
E-H MM, no input: below -90 S/N ratio

The "noise" represents the noise and clock residual signals inherent in the device. Now, there are many ways of interpreting noise figures, but we might notice here that the difference in S/N with no input is approximately 12 dB. If that doesn't sound like much, consider that every 6 dB represents a doubling of power. Lest you think that the figures are saying something that the ears don't hear, I should tell you that when I first tried the unit out (before taking the measurements) and A-B-ed it with the Memory Man Deluxe, the difference in noise was extremely apparent. I then tried the A-B test with some outsiders (non-electronically oriented friends), and they too immediately heard the difference. The conclusion is evident: the E-H MM is audibly quieter.

We also decided to check out the bandwidth of the two devices, and this is what we found. Once again the test conditions were maximum delay, minimum repeat, and full delay mix, with a 0 dB input (see note 2):

FREQUENCY	MM OUTPUT	680 OUTPUT
100 Hz	-21.3 dB	-11.8 dB
200 Hz	-21.6	-11.2
300 Hz	-21.6	-11.7
400 Hz	-21.1	-12.2
500 Hz	-19.9	-12.6
600 Hz	-20.4	-13.2
700 Hz	-20.2	-14.0
800 Hz	-20.2	-14.6
900 Hz	-19.8	-14.9
1 KHz	-19.5	-15.7
2 KHz	-16.9	-33.9
3 KHz	-22.8	-61.2
4 KHz	-42.6	
5 KHz	-62.9	

These results indicate that the DOD 680 has a usable bandwidth of about 1 KHz. The MM, on the other hand, is usable up to 3 KHz (see graph). Now, let's interpret the figures in the light of the technology involved.

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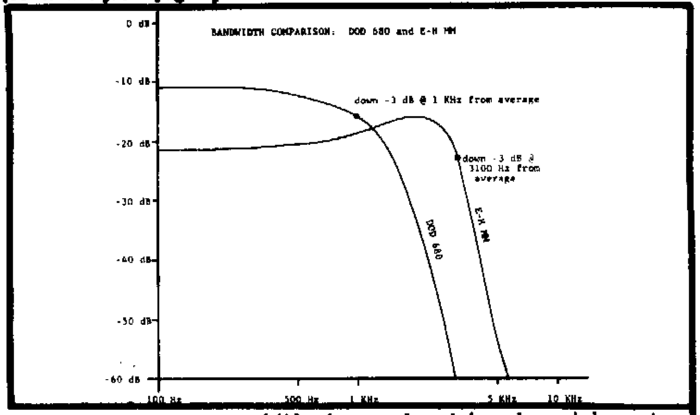

First, it is a fact of life that analog delays have inherent bandwidth problems. This is due to lowpass filtering necessary to keep the clock glitches out of the audio output. There are several philosophies about what concessions and tradeoffs to make. In the case of the DOD 680, the designer has elected to use a long delay line (2000 stages). This means that he can get by with a higher frequency clock, but this at the same time means more noise, for the more stages that the signal passes through, the more dirt and noise is accrued. In addition, the DOD 680 designer has chosen to remove all of the clock by heavy lowpass filtering. But the MM in contrast has opted for a larger passband at the expense of clock leakthrough at long delay times. So, how you react to an analog delay line depends on what you want. Personally, I prefer to hear a little clock as a tradeoff for increased bandwidth (see note 3). In passing, we should note that the intended use has much to do with the suitability of any particular device; recording applications often require less noise than live performance uses, and so on.

CONCLUSION: The DOD 680 is a mechanically well-built delay, but I would personally prefer to pay the extra bucks for the E-H MM in order to gain lower noise and increased bandwidth. On the other hand, those on a budget or those with less critical applications will find the DOD 680 a logical alternative to higher priced units. I'd like to add that my preference is not based on the theoretical measurements given above, but rather on the evidence that my ears detected. And in the end, isn't that the best way to pick any instrument, effects device, or amp?

Also, those musicians who do not like the "sound" of the compansion circuitry added to most analog delay lines will note that the DOD 680 does not use compansion. This trades off noise performance for a more "natural" sound. Whether or not this makes any difference to you is again a matter of personal taste.

ACKNOWLEDGEMENT: I'd like to thank Ron "Arnie" Dick for his great help in setting up and conducting the experiments. It's nice to work with someone in the know.

NOTES:
(1) Test equipment: Hewlett-Packard 400 FL AC Voltmeter, Heathkit sine/square audio generator (Model IG-5218), Tektronix 453 dual scope.
(2) These conditions represent the worst case, in that none of the original signal is getting through, and the clock is at its lowest frequency. Probably you would never play the gizmo this way.
(3) The clock is only audible in the MM at the longest delay settings. However, one seldom uses such a setting unless it is for an "effect". In that case, remember that the MM has an internal "noise gating" action due to the compansion.

hold that flanger!

Unfortunately, several errors crept into last month's flanger article. Some of these were omissions that occurred when transferring the design from circuit board to the printed page; others occurred because I had breadboarded the flanger with a defective NE570, which created some other problems. Luckily, these changes aren't very extensive, so even if you've built last issue's circuit and had a hard time getting it to work (or were disappointed with the performance), you should be able to add these "fixes" in just a matter of minutes. I suggest that you make the following corrections to your schematic now to avoid any future confusion.

1. Add a 10M resistor from pin 1 of IC1a to +15V. This reduces the compression action at low volume levels.

2. Add a 470 Ohm resistor between the 2.5k offset trimpot and the emitter of the 2N3906.

3. Disconnect point Y (the recirculation input) from point B, and reconnect it to the hot terminal of the balance control (i.e. between the balance control and the 2 uF capacitor that goes to pins 10 and 11 of IC1b).

4. Due to the above change, S1a must also be changed. The terminal marked (+) should now be marked (-), and the (-) terminal should now be marked (+). Change the wiring accordingly.

5. Add a 2 uF capacitor between pin 7 of IC1a and the 5.6k resistor that eventually couples into IC2. The positive end of this cap should face pin 7 of IC1a.

6. If you're using the voltage controlled recirculation gain block, change the 2.2k resistor connected from pin 6 of IC9 to ground to 10k. If you're using the manual recirculation control shown on page 4, eliminate the 10k resistor and if you like, replace the 100k trimpot with a 10k trimpot for easier settability.

7. The preamp gain control can be 100k instead of 1M for easier settability.

Note that with these changes, the recirculation signal is now being re-expanded along with the regular signal.

I apologize for any inconvenience these errors may have caused you; this is the most complex project I've presented in do-it-yourself form, and frankly, I was so pleased with the performance and sound of the flanger that I rushed the thing into print a bit prematurely. If you've had problems getting the flanger to sound right, try these modifications. When all the parts are in the right places, this thing sounds just great!

By the way, I have also published a flanger project in Guitar Player magazine that is considerably simpler, and so you might wonder about tradeoffs between the two units. Well, the DEVICE flanger is definitely quieter due to the noise reduction - in fact, several people who have heard the DEVICE flanger feel that it's quieter than just about anything else on the market. However, the circuit is more complex, and requires more extensive calibration, as a result. The Guitar Player flanger is no slouch on performance, though; it gives a real good sound that is still amazingly quiet, and of course only requires three chips. So, both designs are equally valid. What it comes down to is whether you're willing to spend a fair amount of extra effort to reduce the noise level as much as possible.

