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

MODIFYING THE EH ELECTRIC MISTRESS

by THOMAS HENRY

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

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

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

THE MODS

Here are the modifications we'll be adding:

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

(cont. p.2)

IN THIS ISSUE: Pg.

YAMAHA E1010 ANALOG DELAY

by PAUL RIVERA

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been hoped that new analog and/or digital technology could proviode a quiet and dependable electronic delay to replace outdated mechanical delays.

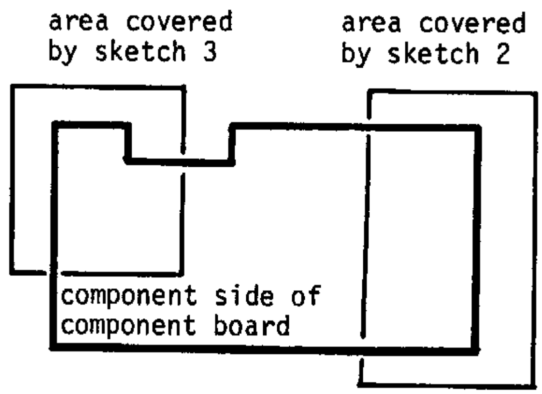
Currently, the echo unit used most by musicians is the Maestro Echoplex. Among other things, the Echoplex is mechanical, noisy, employs poor-quality heads, is prone to wow and flutter, and requires frequent maintenance as well as tape replacement to ensure acceptable sound. Although any one of these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

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

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

straight
and flanged signals
3. increase regeneration for deeper effects.
4. an on-off switch with a power status LED.

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



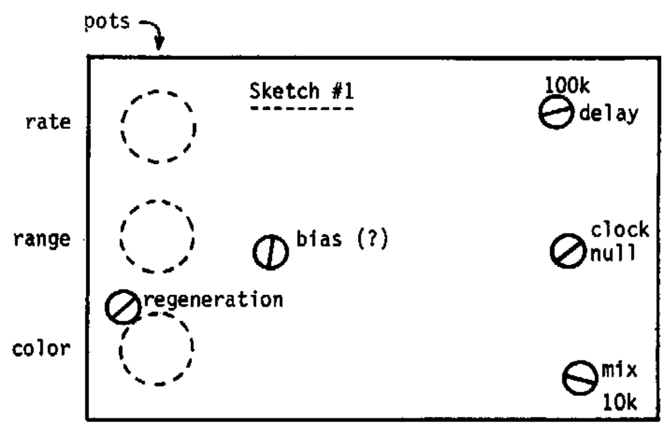
See sketch #1. This is a view of the circuit board from the component side, with the pots oriented to the left side of the board. Note that all the Trim pots are labeled, and we want to concentrate on these. First though, let's see how to get the board out of the chassis without ruining everything. Remember,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

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

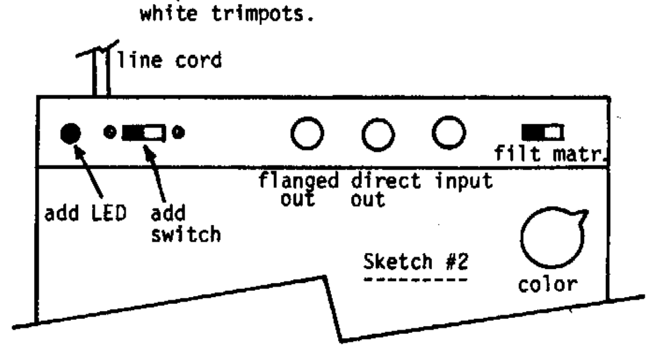
Now, for our first modification locate the delay trim pot. We would like this to be a front panel control. This works out quite nicely, for there is more than enough room in the chassis for several more pots. Simply desolder the trim pots, and run extension wires to a front panel pot. Use a 100K pot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

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

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



Component side of PC board. Note 5 white trimpots.



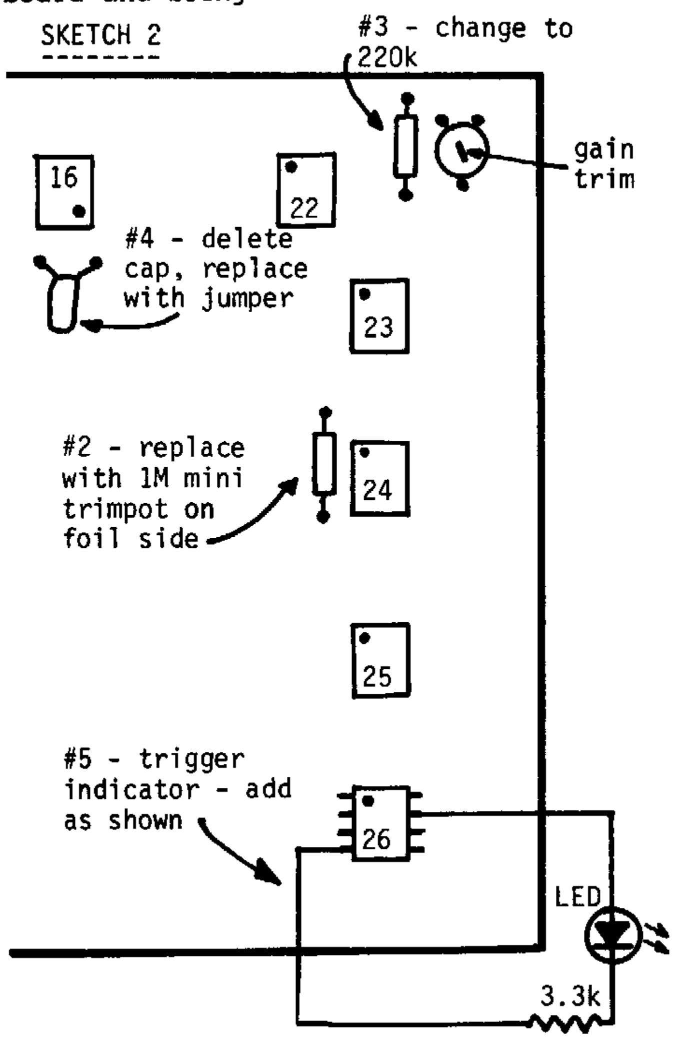
Front panel view of Deluxe Electric Mistress.

