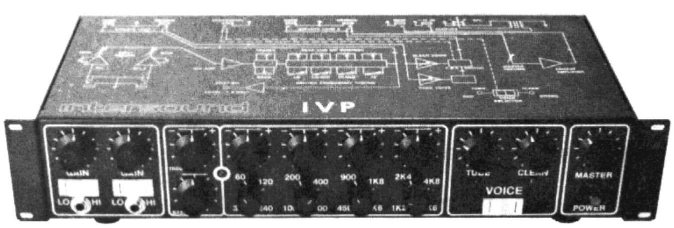
DEVICE the newsletter for the electronic guitarist/musician VOL.1:2-79

REVIEW:

INTERSOUND IVP

by Craig Anderton

People often write me asking what type of amp I recommend. For the past few years, I have been suggesting that they purchase a good quality, high power hi-fi amp like a McIntosh or Crown, build or buy an appropriate speaker cabinet, and finally, make a preamp section by combining several of my projects together (preamp, compressor, tone control, or what ever you like). Probably the most useful way to package these building blocks (except for the speakers)



(*Intersound Instrument Voicing Preamp*)

is the industry standard rack mounting system. There are many advantages to using standard size modules, such as easy serviceability, compatibility with studio-quality effects, and easy upgrading of the system. For example, if you want to install a new flanger, you simply remove the old flanger panel and put in the new one---leaving the other modules intact.

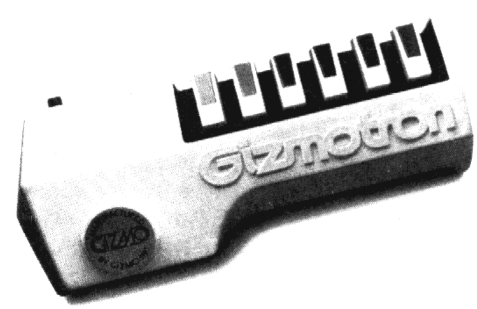
Apparently, the people at Intersound also believe this is the way to go, and their contribution to the system is a rack mounted preamp designed to drive power amps and recording consoles. Actually, calling it a "preamp" may be a bit misleading; something like "control (*cont. on page 2*)

IN THIS ISSUE:

Interview: Godley & Creme's "Gizmo"

by Roger Clay

By now, many of you have heard of the Gizmo, a unique device for guitar invented by Kevin Godley and Lol Creme (formerly of 10CC). Word has been circulating for the last few years that the thing was going to be marketed, and that it was a totally new way to manipulate the guitar...but unless you've been to a NAMM convention, saw 10CC before Lol and Kevin left, or listened to the Consequences album you probably haven't seen what the Gizmo does or how it works. What it does is allow you to "bow" the strings of your guitar by pushing color coded buttons mounted on the case, which in turn is positioned over the strings. These buttons engage rubber-like tiny wheels that rotate against the string itself. The wheels are spun by a motor in the unit (that runs off AC power), and the whole device attaches to your guitar on a specially designed bridge. For something that is so mechanically involved, they have come up with what is, in my opinion, a very neat and uncumbersome package. One of the things that excites me about this device (besides the array of interesting sounds it *(cont. on page 12)*



center" would probably be a better description. It lists for $360 (as of 1/1/79) and often shows up in stores for around $300.

FUNCTIONAL DESCRIPTION. For an understanding of exactly what's included, refer to the block diagram (fig. 1). There are two inputs, each with an associated bi-fet preamp and gain control. Each preamp also has a HI/LO gain switch, which means you can feed the thing signals as low as 50 mV RMS, or as high as 8 V RMS, without clipping the preamp stage. Thus, low level instruments such as guitar, or high level instruments such as synthesizers, can be brought to the same nominal operating level. The input jacks are closed circuit types that ground the input when no plug is inserted, minimizing any noise pickup by open inputs. The combined output from the two input stages goes to a conventional, bass/treble, boost/cut type shelving equalizer (as found on most home stereo units), and then to a 4 band pseudo-parametric EQ system. The output of this section feeds both a level-setting LED indicator and an effects loop, called effects loop #1, which allows you to patch effects in after the equalization but before subsequent voicing circuits. The effects loop jacks are closed-circuit types, so if nothing is plugged in to the loop the signal simply continues down stream towards the voicing section.

The Voice section offers two timbral possibilities, clean (with associated level control) and Tube Voice, which adds distortion. The Tube Voice has one control, which determines the timbre of this voice but only vaguely affects the output level. A panel switch chooses between two voice options, there is also a footswitch option that overrides the front panel control.

The output of the voice section goes through another effects loop (effects loop #2), which returns into the system via a master volume control (not to be confused with the type of "master volume controls" that add distortion). Finally, an output amplifier drives three different outputs: a main, line level ¼" phone jack out, a -10 dB ¼" phone jack out, and a balanced line -10 dB XLR connector out (eliminates the direct box in a studio situation).

Intersound has clearly opted for maximum flexibility within the constraints of competitive pricing. The emphasis on being able to handle widely varying types of inputs, in conjunction with being able to drive many types of outputs, not only attests to the chaos that exists in this industry with regards to standardization of impedances and signal levels, but also shows that Intersound would like the IVP to handle virtually all situations...and it does come very close to meeting that goal. However, as in any piece of equipment, compromises have been made; and naturally, I feel that there are areas which could be improved.

MECHANICAL EVALUATION. This is one of the strongest points in favor of the IVP. The packaging is durable, #16 gauge cold-rolled steel that should hold up well. The jacks are Switchcraft metal collar types, which I feel are much better than the plastic encased "high-density" types favored by much of the industry. The circuit boards are epoxy glass and are well supported mechanically. There is extensive use of shielded wiring, including those wires going to the effects loop jacks. The transformer is enclosed to reduce hum problems, and is mounted as far away from the preamp stage as possible; all AC wires are either supported or tied down, and run as twisted pairs to help cancel hum. The power supply is fused and the line cord is a durable 3 conductor type. My one reservation was an AC bypass cap that's epoxied to the on-off switch, however, I have been assured by Intersound that this method of mounting is not only permanent, but can also absorb mechanical shocks with no problem.

As far as quality of parts are concerned, all pots are CTS (American-made types). While it would have been nice to have something like Allen-Bradleys, there are limits to what you can expect for $360. These pots are basically the least-expensive-yet-still-good pots that are commonly available. There are 10 ICs used on the main circuit board, two LF356 bi-fets from National Semiconductor and eight 4558s from Texas Instruments. A smaller board including two ICs, two power transistors, and two small signal transistors houses the power supply and some other circuitry. The quality of parts is better than, or equal to, what you'd expect for this price; for example, mylar and polystyrene capacitors are used in the EQ circuits instead of disc ceramic types.

