January 1987

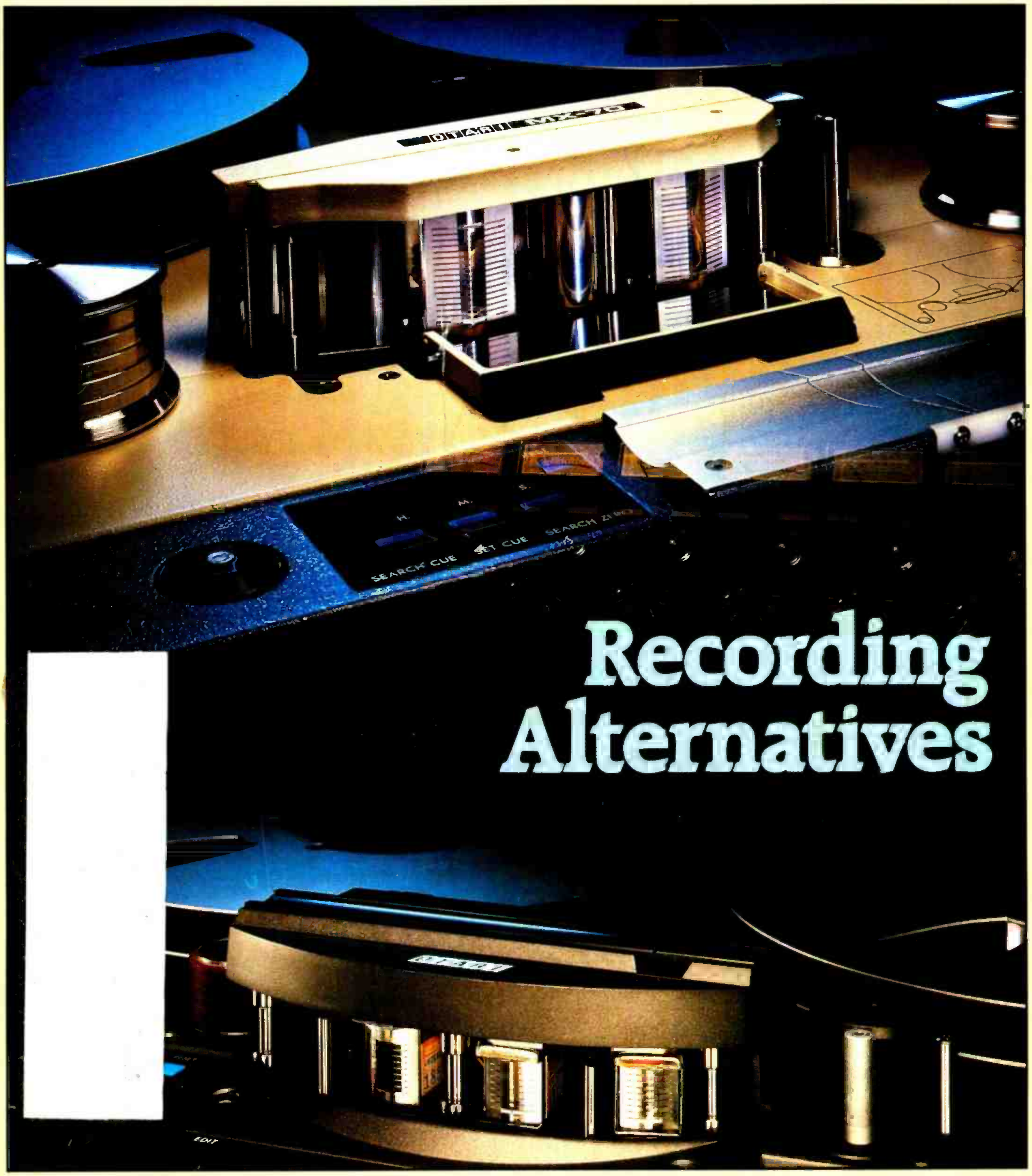
Recording ENGINEER/PRODUCER

T.M.

