DEVICE the newsletter for the electronic guitarist/musician VOL.1:7:79

EDITORIAL:

ON CRITICISM

If you had to make a list of your ten favorite activities, I doubt that "being critized" would be one of them. And yet, criticism is not always a negative force -- if anything, it can lead to very positive results.

By criticism, I don't mean personal attacks; I don't mean trying to boost your self-image by belittling someone else, and I don't mean taking out your frustrations on another entity. Rather, I mean the type of criticism you would call - constructive criticism something that's more related to the word "advice" than anything else. Some examples? Sure...

After the first issue of DEVICE came out, Roger had a long talk with Doug Lynner, the editor of *SYNAPSE*. He loved the idea of what we were doing, but had many criticisms about our format and approach. Now we could have gotten defensive; but our experience has taught us that when someone starts criticizing, the best policy is to shut up and listen... and we learned a lot. All of us benefited from what Doug had to say, and some of the improvements you've noted in DEVICE are directly attributable to his comments.

Another example. John Simonton, PAIA's president, is not only someone I respect, but is also one of my favorite people. Yet, our relationship started because I gave him all kinds of flak about the kind of ads that PAIA was running. John was, and still is, secure enough in his own psyche so that instead of telling me to get lost, he listened carefully to what I had to say and adopted those of my ideas that he thought were applicable. John's ability to listen to advice and criticism from a variety of people has, in my opinion, been one of the prime reasons why PAIA is turning its image around from "well, it's OK stuff for experimenters" to "hey, these people are starting to do some really incredible stuff". *(cont. on page 10)*

IN THIS ISSUE: **PG.**

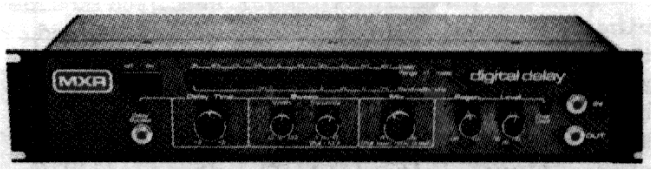
REVIEW:

MXR DIGITAL DELAY

by Cal Stanley

I've had a MXR Digital Delay Unit for well over a year and am still discovering new ways to use it. My unit serves two primary purposes: for doubling and special effects in a multi-track system used to create sound tracks for 3-screen multi-media programs and for various effects in my personal synthesizer recording.

Physically, the machine has a large number of controls on a compact, easy to use front panel. Starting at the left are the power switch, and a bypass/jack and LED indicator. Delay range is selected by pressing one of ten buttons along the top center, in increments of milliseconds, at .31, .63, 1.3, 2.5, 5, 10, 20, 40, 80, and 160. Each of these can be varied from ÷2 to x2 with the Delay Time pot, giving a range of .08ms. to 320 ms. with one memory board as supplied.



Additional control comes from the delay sweep generator. A Delay Width pot controls the portion of the delay range affected (0-100%) and a Sweep Frequency control varies the rate of sweep from .1hz to 10 hz; if the control is pulled out the range becomes 10 hz *(con't on page 2)*

MXR DIGITAL DELAY

(cont. from page 1) to 1 khz giving frequency modulation.

A mix control allows you to vary the levels of the dry and the delayed signal. Pulling this knob out inverts the phase of the delayed signal, especially useful when flanging.

A Regeneration control feeds varying amounts of the delay back into the input and can range from discrete echo to reverb, to runaway echo. Next to the 1/4" phone jack inputs and outputs is the final control. A level matching pot with an overload LED which indicates operation of the internal limiter.

On the back panel are two more phone jacks, one for external delay time control (a remote 50k pot or a synthesizer control voltage) and Repeat-Hold. This is a unique feature, to my knowledge, in which you can play a note or a sequence, depress the foot switch and it locks that sequence (or as much of it as will fit in the memory) and repeat it indefinitely. If you turn the regeneration control up and play a new sequence and click the foot switch on and off quickly you can add to the repeat. Delay range and time and sweep still affect the signal too. There is some loss of high-frequency and some signal degradation, but for the most part it's not objectionable.

Vibrato, cylon-voices, doubling, reverb, discrete echo, echoes that rise or fall in pitch, the possibilities are endless.

I'm experimenting now in using the MXR for a rear channel time delay in a large P.A. system, both for acoustic synchronization and special effects.

On a technical note, the input impedance is high at 200k ohms and output is low impedance at 100 ohms. Noise is 80 db below threshold and is less than .1% at 1kHz, below the overload level. Each memory board, incidentally, has a Random Access Memory chip capable of holding 4096 bits.

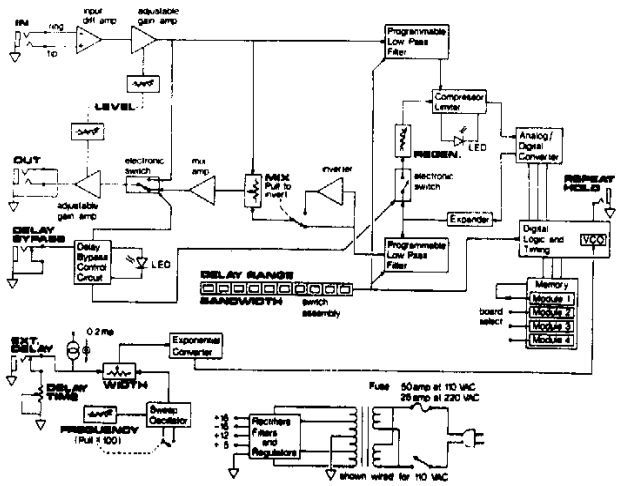
The only real problems I see with the device are (1) High frequency drop-off beyond the shorter delays, (2) A relatively short maximum delay, (3) digital glitches if delay range is switched while in use, (4) lack of a slower sweep speed than .1Hz (5) a slightly over-zealous limiter and (6) only one output.

Problems (1) and (2) were relieved recently when I added two more memory boards giving a maximum delay of 960 ms and 20K response to 120 ms.

The doubling and flanging are excellent and have so many parameters of control that anyone should be able to find the particular sound they want.

There are better DDL's on the market, but I know of nothing of this quality and flexibility for less than two to three times the price, and few that lend themselves equally well to live or studio performance at any price.

Admittedly no digital delay is perfect, but MXR's approach with limiting and a programmable low pass filter (to reduce digitizing noise) all represent a logical, effective and affordable approach to delay.



SOME ADDITIONAL NOTES ON USE:

1. I generally keep the mix control all the way to the delay setting and mix the signal in through another channel, since I work with an 8-track recorder and a 12 channel board I have channels to spare. In a more modest set-up the ability to mix signals inside the DDL might be essential.
2. Doubling with the DDL seems to work as well with voice or guitar, and can be adjusted from a hardly noticeable "fattening", to a clear, slightly off-set second voicing or to more bizarre sounds by increasing the delay or sweep speed.
3. For what I call wah-flanging, punch up the shortest delay, set delay time at ÷2 and boost the midrange EQ on the delayed signal. Different sweeps can be performed by rotating the delay time and width knobs.

Do It Yourself Circuit Boards

by rick norman

After seeing the letters sent to DEVICE and other magazines asking for board art work, or boards themselves, I've decided to do some hunting and pecking on my typewriter, and show you an easy way of laying out your own boards. It is definitely more fun, and there is a lot of satisfaction to be derived from being able to point at your completed effect, and say "I made it", and mean that you made it all.