With respect to serviceability, to get at the main circuit board you remove the 15 knobs and pot bushing hardware. However, no sockets are included in the unit...so if one IC should fail, you are committed to removing the whole board. I really think it's about

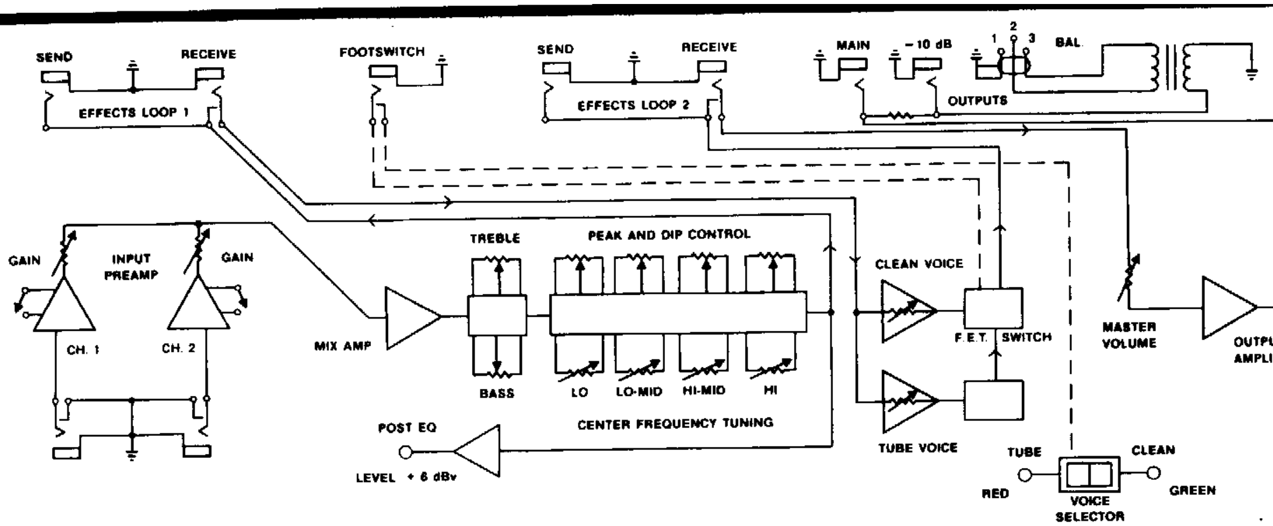


FIGURE 1

time manufacturers started including sockets for all ICs, for several reasons: first, no heat damage during assembly; second, servicing a defective IC takes seconds instead of several minutes; and third, as new and improved parts come out, older versions can be replaced. For example, Signetics now makes a 4558 pinout IC with superior specs compared to the average 4558. Had sockets been installed in the IVP, the user could take advantage of these technological advances much more easily.

As far as "human engineering" is concerned, the IVP scores well. The front panel graphics are not exactly artistic, but they are crystal-clear and easy to follow. Lettering is yellow on black, which shows up well under stage conditions. The controls are logically laid out and follow the signal flow from left to right. As with some other products, the block diagram is screened on the top of the case so you can see what's going on with the signal flow if you're not familiar with the unit. LED indicators help with level setting and show whether the tube or clean voice is selected.

CIRCUIT EVALUATION. The preamp input impedance measured about 45k Ohms, which is fine for most electronic instruments but will produce a "dulling" effect on the high frequency response of the average magnetic guitar pickup. This problem is most noticeable with series connections of pickups, where the output impedance of the guitar is around 10k Ohms; with single pickup connections, the dulling effect isn't really noticeable, and if it is, can be compensated for with a tiny bit of treble boost in the shelving EQ section.

For maximum voltage transfer from a magnetic pickup, for sonic reasons, and to best exploit the maximum potential of a bi-fet type op amp, it would have been nice to see at least one of the inputs have about a 500k inputimpedance. However, high input impedances can lead to greater problems with RF and hum pickup (as well as noise), and also demand the use of top-quality connecting cables. So, it could be argued that the Intersound people made a valid compromise by choosing a relatively low input impedance.

THE EQUALIZATION SECTION. The tone control options presented by the IVP are one of the strongest points in its favor; for those musicians who have been frustrated by having only bass and treble controls to play with, the IVP equalization should make them very happy. Since this section is such an important part of the IVP, I'd like to discuss it in detail.

The shelving bass/treble controls are pretty standard, so there's no use in really dissecting them here. The pseudo-parametric is something else altogether. There are 4 overlapping bands (#1: 30 to 240 Hz, #2: 100 to 800 Hz, #3: 450 to 3600 Hz, and #4: 1.2 KHz to 9.6 KHz), each with a boost/cut control. The first test run on this section was conformance to panel markings, i.e. when it said 1.2 KHz, was it really 1.2 KHz where the action was occurring. This conformance was judged as excellent---in band #1, the markings were right on except for 120 Hz, which measured out as 100 Hz. Band #2 was again extremely close to the markings save the 800 Hz position, which read as 895 Hz---not bad. Band #3 correlated within 10% except for the 1.8 KHz mark, which I read as 2.3 KHz. Band #4 displayed the widest discrepency of them all; 1.2 KHz was accurate, but 2.4 KHz measured out as 2.9 KHz, 4.8 KHz measured as 5.8 KHz, and 9.6 KHz measured as 10.2 KHz. Taking measurement error into account, the overall results are quite impressive...the favorable results are no doubt due to the use of good capacitors, as mentioned previously.

Next test involved consistency of the boost/cut action. In general, the boosting was less pronounced in the lower frequencies of the band; however, this was judged not too important since the lower part of a band usually overlapped with the upper portion of the previous band. For example, in band #3 maximum boost gave +15 dB at 450 Hz, and +19 dB for the other frequency markings. Similarly, the cutting action was also slightly less effective at the lower end; -17 dB for 450 Hz as opposed to -18 or -19 dB for the other control settings. Again, this is mostly of academic interest. The most drastic deviation was in band #4, with only +9 dB of boost at 1.2 KHz and +16 to +18 dB for the other calibrated points. I suppose they could have made the bands more narrow to make the specs look better, but I'd rather have the extra range and not worry about losing a few dB at the extreme low end of the control.

Another favorable aspect of the pseudo-parametric is that I feel Intersound chose the "right" frequencies for their bands. This is a very subjective type of evaluation, but the bands always seemed to fall in the right place for whatever instrument I was using. If I needed a bit of midrange notch, for example, and a little high frequency boost, I never had an occasion where I needed the boost to fall in the same band as the place where I needed the notch. I'm sure there are situations where the controls would not fall in the right range, but I didn't run into them.

Now it's time for my one major complaint about the EQ section: the bandwidth of each band is too narrow for my tastes, and is not adjustable. I feel this gives a "sharpness" and artificialness that a more gentle bandwidth would overcome. However, it could be argued that musicians prefer dramatic tonal changes, and I suppose there is an element of truth in that. Frankly, though, the more one uses EQ devices, the more the novelty wears off and subtle sounds become more important. In some ways this is an example of ear training; when people first use EQ, they tend to go for very obvious types of sounds. After a while these tend to sound over-exaggerated, and then it becomes time for subtler control settings. I prefer low-resonance, low-Q filter effects when equalizing instruments; it's nice to have the ability to do sharp bandwidth stuff, but I find a shallower bandwidth to be more useful and the IVP cannot provide this option. I believe an ideal solution would be to include push-on/push-off type pots for the boost/cut controls, which could change the resonance (damping) of the filter sections. This would approach the flexibility of a true parametric, which typically has a continuously variable bandwidth control for each filter stage. I'm sure this would add to the cost, but in my opinion would be well worth it.

