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

construction:
BUILDING THE AMS-100
by craig anderton

Welcome to DEVICE, and to this series of articles on building the AMS-100. AMS stands for "Audio Modification System", and 100 is the series number. The completed system is intended for processing guitar, but will also work with voice, tape tracks, and other monophonic signal sources. The AMS-200, which we will cover at a much later date, is a polyphonic guitar synthesizer that requires the use of a hex pickup.

The AMS-100 is a modular system. In this age of preset machines, the idea of a modular box may seem antiquated...but I don't think so. What we're aiming for is maximum flexibility, and a modular system can provide flexibility with low cost.

What shall we include in our system? Well, we need voltage controlled, wide range filters; voltage controlled amplifiers; envelope generators; envelope followers; low frequency control oscillators; voltage controlled ring modulators; voltage controlled delay lines; and other standard synthesizer modules. With these, we can process the sound of the guitar way beyond the old fuzz/wa-wa combination...and in many ways, I like the sound of processed guitar better than guitar-controlled oscillator sounds (as used in most common guitar synthesis boxes).

EXTRACTING ENVELOPE GENERATOR TRIGGERS. One of the first modules we need has to take the guitar signal and derive triggers that can drive envelope (or transient) generators (in case you're not familiar with the term, these are devices that generate a programmed change of voltage, on command, for controlling the parameters of voltage controlled modules). "On command" (cont. on page 7)

REVIEW:
ARP AVATAR
by buzz kettles

The ARP AVATAR guitar synthesizer is essentially an ARP ODYSSEY keyboard synthesizer with a Guitar Interface unit replacing the keyboard and its associated interface electronics. The AVATAR is sold as a stand-alone unit complete with a hexaphonic pickup (ie a separate output for each string) that must be appropriately mounted on the user's own guitar, and lists for $2650.

The synthesizer section of the AVATAR has the standard collection of modules for producing single-voiced monophonic melodies or harmonies. This section contains two Voltage Controlled Oscillators (VCOs) one Voltage Controlled Filter (VCF), one Voltage Controlled Amplifier (VCA), one Low Frequency Oscillator (LFO), one Sample and Hold unit (S/H), and two envelope generators. One is an attack/decay/sustain/release (ADSR) type, and the other an attack/release (AR) type.

The signals derived from the Guitar Interface constitute the primary control signals for the AVATAR. The Interface outputs a gate and trigger for the last string plucked. This timing signal is internally patched to the two envelopes (ADSR and AR) so that they will trigger with each plucked note. The Interface then applies the audio signal of this 'last string' (isolated through the use of the hex pickup) into a Pitch-to-Voltage (P/V) converter. The control voltage output of the P/V unit is internally routed to the two VCOs so that they can track the pitch of the guitar. This control voltage can also be applied to the VCF (through a slider labelled 'String CV') so that it may also track the guitar.

An additional control voltage is derived in this Guitar Interface (cont. on page 2)

(ARP AVATAR) section, termed the Envelope Follower (EF CV). The general shape of the EF CV is similar to that of an AR envelope generator; it produces an envelope-style control voltage whose depth, and to some degree shape, are proportional to the loudness of the 'last string' audio signal. The AVATAR allows for this control voltage or its inverted image to be applied to the VCF and/or VCO 2. When applied positively to the VCF, the resonance can be adjusted to yield the customary envelope follower/filter sound (Mutron III et al).

One additional byproduct of the Interface section is the Hexaphonic Fuzz (Hex Fuzz), which fuzzes each string independently and then mixes these signals. This produces a clean sounding fuzz which can be sent directly to the output (entirely bypassing the synthesizer section), and/or mixed into the VCF along with the VCOs.

The VCOs have either sawtooth or pulse waveforms available. One of the more interesting and useful aspects of the VCOs is that VCO 2 can be 'hard' synched to VCO 1 (hard sync corresponds to clamping in Moog terminology). This means that regardless of where VCO 2 is in its cycle, it can be forced to restart each time VCO 1 does. This forces VCO 2 to have the same period as VCO 1 and therefore the same fundamental pitch. Varying the frequency of VCO 2 will cause changes in the harmonic structure above this sync-generated fundamental.

The VCF, VCA, LFO, and envelope generators have no unusual or extraordinary features. There is an additional waveform that can be mixed into the VCF; this is the Ring Modulation product of the pulse waves of VCO 1 and VCO 2. Although there are no specific adjustments for the Ring Modulator, all adjustments on the VCOs will have an effect on the customary 'metallic' Ring Modulator output. The Sample and Hold unit (S/H) has the facility to add variable amounts of lag to its control voltage output. This lag is equivalent to portamento or glide in that it can be used to effectively round off the machine-like characteristics of most S/H units. It also has the ability to sample on every plucked note, or repeatedly using triggers derived from the LFO's square wave.

EVALUATION. That's what the AVATAR looks like on paper; playing it is another matter. I sincerely hope my various criticisms do not make ARP feel as if they alone have been put on the chopping block, since most of my complaints apply to the other guitar synthesizer manufacturers also. However, user feedback is what shapes and improves next year's model; based on the recent questionnaire distributed to AVATAR owners by ARP, I believe the following suggestions will be evaluated with the open ear ARP would like performers to believe they have.

Externally, the AVATAR offers guitar players the important convenient aspect of a one cable and one box unit. The casing is industrial grade (a necessity for road usage), and the internal circuitry is mounted very securely. Unfortunately, the black plastic slider caps are very fragile (ARP says that it has something to do with the particular choice of glue and plastic). This limitation could be overlooked because the slider positions are seen more easily after the caps have broken off (the sliders themselves are white and the casing is dark). The main cable is another source of problems; the ones I've used have deteriorated in the space of a few months' heavy use. Perhaps ARP could supply replacement cables at cost since a backup is an absolute necessity.

My personal feelings towards the Guitar Interface section are mixed. The actual P/V converter works well enough for live performance when the pickup has been properly positioned and the individual trimpots have been (time-consumingly) adjusted. Unfortunately the calibration technique given by ARP is very rough and highly subjective, and I consider this to be one of the major drawbacks of the AVATAR. Too many functions are being calibrated off each of these trimpots; they compensate for bridge curvature, differences between the individual string signal strengths, envelope follower sensitivities, triggering sensitivities, the tone of the clean guitar/hex fuzz for each string, and to a large degree, the amount of sustain available on a held note. Of course, all of these parameters vary from guitar to guitar, but inevitably some tradeoffs are bound to be made. Another important drawback is the fact that if you break a string, you can't just pick up another guitar and get back in action; and if the new string varies from the old one, you may need to re-calibrate.