The method I use is not scientific, or hard. I'm no electronic genius. I'm just a guitarist, who, like all of you, is looking for more sounds from my axe, but can't spend a whole wad of money. So, I've turned, like many others, to Craig Anderton's book, Electronic Projects For Musicians, and learned some basics there. Now I'm exploring new ground with Device.

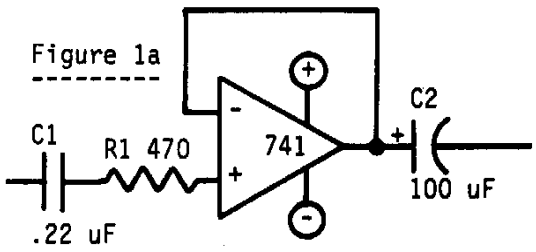
With my method, I've designed boards for all kinds of things: a number of Craig's G.P. column projects, and now the AMS-100. I guess I shouldn't really call this my method, as I'm sure it's not original; there are probably even some of you who use it now. There have been a great number of people to whom I've spoken about this, including some I'd classify as electronic "heavies", and so far no one else has heard of this, but they all think it's neat. Enough background, let's get going.

What we'll talk about first is the how-to, then talk about a number of things to take into consideration before you actually dig in.

Start out by getting yourself a pad of regular graph paper (1/10" grid preferred...don't use the log type), a pencil with a fine tip (like a mechanical pencil) and a BIG eraser. You'll need that last item a lot. Next, get a piece of regular P-pattern perf board, and all the parts you will be using in the circuit. Take a piece of scratch paper, and list the parts along with the space each thing takes up on the perf board; ie; 1/4 watt resistor - 5 holes, .22uf mylar cap - 5 holes, 220pf cap - 3 holes, etc. Once you've got that, tear off a sheet of graph paper, sharpen your pencil and begin.

What the graph paper represents is a piece of perf board. Every place that the horizontal and vertical lines cross would be a hole in the perf board. We use both sides of the paper, one side will be the component side, the other, the traces. Start by making dots on the graph where the pins for the IC go. Do this on both sides of the sheet. I have always found it easiest to start at the IC, and work outwards. Let's say we are building the circuit shown in figure 1A. The next item in line is the resistor. On the component side, move out a couple of "holes", and make two dots, five spaces apart. Then draw in the resistor between them, as shown in figure 1B. Be sure to label the parts as you add them, so you'll know where you are. Now turn the paper over, and make two dots in the same places, and connect one of those to the dot that represents the appropriate pin of the IC. The next component (again, moving away from the IC) is the capacitor, so, again, make two dots the appropriate number of spaces apart, and draw in the cap. Turn over the paper and make the same dots, and connect one dot to the un-used dot from the resistor. Doesn't this sound like a game we played as kids? (see figure 2)

Figure 1a



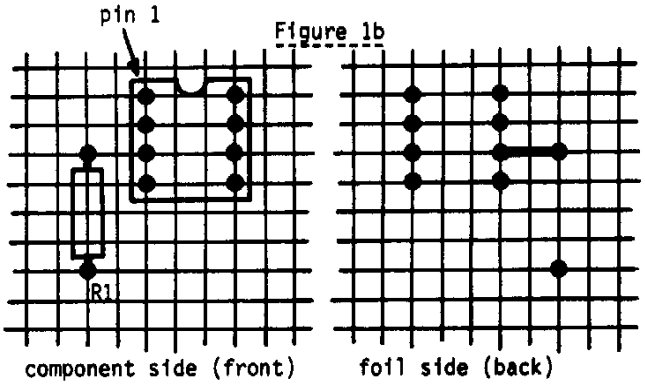
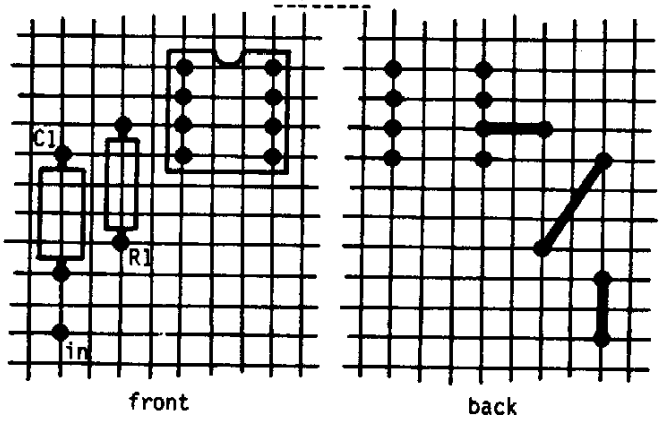


Figure 1b

Continue doing this with all the components until the circuit is completed.

(cont. on pg. 4)

Figure 2



(cont. from pg. 3)

You should end up with something like figure 3.

As you do more complicated circuits, you'll find that the layout gets more complicated. This is when you start appreciating the fact that you bought a big eraser. Things don't always end up where you put them in the beginning. Don't be afraid to erase things, and move them around to make it all fit. Therein lies the beauty of pencil, and eraser. Nothing is final. There are some snags that no amount of component moving will help——the inevitable case where you need a line to get from here to there, but there are a few other lines in your path. Enter the jumper line. I personally don't like them, but in the final evaluation, they work, and are simple to use. Another alternative is to use double sided board layouts, but you'll discover those later, all by yourself. These problems usually arise in circuits having more than one IC, and the need arises for grounding and power connections.

Now that we have the how-to, here are some important characteristics to take into consideration:

1) Compactness. Try to make your board as compact as you can. Remember, this thing has got to fit into your pedal, or whatever, so don't let this thing stretch out like the Great Plains. There are a couple of things that will limit your compacting act. These are discussed below.

2) Functionalism. Don't make the board so compact that the parts are over-lapping, the traces come too close together, and other undesirable things that invite disaster.