Another comment concerns not just the IVP but EQ units in general. These days, there is a certain amount of specsmanship going on---engineers demand that EQs have the ability to boost or cut massive amounts, like +15 or even +20 dB. The IVP fulfills these specs, but it bears repeating that +15 dB of boost is a LOT of boost! Therefore, those people who turn up knobs full blast are going to get some really ugly sounds out of this equalizer. The more flexibility an equalizer has for making good sounds, the greater the chance of producing terrible sounds in the hands of an inexperienced operator. Most of my favorite EQ settings involved boosting no more than a couple of dB, which worked out to only a tiny bit of pot motion. So, if you check this thing out in a music store, be very sparing with the boost and cut controls...a little goes a long way. Also, while most people think in terms of boosting, cutting is just as vital. A bit of lower midrange cutting, coupled with high frequency boosting, made a very full-sounding guitar by eliminating some of the mid-range "honking" associated with my straight guitar sound.

Finally, I would like to see bypass switches for both the shelving EQ and pseudo-parametric EQ sections. I feel in/out switches are mandatory for equalizers, and I hope all other manufacturers start agreeing with me soon. I seldom had to use the shelving controls since the pseudo-parametric does so much, and it would have been nice to cut it out of circuit and eliminate just one more stage of noise when not needed. Similarly, I wanted an in/out switch for the parametric to compare equalized and non-equalized settings, which there is presently no way to do.

Do not infer from the previous comments that I'm "down" on the IVP equalizer capabilities; but adding two in/out switches and a choice of bandwidth would be ideal, and perhaps these features can be included in subsequent models.

THE TUBE VOICE CIRCUIT. The Tube Voice circuit has some strengths and some weaknesses. When I first tried it, I flat out didn't like it; however, this was because I was not familiar with the effect. It took me a while to get the thing "tamed" to the point where I understood what to use it for.

I found that for getting low-level amounts of distortion, the tube voice really shines; in other words, if you want a tube amp kind of "ambience" the thing is ideal. It is quieter than a tube amp, and gives a reasonable facsimile of the sound. However, if you want a long, sustaining rock and roll fuzz effect, you're better off putting a conventional fuzz in the effects loop.

When you put a sine wave through the tube voice circuit and increase the tube voice control, it becomes squarer and squarer (while the corners remain rounded) until past a certain point, you get a square wave with ringing. However, this control only really varies the output level for low levels of distortion. Past a certain point, the output remains the same and the intensity of the fuzz increases.

Probably the best use for the Tube Voice is in the studio. The musician can now get an "amplifier" sound, along with a direct-into-the-board-sound, without having to lug an amplifier around.

Because the Tube Voice does not have an output level control, the best procedure for getting a balance between the tube and clean voices is to set the preamp gain and EQ first (while observing the level-setting LED), then the Tube Voice control for the desired tube timbre, and finally the clean control should be balanced against the level of the tube voice. Adjust the master volume for the desired overall output level.

USING THE EFFECTS LOOPS. Unfortunately, effects are nowhere near standardized in terms of levels and impedances, which can lead to some interfacing problems. Effects loop #1 is easy to match with all types of devices---the preamp gain can be turned down to accomodate low level input effects inserted in the loop, and turned up to drive line level stuff. Effects loop #2 is a different matter. When the Tube Voice is adjusted for an optimum tube sound, chances are there will be too much level coming out of the effects #2 send for many effects that are commercially available. For these occasions, you will need to add an attenuation pot between the effect #2 send and the effect itself. Perhaps subsequent models of the IVP could include a trimpot on the back panel to control the level coming out of the effects loop #2 send.

An additional consideration involves the placement of Effects Loop #1. Under some conditions a better place for this loop would be between the preamps and EQ section, so that effects in the loop could then be equalized. However, there are equally convincing arguments for placing the loop after the EQ section (as the IVP is currently arranged) so that signals going to effects boxes can be equalized; for example, boosting the midrange a bit before going into a phase shifter yields more dramatic results, and removing some of the bass before going into a compressor will often sharpen up the compression action (although the compressor will also bring up any noise contributed by the EQ). There is a way you could have your cake and eat it too, however; a 4PDT toggle switch could switch the effects loop #1 between the pre-EQ and post-EQ positions, which I think would be highly desirable for increasing the overall flexibility of the unit.

OUTPUTS. The multiple outputs are a convenient feature of the IVP; they mean that you can drive just about anything that's thrown at you. In the studio, for example, the balanced output could go into the board for a direct feed while the -10 dB phone jack feeds a standard guitar amp for an amplified sound (if you don't use the tube voice option). At the same time the level output could feed a line level effect that can be mixed in with the other sounds.

I haven't said much about noise performance yet, simply because it isn't that much of a factor. It is considerably better than the "average", but probably not as good as it could be if lower noise op amps were substituted for the 4558s used in the circuit. The only time noise really shows up at all is during high-end boosting with the EQ sections, or with the Tube Voice circuit. However, the Tube Voice is quieter than the average fuzz or tube amp so this is not a relative problem, only an absolute one. Evaluating noise performance is always tricky---what is acceptable to one person may not be acceptable to another---but I would rate the IVP as definitely above average.

SUMMARY. The IVP has, in my opinion, two very strong positive points in its favor.

One of these is mechanical construction, which is well done and lacks only IC sockets to to make it flawless for all practical purposes (especially considering the moderate cost of the unit). Perhaps more important is that conceptually speaking, the IVP is well thought out and highly functional. The idea of including level-matching preamps, comparatively sophisticated equalization, effects loops options, and a footswitch controlled voicing option in one easy to carry package should appeal to many formerly frustrated musicians. Additionally, the IVP is well suited to working with a variety of other pieces of equipment, so it's a good start towards a rack-mounted system. In these respects, Intersound is to be commended for putting out a product that fills a genuine musical need, and not just a marketing need. It's quite clear that they tried very hard to come up with a universal product, and in many ways, they've succeeded. It may be a bit ahead of it's time---musicians are still generally not thinking in terms of modular, component-type systems that recording studios have been using for years---but it is an approach whose time has come, and will doubtless become more accepted as time goes on.

My major candidates for improvement include adding bypass switches for the two equalizers, and offering a lower resonance option for the pseudo-parametric portion. I feel that both of these are important features that should not be overlooked in future generations of the IVP. Additionally, a trim control on effects loop #2 output would help in terms of interfacing to other units. Finally, when it comes to the Tube Voice, only your ears can decide. Some people will like it, others won't; I find it useful for low level, ambience type distortion, but for a rock and roll fuzz effect you'd be better off with something like a Distortion +.