— Craig

Current Events

If you have a new product, information on new components or new suppliers, interesting tidbits, press releases, or new sources of information, tell us. We love a good scoop. Address it to **CURRENT EVENTS,** *c/o DEVICE, 12304 Scribe Drive, Austin, Tx 78759. Or if it's particularly "hot and juicy" call us at (512) 836-3069.*

NEW DEVICES FROM ELECTRO-HARMONIX. Latest from Electro-Harmonix (27 West 23rd ST., New York, NY 10010) is the Bass Microsynthesizer and Space Drum.

The Bass Microsynthesizer is similar to the version for guitar covered in issue 1:6 of **DEVICE**, however it is optimized for use with bass. The Space Drum sweeps an internal oscillator up or down (there are separate controls for the start and stop frequencies), with two separate switch selected decay curves, when the dynamically responsive sensor pad is triggered by a drum stick strike. A trigger output jack (called the Coordinator jack) allows external sound sources to be triggered by rhythms played on the sensor pad.

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NEW LED VU METER CHIP. Joining the LM3914 (10 step linear meter) and LM3915 (10 step log meter) is the LM3916, an LED meter that covers a −20 VU to + 3 VU range. All three parts are manufacturered by National Semiconductor (2900 Semiconductor Dr., Santa Clara, CA 95051).

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GET OUT YOUR VIDEO RECORDERS. The MCA–Sony case has finally been resolved, and Federal Distric Judge Warren J. Ferguson says that it does not violate copyright laws to tape programs off the air. However, the decision does not specify what happens in cases of tape duplication or recording from pay-TV systems; These matters will probably be decided by Congress.

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MORE TI SOUND GENERATING CHIPS. Most of you are probabaly familiar with the 76477 sound generator chip; it's available from Godbout's, Radio Shack, and many other sources. One of the best aspects of this chip for experimenters is that it can stand alone, without the need for external control lines, elaborate interfacing, or outboard audio amps. However, TI also makes a series of chips that are specifically designed to be used under microprocessor control in applications such as games and signalling devices. The newest of these is the 76489AN, slated for introduction in the early part of 1980, which includes three programmable tone generators, four programmable attenuators, and a programable white noise generator. These are controlled by eight internal registers which interface to the microprocessor through 3 control lines and 8 data lines. Like other ICs in this series, the 76489AN uses integrated injection logic and is totally bipolar.

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GR-300 GUITAR SYNTHESIZER. Roland has introduced the GR-300, a polyphonic guitar synthesizer that includes a custom guitar controller and a separate, foot-controlled unit that contains the bulk of the electronics. The guitar includes various switches affecting the filter, oscillators, and LFO, as well as standard humbucking pickups for a "straight" guitar sound to go along with the hex pickup that interfaces with the electronics box. This box contains (among other things) the "hexa-VCO" section with master tuning, LFO modulation, two variable pitch controls (A and B pitch) which are instantly switchable between the two pitch settings, rise and fall time of A and B pitch, and a chorus effect. The VCF includes a sensitivity control with variable attack time, envelope modulation, and envelope inversion. The tracking of this unit is superb, and overall it seems like the GR-300 is a well thought out successor to its predecessor, the GR-500.

NATIONAL INTRODUCES DUAL NORTON AMP. Remember the LM3900? While easy to apply, the AC performance of this part wasn't too impressive. Now National has unveiled the LM359, another Norton (current-differencing) amp with very impressive specs: a gain-bandwidth product of 30 MHz and 30 V/microsecond slew rate at unity gain, along with a noise figure of only 6 nV/root Hz. Both amps are completely independent, and operate from a single supply in the range of +5 to +12V DC. For further information, contact National Semiconductor, 2900 Semiconductor Drive, Santa Clara, CA 95051.

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NEWS FROM SSM. There are three developments from SSM of interest to electronic musicians. First, the SSM2044 dedicated 4 pole lowpass filter chip and SSM2055 voltage controlled envelope generator chips are now available for $6.00 each in small quantities. Contact Eu systems, 417 Broadway, Santa Cruz, CA 95060 for additional information on pricing and availability. Second, the price of the SSM2040 4 pole filter and SSM2030 VCO has been reduced from $10 each to $7.50. Third, SSM now has a data book that includes applications as well as data on the entire family of electronic music chips. It is available for $5.00 (plus 50 cents shipping) from Solid State Music, 2076B Walsh Avenue, Santa Clara, CA 95050.

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INFORMATION FOR SALE. Modern Recording's 2nd annual buyer's guide is now out on the newsstands (cover price: $2.95). This is a real handy reference for finding out addresses of manufacturers, and also includes "short form" specs on all types of audio and music-related equipment. There are also selected reprints from past issues of Modern Recording that relate to mixing consoles, interfacing equipment, and special effects. For more information, contact Modern Recording at 14 Vanderventer, Port Washington, NY 11050.

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NEW FROM ARP. Buzz Kettles filed the following report on ARP's latest instrument, the ARP PIANO, which was unveiled at the Winter NAMM show in Anaheim:

"The ARP PIANO is a digital based polyphonic keyboard instrument. The most important aspect of this combination synthesizer/electronic piano is its 73 key maple - that's real wood, not plastic - keyboard. This appropriately weighted keyboard, along with its key-velocity sensing mechanism, provides the performer with the dynamics, sensitivity, and the elusive "feel" of the most expressive traditional keyboard instrument ever developed - the piano.

"The unit has 16 preset voices, two footpedals, and a handful of front panel controls for real time variations on the selected timbres (voices). These voices include grand piano, clavinet, honky tonk piano, vibes, and 12 others including the metallic ringing tone usually associated with the Fender Rhodes piano. The footpedals are similar in shape and feel to the type of pedals found on traditional pianos. They provide similar functions (soft and sustain) and as an added feature, the left pedal may be used for depth of vibrato. The front panel has controls for detuning one or both oscillators, and for manipulating the filter and vibrato sections. On the final output, the unit features a stereo, six stage (3 notch) phase shifter which can be used to add anything from a subtle ambience to thick harmonic sweeps."

The ARP PIANO should be appearing in stores before too long.

D.I.Y.* power amp

by CRAIG ANDERTON *Do It Yourself

Several months ago, Digital Research of Texas (P.O. Box 401247, Garland, TX 75040) offered the STK-056A, a 25 Watt hybrid power amp module manufactured by Sony. I got a couple of these at the time because it looked like they would make a good amp project for DEVICE; I built up a prototype stereo amp, and it sounded just great (in fact, I'm still using it in my studio). Unfortunately, however, these were a few of the last remaining modules. Since DRC didn't have enough of them left in stock to justify doing an article, I never did write the project up.

A little later the people at DRC wrote to say that they had a new module available with slightly less power, Sony's STK-054. The STK-054 is physically identical and electrically similar to the STK-056A, except that it seems like it's a *true* 20 Watt amp (although the STK-056A was rated at anominal 25 Watts, I measured it putting out +40 Watts under some conditions). Unless you're using very inefficient speakers, 20 Watts per channel should do just fine for most hi-fi and studio monitoring applications.