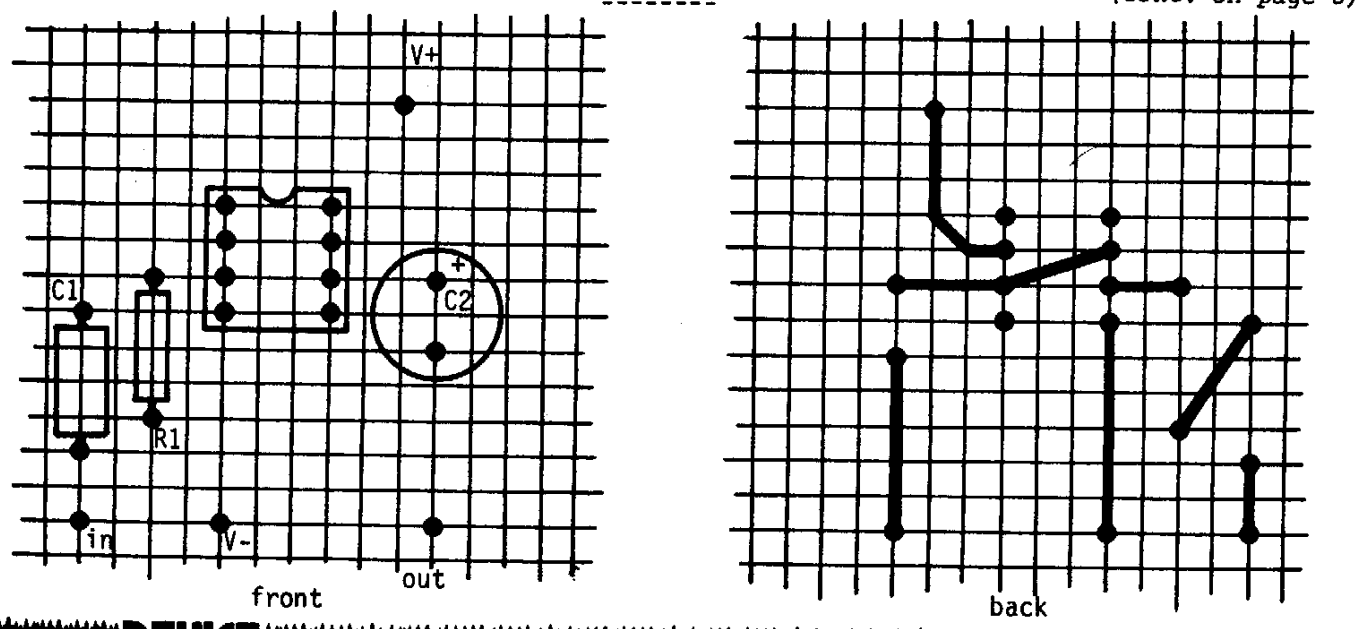
3) Soldering Technique. Realistically examine your soldering skills. If you're not real good at it yet, don't crowd the parts, and hence, their solder pads too close together. You invite solder bridges that way. Even if you can solder well, don't get too carried away. Everyone blows it at some time, so there's no sense in increasing your chances.

4) Mounting Method. If you plan to mount the board on corner studs, or whatever you like, either leave enough extra board so the traces don't come near the stands, or just avoid running the traces in those places.

5) Type of board that you will be making the finished product on. There are a couple of ways to go, here, and I'll discuss them in a little more detail in a minute. You can produce your final product on copper clad perf board, or on a non-drilled copper clad board. Decide

(cont. on page 8)

Figure 3



ADDING SERIES-PARALLEL TO THE EH BIG MUFF pi DELUXE

by thomas henry

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

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

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

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

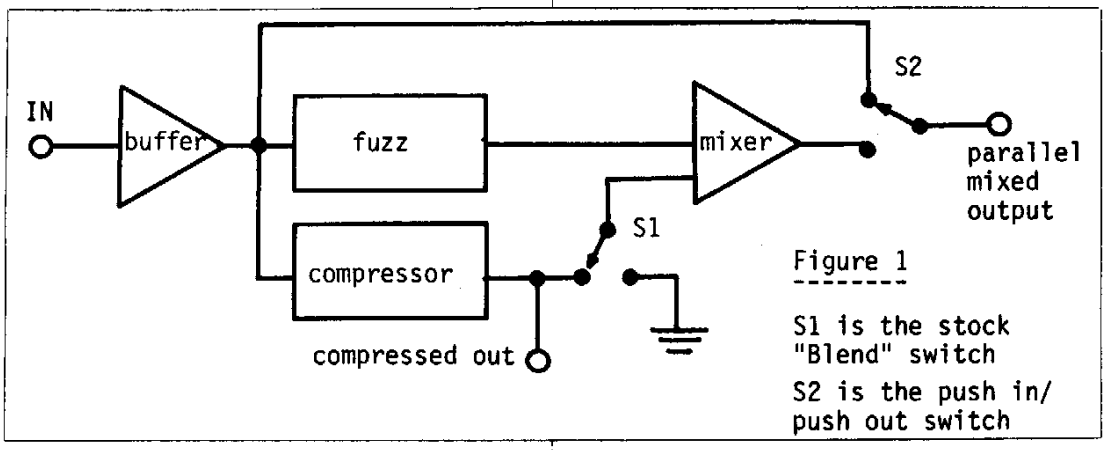


Figure 1

S1 is the stock "Blend" switch

S2 is the push in/ push out switch

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

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

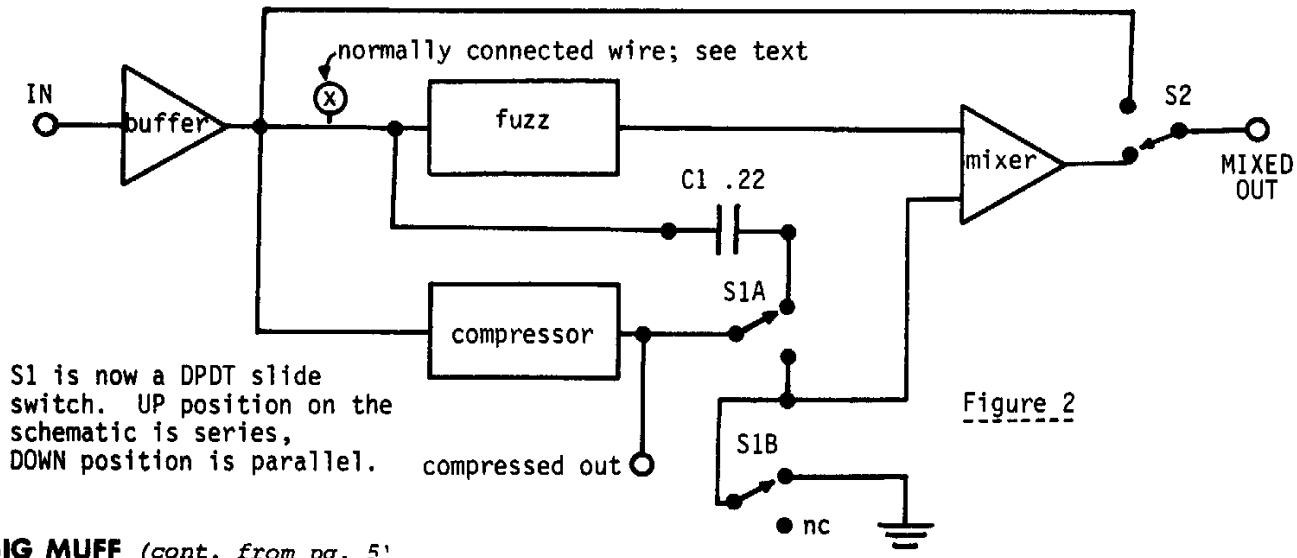


Figure 2

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

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

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

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

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

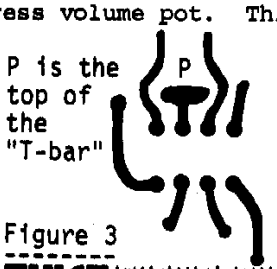


Figure 3

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

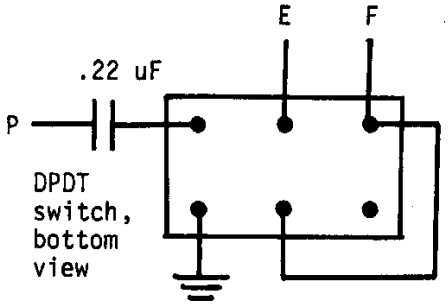


Figure 4

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

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

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

With the switch in parallel mode, the unit is wired exactly as before. Bring up the compress volume for pure compression. Turn it