When testing a unit, I have a nebulous criterion called the "inspiration factor": this measures how much I get turned on by playing a piece of equipment or an instrument. Sometimes I'll plug in to something and enjoy it so much I'll have a great time playing away; with other devices, the inspiration factor will be much lower, and I'll end up spending more time twiddling knobs trying to get a sound that <u>does</u> inspire me. The IVP came out well in this respect...when I plugged in I'd usually get hung up and play for hours, especially in checking out different tone settings. If you're thinking of upgrading to a modular, rack mounted amp system, look into the IVP---it may very well have just the kind of features you'd want in an instrument preamp.

If you have questions about the unit, you can write Intersound at PO Box 1864, Boulder, Colorado 80306. As with all our reviews, any comments are welcome.

WHAT, ME WRITE?

Yes, you! <u>DEVICE wants information from its readers</u>. 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.

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CORRECTIONS: In issue 1:79, page 9, the (-) and (+) symbols for IC4A should be reversed. The pin designations are correct as shown. Also, on page 6 the subscription rate for Polyphony was listed incorrectly as $6/year. It is actually $8/year. We apologize for any inconvenience these errors may have caused, and recommend that you make these corrections in your issue now in case you forget to do so later.

EDITORIAL

The other day we received the following letter:

"I have just received your sample copy, and I am very interested in a subscription. BUT!! As much as I liked your first issue, it was not as grandiloquent in its entirety and presentation as I thought it was going to be. Plus, my issue came to me bent, slightly torn, crinkled and wrinkled. I suggest some exterior covering when sending long distances.

"QUESTIONS: (1) Will DEVICE come to me in much better condition? (2) Will it get more lavish and more than 8 pages (sic) to an issue? Thank you, and do keep up the great work...another soon-subscriber, MJE, Greenfield, MA."

This brings up some important points that need answering. With regards to question 1, you have our sympathy---we've heard horror stories from other publishers about the quality of mail service, and now we're collecting our own set of horror stories (of which yours is one). Although we hope this problem is an offshoot of the Christmas mail rush, we are looking into having wrappers or some sort of covering envelope added to the mailing process. But, you should realize that any extra step such as this creates quite a burden on a shoestring operation like ours (both in terms of manual labor, time spent, and money)... which leads directly into your second question.

Will DEVICE get more lavish and more than 16 pages per issue? That is not in our hands, but in yours. Our intention is to be neither consumer advocate nor manufacturer's mouthpiece; our goal is to be an accurate source of information and in-depth analysis of topics concerning the electronic musician. To do this effectively we need editorial and financial freedom. We decided not to include commercial advertising so that we could speak out freely, whether favorably or unfavorably; as a result, we are entirely subscriber-supported (save for the classifieds, which don't add up to much). Perhaps most readers don't realize this, but the cost of the major magazines is as low as it is only because, in effect, the advertisers are subsidizing the cost of the publication.

At the moment, we are running at a loss. But we believe this newsletter is needed, and so does our investor, who has committed enough funds to allow us to continue in our present format for a period of 1 year. We can only become more "lavish" in three ways: (1) we accept advertising, and simultaneously accept the pressures that can follow from some advertisers as a result; (2) we discover oil on the property where we produce DEVICE and become independently wealthy; or (3) we get lots of subscribers, which allows us lower printing and postage costs, which translates into more money to put into expanding DEVICE. Of these alternatives, (1) is not acceptable to us; (2) is highly unlikely; and that leaves (3). So like we said, the future of this publication is in <u>your</u> hands. We'll keep it going for as long as we can, but we have to start somewhere and until we can get about 1000 subscribers, we can't even think about spiffier packaging. DEVICE is produced in its entirety by the two of us, with some very loving support from our friends...this is not a big bucks operation, in fact, it is a considerable drain on our resources. But we believe in what we're doing. We urge all those who also believe in what we're doing to subscribe, because only in this manner will DEVICE grow into greater things.

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We'd like to add one point concerning reviews. We believe that the accuracy of our reviews cannot be disputed, but "truth", like beauty, is often in the eye of the beholder. Realizing this, we open the pages of DEVICE to those whose opinions differ from those of our reviewers...manufacturer and consumer alike. We welcome discussion and dissension; we feel anything that clarifies problems of musicians and manufacturers can only lead to positive end results. We want to be, and believe we already are, a force that improves and increases our understanding of the instruments of creativity...you are invited to join with us in any way you see fit.

Craig Roger

construction: BUILDING THE AMS-100 - part 2

craig anderton

In last month's installment, we discussed a multipurpose input module that accepts hi or lo level inputs, and gives 1) a preamp out, 2) compressed output suitable for noise reduction use, 3) full wave rectified output (octave doubler/envelope output), 4) peak detector output, and 4) two **complementary** trigger outputs. This month, we'll use the trigger outputs to drive some very simple envelope generator circuits.

Figure 1 shows an attack or decay generator, with DPDT switch S1 determining the mode and R6 determining the time constant. IC2 is a 4016 (or equivalent) quad analog switch; we need to use two sections of it for each generator.

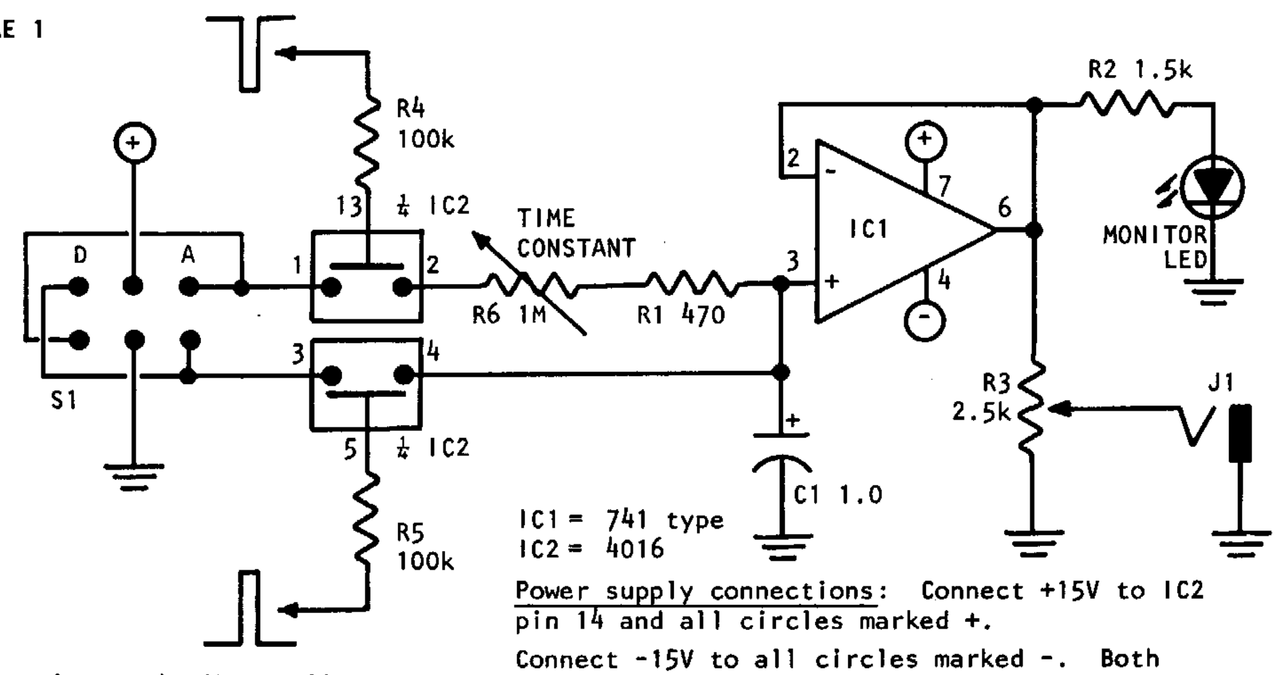
HOW IT WORKS. When no notes are played, pin 13 of IC2 sees a "high" voltage, which means pins 1 and 2 are connected to each other. Additionally, pin 5 of IC2 sees a "low" voltage, which means pins 3 and 4 are not connected to each other. When S1 is in the ATTACK position, current flows through R6 and R1, which keeps capacitor C1 fully charged. The charge on this capacitor is buffered by IC1, which feeds an output level trimpot and monitor LED (more on these later).

