July 1987

Recording ENGINEER/PRODUCER

T.M.

The Technical Journal for Audio Professionals

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Scheduling & Booking Page 52

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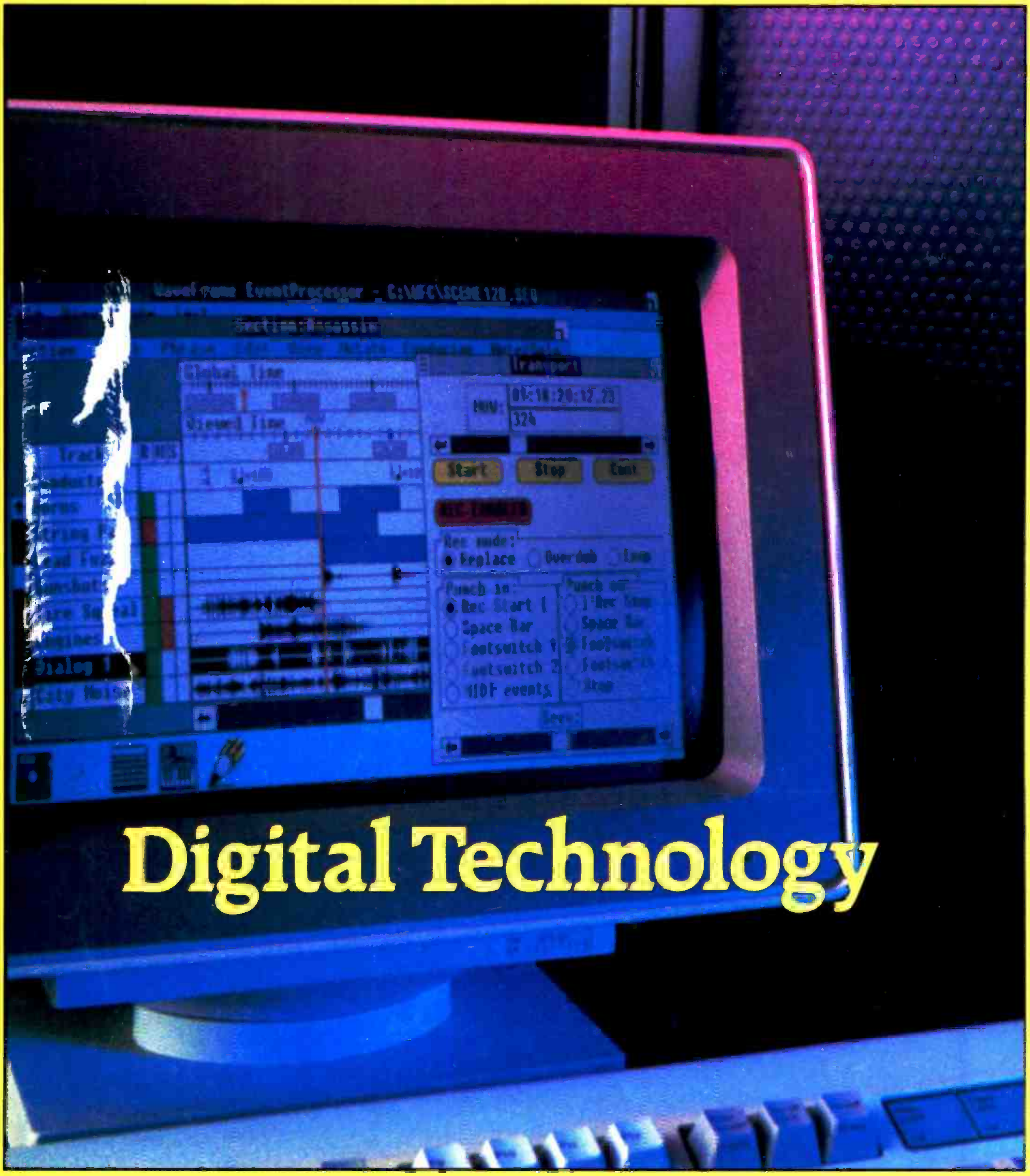

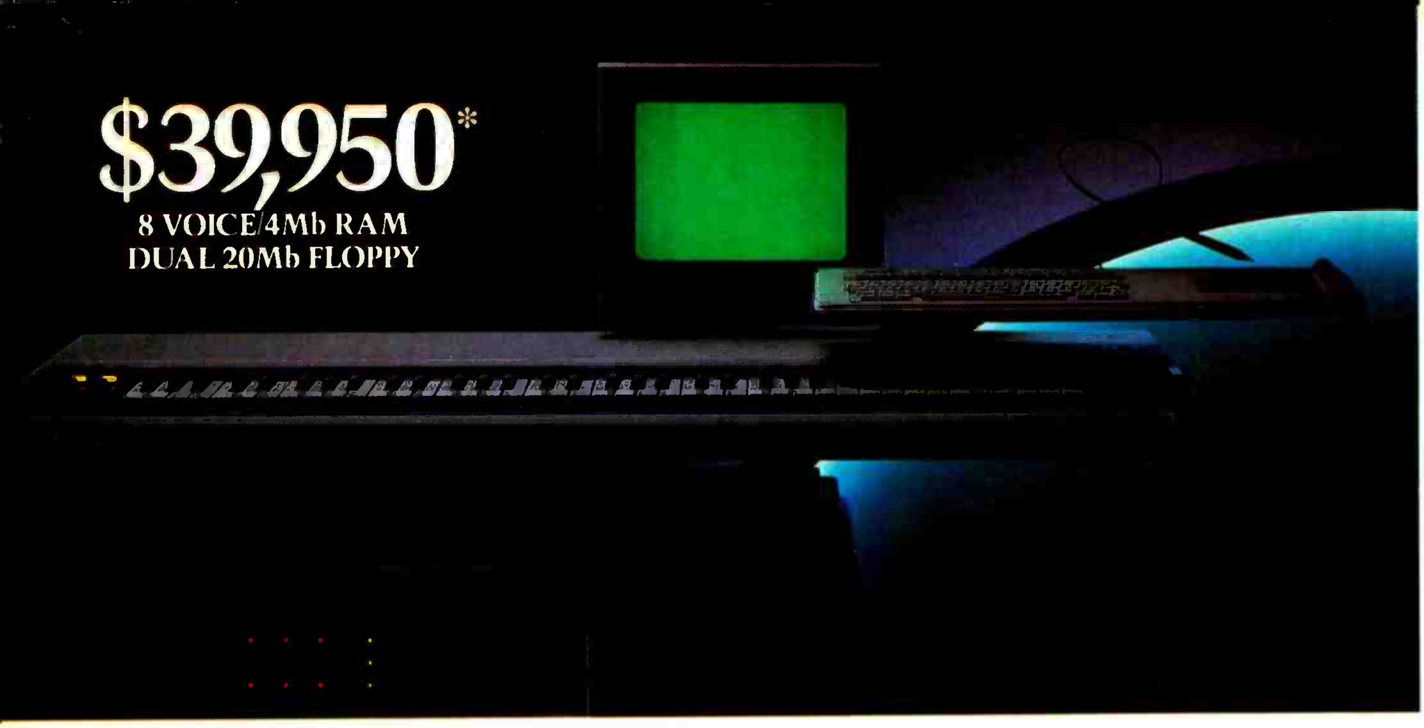
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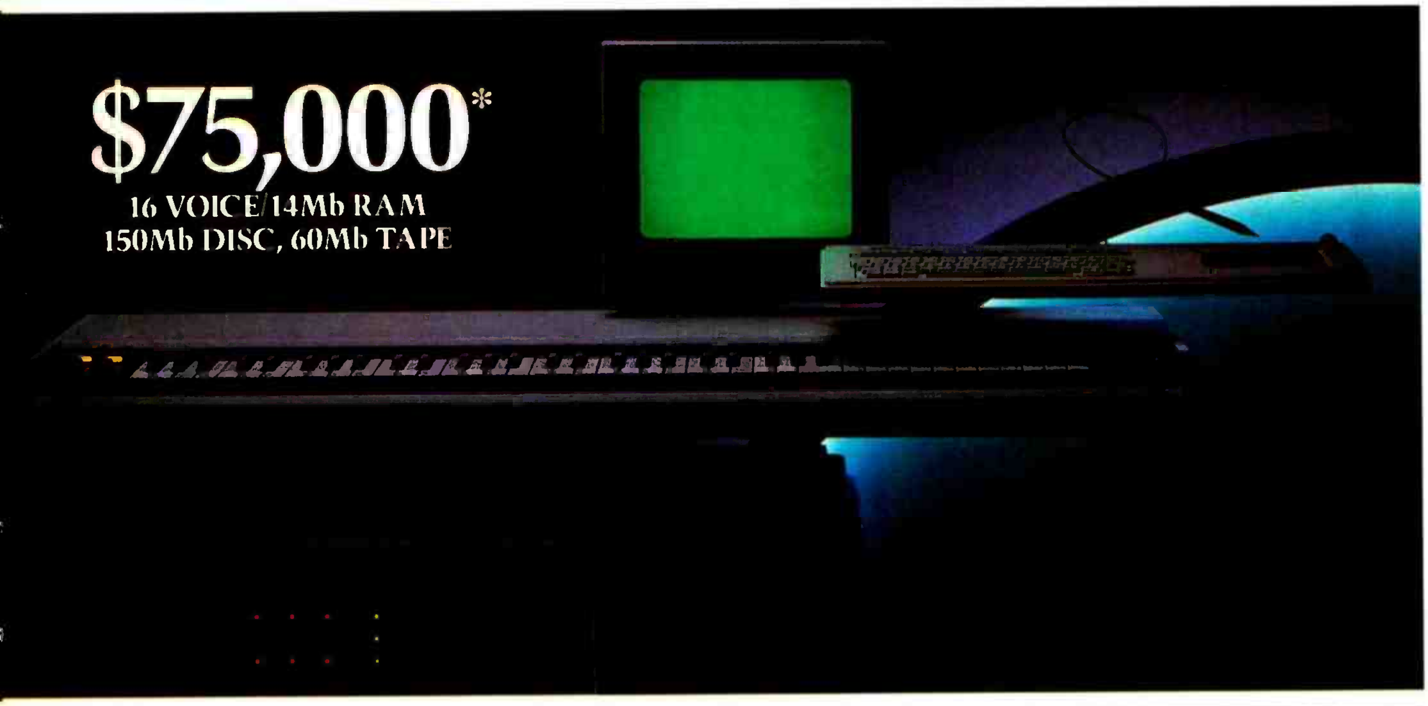
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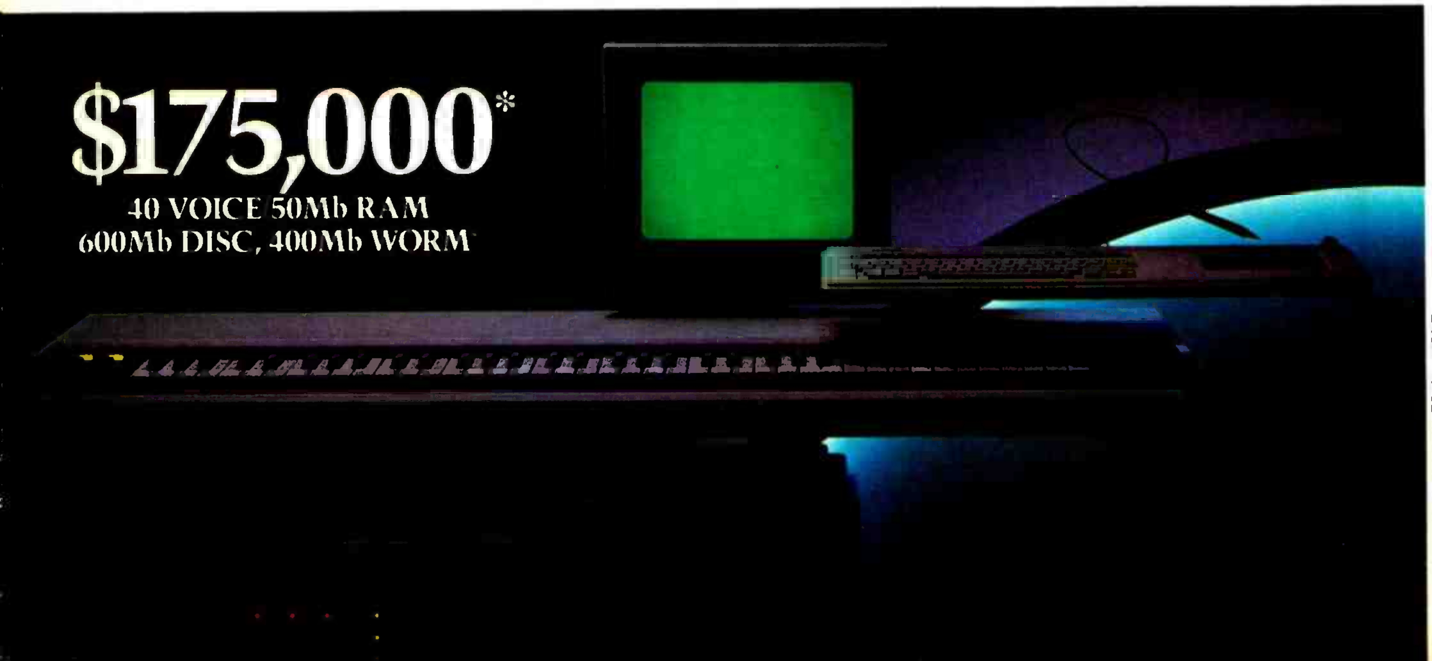
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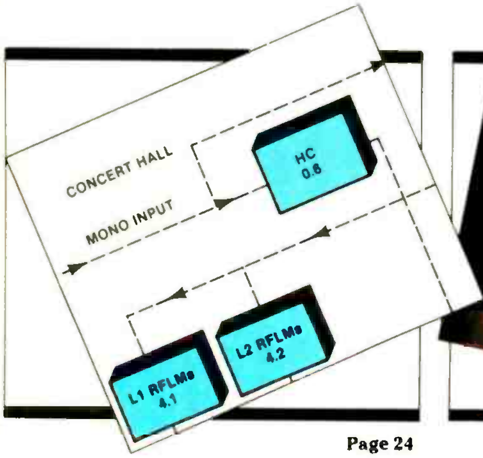

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On the Cover:

An Event Processor Screen display from the newly developed AudioFrame digital audio workstation, which illustrates one of the potential ways in which MIDI performance data might be incorporated with the recording and editing of digitized audio signals. Photo courtesy of WaveFrame Corporation, Boulder, CO.

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Editorial

Increasing Your Options

By Mel Lambert, Editor

Given that, by its very nature, our industry is strongly based upon the communication of musical ideas and concepts, I should possibly be less surprised at the ways in which industry professionals explain their particular visions of the future. Not so much with regard to *what* they are telling me, but rather the changing language and vocabulary we now seem to be using to explain our individual view of technological developments.

Not so long ago, the major flavors of "technological alternatives" were confined, in the main, to developments in analog consoles, tape machines and outboard sound benders, studio techniques being pretty much based on these mature technologies. These days, however, the future of the recording and production industries—not to mention its associated manufacturing base—is tightly linked with digital technology.

The bottom line in all of this is that we are all learning to conceive of the future potential of audio facilities in terms of computers, operating systems and software.

By way of a tangible example, I am reminded of a recent conversation with a well-known producer and sound designer. He and I were catching up on current technology—an endeavor I find myself doing on a regular basis these days—and were comparing notes on random-access editing systems and hard-disk recording systems.

The conversation ranged far and wide that morning, before we started to trade opinions on what we knew about potential manufacturers of digital storage systems, and what thoughts we had about control and display topographies.

While we both seemed reasonably well-versed in the names of the major (and up and coming) audio companies cultivating a definite interest in things digital, it gradually began to dawn on me that very few of the companies we were referring to were names we would have even considered barely a year ago.

Not necessarily that any of the companies had recently demonstrated an expertise in these emerging technologies; rather that some of them didn't even exist 12 months ago, or seemed to be so headed in one specific direction that few of us could have guessed they were quietly cultivating an interest in digital processing, to name but one blossoming area of audio technology.

Turning to practical audio hardware, my producer friend and I found ourselves spending an increasing amount of time dealing with specific computer topics:

- What microprocessors and support chips were the new system built around? Did any particular micro seem to outshine the rest?
- Did the choice of micro and/or operating system affect the system's long-term expansion capabilities?
- What did we know about the software that was expected to spin through this device? (Or vaporware, if hardware commitments were still a ways off—either to allow the company to play chip-gazing games, or because of possible/eventual VLSI decisions.)
- What did we know of the people currently writing or being recruited to write the software? Or, to put it another way: What could we expect in the way of sensible man/machine interface?

Will we be facing boring 4-color high-definition VDUs, or has anybody *really* been spending some time thinking about display technologies and ergonomics theory? Has anyone come up with a more appropriate technique for informing the user of what he or she needs to know about system status and control options than flat-screen displays?

- What did we know (or could make intelligent guestimates) about the system's projected lifetime in the studio? (Or, to put it more prosaically: How long would it take the opposition to fathom out the unit's inner workings and release their own "enhanced" version once they have a better understanding of the market's reaction to the MkI?)
- How would the existence of a digital interface standard or sampling frequency/bit rate standard affect the unit's long-term acceptance? Anybody out there with a reasonable distance at producing cost-effective 18-bit sampling?
- Why do most of the interesting developments seem to be coming from smaller companies and individual-rich companies that have simply taken an off-the-shelf microcomputer system—mentions of the Atari ST1040 Apple Mac II occur all the time now, and probably represent the two ends of the cost, but not necessarily processing power, spectrum—and make it handle elegant digital processing and manipulation tasks?

Have we truly reached the age when the young(er) turks will surprise us with their proclivity for lateral thinking around problems that most of us cannot even begin to understand or solve?

Yes indeed. Times they are a changin'.

R·E/P

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The "Bothy" at the bottom of our garden is where we work on advancing console design concepts.

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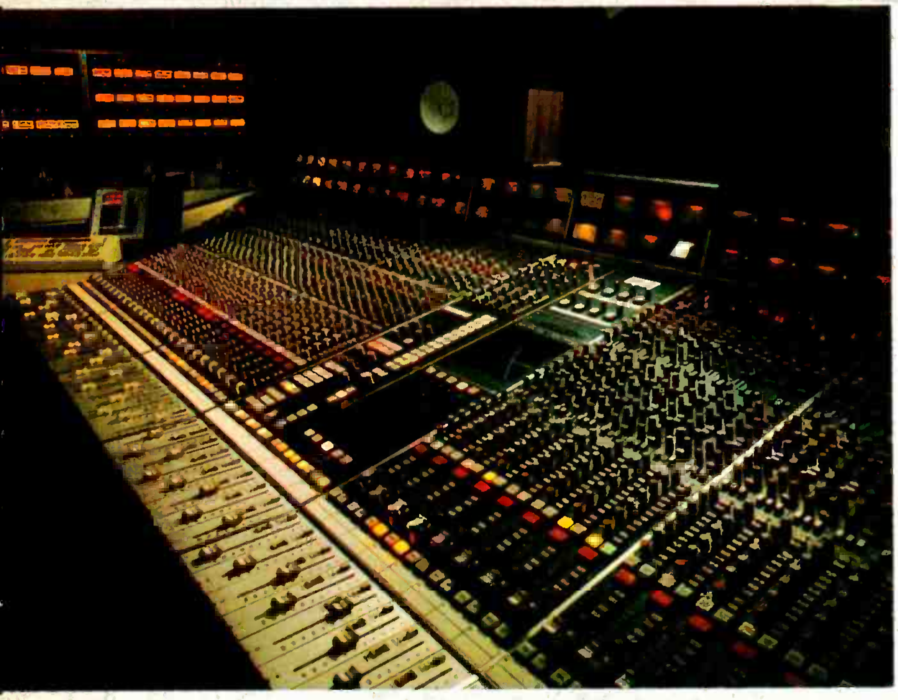

It's just one example of the way we continue to enhance the sound of the SL 4000 E Series Master Studio System to keep ahead in the technology race. And if you already have an SL 4000 E Series you can stay ahead too, as the new EQ is easily retrofitted to all existing consoles.

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News

Sound Genesis introduces computer music services

The company's first sonic software product line for producers is scheduled for introduction in August, and will be implemented on the Fairlight CMI Series II and Series III digital synthesizers.

According to company president Don Webb, "The Master Sampler Collection will contain 20 or more separate packages of documents, software and media, each of which includes 10 'virtual instruments' from one of three categories: instrumental, vocal and environmental.

"Every virtual instrument carries a pedigree that documents its authorship and prior usage. Its purpose is to help professional sound producers rest easy in the certainty of their explicit right to use the virtual instruments in producing their own commercial products, free from any concern of copyright infringement."

The company will also supply a "congenial human interface" that is said to simplify operation of the CMI. This proprietary program is designed to connect an Apple Macintosh to a CMI, simplifying access to Sound Genesis' data base of virtual instruments and signal processing functions.

Included in the sonic software license is six months of free customer support service by voice and by modem. The company's support service not only offers technical problem-solving expertise, but also assistance with the creative elements of musical composition and sound production.

More details from: Sound Genesis, 7807 Creekridge Center, Minneapolis, MN 55435; 612-944-8528.

Revised edition of Cipher Digital *Time Code Handbook*

Subtitled "A Guide for the User from Fundamentals to Technical Specifications," the booklet is divided into two sections. Section #1 describes basics and applications; section #2 provides advance engineering information, technical guidelines and specifications.

The revised edition of the *Time Code Handbook* costs $12.95, plus $3.50 shipping and handling, from Cipher Digital, Inc., P.O. Box 170, Frederick, MD 21701; 301-695-0200.

Air, Sigma and CBS/Sony install GML automation

At George Martin's AIR Recording Studios, London, a 48-channel Moving Fader automation system has been added to a Neve V-Series console in Studio 3.

The second installation was at Joe Tarsia's Sigma Sound Studios, Philadelphia, PA. A newly rebuilt Neve Model 8078 is now equipped with 52 GML Faders modules and computer.

A 48-channel system at CBS/Sony Recording Studios, Tokyo, Japan, went into a custom Neve 4310 Series console.

Although the CBS/Sony order marks the second GML automation system delivered in Japan, the local representative, Soundcraft Japan, has five systems on back order, including a system for each of two new rooms being built by Toshiba/EMI. This pair of systems will be fitted to existing 60-input Neve V-Series consoles.

Recording ENGINEER/PRODUCER

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RECORDING ENGINEER/PRODUCER is edited to relate recording science to recording art to recording equipment, as these subjects, and their relationship to one another, may be of value and interest to those working in the field of commercially marketable recordings and live audio presentation. The editorial content includes: descriptions of sound recording techniques, uses of sound recording equipment, audio environment design, audio equipment maintenance, new products.

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Boogie

News

Studer announces first U.S. delivery of D820X DASH 2-track

The first delivery was made on May 12 to Disc Mastering, Nashville. The DASH-format D820X offers an AES/EBU digital port as standard equipment, and 14-inch reel capacity for over two hours of continuous recording. Transport features and tape handling characteristics are equivalent to those of the A820 analog 2-track.

"I've only had a few days to work with it," admits Randy Kling, owner and chief engineer of Disc Mastering, "but from what I've seen so far, this machine takes the Studer tradition over the digital threshold. It's high-tech in the fast lane, yet the D820X is also exceptionally easy to use."

Within days of installing the new 2-track, Kling also took delivery of a new Neve DTC-1 digital transfer console. His facility thus became, for a few weeks at least, the only all-digital Studer/Neve operation in the United States.

Kling's first session was a CD mastering project for a Dallas production music and sound-effects client. He also anticipates using the hardware for archiving rare old recordings from the RCA vaults, and historic lacquers from the Country Music Foundation.

The Enterprise Studio—An Update

Since the publication of our cover feature on Craig Huxley's The Enterprise facility in the May issure of **RE/P,** the consoles in Studios B and C have been replaced with Solid State SL4000-E Series boards.

Studio B now features a 72-input SL4072 with Total Recall automation and G Series computer and operating system, while Studio C now offers a 72-input SL4064 (64-module SL4000 mainframe plus stereo input modules) with Total Recall, upgraded equalization and G Series computer. R-E/P

Letters

Noise reduction mysteries

From: Michael Cogan, Bay Records Studios, Alameda, CA.

While perusing your January 1987 issue of **RE/P**, and the article "Narrow-gauge Multitrack Applications" by Adrian Zarin, I came upon the following quote on page 25 from Alan Kozlowski [technical director of Visual Eyes Productions, Santa Monica, CA]: "I think everyone knows there are some limitations to noise reduction in terms of what it does to the sound; it tends to compound any frequency-response errors that might be in the signal."

Whereas he is basically correct in shying away from using noise reduction when you can avoid it, I feel that his remark may be based on a much misunderstood myth about dbx—that is, it doubles the frequency response errors of a recorder. This is actually *not* true.

dbx is a broadband system and *cannot* effect the frequency response by itself. It's true that if you measure the response of a recorder using single-frequency sine waves, and then switch in the dbx unit, you will see that any response errors are doubled. This is a fault of the measuring technique, not the noise reduction.

Any system that incorporates signal-dependent gain stages should be measured using *broadband* signals to approximate music.

Of course, response errors in the recorder can cause pumping or other mistracking of the noise-reduction system; with heavy bass notes this may seem to effect the response slightly.

However, it's *not* true that if a recorder plays back slightly bright, it will be twice as bright with the dbx switched in.

Thanks for this opportunity to correct a common audio myth. Keep up the good work with your excellent magazine!

Natural reverberation

From: Leslie Bell, Bang Music Worldwide Communications, West Hollywood, CA.

I found John Eargle's article, "Evolution of Artificial Reverberation" [published in the February issue] fascinating. I too had always tried to figure out why older recordings sounded better, and seemed to have an acoustical life all of their own. It seemed to me that, despite all the advances in recording technology over the years, the quality of recorded sound had gotten worse, not better, from a peak reached in the early to mid-Sixties, most notably on Columbia pop products.

This disparity can readily be observed on the recent double-album sets of *TV Toons*, compilation LPs of various TV themes. In the cases where the producers were able to obtain rights to the original tapes, the fidelity and frequency response may be lacking, but all the acoustic power response and energy of the music is evident in full force.

If you contrast those original soundtracks with their attempts at current recreations, it is painfully obvious how much we have lost over the years. It seems that the human ear will make up for poor fidelity and frequency response (after a while even the most atrociously frequency response-limited material sounds normal). But the ear cannot put back that natural ambience and power response of the music, and the natural acoustical doubling and tripling that ocurred on those early recordings.

I just have a couple of questions for John, in regards to the table of peak acoustic output levels that he included with his article. Where can one find the information to compile a more complete table, with every possible instrument listed? And how do these power levels combine in relationship to each other? For example, how much louder are six saxophones than one? Or four trumpets or trombones, as opposed to one?

I am currently producing a big band that's going for the sound of those early Columbia recordings, and I need to accurately compute the relative power levels in making an ambience mix, to mix back with already recorded tracks. Any information would be greatly appreciated.

Thank you very much for your time, and a great article!

Reply from John Eargle

Lesile Bell has identified one of the significant differences in big-band recordings of the Sixties and those made more recently. In the earlier days, recording engineers had no more than three or four available tracks, recordings and their consoles handled perhaps 16-20 microphones. A lot of pre-mixing had to be done at the time of the sessions, and only major balance decisions, such as vocal and rhythm, could be made

KORG
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CONCERT HALL

Letters

on this at a later date.

Today, an engineer has 24 or more available recording tracks, and his console can handle more microphones than he will ever need for a big-band session. It is easy then for him to fall into the "one-microphone/one-channel" trap, with consequent close-mic placement. The earlier approach favors natural ambiences while the latter, in many ways, represents a misuse of today's technology.

Regarding the acoustical power outputs listed in the table, there is very little reliable data available. To my knowledge, no recent measurements have been made and, of course, peak outputs will vary from player to player. We can assume, however, that in multiples of the same instrument, output levels will increase 3dB for each doubling of instruments.

The intent of the table was basically to underscore the extremely wide range that we can expect to encounter. As a practical matter, I would recommend that the producer and engineer rely on their own judgment and taste in adding reverberation to a set of relatively dry tracks, realizing of course, that brass and percussion will dominate in the power response pictures.

Most important of all is to use reverb settings that really match the acoustics of the relatively live studios (often ballrooms and converted churches) that were popular in the Fifties and Sixties.

MIDI Time Code

From: Chris Meyer, software engineer, Digidesign, Menlo Park, CA.

After reading Paul Lehrman's "NAMM Winter Market Replay" in the March 1987 issue of **RE/P**, I thought I would update Paul and everybody in general on the status of MIDI Time Code, and clear up some confusion on its uses.

MIDI Time Code was indeed approved by the Japanese MIDI Standards Committee (JMSC) just before the January NAMM show in Anaheim. The MIDI Manufacturer's Association (MMA, the American and European version of the JMSC) had approved it at the previous summer NAMM in Chicago. Most manufacturers have been good boys and girls, and not announced or implemented MIDI Time Code and its related features until it was approved. Also, there has been a healthy amount of "wait and see."