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

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

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

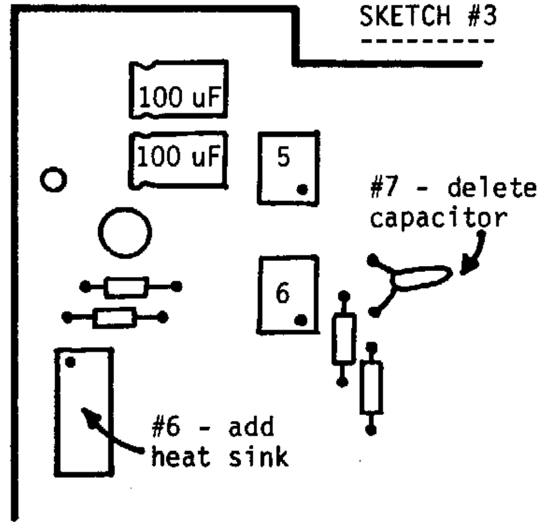
SKETCH 2



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

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

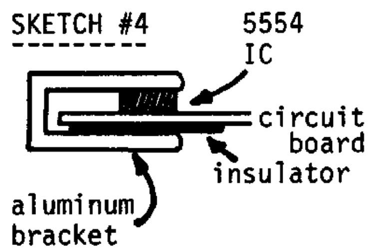
SKETCH #3



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

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

SKETCH #4



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

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

EDITORIAL

THE HALF-WAY POINT

At the half-way point of Volume #1, we'd like to give you an idea of what's happening here at DEVICE.

First of all, we'd like to thank the hundreds of you who are actively supporting DEVICE with your subscriptions, your article contributions, and your letters. This whole newsletter venture has been a gamble for all concerned, but throughout the whole process our strongest desire has been to provide you with information you've always wanted but couldn't find anywhere else. Your positive responses have been very satisfying (and, since we're neophytes at publishing, very encouraging); any negative responses have been carefully considered and evaluated, in the desire to create as responsive a publication as possible.

Now we would like to ask you to do something for us that we cannot do thoroughly enough by ourselves: Promote DEVICE. The staff, here at DEVICE, talk up the newsletter whenever possible, write press releases for other magazines to publish, and occasionally scrape up enough bucks to run an ad or two; but none of this is always effective for reaching musicians on a one-to-one basis. We need more subscribers in order to put us on firm financial footing, to expand our coverage to more subjects, and to allow us to keep putting out DEVICE regularly in the months and years ahead. Judging from your letters, you want DEVICE to continue too. So, encourage your fellow musicians to subscribe and learn about the new musical tools around us. Ask your local music store to consider carrying DEVICE in its magazine rack (for quanity rates on issues of DEVICE, have the store write us direct). If you know any musically-oriented manufacturers, why not tell them about DEVICE---we talk about a lot of subjects of interest to them, too.

The reason why we need you to do our promotion is that we are simply too small to have someone devoted full-time to "public relations" or marketing. You can fill that gap in our operation. Spread the word about DEVICE. We know there are a lot of people out there who would like to know about us, and we're hoping that you'll help us find every last one of them. Together we can make this thing not only happen, but happen right.

Thank you very much for your support in the past few months, and for any help you care to contribute in the future.

Craig Roger

INFO

Those experimenters who have come to appreciate the **SSM/Eu** series of ICs will be happy to learn that there is now another company specifically dedicated to the production of ICs for electronic music. The company is **Curtis Electromusic Specialities** (2900 Maurica Ave. Santa Clara, CA 95051), and they are currently offering the CEM3310, a voltage controlled envelope generator that offers significantly less control voltage feedthrough than the SSM equivalent. This IC is already designed into products by Oberheim and Moog Music, according to Doug Curtis. Additional ICs due to be available in the future include the CEM3320, a voltage controlled 4 pole filter similar to the SSM 2040 but which also includes voltage controlled resonance; and the CEM3330 , a dual voltage controlled amplifier with simultaneous linear/exponential control inputs and better than -80 db of control voltage feedthrough when trimmed. Curtis Electromusic has been around for a few years, but only recently have they started offering their ICs to the public at large; we hope their efforts meet with acceptance.

SSM has not just been sitting back either. According to an informed source, there will be a dedicated lowpass filter IC that is similar to the 2040, but lower in cost and less versatile, available later this year. This sounds like an ideal component for just plugging into a circut when you need a low pass filter with a minimum amount of effort. We've also heard of other improved designs that are on the drawing boards; look for an improved of their transient generator later in the year.

* * * * * * * *

Most reverb devices, whether mechanical or electronic, are less than satisfactory from a musical standpoint (noise, artificial sound, ect.) However Reticon has a new chip, the R5201, which looks very promising. It's a regular BBD

(cont. p. 11)

Q. WHEN IS A CABLE NOT A CABLE? A. WHEN IT'S A CAPACITOR! by Craig Anderton

The other day TEAC gave me a call to tell me about their new line of low capacitance cables. It's not uncommon to have someone from a company call me up and try to get me interested in something they're doing, but this representative was very enthusiastic about the new cords, claiming that the low capacitance made a tangible sonic difference. I was skeptical at first (after all, it seems that the dawn of every day brings some kind of new widget into existence to compete for the consumer's dollar), but I must say after playing around and doing some tests I'm convinced: the cord you chose is very important in giving you the best possilble sound.

WHAT'S THE PROBLEM WITH REGULAR CABLES, ANYWAY? In hi-fi circles, cable capacitance has been recognized for several years as a potential source of problems, but only recently has the subject cropped up among musicians. The problem is simple. Consider a capacitor; this is a component that is made up of two metal conductors, separated by an insulator. Then, consider a piece of shielded cable; this is also made up of two metal conductors, separated by an insulator, and therefore exhibits a certain amount of capacitance. As it happens, in the case of a cord one conductor is usually grounded, so the cord acts like a capacitor between your signal (carried on the hot conductor) and ground. A capacitance to ground such as this presents a low impedance to higher frequencies (e.g. it readily passes high frequencies), thereby passing these high frequencies to ground...and robbing some of the treble response. So, the less capacitance a cable has, the less the chances of losing high frequency response.