A possible solution is the development of a system where the pickup has adjustable poles and the AVATAR has LEDs or some other indicator to tell the user when the optimal position has been achieved. Perhaps a single sensitivity control for the Envelope Followers could be offered on the front panel so that the users could adjust this all- (cont. on next page)

(ARP AVATAR) important CV to their personal picking style and taste. This type of overall sensitivity is already offered for the triggering mechanism. It should also be noted that holding a note for too long is risky at best. As the signal dies away, the P/V can 'hear' the noise in the signal and it gets confused, outputting random notes. If you know you're going to hold a note for a long period of time, you should use the Sustain footswitch. By the way, these various glitches can make recording use of the AVATAR very limiting...not only does the performer need to be pleased with his/her performance, but the machine has to pull it off too.

Getting back to the sustain footswitch, I believe it could have been implemented in a better fashion. As it behaves now, its function is to hold the VCOs and to keep the gate on. However, it also eliminates the use of the Envelope Follower and blocks the triggering mechanism; it's basically a lockjaw for your synthesizer. It also does not function at all when the AVATAR is being controlled through the external control voltage input (located on the back panel). I feel it would have been much more useful if it had sampled the 'String CV' just before it gets applied to the VCOs; then, a switch could have been added to lock either one VCO or both. This would then allow for full synthesizer functioning on one VCO while the other was being held as a drone. The footswitch would then also function with external control voltages.

Another annoying limitation of the AVATAR is the option to use either the Envelope Follower CVs or the Pedal Controller, but not both. When a person plays guitar both hands are occupied, so real time control is more restricted than for keyboard synthesizer. While you are playing the AVATAR you can't control very much; pitch and either Envelope Followers or Pedal. In terms of getting control to the user this is a real drawback. Also, there is no way to apply the Envelope Followers to the VCA. This would have offered guitar players a function consistent with guitar playing, namely, plucking harder makes louder notes. This would be a convenient form of dynamic control.

The next set of complaints stems from the fact that ARP took their best middle-priced synthesizer, added a few simple modifications, and designed a Guitar Interface section for it. This relatively unimaginative approach to the design of a synthesizer leaves, to my ears, a great deal to be desired.

The lack of any octave switching on the VCOs makes tuning the AVATAR a time-consuming task, and also eliminates the option of re-tuning in a live performance situation. Octave switching had been available on the ODYSSEY on the front panel area which was replaced by the Guitar Interface section; I think they forgot the control while transfering from ODYSSEY to AVATAR.

The LFO is synched to the player, since the LFO has been set to start fresh cycles on each new trigger derived from plucking. (Yes, this is the same concept as 'hard' sync. In this case, the LFO is synched to you so if you don't play in a periodic fashion, a random sync fundamental is generated.) Unfortunately, this eliminates the opportunity to use the LFO to churn a VCF in the same way that flangers, phasers, and autopan units can be made to turn in a cyclic fashion. Also, many times LFOs are used in conjunction with a S/H unit to act as a harmonic accent controller for a VCF, but this is also not possible with the AVATAR. I have been told by an engineer that a switch could have been put in, at almost no cost, to either sync the LFO to you or not. There are some advantages to a synched vibrato however, such as more expectable vibrato.

There is a switch for VCO 1 that turns this VCO into an LFO (called the 'low frequency/string off' position), but I find this of little use. On the ODYSSEY, VCO 1 could be used for modulation (audio or low frequency) by using the S/H MIX CV which was available. Without this option, the low frequency position seems far less useful; and there's another limitation added to the S/H unit. VCO 1 and VCO 2 can no longer modulate each other, and sampling the audio pulse waves of the two VCOs produces a random bistable control voltage at best. Instead of using two of the four inputs to the S/H in this way, perhaps sampling the ADSR, AR, or Envelope Follower might have yielded more useful musical applications.

If you require modifications to the AVATAR, ARP has made available a Custom Engineering Group (45 Hartwell Avenue), Lexington, MA 02173). They naturally charge for this service, but then again your warranty is not voided. Many modifications are straightforward enough that finding a nearby musical engineer can save time, money, and will allow more distinct and specific modifications (ARP has 15 different modifications, of (cont. on next page)

(ARP AVATAR) which different ones apply to different ARP synthesizers).

GETTING TO KNOW THE AVATAR. ARP distributes a primer of fundamental patches along with a demo tape; here are some interesting starting points for patching an AVATAR.

For practicing or for beginners, I suggest setting both Attacks to zero, both Decays to about one-third, apply the ADSR to the VCF and the AR to the VCA. Tune both VCOs to unison, sync them, and apply a bit of the +EF CV to both the VCF and VCO 2. Throw in a dash of portamento, and go. This patch offers a very standardized AVATAR sound, with the ability to 'feel' the +EF CV and get used to the way the AVATAR triggers.

Lowering VCO 1 down one octave, mixing in some Ring Modulator, inverting the EF CV (no longer applying it to VCO 2), and shortening the ADSR serves as a nice plucked lead tone.

There are many good bass-type sounds, and they all involve lowering both VCOs and unsynching them. Closing down the VCF pretty much smooths off the bite and adding a little LFO vibrato can make for a more 'rubbery' sound.

As an additional lead sound, I have used VCO 1 tuned down two octaves, and VCO 1 down one octave, with both oscillators unsynched. Adjust the control range for the pedal on VCO 2 to an octave, and set the initial setting for the VCF so that it is relatively closed down with the pedal in the (full back) zero position, and somewhat open in the pedal forward position. When a lot of Ring Modulation is used, this patch yields two basic sounds. With the pedal back, a subdued bassy sound occurs and with the pedal forward, a ripping lead ring sound. Varying the pedal near the high end yields a distortion-feedback-pitch bend sound which I find very Hendrix-like in character and texture.

For those of us who like chords, I have found that setting VCO 1 down an octave, VCO 2 down an octave and a fourth, and mixing them each at 50% with the full Internal Hex Fuzz and 25% outside Hex Fuzz yields a pseudo-synthetic chord sound. The ADSR to the VCF should be set to taste, and the VCA gain should be set at about 30% with the AR applied about 80%.

Also, since most guitar players who are interested in synthesis have probably got a load of 'boxes', these can be used with the AVATAR by coming in the External Input jack on the back panel. I use the Guitar Out, go through my 'boxes', and back into the AVATAR. If an echo or other delay unit (even reverb) is somewhere in this pseudo send/return line, the results can be quite spectacular...for instance, as the AVATAR envelopes open, you hear the delay (space), and as they close you come back to dry again.

If others have any comments on this review, you can write me c/o DEVICE. I'd like to especially thank Dave Torn for his invaluable assistance in the preparation of this article.

info

SWITCHCRAFT (5555 North Elston Avenue, Chicago, IL 60630) has announced a new type of plug with a threaded collar that fastens on to subminiature jacks with a #10-48 threaded bushing. There is also a cable strain relief provision. This seems like one of those simple, but effective, ideas that would also be applicable to standard ¼" phone plugs. Imagine...no more pulling the cord out of your guitar or amp on stage!