HOW TO BUILD IT

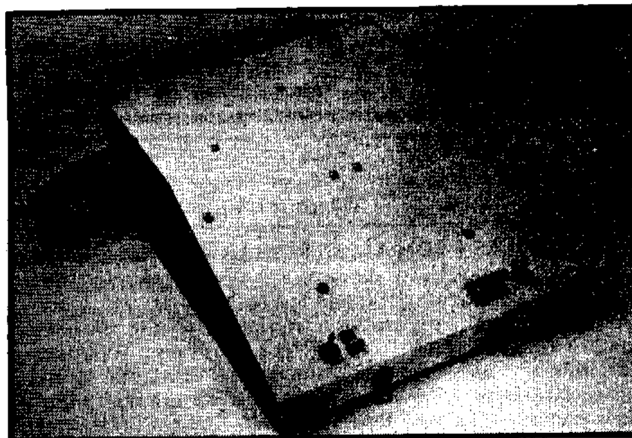
Unlike the documentation included with the STK-056A (which showed you how to build an amp that wouldn't work – I had to do quite a bit of circuit analysis before I got the thing tamed), the documentation for the STK-054 includes a 20 Watt power amp schematic less the power supply that does work. The schematic shown in figure 1 is essentially the same, but redrawn for greater clarity, and shows a power supply along with the rest of the circuitry required to implement one channel of the stereo pair. Those module pins which do not show up on the schematic should be ignored.

The photo shows two of the modules mounted in the top of one of LMB's classier looking boxes. Since the bottom of the module acts as a heat sink, it is important that these heat sink areas contact as much of the metal chassis as possible. A little heat sink compound added between the two surfaces wouldn't hurt either. Note that all the parts associated with each amp mount on a little terminal strip mounted on the front of the amp; this creates a mechanically solid unit, and keeps the leads between the various parts as short as possible.

When wiring up the ground and power lines, follow good wiring practice. This means the amp ground points should all terminate individually at the terminal strip ground point, and all the power supply ground points should also terminate at a common ground lug bolted to the chassis. Although some people would say it doesn't make any difference, I also ran separate V+ and V– leads to each amp from the power supply. Keep the input and output leads separate from each other, use shielded cable for the input leads to prevent hum pickup, dress all audio leads close to the chassis, and mount the transformer as far away as possible from the other parts. I elected to mount the speaker fuses inside the chassis (I never blow them out anyway), but the AC fuse is mounted on the rear panel. You might want to make all the fuses chassis mount types; the choice is up to you.

THE POWER SUPPLY

The optimum power supply for this amp would deliver absolutely no more than +27V DC (the published maximum



rating) with no signal applied to the amp, dropping down to around +23V DC during heavy current loads. This allows for maximum power from the amp. The 36V center tapped (@ 3A) transformer specified puts you close to this ideal; you can try a 40V type if you have one around and see if that gets you any closer, but you're then running the amp right up against the maximum ratings. If, like me, you don't have a 36V CT transformer, you can use the 25.2V CT (@2A) model sold by Radio Shack. While this gives you less output voltage (around $\pm$12V), and therefore less power, there's still enough juice to give any efficient speaker a good kick at moderate listening levels. Be careful with the power supply wiring; you're talking about lots of current and a fair amount of voltage.

If you wish to add an LED indicator to the positive supply, connect the LED cathode to ground and wire the anode to V+ through a 1.5k resistor. To connect an additional LED to the negative supply, wire the anode to ground and connect the cathode to V– through a 1.5k resistor. These LEDs not only serve as power indicators, but also help bleed off any stored charge from the large filter caps after the unit is turned off.

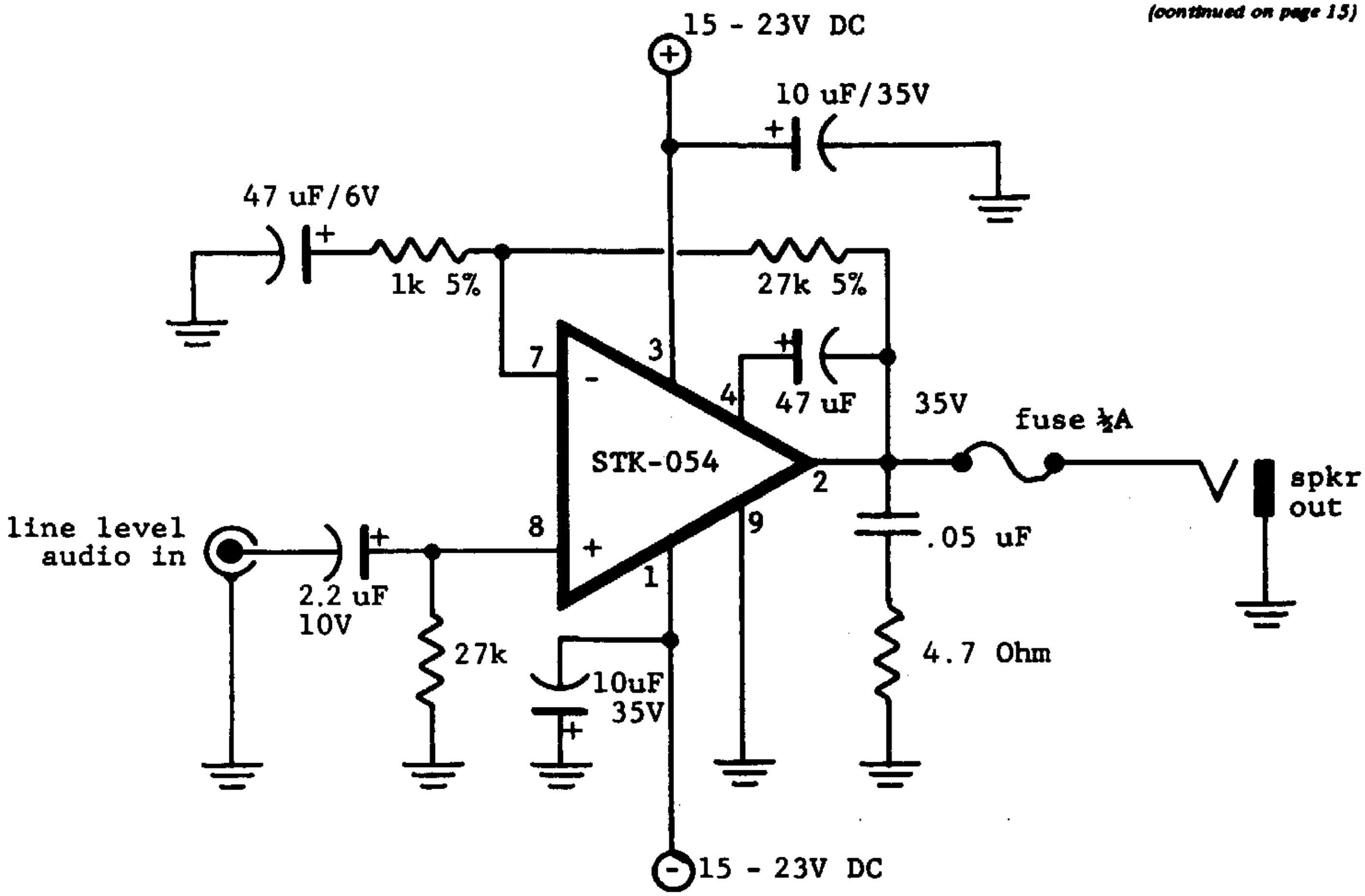
One additional modification I made to the basic schematic was to add a DPDT headphones/speaker switch to switch the output between phones or speakers. You can use 2 or more headphone jacks, but wire a 33 Ohm isolating resistor between each output lead and headphone jack terminal to add some isolation between the different sets of phones. You may wish to increase the value of the resistor a little more if you want less overall volume in the phones.