The Technical Journal for Audio Professionals

Narrow Gauge
Page 24

$4.00



Recording Alternatives

Why Leading from Tape

Finally the true potential of the digital studio can be realized. No longer are you limited to storing and retrieving digital data on media designed for outdated tape technology. The Direct-to-Disk Multi-Track Recording System by New England Digital uses multiple, high capacity, winchester hard disk drives for data storage.

The Synclavier Tapeless Studio™ is available today!

Start with the industry-proven Synclavier Digital Audio System. Now available with a 32 track digital sequence recorder, 32 megabytes of high-speed RAM and on-line storage support for up to 2000 megabytes of sampled sounds.

The Direct-to-Disk System can be added at any time. Operation is simple! The system is controlled by the Synclavier's keyboard control panel. The easy-to-use interface provides all standard tape recorder functions, and more!

The finest quality 16-bit A/D conversion processes and output filtering technology available are combined with variable "stereo" sampling rates of up to 100kHz to offer audio fidelity unequalled by any other system.

The Direct-to-Disk System stores large volumes of digitally coded information on formatted winchester hard disks. Once stored, this information can be accessed randomly at any point in the recorded program material. This random access technology provides virtually instant rewind and sophisticated editing features that would be impossible using conventional technology.

When comparing the Direct-to-Disk System with standard tape-based digital recording there is a dramatic difference. For example, the Direct-to-Disk System does not need error correction. Its negligible error rate contrasts sharply with tape-based digital recorders which require error correction software to compensate for error rates of up to 180,000 bits per hour. This dramatic difference in data integrity illustrates New England Digital's commitment to quality and audio fidelity.

Expanding the system is simple. Start with as few as 4 tracks for overdubbing vocals or live instruments onto your Synclavier sequences; add on more tracks and recording time as needed. With configurations of up to 16 tracks and almost half an hour of recording time large multi-track projects can be easily completed. With the Synclavier's advanced hardware and software architecture, you always have the option to expand.

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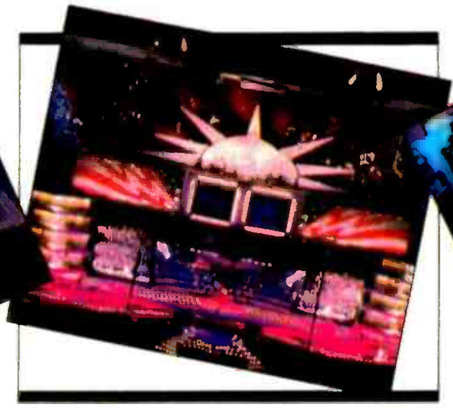

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On the Cover:

In keeping with the editorial theme of "Recording Alternatives," our January front cover shows the headstack layout of an Otari MX-70 1-inch 16-track [top] and a MX-5050 ½-inch 8-track. Photo by Bruce Ashley and courtesy of Otari Corporation.

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Editorial

Audio Concepts

Paralleling recent advances in digital tape machines, signal processors, consoles and random-access editing systems, we have seen changes in the way recording and production engineers need to conceptualize the functionality of latest-generation hardware.

In the past, we could use circuit diagrams and schematics to gain an understanding of an analog console's inner workings. Want to know where the stereo effects returns are routed to the 2-mix bus? Easy. Follow the analog signal flow around the circuit diagram provided in the owner's manual, and it's not too difficult to find what you're after.

Or, suppose you need to interface a new outboard effects unit to an existing console, and are concerned about level and impedance matching? As long as you have a basic understanding of Ohm's Law, an appreciation of signal bandwith, and are familiar with the definitions of dBV, dBu and so on (not to mention the correct polarity for XLR-type connectors), you should encounter few problems in extracting and using the relevant data.

But digital consoles, editing systems and processors are entirely different animals. How do we need to conceive of all the latest microprocessor-based toys that are finding their way into our studios? Would you understand what a certain manufacturer friend of mine meant when he described his latest assignable console design as "an extremely compact, multitasking local-area network?" Maybe not.