TESTING FOR CAPACITANCE. Since I don't have any capacitance meters, I needed to come up with a way to compare cables. I opted for the cable capacitance tester shown in figure 1 as a quick and simple way to calculate cable capacitance. The theory of operation is pretty simple; the 201 is set up as a unity gain amp. Putting a capacitor in parallel with the R2 decreases the high frequency, response, but instead of using a discrete part like a ceramic disc capacitor, we'll use the capacitance of the cable as the capacitor. Then, by observing how the frequency response changes when we add the cable capcitance into the circuit, we can calculate the approximate amount of capacitance. The way I observed this change in response was by feeding a constant amplitude, variable frequency sine wave signal into the tester, then hooking the tester output up to an AC VTVM (an oscilloscope will also do) and noting the changes on the meter.

After breadboarding the cricuit, I went ahead and built a prototype with a phone jack connector so I could just plug in a cable to test its capacitance. This is when I encountered my first surprise---the phone jack had about 9 pf of capacitance! So, I chucked the jack and put in two alligator clips attached to very short pieces of wire. Without having these clipped on to the cable under test, response of the tester was flat from 20 Hz to 20 Khz. Had I left the phone jack in, its capacitance would have influenced the results.

When testing cords, the output of the op amp must go to the ground conductor of the cord, with the hot conductor going to the (-) input. Otherwise, hum induced into the cricuit will give inaccurate readings.

CALCULATING CAPACITANCE. If you're only interested in results, skip ahead to the next section. But if you want to try measuring capacitance yourself, here's how I went about it. Remember I'm a musician, not an engineer, so there could be a chance I'm applying the math incorrectly... if this is the case, hopefully someone knowledgeable in the subject will correct me.

In our tester, the capacitance in parallel with R2 that reduces the high frequencies does so in a very predictable way. As a matter of fact, it's easy to calculate the high frequency response of the cord tester amp for different capacitances; the formula is:

(cont. p.12)

BASICS: TRIM POTS
by RON MINEMIER

Living in a rural area, my early electronic experience came about almost totally through the mail. Craig's first book and his kits from Godbout got me started. Gradually I've branched out and started building from scratch.

When I first started building I realized that the cost of pots was sometimes more than that of the actual electronics, especially if the unit was an ADSR or other synthesizer-type module.

My first wild stab at the problem was ordering a Poly-Paks grab bag of 20 "assorted" pots for $2. When I opened up the bag I found some pots with shafts, some without shafts, multi-ganged pots with strange resistances, and some PC mount types without threaded bushings. Only 1 pot came with a hex nut; that was a pot whose shaft ended just above the bushing and had a screwdriver-like slot for adjustment. My initial reaction was that I had blown $2. Only later, when my building funds were zilch, did I open the bag again and look for some salvageable pieces.

All you need to modify most pots is a very small screwdriver, hacksaw, pliers, and some super glue or epoxy cement. Carbon track pots generally have 5 pieces: the shaft(s), body, bushings, insulator-resistance element, and wiper. By prying up the tabs that hold the bushing plate to the body, you can change the bushings, resistance elements, replace or clean the wiper, and change shafts. To re-assemble put the "sandwich" back together, crimp down the tabs, and you're ready to go.

You can also pry ganged pots apart and use just one of them. Say you want just the back one. Pry the bodies apart. If they have concentric shafts, when you put the top pot's bushing on the back pot, the shaft will have alot of play in the bushing. Cut a section of the top pot's larger shaft and slide it into the bushing - now you have a decent feel for the new pot.

You'll still need hex nuts; Radio Shack sells bags of (cont. p.13)

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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,

Here are some questions regarding your AMS-100. In the 1st issue, the preamp section of the compressor has an input impedance of 1K. This seems to be contrary to your frequent stress for high input impedance. What is the reason behind such a design?

Second, the 570 has a THD trim pin which is left unused. What is the reason again?

Third, why is the compressor output jack coupled with 3 diodes in series instead of two in parallel with opposite polarities to cut spikes?

The last section of the pulse generator is really an ingenious design. I've been trying to solve response delay problems for some time until you pointed out such a new horizon to me.

I've some CA3080's on hand. Can I use them in a compressor circuit? I tried FET's which was a failure; the CA3080 seems an alternative if I run out of 570's.

You seem to be aiming at synthesized sounds; in the future, please give more technical details of the systems so that I can grasp the concepts better. I'm looking forward to learning more.

Lee Wai Hung
Hong Kong

Lee,

These are all interesting questions. In the first question, the input impedance of the buffer (not compressor) stage is actually about 500K. The 1K resistor terminates to ground through lug B of J1; do not forget J1 is a STEREO jack. The resistance to ground at the (-) input does not affect the impedance at at the (+) input directly, so there are no impedance problems.

QUESTION 2: The compressor in the AMS-100 input module is not a superior preformance compressor, rather, it is a "quick and dirty" type that improves the signal to make subsequent processing more easy. Since the THD of a 570 is really quite low, it seems unnecessary to add the THD trim since it didn't really make much of an audible difference. In something like a hi-fi VCA or noise reduction unit, of course, it would be an advantage to include the THD trim control.

QUESTION 3: The signal at pin 7 of the IC2A does not center around the ground, but rides about 1.8V above ground. Thus the real problem is positive spiking, not negative spiking (although there is a little bit of negative spiking, too). Therfore after capacitively coupling the output of this stage, the three diodes remove positive spikes greater than about 2.1V. It didn't seem necessary to take any precautions to remove negative spikes, since they were very low in amplitude.

Glad you like the trigger design; I'm sure there are better ways, but this has served me pretty well so far. However, I hope at some point to come up with an even more reliable triggering circuit, since so far that is the weak link in any guitar/synthesizer system and the AMS-100 is no exception.

About 3080's.... they are an alternative to FET's, but I personally don't like using them due to the large noise level and touchy input characteristics. However, I do like the way 3080's distort, and have used them in a voltage controlled distortion generator that will be featured in an upcoming issue of DEVICE.