When a note occurs and the input module generates a trigger, two things happen: one, pins 1 and 2 of IC2 open up, which interrupts the supply of current to C1; two, pins 3 and 4 close, which dumps the capacitor charge to ground. Remember, though, this trigger only lasts a couple of milliseconds---so as soon as the capacitor is discharged, it starts charging again through R6. The setting of R6 determines the attack time of the generator.

When S1 is in the DECAY position, the reverse occurs: C1 is normally grounded, but when a note occurs, the trigger deposits a charge on C1. In this case, R6 determines the time it takes for the charge to leak off C1, thus giving a decaying envelope output.

IC1 is a 741 type op amp; this op amp has a relatively low input impedance that loads down C1, so if you want to use a spiffier op amp like the LF356 by all means go ahead. The envelope output normally goes from ground (0V) to about +13V. Since the AMS-100 voltage controlled modules (and several other types) require 0 to +10V control voltages, R3 allows you to trim the envelope to an exact +10V on peaks. If you have 0 to +5V equipment

FIGURE 1

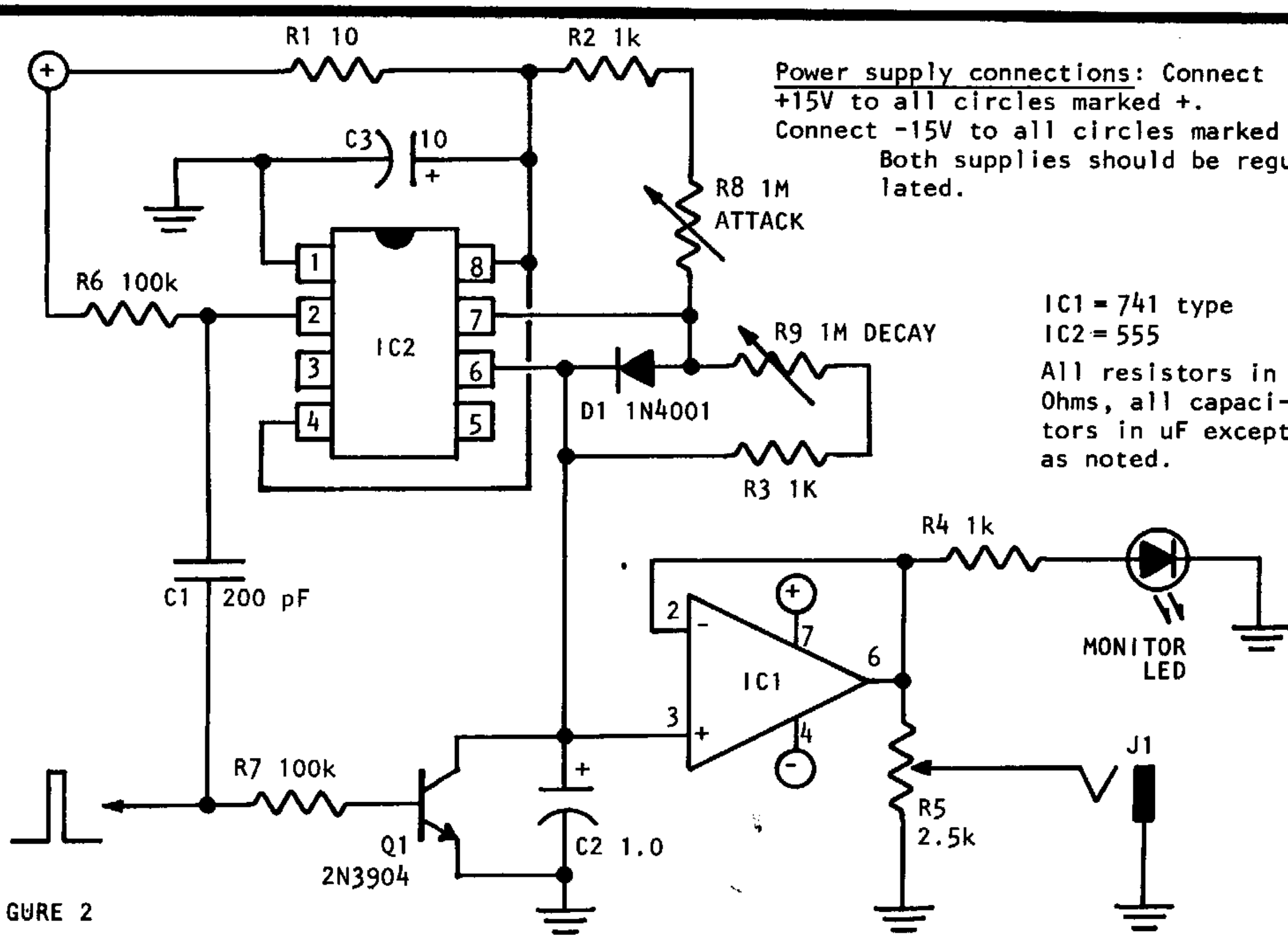


All resistors in Ohms, all capacitors in uF.

Power supply connections: Connect +15V to IC2 pin 14 and all circles marked +.

Connect -15V to all circles marked -. Both supplies should be regulated!

Connect pin 7 of IC2 to ground.



Power supply connections: Connect +15V to all circles marked +. Connect -15V to all circles marked -. Both supplies should be regulated.

IC1 = 741 type
IC2 = 555

All resistors in Ohms, all capacitors in uF except as noted.