I'll concede that this example of "jargonspeak" may be an extreme one. But it is, I would suggest, indicative of the way we need to sneak up on the problem of defining a workable understanding of such hardware.

As I have said before in these editorials, the pro-audio industry is becoming increasingly dependent upon the computer industry to supply the building blocks from which our new technologies will be fabricated. If we are to be able to make the most of the enhanced mode of operation and efficiency such microprocessor-based systems are going to mean to use in the studio of tomorrow, high on anybody's list of priorities, is a basic understanding of, and a ready familiarity with, the terminology used to describe them.

Obviously, design engineers and programmers will be very much more *au fait* with system topographies and data flow throughout the myriad arithmetic and storage processors of which future all-digital systems will be fabricated. Our problem, as users of such hardware, will be coming up with the necessary shorthand definitions and concepts to be able to drive it.

In terms of our interfacing with such systems, I would predict that a familiarity with CRT and liquid-crystal screen displays, mouse-driven "windowing" software and an appreciation of adaptive functionality would be of a high priority. Not only will such familiarity with the underlying potential of such systems enable us to be up and running from the get go, it will also enable us to provide valuable feedback to manufacturers when the time comes to draw up a list of necessary enhancements.

It is often said that any recording or production engineer worth their salt can walk into just about any facility or live-performance situation and get audio from the desk and peripherals within a couple of minutes. With analog technology, this is usually the case.

Substitute an assignable or all-digital console, add a couple of MIDI-based processors, a random-access editing controller, a digital multitrack with timecode, AES/EBU I/Os and video-sync requirements, and the situation becomes a shade more tricky.

And it is not necessarily because the control surfaces bear little resemblance to what engineers are used to with conventional designs. (Few manfacturers can hope to interest potential users in their latest-generation systems if users are going to be forced to work in a way that seems unnatural.) Instead, a ready acceptance of such systems only comes when we can get a handle on the critical design concepts, and not just treat the device in question as a very sophisticated "black box."

It could be argued that audio engineers do not need to worry about bits and bytes, let alone parallel processing architecture, software shells, LANs, etc., to operate the digital studio of the future. What you will need, however, is a clear and precise way of translating what you want to achieve in response to the sound you hear over the control-room monitors, into an action effected by the console, editing system, or effects processor.

To do that, every engineer must begin to cultivate a basic familiarity with the newer examples of pro-audio jargon.

Mel Lambert
Editor

Why your next console should be as difficult to hear as it is easy to operate.

The studio is more complex and less forgiving.

Electronic production techniques using MIDI and SMPTE sync require more control than a "wire with gain" can provide. But as functions and components accumulate, the console's signal path has grown more complex, and its audio performance has suffered. On analog recordings, higher levels of crosstalk, noise and intermodulation were an acceptable price for additional control. On digital multitrack, however, these flaws become glaringly obvious.

Crosstalk blurs the stereo image.

Now that digital recorders have virtually eliminated crosstalk, this is an especially annoying problem. *The AMR 24 matches the channel separation performance of digital multitracks* because it employs balanced buses that eliminate crosstalk the same way mic inputs do. This radical design approach takes full advantage of digital's more coherent stereo imaging.

Balanced buses also eliminate the intermodulation that plagues the sound of conventional "virtual ground" mix amps. *The AMR 24's noise floor is constant whether you route one input to a group, or thirty six.* So you can concentrate on the music without distractions from the mixer, even on digital multitrack.

Features shouldn't degrade audio performance.

Automation widens creative possibilities — and narrows the margin for console error. For example, FET mute switches that are "silent" individually can produce audible glitches when grouped. The AMR 24's carefully controlled switching time constants eliminate this problem.

Every circuit in the AMR 24 has been calculated with equally close attention. Each stage has at least 22 dB of headroom; total dynamic range is over 100 dB. Even so, *unused stages are bypassed to produce the shortest effective signal path in every operating mode.*

Perhaps the AMR 24 is a product of extremist engineering. But as we see it, optimum audio performance, not simply a revised layout, is what makes a console automation- and digital-ready.

