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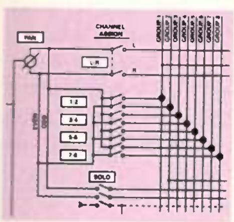

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Dionne Farris "I Know"

Engineer Don McKinzie talks about the crossover hit from the album *Wild Seed-Wild Flower*

EQ: It sounds like the lead vocal is either naturally or artificially chorused. How was this achieved?
Don McKinzie: Dionne is very much into natural effects. When we were cutting, we would often take the rhythm tracks, mix them down to four or six tracks of a second 2-inch reel, and then cut vocals. On some songs we would only mix to a stereo pair and then cut vocals. Dionne might have as many as 20 tracks of vocals — she sang all of the background vocals save for one or two male parts that David Harris sang. So when you hear that doubling you are not hearing artificial doubling, you are hearing Dionne doubling herself. She would spend hours in the studio going over parts just to make sure that they sounded almost like electronic doubles. We could spend up to two hours on one line making sure that the double sounded like the lead. Dionne is a real perfectionist, but she's real easy to work with and she is open to suggestions all the time. She knows what she wants to hear and she won't settle for anything less. Every day she would come in prepared and ready to go.

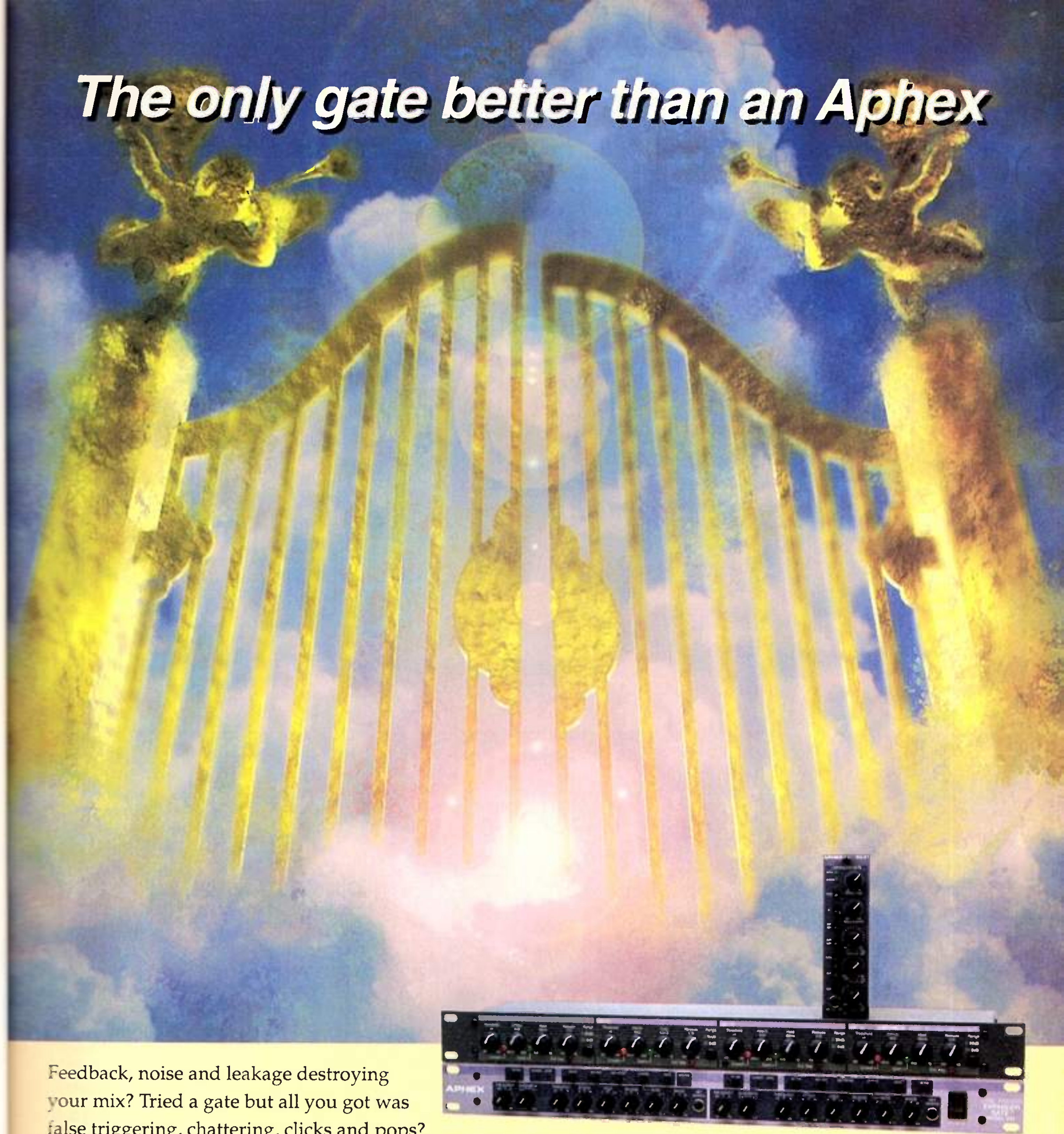
Was she cutting the vocals in the control room?
No, she was in the studio. Most of the vocals were cut here at McMix Production Services, Inc. (Atlanta, GA). We tend not to cut vocals in a booth anymore. We usually put the vocalist in the big room (about 40' x 50') so that we can get some air into the recording. What I would do to minimize reflections is use two six- or seven-foot gobos in a V-shape behind her. But I like getting the air that comes from a high ceiling and large room.