FERRANTI SEMICONDUCTORS is now advertising an ultra-low noise, wide bandwidth preamplifier IC designated the ZN459CP. They claim a 15 MHz 3 dB point, 60 dB of gain, a supply current of 2.5 mA at +5 VDC, a power supply rejection ratio of 42.5 dB, and a noise spec of 800 pV/root-Hertz (which is highly impressive). It seems like this would be a likely candidate for microphone preamp applications. Their address is East Bethpage Road, Plainview, NY 11803.

For those of you who'd like a pre-programmed synthesizer without having to install extra circuitry, BOURNS, INC. (1200 Columbia Avenue, Riverside, CA 92507) sells pushbutton potentiometers. These have thumbwheel-like dials that allow you to dial in an exact, calibrated pot position. For example,one model offers 3 decades of range, giving .1% resolution in standard resistance values from 1k to 1 Meg. Unfortunately these things are not cheap (about $12 in large quantities), but they are precision pots with excellent resettability.

NATIONAL SEMICONDUCTOR (2900 Semiconductor Drive, Santa Clara, CA 95051) has announced a power op amp in an 8 pin minidip package. Said to be able to furnish 250 mA of load current, the LM13080 is designed for moderate power applications including audio amplifiers. It will work off as little as 3 Volts, and costs less than $1 in quantities of 100.

PREPARED GUITAR—A PRIMER

—by mark kata—

Ask any guitarist how he or she changes the sound of a guitar, and invariably the answer will fall into two categories: electronic manipulation with various effects (flanger, fuzz, wa-wa, etc.), or manual manipulation using finger vibrato, right-hand muting, artificial harmonic production, or bottleneck (slide) techniques. Employing any of the options given above still retains the guitar's characteristic sound; for example, a guitar played with a slide through a flanger still sounds like a guitar. However, a third, and rarely used category of sound manipulation exists for all stringed instruments: mechanical manipulation of the strings by preparing them with alligator clips, washers, strips of plastic or paper, and other devices.

Avant-garde composer John Cage (born 1912) invented the prepared piano in 1938 by wedging screws, nuts, bolts, washers, and other assorted pieces of hardware between the strings of the piano. In effect, he invented a totally new musical instrument that would clonk, jingle, pang, and ping just as a percussion ensemble does, while retaining the standard musical notation for the keyboard. With this innovation, any keyboardist could greatly expand the percussive and harmonic avenues available on a piano with a trip to the local hardware store. (For those interested in Cage's music, and for details on the types of preparations he used, refer to the album John Cage: Sonatas and Interludes for Prepared Piano on Tomato Records, #TOM-2-1001---Ed.).

Back in the 1960s, George Harrison brought the sound of the sitar to western ears with songs such as "Norwegian Wood" and "Within You Without You". At that time I was a fledgling guitarist who knew only three chords, but I knew I had to get that sound: the ringing overtones, the swelling envelopes, and the quivering pitch bends. I placed fingerpicks, and later, paper clips, on the strings near the bridge, but all they did was slide along the strings---resulting in a very inconsistent effect. Little did I know I had prepared a guitar.

Within the past six months, however, I have worked on developing the prepared guitar into a consistent and aesthetically pleasing instrument by adapting the techniques used for preparing a piano to the idiosyncrasies of the guitar.

PROBLEMS AND SOLUTIONS. First, most single pitches on a piano are produced by two or three closely spaced strings, thus making it a simple matter to wedge a screw or a paper clip between the strings. However, the relatively large distances between the strings on a guitar preclude the wedging of hardware between them. This necessitates a piece of hardware that clings to a single string. Small alligator clips (available from an electronic supply store) about one inch or less in length meet this requirement. Larger alligator clips will tend to pull too much on the string.

One of the most elementary, and satisfying, preparations for guitar can be obtained by clipping an alligator clip on the low E string at the 24th fret position (normally over the neck/rhythm pickup on a 22 fret guitar). Plucking the string while both lead and rhythm pickups are in operation yields a sound approaching Big Ben.

Muting the length of string over the fingerboard and plucking between the clip and the bridge produces F an octave above middle C.

Yanking the string away from the fingerboard bends the pitch up.

Muting the string between the bridge and the clip and plucking between the clip and the nut (over the fingerboard) produces A below middle C.

Another problem that faces the prepared guitar player is one of amplification. Because the piano is an acoustic instrument, even the most radical preparations (such as a person sitting on the strings inside of the piano) are audible. One solution is a transducer added at the peghead, which can then pick up every sound produced on a guitar without discrimination. Alternately, a conventional magnetic guitar pickup positioned over the strings around the third fret amplifies the tones produced on the strings between the clips (over the fingerboard) and the nut.

I have found that a hybrid system incorporating a transducer and a magnetic pickup offers the best solution, especially when using multiple clips fastened to a single string to produce three of more tones. By regulating the volume and tone controls for each pickup, including those already present on the guitar, the various overtones and pitches can be mixed to (cont. on next page)

(PREPARED GUITAR) provide the desired timbral effect.

Fred Frith, guitarist with the English improvisational group Henry Cow, is the most well-known practitioner of prepared guitar. His album Guitar Solos (Caroline Records #C1508; available through the Jem Records Import Catalogue, PO Box 362, 3619 Kennedy Rd., South Plainfield, NJ 07080, or 18615 Topham Street, Reseda, CA 91335) demonstrates the vast array of sounds available through the preparation of guitar strings without resorting to additional electronics.

My own experiments have been conducted with a Fender Telecaster, a Vibrolux Reverb amplifier, and eight alligator clips, which proves that an arsenal of studio effects are unnecessary for creative sounds.

Finally, the elusive sitar sound can be approached with two separate preparations. The first method involves weaving a strip of paper 3/4" wide by 3" long through the strings very close to the bridge. The second method involves clipping one alligator clip on each string immediately next to the bridge. Both methods produce reasonable approximations of the sitar's jangly sound.

Experiment and enjoy.

DECLASSIFIEDS

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

DEVICE would like to thank several organizations that helped us get off the ground by telling you about them...

SYNAPSE (2829 Hyans St., Los Angeles, CA 90026) collaborated with us on the ARP AVATAR review. If you're into synthesis, we recommend Synapse. Bi-monthly; $8/yr.

POLYPHONY (PO Box 20305, Oklahoma City, OK 73156) gave us lots of help and advice. This is a hardware-oriented magazine that emphasizes construction and modification of commercial equipment. Bi-monthly; $6/yr.

GUITAR PLAYER got us our first subscribers by printing our press release. GP covers all levels of guitar and includes material from interviews to electronics. Monthly; $18/yr. PO Box 615, Saratoga, CA 95070.

ELECTRONOTES (1 Pheasant Lane, Ithaca, NY 14850) is the newsletter of the Musical Engineering Group and offers a wealth of schematics and technical information each month. Write them for rates and a sample issue.

Anyone interested in creating a center for the Advancement of Electronic Guitar contact Buzz Kettles, 1714 San Benito, Richmond, CA 94804. Tel (415) 526-8304.