The feel is familiar, the functions are unprecedented.

The AMR 24 facilitates innovative production techniques within a classically split configuration. Master Input Status switches select mic inputs or line returns on all input channels simultaneously. In its mixdown configuration, the AMR 24 will handle up to 60 tracks, because the 24 Track Select switch changes the monitor returns to line returns normalled to your second 24 track (or to synchronised "virtual tracks" from synthesisers and samplers). The monitor returns have aux buses, solo and mute, plus four bands of EQ and long throw faders, so this flexibility is achieved with no loss of audio quality. For additional effects returns, the Fader Reverse function creates an additional 24 patch points through the cue send faders.

Imaginative design and uncompromising construction give the AMR 24 flexibility and sonic transparency that represent clear achievements: especially clear on digital recordings. For all the facts on this innovative console, send your business card or letterhead to:

DDA AMR 24

SPARS

Klark-Teknik Electronics Inc., 30B Banfi Plaza North
Farmingdale, NY 11735 (516) 249-3660

Unit #1, Inwood Business Pk., Whitton Rd.
Hounslow, Middlesex, UK TW3 2EB

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News

SPARS Studio Business Conference

As revealed in last month's issue, the Society of Professional Audio Recording Studios will host a 2-day conference entitled "Business Plans for the Studio" on March 28th and 29th, at the UCLA Graduate School of Management, Los Angeles. Attendance is limited to 75 individuals.

Topics and speakers are as follows:

- *Overview: step by step construction of a business plan*, by Guy Costa, Motown-Hitsville Studios, Los Angeles.
- *Opening a new studio—a start-up business plan*, by Bruce Merly, Clinton Recording, New York.
- *Entry into video—a diversification business plan*, by Fred Jones, Fred Jones Recording, Los Angeles.
- *Evolution of the multi-studio operation—an expansion business plan*, by Wilbur Caldwell, Doppler Studios, Atlanta.
- *Adding a new location—a diversification business plan*, by Dave Porter, Music Annex, San Francisco.
- *Adding a synthesizer room—a diversification business plan*, by Murray Allen, Universal Recording, Chicago.
- *Getting into the rental business—a diversification business plan*, by Chris Stone, Record Plant, Los Angeles.
- *Summary and Review:* A summary of approaches, review of books, software and other materials available to help construct business plans, by Guy Costa.

The cost of the conference for SPARS members is $130 for registration before March 9, and $180 afterward; for non-members the cost is $230 before March 9, and $280 afterward. For further information, contact the SPARS national office at: 818-999-0566.

3M and Nagra announce new standardization agreement

All newly produced Nagra tape machines for sale in the United States and Canada will be aligned with 3M 808 mastering tape. The two companies have also agreed to intiate a "pack out" agreeement in which 3M product tape will be included with each new machine shipped to these countries.

"We decided to standardize on 3M 808 because it met our key performance criteria—print level and distortion—which are intended to ensure optimum audio quality is achieved with our equipment," says Manfred Klemme, Nagra's vice president of sales. "The product offers the lowest print characteristics that we've seen, as well as low distortion qualities."

The American Comedy Network adds duplicaton system

Described as the only East Coast facility utilizing a dedicated high-speed, reel-to-reel system, ACN's Bridgeport, CT, facility includes a customized Electro-Sound Model 5000. ACN is a major syndicator of radio comedy with 25 million listeners and 151 affiliate stations.

According to ACN executive producer David Lawrence, "We took state-of-the-art cassette duplication technology and built the perfect reel-to-reel system that combines speed [a 32:1 duplication ratio] with broadcast-quality sound.

"The system's speed allows us the opportunity to satirize late breaking national news items.