The structure of the MMA is such that engineers at the various companies collaborate on and finally approve any extension to MIDI; it's usually only after this that the extension is spread to the marketing department. Somebody canvassing the floor of a NAMM show talking to salespeople, as opposed to elusive engineers, could easily get the impression that MIDI Time Code was still just a persistant rumor. (I don't know which group of "voices" Paul was quoting.) And then there's just plain not tipping your hand to your competition.

I have two comments about Paul's statement that "advances in the economy of directly converting time code to MIDI [clocks] and vice versa might just make the format moot."

The first step is to make clear the differences between MIDI Time Code and what Paul refers to as "SMPTE-to-MIDI converters." Currently, a hardware box such as a Roland SPX-80 or Garfield Masterbeat is needed to convert from SMPTE time code to MIDI clocks to run sequencers. These boxes have tended to cost in the area of $1,000.

If these sequences could receive MIDI Time Code directly (a software upgrade that is a lot less expensive per unit than hardware MIDI conversion), they would not need a SMPTE-to-MIDI clock converter.

True, the cost of SMPTE-to-MIDI *clock* converters is coming down, but a pure SMPTE-to-MIDI Time Code converter will always cost less—it does not need the additional RAM or user interface to do the additional mapping from absolute time to a tempo (and those that do convert time to tempo will cost the same).

However, not everything in a system may want to run off the same stream of MIDI clocks at the same tempo—they may need absolute time directly or want to run at different tempi. In these applications, it is not economical to have a SMPTE time code converter for each device. Instead, one SMPTE-to-MIDI Time Code converter could be purchased, and MIDI Time Code sent to all the other devices in a system at considerable hardware cost savings.

Secondly, the "time" portion of MIDI Time Code is less than half of what the specification is all about. The other half has to do with "cuing" information: firing of events, switching of programs, punching-in and -out of record tracks, etc., based on SMPTE times. As mentioned above, some things in our studio work on what time it is, as opposed to what bar and beat it is (ie. videotape); such devices need to know *when* to take actions based on absolute time, not MIDI clocks. In other words, MIDI Time Code is an adopted edit decision list software format using a cheap and common hardware interface. Once a device has this decision list stored, it needs either SMPTE or MIDI Time Code (*not* MIDI clocks) to fire these events.

MIDI Time Code is real and it's finally here. Developments and applications, as always, will take a little more time in coming than any of us would hope, but they are happening. Our company is in contact with several manufacturers who actually are going to put it in samplers, effects devices and tape decks. So can we just sit back and see what develops before striking a death knell? History will prove who was realistic and who perhaps misprojected a trend based on early hints.

[*Editor's note:* Paul Lehrman, **RE/P's** electronic music consulting editor, is currently preparing a feature article detailing the potential applications of MIDI Time Code in recording and production facilities.]

Op-amp slew rates

From: Walter Morton, consulting engineer, San Diego, CA.

The article, "Understanding Circuit Principles," by Terry Pennington and Larry Winter, published in the March 1987 issue was a useful introduction to the subject.

However, I believe there are those who would contend that slew rate must be evaluated with overload characteristics. The bottom line is that sound reproduction may be "colored" in reproduction with slew rates less than 200Vμs.

This contention is supported by listening tests on "sensitive" ears, where both THD and TIM were considered.

Reply from: Terry Pennington, director of technical marketing and development, Rane Corporation:

I fully expected more than just the one response I have received so far on our contention regarding slew rates. The importance of high slew rates has been a very hotly contested subject in the audio community for a long while and is one which, I believe, cannot be resolved. We can apply liberal amounts of science to the question and come up with some very convincing and conclusive evidence, which leads us to the conclusions stated in the article. These are the only conclusions that can be stated without involving the uncertainties of the human element.

This is not to say that the human element is not important; it is, after all, mainly humans that have to listen to what we build. The problem is that each individual will perceive input in a personal way, and this creates a great deal of difficulty when one is trying to quantify a subjective phenomenon dealing with the production or reproduction of audio.

Objectively, we can determine how fast any given circuit must be to reproduce any given waveform at any maximum frequency without any nonlinearities (distortion). Having done that, it is inconceivable that an audio waveform of any kind will cause the circuit to distort because of component speed limits. This will be true as long as the audio waveform does not exceed the maximum frequency used to calculate the required slew rate. This is relatively easy to do, and was carefully attended to in the example included in the article.

No matter how much care is taken in the objective end of things, there will still be perceptual problems when the system is auditioned. When someone listening to the circuit believes they have heard the effects of slew-rate limiting, the waveform that caused it can be analyzed with sophisticated test equipment, and verified to not be outside the frequency limits calculated for the system. The levels can be checked, analyzed, poked and prodded—there can still be *no* explanation to one who has heard the difference. Science cannot explain faith.

It should also be noted that anyone using any sort of digital medium for a source is even more unlikely to hear the effects of slew-rate limiting. This is due to the fact that all analog audio inputs must be filtered before they are digitized to remove any extraneous high frequencies above 20kHz to prevent aliasing (distortion) in the conversion process. Once filtered, we have guaranteed the maximum audio bandwidth and therefore eliminated all possibility of slew-induced distortion (or TIM, if you will) in a properly designed circuit. "Properly designed," in this case, means that the circuit will have two to five times the calculated minimum slew rate based on the highest frequency of interest, which is the corner frequency of the anti-aliasing filter.

It is interesting to note that many of the individuals who were opposed to digital audio at its inception were also those who believed that extremely high slew rates are important. I shall refrain from speculating on this connection.

Some will point to the test developed to prove the existence of TIM, and to quantify its magnitude, as proof that it does indeed exist and is a problem. The problem with this is that all of these tests used either unrealistic frequencies or levels or both to accomplish their tasks. Unrealistic in the sense that audio does not contain such frequencies at anywhere near these levels.

Such tests were accurate in a sense; they were just not relevant to the successful, linear reproduction of any audio source. If such frequencies and levels did indeed exist in audio material, speaker manufacturers would not be able to build their midrange and tweeter elements without voice coils the size of those in low-frequency drivers. They would burn themselves out due to excessive power dissipation (or lack thereof).

I realize that this is a rather lengthy response to a rather short observation. The subject of slew rate, as well as other audio legends, requires more than just a few pages. If clarification of any of the brief points is required, please let me know. I will do what I can to oblige through the pages of **RE/P**. R·E/P

RE/P welcomes letters to the editor concerning articles appearing in previous issues or any pertinent industry topic. Letters may be sent to Mel Lambert, RE/P, 1850 N. Whitley Ave., Suite 220, Hollywood, CA 90028. Please mark on the outside of the envelope that it is a letter to the editor. Letters may be edited for length and clarity.

Managing MIDI

By Paul D. Lehrman

Probably the single most common question people ask me about MIDI is: "What's going to replace it?" To me, that is like someone going up to Alexander Graham Bell right after he invented the telephone, and saying, "Well, that's nice, but what are you going to do next?"

Anyone concerned with replacing MIDI is making the presumption that MIDI cannot last as a standard, either because it's too slow, it's too restrictive or, in some other way, it's just "not good enough." I think that's jumping the gun a bit—MIDI is far from exhausted.

Although technological obsolescence is rampant in the recording industry, it is not necessarily *inevitable*, and that's a good thing. Digital recording has replaced analog for many tasks, but you can still play back a tape made 20 years ago in just about any professional studio.

The same can be assumed of MIDI. New ways of pursuading musical devices to talk to each other will be developed over the next few years, but I hope they will not make MIDI obsolete—rather they will add to its efficiency and usefulness.

Don't get me wrong: I've got the same complaints about MIDI as anyone else. I get just as frustrated at its bandwidth limitations—its propensity to choke when given lots of controller or pitch-bend data—and the fact that certain types of musical events are very difficult, if not impossible, to reproduce with it.

There are, however, ways to get around these problems, and although some of them haven't been invented yet, I have faith. I consider these problems more limitations of the currently available hardware and software, than of MIDI itself—it's the way MIDI is used, not the standard itself, that needs upgrading.

MIDI should not be dismissed lightly. Before it came along, there were dozens of ways for synthesizers to communicate with each other, but each of these methods worked with only a small group of instruments, usually from one manufacturer. Even the smallest variation in a control scheme—whether, for example, control voltages were to be read linearly or logarithmically—would render one unit useless when interfaced with another. MIDI changed all that.

In fact, MIDI is one of the few successful standards that has evolved in *any* area of communications technology that was not bulldozed into existence by one particular company with its own products to push—instead, it was hashed out and compromised over by a number of manufacturers. These companies were enlightened enough to realize that serving their common interests made more sense than pandering to individual desires.

But, besides the historical perspective, there are two factors that figure strongly in MIDI's favor as a medium for the future. One is that it is expandable. There is plenty of room built into the specification for new commands; not only ones that can be used by manufacturers to make their own products perform specialized tasks, but also, thanks to a set of "non-exclusive" System Exclusive instructions that are just now coming into use, commands that anyone can find useful.

MIDI can be sped up not by increasing the baud rate, but by splitting the data into more than one stream.

For example, MIDI Time Code is something that any MIDI device, whether it's a sequencer, mixer, lighting controller or a synthesizer, can potentially take advantage of in one way or another. Another example is the MIDI Sample Dump Standard, which will become more important as studios find themselves with a variety of samplers from different manufacturers that they would like to have talking to each other.

Controllers are another area where lots of room exists for growth. There are plenty of unassigned controllers, and now that hardware and software is available that can easily reconfigure controller data, just about anything can be made to control just about anything else.

The other factor in MIDI's favor is that it can be sped up. Not by increasing the baud rate—the Japanese shot down that idea when someone came out with "high-speed MIDI" for sample transfer a couple of years ago—but by splitting the MIDI data into more than one stream.

A synthesizer that's responding to only one MIDI channel doesn't need to see all 15 others, just so that it can filter them out. If note information on one channel is being used to play a drum machine, while controller data on the same channel is running a mixer, there's no overwhelming need for all of that data to be combined on one cable. MIDI is capable of carrying the most subtle of musical events, as long as you don't try to mix too many of them on a single cable.

As MIDI is used for more and more non-synthesizer control, multiple streams will become more important. In fact, it's already generally acknowledged that *any* device receiving both real-time MIDI information and MIDI Time Code will need two separate In jacks, to avoid bandwidth problems.

There are several things that manufacturers will have to do to accommodate multiple-stream MIDI. One is, obviously, to add more In and Out jacks. Another is to include routines in the operating software to let different channels and/or types of data be assigned to different jacks. (This scheme, of course, also has the advantage of creating more than 16 MIDI channels, which some users have been clamoring for, but which I consider a fairly low priority.)

There are, admittedly, circumstances in which one cable will have to carry information at speeds much faster than MIDI, such as between a computer and a multistream MIDI interface. But the nature of this communication is determined by the hardware itself, and the software controlling it. What may be a fine control scheme for a Macintosh II might be completely impractical for a Commodore 64. Therefore, there is no particular reason to standardize it (which would either render the C64 useless for music or cripple the Mac II), and certainly no reason to call any such scheme "MIDI 2.0."

As everything becomes more complex—with MIDI, MIDI Time Code, SMPTE time code, samples and who knows what else flying out the back of a computer—the need for faster communications will become more and more acute. But in a telephone network, every user does not have to know all the intricate digital details of the central switching station to plug in his phone—he only needs to know where the socket is. Likewise, while there has to be a lot of intelligence passed between the computer and the MIDI interface, how it is done is up to the designer of the interface and the software; the rest of us (and our instruments) don't have to know anything except "where's the MIDI?" And that's enough.

R·E/P

Paul D. Lehrman is a Boston-based free-lance writer, electronic musician, producer and regular RE/P contributor.

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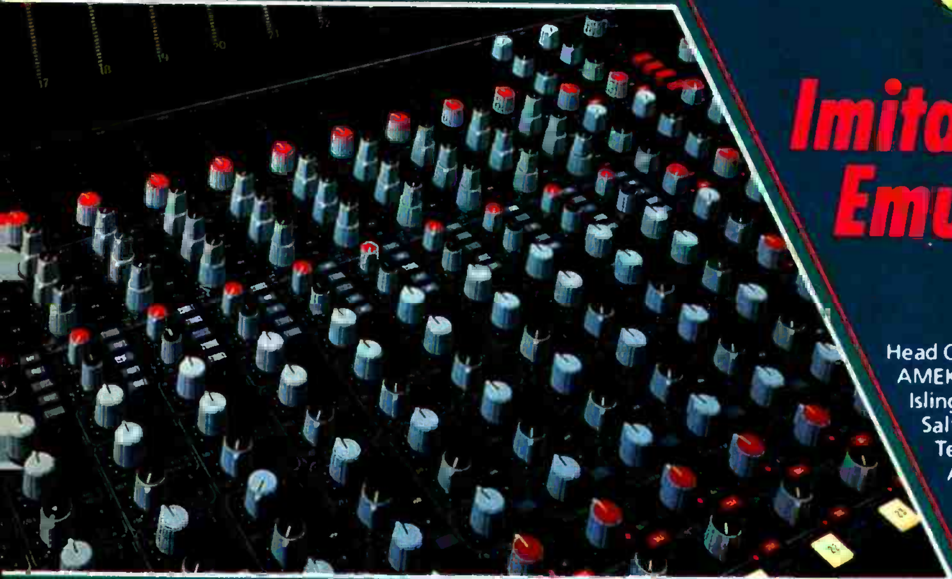

Imitation and Emulation

Today, the mid-range console market is crowded with manufacturers who want to fill the space in the middle of your control room with their console.

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Actually, we admit one concession, but not to our competitors. The new 1987 ANGELA is now finished in grey, to harmonize with the rest of the AMEK product line. Otherwise, ANGELA is still available in 3 chassis sizes each with onboard or external jackfields, giving 28 to 62 dual-channel i/o modules (ie, 56 to 124 inputs, plus returns) and a choice of either 24- or 48-track monitoring in the larger versions. Manual or VCA faders are the standard options, and Audio Kinetics 'Mastermix' is the recommended automation package.

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Sound on the Road

By David Scheirman

The feeling in the air for summer concert touring this season is one of optimism—nearly every recording act from the past 20 years is ready to go out and do live shows. Newly established regional sound companies are finding themselves being taken seriously in bidding for major tour projects. Semi-retired "old roadies" are being lured back onto the tour bus as various musical groups scramble to put together experienced sound crews.

What's behind the increased activity tempo this year is a combination of several different factors, not the least of which has been the fresh input of tour support dollars within the past year or two from both record companies and corporate sponsorship agreements. As these changing financial structures have pumped new blood into the tour business, many concert sound system owners find themselves in the interesting position of having to say, "Sorry, but we're booked up."

This fast-paced summer touring season has come not a bit too soon. The major capital expenditures for new-generation mixing consoles with expanded input sections, and the increasing demand for expensive, studio-quality signal processing hardware, has placed a real burden on sound companies to make some choices that will have a lasting impact.

If console brand "A" is really on the decline in terms of both "trendiness" and quality, then should money and support go in the direction of brand "B" or "C"? Should massive amounts of newly designed, unproven power amplifiers be ordered by the palletfull, or should existing units be upgraded and refurbished? Should the 3-year development project for a new, proprietary loudspeaker system be shelved in exchange for commercially available modular cabinets from a known manufacturer?

Profitable touring projects provide the much-needed cash flow to work out some of these and other dilemmas. There seems to be no shortage of touring groups to accommodate. A primary question this summer, however, would appear to be: Will the early rush of touring acts to take advantage of a music-hungry summer market truly be sustainable, or will it collapse under its own weight? Is the market for concert tickets finite, or is it truly expanding as changes in the youth culture work their way into the mainstream, thus guaranteeing new markets for live music?

Booking agents, personal managers and building facility staff members are perhaps better equipped to offer answers to these questions than are sound company personnel, but one thing is certain: there are a *lot* of shows on the road, with more in the works, and they *all* need a sound system of one sort or another.

The summer season always brings its own set of unique requirements to sound-system work. An outdoor environment, the increased interest in live recordings and multi-media events, and the chance to work in tandem with other sound companies on large festival projects make this time of year an interesting one for live sound.

The demand for high-quality live sound to service this summer's concert tours will be at an all-time high.

Outdoor summer work can be one of the most demanding situations in which to operate a concert sound system. The increased temperatures and exposure to relentless sun and summer showers will instantly make any weak link in a system's chain more apparent. Increased exposure to dirt and dust in such environments as fairgrounds and racetracks means more shop labor afterward for clean-up chores. And the fluctuating ac electrical power sources found at such events can make the sound crew's designated electrical technician one of the most important people on the payroll.

Security also becomes critical in outdoor environments. With as many as a dozen or more musical groups on a festival bill, and the chance for misplaced microphones and direct boxes at an all-time high, knowledgeable sound companies keep a vigilant eye on all parts of the sound equipment present onsite.

Crowd control becomes an issue as well, particulary with the more energetic rock shows. Security personnel should not be taken for granted; they should be consciously brought into production meetings in advance of the events and given information regarding the need for secure mix positions and sound wings.

During the summer, many record companies fund live-recording projects, which means that sound system technicians have the opportunity to work with remote recording trucks. With a bit of well-considered preparation, this can be a painless procedure.

A high-quality program input will benefit not only the recording truck, but also the live sound efforts. Working with the band's stage technicians to clean up the various output signals from electric guitars, keyboards and drum machines helps all the parties involved. A comprehensive, versatile snake and splitter system can mean the difference between a successful show and a miserable one; it can also mean the difference, in the eyes of the event recording and production staff, between professional and amateur.

Cooperation with other sound companies that are often considered to be "competition" during slower seasons makes large outdoor festival dates both educational and challenging for sound company crews. While few companies actually work together in a conscious manner to standardize racks and cabling systems, for the few times that two different firms combine forces for large events, there are certain things that can be done to make these dates go more smoothly. Advance preparation through early communications about such areas as power distribution and stage patching methods should be done *prior* to the event, not while stage hands are on call and waiting for work directions.

As live contemporary music concerts work their way into the cultural fabric this summer as never before, the demand for high-quality live sound to service these events will be at an all-time high. The increasing support of concerts by civic authorities, and the growing number of new outdoor venues that are dedicated to the concert-goers' lifestyle, are guaranteeing that high-level concert sound system suppliers will enjoy an increasing demand for their services.

The 1987 summer season is apparently going to be a time for both the expansion of existing companies, and the creation of new ones. Ultimately, the end of '87 will find the live-sound industry with a larger amount of concert hardware coming into existence than has previously been available. With good luck, the expanding live music market will reach a firm plateau, and not fragment again just at a time when more companies than ever before are making serious commitments to the improvement of audio for live events. R·E/P

David Scheirman is president of Concert Sound Consultants, Julian, CA, and RE/P's live-performance consulting editor.

Film Sound Today

By Larry Blake

Stereo films demand stereo recordings, right? The answer is an almost definite "no" concerning production dialogue, and a hearty "yes" to most ambient backgrounds.

Stereo production dialogue recording has been tried and is generally considered to be more trouble than it is worth. The best-known stereo production technique was the 3-mic setup used during the first five years of CinemaScope. One can only imagine the editorial problems caused by reverse angles and fast cutting, not to mention the headache of mixing while trying to avoid "tennis match" volleys between the actors.

Most of the pitfalls of stereo production recording could be sidestepped with the MS (mid-side) technique, using a front-facing hypercardioid microphone in conjunction with the figure-of-eight side mic. It should be obvious that one should record these signals separately, and not decode them into a left-right stereo pair during the shoot.

Judgment of stereo width, *if any*, is a decision that should be reserved for the final mix, where the flow of the edited scene can be taken into consideration. If the stereo effect is distracting, the side mic can be simply dialed out, resulting in a normal mono dialogue recording.

This is easier said than done, however, and one should take a good, hard look at the compromises involved before opening up the Pandora's box of stereo dialogue. First of all, production and lighting crews usually give the sound team barely enough time to position a shotgun mic; it's a bit much expecting them to allow us to plot out an elaborate move that gives correct left-right positioning relative to the camera's POV.

The use of plant mics, a second boom or radio mics would necessitate that a second recorder be used because the MS signal requires two tracks all to itself. Also, it's hard enough for dialogue editors and re-recording mixers to blend scenes together in mono; one can only assume that a stereo track will "bump" much harder across a cut.

Furthermore, if the "you-are-there" stereo effect works well in the final film, it should be remembered that this will go unnoticed in foreign release unless the production mixer secures a minute or so of the scene minus the dialogue. (This track, of course, will also help the sound editors with the English track.)

Larry Blake is RE/P's film sound consulting editor.

MS production recording would be practical *only* if *everyone*—director, cinematographer, assistant director, picture editor, sound editor, boom operator, transfer person and the dialogue re-recording mixer—was aware of what was going on. The ¼-inch to 35mm mag transfer step is crucial; the control afforded by an in-house transfer bay would help ward off serious problems. Any production mixer who tries to get clever with stereo recording without first consulting the rest of the crew is simply looking for trouble.

There are many sound effects that do not have to be recorded in stereo.

Stereo effects recording

The process of recording stereo sound effects has come into vogue in the past 10 years, and many sound editorial companies pride themselves in an ever-growing stereo library. In spite of the obvious perks of stereo recordings, it should be noted that there are many effects that don't have to be in stereo.

A good example is a car pass-by. While it is possible to shoot an effect for a specific scene in a film, carefully noting stereo position, this usually doesn't pan out (no pun intended). In the long run, a mono recording will suffice and the effect can be easily panned during the mix. An exception during a car series would be an in-car constant, which would do well to be in stereo.

Any production mixer who tries to get clever with stereo recording is simply looking for trouble.

Stereo is also useful for moving "events" like train or plane pass-bys, where there is some length involved. Again, with car movement, fast cutting often makes a stereo effect useful only in mono.

Backgrounds are another matter and, whenever possible, should be recorded in stereo. The ORTF technique, with outward facing cardioid mics, forming an angle of approximately 110°, represents an excellent technique for recording stereo backgrounds. The approximately 7-inch separation between capsules often creates enough phase difference to help the apparent stereo width when mixing to a 4-2-4 matrix; a crossed-cardioid XY recording can often "build-up" in the center. In this regard, if one is recording a stationary background (like a forest), the spaced-mic technique should yield excellent results. Experimentation is the key here, and nothing is carved in stone.

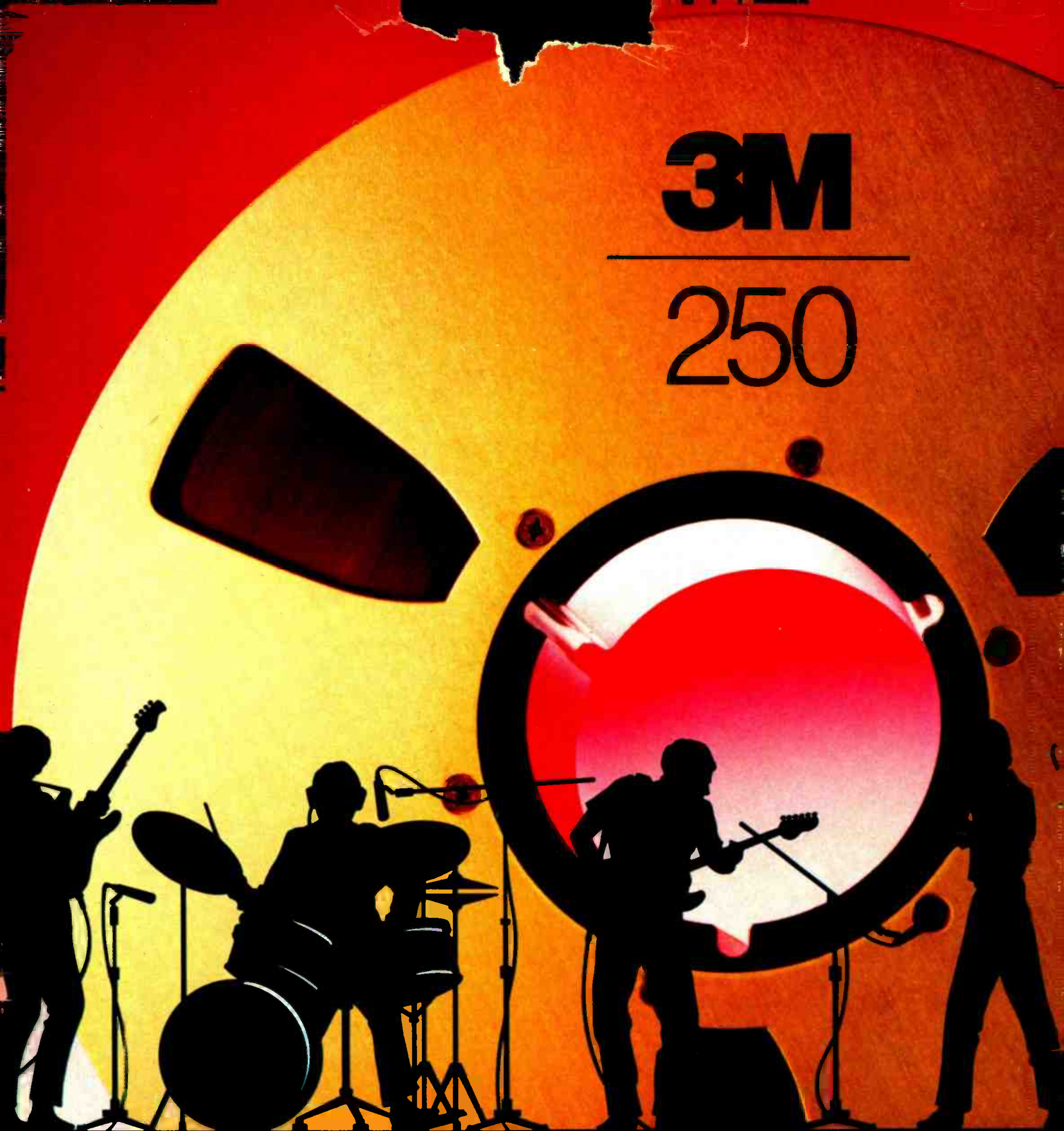
My biggest problem with ORTF and XY recording has nothing to do with the sound: I just find a stereo mic bar with two large windscreens to be a cumbersome and stupid-looking device, not to mention attention-getting. I'm a fan of front-facing MS mics because they make for a mobile and compact recording setup; going to mono is simply a matter of deleting the side information. In addition, the front-facing position is the same as it is in stereo: on-axis.

Reports from friends who have used MS recordings in Dolby Stereo films indicate that MS (decoded into left/right) bleeds more into the surround channel than an original XY recording with an "equivalent" included angle. This difference makes sense as the surround channel is, of course, the difference (L – R) signal, which is neatly provided by the side-facing figure-of-eight mic.

One should realize, however, that the one might then be stuck with this surround information, like it or not, in 35mm stereo, which uses the 4-2-4 matrix. Also, it will not be in the surrounds in a 70mm "discrete" 4-track presentation, unless it was assigned to the rear by re-recording mixers. Again, there are no hard-and-fast rules and, to paraphrase the EPA, your surround mileage may vary.

As I said in last month's column, high-quality cassette recordings, especially with Dolby C, can produce superb results. I'm convinced that the problem with cassettes is not the medium, but that people don't use quality microphones. As is the case with still photography, in the long run the art of sound effects recording has little to do with technology. Research, persistance, setting the scene, willingness to experiment—in any given situation—it is these factors that make the difference.