Is that room very live?
The rooms at McMix have variable acoustic panels that can be moved or removed. There was not really a perceptible delay in the room, but it didn't sound like she was close miked in a closet either. It was all by ear, and we would change that quality from song to song or even from part to part within the same song. On "I Know" we didn't use just one microphone because Dionne also sang all of the background vocals. She can change her voice so when we would do backgrounds, she would change to a darker or brighter sound. To complement that we might switch between an AKG Tube, a Neumann U89 or U87, or whatever worked best.

So instead of just picking a mic and playing it safe, you worked for different timbres on some of the parts.
Sure. That's the job of an engineer. You always want to pick the mic that makes the vocalist sound best, but if the vocalist changes, then that is going to change the transducer that you use on that vocalist. Sometimes she would sing a part and I could hear that it was time to change the microphone.

In general were you using the mics set to a directional pattern or in an omnidirectional mode?
It was almost always cardioid, sometimes supercardioid. There were other songs where we cut live background vocals and I would set up two vocalists on each side of a bidirectional mic and let them naturally blend. I always fed the headphones prefader and with the vocalists at the same level. If you feed the headphones so that each vocalist is at the same level, they tend to blend better by naturally adjusting the level of their voices.

ROCKER ROLL: Check out the intense vocal stylings of Dionne Farris on her new CD.


The only gate better than an Aphex

What was the signal flow from mic to tape machine?

We cut everything either with a tube or high-quality, solid-state condenser mic directly into a tube mic pre, mostly the Drawmer 1960. I have done major amounts of voice-overs and I find that the Drawmer does real well. Sometimes I'll pick it over a more expensive unit. I never really use it for the 48-volt phantom power though — I usually use either the microphone's power supply (as in the AKG Tube) or I use an external Sony 48-volt power supply.

Did you compress Dionne's vocals to tape?

We would compress going to tape and sometimes we would double compress going to tape. On some songs, when I started cutting tracks it was apparent to me that the mixes were going to be very thick with a lot of parts. I knew that I wouldn't be mixing, so I wanted to make it as easy as I could for the mixing engineer. In those cases we would double compress with a couple of tube compressors. It makes mixing more of a set-and-forget type of thing. The compression ratio varied between a 4:1 and 6:1 ratio. A lot of times the threshold would be a low as –10. I'm not afraid to compress as long as it doesn't start pumping and breathing!