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

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

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

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

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

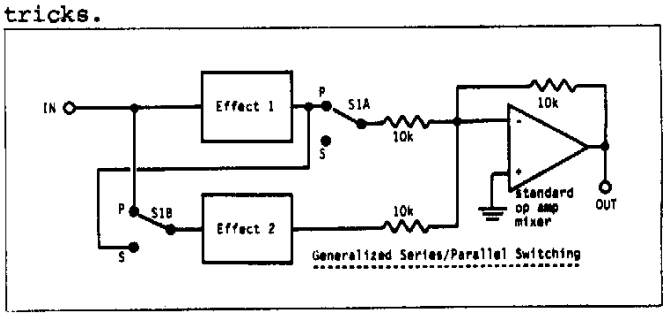
Carefully roughen up the word "Blend" with the end of an ink eraser. <u>Do not</u> scrub the words, just roughen lightly. Then rub some acetone on the word with a cotton ball. Careful! Acetone is highly inflammable, so don't smoke while doing this. Better yet, don't smoke at all and do yourself a favor.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

GENERALIZED SERIES/ PARALLEL SWITCHING

by Craig Anderton

Even though I don't own a Big Muff Deluxe fuzz, I found Thomas Henry's modification article in this issue quite interesting. It started me thinking about coming up with some switching configuration that could put two effects in series or in parallel; additionally, I wanted this configuration to be applicable to any two effects, not only to a specific device.

For those of you who are into collecting switching tricks suitable for pedalboards, the following design makes a good addition to the two switching tricks presented in my June '79 *GUITAR PLAYER* column. What's more, there are no doubt other clever switching ideas out there just waiting to be discovered ...John Blacet's novel switching scheme in the "Phase-filter's" (review in DEVICE 1:6) is a good example of what I ' m talking about. Looking towards the future, John Simonton keeps telling me that everything should be electronically switched and digitally controlled via computer; that makes a lot of sense, and seems very applicable to these various switching tricks.



Anyway, back to the series/parallel switching configuration. As shown in the schematic, you need two effects that eventually feed into some kind of mixer. These effects should have relatively high input impedances, say 100k or more, to avoid loading down any effects located earlier in the signal chain. Under some situations the output mixer may not be necessary, but is generally advisable.

S1 is a DPDT toggle - or foot-switch. The 10k resistors shown in connection with the mixer could also be 100 k resistors or whatever else you have on hand -- however, the→

(cont. from pg. 7)
lower an impedance you can drive with the two effects, the lower the overall noise level.

That's pretty much it; the circuitry should be pretty much self-explanatory. In the series (S) position of S1, Effect 1 feeds Effect 2, whose output then dumps into the mixer. In the parallel,(P)position, Effect 1 and Effect 2 are in parallel, and both outputs feed into the mixer.

What are some good applications for this circuit? Well, using two phase shifters works well...if you're using two, four-stage shifters, putting them in series gives you an 8 stage phaser, and putting them in parallel gives the sound of two paralleled four stage units, which is quite different. Dual analog delays are also useful in this configuration; series mode increases the maximum available delay time, while parallel gives added "depth" when both lines are set for different delay times. One delay time can be sent into the mixer at a lower level when in the parallel mode, thus giving more of a reverberant effect.

Two dissimilar effects can also work well. Of course, the fuzz/compressor duo on which Thomas Henry's modification was based is useful, as is compressor and filter, fuzz and filter, and so on. In any event, you'll no doubt disover certain favorite combinations - enjoy yourself and have fun with this simple circuit. ■

Will the **RESYNATOR** point the way for synthesizer/instrument interfacing? For news on this and other exciting developments watch the pages of **DEVICE**.

(cont. from pg. 4)
which you are going to use before you start the layout. If you are planning to use non-drilled board, you can make the board smaller (most of the time), as you can run the lines closer together. The rows of holes in perf board limit too many parallel runs.

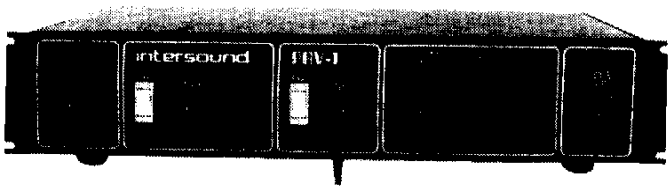
So, you've got your finished drawing? Check it! Over, and over, and over! Be sure it's right now, while you can change it with your eraser. Don't find out after the board is made, and maybe you've blown up some parts, that the board layout was wrong. It's too late, and too hard to change it then!

Now that you've got this drawing of what the board should look like, let's transform it into copper, and fiberglass. The quickest, and easiest way to go is with copper clad perf board. I use this for many one-shot items. Simply cut a piece big enough to hold the circuit, and leave room for the mounting holes. After cleaning off the oxidation, transfer the drawing to the board by simply making a pad with your resist pen (or whatever), any where you have a dot, then connect them as you did on paper. Let the first coat dry, then apply a second. Don't worry yet if some of the traces touch each other where they aren't suppose to. When the second coat dries, take a fine pointed X-acto knife, and scrape away the ink between the offending traces. Got it? Now, toss it in the ferric chloride (figuratively speaking, of course).

The second method is a little more involved. Here, you'll need a piece of non-drilled, copper clad board of the appropriate size, and a piece on non-clad perf board of the same size. Take the piece of perf board and mark the holes that are indicated by the dots on the paper. Then take the piece of copper clad board, with the copper facing you, and clamp the perf board on top of it. Chuck up a #58 - #60 drill bit, and drill through the marked holes in the perf board, which is acting as a template. The tiny drill bits are available from most any hardware store for around 59 cents. Buy a few, as they bend easily, and are pretty cheap. Now you have a board with the holes only where you want them, so play connect-the-dots again, just like you did for the perf board, and etch.

Whichever method you used, the final step is to install the parts as shown on the component side of your drawing. Install in your favorite enclosure and Blast!! If you have never done this before, I don't recommend taking on the AMS-100 right away; start small, and work up. For those of you who want AMS-100 board artwork, I plan to provide mine when all the circuits have been published. Watch Device for more news.

Now, wasn't that easy? ■



INTERSOUND(the IVP people) has introduced the PRV-1 PARAMETRIC EQUALIZER-REVERB a single channel reverberation device with two bands of full parametric EQ. Rack-mountable, the PRV-1 is designed for sound reinforcement, studio, and live stage applications. The PRV-1 features: LED level display, independent level and signal routing controls for both reverb and line signals, variable low-cut filter for feedback control and low end frequency shaping, active balanced and unbalanced inputs and outputs, and external footswitch control of the reverb. The PRV-1 utilizes an Accutronics delay with special drive circuitry tailored for very quiet, natural reverberation. For more information contact INTERSOUND at: Box 1864,Boulder, CO 80306. Look for a DEVICE review in the fall.

Power FETs Getting Cheaper all the Time. Designers look to the power FET as a device that potentially combines the best features of tubes and transistors. Up to now, restricted power and relatively high prices have kept power FETs from massive acceptance; but SILICONIX recently announced the VN10KM, a power FET that handles 80V and 1/2 Amp (40 Watts). Price is 50 cents in large quantities, so it probably won't be too long before manufacturers start using power FETs in medium power amplification equipment.

