

# Letters

## Music software

From: Larry Polansky, assistant professor of music, Music Department, Mills College, Oakland, CA.

Paul D. Lehrman's "Managing MIDI" column in the June issue points out some important questions and issues in the continuing development of music software. He very correctly emphasizes that there continues to be a greater and greater need for expandable, flexible software.

The next generation of software will almost certainly lean more toward language design, and less to restrictive, and inherently limited, applications. Expanded system exclusive implementations in commercial hardware, a more sophisticated user base and the collective creative urge of musicians and producers will, I hope, encourage software designers to more and more often leave behind their imaginary end-user who is, it appears, seen to be impatient with complex machine intelligences, and unwilling to accept open-ended designs which encourage creative user interaction. To implement the kinds of important things that Paul suggests, MIDI software needs to be user-definable and, of necessity, design issues need to be concerned with what many of us refer to as "music languages."

There are already several such environments, including MIDILisp/FORMES from IRCAM, our own HMSL (Hierarchical Music Specification Language). Dan Kelley's MASC, and Ron Kuivila and David Anderson's FORMULA, to name just a few. All of these "languages" are available for standard personal computers (Macintosh, Amiga, Atari, IBM, etc.). All of these environments are also characterized by a high degree of generality and a correspondingly high learning curve. In fact, for three of these environments, the user needs to be a reasonably competent FORTH programmer and, for the fourth, a LISP programmer.

Power and generality are often proportional to ease of use, yet ease of use is also directly related to the general sophistication of the user base. This sophistication will only improve if software designers recognize the tremendous untapped abilities of composers, musicians, producers and engineers, and give them programs and languages worthy of their talents. Software designers will not leave the music world behind; they will bring it forward with them, happily in tow.

## Random-Access Editing

From: Bob Katz, New York.

I just finished reading your remarkable July issue on digital technology. I attempted to speed read to avoid future shock, but succumbed nevertheless. My first reaction to the issue was that my article on advanced 3-machine digital mixing has become instant "primitive" history, in the light of the AMS AudioFile (also reviewed in the July issue) and similar disk-based editing systems.

As a matter of fact, random-access editing *a la* AudioFile would prove to be *exceptionally* efficient in editing the spoken word for commercials, films, radio, etc. One fact that I did not cover in my article was that 1/4-inch editing of spoken-word audio normally involves removing and adding many tiny pieces of tape, containing "lip smack," extraneous noises made by the actor, and room tone. Typical 1/4-inch spoken-word edits contain, on the average, splices about every five seconds, often pulling very short pieces out of the tape.

It is easy and quick to make such splices on a 1/4-inch tape machine. It is almost ridiculous, however, to attempt this type of fine editing on a VCR-based system, with its time-consuming rehearsal process, and difficulty of pulling pieces from within the middle of an already-edited program.

Clearly the AMS AudioFile will provide an efficient, razor blade-less method of cutting spoken word. (We were very lucky that

Christopher Plummer has a pretty "noiseless" mouth, or the voice design alone of *Nutcracker* would have taken several days via the Sony DAE-1100 editing system.)

I would like to know how the price of the AMS AudioFile compares with the unique complement of equipment I assembled for that 3-machine mix: three BVU-800's, three PCM-1630's, one DAE-1100 and a TimeLine Lynx synchronizer. I would also like to know whether the AudioFile can control the gain of each D/A output in the digital domain, because the level-change information it stores could allow the unit to perform automated mixing. [Currently, AMS doesn't provide digital control of level changes, this capability may be added later—*Editor*.]

Then, its eight outputs could connect into a very simple production-type audio console at unity gain, without passing through VCAs or other signal degrading devices. I would be only one step from there to feed a small digital mixer avoiding eight D/A conversions.

The July report of a "Transcontinental Digital Overdub" by Paul Lehrman and David Rideau is also future shocking. However, I should inform you of a recent technical development that will allow multiple musicians throughout the country to simultaneously perform and overdub via satellite, without experiencing the time delay problems mentioned in the article.

The device is called a Digital Advance Line (DAL), now under construction at a U.S. lab and incorporating the latest in superconductivity and time-predictive techniques.

Soon, a singer will be able to send his voice via satellite to a remote site and, by inserting the DAL into the satellite return, can hear his own voice in the headphones mix without echo problems. In fact, the DAL actually *anticipates* what the singer will sing *before* he sings it.

To encourage sales of this time-advancing unit, the manufacturers, in a unique marketing ploy, are asking for their \$1 million price to be paid in 1950s dollars!

Thanks for giving me the opportunity to comment.

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News, continued

### DDA delivers in Europe and United States

DDA has installed DDA consoles at the following locations:

- Capron Light & Sound, Needham, MA; D series with 40 inputs and 8 outputs.
- Sound Rental Services, Parkersburg, WV; two D series consoles for house sound and monitors.

— Saban Productions, Studio City, CA; AMR24 36x24 console and a D series for its post-production room.

— Abbey Road Studios, London; DDA console and D series 16x2 console for mobile digital recording.

— Peter Rafelson, composer/producer; AMR24 36x24 console.

— David Dundas, London; AMR24 28x24 console.

— Tape One Studio, London; S series 6x console.

— Scacco Matto Studios, Lavagne, Italy; ARM24 44x24x2 console with 64-channel Audio Kinetic Mastermix.

— Orinoco Studios, London; ARS24 36x24 console with 36-channel Audio Kinetic Mastermix and remote patchbay.

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