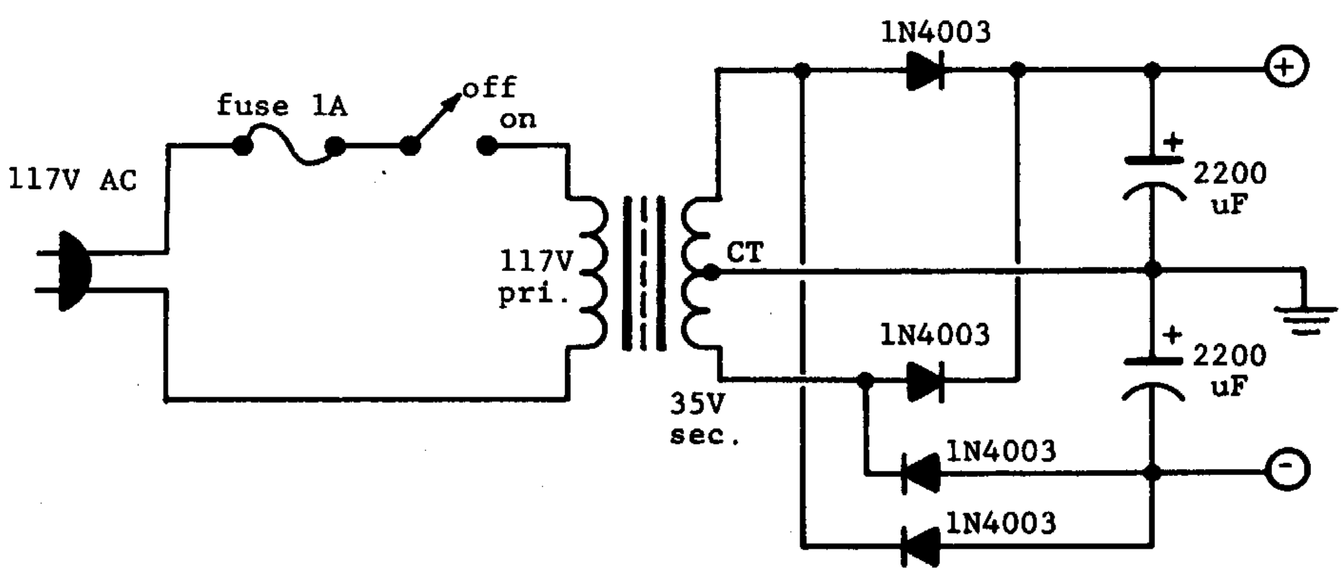
BUT HOW DO THE MODULES SOUND?

In two words, damn good. The amp doesn't need an output coupling capacitor, which not only eliminates that annoying "thump" that occurs during turn-on with some amps, but also controls the speaker in a much tighter way.

This gives a stronger and more satisfying bass sound. But my biggest surprise came in the treble regions; the response was sweet and smooth, with lots of definition. While I don't have the facilities to measure distortion, it seemed very low at normal listening levels. Frequency response was essentially flat from 40 Hz to 20 KHz and beyond. While the entire stereo amp cost me less than $50 to build, it is still a real nice amp that, even when compared with some of the "biggies", offers a very clean sound. If you're looking for a simple and inexpensive medium power amp, it's harder to get much simpler — or much more foolproof — than this design. I do suggest, however, that you act fast if you want these mod-

(continued on page 15)





BUILDING THE AMS-100 - part 11

craig anderton

Rhythm Generator

A NEW DIRECTION FOR THE AMS-100 SERIES. Over the past few months, we've covered control voltage generators/processors and audio modification modules; in fact, we've developed a pretty good set of audio modifiers. While our work in this area is by no means complete (I'd like to come up with a high performance VCA, some filters based on the Curtis filter chip, and some long delay lines), for the next several installments our attention is going to shift to new and different control voltage generators.

I suppose that most people would find control voltage generators to be far less interesting than something like a flanger. I know that I have certainly not been too interested in the subject in the past; however, that attitude is rapidly changing. A couple of months ago I was working on the LFO for the AMS-100...a nice, simple, triangle wave LFO. Then I decided that I might as well voltage control it, but found that a linear response LFO was far less useful than a log response type. So, I got out some SSM2020 exponential gain control blocks and used them to control the LFO. This represented a great improvement, but I thought it would be even better if the LFO could be synchronized to the player, so that initiating a new note would reset the LFO. While not a new idea, this was the first time that I had played with the concept in depth and I was very impressed. Eventually, I ended up packaging two LFOs and a sample-and-hold in the same module; the S/H had the ability to either sample in sync with one of the LFOs, or it could sample every time a new note occurred (indicated by a pulse output from the AMS-100 input module covered in DEVICE 1:1). To make a long story short, the number of intriguing sounds possible with this particular combination made me realize that the more sophisticated your control voltage generators, the more sophisticated your overall sound. Rather than try to pursue new types of filters or similar audio processing modules, I realized that the whole field of control voltage generators really needed to be checked into further.

I got hung up in the lab for about a week, and this resulted in not only the dual LFO/sample-and-hold module, but also in a random control voltage generator and a new type of module called a "pluck follower". All of these modules have been perfected and will run in the next couple issues of DEVICE.

Then a few weeks ago, I went into Radio Shack and saw that they now have a pattern generator chip for electronic drum sets. I thought that there must be better things to do with it than just drive drums, so I brought one home. A couple nights later I came up with the following circuit, and thought I'd put it in DEVICE while the idea was still fresh and publish the other circuits at a later date.

WHAT THE PATTERN GENERATOR DOES. As shown on the data sheet supplied by Radio Shack, the MM5871 requires two "oddball" power supplies (-14V and -27V), and only drives drums. In our module, it runs off a standard 15V bipolar supply, and does much more than drive drums.

The chip includes an on-chip tempo oscillator whose frequency is set with a single resistor and capacitor, and generates 5 separate pulse outputs which, in theory, are used to drive bass drum, snare drum, tom-tom, and similar drum sounds. However, as it turns out when this chip runs from a 15V bipolar supply, you get +15V, 1 ms pulse outputs that are compatible with the various AMS-100 envelope generators. Six additional terminals allow you to choose the pattern that appears at the 5 pulse outputs.