More technical details? What do you think about this, readers? I am planning to give some block diagrams of possible systems after 5 or so modules have been presented, but if the information I'm giving on the AMS-100 is too sketchy drop us a line and we'll see what we can do.——Craig

Dear DEVICE,

Re Thomas Henry's article, "Case Histories": Lifting pads off a MusiKit board indicates that either the iron is too hot (or that the soldering operation is (cont.p.14)

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

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

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

Have fun!

Please note our new address:

DEVICE 12304 SCRIBE DRIVE, AUSTIN, TX 78759

BUILDING THE AMS-100

craig anderton parts 5,6&7

This month is devoted to a trio of control voltage (CV) processing modules that are extremely simple to build

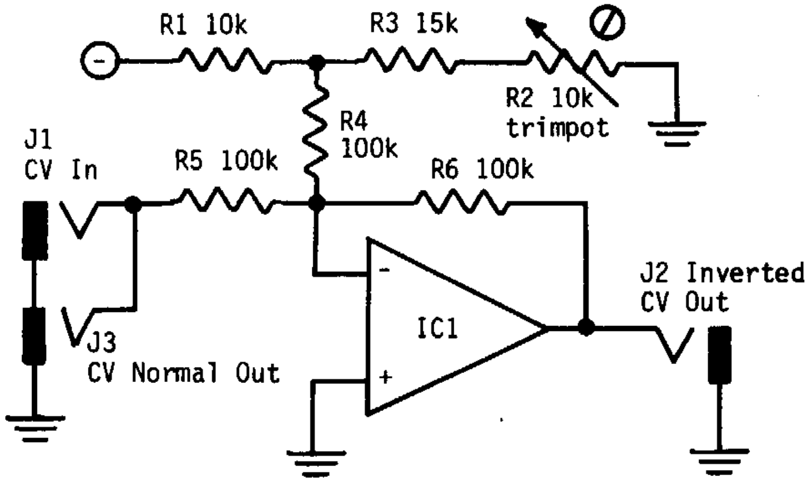
MODULE #5 is a simple control voltage inverter. IC1 can be any 741 type op amp; the purpose of this module is to invert any control voltage presented to it's input. A typical application would be to feed the unprocessed (normal) control voltage into one voltage controlled amplifier (VCA) with the inverterd control voltage going to another VCA. Thus, as one VCA becomes louder, the other becomes softer. This is useful for panning and crossfading effects.

MODULE #6 is a footpedal control voltage generator (this allows you to pedal control last month's phase shifter, for example). Choose a footpedal that uses the full throw of a pot, then mount a 10K linear taper pot in the footpedal in place of whatever is already there. A stereo cord connects the footpedal to the actual control circuitry inside the module. Referring to the module section, IC1 can again be any type of 741 family op amp. R1 should be trimmed so that maximum forward pedal throw results in a +10V control voltage output. Note that on this module, we've included 2 CV outputs so you can control 2 modules from the pedal output. You might wish to add some more output jacks if you have the panel space.

MODULE #7 is a simple lag processor that "slows down" a control voltage signal. For example, if a CV goes from 0 to +10 in 10 ms, after going through a lag processor it will take, say, 20 or 40 ms to go from 0 to +10V (depending on the setting of the lag processor control). The lag also occurs with decaying voltages; a decaying control voltage passing through the lag processor will take longer to decay. A typical application would be to feed the normal CV out into one phase shifter, and the CV lag out into another phase shifter. The frequency of the second phase shifter would then lag behind the first phase shifter by an amount determined by the lag control. Another application would be to feed the peak detector output from module #1 into the lag processor to smooth out any residual ripple left in the peak detector signal.

PACKAGING. Since these CV processors take up very little panel space, you could use an IC like a quad op amp and build a module that contains the inverter, footpedal control voltage generator, and lag processor, all packaged in one panel. Alternately, you could have one module with several inverters, another with several lag processors, and so on. It's hard to give a specific recommendation as to how many of these modules you'll need; for now, I'm building one of each. As time goes on, if I find out that one module (say, the lag processor) is particularly useful, then I'll build some more of them for the system.

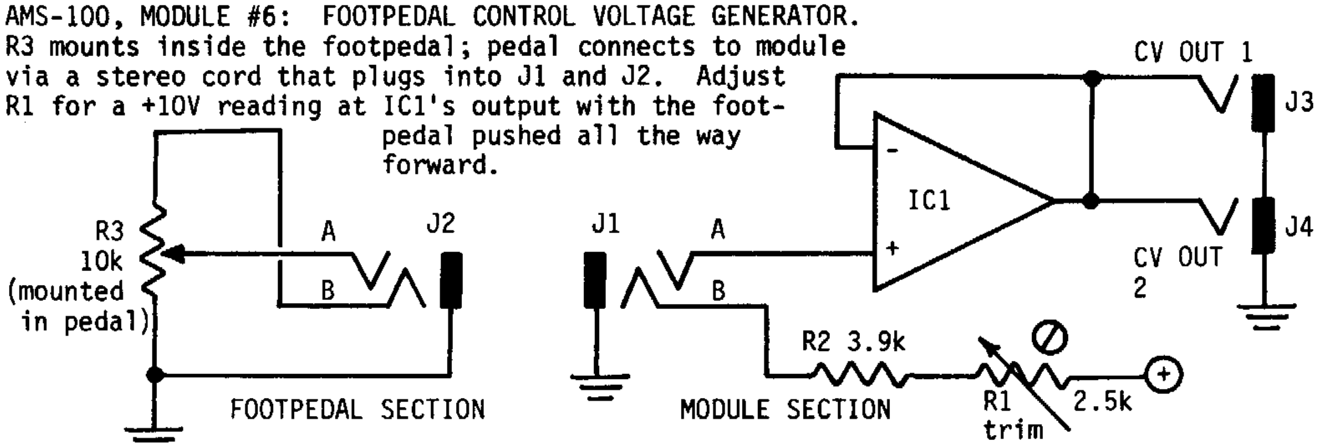
Next month, we'll get back to audio processing. I suspect that in the next few issues we'll have presented enough modules to create a minimal, but effective, processing system. At that point we'll discuss applications and patches before moving on to more modules. As always, questions and suggestions are welcome.