When you would double compress, did you set the units roughly at the same ratio, or did you set up one to compress and the other to limit?

It varied. Mostly so that one would be compressing the other. After a while you tend not to look at the settings and just use your ear. I would check the metering in and out. We would cut directly to tape with no console in between — I have always liked doing that. I think we were cutting at +9 over 250 nanoWebers/m with Ampex 499 (no noise reduction) on an old MCI JH16 24-track, but a lot of it was cut on an Ampex MM1200. I'm a big fan of that machine. You can really smack the electronics hard and it still sounds good. You have to take into account the transients that some sounds can have like samplers or drum machines. If I set the machine up for +3 or +6 and keep the meters peaking at between 0 and +3, I have the headroom for the transients, and it comes out sounding punchier and cleaner. EQ c to

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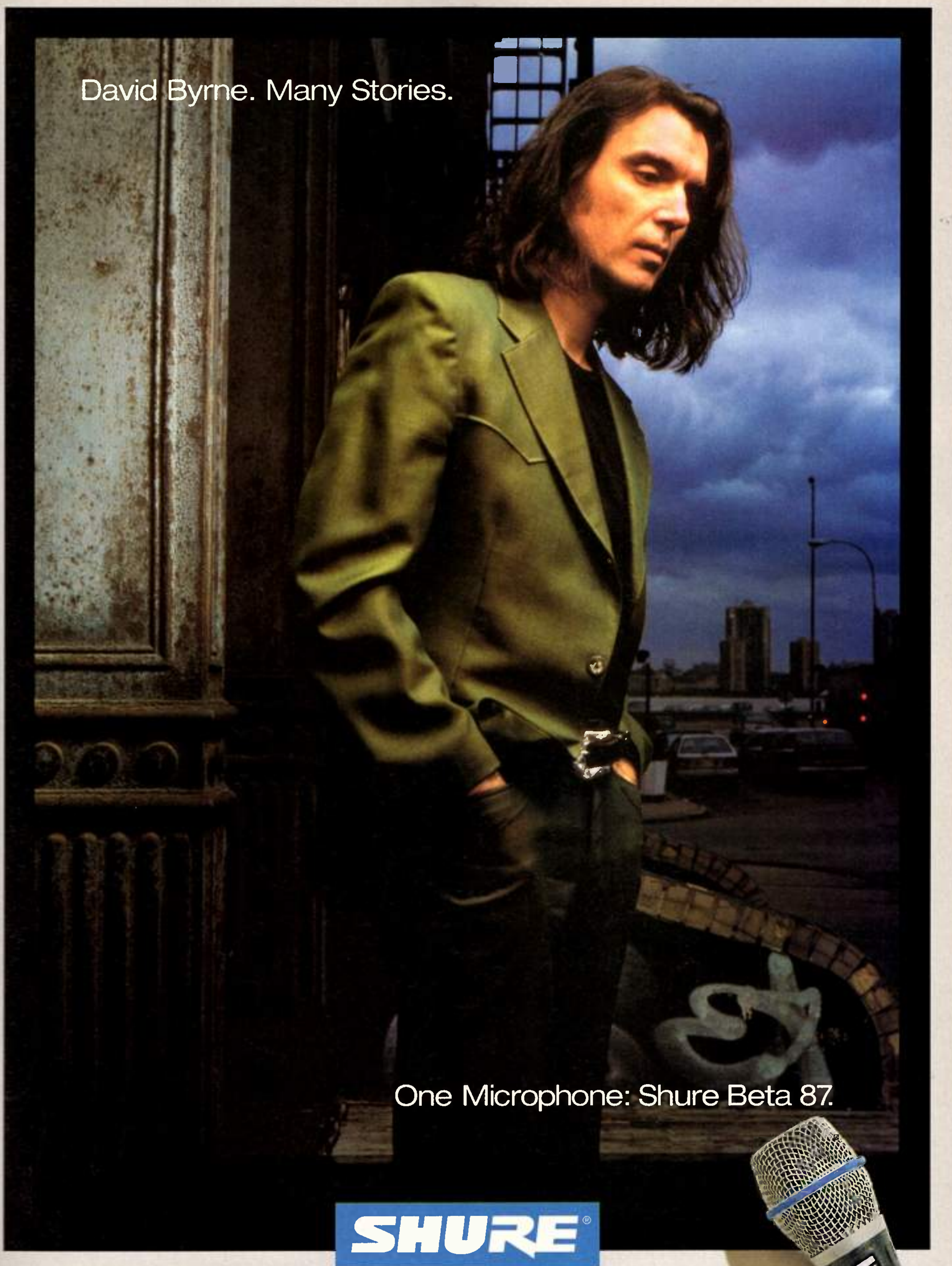
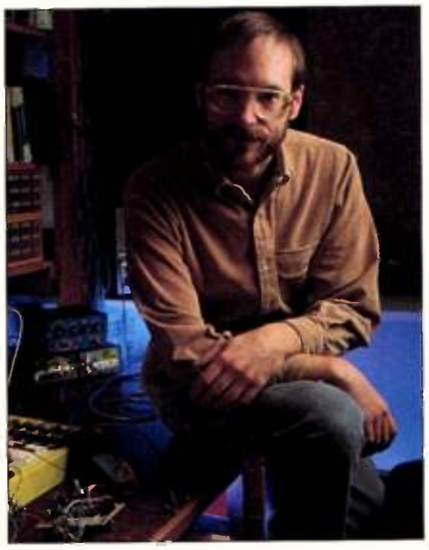