In next month's column I will discuss the concept of a sound library cooperative that would be supported by production companies worldwide. R·E/P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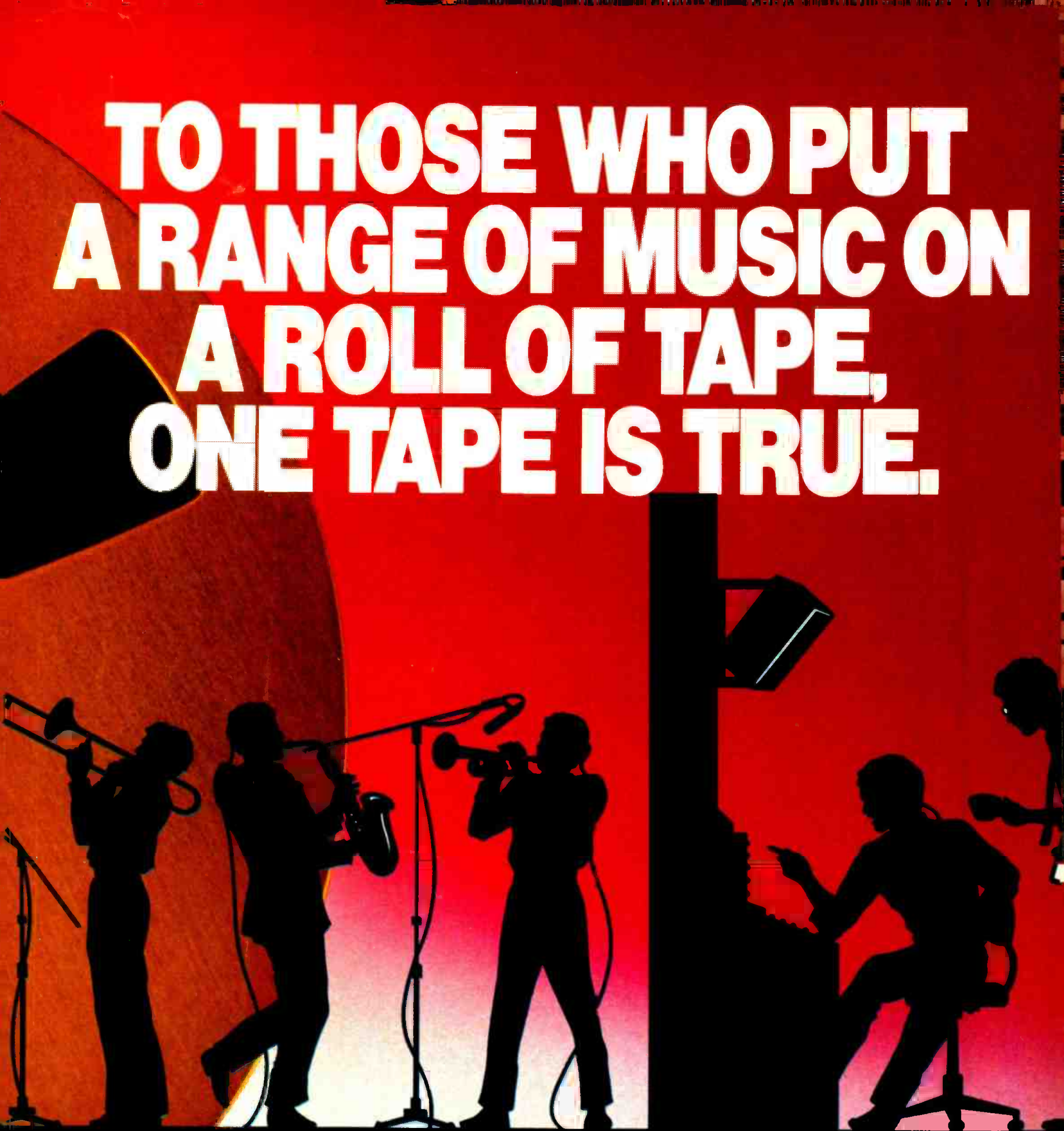
TO THOSE WHO PUT
A RANGE OF MUSIC ON
A ROLL OF TAPE,
ONE TAPE IS TRUE.

Living with Technology

By Stephen St. Croix

I don't know how much you know about computers, but you do know that any system occasionally gets itself into a corner and makes you wait while it works itself through a problem. Some of these waits are barely noticeable, some are a little annoying, while some may result in the computer being sent to its death by leaving through the fourth-story window. (There is an IBM here with a .44 magnum hole in it; I have no idea who could have done that.)

In the course of my consulting, I have done a fair bit of research into just how long a wait is considered acceptable for various conditions. It seems that for a change to appear on a screen, anything shorter than about 60ms (after hitting return) is perceived as instant, while up to 90ms is instant if it takes a while to write the whole screen once it starts. Most people are clearly aware of a delay of anything over 230ms between hitting the return key and having the screen respond.

A system that is capable of responding fast enough to always keep within the 60ms window is perceived as very fast if not instant, and is always preferred, even by those who don't need the speed.

When we start asking our computers to work on audio, a system that looks great running word processing will become taxed to its limit and be unbearably slow when playing with DSP. That 250ms wait every time you insert a paragraph is a lot different than a 250ms wait every time the screen is drawn as you examine an audio splice and drag the picture across the window.

Studio time is expensive; our time is valuable. Audio can do 20,000 different things per second, per channel. Deadlines are twice as close as they seem, and all the associated projects are half as complete as we thought. We need *speed*!

The answer is coming, and it is cool, or should I say cold. Simple new chemical breakthroughs are radically raising the threshold of the onset of superconductivity. This is as important to our industry as was the invention of recording itself.

The first two reasons why computers can't go infinitely fast today are: junction switching speed, and the length of the path that the electrons have to flow to get to those junctions at the speed of light. Light may seem to travel pretty fast; for very short distances it may appear crazy to worry about it when so many other factors appear to be more realistic reasons for the limits of computer speed. But the little time it takes for electrons to travel along a trace accumulates to a noticeable figure after several billion trips over hundreds of such paths.

So, when it comes to developing smaller, more powerful, faster computers, the designers have a double whammy against them: registration (how small can they print those multiple layers on those little silicon chips, and still get everything to line up properly); and *heat*.

The tighter they can print these junctions, the less distance electrons have to travel, and less time is needed for the machine to think of any given thing. This means less time per task, and greater heat per square centimeter. It all gets pretty warm as it is; what will happen when we are able to increase this density? Ever been to Arizona in August?

The real key to 18- and 24-bit audio is superconductive A-to-D converters.

Currently, chips have to be built in such a way as to get the heat out, which limits how small they can be. However, the newest breakthroughs in superconductivity may make it possible to ignore this size limitation. Then, as we learn to accurately make smaller and faster chips, we won't keep hitting the wall of what to do with heat.

Superconductivity makes computers faster. It does nothing, of course, to actually make them faster. (Well, it *"does"* speed up the electron flow along a trace in a chip by allowing the electrons to travel straight along the trace instead of the usual tortured path caused by millions of electron collisions, a phenomenon known as resistance.) It does allow, however, the design of systems that don't have to take heat into account, and can be as small as we are capable of designing and manufacturing.

While all this wonderful speed improvement is now possible, think of this: With true superconductivity, semiconductor junction temperatures will be under control, and trace temperature gains should be zero. With this true superconductivity all temperatures should be stable, which, of course, means chip header designs and that internal real estate can be optimized simply to run, and not compromised to run without melting.

Junction noise would be non-existent. With zero-ohm rails, power supply noise and glitches would be non-existent. (Which means memory without hundreds of capacitors!) Theoretical designs—the ones that come out of the engineering department, or show up in the application books from manufacturers—would actually work.

In addition to all this, with real control over internal chip temperatures the final part of the triple hitter for you (instead of the double whammy against you) is that you can now crank up the old clock speed by 10-100 times. (Speed looks like power by the time it gets to the screen.)

The Cray XMP1 is capable of going quickly (and thereby looking very powerful) because it is run cold. Liquid freon cools this machine. But it is *not* superconductivity. In fact it is nowhere near it. So the XMP1 does generate heat, and this heat does have to be dissipated. This limits its size and clock speed.

With true high-temperature superconductivity (HTS?), it is definitely possible to produce a small desk-top PC with more speed and power than the XMP1.

Booting this Cryo-1000 would possibly consist of starting the pump, waiting three minutes or less for cool down to superconductivity, followed by the system boot. (The superconductivity pump itself can be operated superconductively for even more serious efficiency.)

The little bit of time you lose in cool down would be forgotten the first time you rotate a 3-D color Kline bottle through Nth dimensional space; fly that hot new transdimensional flight simulator complete with animated passengers with the faces of all your friends; watch the waves of a real-time FFT dance across your screen; or whatever it is you like to do with your computer.

Or maybe the way you see it, all this techno-stuff really bytes. Okay, try this: superconductive consoles, summing buses and mic pre-amps: No Noise. The real key to 18- or 24-bit audio is superconductive A-to-D converters. And when this becomes possible at room temperature: HTS cables. Maybe a little help around the studio would be nice; artificial intelligence for the masses can't be more than a few flops away now.

Ever notice how much more powerful (faster) your car is on a nice brisk cool day? This has nothing to do with my column; it is because the number of oxygen molecules available for oxidizing the fuel is higher in the denser charge. But it just goes to show you: Cool *is* Cool. R·E/P

Stephen St. Croix, RE/P's technology developments consulting editor, is president of Lightning Studios and Marshall Electronic, Baltimore.

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SPARS On-Line

By Gary Helmers

This month I would like to take a look at the emerging impact of digital audio in the film and video industries. Traditionally, acceptance of audio progress has lagged behind that of advances in visual technology. This is still true, although the situation is changing and those familiar with digital technology will definitely have a bright future in the film/video industry.

Digital has become a common buzz word in the recording industry. Music studios have been recording direct to digital for the past five years and, with the widespread acceptance and spectacular growth of Compact Discs as a retail medium, the future clearly points to the majority of music existing in digital form from initial recording to home listening.

When you consider the technical advantages that digital audio offers to the film producer, why hasn't it become more commonplace in the recording or re-recording of the soundtracks for motion pictures? There is no tape noise, no wow and flutter, absolutely flat frequency response and the dynamic range is startling. All of these factors result in digital's impressive clarity, from the softest highs to the lowest lows.

Beyond the superior quality of video, there is another advantage digital audio offers for the film medium. In analog, each successive step of post-production drastically raises the noise floor level. It's true that in the last 15 years superior noise-reduction processes have been introduced to the film industry. Sound mixers are still limited, however, in the number of pre-mixes they can make before the noise factor makes any further re-recording prohibitive.

Because a digitally recorded tape contains only numbers, the microprocessors can easily distinguish between the recorded signal and the inherent tape noise. Consequently, digital soundtracks can be bounced—theoretically, at least—through an infinite number of generations, with the last sounding just as clean as the original.

This fact alone makes digital recording extremely attractive to motion picture sound editors and mixers, who routinely premix hundreds of effects, music and dialogue tracks to create the submasters for final mixing. Another advantage is that digital tracks recorded "hot" are not likely to print through.

Some producers have already begun to incorporate digital audio into their projects. In 1982, Disney created the first digital feature-film soundtrack when it recorded the musical score and narration for the rerelease of the animation classic *Fantasia*. Georgia Moroder followed with the release of Fritz Lang's classic silent film *Metropolis*, with a newly recorded digital soundtrack.

Much of the music for major motion picture soundtracks is now recorded digitally and then, in the final stage of post-production, re-recorded to magnetic analog tape. Last year, for *Where the River Runs Black*, virtually all the audio was recorded digitally and the post-production work done in the digital domain.

If the cost of digital cannot be passed on to the film producer in higher hourly rates, the equipment investment will not be made.

If there is such a qualitative and technical advantage of digital over analog, why hasn't digital become the norm in the film industry? The main reason is an economic one. Presently, the motion-picture industry is attempting to keep the costs of features from escalating. The additional costs of recording digitally usually are eliminated from the budget.

Digital machines are expensive, and the final mixing of motion pictures requires at least two recorders. A more complex, or sound effects-oriented motion picture, may call for as many as four or five digital machines. For sound effects editing, dialogue and Foley, more machines will be required.

As you can see, this adds up to a considerable investment on the part of the post-production studio. If costs cannot be passed on to the producer in higher hourly rates, the investment will not be made.

Another obstacle to the growth of digital audio for film is the distribution medium. The producer does not want to pay for the additional cost of the digital technology and then have the film finally released in analog, either on magnetic or optical film.

There have been a number of procedures developed that run dual-system: film in sync with either digital audio stored on videocassette or digital recorders. However, these types of release systems are limited to specialty houses or showcase runs. Until there is a more economical form of digital distribution, and theater owners willing to upgrade their audio systems, there will be little substantial growth in digital audio for the motion-picture screen.

There may be an area where digital will be accepted sooner than in the theaters. As this decade progresses, we are finding that motion pictures no longer exist in the vacuum of the cinema world. Today's feature films are now part of the television and home-video distribution industries. The time lag between the theatrical release of a feature film and its home video release is becoming shorter and shorter. Modular home systems include greatly improved speakers and higher quality reproduction from such formats as VHS and Beta hi-fi which in terms of dynamic range and audio fidelity, comes very close to digital.

The incredible growth of the home video market is revolutionizing the audio environment of the home. Superior audio, whether it be for first-run movies, network television, cable or home video, has never before had such importance. We are dealing with a very sophisticated audience; one that pays attention to developments in noise reduction and appreciates the leap forward created by digital technology. When a viewer is surrounded by a mammoth speaker system and has digital audio at his fingertips, the soundtrack *better* be good.

Despite some of the economic negatives, the next five years will bring tremendous progress in digital technology for the motion picture and television media. A number of major manufacturers are developing equipment that will eventually evolve into tapeless studios. Others are working on methods by which digital audio can be put on optical film tracks, or run in dual systems with CDs or optical discs.

All of these developments, whether in equipment, methods or services, must be economically comptetitive in order to significantly change the direction of the film industry. The visuals have reached a very remarkable stage of development, and the industry looks forward to a future where audio will do the same.

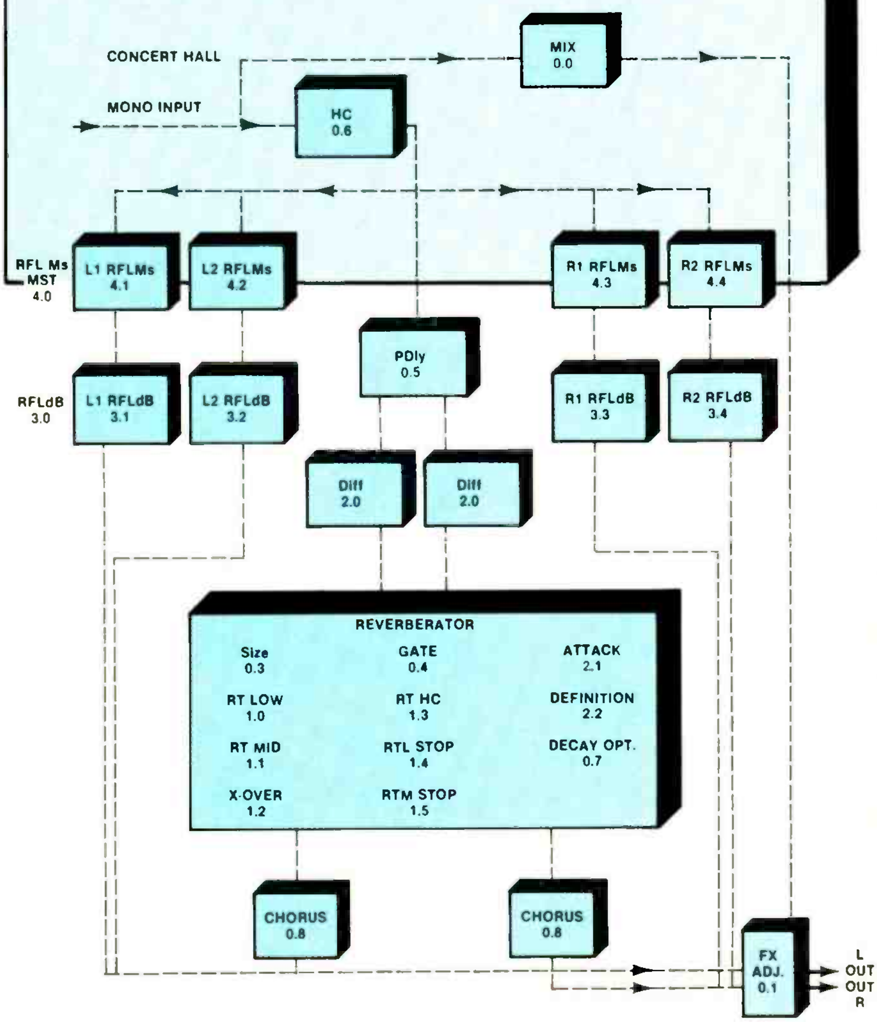
In closing, I would like to express my gratitude to Tom Kobayashi for his help in preparing this column. As vice president and general manager of the Sprocket Systems Division of Lucasfilms Ltd., and former president of Glen Glenn Sound Hollywood, Tom has played a key role in introducing progressive methods of creating sound for pictures. R·E/P

Gary Helmers is executive director of the Society of Professional Audio Recording Studios.

Future Directions in Digital Signal Processing Technology

By Charles Bagnaschi

What technical and operational developments can we expect in the growing field of digital special effects and signal processing units?



Over the past decade digital signal processors and effects units have made dramatic changes in the sound of contemporary recordings, film and video productions. The technology behind those sounds is now evolving at such a furious rate that predictions run the risk of falling behind real developments. Some informed speculation on the directions that future products are likely to take, however, should be of practical interest to all of us. The following discussion will cover five areas that I consider to be of primary importance: signal quality, direct digital interfaces, processor functions, the user's work surface and control interfaces.

Signal quality

We've already experienced an evolution in digital audio formats from 10-bit words, to 12-bit and now 16-bit. The latter now seems to represent a comfortable plateau for most effects processing applications, even though falling costs of digital storage and new conversion techniques will allow larger word sizes to be used in applications requiring lower noise and greater dynamic range.

No doubt many of us would like to see the 20kHz barrier broken. It is not likely that bandwidth will increase beyond this value, however, simply because of the standardization that has evolved around existing 44.1kHz and 48kHz sampling frequencies. On the other hand, there will be improvements in dynamic range and in the more subtle, hard-to-measure areas of sound quality.

Some of these changes will be brought about by an expanded use of oversampling technology. Oversampling is an effective way of dealing with the abrupt phase characteristics encountered near the band limit of sharp cutoff analog filters. The technique is now in digital storage devices such as CD players and random-access editing systems.

Several manufacturers also are working to develop oversampling analog-to-digital converter input stages. These devices will remove some of the characteristics that users perceive to be the contribution of the input filters to "digital sound." Until there's more hardware available for testing and comparison, the exact nature and significance of the effect of such filters will be difficult to establish on an objective basis.

Perhaps a greater benefit of oversampling input A-to-D converters is that they allow conversion to more than 16 bits of dynamic range. By performing A-to-D conversion with a fast converter of less than 16 bits at an oversampled rate, the data can be processed to extrapolate dig-

Charles Bagnaschi is vice president of engineering at Lexicon, Inc., Waltham, MA.

itized representations of 16 or more bits. In this way, dynamic range is increased while still maintaining more than adequate linearity.

Although it isn't necessary to use oversampling to build A-to-D converters that exceed 16-bit resolution, the problems of constructing high accuracy, fast conversion rate A-to-Ds are formidable. With oversampling, however, you can use less expensive analog circuitry and put more of the responsibility for signal quality in the *digital* domain. (Which represents a good trade off because large-scale integration techniques can pack a lot of processing power on silicon. And, as more people use the chips, the costs come down even further, while audio performance goes up.)

Digital interfacing

Manufacturers are beginning to settle on a few sample rates as industry standards. Despite this development, interfacing between units that operate at different sample rates is still quite expensive. There is also demonstrable signal degradation involved in some of the currently available conversion processes.

Another problem exists when you attempt to digitally interface audio processors that output data with a variable sampling rate. (This is a common situation with sampling devices whose data rate changes as a means of altering pitch.) The only readily available solution is to convert back to the analog domain and to resample the data with another A-to-D converter.

Clearly a cost-effective digital solution would be welcomed. What the audio industry seems to want now is a convenient way to connect together various "boxes" in the digital domain, the way XLR-style connectors and standardized signal levels have evolved for analog signals. Whether that involves the use of external devices or internal modifications, the next few years will see a lot of activity designed to remove the uncertainties now encountered when digital audio devices are connected together.

We already have one or two digital interface standards; the one that seems to be emerging as most popular is the AES/EBU protocol. Although it results in a more expensive interface than those proposed by certain individual manufacturers, most companies have conceded that AES/EBU will be universal digital audio language, and have announced intentions to support it.

The cost of the interface will come down as soon as manufacturers begin more realistically pricing their AES/EBU interface chips. With volume usage, the complexity of the interface will be buried in dedicated interface ICs, which should smooth the way for its universal use.

Another digital audio interface protocol gaining favor is the EIAJ standard. It is quite similar in format to AES/EBU, but makes use of low-cost, RCA-type phono connectors. (This latter approach will no doubt gain favor for less expensive processing devices.)

Digital processor functions

Over the past few years, advances in technology have expanded the use of digital processors. VLSI techniques have produced new possibilities for tools that can create new types of sounds. Already, the latest generation of digital processors contain ambience algorithms that produce a closer approximation of reality.

It is OK for one unit to do several things, if it can do all of them well.

Some of the reverb units being built today are capable of producing quite convincing sounds that are interesting to the ear—whether the intended effect is realistic or surrealistic.

The key to the development of new and more realistic sounds is the kind of algorithms that will be written as the possibilities are better understood; these might come from an individual identifying a need for a specific type of processing and writing an algorithm to produce it. Alternatively, a software developer may do something experimental that produces a new class of interesting and usable sounds.

The commercial success of digital signal processors and synthesizers has created a powerful motivation for new developments. Many individuals at university computer music labs and manufacturer's facilities are looking for the next hot item. It's likely that we will see new kinds of emulations of natural phenomena, as well as simulations of effects that owe very little to the types of acoustic behavior we are familiar with.

At the same time, because the hardware to execute these types of algorithms is falling in price, we're seeing experimentation in the home audio market, as well as in professional and semi-pro areas. That development undoubtedly will increase.

Beyond reverb and delay-based effects, which still have room for new enhancements, potential exists for digital dynamics processing (compression, gating and expansion), equalization, sample-rate conversions and noise elimination. Some of these processes are already available, while others will become available in the near future.

As the audio production industry moves toward integrated digital production systems, new possibilities will emerge for techniques that do not need to operate in real time. For instance, it is very difficult to perform high-quality pitch shifting in real time, particularly on stereo or multiple channels. However, if the digitized audio is stored in a random-access medium, a computer can perform the processing in "reasonable time," while the engineers are occupied with other aspects of the production. A pitch-shifted version of the material would be available when needed.

The list of tasks that digital processors can perform is expanding rapidly. Packing more and more functions into every piece of hardware, however, limits the ability of the processor to perform all of them at acceptable quality levels. It also leads to compromises in the human interface that make them more difficult to use than devices that have been optimized for specific jobs.

It is OK for one unit to do several things, *if* it can do all of them *well;* producers and engineers, however, are using more and more effects processors simultaneously. As more units are used in the control room, the need to control them quickly and efficiently becomes more pressing. As a result, there will continue to be a place in the processing rack for boxes that are dedicated to performing one or two high-level functions. Versatility has a place, but not at the cost of audio performance and efficiency in use.

User control interface

Manufacturers are constantly looking for ways to express parameters and effects in ways that allow the user to understand how a particular input interacts with the sound. Computer graphics could be a very powerful presentation medium; price points have fallen to a level where we can start to explore graphics that present the three axes of real space.

Drawing an entire room or hall is not really what's required. A presentation that allows the user to interact with a given class of space to change some of the useful characteristics could provide a viable human interface. At the same time, you want to create a display that allows the user to appreciate key sonic characteristics: the position of the listener, where the sound appears to be coming from, how it might appear to move, how the space is changing, etc.

Video displays of control information are very useful for many aspects of recording, and are going to become more popular as time goes on. For controlling a dedicated digital signal processor, however, something like an Apple Macintosh might be more suitable; a quiet, well-behaved, compact machine that's adept at digital interfacing and offers good graphics.

It might be more efficient to substitute a customized controller for the conventional ASCII keyboard. For most people, tactile inputs, such as sliders, often express parameters better than buttons. We associate motion and "feel" with controlling things, however, which means that a manufacturer cannot remove all of the tactile feedback without making users a little uncomfortable.

Soft-labeled controls have become necessary, due to the explosion of control parameters. If a company builds a dedicated interface into the hardware to do one specific thing, it can be made to do a superb job. But, as people keep pushing the boundaries of the processor with software improvements, a user will eventually run into limitations because of the inability to reconfigure the work surface.

Both remote and wireless controls offer definite advantages. For one thing, the best location for the processing gear is not always where the user wants to be. Separating the control surface from the processing hardware also helps ease the competition for space on consoles and in equipment racks.

While it is important to provide a compact processor and control surface, this should not be done at the expense of ergonomic operation. Current wireless controls still lack good information feedback, part of which is due to the limitation of portable power sources and the expense of providing reliable 2-way communications free from crosstalk and interference.

In terms of the kinds of parameters that engineers and producers will be controlling, we may see some fundamental changes. A few manufacturers have released interfaces that let the user program algorithms directly on an external PC and then load them into the effects processor. To date, this approach has not seen much use beyond universities and in the hands of a few advanced experimenters.

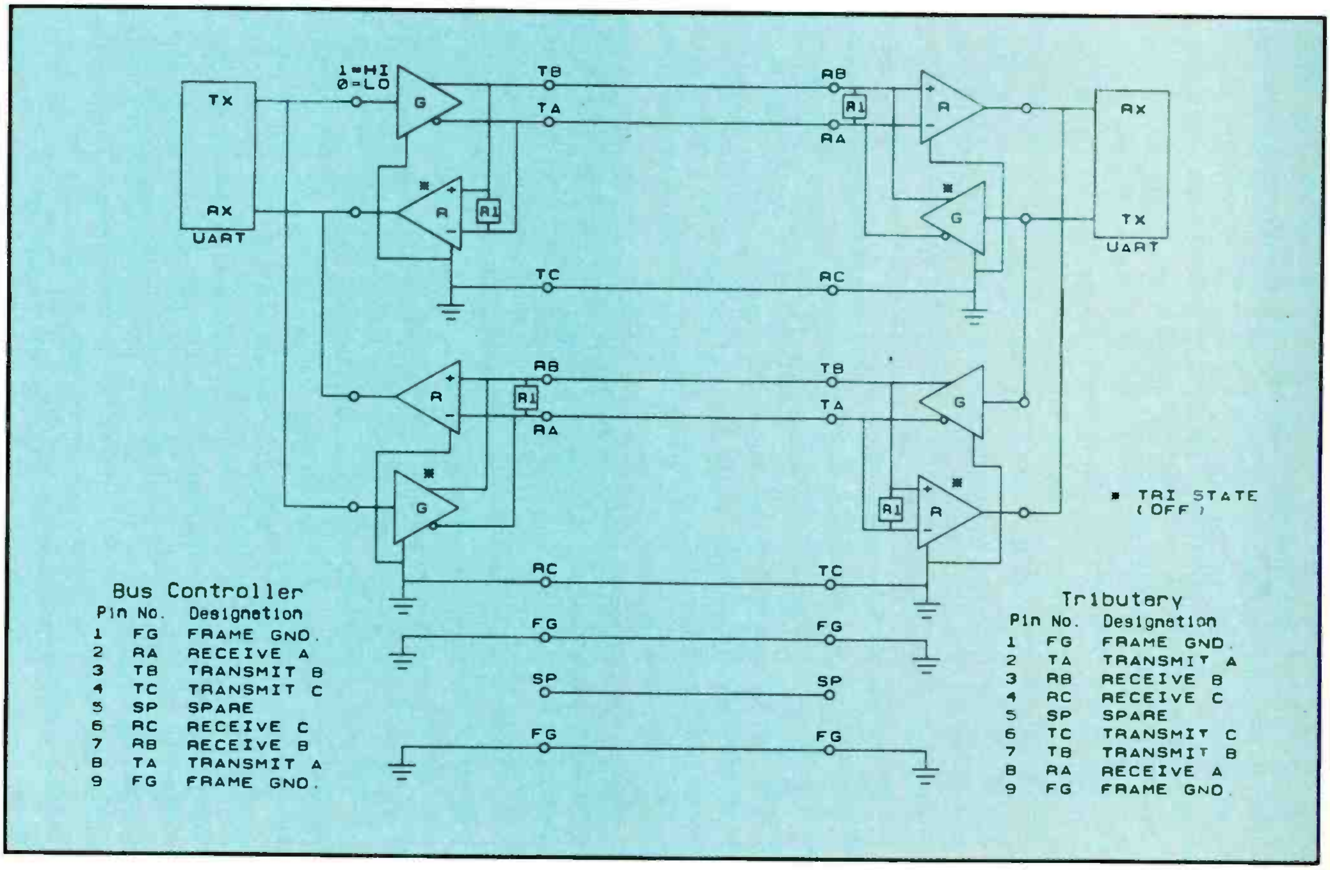
The problem has been the difficulty of developing a good, high-level command syntax, while freeing the developer from hardware-related constraints. Once these problems are overcome we may see a market develop for a programmable effects unit; indeed, it is interesting to speculate on the type of developments that might result from large numbers of users becoming involved in the development of sound modification and synthesis algorithms.

The "program-it-yourself" approach is not for everyone, however. Recall that the first computers were sold as user-programmable devices; they all came with BASIC, and most still do. But how many computer users write their own applications software? The majority of them have been able to get much more out of personal computers by buying off-the-shelf applications software.