FIGURE 2

(PAIA etc.), then you can use this same control to trim the peak to +5V. To calibrate, put S1 in the attack position, and measure the voltage at the envelope output jack (J1). Adjust R3 until you get exactly the peak voltage you require. The monitor LED shows you the intensity of the envelope. Note that in the attack mode, the envelope will go to 0V as soon as it receives a trigger, and then sustain at maximum voltage until the next note occurs.

Figure 2 shows a slightly more complex, but also more versatile, envelope generator. It generates an attack voltage which, upon reaching its peak, starts a decay cycle. There are separate controls for each function.

HOW IT WORKS. IC2 is a standard 555 timer IC. Upon receiving a trigger pulse at pin 2 C2 starts charging through R8/R2. When the voltage across C2 reaches about +10V, then C2 starts decaying to ground through R9/R3. As in the previous generator, the timing capacitor is buffered by an op amp, and again, it wouldn't hurt to use a high input impedance op amp for IC1. The output structure is also very similar---a pot to trim the maximum envelope output voltage, and a monitor LED to see what's going on.

Note that in addition to triggering the 555, an input trigger also turns on Q1, which momentarily shorts out capacitor C2. This assures that whenever a new note occurs, the envelope starts from 0V. To calibrate the generator, ground pin 2 of IC2 temporarily and measure the output at J1. Adjust R5 for the desired peak envelope voltage. Note that we only use the positive going trigger with this envelope generator; the other goes unused.

WHICH ONE SHOULD YOU BUILD? In my present system, I have both envelope generators in use, since they both have their uses. I'd suggest using the Figure 1 version in a dual configuration since we're only using two of the 4016's switches, and there are two other switches in the chip just waiting to be used. Remember, the object of the AMS-100 is to create complex, intricate sounds, and to do that we'll need several envelope generators... not just the one or two normally found in guitar synthesizers. So, since these circuits are quite inexpensive, you might as well go ahead and build several of them; we'll put them to good use further on in the series.

I would like to begin this section with a list of possible problems you should check when you are stuck with a bad unit, and wish to fix it yourself. I have found that problems associated with effects aren't usually caused by bad components (ICs, resistors, capacitors, etc.), but instead by little things such as:

- Dead batteries. Don't laugh, the company I worked for got units back all the time along with nasty letters that complained about the effect. Many times all that was wrong was dead batteries, so always check them first.
- Bad solder joints. This is a common problem that causes a lot of headaches for musicians. If you have an intermittent problem, there's a good chance that it could be a wire or a component that has a bad solder connection.
- Lifted or loose solder pad foil. This is another cause of intermittent operation. It differs from a bad solder joint in that the problem is with the circuit board foil run that is underneath the solder joint.
- Input and output jacks shorting to metal. Jacks have a tendency to loosen up and move around, which can cause problems if the hot part of the jack touches the metal casing when you plug into it.
- A bad mechanical component. Footswitches, pots, and the like wear out occasionally. It's pretty obvious when they go bad, so we won't get into it any further.
- Power supply problems. AC line power supplies have a tendency to fail quite often. Transformers, power diodes, and regulator ICs carry a fair amount of current and can go bad.
- Bad electrical components. Semiconductors are the most likely to fail; resistors and capacitors don't usually burn out, but resistors can crack if the unit has been dropped on a hard surface too many times.

Chances are if you have a defective unit, the problem would be one of the causes listed above. Now, to narrow it down a little further I will list symptoms that your unit might suffer from in order to help identify the most likely source of trouble.

TROUBLESHOOTING

by Gary Kirkpatrick

- No power (if AC unit). Check AC line cord for any cuts or other defects, solder joints or loose connections on the circuit board (follow the line cord into the circuit, check the solder joints in the vicinity of the transformer and power supply parts), and burned out parts (you can probably identify it by the smell---if you turn an effects box on and it smokes, look for a bad transformer or power supply component). Also look for broken parts (cracked etc.) or broken wire connectors.
- No effect (AC power, but no effect). Look for a bad footswitch; since they're used a lot, they go bad occasionally. Also, a misset pot may be problem so make sure all the controls are oriented properly. Check parts and solder connections as listed above.
- No output. First, make sure your patch cord is OK. In case you don't know this already a good way to check the cord is to plug one end into your amp, turn up the volume a bit, and touch the tip of the other end with your finger. You should get a hum without any real crackling or "rice krispies" sounds. Also make sure that the jacks are mounted securely and don't short against other parts in the device. Lifted foil runs are hard to locate, but can sometimes cause the no output problem; use a magnifying glass to check for it.

This should give you an idea of some of the most common problems to look for. Now, here are some final hints on actually repairing a non-functioning device.

First of all, get your test area set up; nothing fancy is needed, just an area with good light. Have your amp set up with good patch cords that reach to your test area. Have your soldering iron ready, along with screwdrivers, wire strippers, etc. Open up the unit (be careful to remember what goes where, it's good to make a diagram to be sure).. If you are working with a line powered effect, be very careful not to touch any of the terminals that carry house current! It can be a most unpleasant experience.

Most important is to determine if you're getting power to the unit. If you have batteries, check them under load with a Voltmeter. Alternately, touching the tip of your tongue to the battery and checking for its "zap" potential is also a pretty good test. If you have an AC powered unit and it has an indicator light that is on, that may mean that the power supply is good, but it also may not. If the indicator is off, it may be a problem in the power supply or in the indicator. The best idea is to use the process of elimination as much as you can...**having a Voltmeter around wouldn't hurt, either.**

If you think the problem involves the circuit board, you should have good overhead lighting (circuit board cracks are really small and hard to see), a magnifying glass, and a small paint brush with alcohol to clean any flux off the circuit board. If you notice any solder joints with too little solder, loose joints, and other problems, touch them up with your iron. You should remove the previous solder (Radio Shack has de-soldering tools) before you add in any new solder. Be careful not to cause any solder bridges in the process.

If after checking all this over you still have the same problem, chances are a component is bad. If you think it is the IC and you intend on replacing it, you might as well add an IC socket while you're in there to make things easier. One thing about replacing ICs is to make sure you have a pin number diagram when you take out the bad IC. No matter what you do, though, don't just start taking out ICs until you are sure you've ruled out all the other possible problems.

Well, that's about it. Let me add one more thing about troubleshooting in addition to using the process of elimination: be patient. It's not hard to fix these gadgets once you get an idea of what to look for, as long as you take your time. If you are a complete beginner in electronics then I would advise sending the unit back to the factory for servicing, if you can't find a simple problem like a loose wire or something like that. However if you can solder and know the differences between the various parts, then go ahead and give it a try.