The Long & Short of Short Delays



Make room for delay

BY CRAIG ANDERTON

The reverb-drenched sound of the '60s and '70s is well behind us, as is the chorused gauziness of the '80s. The sound of the '90s is high-definition and in your face; you hear less and less reverb on records, whether you're talking Red Hot Chili Peppers or cutting-edge dance music.

You might think going direct is one way to get this effect, but we're used to instruments having some "air," both from resonances within the instrument itself and from the room in which it is played. Listen to a drum machine going direct into the board: yes, the sound is clean — but there's also a certain deadness. The stereo is too wide; drums become individual points of sound instead of being part of a cohesive, unified kit. Any electric or electronic sound source suffers from similar problems when going direct.

Some recording engineers even pump electronic sounds through speakers and then mic them (not at all a bad idea, by the way), but there's a more predictable and compact way to give your electronic sounds some air: model a room.

AMBIENCE FOR THE '90S

While "modeling" is a big buzzword these days, the concept has been around for a while. Any electronic reverb is essentially modeling what happens when sound waves run around loose in a room.

Recording in a very tight, sparse, hard "box" of a room is one way to get that in-your-face sound. Back in the early days of digital delay, one technique to simulate this kind of ambience was to put several delays (with delay times of 1 to 10 ms or so) in parallel. Mixing these delays well in the background creates the "comb filtering" effects associated with typical small rooms.

Unfortunately, with today's digital multieffects the room, plate, spring, and hall reverb simulations tend to be optimized for traditional long decays with lots of reflections, not short delays — but there is a workaround.

Many multieffects now offer an easy way to experiment with short delay techniques thanks to multivoice chorus algorithms (also called multitap delay algorithms).

The typical multivoice chorus or tapped delay algorithm looks something like fig. 1. Most inexpensive multieffects combine the stereo input signals into mono. This signal then feeds to three to eight different delay lines (fig. 1 shows a four-voice tapped delay). Each delay has controls for initial delay, feedback, level, and pan. There will also be some kind of dry level control.

Multivoice chorus units will also be able to modulate the delay times. Sometimes this controls all delays; sometimes each delay has its own modulation.

