DEVICE

the newsletter for the electronic guitarist/musician VOL.1:3-79

DEVICE INTERVIEW:

WHO IS STEVEN ST.CROIX AND WHY IS HE SAYING THOSE THINGS ABOUT DELAY LINES?

This is the first part of an interview with designer/innovator Steven St. Croix, V.P. in charge of Research and Development for Marshall Electronics, 1205 York Rd. Suite 14, Luterville, MD 21093 (not to be confused with the amplifier manufacturer). Our interview was conducted in late '78 by our East Coast correspondent, Pete Johnson, at Steve's recently completed 24 track studio tucked away in the rolling hills outside of Baltimore. A man of many talents, Steve leads a busy life apart from his work with Marshall. A working musician, he has done extensive studio work with, among others, Stevie Wonder. His current projects include completing work on albums of his own, the ultimate in-car sound system, audio for a new film, and the rebuilding of various sports cars. His other passions include scuba diving and Chinese food. He also claims to have the fastest car on the road.

Pete: *What's your background, and how did you get into the electronic music field?*
Steve: Well, about 14 or 15 years ago I was living in Europe and playing rock and roll. I was working with some people in England trying to establish ways to get new sounds, but not new sounds as in playing convoluted chord changes; new sounds as in new audio effects. At that time there just weren't any special effects devices happening in music. A lot of people had Echoplexes, and that was great...but then people started finding out that if you played real loud, it sounded better than if you played soft and that the overall (*Cont. on page 8*)

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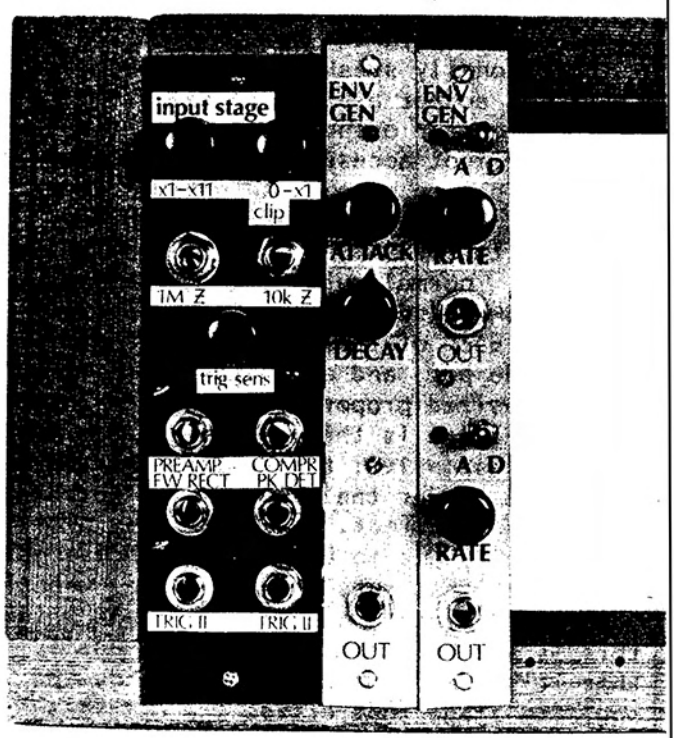
construction: BUILDING THE AMS-100 - part 3

by craig anderton

POWERING AND PACKAGING: Before getting into next month's VCA module, I thought it might be a good idea to discuss packaging and power supply considerations for the AMS-100 modules; let's begin with packaging.

Figure 1 (below) shows the input module from DEVICE #1:1 along with the envelope generators from DEVICE #1:2, built up

(*Cont. on page 11*)



(*Cont.*) sound had a little more balls. I was working with someone from Vox at that time, they were just coming out with the "Super Beatle" amps, so that tells you what time it was. Anyway, there were some people playing in the area I was in who were trying to get something a little more aggressive-sounding, and we had this tape recording that was brought to us by Jeff Beck. He told us he wanted his guitar to sound like it did on the tape; what he showed us was one of those old tape machines with the magic eye level indicators and he said, "if the eye is closed all the time, full on, it sounds great". What he was doing was putting the output of a preamp into the mic input and blowing the Hell out of the front end of the machine. It was, of course, distorting. From that came the Arbiter "Fuzz Face" and the Vox "Mini Distorter"...but the "Fuzz Face" was the one that took off. That's how it all started: in Europe, with effects that gave you a little more sustain and a little more balls on stage.