RIVERA RESEARCH AND DEVELOPMENT CATALOG NOW AVAILABLE. RRD sells replacement wood cabinets for Fender and Music Man amps, as well as extension speaker cabinets, utility cabinets, effects pedals, anvil cases, replacement speakers, and so on. The replacement cabinets are available in oak, koa, exotic woods, and in some cases, plywood. Some of these have to be seen to be believed --- they're beautiful. For more information, send to P.O. Box 641, Tujunga, CA 91042.

INFO

Micropower 555s. Despite its popularity, the 555 has a couple of drawbacks; two of these are high current consumption and generation of spikes during typical operation. EXAR's L555 requires 1/15th the power of a standard 555, and operates down to 2.7V without sacrificing timing accuracy or frequency stability. INTERSIL offers the ICM7555, a CMOS equivalent of the standard 555 timer, with current consumption 1/20th that of the standard part (about 80 uA). Supply voltage can vary from 2V to 18V.

MOEBIUS the L.A. based synthesizer ensemble (that just so happens to include our friend and *SYNAPSE* editor Doug Lynner) will be releasing their first album shortly on MOONWIND RECORDS. We have heard some of the tunes to be included on the album and strongly recommend that you purchase a copy (also that you twist your local DJ's arm till he adds it to his playlist.) For more information on availability watch these pages or contact Doug at *SYNAPSE* 1052 West 6th St., Suite 424, Los Angeles,CA 90017.

ROLAND UNVEILS NEW RACK SYSTEM. ROLANDCORP U.S. (2401 Saybrook Ave., Los Angles, CA 90040) made an impressive display at the Atlanta NAMM convention with the introduction of the ROLAND RACK amplification and effects system. Included in this system are two models of instrument pre-amps (the SIP-300 guitar pre-amp and the SIP-301 bass guitar pre-amp), two power amps (the SPA-240 and SPA-120), and five different signal processing devices: a vocoder, flanger, pitch-to-voltage/synthesizer, digital delay, and a psycho-acoustic effect called Dimension D. The system concept is designed to offer the musician maximum flexibility in designing his own amplification and effects set-up to his individual tastes. The RACK products are built to existing industry standards of 19" in width and 3.5" in height.

(cont. from pg. 9)

DUAL LIMITER FOR RECORDING DUE FROM PAIA. A dual limiter kit designed by Craig Anderton for recording studios will be available from PAIA Electronics in the fall for under $50. Based on the Signetics NE 570 IC, the unit may be configured as two independent limiters or a stereo limiter with gain changes in one channel producing corresponding gain changes in the other channel. Controls include a variable limit point for each channel along with a monitor LED. The kit is rack-mountable. The limiter design will be the subject of an upcoming article in *MODREN RECORDING* magazine.

PANASONIC ANNOUNCES NEW DELAY CHIPS. The BBD3009 is designed for short time delays (up to 12.8 ms), and includes 256 stages. It is claimed to have near 0 insertion loss, a signal-to-noise ratio of 88 dB, and total harmonic distortion less than 0.3%.
The BBD3008 is a 2048 stage device designed for somewhat longer delays. Typical signal to noise ratio is around 78 dB; maximum delay is greater than 100 ms.

CURTIS ICS NOW AVAILABLE TO HOBBYISTS. Doug Curtis of Curtis Electro-music Specialties and John Simonton of PAIA Electronics have agreed to have PAIA distribute CEM chips to the hobbyist market. Initial products scheduled for release in approximately 12 weeks include the CEM3310 voltage controlled transient generator, CEM3330 dual voltage con - trolled amplifier, and CEM3320 voltage controlled filter that features on board linear, voltage controlled resonance. An exponential VCO is slated for early 1980. Prices are not yet set, but are expected to be competitive with the SSM/Eu series of ICs.
This is good news for hobbyist, who now have two sources for high quality musical ICs (prior to now, the only readily available chips dedicated to electronic music have been the SSM/Eu chips, which have been covered extensively in Electronotes and also here in Device). It's also good news for me, since at the meeting I was asked if I'd like to design several new analog synthesizer modules for PAIA based on the Curtis chips. I've always wanted to design a synthesizer from the ground up, and this is my chance to do it. All in all, the distribution agreement between PAIA and CEM should make life just that much easier for all of us do-it yourselfers.

SOURCE FOR DPDT FOOTSWITCHES. Carl Hartman notes that Musician's Supply (PO Box 1440, El Cajon, CA 92022) carries a DPDT push-on push-off footswitch for $3.25. The stock number is 111006. Write to Musician's Supply for additional information.

BOURNS INTRODUCES SWITCHABLE SIGNAL ATTENUATOR While intended for telecommunications, if you have been looking for a precision mic attenuator this might be the device for you. It is housed in a trimpot-sized package, and includes 4 binary-weighted switches that set the attenuation from 0 to 15 dB in 1 dB steps.

COMPLETE CONTROL. Just as there seems to be more interest in LED VU meters, there is also a lot of interest in remote control devices. National's latest entry into the field is the LM1872 (reciever) and LM1873 (transmitter) chips. The receiver is a true superhet type, and includes digital decoding on chip. It shouldn't be too long before you switch effects by wireless, as well as broadcasting your signal by wireless.

ON CRITICISM *(cont. from pg.1)*

Today I got a call from the president of a well-known effects company who had just read the results of the effects opinion poll. Our readers' views of his products clearly did not coincide with how he thought people looked at his products; he was taken aback by some of the criticisms we printed, and wanted to get more background on the survey. Interestingly enough, it seemed like he didn't see the favorable remarks...but then again, I suppose that most people are far more sensitive to critical comments. Far from being abusive or complaining, he wanted to know more -- so I did my best to help, and gave some suggestions that I thought he would find useful. The call ended with him feeling better about the whole thing, and recognizing that you can't be complacent -- you have to keep on your toes, always try a little harder, and not expect momentum to carry you into the future. Frankly, I was impressed by the person's attitude and delighted with the situation in general, because it meant that a manufacturer had really listened to what you had to say. And better yet, instead of contacting us to call us jive turkeys (or worse), he took your collective comments very seriously and respected what you had to say. So, although the survey results may have caused this person some initial distress, they also shook him up and got him thinking...which can only benefit his company, and therefore benefit the musician who buys the products. I think this is a case where, despite the intitial sting of a piece of criticism, in the long run everyone wins. I'm not so naive as to think you can please all the people all the time; but, any way you can please more people is certainly welcome.

It takes a certain amount of courage to face critical comments, we all know that. But,

(cont. on pg.16)

build it!

5 STEP LINEAR LED BAR GRAPH VOLTMETER

The following circuit was submitted by John Blacet of Blacet Music Research (18405 Old Monte Rio Road, Guerneville, CA 95446). It is a linear, 5 step LED meter circuit that has four switchable ranges and responds to either AC or DC. It makes an excellent AMS-100 module, since it uses the AMS-100 standard ±15V supply and can monitor the system's control voltages or give an indication of audio signal levels. It also makes a useful general purpose module for checking battery levels, telling whether something is putting out a signal or not, monitoring logic circuits (logic probe), and so on. A kit of parts and circuit board is available from Blacet Music Research; write them for pricing.