FET - abbreviation for fetish. For further information contact Jay Lee, 4517 Longridge Avenue, Sherman Oaks, CA 94804. Tel (213) 995-6869.

Bill Godbout Electronics sells parts of interest to electronic musicians. Send name and address for free catalogue to PO Box 2355, Oakland Airport, CA 94614 (for 1st class delivery send 41¢ in stamps).

Magazine back issues available (Popular Electronics, Guitar Player, Byte, Contemporary Keyboard, others). Send SASE for listing to Craig Anderton, PO Box 480, Clayton, CA 94517.

We sell everything: keyboards, guitars, pro sound, repairs, modifications. Custom pedalboards and effects accessories by Paul Rivera. Valley Arts Guitar, 12162 Ventura Blvd., Studio City, CA 91604.

(AMS-100) means whenever we strike a note, or chord, on the guitar. There are several ways to do this; let's look at the most common method.

The general procedure is to first feed the guitar through an envelope follower. This is a module that produces a DC voltage that corresponds to the overall amplitude of the instrument; i.e. harder plucking produces a greater DC output voltage. As a result, if you pluck a guitar note, the output of the envelope follower looks like fig. 1.

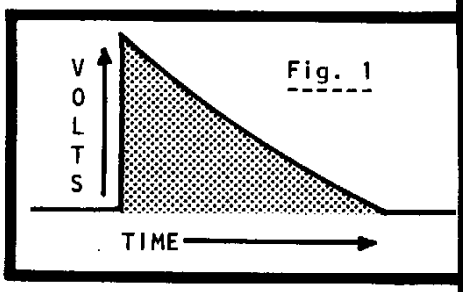


Fig. 1

This circuit feeds a comparator, which gives an output (turns on) when the envelope exceeds a certain threshold, and turns off when the envelope drops below this threshold.

The comparator output is called a "gate" signal, and is suitable for triggering synthesizer type envelope generators. As shown in fig. 2, with an attack-sustain-release envelope generator, the gate initiates the attack and sustain parts of the envelope. When the gate stops, the release section starts. An ADSR responds similarly, with the addition of an initial decay time and adjustable sustain level. When the gate stops, the release section starts. Often times, an additional trigger pulse will be available at the leading edge of the comparator output for additional triggering functions with ADSR type generators.

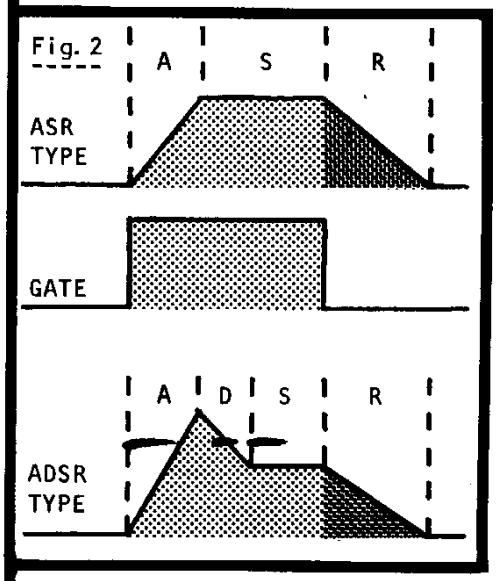


Fig. 2

This all sure sounds simple, but there are problems with this type of system that make it much more difficult to successfully implement than it would initially appear.

Problem #1 is that the envelope follower output will actually look more like fig. 3 than the ideal of fig. 1. We can see there is some rise time, i.e., even though the string starts vibrating immediately upon plucking, the follower takes a bit of time to respond. Problem #2 is called ripple. This means that the conversion of an AC guitar signal to a DC control voltage is not totally efficient, and some of the AC remains. This causes a problem called false triggering; if the comparator threshold is set as shown by the dashed line, then the output of the comparator will give two or more gate signals for just one plucked note. If the ripple is bad enough, it will trigger at an audio rate. You can overcome the ripple problem by adding hysteresis (producing noise immunity) to the comparator, however, this can reduce sensitivity and up the response time.

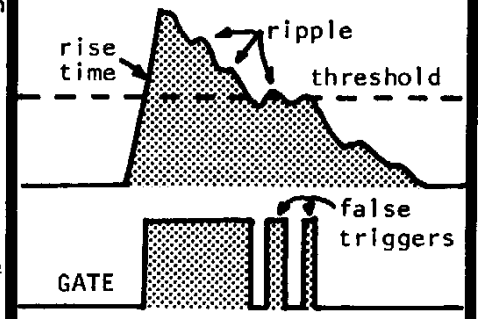
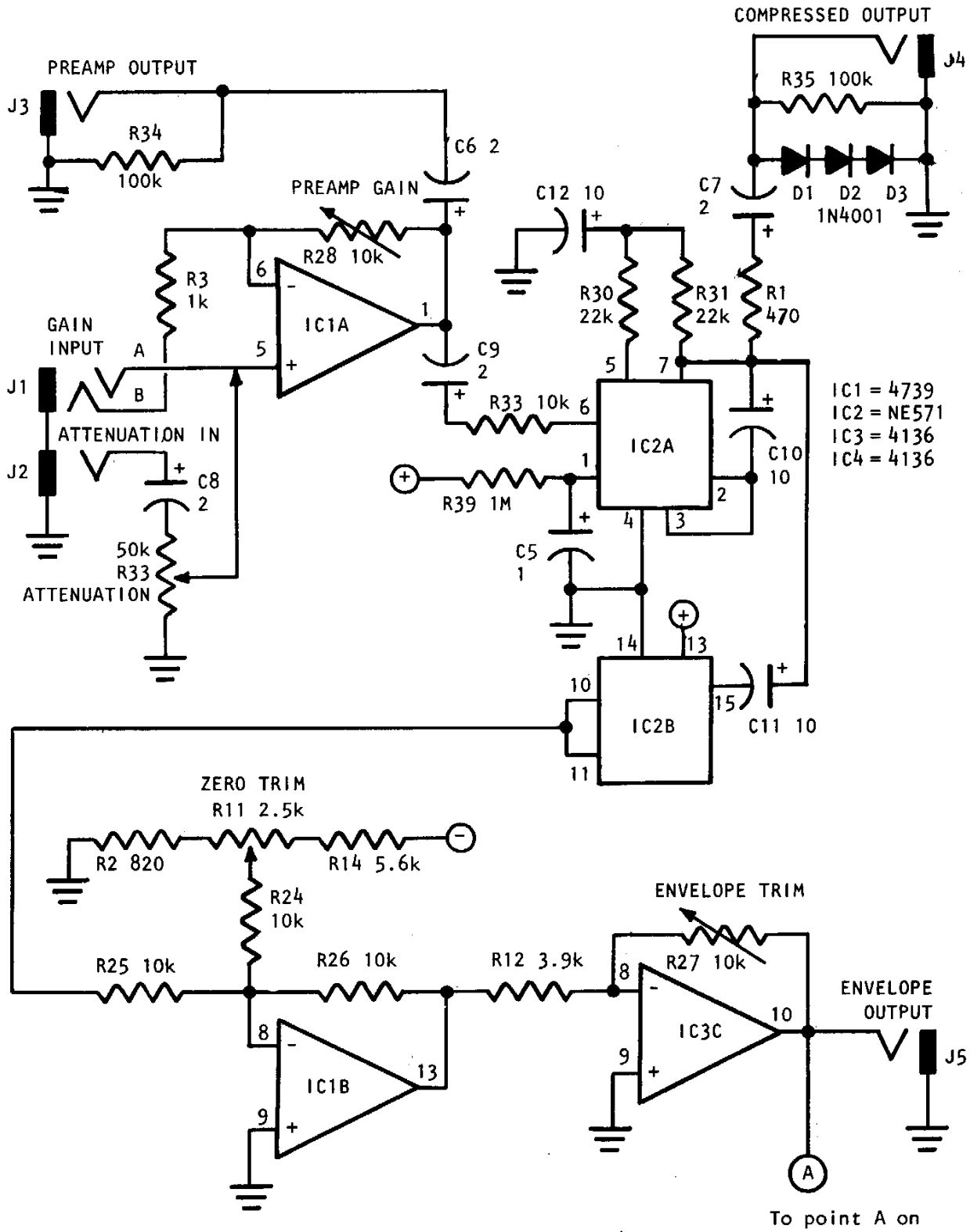