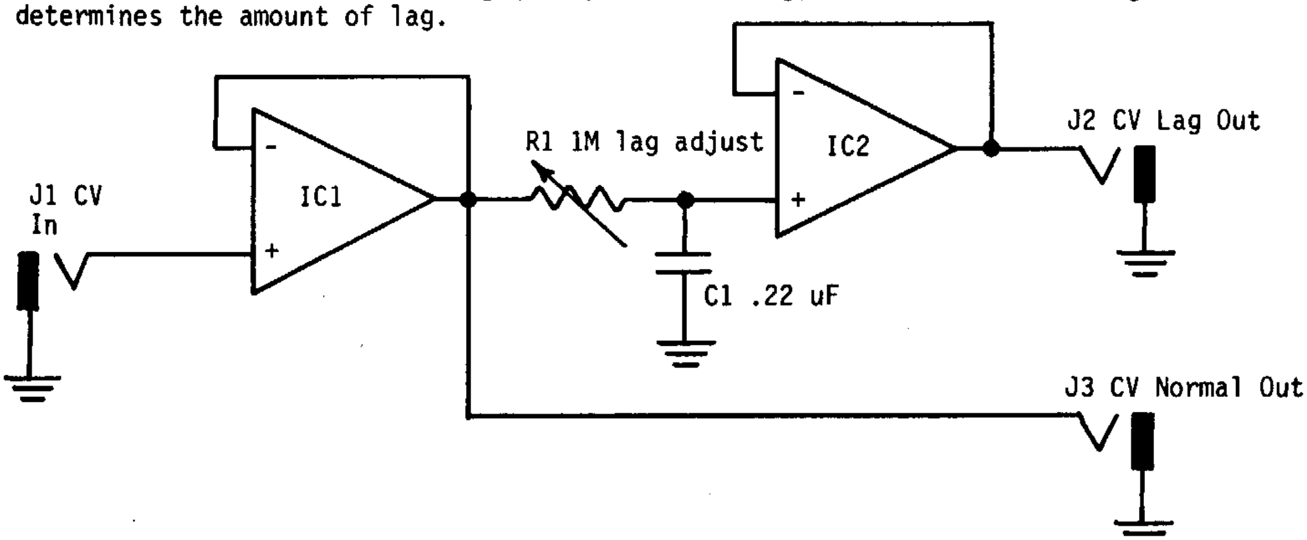
AMS-100, MODULE #5: CONTROL VOLTAGE INVERTER. This module inverts any control voltage presented to its input, i.e. +10V in gives 0V out, 0V in gives +10V out.

Adjust R2 by shorting J1 to ground and measuring the output of IC1. Adjust this trimpot for a +10V reading at IC1's output.

AMS-100, MODULE #6: FOOTPEDAL CONTROL VOLTAGE GENERATOR. R3 mounts inside the footpedal; pedal connects to module via a stereo cord that plugs into J1 and J2. Adjust R1 for a +10V reading at IC1's output with the footpedal pushed all the way forward.



AMS-100, MODULE #7: CONTROL VOLTAGE LAG PROCESSOR. This control voltage processor adds an adjustable amount of lag (low pass filtering) to the control voltage. R1 determines the amount of lag.



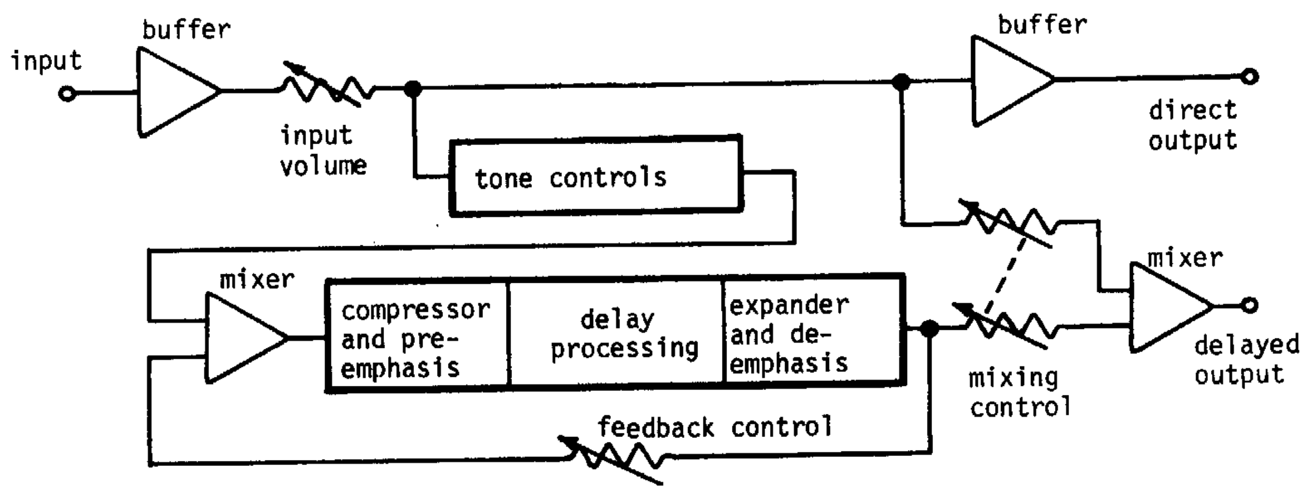
YAMAHA E1010 cont.

the point of dimishing returns: There is nothing inexpensive that remains to be done to mechanical delays that can rectify their problems.

In analog/digital, every day brings new news of the latest developments in IC design — developments that mean lower cost and better performance for future units. In my opinion, the first of the analog delay units with acceptable performance and reasonable cost is among us, and it is the Yamaha E1010.