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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INTERTEC PUBLISHING CORPORATION

News

Dolby ST used on 70mm "Star Trek IV" prints

Star Trek IV: The Voyage Home, released on Nov. 26, was the first Dolby Stereo film to use Dolby Spectral Recording on 70mm release prints. Although sound for the film was edited and premixed onto 35mm mag, the final stems and the 6-track printing master were recorded on a Sony PCM-3324 digital multitrack at Todd-AO Glen Glenn Stage One. The digital 6-track printing master was then copied to an SR-encoded 35mm mag printing master for use in the sounding of over 150 standard 70mm prints that employ Dolby A-type noise reduction.

Later, two SR-encoded 70mm prints were sounded directly from the 6-track digital printing master. These prints played in Los Angeles at the Village Theater, Westwood, and the Cinerama Dome, Hollywood. During a presentation on Dec. 15 at the Academy of Motion Picture Arts & Sciences, Beverly Hills, the film sound industry had the opportunity to compare a 70mm Dolby A print of *Star Trek IV* with one of the SR-encoded 70mm prints. In addition, the 35mm mag SR-encoded printing master was A/B'd with the 6-track master running in interlock with a digital multitrack.

This fall, Dolby Laboratories announced the release of the 1,000th film to use the Dolby Stereo film process, Clint Eastwood's *Heartbreak Ridge*. Since the introduction of Dolby Stereo optical in 1975 for *Lisztomania*, use of the process has grown from 12 stereo optical films in 1977 to more than 225 worldwide in 1986. Additionally, over 6,000 theaters in North America and 4,000 theaters overseas are equipped with Dolby Cinema Processors.—*By Larry Blake,* ***RE/P's*** *film-sound consulting editor.*

Mitsubishi announces new leasing plan

In these days of capital-intensive recording and production equipment, the cost of purchasing a new digital multitrack or console sometimes causes problems for facility owners interested in offering current-generation technology. To offer a potential solution, Mitsubishi Pro-Audio Group is offering financing through its new Diamond Leasing Plan, a service of RediVision Leasing.

Established several years ago to meet the leasing demand for Mitsubishi's various video display systems and other capital electronics equipment, RediVision is now also offering "one-stop leasing" services for the company's pro-audio products.

According to Tore Nordahl, MPAG president, "To my knowledge there is no publicly advertised program by a major manufacturer to offer internal financing on a scale as large as this. Any of the major pro-audio companies that expect to see strong growth in the marketplace would have to have a good leasing company connection.

"Although you can't sell equipment with a good leasing plan alone, once you've established yourself in the market, a good in-house leasing plan can be of enormous benefit to both your clients and your own sales growth."

Traditional bank financing generally offers better lending rates, Nordahl continues, but getting the loan approved quickly and easily can pose problems.

"As a result, many of today's studio owners wind up arranging lease financing for their larger equipment purchases, but spend a great deal of time shopping for the best rate of interest," he said. "These two factors were a big part of Mitsubishi's decision to offer in-house leasing services.

"We know the studio business, the people, the equipment and its inherent resale value, so we can make a decision quickly, and provide the best possible lease rate."

Typical lease periods for the X-850 digital multitrack, larger Westar consoles and packages of Westrex film-sound systems are up to 60 months. Smaller equipment packages, such as the new X-400 16-track, can be financed for up to 48 months, and the new X-86 2-track for up to 36 months.

For more information about the Diamond Leasing Plan, contact Cary Fischer at 818-898-2341.

Lynx compatibility for Boss editor

Interfaced via the TimeLine Lynx VSI synchronizer's RS-422 ports in a daisy-chain configuration, the Alpha Audio Boss system will now handle audio editing using the Lynx as a lock-up device. The microprocessor-based editing system also supports Adams-Smith 2600 and Cipher Digital 4700 Shadow synchronizers.

According to David Walker, Alpha's director of marketing, "We've come a long way in trying to be all things to all people by including the Lynx in our system's functional orientation. We eventually hope to support mixed racks of synchronizers to accommodate a user's preference for any combination of Shadow, Lynx or AS-2600 units."