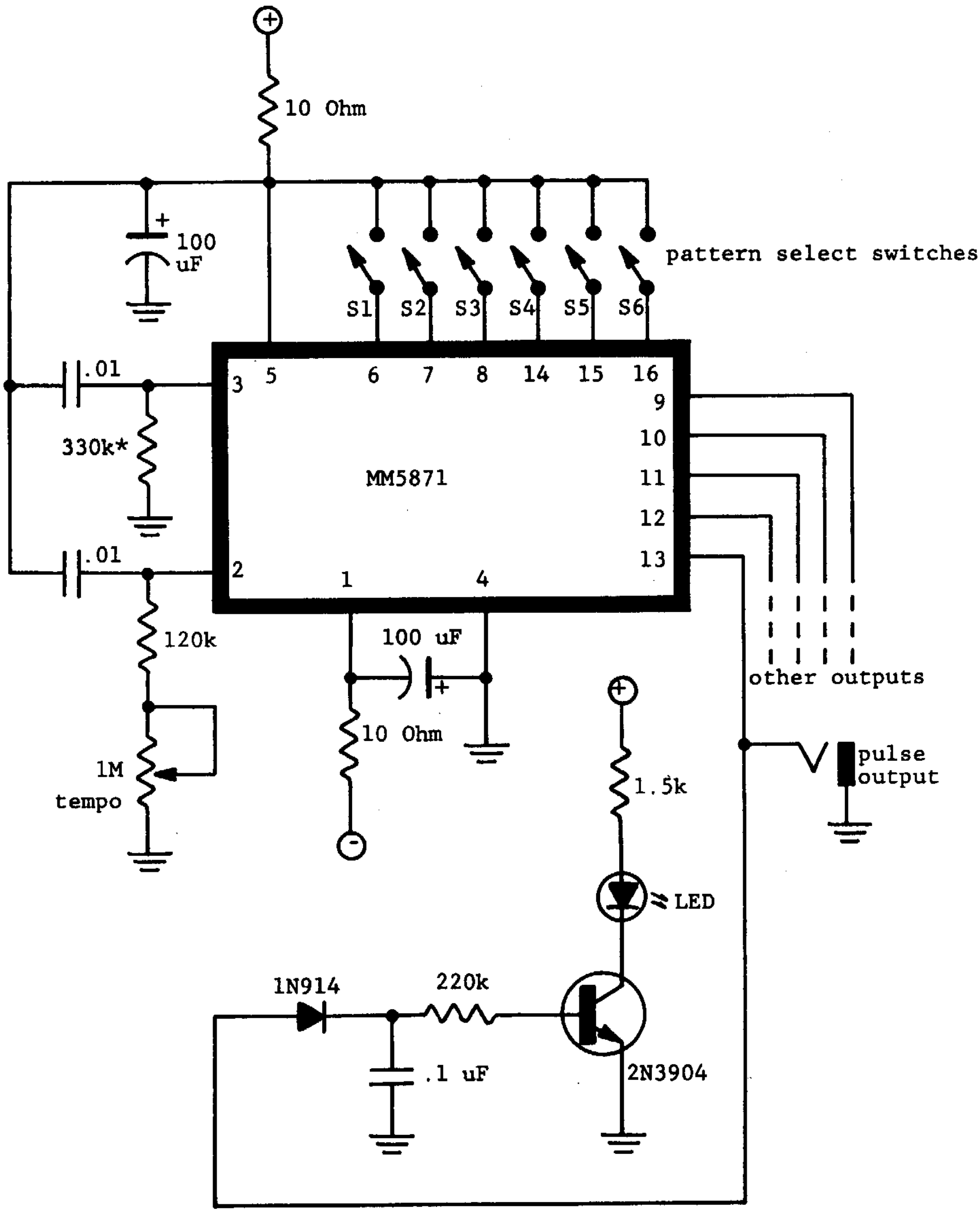
What good is this thing? Well, imagine imparting rhythmic qualities to a guitar, keyboard, or vocal track. You can feed your instrument through a number of paralleled modules; the pulse output that would normally trigger a bass drum in the drum set could trigger an envelope generator that lets through a brief burst of fuzz on the down beats, while the pulse that would normally go to something like a snare drum could sweep a filter in a rhythmic manner. Other pulse outputs could drive envelope generators connected to VCAs, flangers, and the like. The result is an instrument sound that acquires a uniquely rhythmic quality. In fact, if you add this kind of processing to something like a bass it becomes a percussive - as well as melodic - instrument. For several years now, I've been synchronizing instruments to each other using envelope generators, click tracks, and similar techniques; the results have been most satisfying from a musical standpoint. This module allows you to experiment with pattern generation with a minimum of fuss and effort, since all you need is one chip and a few other parts.

ABOUT THE CIRCUIT. Let's look at the pulse output coming from pin 13. First it goes to a pulse output jack, capable of triggering our AMS-100 envelope generators (or triggering drum circuits, if you're so inclined). The remaining circuitry connected to the pulse output is based around a 2N3904 pulse stretching/indicator circuit. Since each pulse is only 1 ms wide, if you use that pulse to directly drive an LED then that LED is on for such a short period of time that you can hardly see it blink. With the pulse stretcher, the pulse charge gets dumped on the .1 uF capacitor, which then discharges over a much longer period of time. As it discharges, it forces the 2N3904 to conduct, thereby illuminating the LED. While this pulse stretcher circuit isn't necessary, it is nice to have LED monitors and I'd advise that you include it. Driving an LED directly from the pulse output will clip the pulse at about +2V, which makes it no longer compatible with the AMS-100 envelope generators.

Although I haven't shown jacks and pulse stretching circuitry for the other outputs, that's only because I didn't want to stay up all night drawing identical pulse stretcher circuits. However, all the pulse outputs are dealt with in the same way.

The data supplied with the chip tells you what patterns are created by closing switches S1 - S6. Basically, three of these switches give 3/4 time patterns and the other three give 4/4 time patterns. However, you can also close more than one switch at a time to modify these basic patterns,

(continued on page 10)


10 Ohm
100
uF
pattern select switches
S1
S2
S3
S4
S5
S6
.01
3
5
6
7
8
14
15
16
9
10
11
12
13
330k*
MM5871
.01
2
1
4
120k
100 uF
other outputs
10 Ohm
1M
tempo
pulse
output
1.5k
LED
1N914
220k
.1 uF
2N3904

(continued from page 2)
cause you need 4 transistors per delay stage instead of 2, but it's worth it in terms of sound quality. Our CCD chip, combined with the way we use it in the dual differential mode, has turned out to be very good. I'm really pleased with it.
P: So you use your own IC design...
S: Yes, we've been working on our own CCD chip for years. We're using a third generation chip that's pretty simple, not too many pins...easy to live with and super quiet. We've got a 1 second delay now with 95 dB signal-to-noise ratio, and no quantizing noise or quantizing error (unlike digital delays).
P: Is that delay line being built into the Time Modulator?
S: No, the Time Modulator at present has a 400 ms option, so you're just under half a second.
P: Do people overdrive the Time Modulator for an effect?
S: Yes, and the sound is much better when you overdrive an analog delay than when you overdrive a digital delay. By the way, you'll notice that most digital delay lines cannot do flanging because they don't have the ability to get very short delays. Typical minimum delay time with most digital units is around 1 millisecond, which is right in the middle of the optimum range for flanging - but that's academic anyway, because the digital stuff won't sweep much more than a 2:1 range, and you need much more than that for flanging. It's a bitch to spend that kind of money on a delay line, and not be able to flange; most of the special effects nowadays are done with analog units.

There's another consideration for flanging, and that is the intensity of the flanging effect. As you probably know, flanging produces a comb filter effect with a series of notches (cancellations) cut into the signal. The depth of these notches contributes directly to the quality of the flanging sound. With tape deck flanging, you can get about 45 dB notches if the equipment is in good shape; most bucket brigade devices will give you about 25 dB. The reason why you can't get deep notches is that the delayed signal is not full bandwidth, therefore the high end of the frequency spectrum is not flanged simply because there is no high end signal present to be flanged against. Also, most commercial flangers don't go to much trouble to match the amplitudes of the delayed and dry signals, so that means an even shallower notch. Even if you do match the levels, distortion, phase shift, and other instabilities in the delayed signal still contribute errors. With the Time Modulator, we've gone to the trouble to design super-matched circuits so that the straight and delayed signals are matched within 0.1 dB; additionally, the increased bandwidth means you get full flanging up to 15 KHz. And because it's quiet - well, if it wasn't quiet it wouldn't matter how deep you cut the notches because the noise would fill the notches up. Finally, we do compansion which doubles the cancellation and gives a very deep flange. By the way, with most delay lines the amplitude changes when you change the clock frequency, which contributes another possible source of error. At high clock frequencies the switching stages of the delay line become slew limited, so you lose amplitude and the chip runs hot. We went out of our way to have amplitude stability over the full bandwidth of the device.