The specifications of the E1010 are impressive, partially due to its excellent inter-stage gain structure that preserves headroom, and thus signal-to-noise ratio, throughout. Total harmonic distortion, less than 2%; IM (inter-modulation) distortion, less than 3%; hum and noise, -87dB with the input volume conrol set at maximum; frequency response, + or - 3dB from 30 Hz at 10 milliseconds (ms) + or - 3dB from 30Hz to 2kHz at 300 ms.

and have 12dB of cut or boost available. The tone controls feed a buffer amplifier that sends the signal into the first of many low pass filters (LPF). These LPF's all have a cut-off frequency of 13kHz, and a 24dB/octave slope. After going through the first LPF, the signal enters a pre-emphasis network, and then a compressor. This helps maintain fidelity, and makes it easier to keep the noise down. The compressor drives two different delay circuits. One of the circuits is a 512-stage bucket brigade delay coupled with a LPF. This is strictly for the 10ms output. Longer delay times are achieved from the other delay circuit, which consists of four cascaded 4096-stage BBD's, each with a LPF following it. Taps taken after each LPF, but before the next BBD, are fed to a switch, which determines the delay time range. For 10ms or less of delay, the signal is fed only through the first of the 4096-stage BBD's. From then on, corespondingly longer delayed time are



Simplified block diagram, **Yamaha E1010**

Equally impressive is the fact that the E1010 does a number of additonal things, all well. And, although it is not possible to change instantly from one effect to another, this versatility saves the E1010 from being a "dedicated" device.

The input of the E1010 feeds a buffer amplifier, then the input volume control, and then another bufer amplifier. At this point, the signal splits three ways: One send feeds yet another buffer amplifier that drives the nine-sction LED peak meter, one send goes to a "direct only" output jack on the back panel, and the final send drives the tone controls (bass and treble). The tone controls center at 70Hz and 7kHz,

achieved by utilizing more of the cascaded BBD's.

Each of the BBD's, including the 512-stage BBD, has two control ports; these control ports are fed from individual buffers driven by a voltage-controlled (vc) clock generator. This is how the delay time is varied within the selected range of delay, from short to long. The VC clock generator serves another function, as well, for the depth control, low-frequency oscillator and the LFO control are part of this circuit, too. These allow the E1010 to act as a flanger, chorus and vibrato unit as well as an echo.

The output of the final LPF in the

delay string is fed back into an expander. Actually, the compressor and expander are on one IC, an NE 570. From the expander, the signal is fed into the de-emphasis network, and again, into a LPF.

At this point, the signals splits again. One portion feeds back into the delay circuitry via a feed-back control; this determines the regeneration, or number of repeats. The other portion fo the signal with the buffered direct signal. The output of the mixing circuit is automatically compensated for noise through a LPF that is switched coincidentally with the delay range. The output of this network feeds another buffer amplifier, and this final buffer drives the outputs.

It's clear to me that great attention has been paid to the final sound. LPF's are used throughout the circuit to minimize HF noise, and buffer amplifiers abound to cut down on any mismatches between stages.

Setting up the E1010 is quite straightforward. For echo, one simply presses a button to select the delay range, fine-tunes the actual delay time with the delay control, regulates the number of repeats with the feedback control, and then mixes the direct and the delayed signal to taste.

Setting up other effects with the E1010 is equally easy. For those familiar with synthesizers, the frequency modulation section of the control panel will be immediately understood. For others, a quick look at the owner's manual and a few moments of experimentation should be sufficient to acquire at least a working knowledge of the controls and what they do.

Like other Yamaha professional products, the E1010 is rugged, well-built, reliable, well-packaged, easy to repair, and uses quality components. The only change that could be made would be the substitution of fiberglass PC board for the phenolic board.

Also like other Yamaha pro equipment, the E1010 is designed to be rack-mounted. Although rack-mounting is the standard for PA and studio gear, the E1010 is obviously aimed at a broader market. Apparently, Yamaha is expectng players to start accumulating rack-mounted effects. This would be a welcome move, and could lead to modular racks containing everything from pre-amps to remote effects switching units.

Even if you aren't planniing to acquire a rack full of equipment, the E1010 is well worth considering. It is quiet, and its flanging, chorus and vibrato (not tremolo, mind you) are equal to or better than most other seperates.

How does the E1010 stack up against an Echoplex? The only catagories in which the Echoplex surpasses the E1010 are sound-on-sound and length of delay at maximum setting (the E1010 can simulate even the wow and flutter present in the Echoplex!), and for those who need either feature, there is still no substitute. Others, however, who wish to kiss goodbye to tape changing, head cleaning, fiddling, and cursing should check out the E1010. Electronic delay is here to stay.

-Paul Rivera

INFO cont.

but has a number of taps that are not evenly spaced. Therfore, adding feedback causes regeneration that closly simulates room reverberation. As soon as a chip is obtained for evaluation, we'll write up the results in DEVICE. For more information on the R5201, write **Reticon** at 910 Benicia Ave., Sunnyvale, CA 94056.

* * * * * * * *

National Semiconductor (2900 Semiconductor Dr., Santa Clara, CA 95051) has announced the LM13600, an IC that contains two current controlled transconductance amps (each with differential inputs and a push pull output). It is claimed to significantly reduce distortion while allowing higher input levels, thereby giving a superior signal-to-noise ratio. This looks like a good part for replacing the 3080 in applications where low noise is important.

* * * * * * * *

Sounder Electronics (21 Madrona St., Mill Valley, CA 94941) has announced a phase checker, designed to catch phase errors in studios, PAs, hi-fi stores, and the like. The phase checker consists of two units; a pulse generator and a phase detector, which when used together check out the phase integrity of a given audio system. Phase problems are quite common, even in professional audio situations, since non-standard speaker wiring and miss wired connectors are certainly not uncommon; this looks like one way around the problem. We're trying to get our hands on one for evaluation, at which point we hope to do a full review.