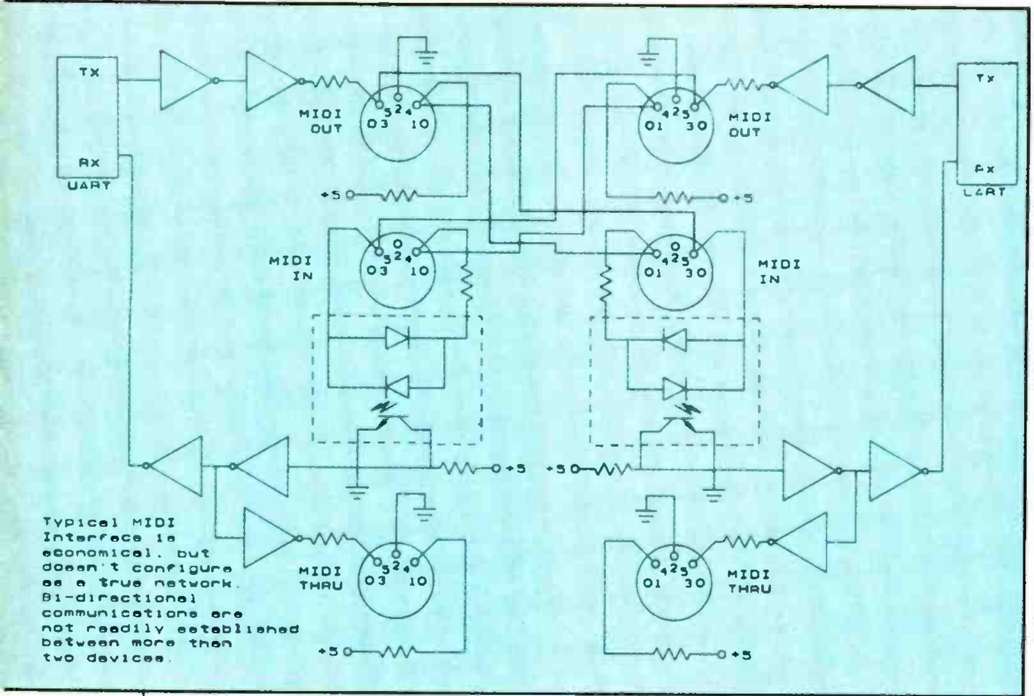
Similarly, DSP applications are developed by relatively few companies. Most users just don't have sufficient time to devote to the complex process of developing reverb or effects algorithms. It could be argued that musicians, producers and engineers have already gone as far into programming as they want to with some of the MIDI sequencers, mappers and sound modifiers currently available. It's unreasonable to ask the broad base of users to go much deeper into the process than that.

Digital control interfaces

The idea of one standardized control interface and protocol that would allow



Schematic of a typical RS-422 serial interface for remote control of digital hardware.



Schematic of a typical MIDI interface which, although economical, does not configure as a true network. Bi-directional communications are not readily established between more than two MIDI-equipped devices.

users to construct universal controllers for digital processors is an attractive prospect, at first glance, but an adequate universal control doesn't seem likely to emerge in the immediate future. It is a formidable challenge to develop a control interface and protocol that will provide for all the eventualities in the fast changing domain of effects processors.

In the future we will start to see new types of control interfaces. As now defined, MIDI is very versatile but does run into limits for certain control applications. The control bandwidth is limited, and MIDI is a daisy chain rather than a true wideband network. MIDI is able to prove some spectacular control capabilities, considering its origins and the limited function it was intended to perform. The interface is very well suited to low-cost gear, and for use by people that need to exchange only a fixed amount of control information in real time. It will also be around for a long time.

At the same time, there are other needs in the studio and already other control buses are being used to meet them. For example, the SMPTE/EBU ESbus interface based on the RS-422 electrical standard has been adopted by broadcast and pro-audio manufacturers. ESbus offers more of a networking standard; a user can set up a master controller device and a group of listeners, any of which can change roles and take over the network.

RS-422 interfaces are somewhat more expensive to implement from a hardware point of view, but standards rather than cost present the major problem. All interface protocols consist of three layers: hardware definition, message format, and message content. The SMPTE/EBU interface succeeds with the first two layers but, as it now stands, it is still difficult for products from different manufacturers to communicate with one another because of inconsistent message content and meaning.

In the future, it would be very useful to have available a good network standard for processing devices that could combine control real-time audio data and perhaps some video channels. Although this sounds like a difficult task to achieve, much of the basic groundwork has already been done within the computer and broadcast industries. Such a network is likely to use either a coax loop or even an optical data path. (Fiber optic links will appeal to anyone who has struggled with ground loops, noise and electrical safety problems when working in demanding performance situations.)

Developments worth watching

DAT will accelerate the demand for "studio quality" audio at "home recording" prices. In the past, most processing gear was used as an enhancement on the stereo mix. Unfortunately, for years we've been using hardware that did not offer the same technical specs as our stereo mix, but was fine for sweetening and enhancement. The resultant audio quality compromises were, to a large extent, economic decisions necessitated by the competitive aspect of the market.

The good news is that the widespread acceptance of Compact Disc technology has provided a spectacular cost reduction in D-to-A conversion hardware. Low-cost technology will now percolate up to the lower-volume effects processing markets. In addition, new technology now being developed for high-end products will soon work its way down to lower-priced devices.

Another development in audio processing is the evolution of low cost mass-storage devices. In the next few years we can expect advances in both magnetic and optical storage technologies. Although these advances have been driven largely by the computer industry, they fit very nicely with the need for large-scale digital audio storage.

Another area to watch will be the development of both general-purpose and application-specific signal processing ICs. Smaller geometries have reduced power requirements and increased speed to the point that real-time signal processing is economically viable for many more audio applications. At the low end of the market we are seeing increased reliance on full custom LSI circuits to produce very capable and cost-effective signal processing devices. This movement has been helped by improvements in computer-aided design tools for IC production, and by rapid improvements in integrated circuit manufacturing techniques.

It's important to recognize, however, that improvements in signal processing and data conversion do not, in themselves, produce better, more usable products. Both audio quality and good ergonomics are going to be demanded by a growing group of users. At the moment, many manufacturers are putting a lot of emphasis on bells and whistles, at the expense of performance.

A lot of people are smart enough, however, to know that good sound and ease of use are really what you want out of a box. If those needs are met, the box is going to be a good tool.

Good products are not developed in a vacuum. The design process for any product is structured to meet the needs of the ultimate user. If they are to be economically viable and musically useful, digital effects processors must satisfy the desires of studio owners, musicians, engineers and producers. Effective, reciprocal communication between users and designers can shape the many decisions and compromises involved in the development process to yield the most effective tools for the future.

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AMS Audiofile Digital Random-Access Recording and Editing System

By John Binder

Let's face facts. When picture elements originate from film, the majority of audio is produced using traditional film mixing techniques.

Why? Because, up until now, nothing has been able to improve upon the convenience of mag film dubbers when it comes to changing the timing relationships between audio elements.

John Binder is a staff audio engineer at Editel/Chicago.

However, a number of audio post-production facilities, including Editel, have been sweetening video on a 24-track for quite some time. When a track needs to be slipped against the picture, it is re-recorded with the appropriate time code offset. This procedure takes time—time that the client doesn't always have. We needed a fast, effective way to slip tracks, while maintaining audio quality. The ability to work with time code numbers from a CMX edit decision list was also a priority.

One solution we toyed with was installing a video editor, five VHS videocassette recorders and five EIAJ-format digital audio processors. This setup would meet all our quality and track slipping criteria, but we'd still waste a lot of time winding tape. Maintaining five individual transports was another drawback. Eventually, we decided that a random-access, disk-based recording and editing system was required.

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Roy Thomas Baker

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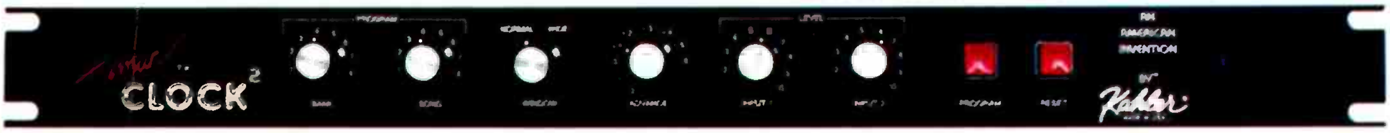

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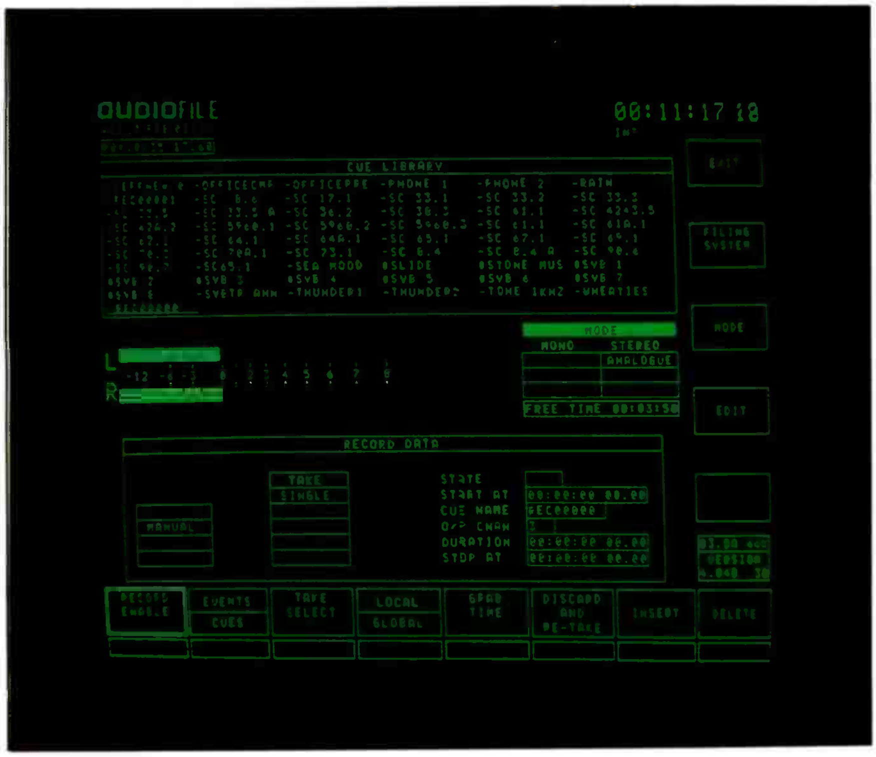

Figure 1. *Cue Record Screen.*

The AMS AudioFile uses a different approach to video sweetening than standard machine control/list management systems. It is not a synthesizer, but rather a powerful post-production tool that uses time code as the basis for simpler audio manipulation. We didn't need to buy a system that would also process the audio; we already had a roomful of available signal-processing equipment. More than anything else, we needed control—AudioFile gave us all that we needed.

The system can sample up to two hours of full-bandwidth mono audio, or one hour of stereo; four hours of record time is available upon request. All data concerning the audio cues is contained in an events list. The timing relationships of each individual cue to other cues can be redefined quickly and easily at any time during the session, with minimal keystrokes. Editing is accomplished in the digital domain, and is non-destructive. Playback of any cue (as AMS refers to each individual recording) is possible over any of eight analog outputs.

AudioFile allows for control of a ¾-inch U-matic VCR through an RS422/RS232 serial port, which operates in the range of 9.6kband to 76.8kband. Audio can be synchronized to this VCR master, or any other source of external time code.

Hardware configuration

The system consists of three rack-mountable units: an external power supply, mainframe and control panel. The power supply and mainframe can be installed over 200 feet away from the control panel, to prevent fan noise from interfering with audio monitoring.

The mainframe itself is remarkably small, consisting of only 5U (8¾ inches) of rackmount space. Located inside are the system's two 380Mbyte hard disks on which all directory and audio information is stored. (An additional mainframe is necessary in the 4-hour recording system.)

Audio is sampled into the system via two A-to-D converters that use a 16-bit linear PCM encoding format. Sampling rates can be supplied as either 44.1kHz or 48kHz; other rates are also available through the factory. Output channel assignment is implemented in the mainframe unit and routed to any of the eight selected D-to-A converters. Although all eight D-to-As are active, currently only five simultaneous outputs are possible. (AMS projects that eight simultaneous outputs will be available in mid-1987.)

The control panel interfaces with the mainframe unit via an RS422, 38.4kbaud serial port. A high-resolution video monitor displaying 62 lines of text provides the user with necessary information on the parameters to be controlled. To the right and below this video monitor are 13 softkeys whose functions change as different display screens are called up. A set of six dedicated transport controls allow the system to be operated like a standard tape machine.

In "global" or "external" mode, the digital audio is synchronized to time code from the external ¾-inch U-matic. A numeric keypad is available for addressing time code numbers, output channel assignment and offset entry. Two continuously variable knobs on shaft encoders control cue and events list scrolling, edit point trimming, and jog and shuttle control of the external U-matic.

Below the video monitor are eight cue triggers, labeled T1 thru T8. Each key can be assigned to a specific audio cue and the latter replayed at a specified internal or external time code location. The start and stop times of each trigger can be logged automatically into an events list. AudioFile also allows for remote triggering via eight optically isolated switch closures.

A 3½-inch floppy disk drive enables software updates, as well as storing and retrieving events lists. Cue and event lists labeling is handled on a conventional QWERTY keyboard. (It should be mentioned that AMS will be able to supply details of its RS422 interface protocol to allow a user or manufacturer to interconnect with other computer-based systems.

Technical Specifications

Audio inputs: *two, electronically balanced, 10kΩ, line level.*
Audio outputs: *eight, electronically balanced and ground compensating, line level, 100Ω.*
Dynamic range: *96dB*
Frequency response: *10Hz to 20kHz, ±1dB at 48kHz sampling rate.*
Distortion: *better than 0.03% at 1kHz, full output; 44.1, 48kHz, others available.*
Tape transport control: *various Sony and JVC U-matics via RS422/RS232 output port from 9.6kbaud to 76.8kbaud.*
Audio triggers: *eight independent switch closure inputs per channel.*
Backup: *Sony PCM-701/601 format; others are available.*
Time code: *read input levels −50dBm to +10dBm; 24, 25, 30 fps and drop frame; outputs all frame standards, selectable level and rise time; internal generation or slave to external references.*
Sync time: *1 second, approximate.*
Control panel dimensions: *standard rack, 19-inch wide, 10½-inch (6U) high, 12½-inch deep.*
Main electronics: *standard rack 19-inch wide, 8¾-inch (5U) high, 17-inch deep; may be mounted up to 200 feet from control panel.*
Power supply: *standard rack 19-inch wide, 3½-inch (2U) high, 12½-inch deep; must be mounted near main electronics unit.*

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Figure 2. Events Record Screen.

Figure 3. Cue Edit Screen.

Figure 4. Events Edit page.

Figure 5. Cue Library Screen.

(We've found the standard panel to be very easy to work with.)

Software-driven screens

Each major function is addressed by its own menu-driven screen display. Presently, these include Cue and Events Record; Cue and Events Edit; Cue Library; Events and Filing System screens.

The *Cue Record Screen*, shown in Figure 1, is used to enter audio cues into the system. Several input modes can be selected including analog mono, analog stereo, EIAJ-format mono (PCM digital), or EIAJ-format stereo. In the latter mono/stereo digital modes, the audio is input directly via a video-format digital bitstream, thus avoiding additional D-to-A and A-to-D converters.

Record levels are monitored with two peak program meters displayed in the center of the video screen. Should audio levels ever exceed the system's headroom, a visible clip graphic is displayed. Record time still available on the hard disks is updated dynamically throughout the recording process. Cues may be recorded with either local or external time code labels.

Once the recording is completed, a name may be assigned to the cue via the keyboard. If no name is given, AudioFile defaults to a cue name of "rec" [number]. The output channel designated for the cue may also be specified on the cue record screen. Re-recording "blown" takes is a simple procedure, involving the use of a Discard and Re-take function. Given the speed of random access provided by the system, any cue may be auditioned instantly, to determine whether or not it's a "keeper."

The *Events Record Screen*, shown in Figure 2, allows the operator to record while playing back from the events list. In this way, a final mix can be recorded into AudioFile as a separate cue. This task can be accomplished with a programmed time code start or a manual punch-in. Another use is to store console automation data with the events list.

The *Cue Edit Screen*, shown in Figure 3, allows for non-destructive editing of recorded cues. On this screen display the name of the cue to be edited is highlighted, and its audio content is represented by a graphic located at the bottom of the screen. The graphic takes the form of a piece of tape that moves across a "reproduce head" while it is being played. User-friendly editing provides the operator with full transport control over this piece of virtual "tape."

Extraneous audio may be deleted from the head Cue In and tail (Cue Out) of each cue. (These terms are important in discussing audio editing.) The portion of the cue to be kept is represented by a shaded portion of the tape graphic described above. Currently, edits are performed with a Grab Time feature. By using the Play Up To and Play From softkeys, edits can be trimmed to an accuracy of 0.1 frame (3ms).

Future software updates will include "rock-and-roll" location of cue edit points with the control pots; variable ramp-in and -out slopes; level adjustments; and balancing of stereo cues.

As mentioned, the Cue Edit Screen includes a Cue In and Out function that governs what portion of the cue is to be replayed. The *Events Edit* page, shown in Figure 4, adds "List In" and "List Out" to these capabilities.

The difference between the designations Cue In and List In is the same as that between video EDL categories Play In and Record In. The similarity is not accidental; one of the major advantages of AudioFile is, that the system like video editors, uses time code numbers to perform digital edits of the audio material.

[For the sake of clarification, it should be remembered that a "cue" in this context represents the original source material recorded onto the hard disk; an "event," on the other hand, is the same or shortened material from the cue now contained in a list of related events. In other words, the same cue might appear in the same or edited versions throughout a list as discrete events. The advantage offered by a disk-based random-access editing system, of course, is that events comprising of identical cues need only be recorded onto the hard disk *once* and will simply be replayed at the appropriate time code or trigger points—*Editor.*]

Once an events list has been created, cues contained in that list are now known as events and can be re-edited several different ways, while leaving the original cue edit intact. This task is done in the Events Edit Screen, which is one of the system's most powerful utilities. In many ways, the Events Edit resembles the Cue Edit Screen, because it displays the tape graphic and has the same editing functions.

The Events Edit page allows the operator to control when and what portion of the cue will be played back into the events list by manipulating the cue, and using List In and Out times. Because the editing process is non-destructive, List In and Out points may be easily trimmed per client request.

The *Cue Library Screen*, shown in Figure 5, lists all the cues, in alphanumeric order, that are presently stored on the hard disk. Any cue that is auditioned will play back only the portion selected in the cue edit process. From this screen,

cues may be assigned to trigger keys for playback. The "record trigger" function logs the start and stop times of cues played from these triggers, while using either internal or external time code. Cues may also be manually inserted into the events list with the insert key.

The Events Screen, Figure 6, is where all these operations come together. The upper portion of the screen displays the events list in a vertical format. Named here are list start time, cue names, output channels, cue durations and list stop times. When a cue is slipped, the stop time changes automatically, depending on the duration. The output channel can also be re-assigned at this point.

In the lower portion of the display screen in horizontal graphic time line representing all eight channels. Cues are represented by shaded blocks that move toward a "reproduce head" while the events list is played. A visual representation of the location and timing of cues has proven to be an invaluable aid during mixdown. While the system is in global or external mode, transport control is assigned to the VCR and the events list is locked to its time code track.

To add a sound effect, the VCR can be jogged to the appropriate video frame and the external code stored by using the Grab Time function. Any cue in the cue library can be inserted into the list, and its location specified by the grab bed time code value. AudioFile's random-access ability makes this process clean and quick.

When the events list has been completed, it can be stored on a 3½-inch floppy using the Filing Screen, shown in Figure 7. Each list file is stored with its own name and can be recalled at any time. Other functions available on this screen include event list merging, event deletion and list time code offsetting.

Figure 6. *Events Screen.*

Figure 7. *Filing Screen.*

Author John Binder in Editel editing room.

Audio editing for picture

Maintaining high quality audio for video has always been a rigoruous task. Since Editel took delivery of the AudioFile system, our task has been simplified considerably. The unit has proved invaluable in our Express Transfer System, a post-production method that allows clients to shoot on film but edit with the speed of tape.

In the past, audio had always been the slowest step in the process, which led to a dangerous temptation. After the film-to-tape telecine transfer, we synchronize the audio-shoot elements with the videotape. These "scratch elements" are then taken to an off-line video editing system, where an off-line master and a video EDL are produced.

At this point, the temptation might be to pull the sync lines off the video master and sweeten them. However easy this process might be, it adds generations to the process; tape generations and added noise that we want to avoid.

Now we can avoid the tape generations, retain original quality and reduce editing time. At Editel, we go back to the original shoot elements and load "keeper" takes into AudioFile with external time code. Using the video EDL, we conform these audio takes to the off-line video master. This sequence is handled in the Events Edit page, simply by entering the video EDL numbers directly into the Cue and List Cut points. All conforming of audio tracks takes place completely in the digital domain.

Because a butt-splice editing routine is not available in the system's software, sync lines need to be checkboarded across two channels. After the sync lines are cut, the music, sound effects and voice-over are added. We mix back into AudioFile using the Events Record screen. That way, when the time comes to lay back onto the 1-inch video master, the audio can come directly from AudioFile.

We are awaiting installation of an archiving update that will allow us to download the digital audio cues associated with an events list. Cues and directory information will be stored directly onto videocassette using in the Sony PCM-701/601 digital format. Currently, we back up audio jobs onto our analog 24-track. A CMX interface would also be advantageous, eliminating manual data entry. AMS has informed us that it is working on editing butt-splice and cross-fade routines that will allow cues to be merged; these upgrades will be incorporated into a Cut and Paste screen.

In terms of storage time, we have already handled 30-minute industrials with disk time remaining. Although additional storage has been promised in the future, we have found the present system configuration to be sufficient.

When Editel/Chicago purchased the AudioFile we did so with excitement, but also with apprehension. Because our system represented the first U.S. delivery, we feared that it might be racked with flaws and not live up to our standards. We also knew that the manufacturer was based in England and wondered if the support we needed would be available in the event of a failure. As it turned out, there was little need for concern.

AudioFile worked out of the box, and was put directly to work on some spots for the Super Bowl. A conservative estimate of the time saved to do the job was 30%. Needless to say, because our clients were happy, we were happy that AMS has been superb as far as support goes. Software updates have been provided at no additional charge.

Clearly AudioFile's forte lies in the fact that it provides us with incredible speed and flexibility, while actually increasing the quality of our final product. With this production tool, the audio quality we are able to supply to our clients on their 1-inch C-format video master is just one digital generation away from the audio elements that come into our room.

The most appealing aspect of the AMS AudioFile is its versatility. At Editel, we have used it for digital film mixing, Express Transfer, industrials and audio laybacks. This one unit allows us to record, edit and conform audio entirely in the digital domain.

What makes me appreciate the AudioFile system the most, however, are the years we've had to do recording and editing on tape. R-E/P

Photos by Ellen Pinkham.

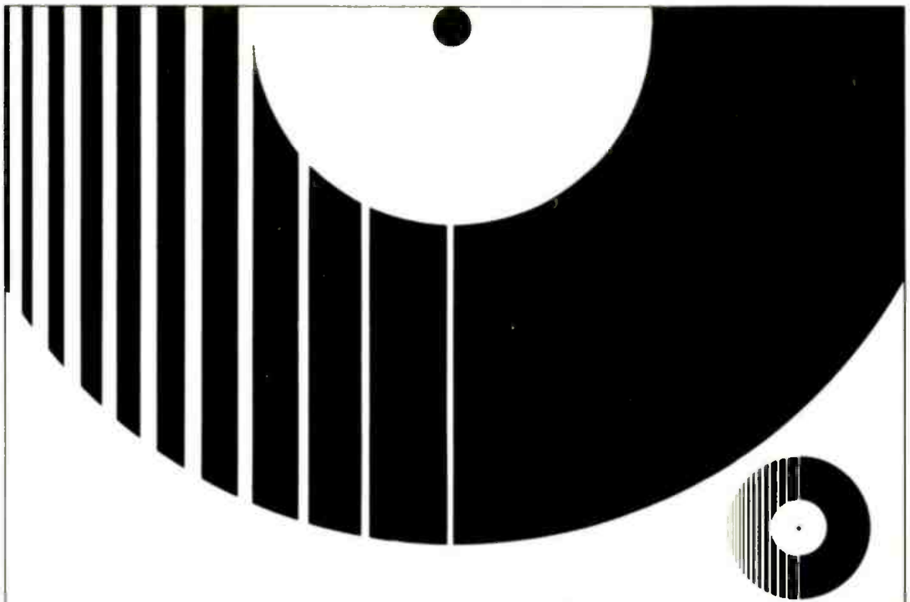
Stop Press: System enhancements announced by AMS

Since John Binder of Editel wrote his in-use assessment earlier this year, all updates mentioned in the report are now available in the United Kingdom, and will be made available shortly to AudioFile owners in the United States.

The maximum storage time currently being offered is just short of eight hours, although the company claims that technically the storage can be increased in 2-hour multiples to whatever maximum a customer may need. Systems released earlier in the year were able to offer up to five simultaneous outputs, whereas now all eight are available.

A new Cut-and-Splice edit page has also been added, which permits the type of butt-splice editing John Binder refers to in his report. This page has been developed with the help of several British mastering facilities who are now using it as a stereo editor for record and CD preparation.

Back-up or archiving is also available with Data-Dump. The entire cue library, individual cues or an events list can be dumped from AudioFile via a slightly modified Sony PCM-601/701 processor to videocassette. Along with the audio cues, titles and time code information is also dumped, so that on reload, work can begin exactly where it ended at the finish of a previous session.



An All-Digital Nutcracker:

Adapting Digital Technology for 3-Machine Mixing and Editing

By Bob Katz

Conventional digital editing systems are normally configured to handle a single master and slave VCR. It represents something of a technical challenge to modify such a system to accommodate a pair of replay VCRs working with a master deck in a 3-machine mix and edit session.

Executive producer Ward Botsford (left) and session engineer Bob Katz at the Neve console and DAE-1100 editor controller during production of The Nutcracker Suite *in Caedmon Record's control room.*

Caedmon Records is the oldest spoken word record company in the United States. Although Compact Discs are routine for Arabesque Records, Caedmon's music division, they are still rare birds for spoken word product. That's why I was both pleased and surprised to hear executive producer Ward Botsford's request: Could digital equipment be adapted to mix and edit a proposed Caedmon CD of *The Nutcracker Suite*?

Botsford had digitally recorded Tchaikovsky's music in London, masterfully performed by the Philharmonic Orchestra led by Michael Tilson Thomas; in Caedmon's New York studio, he had digitally recorded E.T. Hoffman's *Nutcracker Story*, delightfully read by Christopher Plummer into our M-S stereo microphone.

Now he wanted to know if I could put together a system which would control one VCR replaying the PCM-encoded voice track, another playing the PCM-encoded music, mix the two together, and even perform pickup edits on the mix when necessary—all in the digital domain. Suddenly I was involved in an uncommon engineering challenge. In fact, I soon learned that this was to be a digital first: no one had ever taught the Sony PCM-1610 and DAE-1100 editing system to dance like this.

Before I was ready to say "Yes, we can

Bob Katz is a New York-based independent recording engineer specializing in classical, jazz, folk and spoken-word recordings.

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In somewhat colorful comparative terms, Peter Mengaziol of GUITAR WORLD wrote, "The ESQ-1's sound combines the flexibility and analog warmth of the Oberheim Matrix-6, the crisp ringing tones of a DX-7, the realism of a sampler, the lushness of a Korg DW-8000 and polytimbral capacity of the Casio CZ-1".

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MUSIC TECHNOLOGY's Paul Wiffen had a great time mixing colors with the ESQ-1's 32 on-board waveforms and 3 oscillators per voice. "After a few minutes of twiddling, you can discover that, for example, an analog waveform can make the piano waveform sound more authentic, or that a sampled bass waveform can be the basis for a great synth sound. Fascinating stuff!"

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Circle (21) on Rapid Facts Card

Three Sony BVU-800 U-matic decks and a Time Line Lynx time code synchronizer (top) used during the digital editing and mix stages.

Stacked in the rear of Caedmon Records' control room (top to bottom): Sony PCM-1610, 1630 and a second 1610 digital audio processor, plus the electronics cabinet for the Sony DAE-1100 digital editing system.

do it," I inspected the video, audio and digital patching of the digital equipment rented by Arabesque, and made a cursory block diagram of a proposed modification. I then explained our proposal to Sony's field representative, who agreed that my theories seemed sound. But, until tested, there would be a degree of uncertainty, since no one had ever tried this before.

My idea was to use the DAE-1100 editor to the limit of its abilities. Although the editor can control two playback VCRs, it can only start one player automatically in preparation for an edit. I could put the PCM-encoder voice tape on this VCR, which would provide repeatability of entrance (to the microsecond, as a matter of fact) as well as flexibility—we would be able to "slip" the voice track against music.

It was untenable to consider manually starting either the voice or the music in a digital 3-machine mix situation. First of all, there was the unpredictable delay between starting a VCR and the onset of sound. And, more important, we planned to use the digital editor to perform pick-up edits during the mix.

In order for the DAE-1100 or any other digital editor to perform an edit, there must be an editing rehearsal, during which the operator examines the digital bit stream before and after the edit point. This digital information is stored in internal RAM and used to perform a crossfade at the actual edit location.

It is absolutely necessary that the editing rehearsal *exactly* match the real edit, or there can be a glitch at the edit point. In our case, we would have to ensure that the music and voice before the edit point were identical both during the rehearsal and the real edit.

The only solution, therefore, was to control the second VCR holding the PCM-encoded music with a time code synchronizer.

Ultimately, these turned out to be relatively trivial problems, but one question would haunt us until the end: Could we make the digital editor cut into a composite audio mix, as opposed to the straight output of a single VCR?