What makes our flanged sound really unique is the 72:1 sweep range; you can sweep all the way down to an echo in one continuous sweep. In addition, there are two delay lines in a Time Modulator so that one can flange against the other. This creates some incredibly complex harmonic structures.
P: Will the Time Modulator do harmonizing effects?
S: No, it's not designed to; and besides, the idea of using a hard harmony - a fifth, a third, or whatever - is musically invalid. Harmonies simply don't remain fixed throughout a piece of music. You can feed a sawtooth into the external control voltage input and get harmonizing if you really insist on it, but it sounds just as bad as most of the other harmonizing devices on the market.
P: How do you do double and triple tracking with the Time Modulator?
S: There is a right way and a wrong way to do double tracking. The wrong way is to delay te signal, mix it in with the original signal, and call it double tracking. Ha! We all know better than that. When someone goes into the studio and puts down a track, rewinds the tape and puts down an overdub, it's never perfect; it's always just a little bit out of tune and out of time. Those errors have to be introduced for the thing to sound right. So, we modulate the 2 delay lines independently; additionally, we don't use time-spliced detuning but Doppler detuning to eliminate glitches.
P: That should be enough for the interview; thank you very much for your time on behalf of the readers of DEVICE.

RHYTHM GENERATOR (continued from page 9)

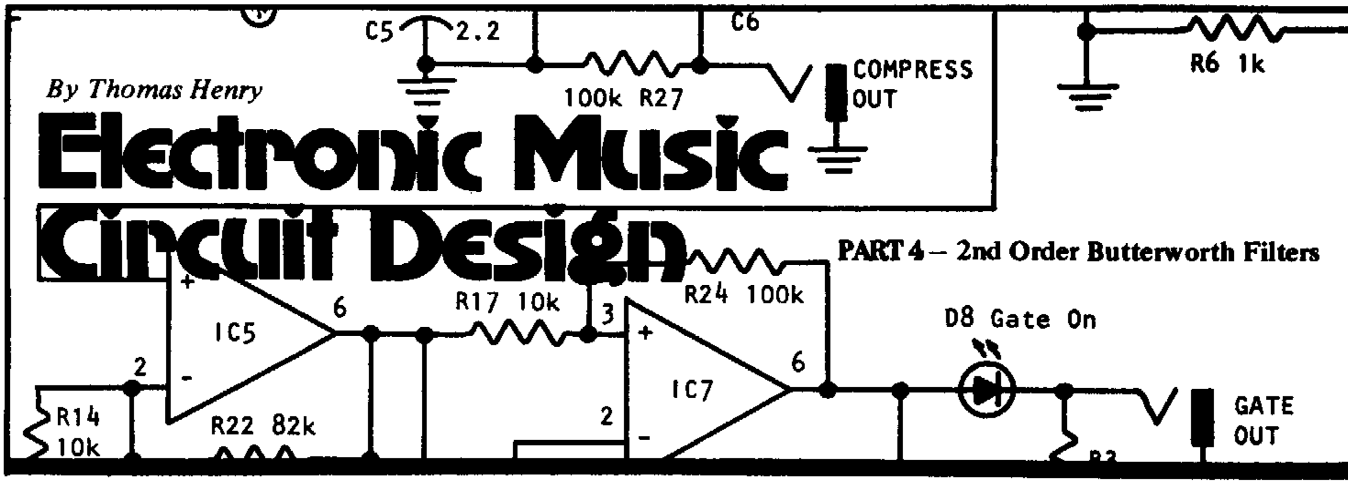
and there's no reason why you couldn't multiplex the various pattern input terminals to achieve more complex rhythmic patterns. I'd suggest not thinking about these switches in terms of drum patterns, but rather, just watching the LEDs and choosing whichever pattern seems most suited to the music you're doing.

The resistor/capacitor combination connected to pin 2 determines the overall tempo. You can increase the capacity to about .05 uF for slower tempos, but past this point the oscillator fails to respond (the data sheet recommends using a .005 uF cap). Similarly, the resistance to ground shouldn't be changed too much from the schematic since too much resistance will also cause the clock to stall. Incidentally, I spent quite a bit of time trying to inhibit the internal clock so that I could synchronize the MM5871 to other clock sources. So far, I've had limited success. You can drive pin 2 directly from the output of something like a CMOS gate or oscillator, but unfortunately that increases the amount of clock feedthrough at the outputs. If you come up with any way to get reliable synching, let me know. Until then, use the internal clock.

The resistor/capacitor combination connected to pin 3 controls the pulse width. Varying either the resistor or capacitor will change the pulse width over a wide range; the values shown give an AMS-100 compatible 1 ms pulse, but feel free to experiment.

This chip is like a top octave divider or similar complex digital chip in the sense that it puts all kinds of spikes and garbage on the power and ground lines. Therefore, you should keep things bypassed and your grounds should be well laid out. Aside from these notes, the chip is remarkably uncritical and can take just about anything you can dish out to it.

APPLYING THE CHIP. Of course, the more envelope generators and other modules you have in your system, the more games you can play with the pattern generator. To describe patches here would be futile; your experiments will teach you a lot more than the printed page ever could. However, rhythmically sweeping filters, voltage controlled distorters, and flangers all give great sounds. I haven't tried out too many combinations, but then again, this is one of those modules that keeps suggesting new uses as you go along. Another possibility is to feed your instrument through a filter bank, with each output of the bank going to a separate VCA. By controlling each VCA with an individual envelope generator, and triggering the different generators with the different pulse outputs, you get a rhythmically filtered sound that is percussive and very interesting. Additionally, the pulse outputs can be used to reset the reset lines present on some of the AMS-100 modules. (continued on page 14)



By Thomas Henry

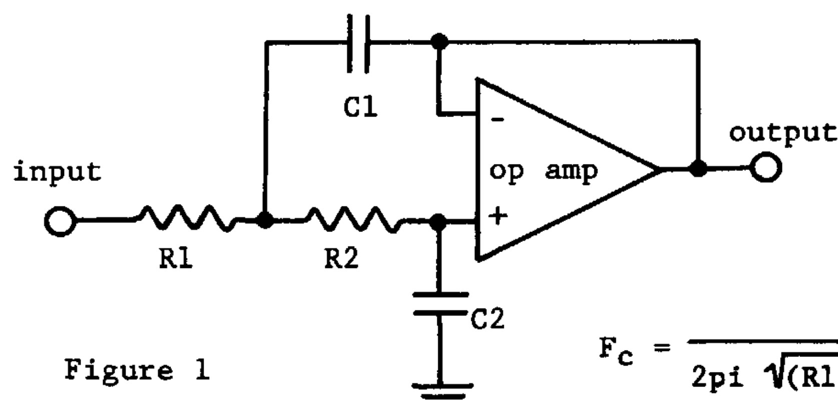
Electronic Music Circuit Design

PART 4 – 2nd Order Butterworth Filters

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

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

2nd ORDER BUTTERWORTH LOWPASS



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

Figure 1

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

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

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

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

A good design procedure would then be:

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

(continued from page 11)