Digital Magnetics, Hollywood, recently received the first Lynx-compatible Boss system.

Benchmark adds architectural and construction services

Benchmark Associates, designers of recording studios, control rooms and performance stages, recently joined forces with architects Downtown Design to form a new company that will specialize in designing and building complete facilities.

"We've had great success as studio designers," says Vin Gizzi, of Benchmark Associates, "but we realize that it's the end product—the studio itself—that our clients are investing their money in. Now Benchmark-Downtown Design can produce that studio for them and maintain tight control over the economic, technical and aesthetic aspects.

"Most projects include reception areas, offices and the customary support facilities for a studio. It's far more convenient and more efficient to have the job handled by one company, rather than an architect plus a studio designer plus a contractor."

The new company is currently working on expansion projects for Greene Street Recording and Power Play Studios in New York. Also under consideration are studios for Andre Perry Video, Don One Recording, Sound on Sound, and London By Night Productions.

For more details, contact Vin Gizzi at 212-688-6262.

Sony to sponsor CD mastering seminars

Intended for music and recording industry personnel interested in Compact Disc mastering, dates of the 2-day seminars are Jan. 13-14, 15-16 and 19-20 in Los Angeles, and Jan. 27-28, 29-30 and Feb. 2-3 in New York.

The seminars are designed to introduce the company's mastering system, and will cover 2-channel digital recording and playback systems, CD mastering systems, digital editing, CD subcode editing and CD cutting.

Neutrik connectors now available in U.S. from HHSmith

The line of Swiss-made connectors includes XLR-type plugs and sockets, goosenecks, speaker connectors and audio accessories.

Of special interest are said to be Neutrik's X Series of XLR-type connectors, which consist of only four parts and eliminate the need for setscrews or crimping. R·E/P

Letters

Advancement of the Byte

From: Susan J. Alvaro, administrative director, Digidesign, Inc., Palo Alto, CA.

With reference to Stephen St. Croix's October "Living with Technology" column, please sign me up immediately as a charter subscriber to his "Synth-Of-The-Month-Club." It is an idea whose time is long past due.

As the producers of some of the more advanced "little blue squares" available, we at Digidesign have excitedly accepted the challenge of transforming the Apple Macintosh into as many "somethings new" as possible, and we're pleased at the response we're getting from users.

Since introducing that "small, blue 16-bit synth" Stephen refers to in his October column, we have had a number of calls from synthesists who were amazed that they were holding more power in the palms of their hands than they had on their studio racks. And, because this power was portable to no less than six different existing samplers, they didn't need to go out and buy new hardware to run the program.

As to his suggestion of standardized communication protocol and faster ports implementation (like SCSI), we're all for it. The thought of what software writers could do, if they weren't continually having to write new ports to accept newer synths, is mind-boggling.

Portable R-DAT

From: Flawn Williams, production engineer, National Public Radio, Chicago.

I was pleased to read Larry Blake's "Film Sound Today" column in the October issue of **RE/P**, where he discussed the requirements of a "digital Nagra" in the R-DAT format. I've been putting a lot of thought into that same topic, anticipating (for what seems like years now) the consumer and eventual professional versions of a portable R-DAT recorder.

I agree with virtually everything Larry specified for the theoretical machine, and have many other ideas I'd like to see considered. First though, a comment on one of your assertions that, to ensure compatibility, R-DAT should record at a sampling frequency of 44.1kHz. I've seen in at least one published reference that the data structure of R-DAT is different from that of Compact Disc. So, even if the sampling rates are the same, some kind of data conversion would be required, right?

[*Editor's note:* Differences between the encoding and error-detection schemes used for CD and R-DAT would not affect the direct digital transfer of audio from Compact Disc to an R-DAT recorder, simply because the 16-bit/44.1kHz data stream would be accessed *after* the replay circuitry. It remains to be seen, however, whether standardized AES/EBU or similar digital input and output ports are provided on all professional R-DAT machines, and whether this feature is offered on CD players for use in video and film post-production facilities.]