The next step was to set a date for the mix/edit session, and arrange for A.T./Scharff Rentals to deliver the equipment needed to supplement the existing PCM-1610, DAE-1100 editor and two BVU-800 U-matic VCRs. We ordered an additional BVU-800, a PCM-1630 processor and a Time Line Lynx Synchronizer. Caedmon's Neve console and EMT 245 digital reverb would be used to mix and sweeten the voice and music material in the analog domain.

With the technical arrangements under way, I sat down in Botsford's office to hear about the project. An analog version of *The Nutcracker* had already been produced for the 1985 Christmas season. The 3-record (or 3-cassette) set was turning out to be one of Caedmon's best sellers, with 45,000 copies sold as of January 1986. Now the label wanted to make a Compact Disc version, a deluxe 2-CD set. Naturally, I asked him why he hadn't thought about mastering the CD from the analog ¼-inch (which would also save 100 potential man-hours of 3-machine mixing and editing)? He answered my question by explaining that he wanted to avoid the accumulated tape hiss from an analog voice tape and an analog music tape. It would be improper to dub that to CD, he said.

The first step in producing the all-digital *Nutcracker* would involve re-editing the 1610-encoded originals of Christopher Plummer's voice, conforming them to agree with the studio takes used to produce the original, edited ¼-inch master. The next step would involve the music replayed from Botsford's edited 1610 master tapes of the score (and sold to CBS Records for a music-only version).

For the voice-over version, the music would have to be reorganized and re-edited to suit the story, and perserving Botsford's original creative mix. We mapped out the following schedule:

Monday and Tuesday: Ward Botsford would digitally edit the voice master tapes during the day and in the evenings I would patch in the extra equipment and experiment until it works.

Wednesday: Botsford and Katz would edit the music, using the original ¼-inch as a reference for where the music pauses occur.

Thursday and Friday: We would mix.

The Weekend: Backup days.

It turned out we would need all seven days and nights, plus the following Monday to dub the CD master videocassette that was sent to Japan for Compact Disc manufacture.

On the appointed Monday, the equipment arrived. I patched in the normal DAE-1100 system and left for other chores. When I returned at 5 p.m., Botsford reported that he had only managed about two hour's worth of voice editing, what with constant interruptions by office business (does this sound all too familiar?). Obviously, we would never get the project done that way, so he promised to install a "Do Not Disturb—This Means You" sign on the door of the control room for Tuesday.

Monday evening was to be the ultimate test of my theories: Could I make the DAE-1100 edit the output from Caedmon's Neve console? The equipment certainly would accept a mix, for I could


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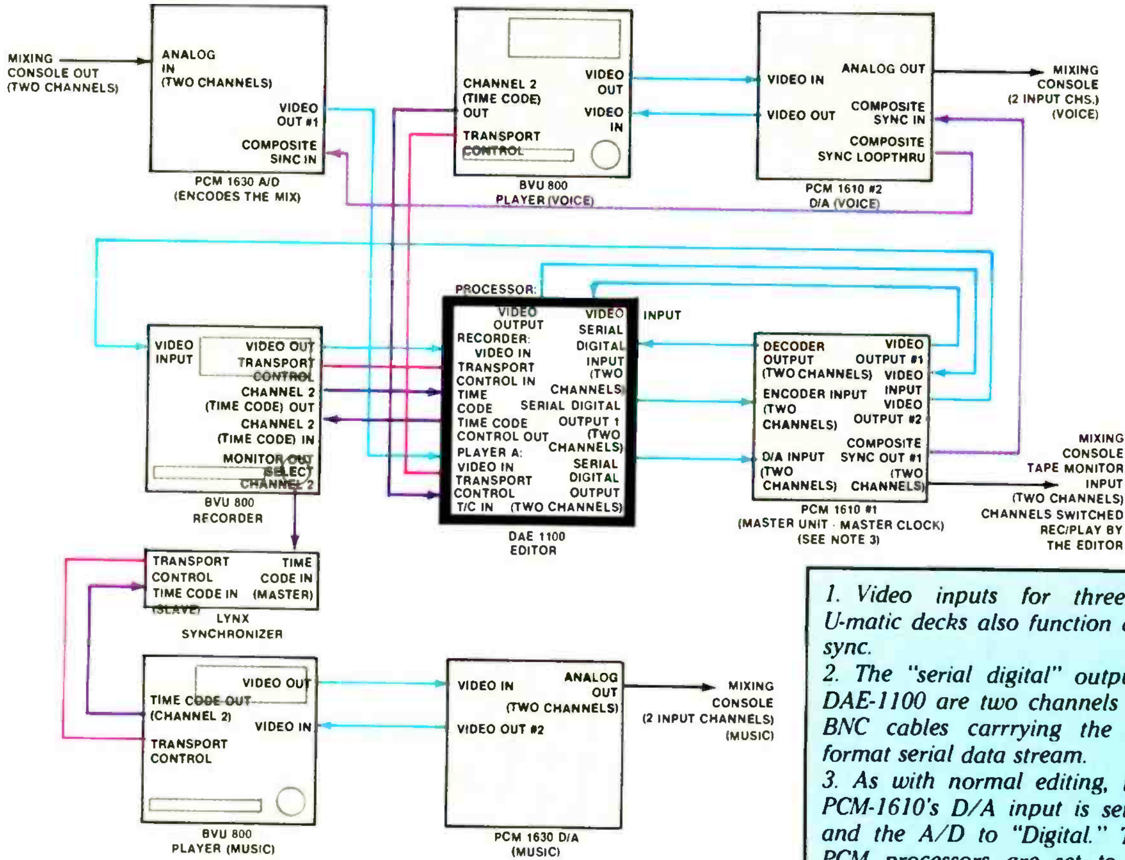

Block diagram of three PCM digital processors, companion U-matic videocassette recorders and master editing system used during the production of The Nutcracker Suite.

1. Video inputs for three BVU-800 U-matic decks also function as external sync.
2. The "serial digital" outputs on the DAE-1100 are two channels of BNC-to-BNC cables carrrying the AES/EBU-format serial data stream.
3. As with normal editing, the master PCM-1610's D/A input is set to "Ext," and the A/D to "Digital." The replay PCM processors are set to "Int" and "Analog."
4. The Time Line Lynx synchronizer is set to "Code-only Master" mode.
6. Composite Sync Output 2 is identical to output 1, and available for expansion.

feed console tone into the 1610's analog inputs. When I tested the editor, however, my digital theories came up all zeros: the 1100 kept locking up, putting out nasty error codes in little red letters.

To say the least I was disappointed. Nevertheless, I spent the rest of that evening studying the DAE-1100 and how it worked, pulling cables, trying all kinds of cross patches. Eventually, I learned more about the internal block diagram of the editor than the average field service engineer is expected to know.

Tuesday morning I gave Botsford the news. I found a way to mix, but we couldn't put together the pieces in the same pass, a very dangerous proposition for such a long composition. But I also told him that my Monday night research revealed a possible solution: All we would need was another 1610, bringing the total to three units. A call to A.T./Scharff revealed that another rental unit was absolutely unavailable, but the firm's Josh Weinberg volunteered to call Sony for us to see if we could arrange a loaner—after all, no one had anticipated that we would need three digital audio processors.

While waiting for word from Sony, Botsford and I proceeded according to schedule. By 3 p.m. Wednesday afternoon, he had finished editing the voice segments and called me in to edit the music submaster. My plan on cutting the music was very simple: conform everything to the already mixed ¼-inch tape, duplicating as closely as possible all the original pauses. Why fiddle with what worked musically the first time around?

While the music was being digitally edited, I replayed the ¼-inch master via the console's built-in cue speaker, manually varispeeding the tape for sync with the music that was being digitally edited.

All the music was laid down at unity gain, allowing us to ride level in a subsequent mix. In certain places where the music was to be faded out, we gave the digital about 10 more seconds at unit gain and then faded.

Laying down the music was not a very complicated process, except at certain points in the work where the music ended or was faded and it was intended for the voice to be heard alone. For those occasions, I kept the ¼-inch reference tape rolling (in cue) as well as the 1100—in effect editing silence. As soon as I heard the music on the cue speaker, I punched a button on the 1100 that marked the time code location at which the next piece of music would begin, or fade up. In this way we constructed a music submaster complete with silences, whose spacing and timing would match the ultimate length of the two CDs (about 72 minutes each).

Because we did not want to produce our submasters on U-matic C-75 videocassettes (danger of dropouts), we employed three C-60s of about 50 minutes each, counting overlaps required to cut them together later. I knew that there could be some minor timing error in this conforming process, but made sure that the spaces would always be a little "long." That way I could correct any errors in the music track spacing by adding small pauses (or room tone if necessary) to the voice segments.

Later, during the mix, it turned out that the spacing was perfect 90% of the time. The other 10% was fixed by using the synchronizer's offset function to tighten the music closer to the voice.

By early Wednesday evening we had

edited the music and, amazingly, were back on schedule.

However, our loaner 1610 was not yet confirmed, which meant we might not be able to pickup-edit while mixing. Botsford and I agreed it would be too dangerous to stop the mix at each of the numerous voice entrances for, as mentioned above, you cannot manually cue a VCR-based digital tape the same as an analog ¼-inch. We decided it would be better to produce a spaced voice-only digital submaster, to be coordinated against the music submaster we had just edited.

Keeping the DAE-1100 patched for normal editing, I placed Botsford's pre-edited voice segments on the playback VCR. I then moved our music submaster to the VCR that was slaved to the record VCR and set up the console to monitor a mix of voice and music. I also played the reference ¼-inch via the cue speaker to determine what musical points Botsford had originally chosen to hit the voice. It took me the rest of Wednesday evening and Thursday day to digitally edit a properly-spaced voice tape (actually on three C-60 U-matics) and produce a time code-based cue sheet for all the voice entrances and exits.

The mix proceeded without a hitch; at one point we mixed for 55 minutes without a stop or an edit.

3-machine mixing

The good news came mid-Thursday: Sony would be able to loan us another 1610, scheduled to arrive Friday morning. The other bit of good news was that my spaced voice tape would almost let the piece mix itself. In theory, we would just have to start the machines and move faders for two hours. Friday morning we hooked up the third 1610 according to my latest block diagram and attempted an edit. To my relief, the edit worked the first time. We could mix two PCM-encoded tapes under programmable control, as well as pick up anywhere in our mix and digitally edit.

However, the processor playing back the music tape became intermittent, and there were occasional glitches at the edit points. By this time I had developed a nose for digital problems and my nose was telling me that the trouble was probably in timing. Botsford let me have Friday night to tidy up the situation, as long as I promised we would be mixing by Saturday morning, come what may.

That evening I found that the word sync (44.1kHz) patches I was using were probably causing the glitches. Changing over to composite sync and routing the sync (as shown in the accompanying block diagram) proved to be the perfect solution. Saturday and Sunday the equipment performed without failure.

The mix proceeded without a hitch; at one point we mixed for 55 minutes without a stop or an edit. In addition, the 3-machine mix approach was well worth the trouble; it allowed us to slide tracks whenever necessary, making the musical result even more satisfying.

The sonic result? Very pleasing. . . it was certainly the quietest voice-over mix I've ever done. If you ask me if I would do it again I'd say: "Yes, with certain changes. . . but that's another story."

R·E/P

Photos by Mary Kent.

Session Report: A Transcontinental Digital Overdub

Photo by David Rideau.

Nile Rodgers in Master Sound Studio, New York (left) and Stevie Wonder in Los Angeles, as they appeared on video monitors at Wonder's personal-use Wonderland Studio.

A recent digital overdub session involving Stevie Wonder, Nile Rodgers and Quincy Jones linked East and West Coast studios via a transcontinental satellite link. What technology was involved, and how did the participants handle the inevitable time delay inherent in digital audio connections spanning some 3,000 miles?

By Paul D. Lehrman

In early March, a highly unusual recording session took place simultaneously at Master Sound, the new recording facility at the Kaufman Astoria film-studio complex in Queens, New York, and Wonderland, Stevie Wonder's personal-use studio in Los Angeles. The clients were Wonder, Quincy Jones and Nile Rodgers.

Although, without doubt, the session became something of a media event, it also served a serious purpose: to demonstrate that a high-quality session could simultaneously involve two studios, located thousands of miles from each other, using digitally-processed audio and commercial communications satellites.

The session was actually two dates, with each studio in turn acting as the master. Rodgers, in New York, conducted a chorus of teenagers in an overdub of Wonder's anti-crack song, "Stop, Don't Pass Go", the multitrack master for which was laced up on a machine in Los Angeles. After that, Wonder played a harmonica solo on a single version of the theme to the TV show *Moonlighting*, being produced by Rodgers in New York.

In each case, a stereo monitor mix from the originating studio was digitally encoded via a PCM processor and set up as a video signal to a satellite, from

Paul D. Lehrman is a Boston-based free-lance writer, electronic musician, producer and regular RE/P contributor.

Photo by Media Photo Group.

Producer Nile Rodgers (left) and Master Sound co-owner and chief engineer Ben Rizzi at the Trident TSM console in New York setting up a stereo monitor mix of Rodgers' Moonlighting *theme for satellite transmission to Wonderland Studio, Los Angeles.*

Photo by Media Photo Group.

After the basics for "Stop, Don't Pass Go" had been laid down, a New York teenage choir led by producer Nile Rodgers gathered around a Calrec Soundfield microphone for overdubs onto the digital master tape in Los Angeles.

Photo by David Rideau.

Following the recording of overdubs to his anti-drug song, Stevie Wonder returned the favor by adding solo harmonic overdubs to Nile Rodgers' score for the Moonlighting *TV series, the basics for which were replayed from Master Sound, New York.*

which it was received and decoded by the overdubbing studio. The overdub tracks were then themselves PCM encoded and sent back over a different satellite transponder to the originating studio for the monitor mix.

In addition to the two digital audio paths, a pair of satellite channels were also used to provide audio-video talkback between the two locations. In this way, participants on both ends could see as well as hear each other, and the communications and musical pathways could be kept completely separate.

The music path was always on-line; whenever tape was rolling, a signal was going out over the satellite, with no special intervention from the engineers. On the other hand, the audio talkback system, as in a conventional session, was operated at each end by pushbuttons on each console, so that participants in one studio could confer with one another without being overheard by the team at the other end.

Cast of players

The participation of many companies and individuals, all of whom donated their services free of charge, was necessary to pull off the event. Although the complexity of the operation was constantly being pointed out, the message was also being conveyed that the session was *not* to be considered a 1-shot event. Instead, it represented the inauguration of a new commercial service that the principals hoped would be used often by the professional recording industry.

Satellite links on the East Coast were provided by Teleport Communications, a company that supports a 150-mile-long-fiber-optic communications network in the metropolitan New York area, with financial services, broadcasters and inter-city telecommunications carriers among its 40 or so customers. The network connects to the company's own satellite uplink and downlink facilities on Staten Island, the southernmost borough of New York. A permanent node of the network is installed at the Kaufman Astoria complex.

On the West Coast, the satellite links were handled by IDB Communications Group, a national supplier of satellite transmission and distribution systems, whose headquarters are located in Los Angeles. IDB parked one of its mobile trucks outside Wonderland, adjacent to Wonder's new remote-recording unit, dubbed "1-DER-1," which was seeing service for the first time.

The satellite transponders themselves were provided by GTE Spacenet, who donated time on both its GSTAR II and SPACENET I satellites.

Among the independent consultants involved with the project was Harry Mendell, a New York-based inventor and technical consultant, and a longtime collaborator of Wonder's. Mendell has become a vocal proponent of linking studios via satellites, and was a major motivating force behind the event. Also on hand was Mark Schubin, another New York-area consultant, and probably best know for his work as a system designer and broadcast supervisor at New York's Lincoln Center.

Satellite overdubs

During a pre-session reception at Master Sound, Mendell shared his vision of a worldwide network of studios, wherein "an artist in London could book a session in New York as easily as if he were booking one in his own town."

Master Sound co-owner and chief engineer, Ben Rizzi, explained that the major purpose behind the new service is to provide producers and artists faced with time-scheduling problems—which could make it impossible for them to be in the same place at the same time, for example—"an opportunity to work together." He also lauded the fact that the sound coming from the remote location is not a copy; "It's a clone of the original sound," he offered.

Stevie Wonder talked about the ability to send tracks back to his home studio when he's on the road.

"If I'm on tour and I'm writing a new song I want to record," he explained, "my voice is going to sound very different, depending on if I do it when I'm

Stevie Wonder (left) and Producer Quincy Jones at the console in Wonder's personal-use Wonderland Studio.

Photo by David Rideau.

Part of Stevie Wonder's keyboard setup at Wonderland.

performing every night, or if I wait until after I get back. It's important for an artist to preserve the feeling he gets when he's performing."

According to Quincy Jones, "This [digital satellite link] solves a problem we've been dealing with for years. When I was doing the soundtrack for *The Wiz*, we had a situation where I had to ask Michael Jackson and Diana Ross to redo a track. I was in New York and they were in Hollywood, and there was no way any of us could get away to meet the others.

"So, I made a copy of the master tape and gave it to an assistant, who flew out to the West Coast, got the tracks and then flew back. It took three days. If we had this system, we could have done it in a couple of hours."

In addition, because Jones wasn't able to be at that particular session, he had to leave it up to others to make sure the track was recorded correctly.

Jones also pointed out that the system's potential worldwide capabilities could open up some interesting cross-cultural opportunities.

"It's always difficult to get musicians from Africa or Brazil into studios in the United States," he said. "And it's just as hard for us to go there to work with them. Now we won't have to."

Although the Wonder/Rodgers session was concerned only with 2-track audio, Ben Rizzi was asked if there was a way to carry 24 digital tracks on a single satellite transponder channel. He replied that it could indeed be done, using an appropriate data-compression scheme.

"We just haven't done it yet," he confided. He also explained that the satellite service was mostly designed for overdubbing to previously recorded tracks—that there was no way, because of the unavoidable time delays in the system, to do a transcontinental "jam session" with tracks from different locations being recorded simultaneously.

"I suppose you could have two musicians doing overdubs at the same time to the same backing track," he said, "but they wouldn't be able to hear each other in sync."

Laying the tracks

Finally the session began. First, Wonder recorded a chorus of professional singers in his Los Angeles studio. Then Rodgers assembled his teenage choir around a Calrec Soundfield microphone in one of Master Sound's isolation booths. Except for the fact that the master tape and producer were 3,000 miles away in Los Angeles, it was a typical recording session.

"Shall we send the track over to you dry?" Rizzi asked Jones.

"Yeah, we'll figure out what to do with it when it gets here," came the reply.

Behind Every Synclavier® There's a Success Story

Profile: André Perry

C.E.O. and Chairman of the Board of **The André Perry Group** and **Le Studio**

Visionary producer André Perry heads one of the most sophisticated music and video production facilities in the world. Located in a beautiful and secluded Quebec setting, LE STUDIO and THE ANDRE PERRY GROUP have proven that, with the right personnel and equipment, a studio doesn't have to be in a major urban center to stay on top. With his facility constantly booked, it's clear that André's philosophy of total service has paid off. Recent projects range from network TV series to records and videos by such leading artists as Chicago, The Bee Gees, David Bowie, and The Police.

He comments on the success of his first Synclavier Digital Audio System and his future plans: "The Synclavier was so simple to learn and use that two weeks after installing the system we did the music and sound effects for a major network Movie of the Week. It's been so cost-effective that we're already ordering a second system for our new Washington, D.C. facility."

André Perry

New England Digital

Circle (24) on Rapid Facts Card

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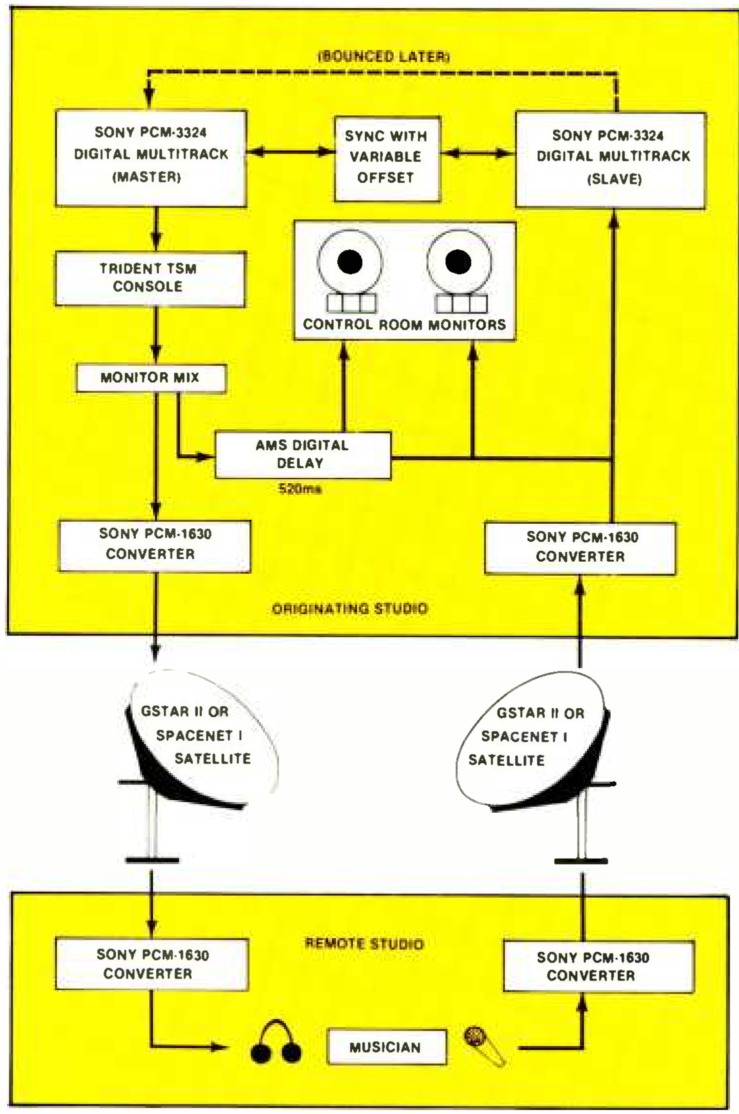

Graphic concept by Paul D. Lehrman.

Figure 1. *Schematic of a technique for linking two distant studios via digital audio signals carried in conventional satellite channels.*

"Can we get more of the lead vocal in the chorus' mix?" requested Rodgers.

"No," replied Wonder, "I don't really want them to hear it!"

After they found the right place to come in, Rodgers and his choir ran through several takes. Wonder spent a few minutes tightening up the group's rhythm, with Jones occassionally calling for them to "Hit it harder!"

Halfway through the session, something went wrong. The monitor mix being sent from Los Angeles began to distort in the Master Sound control room in New York, as if it were being scrambled through some misaligned delay line. Looks of concern passed among the engineers.

As various parts of the signal path were checked over, Rizzi asked his counterpart in California to turn off the master tape and send tone. At the same time, a couple of cables behind the PCM-1630 converters were replaced. After a pure, 1kHz tone emerged from the speakers, Los Angeles was asked to turn the master tape back on. When the audio came across clear as a bell, there was a collective sigh of relief.

Mark Schubin was asked if the problem was the cables.

"Well, no," he replied, and then lowered his voice. "It was the sampling rate," he whispered. "I think someone in Los Angeles must have inadvertently changed the sampling-rate switch on the 1630 prior to uplink. So while they were changing the cables, I flipped the switch on our downlink unit to the other rate."

The track done, Rodgers took the chorus outside for cold drinks, and Rizzi reconfigured the Trident TMS console to deal with a multitrack mix. The master tape for the *Moonlighting* theme was cued up on a Sony PCM-3324 digital 24-track, and a second PCM-3324 loaded with another tape, blank except for a SMPTE time code track.

Dealing with delay

The foremost question in just about everyone's mind was how to overcome the time delay inherent in any transcontinental satellite transmission. After all, if the signal from a tape went up to a satellite, then down to another studio and a track overdubbed to that received signal, which was itself sent up to a satellite and then back down, there would be a large difference between the time the original track was played and the time that the overdub made it back to the originating studio.

The answer was actually fairly simple. As can be seen from Figure 1, the original 24-track output was mixed to stereo in New York, encoded with a PCM-1630 and sent up to the satellite. The stereo mix, however, was not monitored directly at Master Sound; instead, it was first passed through a digital delay set to 520ms and only then routed to the control room monitors. Therefore, the

Photo by David Rideau.

To handle replay and recording duties at the Los Angeles end of the transcontinental overdub session, a pair of Sony PCM-3324 DASH-format digital 24-tracks were housed in Wonder's new mobile recording truck, named "1-DER-1."

Photo by David Rideau.

Also located aboard the 1-DER-1 mobile were several Sony PCM-1630 digital processors for converting stereo analog signals to video signals suitable for uplinking to New York via conventional satellite channels, and also reconverting incoming video feeds.

mix would appear in the monitors at precisely the *same* time that the sound of Wonder's overdub harmonica was arriving in New York.

The harmonica overdub was recorded on the second digital 24-track in New York. When the session was over, the two tape decks could be lined up—with the second one offset 520ms back from its original start time to compensate for the delay—and the harmonica track copied (in the digital domain) onto the original master.

The exact delay setting was worked out during tests the previous week, according to Harry Mendell, by sending out a signal from one track of the 24-track, recording it on another track as it returned from Los Angeles and then measuring the difference in time code between the two events. An AMS DMX-1580S DDL was called into service to provide the necessary delay.

Over the next hour, Wonder meticulously assembled his harmonica solo and then the work was done. Someone in Los Angeles put on another of Wonder's tapes, Rodgers pulled out an electric guitar and started jamming along with it.

"They're going to cut us off soon," came the word. "CBS needs the transponders!"

And just about 3½ hours after the session started, the video screens went blank and the music stopped.

A usable technology

Although the session was considered by many to be a technical success (and they hope a musical one as well—we'll know when the records come out), there remained the question of how useful the idea was. After all, there was no radical new technology at work; what *was* unique about the session was the number of different companies, each with its own crucial part to play, who cooperated to pull it off.

At the beginning of the event, one observer wanted to know what a session like this would cost if everyone involved were charging normal fees?

"About $12,000 for a 4-hour session," replied Harry Mendell. Representatives of the various companies then rushed up to the questioner to qualify Mendell's statement. Each satellite transponder costs $400-$500 per hour, they said; more if an overseas link is involved. The cost of getting the signal to the satellite using Teleport's fiber-optic network and dish is $170 per half-hour. Figures weren't immediately available for the IDB mobile satellite unit but, when one

The Los Angeles Connection
An on-the-spot report from engineer/producer David Rideau

From a West Coast perspective, the session began with Stevie Wonder leading an overdub in the traditional sense, with a young vocal choir in his studio performing backgrounds on Wonder's anti-crack anthem "Stop, Don't Pass Go." This was followed by a group of young people led by Nile Rodgers in New York "beaming in" additional BGs on the same number. Apart from a short break while the New York crew sorted out a slap effect in the headphone mix, the initial segment of the world's "first bi-coastal simultaneous recording" went smoothly.

After a short intermission, during which technicians repatched the studio for a Los Angeles overdub, Stevie Wonder showed his harmonica expertise as he "blew" over Nile Rodger's tracks of the new Moonlighting *theme.*

I was particularly impressed by how short a time it took to switch from doing overdubs in New York to doing them in Los Angeles. I was left with the impression that the session could just as well have involved two rooms in the same recording complex using standard tie lines. In fact, the session went so smoothly that there was extra time to reverse the overdub roles once more, so Rodgers could handle a guitar overdub to one of Wonder's songs.

Future Potential

As a free-lance engineer/producer, my imagination was running wild throughout the entire session. All that is now required for a remote, high-quality audio connection is an A-to-D converter, a small console and a satellite dish—a setup that could easily destroy many geographical and political barriers.

As with any technology, of course, such technology is going to be too costly for most of us for a while yet. It was estimated that the overdub session cost about $3,000 per hour. In the not so distant future, however, I can imagine a growing number of studios being prepared to handle satellite sessions. Such multi-city global jams would be possible, promoting what Quincy Jones referred to as a "fusion of musical cultures."

A last-minute mix for a video shoot could be transported coast-to-coast in the time it takes to play it back. The London Philharmonic via satellite for that afternoon string date? Or what about group members repairing vocals while on tour?

To further ease the initial setup costs, studios in major cities could be linked to local earth stations via fiber optics (as in New York for this session), or microwave. Not only would these links save the studio the cost of permanent satellite uplinks, but also free them from maintaining the licences needed to broadcast on satellite frequencies.

Overall, I'd say the Wonder/Jones/Rodgers session could be considered nothing but a success; it proved, at least, that a high-quality audio connection is available to almost anywhere in the world. It's also very encouraging to see so many people investing energy—practically everyone involved on the project donated their time and equipment—in what, even a short time ago, would be considered a wild concept.

also figures in the cost of booking two state-of-the-art recording studios, the price tag starts to add up.

In addition, it turns out that, in actuality, *six* satellite transponders were used for the transcontinental session.

"We're double-feeding to Los Angeles," said Paul Dujardin of Teleport Communications. "Our first plan was to go all Ku-band, and just use GSTAR II. However, the installation in Los Angeles was basically just an ENG truck with a small-aperture dish, which was fine for transmitting, but the receiver performance on the Ku-band was borderline. So we got a pair of C-band transponders on SPACENET I for security."