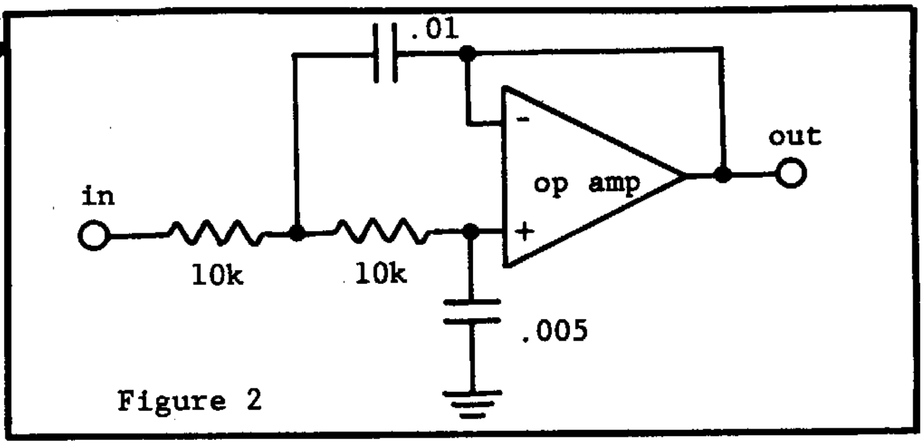


Figure 2

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

1) Pick C2 = .1 uF

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

3) Solve for R from the following equation:

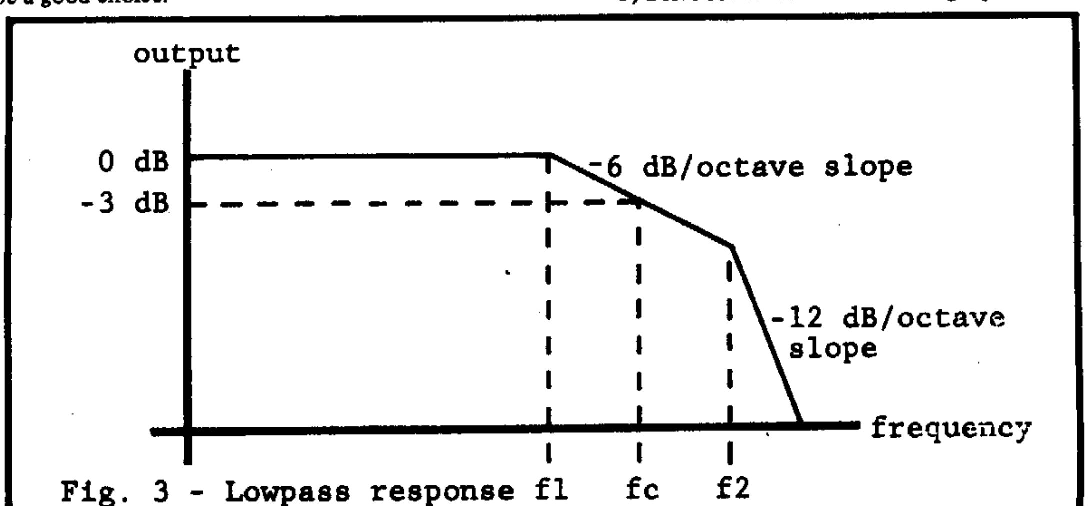


Fig. 3 - Lowpass response

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

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

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

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

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

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

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

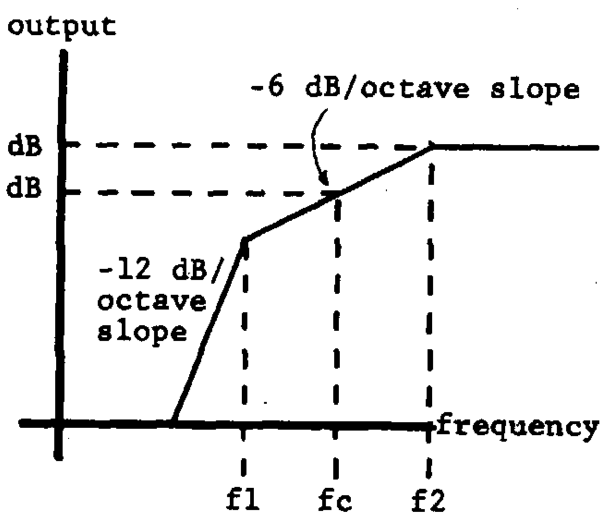


Fig. 4 - Highpass response

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

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

NOTES:

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

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

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

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

2nd ORDER HIGHPASS FILTER (use same equations)

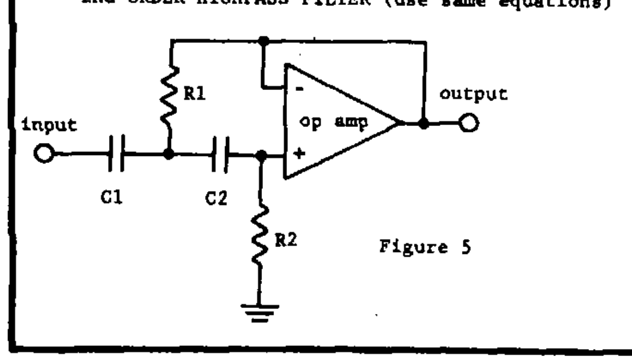


Figure 5

SUGGESTED READING:

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

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

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

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

NOTES FROM NAPA *(continued from page 2)*

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

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

(continued on page 16)

76477 Mixer Tip

BY CRAIG ANDERTON

The 76477 chip is lots of fun to play with, although it does have numerous limitations – but luckily for us, these limitations can usually be overcome. One inconvenience is the way you select the different sounds available from the chip; Texas Instruments' data sheet shows three programming pins (25, 26, and 27), and the voltages on these pins determine which sounds you'll hear at the output. The truth table is as follows. Note that logical 1 is +5 Volts, and logical 0 is ground.

Pin 27	Pin 25	Pin 26	Mixer Output
0	0	0	VCO
0	0	1	SLF (low freq osc)
0	1	0	Noise
0	1	1	VCO + Noise
1	0	0	SLF + Noise
1	1	0	SLF + VCO
1	0	1	SLF + VCO + Noise

On my first 76477 device, I used three toggle switches to program these different sounds. However for most people, thinking in binary terms is difficult, and besides, it's easier to forget exactly which combination of switches produces which sounds.

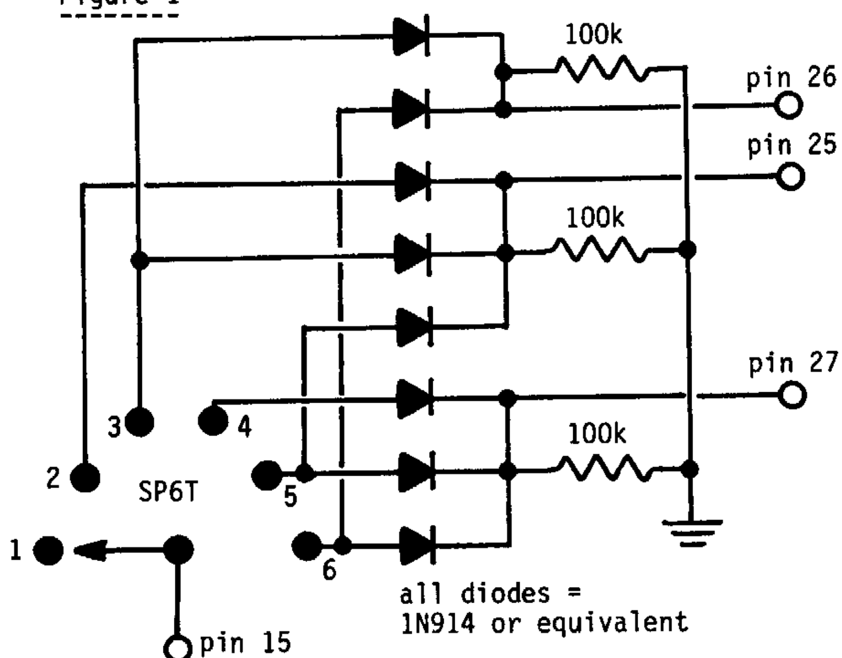
Figure 1 shows a simple decoder scheme, consisting of 8 diodes, three resistors, and an SP6T rotary switch. The circles connect to pins of the 76477, as indicated on the schematic. The switch position table is as follows:

Position 1	VCO
Position 2	Noise
Position 3	VCO + Noise
Position 4	SLF + Noise
Position 5	SLF + VCO
Position 6	SLF + VCO + Noise

Note, however, that with this selection scheme we cannot hear the sound of the SLF by itself; I don't judge this to be too great a loss, and I doubt that you will either.