I hope I was able to help any one with a bad effect...if you have any questions, problems or comments on this article feel free to write, I would be glad to hear from you. My address is: Gary Kirkpatrick, 8443 Beacon Avenue, Seattle, WA 98118.

info

A new company has entered the guitar synthesizer/processor market, **SCD TECHNOLOGY, INC.** (285 K Sobrante Way, Sunnyvale, CA 94608 - tel. (408) 739-2880). Their product, the NEBULA, makes extensive use of the SSM chips designed by Dave Rossum and Ron Dow. The guitar signal is processed directly (no hex pickup, no PVC) and is modified by way of: an input processor (consists of compressor, fuzz, and octave divider/multiplier), a VCF, a VCA, envelope generators, and a parametric equalizer. List price is $795 + options.

SIGNETICS (811 E. Arques Avenue, Box 9052, Sunnyvale, CA 94086) has introduced the low noise NE5532 dual op amp. It is pin-compatible with the MC1558 type op amps, but can drive a 600 Ohm load with 10V RMS of signal.

Rockland Systems Corporation (Rockleigh Industrial Park, Rockleigh, NJ 07647) has announced the Model 752A dual channel, low pass filter. It's tunable from 1 Hz to 100 KHz, by remote digital control or manually. Each channel has passband flatness within ½ dB p-p, with a 115 dB/octave rolloff. By cascading sections, you can achieve a 230 dB/cctave rolloff. The Model 751A is a tunable high pass/low pass (adjustable bandpass) elliptical type filter. These filters are intended for precision laboratory use.

HEAR, Inc. (1122 University Avenue, Berkeley, CA 94702 - tel. (415) 848-6262) is expanding the ZETA line of guitar synthesizer equipment with the addition of the ZETA MODULATOR. The unit provides a multi-mode VCF, externally triggerable envelope follower, and a VCM (voltage controlled modulator) which is similar to an LFO. May be used with the Zetaphon synthesizer, or as a signal processor for guitar, keyboards, etc. Contact HEAR for price and availability information.

You may have heard about the new LED VU meter chips...now **Litronix** (19000 Homestead Rd., Cupertino, CA 95041) has lined up ten rectangular LEDs to make a single 1" array. All LEDs in the array are individually addressable for both anode and cathode. This looks like a good part to have if you want to add inexpensive bar graph displays to musical (and other) equipment.

Due to space limitations, the pedalboard article and analog delay interview have been pushed ahead to next month's issue.

(GIZMO cont.) *produces) is the fact that it was developed by two professional musicians, who took a basic concept and with very little technical background turned it into a legitimate, working product.*

At the recent NAMM (National Association of Music Merchants) convention I was fortunate enough to conduct this interview with Kevin and Lol, and finally see the Gizmo in action. As we join the proceedings, the three of us are gathered around the table at one of the Disneyland Hotel restaurants, nursing drinks and recuperating from the previous night's rabble-rousing. For some reason this all seems fitting.

ROGER: All right, it's interview time. What's happening with the Gizmo as far as getting it into full production?
KEVIN: We should be shipping within 4 to 6 weeks (as of the end of January), when it will be officially available. List price is $250 - full mark-up.
LOL: What do you mean by full mark-up?
K: I'm not sure... oh yes, that means the dealer pays about half.
R: What have been some of the problems of bringing it into production? I thought the Gizmo was originally due out a couple of years ago.
K: I think the main problem has been consistency. When we invented the thing it worked fine, but it was inconsistent; some days one string would be working, some days another. There also had to be a lot of research to find the right materials. We had to make sure there were no eccentricities in the wheels, which was quite crucial.
R: Were these quirks occuring at the time of the <u>Consequences</u> album?
K: Yeah! A lot of the tracks on that were recorded string by string, which is one reason why it took 14 months to record. The other reason for the delay in production was because we're in charge of quality control of the thing, 'cause it's our baby, and we didn't want to be unhappy with the integrity of the product. Not only did it have to work and work well but we had to fit it into a specific price range...this was something we had decided to do from the beginning.
R: With guitar effects that seems to be a very important factor; guitar players aren't used to spending lots of money for one piece of equipment. Perhaps that is one of the reasons why the guitar synthesizer has taken off so slowly.
K: Ah, that's a problem. But remember, a Gizmo is not supposed to do what a synthesizer does. It's really a different approach, an alternative way of getting something different happening.
R: I liked the "cello" sounds Lol was getting better than a majority of the synthetic string sounds I've heard.
K: Oh yeah! Well, first of all it's not synthetic. It's similar to what a violin bow does to a string on a violin.
R: What was the basis for the collaboration with you and Lol in the original creation of the Gizmo?
K: It happened on a session a long time ago. We were recording a track called <u>Old Wild Man</u> which eventually ended up on the <u>Sheet Music</u> album. At that time the only thing you could do to get string sounds were the Mellotron and, of course, real strings; the string synthesizers hadn't been fully developed yet. Since orchestras are <u>so</u> expensive and such a time consuming operation, we began to wonder if there was a way to get a guitar to make the sounds we were after. So we fucked around for an afternoon , strapped an old Stratocaster to the wall, and I got this electric drill and stuck an eraser on the drill bit and held it up to the guitar. We got this horrible noise, but it gave us the idea that eventually became the Gizmo. We realized after a bit that the idea would work and we spent a year and a half getting involved in the engineering side; the development side, which was very interesting. Neither of us had been into engineering at all before this. We weren't into mechanical things at all, we were writers up until then.
R: This is one of the things that interested me about the development of this instrument: the fact that you were coming from an artistic background. This is not the norm.
K: The thing was, though, we could only take it so far by ourselves. We made up a pretty flaky prototype, all strapped together with gaffer tape, which worked after a fashion. Then we found out about this program run by the British government, whereby certain British universities have a thing called "industrial liason". The university provides assistance to people who have new inventions but don't have the facilities. So, we approached these people and got involved with two gentlemen ; one headed the physics department, and

another in the acoustics department. They became very involved with us. They had all this mechanical equipment there, and they gave us information on how to approach things... such as the use of lathes and the like, in order to get a truly working prototype. When that was done, we used it in the context of 10CC for four years. In '76 we visited Musitronics at the end of our U.S. tour, and made a deal with them about producing it. So the story has been pretty placid from beginning to end; that is, our ability to put time into the project has been very on and off due to our musical schedules.

R: Did you use it on stage at that point?

K: We did. Who knows what the audience thought, though...every time a string sound came out, the spotlight would go to Lol with the guitar.

R: Lol, I noticed during the demonstration that you don't use a pick. Do you feel the Gizmo is better suited to those who use their fingers?

L: Yes, it probably is actually. Someone who is more finger oriented will find it easier to use the Gizmo's different techniques.

R: How easy is it for someone to just pick it up and start playing?

L: It's just a question of practice, as with anything else. Most of the people who have picked it up have been pick-style guitar players. Sometimes a person picks it up and within seconds they've got it happening properly, or it takes a guy about 5 minutes to get a handle on what's up. Then again, there are people who come over and you know they'll never get it right. There is a problem with where to put your pick when you're using the Gizmo, but the production people are putting in a place to hold the pick...maybe some sticky spot on the Gizmo itself using a new type of glue so that things don't get all gummy. Using the Gizmo is a specialized technique for guitarists, like playing pedal steel. Also, people are generally so excited when they come over to try it, that they get too anxious and heavy-handed. They start trying to do all this flashy stuff right away instead of trying to understand and get used to it.