If enough people use the proposed service, it will inevitably become easier to set up and the price will come down. There's some question, however, as to whether that will happen. In an age of tightening record-production budgets, it seems that spending several thousand extra dollars so a superstar can do an overdub (which doesn't include what the superstar gets paid) may be viewed as a frill.

As one invited guest—a rather well-known free-lance engineer—said: "If they can bring the cost down it'll be great. But, right now, it certainly isn't for everybody." Time will tell. R·E/P

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Scheduling and Bookings Policies
for Recording and Production Studios

By Charles R. Mills

How best should a facility establish, and then effectively implement, a viable policy toward scheduling a session and ensuring that clients understand its booking procedures?

Recording and production studios are a part of that large animal that lately has become known as "The Service Industry," as such they must be able to respond to a great variety of situations, circumstances and types of people. Effective booking and scheduling policies and procedures are crucial to the smooth, efficient operation of the studio, and successful longtime client relationships.

There are three main areas that should be addressed when setting up a booking office: defining policies; establishing procedures; and addressing client needs.

Negotiating the Session

During negotiations with prospective session clients, consider the following:

- ☐ ***The type of project.***
- ☐ ***Necessary staffing levels.***
- ☐ ***Times of day selected for the session.***
- ☐ ***Who pays the bills.***
- ☐ ***Cash payment or credit terms.***
- ☐ ***How to bill the extras, such as rental items, recording tape and meals.***

Defining bookings policy

Successfully addressing the subject of scheduling and booking requires that, first of all, you have a clear understanding of the studio's goals. In many small businesses this goal is never stated, and exists solely as a body of past history and current attitudes. However, a relatively simple internal statement of purpose from the owner goes a long way in shaping and defining the way the studio does business. The policy statement should address business priorities (profitability, longevity and growth); the general nature or interest of your company (recording, post-production, advertising, films, and so on); and any major personal goals for the firm.

Next, define such basic areas as the

Charles R. Mills is studio manager of Clinton Recording Studios, New York.

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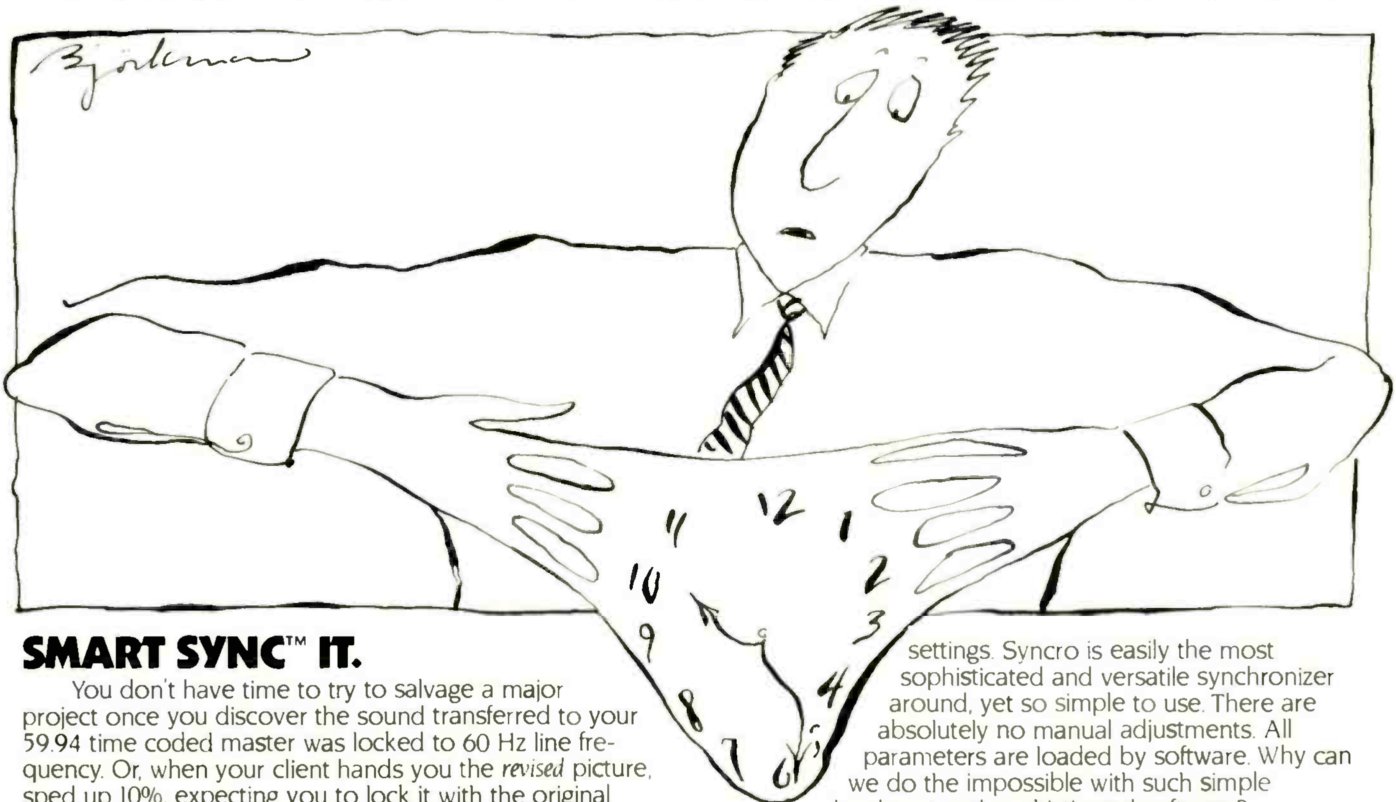

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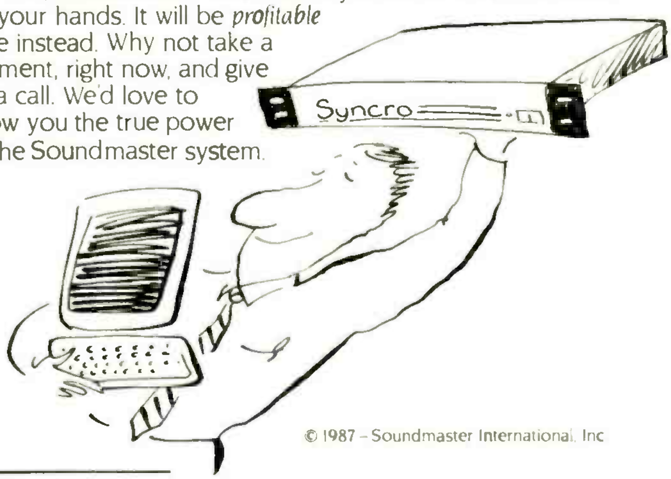

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kind of service you will and will not provide them; for how much; and in what manner. A concise statement addressing these areas will provide the framework for developing the rest of your booking and scheduling policies and procedures.

If you use a rate card, you should realize that it is your most obvious public statement of policy. Develop it with great care, *after* you have developed a general policy statement for yourself.

• *The Rate Card.* The rate card provides one of the earliest signals to prospective clients about your studio's ability to perform whatever tasks they might need, as well as what you expect of them. Generally speaking, prospective clients would not expect to walk into a $25 per hour studio in New York or Los Angeles, and be able to use two digital multitracks locked together with video playback or 35mm projection included in the rate. Inversely, they might expect all of this and more from a $325 per studio hour.

It is, therefore, important that the rate card clearly detail the items that are included in the basic rate, and what can be made available to them at additional charges. Items such as petty cash, food, phone calls, etc., may not be listed on the rate card, but a clear policy must exist.

Keep in mind that having clear-cut policies does not necessarily render you inflexible, but simply gives you and your client a starting point for negotiations.

During negotiations, some of the items that should be taken into consideration are as follows: the length of the session(s); the type of project and necessary staffing; times of day; who pays the bills; cash vs. credit; billing extras such as rentals, tape and food.

Demo rates are also to be considered under the topic of negotiations. Some studios will negotiate demo rates with jingle houses if they are promised that the finals will also be booked at that facility. Such deals require an act of faith in which you may or may not wish to become involved.

Another consideration when establishing policy is whether or not you wish to become involved in "spec deals." These involve the case where someone wants to produce an album at your studio for less than the normal rate, in exchange for considerations such as points, paybacks, future projects, etc. Remember that spec deals offer high risk for little cash up front, and require a great deal of faith on your part. They are also very difficult to enforce. [See Rosanne Soifer's article, "Independent Production Contracts," on page 82 of the April 1987 issue—*Editor.*]

• *Hold Firm and Cancel.* The areas in which your policies should be fairly explicit are matters related to the session status. Is a first-hold a request or a commitment? When should it be either removed or firmed? What about second (third, fourth, etc.) holds? Once a date is "firmed" can it be cancelled without penalty? Under what circumstances? What should you do with a client who won't firm or release a date until the last minute?

Although some of these items will be discussed in a later section on the care and feeding of your clients, keep in mind that clear policies make it easier for you and your client to know what the deal is. Other areas for determination are: length of bumpers, (how closely is the next session booked); time not used (do they pay all or part); what constitutes overtime and what do you charge?

• *Cash and Credit.* If your studio is in the tax write-off business, you can skip over this section. If you're in business to make money, however, that goal must be reflected in your policies and practices. Credit and collection processes are difficult for some people to deal with. Policies that state you will run credit checks on *anyone* who is unknown to you makes this procedure less of a personal issue.

Some things that you might request (or insist upon) of your unknown client include provide a deposit before the date is considered firm; pay in cash (the green stuff) at the end of the date, or use a major credit card (if you are set up for that). No third party post-dated out-of-state checks, please.

Purchase orders can be lovely, but remember that they too can become as worthless as the aforementioned check when you're dealing with an unknown client. POs can also work against you in a dispute, for example, where a producer has you perform more work than is specified on the PO. At certain times, Cash = Credit.

• *Rate Structure.* In determining your rate structure, consider what you will charge per hour using staff engineers and what you will charge using independent engineers. Also, consider daytime vs. nightime rates vs. costs. What will you gain or lose by allowing for day rates or lockouts?

Assuming that your goal is to maximize your income, how does the ideal of fairness affect your decisions when considering the following scenario? Two different clients are trying to book the same time—one long and the other short sessions. Who do you choose? And what about cash vs. credit; regular or new client; and payment record, etc.

Last but not least, you should have a policy related to your dealings with other studios. How much information will you share? Will you grant them credit? Some studios have strong business relationships with their colleagues; others are notorious credit risks.

The Policy Statement

A production facility's policy statement, which would constitute a clear understanding of its business goals, should address the following areas:

- ☐ ***Business priorities—profitability, longevity and growth.***
- ☐ ***General nature or interest of the studio—recording, post-production, advertising, film sound, etc.***
- ☐ ***Major personal goals for the business.***
- ☐ ***Kinds of service that you plan to offer prospective clients.***

Booking procedures

The book should be set up so as to present a clear, concise, complete picture of what is going on at your studio. It should be large enough for all information to be clearly visible, and its placement should make it accessible to all appropriate viewers.

It should convey as much basic information about the sessions as possible including: the client, the hours booked, status (firm or hold), project title, contract(s), engineer, assistant engineer(s) set-up, special requests, producer, equipment alignment, rentals, and other information that various members of your

staff need to know to get the job done.

The book should be visually organized in such a way that you can maximize the potential of your rooms and your staff by knowing who's where, and when. Its layout should give a good visual impression of all activity and interaction of clients, staff, sessions and facilities for a given period of time, whether it be a day, a week or a month.

Besides maximizing income potential, a good book graphically helps you control costs such as overtime, multiple equipment rentals of the same item, set-up/breakdown time among rooms, etc. It should *not* contain extraneous information that will only serve to confuse or detract from its purpose. It also should not become a message center for non-business matters. You should have strict rules that clearly identify who sees and who writes in the book.

If equipment and/or instruments are to be rented, it is useful to note that in the book, together with information regarding who is acquiring the rentals and which vendor is supplying them.

The person at the book must have good listening and information-gathering skills, in order to obtain as much useful information as possible about the upcoming session. This information not only includes basics, such as accurate set-ups, who obtains what rentals, etc., but, by observing the client's pauses, constant reiteration of a question, use of words like demo, limited budget, and other subtleties (adding time to the bumper or changing the time booked).

Accurately hearing what is being said and learning to interpret the meaning can often forewarn you of potential problems. It can also provide excellent opportunities for selling and marketing at the book. By using problem solving techniques, you may be presented with opportunies to inform prospective clients of tasks your studio can perform that they had never thought of.

• *Bumpers and Overruns.* Once your policies are cleary expressed and the session times clearly recorded in the book, the issues of bumpers and time booked but not used become fairly easy to deal with. When clients run past their bumper, when necessary let them know that another session is booked into the room after them, and politely but firmly let them know when you need the control room cleared.

Remember: Your goal is *not* to get them out, but to get the next client started on time. It may be necessary to come into the room at the end of the session in order to clear it. (As the police have discovered, sometimes your mere presence can accomplish this task with a minimum of angst.)

In order to accomplish all of this, it's essential that you establish and maintain a good flow of information with your staff. By keeping accurate records, you will be better able to resolve disputes, track past events, plan for future events and generally serve your clients better. If you have good manual systems, fine. If they need some improvements, it might be a good time to investigate the use of a computer to help organize the mounds of paperwork generated by your studio.

• *Automation.* A number of companies now offer office automation systems for recording-studio operations. Such systems vary greatly in scope, modual integration, power, flexibility and cost. If you have a multiple-room operation with a sizeable paperwork load, or if you want to be able to format and analyze large amounts of administrative data, or if you are single-handedly trying to manage bookings, accounting and management of your studio, you should investigate the use of computers in your front office.

Some basic aspects that seem common to the existing, prepackaged studio systems are that the more integration you seek, the less flexible the system is likely to be. In addition, although many operations are common to all studios, few prepackaged systems will adapt fully to your way of doing business. Also, the old adage of "garbage in = garbage out" still holds true.

Consider multi-user systems from the outset, as well as the future operations you might want to add to the system. One powerful advantage a manual calendar or book offers over a computer screen is the ability to present an enormous amount of data in a powerfully meaningful visual format; as a result, many studios have automated office systems while retaining manual booking calendars. Longer data terminal experience and more powerful software developments will likely change that situation in the future.

[For further information, see Robert Carr's article, "Automating Recording Studio Operations," on page 38 of the December 1986 issue—*Editor.*]

Care and feeding of clients

The first task to perform in this area is to define the types of clients and then to determine their needs. Advertising clients often operate under a lot of pressure; their main concern is to get the job done and move on to the next project. Record producers often "live" with a project for a period of time and may not be as concerned with the moment but more with the overall project. In other words, producers may be less frantic on a day-to-day basis, but more attentive to details of technical quality and budget.

Film scoring clients are in some ways a combination of the above two types. There are often several people in charge; they are under time pressures; the sessions are often spread over several days or weeks; and thus they become a part of

Keeping "The Book"

The day calendar should convey as much basic information about the recording session as possible, including:

- ☐ ***The client's name.***
- ☐ ***The number of hours that have been booked.***
- ☐ ***The status (firm or hold) of the booking.***
- ☐ ***The project title.***
- ☐ ***Appropriate contacts for the session, plus telephone numbers.***
- ☐ ***Session engineer to be used (staff or freelance).***
- ☐ ***Assistant engineer(s) for the project (staff or***
- ☐ ***freelance).***
- ☐ ***Special requests.***
- ☐ ***Session producer.***
- ☐ ***Equipment to be rented for the session.***
- ☐ ***Musical instruments to be rented for the session.***

...Plus any other information that various staff members might need to know to get the job done.

your place during their stay.

Learning the rhythms of your clients can save untold grief and help reduce the number of surprises. Each of them will have their own unique technical, administrative and social requirements. The impression you should give to every one of your clients is that you are there for them; that you will do whatever you can to help solve a problem. Make time available for them to ensure that sessions run smoothly.

New clients often bring with them attitudes that have been cultivated through their dealings with other studios they either love or hate.

• *Required Skills.* In order to do a good job at the book, you should have excellent organizational and communication skills. You must be able to effectively communicate verbally with your clients and staff, and you must develop consistent writing patterns in the book so that the rest of the staff can glean the information they need in order to best serve the client.

At least some technical background is essential so that you can answer questions about the studio's abilities to perform (or not to perform) various tasks. A basic knowledge of music and its terminology is also very important.

"Networking" establishes you as a source of information, as well as the booking person; this in turn enhances the studio's value to the clients. Schmoozing can become part of a personal style that can help smooth over difficult situations; it is imperative, however, that it be done at times appropriate to the circumstances, lest it be perceived as wasting someone's time.

It is also important to understand and appreciate the telephone in all of its guises and uses: as a marketing tool, for exchanging information and, sometimes, as a weapon. Some people who don't engage in face-to-face confrontation will return to their office, pick up the phone and let you have it. Understand that this ieir escape valve and may have noth- to do with anything over which you e control. (When they get you rattled may want to use the hold button—af- olitely asking them to hold—scream, ack on line apologizing for the delay hen allow them to continue their gue.)

Which brings us to: "The customer is always right!" Right? Well, the answer is pretty obvious. What to do when the customer is *not* so right? Tact is always in good order. Remember: If you are in a dispute with a client and you have good backup that makes it clear that the studio is not at fault, don't sell the facility and the staff short. Such an attitude will come back to haunt you later.

• *New Clients.* Clients who are new to you may require some additional care and handling. No matter how good your reputation, they may be somewhat nervous about dealing with a new situation. A little more hand-holding, some reassurances about your studio's ability to perform complex tasks, coupled with your "We've been hoping for your call" attitude makes for a nice beginning.

Most often new clients bring with them attitudes that have been cultivated through their dealings with other studios they either love or hate. Either way, part of your job is to overcome such attitudes.

When you are providing new clients with quotes or proposals, it is important to be as thorough as possible in describing what you offer and at what costs.

• *Studio Personality.* Does your studio have a personality? Yes! The people on your staff comprise part of its personality, but the physical space also contributes to it. By being aware of your studio's personality, you can answer questions to yourself, such as: Which engineer and assistant would be best for this date? Which room would the client probably be the most comfortable in?

By understanding *your* studio's personality placed next to the *client's* personality, you can sometimes answer unasked questions and solve seemingly inexplicable, non-existent problems.

• *Dangerous Trends.* Trends often turn into habits; today's favor can become tomorrow's habit, which may then be presumed to be policy. Part of your job, when clients lose sight or "forget" how you operate, is to bring them back to reality concerning *your* policies. This situation can become still more complex when dealing with clients that are also competitors.

Today's recording industry has created a situation where many music houses and producers have established their own in-house studios, even though they still book time at commercial facilities. Their policies will undoubtedly be different from yours, and their perceptions about you will be different from other clients. Consequently, you may have to determine how and when to apply or modify some of your specific policies and procedures.

• *Care in Feeding.* When it comes to the care and feeding of your clients, there are times when they seem more concerned with the quality of the corned beef than with the quality of the sound on their masters. You should pay close attention to their eating and drinking needs; it can often make the difference (in their minds at least) between a good and a bad studio.

There are times when clients seem more concerned with the quality of the corned beef than with the quality of the sound on their masters.

• *Manners.* There have been several magazine and newspaper articles written recently about America's lack of unqualified success in the burgeoning service sector of our economy. The articles cite many reasons for this poor showing, including too much television, the "me" generation, bad breeding at home, the breakup of the nuclear family, working mothers, uncaring fathers, etc., etc. Finally, you inherit people who have a wide range of ideas concerning how to act. Demand and, if necessary, teach them good manners.

When in doubt, common sense should prevail. Think about how you feel when someone picks up the phone at the other end of the line while finishing their conversation with someone in their office. Or you are speaking with a client and a staff member stands between you. Or an employee who insists on airing dirty laundry in front of a client, making everyone uncomfortable.

Even from the point of view of booking and scheduling, employee training and standards of behavior are of paramount importance. Many corporations consider telephone skills and manners a primary ingredient of new employee training and orientation.

Booking and scheduling can be as complex or as simple as your business needs require, and your resources allow. But, no matter how you approach it, for recording studios such considerations are the nerve center of the business—the focal point through which all information passes and from which all activity begins.

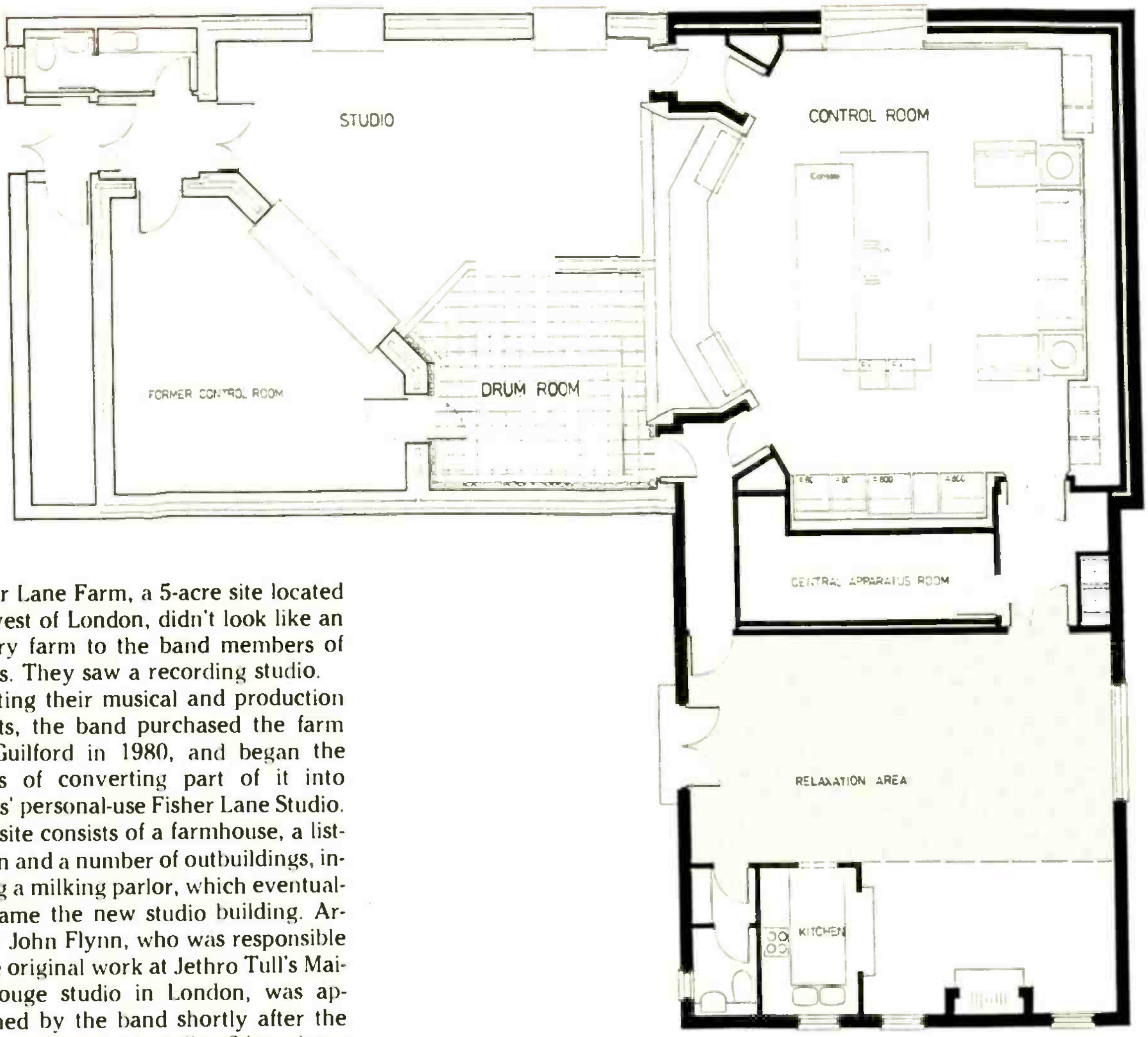
The care used in developing your booking and scheduling policies, and the skill with which you implement your procedures, will be major contributors to your business success. R-E/P

Facility Spotlight:

Genesis' Personal-Use Fisher Lane Studio

By Richard Elen

The conversion and upgrading of a farm milking parlor into a versatile recording facility presented some unique challenges to architect John Flynn and acoustic designer Sam Toyoshima.



Floor plan of the new Fisher Lane Studio, showing the location of the existing and added control room areas.

Fisher Lane Farm, a 5-acre site located southwest of London, didn't look like an ordinary farm to the band members of Genesis. They saw a recording studio.

Trusting their musical and production instincts, the band purchased the farm near Guilford in 1980, and began the process of converting part of it into Genesis' personal-use Fisher Lane Studio.

The site consists of a farmhouse, a listed barn and a number of outbuildings, including a milking parlor, which eventually became the new studio building. Architect John Flynn, who was responsible for the original work at Jethro Tull's Maison Rouge studio in London, was approached by the band shortly after the farm's purchase to install a 24-track recording facility on the site.

The construction was carried out in two phases. The first phase—to install the studio in the existing building—was completed a few months after the purchase.

Thanks to the rural nature of the property, earth-born vibration was minimal.

The second stage, which included doubling the size of the building and constructing a control room designed by acoustic designer, Sam Toyoshima, was completed last year.

Despite the fact that it would have been more economical in the long run to construct an entirely new studio building on the site, the vagaries of British building license requirements meant they they had to work within the existing building envelope—at least to begin with. By the time *phase two* was about to begin, the building permits had all been sorted out.

Original studio

In the early days at Fisher Lane Farm, the band wanted to be up and running as soon as possible. As a result, a contractor was on site, ripping out the old interior of the milking parlor—it had been used as a garage in recent years—while the initial drawings were still being complete. The building didn't have sufficient height for the intended purpose, however. So the first task after clearing the interior was to dig out the floor and relay a new one, virtually at foundation level, two feet below the previous floor surface. As a result, it became necessary to dig a trench all the way around the building to ensure good drainage. Then remained the matter of placing an entire acoustic shell inside the building.

The acoustic consultant for the first phase decided that there was no need for a floating construction. Thanks to the rural nature of the property, earth-born vibration was minimal; consequently, it was just a matter of laying a concrete slab for the floor of the facility.

According to architect John Flynn, "The acoustic treatment in the studio area was virtually nil. There was a separate drum room for Phil [Collins], which we lined with stone. He's been very pleased with it."

As can be seen from the accompanying floor plan, the studio area was divided diagonally during the original phase of construction, with the drum room off the end of the partition between studio and control room—a division that still exists.

"The acoustic treatment in the control room was based on Audio-Kinetics modular boxes," Flynn says. "It was pretty small, and there were a few limitations, largely stemming from the size and shape. But, overall, it worked well."

Richard Elen is a U.K.-based free-lance writer, producer and session engineer.

After the completion of the original phase, the studio was used by the band for four years: it was the location for several solo and band albums during the period. Then, in late 1983, Flynn was asked to carry out a feasibility study on a new control room, simply by placing a new room on the end of the studio.

Toyoshima can test his designs at JVC's acoustic facility with up to full-scale models.

However, plans began to take on a rather dramatic shape as time went by. This was due partly to suggestions from Genesis producer Hugh Padgham, who was well-acquainted with developments at Town House, Virgin Record's west-London studio. Eventually, they called in studio designer Sam Toyoshima, who was responsible for Town House Four. The result is the present design, which includes a new control room, equipment room, relaxation area, kitchen and other facilities.

Phase two construction

The 6-month time span for construction may not be so surprising, since major additions had to be made to the building.

"Sam was primarily responsible for the design of the interior acoustics of the control room," says Flynn. "We were already fairly committed on the actual construction of the soundproof 'envelope'—but he also made some suggestions about the treatment of the existing studio area.

"We had the physical size of the rooms already set up, and the interior shapes laid out. Sam came up with conceptual drawings containing the essential geometry and acoustics. He then came over to look at the site."

Some of Toyoshima's design required modifications to the new building works that had been constructed before he had become involved with the project.

"Then Sam went back to Japan," Flynn recalls, "and we got on with the shell. We had completed the shell just in time for his next visit—and it was then that we spent three days, often until two or three in the morning, with him feeding me details on the traps, the fabric lining details, and so on.

"We sketched them out, on the backs of drawings, usually, and from there I drew them up into something we could build from," Flynn says.

"Sam sketched acoustic details for every nook and cranny of the new room, and I had to take it from there into drawings and get the studio built. And, in view of the building contractor's lack of experience in this type of work, I was on site for a great deal of the time.

"There was some overlap between the design stage and the construction."

The bottom line is that Flynn was largely responsible for the practical elements of the building work, including locations of the lighting circuits and their resistance dimmers.

The completed control room can be considered typical of Toyoshima's designs to date, with the notable exception that the original partition between the drum area and the studio butts up to the new, triple-glazed control room window.

"It's a quirk of the design," says Flynn, "but it works very well. The partition can be removed later if necessary, without affecting the viewing window, but the amount of separation is excellent."

The two SSL display monitors for the Primary Computer and Total Recall automation systems are located in the area above the central window.

Acoustic treatment

Drapes cover part of the stone walls in the drum room, which can be added or removed as required to alter the acoustics. A similar technique is used in the rest of the studio area, where the only additional treatment is a series of angled plywood ceiling panels, backed with absorbent material, that reflect sound back to the walls. (Originally, these panels were to have been faced with modular absorbers, as in the first control room. These were omitted, however.) The windows are triple-glazed throughout, the outside consisting of a sealed, double-glazed unit to minimize potential condensation problems.