Because I was a working musician, I could tell right away if an effect was musically useful or not, while a lot of other people who were doing engineering were engineers only. They could come up with all kinds of hot stuff...but it was stuff that didn't necessarily apply to music. This is still a problem today; you've got musicians and engineers in a lot of companies, but they're usually not the same people. The engineer may design this really beautiful thing but in a real performance situation it has little musical value...you'll notice this a lot in console equalization modules. Anyway, our experimenting went on and we kept developing more toys. During all this I had been doing medical electronics, going to school in Switzerland, and playing music full time. It was a pretty packed period of time, so it didn't really occur to me to do gizmo manufacturing. I was doing designing and design licensing for other people, and that seemed enough. It stayed that way until about 4 years ago.

P: Was that about the time you developed the Marshall Time Modulator?

"(ENGINEERS) COULD COME UP WITH ALL KINDS OF HOT STUFF... BUT IT WAS STUFF THAT DIDN'T NECESSARILY APPLY TO MUSIC. THIS IS STILL A PROBLEM TODAY; YOU'VE GOT MUSICIANS AND ENGINEERS IN A LOT OF COMPANIES, BUT THEY'RE USUALLY NOT THE SAME PEOPLE."

S: Yeah. Remember, this was before there were any flexible time-based manipulators. A friend and I were having a discussion about manipulating time and the overall ramifications as far as communications was concerned. The conclusion we came to was that it would be a lot of fun to figure out how to do audio time modulation and hear the results. I had four days before I was due to go out on a gig, so that night I sat down and was determined to show my friend that I could back up all my outrageous statements about time modulation. I had been recently doing work with charge transfer systems, doing a lot of R & D, and I decided to incorporate this research in the design of this new idea. I stayed up all night, really cracked on it, and by lunch the next day I had a working prototype. It was literally an overnight device.

P: Could you explain "charge transfer systems"...

S: An analog system of clocked delay. I knew we had to go analog on this idea, and the only analog system that could be practical would be a clock transfer system with a FET (field effect transistor) and a capacitor. The cap would remember the voltage, and be switched to the next cap by a clock. We didn't want to take a digital approach, because the inherent limitations would have eliminated many of the possibilities I was heading for.

P: Is charge transfer similar to bucket brigade?

S: Well it's similar, but there were some differences.

P: Back to developing the prototype...

S: Well, the idea was to get a transfer system working, which I did. It was incredibly unwieldy and super-touchy, but it worked. The afternoon I'd gotten it working I had a session in the studio, so I took the device along to try on a song I was doing. There were a bunch of folks hanging around including the studio owner and I said "I've just got-

ten this thing working, you should listen to it". I'd never really tried the thing out with music, I'd just talked through it...I'd say "hello" and the thing would go "whuaguaula". I thought, "wow, this is great!" It had a dynamic range of about 1 dB and a bandpass of about 1 Hz, not real impressive in terms of performance! Conceptually though, it was a mind-fucker. I brought it into the studio and played with it a couple of minutes while I was working on a Clavinet part...those couple of minutes turned into hours. It wasn't intended to be a product; it was made to win a bet, quite frankly. But as we played with it, it dawned on us that this was an audio product that could really be useful.

In the meantime, the AES show was a couple of days away. Now I'd never even heard of the AES, let alone gone to an AES convention. But the guy who owned the studio suggested we make the thing look a little more realistic and take it to the show in Los Angeles, rent a booth, show it to some people, and see what happens. That seemed like an interesting idea, so I went home and went crazy getting the thing ready. Did a silk-screen front panel, added something like 8 times the number of delay elements to bring the bandpass up, and some other stuff. We packed the whole thing up in a paper box, we didn't even have a chassis; printed some real fast brochures at a jiffy overnight printer, rented the smallest booth we could find and walked in cold. I mean, I had hardly even heard of the AES and here I was showing at their convention. I had no fixed idea of what to do, I just wanted to show people the toy and see what they thought. It was still light years away from production, but it was an interesting concept. So we showed it...and all Hell broke loose. It just went crazy, the booth was constantly crowded. People were telling other people to come in and see it...it was packed! Well, we walked away with two years worth of orders. And it took those two years to fill 'em. We decided not to ship them until we had finished developing this machine into a truly professional device.

We got a lot of yelling and screaming for pulling the thing off the market during this period, but I had been looking at the marketing mistakes of other audio products and came to realize you have to make a basic decision with a new product---let it out like it is and try to make a whole bunch of money, or wait and make it absolutely the best you can and then let it out into the world. You may lose some money or customers at the start, but the theory is it will really do you good in the end because even your oldest products will be really excellent. We ran several hundred prototypes - the PT series - they worked great, but they were still prototypes. They were touchy and didn't have the Kepton flex circuits we use now; they didn't have the heavy weight metal and welded studs, but they were very similar in appearance and performance to the ones out now. Those were the one you'd call the original models; those were the ones Leo Kottke and Stevie Wonder got.