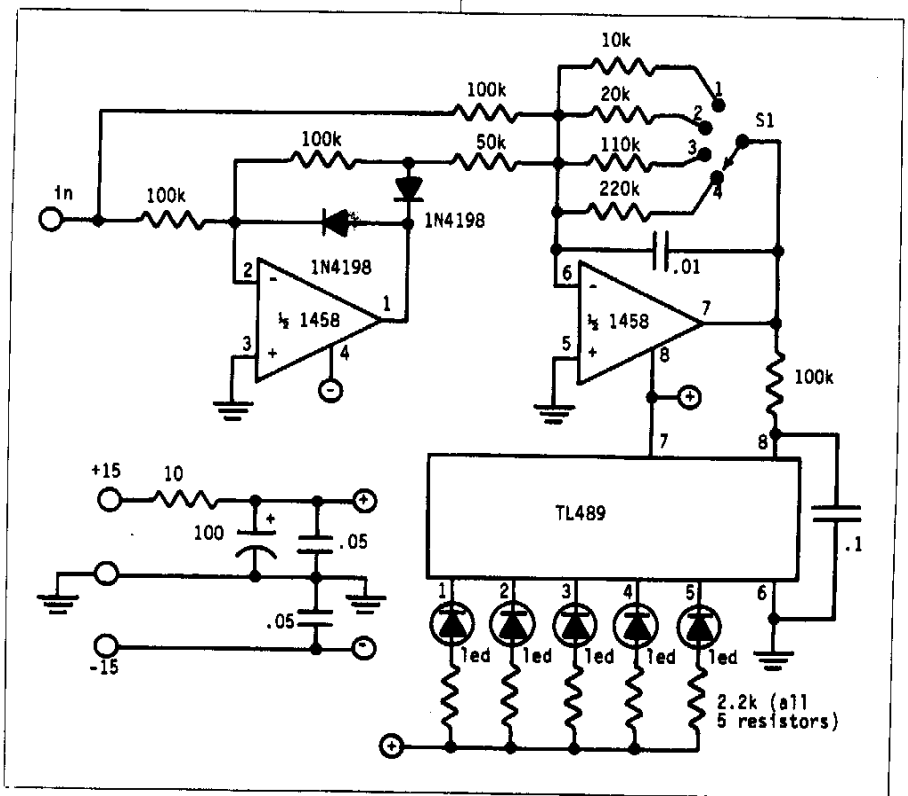
HOW IT WORKS. Half of the 1458 dual op amp is a precision half-wave rectifier; summing the input signal and half-wave rectified output in the second 1458 products a full-wave rectified output. S1 is the range selector switch, and selects one of four feedback resistors to give different sensitvities. The following table lists your choices...

Switch Pos.	Range	Volts/LED
1	0-10V	2V
2	0-5V	1V
3	0-1V	.2V
4	0-0.5V	.1V

The output of the second op amp feeds into the TL489, a 5 step linear level detector from Texas Instruments. Each of its five outputs sinks current through an LED, so that as you increase the voltage more LEDs light up. For example, in the 0-10V range no voltage means no lit LEDs; exceeding 2V lights up one LED, exceeding 4V lights up two LEDs, exceeding 6V lights up three LEDs, and so on until all 5 LEDs are lit.

While a linear response is good for monitoring control voltages (hook this up to the output of any of the AMS-100 envelope generators, for example), it may also be used with audio signals. In this instance, the display indicates the approximate RMS voltage. In the 0-1V range, turning up a sine wave so that the fourth LED just comes on means you're just a bit above 0 VU (.775V AC RMS).

FINAL COMMENTS. That's really all there is to it. You'd be surprised how handy it is to have a little Voltmeter like this hooked up as a synthesizer module; it can tell you a lot about what's happening in your system as you patch it around various points. ■



electronic music circuit design | | part 1 - the op amp

CRAIG ANDERTON

So many of you have asked questions about circuit design that Roger and I figured it was time for me to institute a semi-regular column on design techniques. We'll start off with op amps, since they are pretty universal circuit building blocks.

OP AMP BACKGROUND. Figure 1 shows the typical schematic for an op amp such as the 741. Note that there are two inputs: Applying a signal to the (-) or inverting input gives a 180 degrees out-of-phase signal at the output; applying a signal to the (+) or non-inverting input produces an output signal that is in-phase with the input.

Op amps like to work with bipolar power supplies; that is, two voltages that are equal and opposite to each other. The precise value isn't too important, so ±9V is a convenient choice for battery power, while ±15V is often used in synthesizer systems. We'll see the significance of different voltages in later installments of this series. Most op amps will not function properly below ±5V or above ±18V; these are considered the maximum and minimum allowable supply voltages for the majority of op amps.

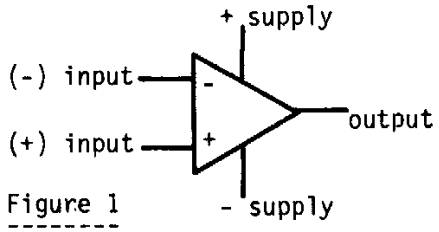


Figure 1

DC APPLICATIONS. The op amp was originally designed for use with analog computers, and only recently have op amps been designed with audio applications in mind. As a result, to fully understand how op amps work we need to look at their "roots" for a little bit, and how they are used in DC applications.

An analog computer is really more like a slide rule (remember those?) than something like a pocket calculator. By using a calibrated voltage source, an accurate op amp, and a Voltmeter, you can actually perform multiplication, addition, and other mathematical operations. For example, suppose you wanted a circuit that would compute the result of multiplying a number by 2. You would set up the op amp to give an exact gain of 2, then set up a Voltmeter at the output and a calibrated voltage source at the input. To multiply 2 times 2, you'd apply a 2V input to the op amp. Looking at the output of this gain-of-two amp, you'd end up with a 4V signal. So, 2 X 2 = 4. Or, you could apply 1.5 Volts to the input of the op amp, and the meter at the output would read 3V -- so 1.5 X 2 = 3. If you wanted to find out the results of multiplying a number by 3, you'd set up the op amp for a gain of 3, feed in your voltage representing a number, and look at the output voltage (which would represent the answer).

Pretty boring stuff, eh? Well, not really. Think of all the times you need to multiply voltages in electronics: After all, "amplification" is the audio equivalent of the mathematical word "multiply"...and we certainly need to amplify signals a lot in the course of doing electronic design.

The example we gave is a non-inverting application; in other words, if we feed a positive voltage into the input, we want to see a positive voltage at the output. Figure 2 shows the basic configuration of a non-inverting DC amplifier using an op amp (power supply connections are assumed, which is often the case in op amp schematics). That's all you need...breadboard it, feed in some voltages, and look at the output; you'll find you have a very limited analog computer on your hands. The gain (multiplication factor) is given by the formula:

$$\frac{R1 + R2}{R2} = \text{gain}$$

So if R1 and R2 are both 10k Ohms, then

$$\frac{10{,}000 + 10{,}000}{20{,}000} = 2$$

...which means we have an amp with a gain of

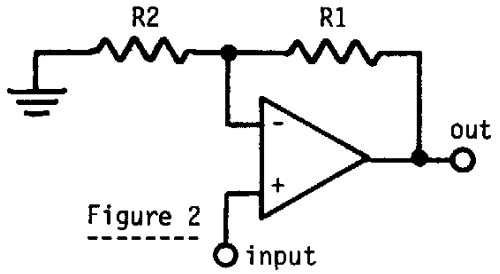


Figure 2

two. For a gain of 11 amp, we can make R1 = 100k and R2 = 10k, giving us:

$$\frac{100{,}000 + 10{,}000}{10{,}000} = 11$$

Note that you can't have lower than unity gain (gain of 1) with this configuration; if R1 is a short circuit, you get unity gain regardless of the value of R2 (plug the numbers into our gain formula and you'll see why...R2 + 0 divided by R2 is always going to give a gain of 1).