There is the feeling that his designs are "known to work" before they are ever realized on the client's premises.

The doors feature an internal sandwich construction of lead sheet and Rockwool, and use magnetic seals. They provide a quoted separation of at least 40dB. In the lock between studio and control room, both doors open inward (toward each other) making the seal foolproof.

Rear view of the new control room, showing tape machine soffits and client monitoring areas.

Traps are constructed of plywood with Rockwool on both sides, and run down from the structural ceiling. The ceiling itself features a stepped construction, with additional treatment beginning above the slope behind the console. About a foot behind that is located a precast concrete slab forming the structural roof. The outside roof of the building is comprised of corrugated sheeting—an extension of the original milking parlor.

Air conditioning is based on three heat-pumps; two supply the studio plus the old control room, and one the control room. Separate air conditioning systems cool the equipment room (located behind the tape machine alcove) and the lounge area. Because of the confined space in the ceiling, the routes taken by the air conditioning ducts, with their baffles, are complex.

"Although the room is large," Flynn says, "the acoustic characteristics are

Without the ability to test a design in advance, it is often the client who provides the laboratory.

Control Room Equipment

Console: *56-input Solid State Logic SL-4056 E-Series with Primary Computer and Total Recall automation systems.*

Multitracks: *Two Studer A800s with Dolby SP24 noise-reduction racks.*

Effects processors: *Audio and Design Scamp rack; Lexicon PCM-41 and two AMS DMX-1580S delays and pitch shifters; Valley People Dyna-Mite, Kepex II and Gain Brain compressor-limiters; UREI LN1178 stereo limiters; Neve model 1078 equalizers; Friend Chip SRC time code and MIDI synchronization system; Drawmer noise gates.*

Monitoring System: *Westlake HR-1 cabinets driven by six Crown PSA-2 power amps; Acoustic Research AR-18LS, Yamaha NS10M and Auratone 4C close-field monitors powered by Quad 520 amps.*

Reverb systems: *Quantec ARS Room Simulator, Yamaha REV-7 and AMS RMX-16 digital reverbs; EMT tube plate.*

Mastering transports: *Two Studer A80RC 2-tracks; Sony PCM-1630, digital processor, DAE-1100 editing system and companion DMR-2000 U-matic VCRs; various cassette and Beta hi-fi videocassette decks.*

similar wherever you are in the room."

The control room is large, with a seating area at the back. Cassette machines and tape storage is located at either side, set into the rear wall. The carpeted space between the seats and the wood-paneled floor area is occupied by the 56-input, customized Solid State Logic SL-4000 console. Space is provided on all sides of the console, with a window to the right and the tape machines in an alcove on the left. Mobile, wood-faced rack units house outboard gear next to the patchbay on the left-hand end of the console. There's sufficient space for synthesizers and other equipment.

Sam Toyoshima, unlike most other studio acousticians, is in the unique position of being able to test his designs at JVC's acoustic research facility in Japan with up to full-scale models. Toyoshima, in fact, spent 10 years of research before he tackled his first studio: There is the distinct feeling that his designs are "known to work" before they are ever realized on the client's premises. Without the ability to test a design in advance, it is often the client who effectively provides the laboratory. As a result in this case, though, very few modifications were needed once the construction had been completed.

Responding to the continuing trend toward recording systhesizers and direct-inject instrumentation in the control room, additional space has been provided in back of the console for musicians to be able to "play into the mix" during overdub and tracking sessions.

Very few modifications were needed once the construction had been completed.

Control room equipment

The original equipment installation includes a pair of Studer A800 multitracks and a pair of A80 stereo machines, plus a Sony PCM-1610 digital stereo system and DAE-1100 editor, complete with three U-matic machines. In addition, there's a full complement of cassette and Beta hi-fi video machines.

The main monitors are Westlake TM-4s, with two bass drivers, driven by Crown PSA-2 amps. Close field monitors are Acoustic Research AR-18LS units.

The outboard complement features a Quantec QRS, Yamaha REV-7 and AMS RMX-16 reverbs (the latter including keyboard interface); an original tube EMT plate in a back room; Audio and Design Scamp rack; Lexicon PCM-41 and two AMS DMX-1580S units; Friend Chip SRC time code interface and synchronization unit; four Valley People Dyna-Mite signal processors; a pair of Drawmer noise gates and assorted Valley People Kepex II and Gain Brain units; a UREI 1178 stereo complimiter; dbx models 160 and 163; and a pair of Neve 1078 equalizers.

Sam Toyoshima and John Flynn are now regularly collaborating, offering a complete architectural and acoustic design service for the studio industry, via their jointly formed UK-based company, Acoustics Design Group. The group has also been joined by Hugh Padgham, who acts as consultant.

R·E/P

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SPARS
Business Conference Replay

By Lee Murphy

A report of a 2-day conference that provided details of various schemes for developing and implementing a successful business plan for recording and production studios.

Conference coordinator and keynote speaker Guy Costa, Motown Hitsville Studios, Los Angeles.

The Man Who Mistook His Wife for a Hat, Olives Sacks' national best-seller, describes a person who could not identify even the most common objects (like hats), but who instead approached people and things as abstract puzzles or tests which, without some striking feature to key him, the man invariably failed.

Don't laugh. The disorder afflicts thousands of otherwise sound individuals who call themselves executives, but who don't truly understand a basic foundation of good business: *Planning*.

Some 60 of these lost souls traveled to Los Angeles in late April, to attend the 2-day SPARS 1987 West Coast Business Conference at UCLA, where seven born-again industry leaders shared their knowledge and experiences. Some, like Chris Stone, president of the Los Angeles Record Plant, even opened their financial records to illustrate the points they were making.

It was a pilgrammage well worth the time and money.

Day one: morning session

Guy Costa's keynote presentation, entitled, "A Business Planning Guide," is summarized in Table 1.

Some of these elements are considered by studio owners planning for a new room or new business. However, Costa, who is vice president of operations at Motown, contends that the "realities" of being in business are too often *not* perceived. Planning, helps put the emotional aspects of a business into proper perspective Costa says.

"This is especially important in our 'creative' environments, where the desire/-drive is typically greater than the availability of resources required for success."

The "definition of business," in historical terms, may be easy. But, when it comes to a proposal, you need to provide an answer to consider the question: Why do you want to get into this venture? Securing an answer begins to call for a level of analysis few of us are prepared to undergo, though some do it intuitively. Then look at hard-edged questions such as: Is the planned objective needed and how will you accomplish it? And, why do you think it will be successful?

You now need to evaluate the marketplace. Costa advised that studios look closely at an *overview*; Is the market growing or shrinking? You should have few problems listing your major clients, but it becomes more difficult to define the size of your potential market. How many customers? How many dollars?

Then consider the *potential*: What breadth and depth of services will you provide? What are your strengths and weaknesses? And, what is your market integration? Are you at the beginning of the production chain, at the end or preferably somewhere in the middle, going full circle to control your own supply channels at the start, and distribution channels at the finish. Will your facility be a turnkey operation, producing all elements of a client's project?

Next, Costa suggested, your integrity will be severly tested, as you evaluate the competition. For instance, how do your operation costs compare with the competition?

"If you're receiving lots of resumes," he quipped, "you're probably paying too much in salaries."

Service and support are essential, so you must determine whether you can match or better the quality offered by your competition on these critical points. How does your facility compare in your marketplace for location, capacity and efficiency?

Finally, Costa calls for a critical analysis of the competition. Who is the competition? How do they function? Who are your customers? What does your competition do to win them over? Costa says that owners often get choked for price with a single client. Which is why Motown, its own best client, is opening its doors for outside audio-post for video business.

Consider management analysis: Who are they? How will they be compensated? Are individual responsibilities clearly defined and are their functions adequately supported? Sales, service, marketing and technology also figure in this analysis. Who said it would be easy?

Analyzing the finances

The buck starts here. Where's the

Lee Murphy is owner/mixer of Brigg's Bakery, a single-room audio-post for video facility in New York, which caters primarily to broadcast and corporate clients. He is also a co-founder with his technical director, Jim Stephen, of a product line known as Stable Cables.

money to come from, and how is it to be used? Costa suggests that professional accounting services be sought for the preparation of monthly income statements and balance sheets.

Next follows the definition of objectives stage of planning: start-up, short-term and long-term goals. This is followed by definition of strategy, where the studio owner details the conditions of a specific plan and how it's to be executed.

Costa suggests, this is a good time to create a matrix of potential risks and their dollar values. Also, he advises to prepare a survival strategy to deal with potential downside realities—for example, what happens if your building burns down?

The final phase of creating a business plan is the summary where all information is written into a concise statement of purpose. Costa provided the following example of a business plan summary:

The Whodunnit Studio, a closely held New York corporation, is seeking $15,000 in additional financing to purchase electronic recording equipment to duplicate the new R-DAT tapes that are expected to hit the market within the next two months. This sum, together with an additional $25,000 invested by the principals, Joe and Sally Cheap, will enable the business to go after a niche that is not expected to be filled at least for the next 12 months. Whodunnit believes that they have the distribution and manufacturing channels and that they can lock up the Spring Valley City marketplace for an extended period of time and become an overnight success.

The final stage—completing the plan—entails packaging the business plan. Design a cover and binding, and also write a title page, introduction and table of contents.

Table 1. Costa's Catechism

1. Define the business.
2. Evaluate the marketplace.
3. Evaluate the competition.
4. Analyze the management.
5. Analyze the finances.
6. Define the objectives.
7. Define the strategies.
8. Prepare the summary.
9. Complete the plan.

Opening a new studio

Now, let's shape Costa's tutorial into reality—the process whereby Bruce Merley put together Clinton Recording Studios in New York. Merley's presentation entitled "Opening a New Studio" focused on his own experiences.

There are a number of obstacles to be overcome by anyone opening a new studio in the bureaucratic swamp of a large city. (It may not be much easier to do it in a smaller community either.)

Like other speakers at the conference, Merley was able to look back on his experience and use that hindsight to benefit other studio owners. Merley brought exceptional administrative talents to his task from working at the Yale School of Music, and later beginning his own, independent record label.

In the end, Merley slogged his way through a 30-month minefield of municipal and institutional barricades. The creation of Clinton Recording depended upon state financing, federal guarantees and personal commitment.

"I hired a Harvard student," Merley said, "to create a report that recording-studio business would increase 25% over the next two years. She did, and our loan package was approved.

"We then went through six months of management relations training, because I wasn't sure I wanted to work with my partner-to-be."

"We incorporated clients' needs into our facility; showed them the plans; took them on-site to confirm that Clinton was addressing their specific desires. We made sure our credit relationships were *strong* even before opening."

Day one: afternoon session

If there was anything daunting about Merley's presentation, Wilber "Pete" Caldwell, of Doppler Studios, Atlanta, made conferees feel right down-home again. Beneath Caldwell's good ol' boy exterior, however, lay the sensibilites of yet another keen businessman, and one that could relate the trials and tribulations during his presentation, entitled "The Evolution of the Multi-Studio Operation."

First, Caldwell weighed the virtues and vices of various sized facilities. Three or four rooms provide flexibility, but a larger operation becomes less personal. The bigger the facility, the more services you can provide to clients (and the longer the lines of communications to accomplish this). Diversification permits rock music, advertising, audio post and so on (while it precludes the charm of a so-called "boutique" business). Economies of administration and labor accrue in a larger studio (and so does traffic flow).

As for Doppler's recent expansion into a multi-studio operation, Caldwell says he developed cost estimates with a fairly high degree of accuracy. His thoughts, in Table 2, offer some sage advice.

An interesting plus that Caldwell has found in the wake of Doppler's expansion is what he calls "Coat-Tail Technology:" an older installation benefits from its proximity to a new, state-of-the-art room. At the same time, he warns, a new control room can spoil clients and staff to work in older rooms.

Among much of what seems obvious (after you've been told it) are some things which are not. There are two cost areas that Caldwell thinks deserve special consideration in the years ahead: Utilities and insurance, both of which are rising rapidly.

Other wisdoms: Schedule seminars to educate and sell clients on new technology; start each day with one open studio, not prebooked—you'll fill it; and, finally, occassionally create a business plan, even if you don't plan to expand.

If Merley and Caldwell did everything, or most things, right, Dave Porter, president of Music Annex, didn't do everything, or most things, wrong. He owned a going business right? But during his presentation, entitled, "Adding a New Location," came the admission that at the very least, his primary location in Menlo Park, CA, left something to be desired.

"It is," as Porter pointed out, "20 minutes from everywhere."

After eight years in Silicon Valley, Porter found that his computer-making client base was no longer in the chips; business was flat if not descending, and the strongest opportunity for growth and diversification lay "20 minutes away" in San Francisco.

He conducted the obligatory market research, to determine the capabilities of competitive facilities, as well as the availability of clients. He comparison-shopped other San Francisco studios and located himself in a hub of networks and

Table 2. The Doppler Effect

1. Estimate overall costs, then double them.
2. Estimate installation cost, then multiply by 2.5.
3. Expand studios and maximize compatibility.
4. Construction cash requirements mandate super attention to recelvables.
5. Own your own building, or the landlord will own you.
6. Although it's a corporate (business) loan, bankers always lend to people.

Table 3. Porter's Law

1. A studio's gross billings should rise by the same amount as is invested in hardware during any given year.
2. Labor costs should not exceed 25% of gross.
3. Ancillary services, such as tape duplication, can help buffer a studio from volatile swings in music recording.
4. Banks want to know how big a risk-taker you are.

ad agencies. The space itself was then designed to be almost entirely billable (providing direct client services), which was essential, given the city's relatively high rent levels. The San Francisco site houses two studies: Studio I is a 24-track audio-for-video room; while Studio II functions as an 8-track media and jingle room.

Interestingly, the new location does not compete with the original Menlo Park site, where billings, in fact, have since increased by 15%.

Porter's business plan, summarized in Table 3, is self-disciplinary insurance against what he calls "LED Fever," the sometimes chronic compulsion to buy new equipment. Planning can also help an owner compete effectively against what he dubs "Hot Tub Studios"—"set up by Daddy Warbucks who give their kids \$300k to shelter money, " he says.

In all, Porter filled a 12-page presentation with savvy and inventiveness, concluding that as long as financing is available for the acquisition of equipment, and expansion into new areas of audio and audio-for-video, his facility will continue to thrive.

Day two: morning session

In his presentation entitled, "Entry into Video," Fred Jones, president of Fred Jones Recording Services, Hollywood, discussed methodologies that closely tracked those of others who had spoken before him: objectives, proposals, strategies, marketing, facility, utilization, financials and so on. Then he turned the session into a call-in radio show.

Jones claimed a 1-inch VTR is essential in the Los Angeles market to satisfy clients that audio laybacks are accurate. Other conference participants disagreed, however, offering that an audio layback machine created a better quality sound on videotape. And Pete Caldwell responded that access to a 1-inch video machine also opens the door for dub business at the end of a sweetening project.

Console flexibility for video sweetening was discussed, centering on a board's capabilities and not its brand name, at least for audio post-production.

On the subject of client relations, Jones said that he's increased bookings 80% with one agency that traditionally booked "on-hold," but then often cancelled a short time later, presumably after confirming space and time in a first-choice studio. Jones broke the client of this nasty habit by having his traffic person call the agency person back soon after the original 'on-hold' booking, to report that "another client wants the time being held for you."

Guy Costa confirmed the point by saying, "Don't be at the affect of your market," a situation he indicated could happen to even the best studios. He pointed to audience member Skip Saylor, whose Los Angeles studio caters to what Saylor himself describes as musicians who've been "86'd" by every other Hollywood facility.

"Skip probably makes more in his studio than we do at Motown," Costa offered.

"Yes, and with your artists!" Saylor replied.

The baseline in business, Jones offered, is mirrored in a simple statement that probably should be framed and placed directly in front of all studio owner/managers: "If you're coming to the party, come prepared." Which is a kind of an inverse, positive variation on the old saying: "If you can't stand the heat get out of the kitchen."

Where do studio owners learn all they apparently know? Murray Allen, president of Chicago Recording, matriculated through the studios of Chicago as a session player, and brought to the conference a curious and charming mix of street smarts and board-room intellect, heavily seasoned toward the former. Allen's business wisdom is summarized in Table 4. The last of these three maxim's sparked his determination to "Add a Synthesizer Room," the title of his presentation.

Table 4. Murray's Maxims

1. I never met a banker I didn't like.
2. However, never let the person you're doing a deal with introduce you to *his* banker.
3. "Rewind" is wasted time.

Universal, by Murray's own account, doesn't use a common form of business plan, finding most of them to be too generalized for the specific needs of the recording industry. Business plans that cause a change in the marketplace should be implemented, he considers, rather than those that simply capture some share of a marketplace which already exists.

Given Universal's successful track record in the \$20 million per year Chicago audio market, it seemed sensible that Allen's studio should create ways to improve client services *and* productivity. He committed his facility to purchase a disk-based recording system. Equipment costs for the Synclavier, console, video and outboard audio equipment were \$400k. Universal's cost-of-money formula is 2% per month, or \$8,000 a month to break even.

At three hours a day utilization, five days a week, it was determined that the basic session price would be \$120 per hour. Additional time booked becomes profit, as do additional sales such as video lock-up at \$50 per hour, 24-track rental at \$50 per hour and tape and floppy disk sales. A total of \$50k in leasehold improvements came out of cash flow, and all equipment is leased at 2% above the prime.

Was it worth the investment? Allen says that, so far, his new room has booked \$200k—only 10% of which was not directly attributable to the Synclavier system. What is more, Allen concludes, "I don't have to rewind tape!"

Second day: afternoon session

The following may take you a few minutes to accept, but if you started an equipment rental business, and then rented all or most of your specialized gear to your *own* recording studio, you'd probably be making a lot more money than you do right now. Maybe even as much as Chris Stone, president of the Record Plant, Los Angeles, whose presentation entitled "Getting Into the Rental Business" confirms the old saying that the best defense is a strong offense.

"Five years ago we said diversify or die," Stone recalls. To avoid the latter, he set up Livingstone Audio Rentals as a separate yet equally and fully owned extension of the Record Plant.

Why the rental business? It seems that Stone's business planning led him to conclude that his studio (and not an outside rental company) should derive income from all the toys requested by clients. In effect, the Record Plant would be its own best customer. Clients are aware of the corporate relationship, and even support it—because a studio's hourly rate charge goes above the line (is fixed) in most budgets whereas rentals go below it (are not fixed).

Stone told the SPARS conference that he encountered stiff resistance when he attempted to add $25 per hour to the price of a room into which he installed a new digital multitrack. Clients were refusing to pay a premium of $250 per day, based on a 10-hour session, yet were ready, able and willing to rent the same unit for $1,000 per day!

And when Stone doesn't have a particular unit on hand to rent, he re-rents from a friendly competitor and splits the fee. In addition, when he rents to others, the cost of trucking is tacked on as an extra charge.

In 1986, the rental company returned a 30% net profit on total sales, with a large part of that activity based on 4-day weeks and 3-week months. In six months, Stone had recovered 144% of his original investment on much of the gear. His collective wisdom is summarized in Table 5.

If some studio owners find it difficult to owe money, Stone says the only way he can sleep at night is to owe a lot of money.

"The banks become your partner" he confides, and they're not going to let anything [bad] happen to you." Stone says he's sleeping very soundly these days.

Martin Polon, president of Polon Research International, probably knows more about the economics of the recording service industry than any other civilian. His presentation was entitled: "An Economic Forecast for the Remainder of 1987, with a Focus on the Studio and Recording Industry."

Polon's forecast, I must report, was based on a negative short-term premise: How do we all survive until the Nineties? After which Polon predicts smoother sailing and surer times.

Meanwhile, there's the federal deficit to overcome; its effect on currency rates clearly impacts studio owners.

"Remember that recording studios are importers," Polon stresses, "that buy equipment from Japan, Germany and Britain, to name three supplier nations."

What happens to the value of the yen, mark and pound, relative to the dollar, is *critical* to studio owners.

Interest rates rise and fall on currency and deficit movements. Polon considers that interest rates are not going to fall again soon. In his view, they'll probably hover in the average 10% range, and could rise further. As a result, waiting for prices to drop before purchasing equipment is probably not a valid proposition—not now anyway.

Polon's strongest thrust centered on taxes and the new federal tax laws.

"Learn them," he advised. "Consider the way your business is configured. Are you operating as a DBA? Then your tax bite could be a great deal bigger than if you formed a sub Chapter S corporation."

Polon estimates that the latter type of business, earning under $100k in profit, might pay a tax rate of 17%.

Table 5. Stone Cold Logic

1. Rent before you buy.
2. Cartage is a revenue producer, not not an ancient city.
3. The three Rs are: Risk, Reward and Ration, not all that other stuff.
4. Hot new gear can return 4%-5% of original cost per day.

Regarding equipment acquisition, he pointed out that before the new tax laws went into effect it was probably more advantageous to own equipment and lease the space to house it. However, with the loss now of Investmant Tax Credits it is almost certainly more prudent to lease equipment and own the facility.

What items do you lease and which do you purchase? Polon favors leasing of the more disposable hardware, such as computers. A lease on a high-ticket, long-term investment, such as a console, leaves the lessee paying for it twice—in big principal and big interest payments.

Not everyone agreed on this latter point. Murray Allen of Universal Recording raised another consideration: If the cost of leasing may be higher (and remembering the loss of ITCs), then taxes levied upon purchase at the end of a lease period are computed on the basis of the unit's lower value at that time, than when it was brand new.

Motown's Guy Costa, who moderated the conference, closed the 2-day event with the recommendation that studio owners consider two sources of planning design information: *Venture Plan*, a software package by Cliff Allen; and *Develop Your Business Plan*, a book by Richard Leza and Jose Placencia.

In summary, the conference stressed that planning is an essential element of business. Without it, studios survive by sheer good fortune or sheer energy. But these abstracts are *not* appropriate substitutes for planning.

All too often, studio owners do not plan so that they might properly identify the most basic business realities. Clearly a case of the Man Who Mistook His Multitrack for a Client! R·E/P

The Audibility of Electronics

By John Eargle

Listening tests can be useful in evaluation of a variety of pro-audio equipment, including monitor amplifiers. However, it is important to separate myth from reality when making sonic judgments of electronic hardware.

The range of listening experiences that take place in a pro-audio dealer's demo room are normally quite different from those that occur in a typical high-end hi-fi store. A recording or production engineer is concerned with a large palette of sound differences, beginning with microphones and progressing on through equalizers, limiters, compressors and, of course, consoles. While auditioning a new piece of studio equipment, an engineer is certainly concerned with what it sounds like; but he may also be as concerned with its reliability and stability during a long tracking or remix session.

When it comes to power amplifiers, engineers may be inclined to take matters more or less on faith, feeling that the unit's specifications pretty much tell all. Or, they may feel that any sonic differences between amplifiers are a good order of magnitude less than those resulting from microphone choice, microphone placement and signal processing, and may thus be ignored.

On the other hand, the hi-fi specialist store deals with a very simple transmission chain, consisting basically of source, pre-amp and power amplifier. It is worth noting, however, that many of the developments taking place in high-end consumer audio have found their way into pro-audio systems. Many studios routinely specify exotic consumer amplifiers to power monitor loudspeakers, and the present concern with high-grade cabling and hookups had its origins in the hi-fi world. Certainly, we have witnessed the impact of high-grade consumer loudspeaker systems in classical recording.

John Eargle is president of JME Consulting Corporation and a regular contributor to RE/P.

It is a rare listener who will freely admit that they hear no differences at all.

Subjective differences

What about the differences reported to exist between amplifiers having substantially the same measured performance parameters? Does a $6,000 stereo amplifier necessarily sound better than a $600 model? Controlled listening tests generally lead to the conclusion that any differences are minimal—as long as both amplifiers are operated below their clipping point.

Going beyond power amplifiers, what sonic differences might exist between microphone pre-amps, or between a straight-through, line-level input/output path through a pair of consoles of similar architecture but of different manufacturer? Where measurable differences exist, we would expect some listeners to hear differences some of the time. But, it is often surprising how much distortion may go undetected by many listeners on certain types of music.

I will attempt to sort out myth and reality associated with sonic judgments of electronic hardware.

The testing environment

Too often, power amplifiers may be compared under the most informal of conditions. In a typical dealer showroom, either professional or consumer, an amplifier may be auditioned for 15 minutes. It is then removed and another amplifier put in its place. The listeners usually know which amplifier is which, and often they will be virtually unanimous in their judgments of sonic characteristics. It is a rare listener who will freely admit that they hear no differences at all.

The problem is that there's often an expectation that a certain model will sound better—and so it will. Even the mere tactile aspects of an amplifier can influence that expectation. Extra heft, finely turned metal knobs, big handles, meters and the like may all be calculated to produce this reaction.

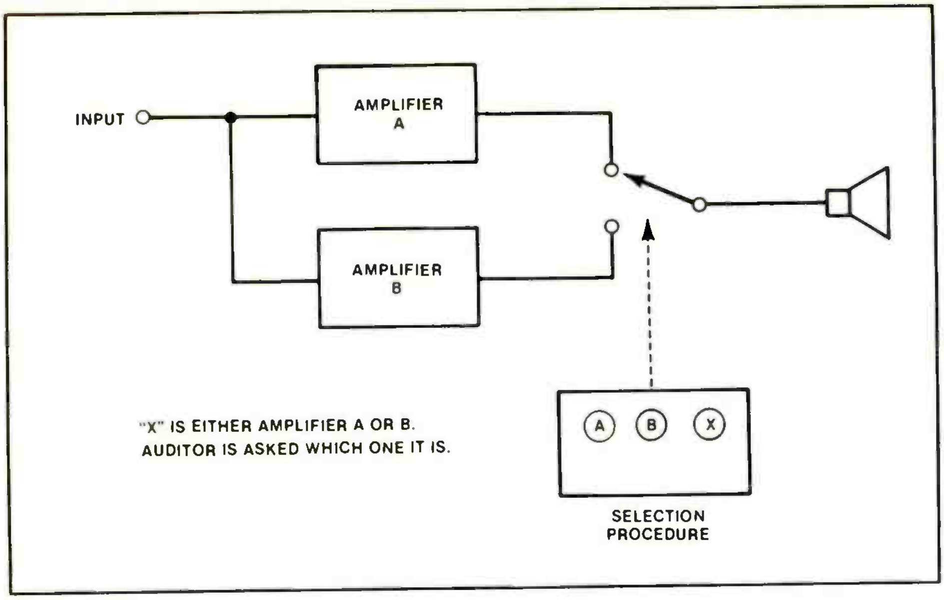


Figure 1. Double blind testing

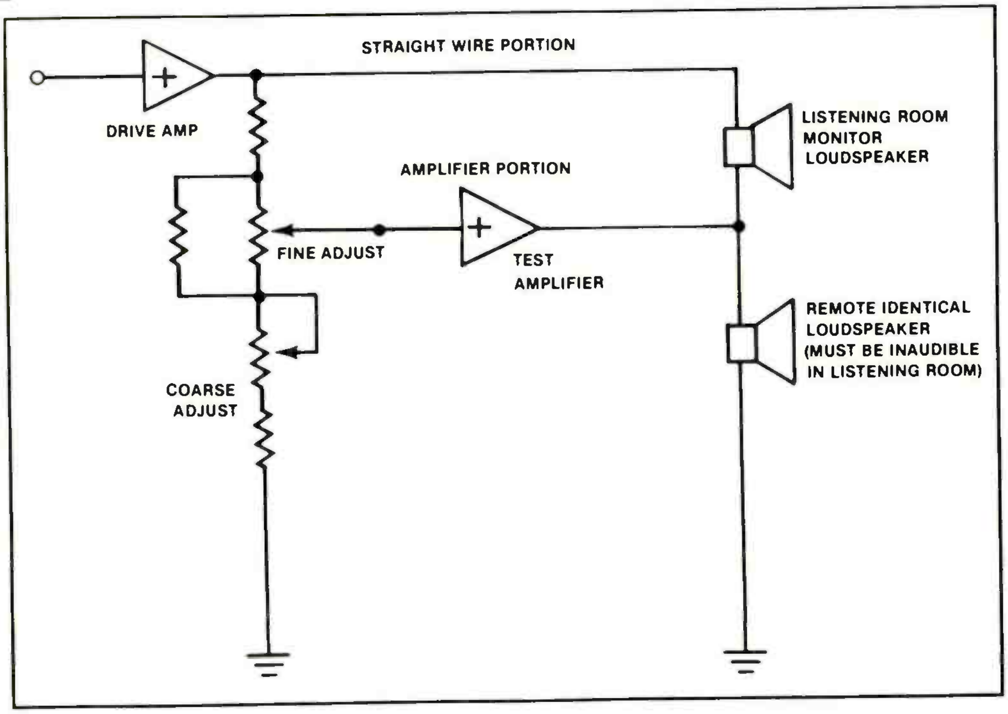


Figure 2. Hafler's straight wire differential test.

Even when there is no particular expectation, the first person to comment on sonic differences in an articulate manner tends to sway the whole group.