Fig. 3

The third major problem is that in order for the envelope generator to start a new cycle, the gate must shut off for long enough to reset the envelope generator. With some envelope generators, such as the Eu/SSM chips, you can retrigger the A and D functions by feeding in a trigger pulse, even if the gate doesn't change; while a definite improvement, this still doesn't solve all problems.

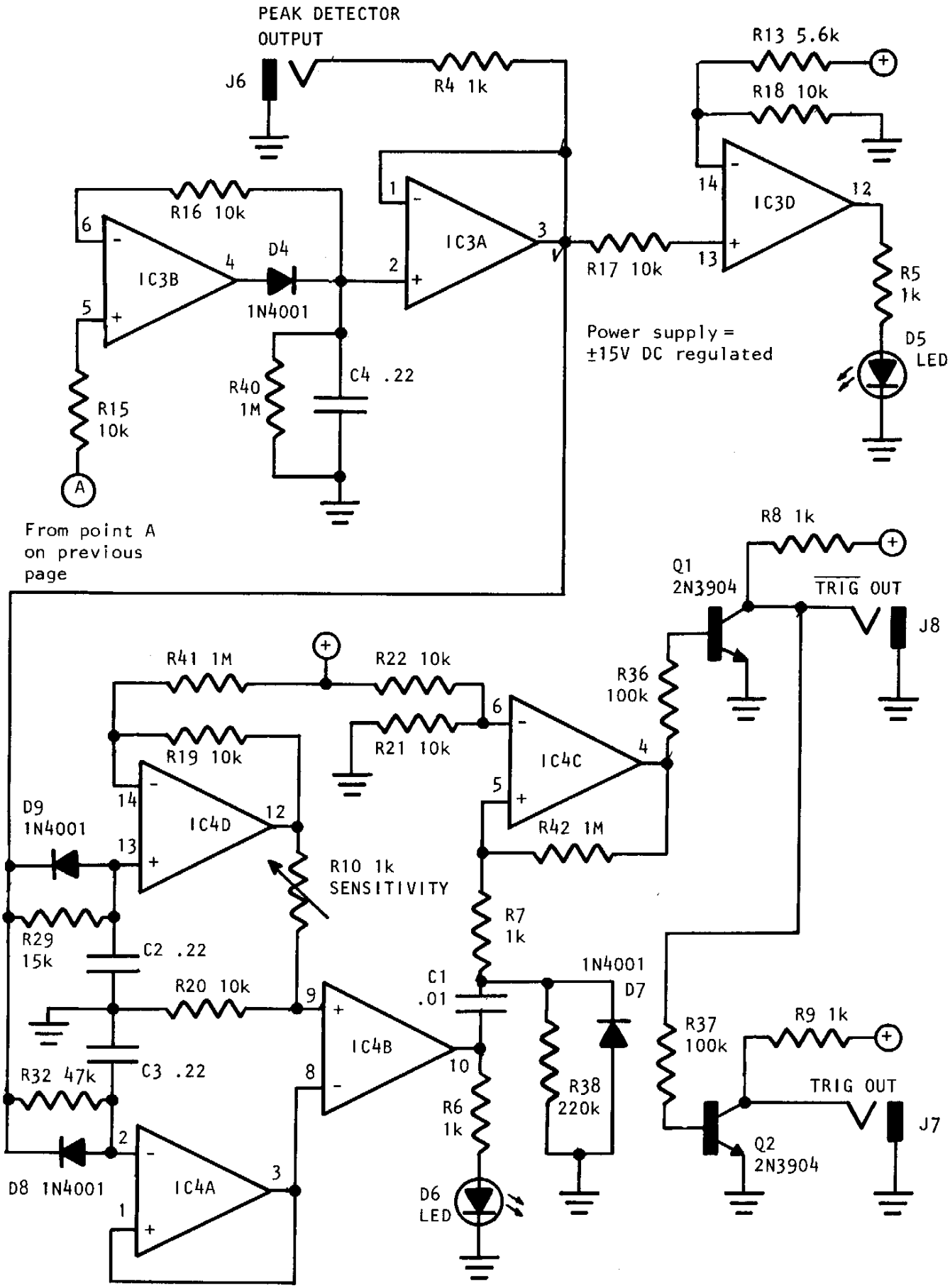
To complicate matters further, if you instantly mute the strings, it will take a finite amount of time for the envelope generator to return to zero output (the response time problem again). Finally, let's say you want to have an attack that goes from 0 volume to full volume over a certain period of time. If you're in the middle of the release section, the next attack cycle will not start from 0 but instead from wherever you were in the release section when the gate output stopped. You can improve the situation by leaving enough of a space between notes, but only at the expense of interfering with a natural style of playing.

While this system approach is usable, I wanted to come up with something that was as natural as possible, and required as little a change as possible in the technique of the musician---even if that meant limiting flexibility a bit. This is (cont. on page 10)



Power supply connections: Connect +15V to IC1 pin 14, IC2 pin 13, IC3 pin 11, IC4 pin 11, and all circles marked +.

Connect -15V to IC1 pin 7, IC3 pin 7, IC4 pin 7, and all circles marked -. Both supplies should be regulated!