Without R1 or R2 in place, the op amp is said to be running at open loop gain -- a figure that can literally be in the millions However, we seldom (if ever) need that much gain, so apply negative feedback (180 degrees out of phase) from the output back to the input via R1. This has a canceling effect that reduces the gain. If R1 is small compared to R2, then there will be much negative feedback and the gain will be very small. If R1 is large compared to R2, then there will be little feedback and the op amp will run closer to the open loop gain. R1 and R2 form what is called a "voltage divider" -- we can draw our non-inverting configuration just a little differently (see figure 3) and relate our voltage divider to a standard volume control. When R1 = R2, you are in effect halving the amount of negative feedback by picking up the output voltage at the halfway point of our volume control. This results in a gain of two. If we were only feeding back 1/4 of the output signal, we'd have a gain of 4; if we were only feeding back 1/100 of the output signal, we'd have a gain of 100...and so on.

EXPERIMENTS YOU CAN DO. Try breadboarding a 741 op amp configured like the setup in figure 2 with jacks at the input and output. Apply power with two 9V batteries (one positive with respect to ground, the other negative); note that the junction of the two batteries (ground) does not connect to any point on the op amp, but does connect to the input and output jack grounds. Plug an instrument such as guitar or a microphone into the input while listening to the output. As you change the values of R1 and R2, you will hear changes in the gain. You can create a unity gain buffer by making R1 = 0 Ohms (you may then eliminate R2 since it serves no useful purpose), or make a preamp by making R1 large compared to R2.

ENTER THE VILLIANS. We've been assuming all along that our op amp is perfect -- specifically, that when left alone the (-) and (+) inputs are at ground potential (zero Volts), just waiting for us to do something with them. Unfortunately, this isn't always so; there is a small voltage discrepency, or input offset voltage, that keeps these inputs from being at exactly ground. Normally, these offsets are in the milliVolt range...so you might not think that's too bad. But wait a second; let's say that we have our op amp set up to give a gain of 100, and there's a 10mV offset present at the (+) terminal. That means that even if we don't apply any voltage at the (+) input, the op amp thinks we've appl-

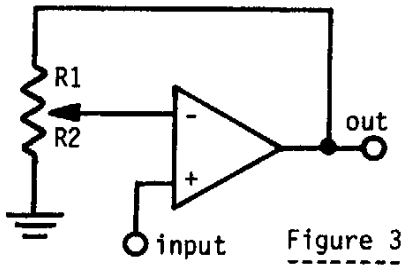


Figure 3

ied a 10 mV signal -- and promptly amplifies it by 100, giving us 100 X .01V or a 1 Volt signal at the output. Needless to say, this screws up the accuracy of our analog computer! Luckily there are ways to eliminate offset problems, which we'll cover later on.

Another villain is feedback -- the same feedback that plagues PAs and such, but under a slightly different guise. At higher frequencies, the gain of the op amp tends to fall off (remember, they were intended for DC applications and didn't need a lot of high frequency accuracy) and the phase of the output signal starts shifting. In fact, at some high frequencies although you may think you're still applying negative feedback to the op amp input through R1, you may actually be applying positive feedback due to the phase shift, thereby creating oscillations and other foul problems. In some cases the high frequencies won't have enough power to cause oscillations, but with high performance op amps that do have a good high frequency response, it's easy for the phase shift problem to occur and cause oscillations. The usual way to solve this problem is to limit the bandwidth of the op amp by filtering out the high frequency range where problems can occur (usually well out of the audio range, so the filtering doesn't mess with your sound). In our next column, we'll discuss how to band-

(cont. on pg. 16)

DIALOGUE

We'd like to thank you for your comments and suggestions, as they help us to create a publication that truly answers your needs. Send ideas, love letters, crank mail, or whatever to DIALOGUE, c/o DEVICE.

Dear DEVICE:

Have received a couple of issues of your new publication "DEVICE" and enjoyed looking through it very much. I hope that possibly you can be influential in getting some sort of informal "standardization" happening amongst all the folks who are making "devices".

For us people who are trying to provide interfacing connections for our amplifiers, it's just damn difficult to provide "loop" facilities that will suffice for any possible load/level situation. Of late, there are increasing numbers of both line level and and guitar level effects devices and many of the guys trying to use these are absolutely going crazy in the field. Frankly, as an amp manufacturer, it frosts my ass to get the blame for some device which is either poorly designed, poorly manufactured, or just simply misused causing terrible results when used in our product.

I certainly hope that yours can be a "voice" of reason" among the chaotic claims and counterclaims presently being batted around in the marketplace.

In any case, best of luck with your new publication and if there's anything we can ever assist you with, please feel free to contact me.

Sincerely,
Hartley D. Peavey, President
Peavey Electronics Corporation

Dear Hartley,

Your support is greatly appreciated. We are developing some propose standards for efects and will present our ideas in DEVICE. Hopefully others will pick up on this and offer comments, both pro and con, about what is needed.

As for amp effects loops...off the top of my head, it seems a low impedance (less than 1k), line level out effects send would be the best overall compromise. This way, you could plug your effects send output externally (I figure it's always easier to pad down than amp up). Sure, someone will plug a low level effect into the effect loop at some point, and the thing will distort...but perhaps you could have a little printed sticker next to the effect send jack, stating clearly that it is for line level devices and give its specs. What do you think?

As to if there's anything you can do to assist us, you already are by writing and getting involved...and putting your opinions on the line for comment. Thanks again.

Dear Device:

I have a couple of questions regarding a possible career with electronics and music.

I am 20 years old, play bass, am heavily into electronics, and am going to a credited school. I am interested in going into the field of musical electronics, such as recording studio, working with electronic effects and amps, possible design work, etc. However, talking with a few instructors at my school makes it sound like my chances are slim to none. They say I have to go to audio school, have a superb ear, gain plenty of experience, and even have good connections.

Any answers? How do I go about it?

Don Morton
Oak Park, Michigan

Don,

Well, you're subscribing to DEVICE, so you're off to a good start. And I'm not being flippant -- you're going to need to know at least as much as the Next Guy, and hopefully more if you want to move beyond that: publications such as this, Modern Recording, Polyphony, Radio-Electronics, the various GPI publications, Synapse and all the others are an excellent means of continuing your education.

I know many people would like to make their living working with music, and since I've managed to do it, I hope you find the following advice helpful.

1. Decide what you want to do. It can be a list that includes 91 different things, but evolve a plan.

2. Educate yourself. Read, learn, use real equipment in real situations. If you're serious about being a recording engineer, you'd better at least have a 4 track studio

DECLASSIFIEDS

[Any individual or company may advertise in **DEVICE** Declassifieds. Rates at present are $2.50/25 words or less - name, address are free. Over 25 words: 75 cents/word, since we're trying to keep things brief. Ads recieved before the 1st of the month appear in the next month's issue. Editors reserve the right to reject ads deemed inappropriate to **DEVICE**, and cannot accept responsibility for claims made in this section.)