The testing dilemma can be solved through what are called "double-blind" procedures. Neither auditors nor testers know which of two amplifiers is playing at the time of the trial. Auditors are merely asked to identify differences, not preferences. For example, in a given trial, an A-B switch is in the hands of an auditor. He is asked to switch back and forth between the two amplifiers, noting any differences that may exist between A and B.

When he is ready for the actual trial, he presses a button labeled X: X may be either A or B, and the auditor is asked to determine which it is. After he makes his judgment, he goes on to the next trial. X is randomly varied between A and B so that the listener approches each trial fresh. The basic test setup is shown in Figure 1.

The key here is that the auditor is listening for differences, *not* preferences. When levels are carefully matched between amplifiers, and when the amplifiers are operated within their linear power limits, then most auditors—skilled as well as unskilled—fail to reliably detect differences.

Such a listening test, conducted by David Clark and Ian Masters, was reported in the January 1987 issue of *Stereo Review* magazine. Five amplifiers were used, ranging from a low-cost receiver to a pair of highly regarded, expensive mono tube amplifiers. Between these extremes was at least one amplifier widely used in driving control-room monitor loudspeakers. A total of 25 auditors, on an overall statistical basis, *failed* to detect differences between the amplifiers. The amplifiers were carefully operated within linear limits, so that the nature of amplifier overload and recovery were not apparent.

While no single test is ever definitive enough to make the sweeping statement that nobody can reliably hear differences—ever—note that the *Stereo Review* evaluations are in accord with double-blind tests that have taken place elsewhere.

There is a tendency to "go with the crowd" and hear things that are not really there.

What differences can be heard?

Any slight difference in gain setting between amplifers A and B in a double-blind test can be spotted easily by skilled auditors. Usually, the gain difference may not be commented on for what it is; rather, the louder amplifier may be judged as brighter, or having more detail.

In double-blind tests of phono pre-amplifiers, it is common to use an inverse RIAA filter so that the response of the phono section can be tested with an external program source (not a vinyl disc). Under these conditions, it may be possible to identify slight variations in RIAA response of different pre-amps. Typically, gain is matched at 1kHz, and should a given pre-amp phono section exhibit a slight rise in its low-frequency boost (a not-uncommon occurence), then many auditors will judge it as sounding warmer.

In a typical recording studio evaluation, microphone pre-amplifiers sometimes can be identified by the spectral characteristics of their noise floor, often near a subliminal level, or by their susceptibility to clipping at high mic input levels. (Both of these aspects were important in the tube-vs.-transistor comparisons made a few years ago.) Such problems as these are largely operational in nature, but they lend credence to the belief that consoles can sound quite different from one another.

Another concern here is the effect of excessive signal transmission through a particular console. Many engineers routinely switch out, or patch around, any part of the console architecture not actually in use. Is the result audibly better? Double-blind testing would certainly determine that answer but, even on general principles, and in the absence of more compelling reasons, it is a good idea to bypass the unnecessary.

The critical loudspeaker-amplifier interface

In the double-blind amplifier tests mentioned earlier, the loudspeakers used presented a fairly smooth resistive load to the amplifiers, making their job a bit easier. Conventional studio monitors may be quite different. Because of their relatively high efficiency, the unit's motional impedance curve may show the effect of considerable load reactance.

At the AES Convention in Los Angeles last Novemeber, Stanley P. Lipschitz and John Vanderkooy presented a paper entitled "Computing peak currents into loudspeakers" (preprint number 2411). What they and others have noted is that under certain transient-drive conditions, a loudspeaker may actually present an effective *resistive* load to an amplifier that may be less than *one-half* its steady state impedance minimum value.

In particular, they measured a theater loudspeaker system with a rated impedance of 4Ω. From their tests, they discovered that the LF section had an actual minimum of 1.48Ω, while the HF section had an actual minimum of 2.4Ω. This is the type of information that an engineering consultant laying out a motion picture sound system should have access to, but which is not present on any manufacturer's specification sheet.

In a typical recording studio evaluation, mic pre-amps sometimes can be identified by the spectral characteristics of their noise floor.

While most modern solid-state amplifiers may voltage clip rather gracefully, the onset of internal current limiting in the amplifier may be quite audible. It is not uncommon to see bridged amplifiers which, on a voltage basis, may have a steady-state output capability of, say, 400W. Often, a 100W amplifier, if it has generous current capability, may actually sound better and play louder—simply because it does not go into current-limiting mode as easily as another model.

Bi-amplification is certainly the preferred way to implement loudspeakers in a control room, and it will minimize (but not get rid of) many of these problems.

Yet another recent paper draws attention to the audibility of amplifiers in a different way. David Hafler, writing in the February 1987 issue of *Audio*, describes a method for "nulling out amp distortion." Again, this is a type of A-B test, and is based on the premise that A and B are presented together and subtracted. What remains is simply the difference between the two. In this particular test, a comparison is made between an amplifier and a straight wire bypass around it.

Figure 2 shows the basic setup for such a procedure. The main loudspeaker is located between the output of the driving amplifier (the straight-wire portion of the setup) and the output of the amplifier under test. Because the output of the driving amplifier is common to both, its action, good or bad, is effectively eliminated. What you hear out of the main loudspeaker, after the null setting has been made, is simply the residual distortion and phase-amplitude anomalies produced by the amplifier under test.

The better an amplifier is, the less sound you will hear out of the main loudspeaker.

Obviously, the better an amplifier is, the less sound you will hear out of the main loudspeaker. Hafler states that the residual output from the loudspeaker varies anywhere from sheer distortion (with bad amplifiers) on up to clean artifacts, due only to phase and amplitude deviations in the test amplifier.

Although these tests are rather critical and demanding of careful setup, they do isolate the actual distortion products that may be generated by an amplifier. In this sense, they offer a degree of ear resolution that the double-blind tests discussed earlier simply cannot attain. It seems obvious that this kind of testing is the way to go, as it will isolate differences between amplifiers well below the acuity of even the best trained listeners. Those recording engineers interested in this testing method should study Hafler's article in detail.

As we have seen, there are many aspects of electronics design that relate to audibility, both subtle and bold. In

Many engineers routinely switch out, or patch around any part of the console architecture not actually in use.

general, these can be detected during impartial A-B testing, however difficult that may be to carry out.

The high-end hi-fi world has no monopoly on dogmatism and the doctrinaire. There are sonic high priests in many places waiting to be defrocked.

There is a tendency to "go with the crowd" and hear things that are not really there. The best advice we have for any engineer, experienced or new to the field, is to be unremittingly honest in all listening tests.

From time to time, all of us have had to invest in certain pieces of new equipment merely to remain competitive in our trades. Sometimes the demand is genuine, and other times it is nothing more than industry faddism. Let us be sure we know the difference. R·E/P

Studio Update

Northeast

Effanel Music (New York) has recently expanded its remote recording services to include equipment and facilities ranging from its compact portable multi-track system to a 45-foot 18-wheel mobile recording studio.

The 3-room mobile unit is equipped with dual **Otari MTR90/2** 24-track recorders, two **Otari MTR-12** 4-track recorders, a 40-input **Series 34 Sound Workshop** console with an extensive patching system.

The portable system can be set up in the lounge of the mobile truck providing two independent control rooms in one unit.

Randy Ezratty, owner-engineer, says he sees the new truck's lounge as an overdub/MIDI room for location album tracking projects.

Ezratty and his assistants have recently completed a 3-week project with Paul Simon in Zimbabwe for the *Graceland Live* concert video. They were contracted for the 64-channel audio portion of the video. *66 Crosby St., 4B New York, NY 10012; 212-807-1100.*

Stardust Recording Studio (Upper Montclair, NJ) has recently installed a **Trident Series 24** console with 36 inputs and short loaded to 28 x 24 x 24. This is the first Series 24 console installed in a studio in the United States. *615 Valley Road, Upper Montclair, NJ 07043; 201-746-2359.*

39th Street Music Productions (New York) has recently taken delivery of two **Timeline Lynx** modules, **Yamaha SPX-90** digital effects unit, **Yamaha DX-711FD** synth, **Yamaha 81Z** and a **Yamaha FB-01.** *260 W. 39th St., 17th Floor, New York, NY 10018; 212-840-3285.*

AudioLink (Boston) a 16-track audio post-production facility has added a **Mac Plus** computer running **Performer** and **Sound Designer** software.

The facilty has also added to its **Sound Ideas Series 1000** Compact Disc sound effects library with the purchase of the **Series 2000 Compact Disc library** of 22 new all digitally recorded Compact Discs. *1380 Soldiers Field Road, Boston, MA 02135; 617-783-0433.*

Trod Nossel Recording Studios (Wallingford, CT) has recently added a **Sony MXP 3036** console. It features a new time code driven automation system that uses hard disk storage and downloads to a 5¼-inch floppy for transfer to another similarly equipped facility. Automation functions are handled at the console via an infrared remote key control unit. *10 George St., Wallingford, CT 06492; 203-269-4465.*

Dreamland Recording (Woodstock, NY) has recently added keyboards including a **Steinway B** and a **Yamaha DX-711FD.**

Other new equipment includes three stereo **Drawmer DS201** gates, **Tube Tech PE1A**, a **Roland DEP5** and a **Neumann M40** mic. *P.O. Box 383, Bearsville, NY 12409; 914-338-7151.*

Midwest

Streeterville Studios (Chicago) has added Bob Miller to its staff as engineer. He has worked at Universal and Genesis studios. *161 E. Grand, Chicago, IL 60611; 312-644-1666.*

The Sound Factory (Olathe, KS) has recently updated their software library to include **Macintosh Plus** with **Southworth Total Music** MIDI sequencer SMPTE jam box, **Digidesign Soundesigner 2000**, and soft synth and **Opcode DX/TX**.

Also featured are a **Yamaha QX-1** 8-track event MIDI sequencer, **Emulator I** 8-bit sampler, **ESQ-1** digital synth with 10k note MIDI sequencer.

Studio A features a **Trident Series 80B** console, **Sony MCI/JH** 24-track and **Studer A810** 2-track. Outboard gear includes a **Lexicon 224X**, **Yamaha REV7**, **Eventide 1745M**,and **UREI Cooper Time Cube**. Monitors available include **Yamaha NS10Ms** and **JBL 4435** with **Crown** amps.

Studio B features a **Tascam M-520** console and a **Tascam MS-16** 16-track, **Studer A-62** 2-track, **Otari 5050** 2-track, **Tascam 44** 4-track. **Aurotones, JBL 4411s** and **Yamaha NS10ms** are also available. *14804 W. 117th St., Olathe, KS 66062; 913-829-2727.*

North Central

Royal Recorders (Lake Geneva, WI) recently took delivery of a second **Mitsubishi X-850** 32-track digital recorder. Locked with its identical predecessor, the new Mitsubishi will help provide a full 64 tracks of digital recording capabilities.

The studio has also added eight **Neve Focusrite ISA 100 EQs** for use with their SSL console. *Highway 50, Lake Geneva, WI 53147; 414-248-9100.*

Southwest

Omega Audio and Production (Dallas) has become the first post-production company in the Southwest to install the **CMX CASS-1** audio editing and console automation system.

The CASS-1 computer aided sound

Pictured (left to right) are Charles Pell, Stephen Potter, Rick Larson and Bruce Bell of Larson Technologies, Burbank, CA.

Studio Update

Pictured in Studio Two of Metropolis Audio Pty. Ltd. are directors [left to right] John Horwood, Ernie Rose, Roger Savage, Ian 'Mack' McKenzie, Ian Robertson and Ted Gregory.

sweetening system is an integrated time code-based audio editing and console automation system that can simultaneously control up to six audio tape recorders and 15 additional sources, permitting precision mixing of an entire soundtrack. It can also automate 32 faders of a VCA control console. *8036 Aviation Place, Dallas, TX 75235; 214-350-9066.*

Southern California

Craig Harris Music, (Studio City, CA) has added a **Massenberg EQ**, a **Roland MKS-20** digital piano module, and a **Yamaha SPX-90** to their facility.

The studio is a 24-track facility augmented by a complete time code locked/MIDI compatible **Synclavier Digital Music System** and features a **Sony 5800** ¾-inch and **PCM F1, Lexicon AMS** and **Drawmer** gates. *P.O. Box 110, North Hollywood, CA 91603-0110; 818-508-8000.*

Larson Technology (Burbank, CA) has purchased seven Otari MTR-900IIs through Everything Audio.

The facility specializes in audio post-production for the television and motion picture industries and provides ADR and Foley post-production capabilities, audio and video transfer services, electronic sound effects editing for TV shows and a systems design/fabrication division. *4109 Burbank Blvd., Burbank, CA 91505; 818-845-4100.*

England

Tape One Studios (London) has added a **Neve DTC-1** digital tape transfer console to be used in conjunction with an **AMS AudioFile** digital random-access recording and editing system. *London, England.*

Battery Studios (London) has ordered a 36-channel **Mitsubishi Wastar** console equipped with **Compumix PC** automation for Studio 5. The new room is being developed to house a synthesizer package and will feature a **Mitsubishi X-850** PD-format digital 32-track and a **Fairlight Series III.** *14/16 Chaplin Rd., London, England 01-459-8899.*

Australia

AAV Australia (Victoria, Australia) has announced a new joint venture with three of Australia's leaders in the recording industry.

Roger Savage, former original partner in Bill Armstrong Studios (later to become AAV Australia); **Ernie Rose,** manager of Audio at AAV Australia and **Ian "Mack" McKenzie,** former manager of Platinum Studios and an original engineer with Bill Armstrong Studios are partners with AAV Australia in the new venture called **Metropolis Audio Pty. Ltd.**

Directors of metropolis will also include **Ted Gregory, John Horwood** and **Ian Robertson.**

Gregory, general manager of Metropolis, says, "Success in todays fast-moving music and sound recording industry is depending more and more on a mixture of entrpreneurial flair and skilled operators with the best facilities," he added, "they all have reputations in these fields which are second to none. This venture affords them an opportunity to share in the future of music recording in Austrailia."

Over the years, the studio has recorded acts such as David Bowie, Mike Brady, Joe Cocker, Men at Work, Leo Sayer, John Farnham and the original soundtrack score for the Australian film *Crocodile Dundee.*

McKenzie, who won the 1974 "Engineer of the Year" award for Renee Geyer's album *It's a Man's, Man's World*, has recently worked on albums/singles for performers such as Pseudo-Echo, Kids in the Kitchen, Geisha, Little River Band and John Farnham.

"Recording is a unique kind of business and it's important that those in management are actually involved in the recording process to understand the needs of a studio. This formula worked at Platinum and I'm confident that it's going to be the same at Metropolis," McKenzie says.

Future plans include rebuilding their audio facilities, a refurbishment program for Studio Five and adding new meeting and lounge areas. *180 Bank St., South Melbourne, Australia 3205; 03-699-1844.*

R·E/P

New Products

Maxell R-120DM digital audio tape

The new R-DAT cassette features durable magnet particles and binder system to produce high output level, high carrier to noise ratio and low dropouts, the company claims.

The R-120DM can record up to two hours of program material.

The company is also introducing a PCM ¼-inch tape available on 10- and 7-inch reels.

Circle (81) on Rapid Facts Card

Pristine Systems Sound Manager software for CD libraries

The software allows the user to find and play any CD music cue or sound effect in a sound library within seconds, the company claims.

Other features include the ability to locate and retrieve sound effects and music cues using the Sony CKD-006 multiple disc CD changer.

The software runs on an IBM-PC and compatibles.

Circle (86) on Rapid Facts Card

Forat Electronics F16 sampling drum computer

Based on 16-bit digital sampling technology, each of the unit's 16 voices has independent tuning, volume and pan controls, and is available through direct outputs or a built-in stereo mixer. Sounds are stored on a built-in floppy disk drive.

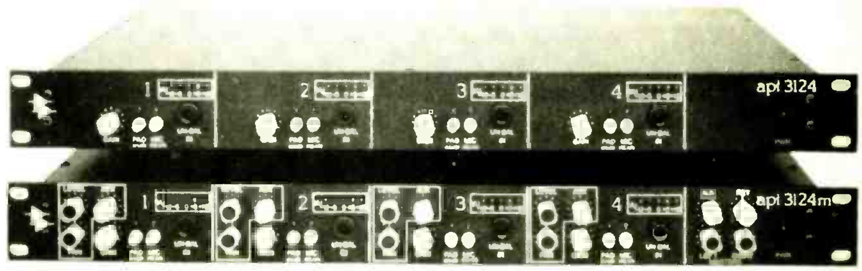
Each sound comprises 500,000 bytes of data, which provides individual samples of up to six seconds at full audio bandwidth, or 25 seconds with reduced HF response.

Circle (106) on Rapid Facts Card

API model 3124 mic pre-amplifier

The new mic pre-amp comprises four model 312 mic cards housed in a 1U self-powered unit. It features balanced XLR connections on the rear, unbalanced high ¼-inch inputs on the front panel, gain control, LED metering, 20dB pad, Jensen input transformers, and all-discrete circuitry using the API model 2520 op-amp.

Circle (102) on Rapid Facts Card



Richmond Sound Design Command/Cue 4096 console

The computer-controlled console is designed for theater sound and features levels and routing control for up to 4096 faders.

The unit can be configured into different formats to handle various sound-effects matrix systems. It can also be used with live PA mixers running mics while the computer is handling taped effects and automating level control.

Circle (101) on Rapid Facts Card

JL Cooper PPS-1 sync box

Designed to allow MIDI syncing between multitrack machines and equipment that can respond to Song Position Pointer, the PPS-1 converts MIDI sync to an FSK sync tone that is used to stripe the tape.

MIDI Song Position Pointer enables the unit to "chase" tape and eliminates the need to return to the start of the tape when corrections or overdubs are made.

Circle (97) on Rapid Facts Card

Cetec Vega model R-33 wireless mic system

The new portable wireless transmitter can be mounted on a camera, sound cart, shirt pocket or belt.

The receiver is designed for applications requiring ultra-portable equipment, such as field TV and film production; camera-mounted ENG wireless; and sound-system feeds to roaming cameras and recorders.

The R-33 Pro Plus can run up to eight hours on a 9V battery. Signal-to-noise ratio and dynamic range are a quoted 104dB.

Circle (90) on Rapid Facts Card

Community Light and Sound CS60B sub-woofer loudspeaker

The new base reflex loudspeaker features four proprietary ferro-fluid cooled 15-inch drivers and a large ducted port. Operating frequencies are a quoted 35Hz-800Hz; power handling capacity is rated at 600W rms/1.5kw program, with a maximum SPL of 132dB measured at 1W, 1m.

The cabinet also features an internal 150Hz crossover network with dual high-pass output jacks.

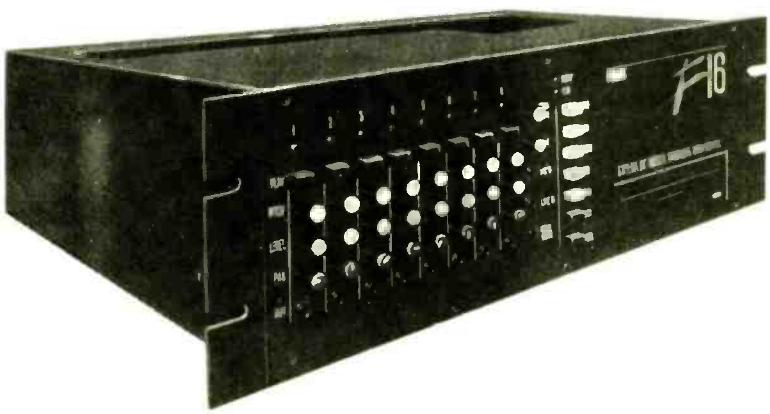
Circle (111) on Rapid Facts Card

Shure model 839 condenser lavalier mic

The lavalier mic features a wide-range frequency response tailored for lavalier operation, low distortion and low RF susceptibility.

The model 839 can be used with either battery or phantom power.

Circle (109) on Rapid Facts Card



New Products

Gold Line ASA 30B real-time analyzer

The new third-octave analyzer features 30 bands on standard ISO centers, a built-in microphone SPL reading from 30dB-123dB and external line input levels of -85dBm to +4dBm.

Circle (103) on Rapid Facts Card

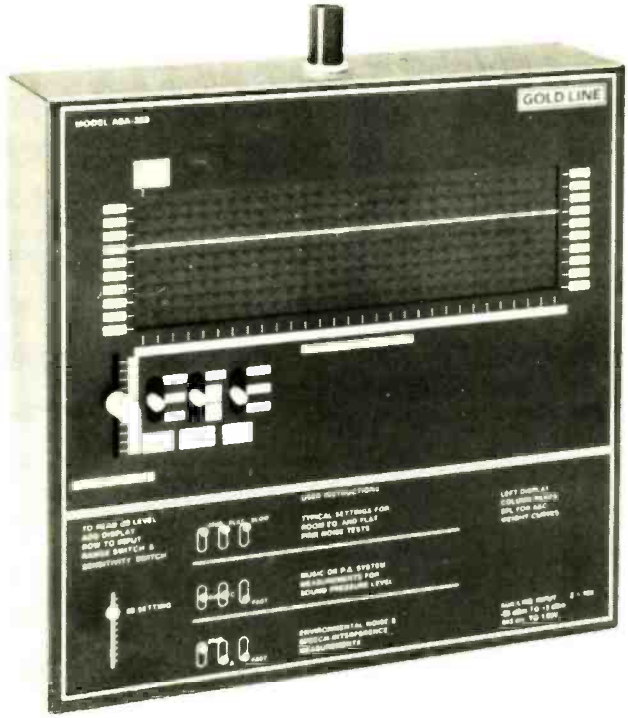
Soundcraft Series 200BVE production console

The new mixer is designed with an optional linear cross-fade depth control, for linking to external video editors to provide automated level control during assembly editing of audio material to picture.

Three types of input modules are available, including a mono input with 4-band, fixed frequency equalizer; a mono input with a 4-band equalizer and variable frequency of the two mid-bands; and a stereo input module with 4-band fixed frequency EQ.

Frame sizes accommodate 8, 16, 24 or 32 channels.

Circle (107) on Rapid Facts Card



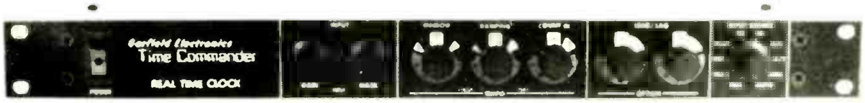
New Products

Garfield Time Commander real-time clock and synchronizer

The unit syncs to live input, such as MIDI note data, and provides seven simultaneous sync outputs.

An input source control selects sync to 24 or 48 ppqn clocks. Outputs include 24, 48, 96 and 384 ppqn clocks. The unit also features zero to eight beat and adjustable count in or manual cueing, on beat punch-in/out, real-time adjustment of lead/lag and a SMPTE/MIDI Time Code option.

Circle (117) on Rapid Facts Card



BGW model GTA amplifier

The modular unit, which is designed for live-performance and touring applications, features low feedback circuitry, built-in dc speaker protection and oversized indicators.

Each input features XLR balanced connections and two ¼-inch TRS phone jacks. Front-panel controls comprise detented gain controls with 41 steps for each channel.

Circle (104) on Rapid Facts Card

Aphex Type E Aural Exciter

The new unit is designed for portable stage and studio use. Instruments or mics can be plugged directly into the Type E, negating the need for a dedicated pre-amp or mixers.

The Type E can also serve as a low-noise, pre-amp and direct box, the company claims.

Circle (110) on Rapid Facts Card

AKG Acoustics V3.0 software for the ADR-68K effects processor

The new software provides five new programs including Dual Digital Delay Lines, Multi-Tap Stereo Processing, Multi-Effects, Poly Chorus and Stereo Sampling.

Also featured are MIDI preset send and receive, 100 factory presets, a register recall function and a mode for computing delay time in terms of beats per minute and note value.

Circle (113) on Rapid Facts Card

Shure FP51 gated compressor-mixer

The new device combines a gated memory compressor with a 4-input/mono mic mixer in a portable unit.

The unit provides a 40dB compression range with a compression ratio of approximately 10:1 in normal operating range.

The FP51 also features four transformer-coupled XLR inputs and one output, each switchable for mic- or line-level operation. It features phantom power plus a built-in oscillator for line or level checks.

Circle (105) on Rapid Facts Card

Community Light and Sound CS52 3-way cabinet

The unit has a quoted bandwidth of 40Hz-20kHz and uniform wide-angle dispersion, an extended frequency range, and a coherent wavefront design, the company says.

Midrange frequencies are handled by a 6½-inch cone drive coupled to a short compound horn. High frequencies are handled by a PZT driver mounted on a wide-angle pattern control horn.

The unit uses a fuseless protection circuit to guard against excessive input levels.

Circle (115) on Rapid Facts Card

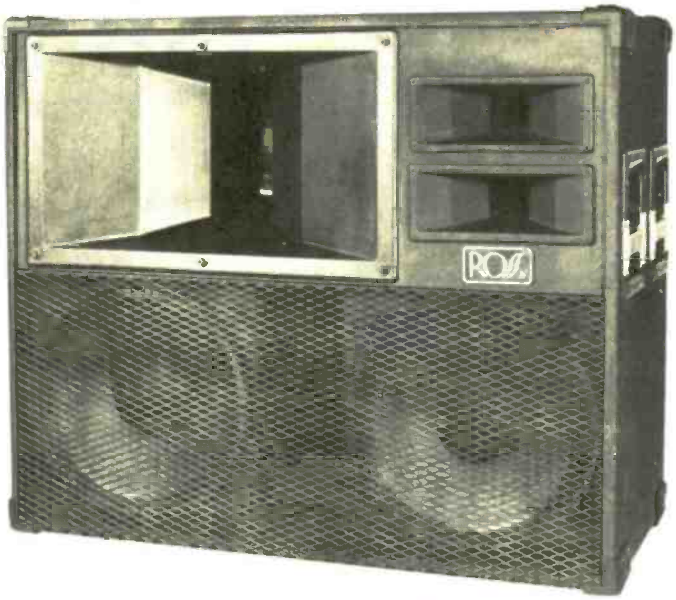
New Products

Ross Systems Hurricane speaker systems

The new speaker system series can be bi-amplified and features caster sockets, stacking corners, edge extrusions and heavy-gauge expanded steel grills. A computer-optimized crossover working at 300Hz and 4kHz was developed for each model.

- HF01 constant directivity mid-horn covers the range from 300Hz-4kHz and is said to eliminate crossover distortion, phase cancellation, and horn throat distortion.
- H115CD is designed for high SPLs in a limited space or as close-field stage monitors. The 3-way system provides vocal projection and full-range response, the company says.
- H118CD is designed for high SPLs with extended base response. Frequency response is a quoted 40Hz-18.5kHz, ±4dB.
- H215CD is designed for high SPLs and extended low-frequency response. Frequency response is a quoted 37Hz-18.5kHz, ±4dB.
- H218CD is designed as a single-cabinet sound system that features dual LF18 drivers, dual HF01 high-frequency constant directivity horns.

Circle (116) on Rapid Facts Card



New Products

Kurzweil
Sound Modeling program for K150

The new program allows the user to design new sounds for the 150FS synthesizer using an Apple IIe PC.

Frequency and amplitude envelope of up to 64 partials per sound model may be designed in data table or graphic display form.

The program requires Version 1.6 software for the K150FS synthesizer.

Circle (112) on Rapid Facts Card

Applied Creative Technology
DB8 direct box

The 8-channel transformerless DI box is designed for drum machines and keyboards, and features line-level, high-impedance unbalanced inputs and mic-level, low-impedance balanced outputs.

Frequency response is a quoted 10Hz to 25kHz, ±0.2dB, and maximum input level +10.5dB. Red LED clipping indicators are provided for each channel.

Circle (118) on Rapid Facts Card

Sound & Vision
MICRO 1 tape controller

The new unit features direct search-to-cue, auto punch in/out, tape and record loop with pre-roll, rehearse mode with pre-roll and cue tone output.

Also featured is shuttle speed, digital read out, full transport controls and trigger out.

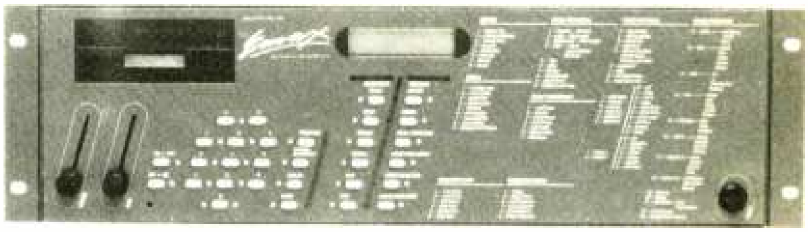
Circle (114) on Rapid Facts Card

E-mu Systems
Emax rack digital sampler

The Emax Rack Digital sampler is a rack-mountable version of the Emax, a digital sampling keyboard. The unit can add power to MIDI keyboard-based versions and offers sampling via a MIDI-equipped guitar.

The unit features MIDI Overflow Mode and multi-timbral capabilities make it suitable for MIDI keyboard and sequencing setups.

Circle (100) on Rapid Facts Card



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