CABLES cont.

$$F = \frac{1}{6.28 \times \text{capacitance in uF}}$$

To plug in some numbers and see how this formula works in practice, if the capacitance is 100 pf (or .0001 uF) and we put this capacitor in parallel with R2, then F works out to:

$$F = \frac{1}{6.28 \times .0001} = 1592 \text{ Hz}$$

By definition, this frequency is the frequency where the response of the amp will be down by 3 dB. We can mutate the formula to derive the capacitance if we know at what frequency the response is down by exactly 3dB. For example, if we read on the VTVM that response of the tester is down by 3 dB at 1592 Hz, we can use the following formula to measure the capacitance:

$$C = \frac{1}{6.28 \times \text{frequency in Hz}} = \frac{1}{6.28 \times 1592}$$

(cont. from above) **.0001 uF = 100 pF**

Another example: if we read that response is down by 3dB at 2000 Hz, then using the same formula we know that the capacitance is .000079 uF or 79 pF. Actually, these formulas are somewhat more complex than I've indicated; the simplified versions shown here only hold if R2 is equal to 1 Meg.

Well, I hope you found that interesting...if you didn't, then without further ado let's get into the ramifications of all this.

RESULTS. Table 1 shows the length and type of cord, -3 dB frequency according to my test equipment, calculated total capacitance, and calculated capacitance per foot. What does it all mean? Well, all of these cables have quite a lot of capacitance, certainly enough to seriously diminish high frequency response if you're feeding a very high input impedance from a relatively high output impedance. The results suggest several possible conclusions:

1. Always use the shortest possible cable length in high impedance output systems. This has traditionally been considered good practice, but it is much more important than I had realized in light of these capacitance measurements.

2. TEAC's low capacitance cord did, in fact, test considerablyl lower in capacitance than the various non-low capacitance cords, so there is a difference between different cords.

3. When connecting your guitar to a high impedance buffer stage, use the highest quality piece of cord you can lay your hands on...and keep it short.

4. Everything seems to have a certain amount if capacitance. For example, I tried testing various connectors. A standard 1/4" phone jack had anywhere from 9 to 11 pF of capacitance between the hot and ground elements. The least capacitive jacks were the so-called "high density" jacks that are encased in black plastic (although the contacts are not as hefty as some other types). The capacitance for one of these (stereo version) measured only 7 pF from hot conductor to ground.

5. A lot remains to be done in this field; there must be a zillion different cords out there, just waiting to be checked out for a minimum capacitance. But, let's not be too hasty in jumping to conclusions. Using thinner wire in an attempt to cut down on capacitance may also make the cord less reliable, and increasing the space between the ground and the hot conductors to reduce capacitance could also decrease the shielding effectiveness...you get the idea. Everything seems to involve a tradeoff, and although searching for low capacitance is probably desirable, there are other requirements that go into making up the "ideal" cable that must also be taken into account.

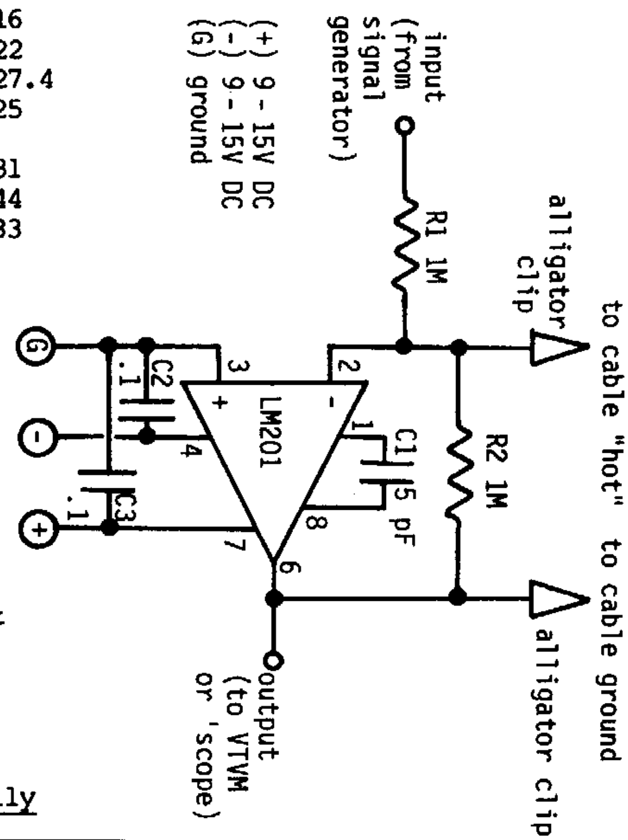
6. Throw away your coil cords! (If you haven't already.)

I don't pretend to have answered all the questions; this excercise has been dedicated towards increasing the awareness of yet another part of the audio system. Cords tend to get ignored, because they don't consume power or make noise.[I don't know about that Craig, have you ever heard the snap, crackle and pop from one of those plastic insulated cables? I sure have!--Roger] But, they obviously have quite a bit of influence on your overall sound in terms of frequency response. If your're feeding a bunch of stages with 1 Meg input impedances from stages with 20 t0 50K output impedances (or more) with coil cords, then your're just not going to get a

TABLE 1.

LENGTH	CODE	-3dB POINT (Hz)	TOTAL pF	pF/ft.
5"	A	1980	80	16
6"	B	1199	132	22
5"	C	1155	137	27.4
12"	D	520	306	25
6"	E	850	187	31
7"	F	512	310	44
8"	G	241	660	83

Figure 1



Code Explanation

Cords with RCA phono plugs at both ends:

A = TEAC low capacitance cord
B = Radio Shack standard hi-fi cord
C = Pioneer cord included with cassette deck
E = TEAC low capacitance cord

Cords with 1/4" phone plugs at both ends:

E = Surplus plastic cord with plugs molded on at each end (a true el cheapo)
F = Home made cord with Belden 9271 Twinax and "military style" plugs
G = Coil cord. 8 foot length refers to the cord extended to its maximum usable length, not the total length of the cord if it was totally uncoiled.

nice, crispy sound.

Still, other questions remain...do some connectors have more capacitance than others? This could very easily influence the capacitance figures I came up with. It seems that RCA plugs, for example, have less capacitance than 1/4" phone plugs. And what about comparing the pF/foot readings of cords E and F? It's very possible that those sturdy, military strength plugs have a lot more capacitance than the plastic molded kind...but again, what about the strength and reliability tradeoff?