When patching into a mixer, you would generally drive the multieffects inputs from an aux (or effects) bus, just like you would a standard reverb, and feed the outputs into the aux returns. Turn up the send controls for the channels you want to process, and regulate the overall level with the aux bus return level control (remember that a little bit goes a long way).

TYPICAL PARAMETER VALUES

Following are two programs for the Alesis Midiverb 4 designed specifically for processing drum-machine sounds to make them sound more "live." (If you don't have a Midiverb 4,

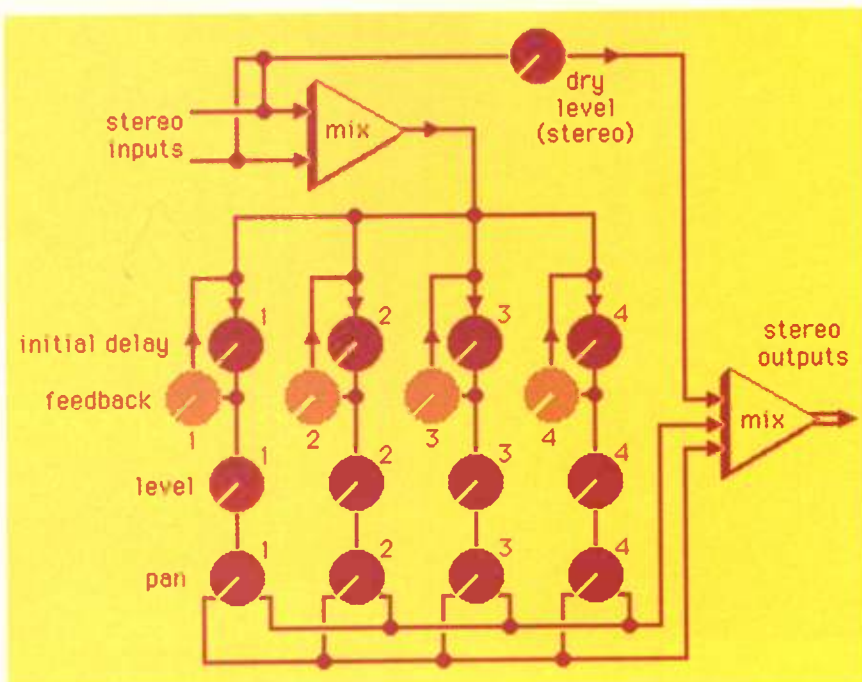


FIGURE 1

Even with 512 user-adjustable programs, 2-octave pitch-shifting, true stereo operation and more, the most amazing thing is how it sounds.

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†Yes, we know you've heard this before, but definitely not in a reverb costing $249.99.

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the principles also apply to other units.) [*See this issue for reviews of the Alesis QuadraVerb 2 and Midiverb 4.*]

Typical parameters for a program based on the Four-Voice Chorus algorithm are:

```
Page 1 (modulation): Rate = 0.2
 Hz, Depth = 255, Feedback = 7%
Page 2 (delay times): Predelay 1 =
 1 ms, predelay 2 = 2 ms, prede-
 lay 3 = 3 ms, predelay 4 = 4 ms.
Page 3: Dry only (—)
```

This creates a dry, tiny room. The very slow modulation effect adds a bit of animation that dynamically colors the sound. For a more static sound, set the depth to 000. To change the room characteristics, try various other predelay times between 1 and 10 ms.

The next program uses the Tapped Delay algorithm. Although with the Midiverb 4 this has only three taps compared to four for the chorus, each tap has more flexibility. Use these values as a point of departure:

```
Page 1: Time 1 = 005, Level 1 =
 99, Pan 1 =  -50, Feedback 1 = 00
Page 2: Time 2 = 007, Level 2 =
 99, Pan 2 =  50, Feedback 2 = 00
Page 3: Time 3 = 003, Level 3 =
 99, Pan 3 =  00, Feedback 3 = 00
Page 4: Master Feedback = 50%,
 LowCut Filter 177 Hz, Hi Cut
 Filter 15.1 kHz, Mix = (—)
```

Some notes: Master feedback is set to 50 percent, so if you want to experiment with individual feedback taps, turning them up just a little bit gives an audible effect. The low-cut filter is handy for keeping the kick drum out of the ambience effect, thus giving a stronger "thud."