FOR SALE-"Organ Guitar" Immaculate MCI Guitorgan Model B-300. Black/Chrome. $950. Plush case and volume pedal. Call (305)886-4906 or write Charley Deitz, 1215 Ustler Road, Apopka, FL 32703.

FOR SALE-SWTPC Stereo preamp, 25 W X 2 power amp $175; SWTPC FET 6 in 1 out mixer with one ±5 - 15V variable, ±5, ±15V three-terminal regulators large LMB enclosure $55; PAIA phlanger (C.A. modifications) $70; small electronic drum kit $40. All assembled calibrated, manuels, ppd. Ron Minemier, P.O. Box 153, Ripon, Wis. 54971. (414)748-2793.

FOR SALE-Sequential Circuits programmer. In mint condition. Lists around $950; will sell for $600. Great for use with guitar/synthesizers. Call evenings - (512)836-3069.

WANTED: Do you have a working knowlege of synthesis techniques? Are you able to transfer this knowlege to others in an interesting and informative manner? If so we're anxious to talk to you! Contact us C/O DEVICE. ACCESS-the synthesizer school.

FOR SALE-Mutron Micro-V envelope follower, $26. E-H Zipper envelope follower, $32. Both A-1 condition and with original boxes and instructions. Postpaid. T. Henry, 249 Norton St., Mankato, MN 56001.

(cont. from pg. 14)

at home. This requires money; it is the capital required to start your "business". Work two jobs, figure out a supplemental gig, take a loan, whatever it takes as long as it's ethical -- but you'll need some bucks to gather together the tools you're going to be working with. Really want to get into design? Buy data books, parts, some breadboards, an old scope... and do it. Don't depend solely on academians to educate you in a totally satisfactory manner; colleges often run on different rules from the real world. I don't have a college diploma, but no one has ever turned me down for something because I didn't. If you have a unique talent to offer or if you can do somebody some good, they aren't going to complain because you don't have a piece of paper.

3. Start at the bottom if necessary,but start. Want to be a recording engineer? Offer to work part-time at ridiculously low wages as a set-up, clean-up, and "gopher" person at the studio. Go out and get the coffee, or the pizza; but when you're not, observe everything like a hawk. Learn how to align the machine, how to clean it, all that stuff. Learn about good mic placement by watching a pro. At some point you'll become a tape operator, and then perhaps collaborate on a mix or two when your talents are needed ...and then one day the engineer will get offered a job somewhere else, and you studio will need an engineer... and you 're on. If they like what you do, you'll do the next date. And if you've been observing, and educating yourself, they will like what you do.

Want to do design work? Start out as a product tester or quality conrol person. Find out what things go wrong, what doesn't work, what does. Move you way up, and what you learned on the way will come in handy more than once during your trip.

4. Perseverence is the key. You can't give up, even during the year when you make $2500 and your girlfriend's part time job allows both of you to eat. The fates have a strange way of working, but at the least expected moment someone will call up with a job, or someone will have heard about you from someone else --- you get the idea.

Finally, in many respects your teachers are right. You have to be educated (but you can do that yourself, and don't be afraid to ask questions of people who know more than you do); you have to have a good ear (but good ears are made as well as born, and nothing develops an ear faster than working in the studio!) and experience and connections don't hurt. But, you are the one who gathers experience by your actions, and those actions create connections.

Perhaps your teachers don't understand that faith in yourself, honesty in you dealings with people, and relying on yourself to as great a degree as possible can overcome virtually all obstacles placed in your path. Oh yes, and expect things to start really popping for you about 10 years after you start in earnest. If you can't wait that long, I'd advise looking to something more secure. -Craig ■

(cont. from pg.10)

the rewards are considerable. People who snub critics and think they have the answer find that times change, people change, tastes change ...and you have to stay in touch with these changes if you want to survive. Some of the retail giants of yesteryear are out of business, or floundering, because they refused to listen to criticism; as Bill Godbout said about a computer company that went bankrupt, "you can't make 8080 products in a Z-80 world". He's right. Then again, Bill is another one of those people who views every criticism as a potential gold mine of valuable information. He knows that only by addressing criticisms can a product be perfected.

So far, I've been stressing the ability of the person being criticized to accept criticism and learn from it... but the critic must also accept an equivalent amount of responsibility, or the process will not work. One responsibility is to be quick with praise as with criticism when someone is doing a commendable job. The other responsibility is to avoid any personal animosity, to be as understanding as possible when picking apart someone's creative brainchild, and to be sensitive to the fact that most people are not secure with themselves, and may interpret criticisms as personal attacks. The critic must understand that he or she should be providing a service, and to be as gently -- and truthful --and constructive -- as possible. That way, we'll all grow: Not just as musicians, but as compassionate human beings...and we could certainly use a few more of those on this planet. by Craig Anderton ■

(cont. from pg.13)

width limit a non-inverting op amp configuration for both high and low frequencies -- and how to optimize it for lowest noise. ■

THE BACK ISSUES

1:1 AMS 100 — Part 1
The Famous Arp Avatar Review
Prepared Guitar
Evaluating Effects

1:2 AMS 100 — Part 2
Godley & Crème Interview
Intersound I/P Review
Troubleshooting Effects

1:3 Steven St. Croix Interview
AMS 100 — Part 3
EH Micro-Synthesizer Review
Schecter "Tele" Pickups Review
Prelude to Pedalboards

1:4 ADA Harmony Synthesizer Review
Results of the Effects Opinion Poll
AMS 100 — Part 4
Case Histories
Notes from West Germany

1:5 Yamaha Analog Decay Review
Mods to the EH Electric Mistress
Cable Capacitance
Budget Pots
AMS 100 — Parts 5, 6, & 7

1:6 Double Coil Pick-Up Mods
Phasefilter Review
AMS 100 — Part 8
Mods to the EH Micro/Synth
On Location: Hong Kong

$1.50 per issue for more than one issue. Please add .40 postage and handling per issue.

DEVICE

PUBLISHER
Roger Clay
EDITOR
Craig Anderton
CIRCULATION
Deanna Holohan
PHOTOGRAPHY
Vesta Copestakes
Rick Reed
LAYOUT
Roger Clay
Terry Lenthecost
CONTRIBUTING EDITOR
Thomas Henry

SUBSCRIPTIONS

DEVICE is published monthly and distributed during the last two weeks of the month. SUBSCRIPTION RATES are $15.00 a year in the U.S., $15.00 in CANADA and MEXICO, $18.00 for all other foreign countries at international surface rates and $24.00 at international air rates.

NOTE

All circuits published in DEVICE are intended for the private, non-commercial use of our readers and are believed to be original and free from patent infringement. However, patent protection of which the authors are not aware may exist for some circuits published in these pages. The publisher does not accept responsibility for failing to identify such patent protection where it may exist.

Send correspondence, products for evaluation, press releases, manuscripts, etc., to: 12304 Scribe Dr., Austin, TX 78759.

U.S. POSTAL SERVICE TX 787 SEP 6 PM 1979

DEVICE
the newsletter for the electronic guitarist/musician
12304 Scribe Drive, Austin, TX 78759

send to:

020180

Mark Hammer
Dept. of Neurosciences
McMaster University
Hamilton, Ontario, L8S 4K1
CANADA

FIRST CLASS

PRINTED MATTER