Well, that's pretty much it. There are lots of other tricks that are applicable to this "universal" chip, which we will cover as space permits.

Figure 1



RHYTHM GENERATOR *(continued from page 10)*

You can also use the pulse outputs to drive electronic drum circuits. If you wish to experiment along these lines, PAIA sells the EK-2, a unit that includes 6 separate drum voices. Should you build it, though, you'd better replace all the ceramic timing capacitors with mylar and polystyrene types if you don't want the drums to drift all over the place with variations in temperature.

WHERE DO WE GO FROM HERE? Unless something new and interesting comes up to grab my attention, next month we'll do another control voltage generator. In the meantime, if you're up for something new, different, and simple, build this pattern generator. It will open up a whole new world of rhythmic sounds for you.

DIALOGUE

DIALOGUE is the place in DEVICE for *you* to get your two cents in. The comments and suggestions have been instrumental in us creating a publication that truly answers your needs *and* gets your ideas and interests out to those of a like mind. Tips, love letters, crank mail, or whatever should be submitted to DIALOUGE, c/o DEVICE, 1085 Broadmoor Dr., Napa, CA 94558

Dear **DEVICE**,

I have run into some drawbacks as the electronic stores here must have IC letters and reference numbers in addition to the basic IC identification number. If you could provide me with any information, I would appreciate it.

By the way, is it at all worthwhile to try to buy directly from the manufacturer? Or do they discourage personal orders for 1 or 2 ICs?

I recently put together the ring modulator but have trouble getting any guitar signal with the modulated signal. Can you suggest any troubleshooting procedure?

Finally, thank you for your time and consideration in these matters and I hope your move to Texas has proven enjoyable. Now that you're in SWTPC country perhaps you can get them to give you some profiles and feedback.

Martin Lindsay
Madison, Wisconsin

Martin –
For the 4739, try XR4739 or RC4739. Use the same prefix for the 4136. Put an LM in front of any of the other numbers and see what happens; National Semiconductor makes lots of different parts, and they probably make the one you're looking for with the exceptions noted above. I don't like to recommend prefixes, since often times several different manufacturers make the same IC, and each applies its own prefix and suffix code.

As far as factory direct orders go, forget it! Companies want to sell 1 or 2 ICs for personal use about as much as they want to get swallowed up by the earth. Your best bet for neat parts is mail order, since mail order companies are the only ones with a broad enough customer base (i.e. anyone with a mailbox) to justify carrying a wide range of parts. They will also not put you through the trouble of finding code numbers.

Re the ring modulator...well, there isn't really suppose to be any guitar signal mixed in with the modulated sound (except for a little leakage). Your best option is to use a mixer, and split your input signal into a straight channel and ring mod channel. Then you can vary the blend to suit your desires. I'd suggest using about a 50-50 mix for starters.

Concerning Texas: Roger says it's great, but I wouldn't know...I'm still here in California, which is where I do my work. As far as being in SWTPC country, they're more oriented towards audio than music, but you're right...we should try to open some lines of comminication there and see what happens. –CA

[As you will all note from last months issue, DEVICE has moved again and we are back in Northern California. Roger]

Dear **DEVICE**,

As you stated, *Audio Handbook* from National Semi is definitely out of print. Digi-Key (Thief River Falls, MN 56701) has some left in stock as of October for $4.00 in case someone needs a copy.

Leroy Markman
San Antonio, TX

Leroy –
Thanks for the tip. I hope other readers follow your lead and turn us on to sources for rare or particularly interesting parts. –CA

Dear **DEVICE**,

I have a problem and hopefully one of you out there can help me. I have a *Sound* amp, model No.1200, with a busted filament transformer. At least, someone told me that's what it was. I really like the presence and brightness of this amp, and would like to get it working. Can anyone help me with info about the transformer, or get me a schematic? If so, it would be greatly, greatly appreciated.

Gratefully,
Jerry Helm
762 Rugby Rd.
Bryn Mawr, PA 19010

Dear **DEVICE**,

I am very glad to see the series "Electronic Music Circuit Design", and particularly glad that you are giving attention to the noise/hiss problem in op amps. I'd like to offer a couple of points in reference to the installment of the series in **DEVICE** 1:8.

Use of 1M source resistors: When I use to build tube guitar amplifiers, I found that you simply had to do something about the grid resistor at the input. The best solution seemed to be to use a good sized resistor (maybe a 1 Watt) and shield it with apiece of grounded braid. Shielding a metal film ¼ Watt resistor in this way might help a lot with the hum problem. However, I have not had hum troubles at a (+) input with a 470k resistor, provided the circuit was in a metal box (even one with the top open).

Additionally, I have found that with bifets the value of the resistor going from the (-) input to ground greatly affects noise. This is the 1k resistor on your diagram on page 9. Some of my problems were solved just by changing from a 10k to a 1k resistor at that point.

The series you are doing is helpful, and I am very glad to see it. One last question: In your test setup, was the input open? If so, how do you know you aren't just measuring hum (rather than hiss) with higher source resistors?

Carl F. Hartman
Newport Beach, CA

Carl ––
Thanks for yet another DEVICE *contribution. With regard to your last question, the input shorts to ground through the resistor. Even so, there are components of hum which can be differentiated from hiss when looking at an oscilloscope. By decreasing the timebase frequency, you can see the hiss actually "riding along" on the hum. – CA*

D.I.Y. POWER AMP *(continued from page 7)*

ules, since they are surplus and stocked in limited quantity. As of this writing, the cost of each module is $8.99; there is a 50 cent postage fee, and a handling fee of 75 cents on orders under $15.

Postscript: About 2 weeks after I got these amps up and running they poped a fuse and went dead. I checked the transformer and modules; they were both extremely hot to the touch. After quite a bit of time spent troubleshooting, I found the problem was due to oscillation in the preamp feeding the amps. Apparently a tone control circuit had become defective, and turned into an oscillator. So, it's pretty clear that these amp modules have *exellent* high frequency
(continued on page 16)

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D.I.Y. POWER AMP *(continued from page 15)*

response – high enough to pass supersonic oscillations from previous stages. These amps shouldn't run hot at normal volumes, so if they seem extremely hot to the touch, check the rest of your equipment before you suspect the amp. By the way, what caused the fuse to pop was that the transformer had become so overheated the insulation started to melt; but it's comforting to know that the amps managed to withstand lots of abuse and still work well.

Finally, about the transformer. I got a hold of a 40V CT transformer, and the quiescent voltage was a couple of Volts over the recommended maximum rating (±29V). So, I'd recommend using a 36V CT type to be on the safe side if you want more power than would be obtainable with a 24V CT type.

– *Craig*

THE BACK ISSUES

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NOTES FROM NAPA *(continued from page 13)*

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Finally, DEVICE readers, we love ya! Your support and enthusiasm have really sustained us through some trying times with this little publication. If we haven't said it enough – THANKS!! *–Roger*

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