P: How were they different from the ones out now?

S: They only had one delay line instead of two, they only had a sweep rate of 40:1 which is now 72:1, the dynamic range was only 85 dB - now we spec 95 dB unweighted at the longest delay. They didn't have custom pots, the metalwork wasn't strong, things like that...we've put a lot of effort into making a more roadworthy product.

P: What is the basic principle of the Time Modulator's operation?

S: There are two main families of ways to delay; digital and analog. Due to developments in the past year digital now falls into two basic categories: conventional digital, where the signal is converted by an analog to digital converter to a bunch of numbers. These numbers are then clocked down a series of delay lines. If it's, say, a 10 bit digital line then there are 10 parallel delay lines in the device, whether it be RAM (random access memory) or shift register. Each line passes a bit down the line in parallel. At the end, or in the middle if you want less delay, the line feeds a digital to analog converter and is translated back into analog. That's conventional old telemetry digital that they've been using forever. The advantages are: if you want to spend the money, you can get as much delay as you'd like without degradation of quality because digital is a regenerative process. If a bit of information starts to get "sloppy" it doesn't matter, because a bit is either "on" or "off"; the next time it's transferred to the next stage, it sharpens right up again. So there is no transfer loss in digital. There are dropped bits and mistakes, but there is no necessarily cumulative loss in digital, which is the big advantage of digital technology. If you want 5 seconds of delay, you just whip right

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(Steven St. Croix cont. from page 9)

out and buy yourself 5 seconds of digital delay line. But there are also disadvantages. Digital lines are expensive, they run hot, they can't be swept over a wide range, and it is a bitch to generate a lot of the special effects you can theoretically do in analog. And, digital lines sound digital; when you convert in and out of digital you only have so many numbers or steps. Remember, the digital line doesn't remember pitch, it remembers samples of amplitude...hopefully at least 15,000 samples per second to get a realistic picture of the waveform. To get an anywhere near accurate sample of a signal you need to have at least twice as many samples as the highest frequency, so you're talking about some pretty high clock rates. What actually happens is this: the delay line takes a bunch of samples and looks at a waveform. There are certain numbers of levels it can remember, and if the amplitude sample falls between two of those levels the A/D converter averages and makes a decision. As an example, when the signal is bigger than level 500 and smaller than level 501 but it's closer to level 500, the converter rounds it off to level 500 and that's transferred down and later converted back to audio. Now if the signal happens to be exactly 500 that's great, but if it happens to be 500.4 then there's an error. Note that it had to make an error; it didn't make a mistake like dropping a bit or something, it made an error it's designed to make. Now, if you have 16 bits of computer information defining the amplitude of a signal you have much smaller increments, so it can be much more accurate...which is part of why 16 bit lines are in the majority.

To sum up, a signal is averaged up to a bunch of numbers; these numbers are transferred down the memory bank and then converted back to audio again. This rounding off is called quantizing. Let's say you have 15,000 samples of a 1 KHz sine wave; that's 15,000 samples per second, and you're remembering 15,000 instantaneous amplitudes. If, after being delayed you put all those samples back together in the right order you get your sine wave back again. The fact that the analog to digital converter rounds off, though, means that it makes errors and this is what quantizing error is all about. Digital delay lines have this error, you can hear it, and this is one of the big problems with digital delay.

"WE LIVE IN AN ANALOG WORLD, A WORLD THAT DISTORTS THINGS IN A FASHION THAT MAKES SENSE...THEN CAME DIGITAL WITH DISTORTION PRODUCTS THAT DON'T MAKE SENSE, AND WITH REVERSE DISTORTION LAWS, SO IT'S NO WONDER DIGITAL SOUNDS JUST PLAIN WRONG TO A LOT OF PEOPLE."

P: What does this error sound like?