PEAK DETECTOR
OUTPUT
J6
R4 1k
R13 5.6k
R18 10k
R16 10k
IC3A
IC3B
IC3D
D4
1N4001
R17 10k
R5
1k
D5
LED
Power supply =
±15V DC regulated
R40
1M
C4 .22
R15
10k
A
From point A
on previous
page
R8 1k
Q1
2N3904
TRIG OUT
J8
R41 1M
R22 10k
R36
100k
R19 10k
R21 10k
IC4C
IC4D
D9
1N4001
R42 1M
R10 1k
SENSITIVITY
R7
1k
R29
15k
C2 .22
C1
.01
1N4001
D7
R20 10k
IC4B
R37
100k
R9 1k
R32 47k
C3 .22
R38
220k
TRIG OUT
J7
R6
1k
Q2
2N3904
D8 1N4001
IC4A
D6
LED

(AMS-100) what led to the development of the AMS-100. I set the following design objectives:

1. A new envelope had to be created with every pluck on the guitar. An attacking envelope would have to start from zero under all circumstances when a pluck occurred.
2. No false triggering.
3. No sensitivity problems---usable at very low picking levels, or very hard picking.
4. Zero rise time. With our desire to derive a new envelope from every pluck, we've opened ourselves up to a problem that's not a consideration with traditional systems. Referring to fig. 4, let's say you pick two notes in succession and you want an "attack-that-sustains-at-peak-level" type of envelope. With our system, you get a new envelope for each note (as opposed to the conventional system, where you'd only get one gate output, and thus envelope, because there's no pause between notes). But note that because of the response time, shown exaggerated for clarity, the envelope doesn't start until a fraction of a second after the note commences. In the case of the first note plucked, this presents no problem. But in the case of the second note, an "attack blip" from the second note is tagged on to the end of the first note's envelope, and only then does the attack start. So what we hear is a very brief (the length of the response time) blip from the second note, then the attack begins and all is well again. But that blip sounds terrible!
5. Multiple outputs for a variety of functions.
6. Operation at nominal 0 VU levels.
7. Nothing too expensive, complex, or liable to go out of adjustment.
8. Compatible with any type of guitar, and most other monophonic sound sources.

Well, that's a pretty stiff set of conditions, and I didn't meet them all...here's the rundown:

1. This requirement turned out pretty successfully. There are still some conditions when the thing won't retrigger with a new pluck, but that's fairly rare. You can strum a chord, for example, and once you've got the knack of the system you'll get a trigger output for each pluck of each string. You can play pretty fast with the thing and get triggers out, too. While you do have to play a little more attention to your playing style than normal, it's really pretty responsive.
2. Yes and no. Open strings mess up the works, but that's the only majoy problem. If you hit a barre chord, you'll get a trigger out at the beginning of the chord and 95% of the time it won't retrigger after that if you keep your fingers still. Once you get good at it, you can hit open strings, get a trigger, then while those strings are still ringing pluck a few more notes and get more triggers for those notes to initiate new envelopes.

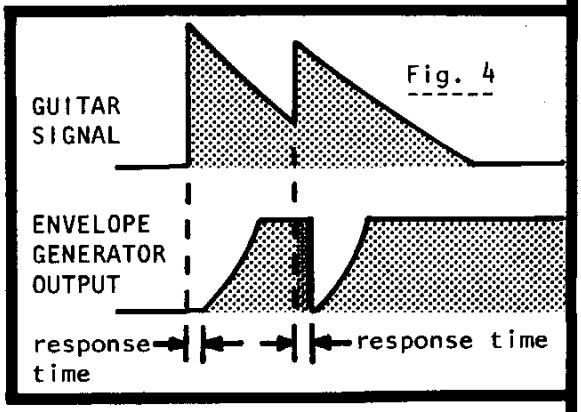


Fig. 4

3. This requirement worked out very well. It's sensitive enough so that glissing your hand along a string or hitting a string will create a trigger; but if you're hitting pretty hard and hit harder, you'll <u>still</u> get another trigger out. Problems happen more at the loud end of the range than the soft, so if you're a delicate to moderate picker you'll like the response a lot. For hard pickers, there's a sensitivity control to accomodate that type of playing style.
4. I tried to reduce the rise time to an absolute minimum, but still couldn't get it down much below a couple of milliseconds---and you can hear even a single millisecond as an objectionable pop under the conditions described under (4) above. This really stumped me (to say the least!!) until I came up with a somewhat sneaky solution: delay the guitar signal a bit before sending it to any of the other processors controlled by the envelope generators. So, if the trigger has a response time of 3 milliseconds, you simply delay the guitar signal by an equal amount of time so that the guitar pluck and trigger output are coincident. Sounds simple, but unfortunately, the rise time isn't constant at all levels and frequencies, which screws things up a little bit; but the added delay makes the "pop potential" go from unacceptable to much-better-than-acceptable. We can also use this delayed output signal for other fun and games---like running another delay line in parallel to get flanging effects that go from no delay (cancellation) all the (cont. on next page)

(AMS-100) way down to 100 or 200 milliseconds of delay.

5,6,7: All of these conditions weren't hard to meet and will be discussed later.

8. I've tried batches of guitars with the thing and it works...no special pickup, cord, or anything else is required. There's no way I can test everybody's axe and playing style, but I'm pretty confident about the circuit. It particularly likes series connections of pickups.

HOW IT WORKS. IC1A is ½ of a 4739. J1 is the gain input; when used with a mono plug, R3 shorts to the ground of the plug. This turns IC1A into an amplifier (if the input jack is wired up correctly!), and the input signal goes directly into the (+) input. This input is intended for use with a standard, non-preamped guitar. R28 controls the gain, from X1 to X11.

If a signal is plugged into J2, the attenuation input, then R3 is no longer connected to ground---turning IC1A into a unity-gain buffer. R33 controls the attenuation. So between these two inputs, you can either amplify a weak signal, or attenuate a strong one, in order to condition the input to a nominal 0 VU signal level.

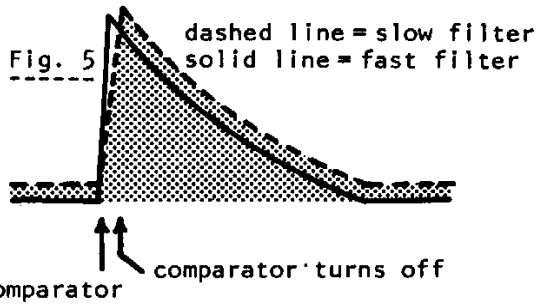
From here the preamp feeds a compressor, IC2A, made up of ½ an NE571 compander IC. This compressed output is compressed in an exact 2:1 ratio, and the signal present at J4 may be used to feed the delay line. Note that by utilizing 1:2 expansion at the output of the delay line, we can restore the signal to normal range and attenuate the delay line noise (tra la). Or, you can just use the compressed output as an audio signal. Diodes D1 - D3 clip a nasty spike that happens when a strong signal hits the compressor after a period of no signal. Compression of the signal makes it much easier to extract consistent trigger signals further on down the line.