I realize that this article has been long on theory, and short on practical results (i.e. which cable is "best", which connector is "best", and so on). But I thought this whole subject was indeed of significance (the TEAC people were right!) and that DEVICE readers would rather get the scoop now and do some experimenting, than wait around a few months while I collected cords and plugs and did some exhaustive testing. So, the discussion here is only a beginning...as more data comes in on the subject of cable capacitance, the discussion will continue from where we left off. -Craig Anderton

POTS cont. these and though the machining of the nuts isn't very good, the ones that don't fit can be used as spacers.

For those pots with short shafts you can either change shafts entirely or simply glue a piece of shaft from the excess of another.

There are ways to use PC mount pots without modifications, but this involves integrating the pots with the actual circut board or making a pot board with a few threaded pots mixed in to secure the sub-board to your panel. Both of these methods are good but may limit your panel layout and will take extra time to construct.

With a little mechanical manipulation and just a couple minutes, my initial $2 investment has yielded 8 useful pots. I've actually made $6 on the deal plus I still

(cont. p.14)

DIALOGUE cont.

taking too long) as those traces should not lift off under normal soldering conditions. Using an iron with a 600 degree tip should solve the problem.

Bill Godbout
Oakland, CA

Dear DEVICE:

I was surprised and delighted to read of your newsletter (I've got my request in for a sample issue, in fact). The particular chosen name was delightful and rang a few bells, because at one time, I was a member of an ensemble called DEVICE. Since any moderately advanced culture knows the magic of names, I can say that things look good for you!

Anyhow, we had an old logo we never got around to using that I've always thought was quite handsome. My question is: could you use it? Here's the nice part: it is free, if you wish to use it, go ahead. About the greatest compensation I can think of is just having it put to good use. If you are a standard white liberal like me (born to guilt and obligation), just autograph an issue and call it done. Good luck.

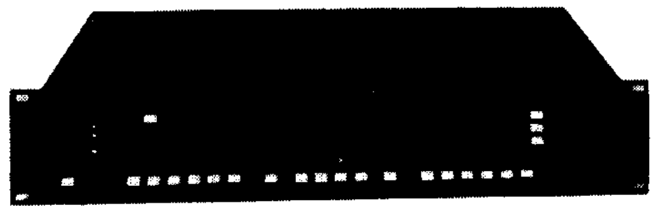
Greg Taylor
Salt Lake City, Utah

Dear Greg,

We don't know about the guilt and the obligation, but how would a tee-shirt with the logo sound? Anyone interested?

Ed.

Here's Greg's logo design.



Here is a picture of the new EVENTIDE H949 HARMONIZER. As well as pitch change and delay, the H949 offers flanging, repeat, random delay, and an entirely new effect - REVERSE. The HARMONIZER has two different algorithms to handle pitch change 'glitches'. As soon as we get out hands on one, you'll see a review in DEVICE.

POTS cont.

have many units left and I haven't had to change any of my construction techniques.

Don't overlook these "assorted" grab bag specials - they can be a virtual gold mine with just a small effort on your part.

WHAT, ME WRITE?

Yes, you! DEVICE wants information from its readers. A lot of you people out there have tricks, ideas, information, and advice that other readers want to know about...but they'll only find out about it if you send it in.

Don't worry about your writing, or style, or any of that stuff; that's what editors are for. If your ideas are good, we can make them readable without much difficulty. With some authors who have something special to say, we'll even accept cassettes as long as they don't ramble on too much.

By now, you're probably asking what's in it for you. Well, we can't pay princely sums of money just yet, but we will extend your subscription for 1 year for each article published in DEVICE that runs more than 1 page. You also get a free classified for every printed article. With this comes the additional confidence of being a published author, the appreciation you'll get from other readers who are into what you're talking about, and maybe even the chance to make some new contacts and friends, or hear about improvements to your ideas. All in all, you have nothing to lose and a lot to gain by sharing your knowledge with like-minded people...so, write for DEVICE.

OPINION POLL:
GUITAR/SYNTHESIZERS

This time out our opinion poll will require some homework from some of you. We realize that everyone hasn't had the opportunity to try a guitar/synthesizer system but we need the input from you for some future articles. Sooo..bop on down to your local music dealer and see what you can get your hands on then fill out the poll. We'll let you know the results in a couple of months. If you can't locate one, try to formulate an opinion with your present knowlege. We're waiting to hear from you!

1. Do you own a guitar/synthesizer?

2. If not, are you interested in acquiring such a system?

3. Have you experimented or used a guitar/synthesizer? If yes, what was the system and what was your opinion of its performance?

4. How important is foot control of synthesizer parameters? How important is programming or microprocesser control?

5. Specifically what features of guitar/synthesizer systems do you like? What features do you feel miss the mark?

6. Understanding the costs of development, marketing and the need for a profit incentive, what price do you feel would be fair for the system you would want?

7. What is your opinion of the idea and implementation of the hexaphonic pick-up systems?

8. What kind of developments would you like to see in the guitar/synthesizer market?

9. Understanding the costs at present, how important is the development of polyphonic systems to you? What features do you want to see on such a system? What price range do you feel would be fair?

10. What is you opinion of the following systems of guitar/synthesizer interfacing as you understand them:

good	bad	no opin.	
			a. self-contained hex pick-up units (mono)
			b. slavedriver hex pick-up units (mono)
			c. slavedriver fret-wired systems (mono)
			d. pitch-to-voltage converter using guitar pick-up output (mono)
			e. systems requiring the purchasing of a special guitar
			f. systems requiring modifications to your guitar
			g. polyphonic systems

WHAT IS DEVICE?

DEVICE is a subscriber supported monthly newsletter dedicated to those interested in the latest developments in signal processing and guitar-related electronics. **DEVICE** offers maximum editorial freedom and indepth views of the current status of our field. **DEVICE** plugs you into a network of information and communication that puts you closely in touch with the rapidly changing electronic music industry. Every month **DEVICE** brings you a variety of features including news, accurate reviews, construction projects, modifications, interviews and much more. **DEVICE** needs your support to make it work. Subscriptions are one way, the others are your submitting ideas, articles and opinions. Then **DEVICE** becomes the forum we all desire. What is **DEVICE**? **DEVICE** is your publication!

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