This program seems to work best with the short delay panned center and longer delays panned left and right. Of course, more expensive units with more taps let you create correspondingly more complex ambiences.

WATCH OUT FOR...

If you turn up the aux returns to obtain lots of processed signal, beware of phase cancellations. Although the whole point of this exercise is to add the phase cancellation/addition effects found in the average room, high levels of processed signal can cause excessive cancellation. Check the signal for mono compatibility.

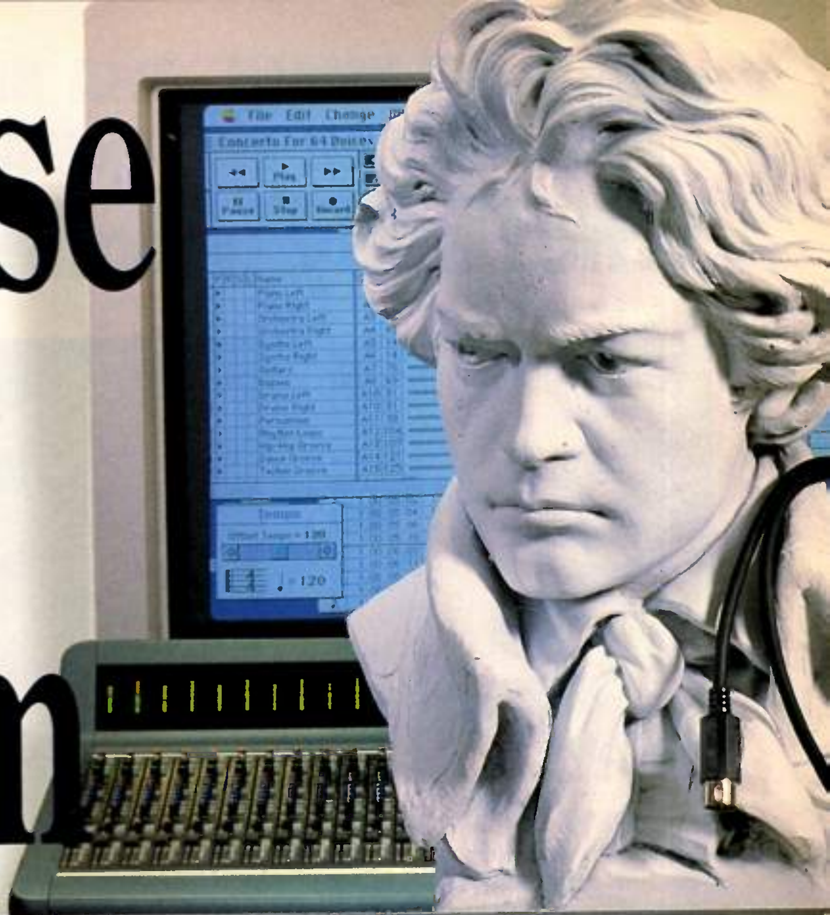
Another consideration is that this technique will tend to "monoize" the signal and make the stereo spread less obvious. Frankly, I consider this a benefit as it provides an overall sonic ambience for the drums.

Finally, note that these delays can sound good on vocals, but there's still nothing like a nice, warm chamber for wrapping around a voice. I always have at least two reverb devices available — one to create these short, ambient delays, and the other a more traditional plate sound for vocals. This gives the best of both worlds.

(Download an AIFF or WAV file from America Online comparing drums processed through short delays compared to straight drums. Use keyword SSS then the following path: EQ Online > Demo Sounds > Short Delay Example.)

Craig Anderton would like to take these few column inches to plug his new book "Multieffects For Musicians" (published by AMSCO), which tells you just about all you need to know about these critters.

Compose Record Perform



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