The other half of the 571 is set up as a full-wave rectifier (FWR) that is the first step in deriving an envelope follower output. However, the FWR output is not ground referenced, nor is it strong enough. IC1B sets the ground reference via R11, and IC3C sets the overall envelope output (we'll cover these trimpot adjustments later). The FWR output is also available at J5. It can be used to drive traditional envelope follower filter sections and gate extractors, or used as an audio signal (with guitar, it produces a unique type of octave-doubling fuzz---especially on the higher part of the neck when using the rhythm pickup).

In our system, the FWR output goes to a peak detector, IC3B. This is preferred over an envelope follower because of the faster potential attack time. R40 and C4 smooth out the decay part of the envelope for minimum ripple. Additionally, a level-setting LED taps off this output whose use will be explained later.

The next circuit block is the pluck detector, comprised of IC4A, IC4D, and IC4B. The peak detector output feeds two filters, one of which is "fast" (IC4D), and one of which is "slow" (IC4A). The outputs of these filters feed comparator IC4B. R41 offsets the fast filter so that with no input signal, pin 3 of IC4A has a higher voltage than pin 12 of IC4D which means that IC4B is turned off. When a pluck occurs, here's what happens. The fast filter output hits a high voltage almost immediately, while the slow one takes a while longer. So, the comparator senses that the fast filter output is higher than the slow filter output, and turns off. But, because the signal coming from IC4D to the comparator is attenuated somewhat by R10, as soon as the pluck is over the slow filter climbs over the fast filter and turns the comparator off (see fig. 5 for summary). Since the period of time during which the comparator is on varies, in order to get a nice, clean, thin pulse, we take the comparator output and feed it to differentiator IC4C. This IC delivers a positive going spike that Q1 turns into a "not trigger", or in computerese, trigger. In other words, this output is always high or on, and turns off when a pluck is detected. Q2 inverts this spike to create a trigger signal that is normally off, but which goes high for a millisecond of two when a pluck occurs. It would be nice to monitor these spikes with an LED, but they're so short (cont. on next page)

Fig. 5



(AMS-100) they're very hard to see. So, we connect our monitor LED to the output of comparator IC4B so we can see when a trigger occurs more easily.

CONSTRUCTION. The AMS-100 requires a regulated, ±15V power supply, and input leads going to J1 should be shielded.

There are a lot of time constants in this circuit that are critical. C5 sets the compression response and is a tradeoff between distortion and speed of response. 1 uF seems just about right, although 2 uF is OK...but not preferred. R40 should be a 5% resistor, and C4 must be an accurate mylar capacitor. These set the decay time of the peak detector. Too short a decay time gives false triggering; too long reduces sensitivity under hard picking conditions.

R29/R32 set the attack time constants of the filters and should be 5% resistors. C2/C3 are also part of this process and should be close-tolerance mylar capacitors. R41 determines the initial offset of the fast filter. This sets the output of IC4D just below IC4A with no signal input; if the thing is too sensitive to your initial pluck, you can lower R41's value to 470k. I like it pretty sensitive, myself.

The above mentioned parts are the critical components in the operation of this module. They represent tradeoffs of sensitivity, false triggering, etc.; naturally, all compromises are somewhat subjective. If you get the chance, try breadboarding this circuit and experiment with these different time constants to see if you can come up with something that is exactly matched to your style of playing. The circuit as it stands is calibrated to my playing style, which incidentally is pretty normal except that I hit the strings a bit harder than some people do.

Another tip: I mounted IC1 and IC2 on one board, with IC3 and IC4 on another board. That way if the pluck detector section, or input module, get revamped or improved at a later date, one may be replaced without trashing the other.

MAKING IT WORK. Plug your signal source into whichever input is appropriate (remember, you can only use one at a time, not both) while monitoring the preamp output jack with an AC VTVM, VU meter (like the kind Radio Shack sells), or a calibrated tape recorder input. Adjust R28 or R33 for a signal that averages 0 VU but doesn't go over +3 dB on peaks (this will make the compressor happy).

Next, monitor pin 13 of IC1B with a sensitive VOM or scope and, with NO SIGNAL APPLIED at the input (turn down the volume control on your instrument all the way), adjust trimpot R11 for a reading of exactly 0 Volts. Now turn your volume control back up (the preamp output should be back at its nominal 0 VU now). Adjust trimpot R27 so that D5, the level setting indicator, flashes on peaks of your playing. This results in a peak detector output that goes from 0 to +10 VDC, making it compatible with ±15V synthesizer systems (for 0 to +5V systems, add a couple of equal value resistors or a trimpot to cut the output in half). You can use this output as a sort of envelope follower control signal.

Now is the time to set R10. I suggest putting it initially at the half-way point. For more sensitivity, decrease the resistance; for less sensitivity, increase the resistance. Increased sensitivity gives the tradeoff of possible false triggering, but makes the thing really respond---even to ultra-fast leads and rapidly strummed chords. Chances are you'll probably leave the control at a compromise setting, and crank it to less resistance for special passages. Remember, you need a little resistance in there or you'll have false triggering problems; you might want to add a 100 Ohm resistor between R10 and IC4D pin 12.

So, now we have a trigger for each pluck, which we'll use to trigger the AMS-100 envelope generators slated for next month. These envelope generators are SIMPLE because I want to have lots of them in the system for controlling large gobs of stuff inexpensively...but they still do the job nicely. Note that our system is not compatible with ordinary, keyboard oriented envelope generators, although if there's a hue and cry I've got a standard type circuit for doing that and it can be published at a later date.

One more thing: about that delay line required to prevent popping problems...we'll be covering delay lines in the months ahead, but for now, you can use a flanger-type device set for no modulation and maximum delay. Then, experiment with the amount of delay time and use the minimum amount of delay consistent with good results.

Well, it's goodbye typewriter, hello lab, and time to finalize the envelope generators. If you have problems, questions, plaudits, or suggested improvements, don't be shy---tell use about it! Catch you next month.

BASICS: EVALUATING EFFECTS

by Gary Kirkpatrick

This is the first part of a two-part article. Part 1 will cover general effects evaluation methods to help a musician who is in the market for purchasing an effects device. Part 2 will get into troubleshooting methods to help any unfortunate people who get stuck with a bad effects unit.

For this first section, let's assume that you are a musician who is interested in buying a couple of effects to enhance your playing. Make sure you patronize a music store that you feel offers the best combination of price, service, and selection. The best way to judge the service reputation of a store is by your own personal experience and the experience of other musicians who have purchased equipment from the music store in question. If a store with a real good service reputation has somewhat higher prices, pay the couple extra bucks---it could really be worth it later on. By the way, don't be pushed into buying anything by a salesman. Use these testing methods and decide for yourself what effects you like the best. When you walk into a music store with the intention of possibly buying an effect, ask them to provide you with a guitar or whatever, amp, patch cords, and the effects you want to try. Don't be afraid to ask for more than one unit; in fact, it's easier to lay your requirements on the salesman all at once so that he doesn't have to run back and forth grabbing effects off of the shelves. The salesman will leave you alone and won't give you the "sales pitch" lines if you appear to have a good working knowledge of effects and their applications.