S: This is of course very subjective; sometimes, it's a very annoying "gricky" sound. People who have used digital delay for a long time realize that you get a gritty kind of coarse sound. If you operate a digital line at 0 VU with a full signal sweep, you're using the majority of all those little samples and things aren't too terrible. This will give you usable distortion levels at full amplitude levels. But as the amplitude goes down in digital delay lines, the distortion goes up...a very unnatural effect. The reason the distortion level goes up is because you have your signal excursions over less of a step so each amplitude step becomes coarser and coarser until at the end, in many digital delay lines you can actually hear it going (makes sound similar to that of a sputtering Woody Woodpecker) as it quantizes the relatively giant steps towards the end. There's also quantizing noise, which is the switching transient noise that's transferred down, and little errors at the bottom that we hear as kind of a grit. What happens in a digital delay line is as you approach lower levels you literally approach 100% distortion...and, this distortion is not particularly pleasing. When you splatter a digital delay line you don't get harmonic distortion; you get bastard harmonics, harmonics that are the product of the input information's harmonics plus the clock frequency. For example, if you're sampling at 40 KHz and you splatter a 5 KHz tone, you get a 10 and a 15 KHz harmonic...but it's not just that simple. You get 10 and 15 KHz multiplied and divided by 40 KHz, the clock rate, which gives you a totally unnatural series of sideband harmonics. Now, you can accumulate a great deal of "natural", harmonically related distortion and it will sound acceptable, since your ear has grown up with harmonically related distortion. Your ear has also grown up with another law: the louder it is, the more distortion...whether it be your ears distorting or the equipment. For example,

if I play a record and it's distorting 5%, you'd go "that's really loud". So if I say "OK, I'll turn it down", then go over and turn a knob that takes only the distortion out you'd probably say "that's much better". You'd have to be very well trained to realize that I have not done anything except turned down the distortion...you correlate distortion that much with amplitude. We live in an analog world, a world that distorts things in a fashion that makes sense. Wellll...then came digital with distortion products that don't make sense, and with reverse distortion laws, so it's no wonder digital sounds just plain wrong to a lot of people. Now, with the new PCM tape recording systems and people going to 16 bits and doing floating point conversion it's getting fancy enough so that all these awful things we just talked about have to be weighed against the inherent awful qualities of tape...but that's a whole other subject.

There's also delta digital. Delta is kind of a cheating system where everything is put through in serial; information is not converted to a number of parallel digital bits. There's one delay line and all the information is spit out in serial fashion. I won't go into Delta too much except to say that with the present technology it's extremely difficult to push Delta to anything that even resembles professional quality. Delta lines can't slew, and that's the key: you can't get things to happen fast enough to make delta work under true professional conditions. There are some advantages to the manufacturer with delta, though; for example, it's a snap to convert from digital to analog in delta. Since delta is a series of numbers formatted so that they come out as steps of amplitude you hang a smoothing filter on the output and bang---you've got audio. This is great for the manufacturer because it's super cheap to make. But it's still digital and it's still subject to digital error, along with some of the worst characteristics of analog...high frequency problems and the like. OK, now I've slammed around some of the competition, on to the analog delay lines. There are two types of analog lines: bucket brigade and a new type of CCDs...

P: CCDs?

S: Charge coupled devices, bucket brigades being BBDs. BBDs are pretty simple to use; they can also be swept over a pretty wide range. But frankly, the performance disadvantages are so severe, they cause many, many problems...BBD delay lines are extremely noisy; all kinds of noise gating and companding concepts have been designed to try and quiet them down, but they're still noisy. It's also unrealistic to try and achieve very long delays with bucket technology. And, as you increase delay you lose bandwidth; i.e. the longer the delay, the more the high frequency rolloff. More disadvantages: as you increase delay BBDs become noisier and I'm not talking about something subtle, I'm talking about becoming virtually unusable. They're also temperature sensitive. Another problem shows up when you try to develop a BBD product: quality control. You can buy a hundred chips and you get five different types of performance parameters...they're just not that tightly controlled. In spite of all these problems, there are quite a few bucket brigade analog delay lines out. Almost every analog delay line on the market uses some form of bucket brigade and they're all subject to the same problems I've just mentioned. But, there is a new type of analog transfer system and it's CCD. A well designed CCD element, used in a well designed delay line format, can be free of these problems. The Marshall Time Modulator and our other products use this new system.

(to be continued)

(AMS-100 cont. from page 1)

on plexiglass panels that mount into a wooden frame. The frame is approximately 19" wide so that it is about the same width as a standard rack panel.

I realized as I got into building the AMS 100 that I would need to go through two packaging stages. In stage 1, I would be able to build up all the modules and try them out with each other in various combinations. However, since I might want to change a module, make revisions in the light of reader suggestions, and just generally need a certain amount of flexibility, I opted for a packaging scheme that would be functional but not permanent. In stage 2, after finalizing the layout and number of modules, I could then repackage them into a standard rack mount panel.

To simplify construction during stage 1, I built the circuits on plexiglass panels because plexiglass is easily drilled (with the proper bits), colorful, and you can get scrap plexiglass inexpensively from a distributor. Figure 2 shows a typical module (the dual envelope generators). Note that the circuit board is *(Cont. on page 12)*