K: Guitar players seem to be more used to putting the guitar <u>into</u> things.

R: Yeah, instead of having to manipulate something attached <u>on to</u> the guitar. The machine does seem to have a very nice touch, though.

L: Yes, we're pretty pleased with it. The design is now very low and away from where your arm normally rests. Did Kevin tell you who did the final package design?

R: He said the man who designed the Studebaker, but I thought he was putting me on.

L: No, it's true. His name is...

K: Herb Ross!

R: It does have sort of Studebaker lines to it...

K: That's the second guy to be involved with us who was originally involved with automobiles. The physics professor used to work at Rolls Royce doing design.

R: The old "cars and music" theme again. During your demonstration I noticed a raspy or grinding sound on the low A string. Are there times if a guitar player is being particularly violent with his guitar that the Gizmo's wheels may miss a string?

K: That raspy sound is because the model we have here doesn't have the finished wheels that are going on the ones in production. The ones on this model have their little eccentricities.

L: I know what you mean, but I think the thing sounded fantastic overall. There was a slight overtone on the "A" string but that's because the thing has fallen out of adjustment from all the use it has been getting here. You have to understand that at a convention like this you get all manner of guitar players coming up and attacking the thing in all sorts of crazy ways. And on a couple of occasions, there have been people giving it a good wacking. Naturally, with this kind of abuse it's not going to stay fine tuned forever. This is a common problem with demonstrating in a convention context: if you let one guy try it, you've got to let everyone try it. And it always seems the first or second person up is "Mr. Heavy Hand". It's like any instrument; you have to have a feel for the thing.

R: What's involved in mounting the Gizmo on a guitar?

L: Well, the Gizmo does not actually attach to the guitar, so you don't have to screw a mounting plate into the guitar. What you do is get a replacement bridge for your guitar from the dealer (it's included in the price)---say, a replacement Strat bridge. The Gizmo Strat bridge is the same as a regular Strat bridge except that it has a mounting plate for the Gizmo molded on.

R: What about other types of bridges, like the "Tune-O-Matic"?
L: We've designed something like 12 bridges for the 12 most popular guitars. And if you don't happen to have one of those guitars, there's the universal bridge which is on the Guild here at the show. It's a separate thing that sits beside the bridge. We're also using a special new adhesive that's just come out; you stick it on the guitar in the right position and it stays there forever. But, if you want to take it off it won't leave any marks...which I think is amazing. Then there is an adjustment for string spacing. When you take the casing off there's a key, and behind that, a little button that you unscrew. You then move the key along and press a button down until it sounds as if the wheel is moving smoothly against the string. You see, no string will sound beautiful musically unless the wheel hits it correctly. If you drag a bow across a violin string incorrectly it can sound horrendous; that's what the Gizmo can sound like if you don't adjust it. If it's too near the string it grates, and if it's too far up the string you won't hear anything at all.
R: Do the wheels or strings wear down after much use?
K: When we first designed it we thought we were going to have a lot of problems with that, but actually that doesn't happen because the wheels are just tickling the strings. They're not plucking like you do with a pick, so they really get very little wear. The original prototype has had the same wheels for four years.
R: The last question is slightly unrelated, but in keeping with the electronics and music theme. What would you do if suddenly the "big plug" was pulled?
L: You mean if there was no more electricity?
K: I'd be thrilled.
L: Yeah! I've always been partial to steam, anyway. I'm very into steam. Besides, we could go back to an "au naturel" style of living.
R: What about music?
L: Fuck music (laughs).

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"ANSWERS"

Dear DEVICE: I am working on the AMS-100 and have run into a stumbling block. I have been unable to locate IC cross reference numbers for the 4739, 4136, and NE571. Could you give me an SK or ECG part number for these ICs? Also, is J1 a stereo or mono jack?
--Mark Holland

ANSWER: J1 is a stereo, closed circuit jack. The reason for using a stereo jack is to accomplish some switching functions; do not, however, plug a stereo plug into this jack. It is designed to work with a standard mono cord/plug combination.

The question of where to find parts is a constant problem for experimenters, but it need not be. Unfortunately, the SK and ECG type series of parts do not cross-reference some of the more interesting/exotic chips that we use in the AMS-100. I would strongly recommend getting these parts from reputable mail order companies. The 4739 and 4136 are available both from Bill Godbout Electronics (Box 2355, Oakland Airport, CA 94614) and from Jameco Electronics (1021 Howard Avenue, San Carlos, CA 94070). The 571 is virtually identical to the NE570, but has looser specs and costs somewhat less. The 570 is available from Jameco; the 571 is available from E-Systems (Box 5305, Berkeley, CA 94705). I suggest writing to these companies for current information on pricing and availability--CA

Dear DEVICE: Regarding the AMS-100, Craig says to minimize the "attack blip" he ran the guitar signal through a delay line. I fail to understand how that solves the problem. Also, how is the delay line placed in the line of circuit modules? I assume a tape echo would work equally well. Finally, will it be possible to run the entire AMS-100 system from a ±15V supply, 1A per side? --Mike Fusion

ANSWER: First of all, power supply requirements will vary depending on the number of modules you use to implement the system...obviously, a system with 40 modules will draw more current than one with 10 modules. However, a supply like the one you describe will handle all the basic modules as well as some duplicates. Next, refer to page 10 of the 1:79 issue of DEVICE. Note in figure 4 that a plucked note on the guitar occurs just before the start of the attacking envelope, due to the response time. So, while we want to use the regular, undelayed guitar signal to feed the envelope triggering circuitry, we'll need to delay the audio signal somewhat before feeding it to modules controlled by the envelope generators covered in this issue. Adding this small amount of delay insures that the guitar note pluck occurs at the same time the envelope generator creates its envelope, thus eliminating the blip. I should perhaps have pointed our that this is only a problem under some conditions, but when the problem occurs it is annoying. As far as placement is concerned, run your guitar into the AMS-100 input module, but run the output you choose (preamp out, compressed out, or full-wave rectified out) through about 10 ms of delay before feeding it to the other audio processors in the system. A tape echo unit might be acceptable, but I doubt it because the delay is long enough to throw you off while you're playing. The advantage of using a short delay is that it appears to the ear that the guitar is playing in real time, but the modifiers see a suitably delayed signal. There will be more on this whole subject when we get to the delay line part of the AMS-100 --CA

Dear DEVICE: Will there be parts kits and boards available for the AMS-100? (many readers)

ANSWER: At the moment, there are no plans to offer AMS-100 parts kits. However, John Simonton at PAIA has offered to provide kits if I will provide them with documentation, board layouts, etc. Sad to say, I don't really have the time to do that; so unless there is a lot of demand for parts kits, you'll have to go it alone for now. I am sure, though, that some of the AMS-100 modules will show up in other projects I'm working on for which there will be parts kits --CA

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