Here are a couple of general things to do when testing any type of effect; I'll get into a more specific breakdown of some of the most commonly used characteristics later.

CHECK ALL CONTROLS ON THE UNIT. Pots: Rotate them to see if they have a smooth action and are not scratchy. Footswitches: Check them several times to make sure the bypass mode (i.e. the no effect position) works, as footswitches tend to fail more often than some other parts. Also, check them with the amp up loud and listen for a pop when you engage or disengage the footswitch. Don't play when you check for the popping noise, simply because it's easier to hear the pop if you aren't playing. You will find a lot of lower priced effects have this popping noise so you have to determine for yourself what is acceptable and what isn't. This pop could be very annoying when playing live or recording, especially if you like playing with a lot of reverb. Switches: Make sure they actually do what they say they do. Also check all switches for popping noises, as described above; however, these noises are not as critical because you usually don't change effects switches while you're playing. LEDs: Make sure they relate to the proper controls. Sometimes wires get switched around by mistake and this causes the LED to light up with a different control function than that which was originally intended. Noise: Test for this the same way you check for pops, turn the amp up loud and don't play. Make sure the effect is in the effect mode; if the effect is in the bypass mode, you will bypass the effect circuitry containing the noise. Most effects have some degree of noise, so you'll have to determine for yourself what you can live with. When testing for noise, make sure the noise you hear is coming from the effect and not the amp. Do this simply by unplugging the input to the amp, then listen to the degree of noise coming from the amp.

To summarize all this, take into consideration the quality of the parts used, the usefullness of the controls included with the unit, and the degree of noise and pops associated with the unit. These factors usually determine the price of the effect, so it's up to you as a consumer and a musician to determine what your needs are and what you can afford.

So much for the general stuff...now I'd like to get into a more specific breakdown of different types of effects and their characteristics. I'll start out with the simplest device, and probably the most popular among guitarists. (cont. on next page)

(continued from previous page)

1. FUZZ BOX (OR DISTORTION UNIT). The purpose of this type of effect is to add an overdriven sound to a guitar. It's not a particularly complicated device, but different makes have different characteristics. A few key factors to consider when buying a fuzz unit are:

--The versatility of distortion control. It's nice to be able to get a high bright sound, or a warm mellow fuzz.

--Sustain characteristics and control. You should have the flexibility to choose from no sustain to a lot of sustain.

--Feedback. This is something some guitarists want and others don't. I personally do not like it but some guitarists like to toy with it, which can be interesting if done right and assuming no one's hearing gets damaged in the process.

--Noise. This type of effect tends to be very noisy, so watch carefully for differences when comparing different units.

2. PHASE SHIFTER. This device gives you a sweeping type of an effect that's useful with guitar, keyboard, drums, and my favorite, string synthesizer. A few things to consider when buying a phase shifter are:

--Sweep. Check for a smooth and even sweep from high to low frequencies. By this I mean no "glitch" points where the sweep jumps abruptly from one point to another.

--Rate of sweep. Different units offer different sweep ranges. Ideally, you'd want to be able to go from a very slow sweep to a very fast sweep for "bubbly" effects.

--Feedback. This is an additional function found on some units (also called "resonance"). It changes the sound of the phase shifter to a peakier, more filtered sound that is more intense than regular phasing.

--Depth control. Another additional function that varies the subtlety of the sweep effect, from a light kind of shimmer to a very deep sound.

--Noise. These units again tend to be noisy, so listen carefully.

3. FLANGER. Another sweeping effect, but it creates a more versatile and thicker sound than a phaser. These units tend to have the same type of controls as a phaser, however, you might keep an eye out for flangers that include companding and noise gates as these help to keep things relatively quiet. Also, the "depth" or "width" control should be able to give very subtle effects (like a phase shifter) or very dramatic effects. New types of flangers are being introduced every day, so look into the market carefully before committing yourself to a particular model.

Before moving on, let me describe a good method for testing the sweep of a phase shifter or flanger. Turn the guitar up full, the amp to a moderate listening level, and the slowest possible sweep. Strum the guitar once and listen to the full cycle of the sweep. This sweep should be a gradual succession of low to high frequencies with no jumps or skips. It should reach a certain high point at which it starts back to the low point of the sweep cycle. Some phase shifters have very irregular sweeps, which can be really distracting.

(to be continued next issue)

DEVICE: NEXT MONTH

NEXT ISSUE: Craig Anderton reviews the Intersound "Instrument Voicing Preamp", and assesses their component approach to assembling a guitar amplification system. Buzz Kettles and Roger Clay start a series of articles on pedalboards...problems, implementation, and the advantages of hex processing. We'll be having a special interview on the subject of time delay processing, as well as info on building the AMS-100 envelope generators.

FUTURE ISSUES: Watch for reviews of products from A/DA, PAIA, and Electro-Harmonix, as well as in-depth reports on the Zetaphon guitar synthesizer and Marshall Time Modulator... along with our regular features, technically-oriented interviews, and news of new components of interest to electronic musicians.

OPINION POLL: EFFECTS BOXES

One of the reasons for distributing the first issue of DEVICE free of charge is to obtain as many replies as possible to this poll. Please take the time to answer the following questions; it will help us and your fellow readers. We suggest you photocopy this page or write the question numbers, along with your answers, on a separate sheet of paper.

Your opinion is vital...thank you very much for your help. Results will appear in a future issue of DEVICE.

#1. Are there any effects boxes for either studio or stage which you feel represent either exceptional value or offer exceptional performance, and which are they?

#2. Are there any effects boxes which you feel are particularly shoddy in quality of performance?

#3. Is there any one manufacturer that you feel deserves a reputation for reliability and quality?

#4. Is there any one manufacturer that you wish would disappear from the face of the earth?

#5. I feel that specifications published in the majority of ads are:
Very helpful_____ Slightly helpful_____ Meaningless_____

#6. The qualities that are most important to me in a device are (enter 1 for very important, 2 for somewhat important, 3 for not important):

Low power consumption/energy efficiency_____
Easy availability of service _____
Low cost _____
Durability of packaging _____
Availability of AC adapter _____
Balanced line inputs/outputs _____
Low noise _____

#7. Are there any companies with which you have had good experiences in getting a piece of equipment serviced?

#8. Are there any companies that have given you the runaround when getting something serviced?

#9. How important do you consider standardization of effects with regards to signal levels, input/output impedances, methods of determining specifications, and the like?

Absolutely vital, hope it happens tomorrow _____
Nice if they get around to it, but I'm OK as is _____
Standards are unnecessary, and may actually stifle creativity_____

Please feel free to add any additional comments, or copy versions of this poll for friends to fill out. Send completed poll to DEVICE OPINION POLL, PO Box C, Carmichael, CA 95608.

ADDITIONAL COMMENTS:

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