Now, on to the wish list:

1. Off-tape monitoring is a must but, in addition, some kind of data comparison circuit should be feasible, to check the digital data coming back off the tape against the data going in. This circuit could activate some kind of icon, similar to the Nagra's speed sensor.
2. Metering should have some kind of peak-storage capability, similar to the Sony PCM-F1 system. If a moving-coil meter is used, I like the twin-needle format used on the Nagra.
3. The recorder should have flying erase heads, and circuits suitable for insert and assembly editing.
4. Digital-in and -out should be provided, as well as analog connections.
5. A real-time counter.
6. Some kind of reasonable peak limiters (optional and switchable). Although I seldom use the ones on a Nagra, they've saved a few situations.
7. Mic-in connections on XLR-type female connectors, with optional 48V phantom power, and phase-reverse, as on the Nagra; plus switchable pads, to ensure compatibility with current mics and the new generation of hotter, almost-line-level mics. The ability to have one mic-level and one line-level input would help.
8. The Nagra's headphone monitor system is great, and should be emulated. I'd also like to see an M/S decode function on the headphone output.
9. Will there be room in R-DAT's digital format for any kind of subcode "tab marker?" This would come in very handy for broadcast production work.
10. Finally, what about the fabled capability for 4-channel recording at 32kHz sampling? Or does this get relegated to the same shelf of esoterica as the idea of taking the audio-only format or an 8mm video system, stacking up a bunch of A/D circuits, and making a 12-channel field recorder out of it?

Does the recent admission by Sony that it is working on developing a digital studio at 20-bit resolution mean that, in the long run, R-DAT (and CD?) will be relegated to the consumer market?

To give you some background on where these ideas are coming from, I've been using the Nagra IVS for eight years to make music and documentary recordings for NPR. For the past three years, I've also been using a digital system based on the PCM-701ES to record nearly 100 concerts for radio and tape/disc release, mostly in the field of traditional music. I'm also involved in getting public radio stations to use digital recorders for time-shifting of programs from the NPR satellite services. Many is the time when I've wished that manufacturers would bring out R-DAT in a pro-audio format **first**, so that we can prove the waters for them.

Please keep up your attention to this pending basic advance in the tools of our craft.

*Reply from Larry Blake, **RE/Ps** film sound consulting editor:*

I agree with you on most of your ideas, especially M/S decoding on the headphone output, and the use of a "flag" like that on the Nagra to verify the data coming off tape.

Deadlines prevented obtaining answers to your more technical questions, but I'll be looking into them.

My first sight of a professional, portable R-DAT machine occurred at the AES Convention last fall, where Sony was showing a prototype unit. I'd like to recall one of the specifications listed in my October column, regarding my ideas for the design of R-DAT location recorders. I spoke of the need for a "rugged, one-hand" (read: manual) function control knob. What I perhaps should have requested is the ability to go into record mode by pressing one switch; also needed is the ability to lock out, on a separate switch, the record function.

I have to admit that there are many uses in the field for some type of semi-automated controls that would not be possible with a manual knob. One example is the "search" function, which allows the head of takes to be found using the "ID marker."

Since the October issue was published, the idea arose of distributing a questionnaire regarding the design of portable R-DAT machines. The results would be a "wish list" reflecting the desires and experience of perhaps dozens of working professionals. Anyone interested in participating should contact me at the address given at the end of this month's "Film Sound Today" column, page 16.

0-120 in 3.6 seconds

Synthesizer

8-voice polyphonic and polytimbral
32 synthesized and sampled waveforms
40 internal, 80 cartridge programs
80-character lighted display
Each voice features:
3 digital oscillators
3 multi-waveform LFO's
4-pole analog filters
15 routable modulation sources
4 complex envelope generators

Sequencer

8 polyphonic tracks
Auto-correct, auto-locate, step edit
Internal storage—2400 notes
Expandable to 10,000 notes

MIDI

Poly, omni, multi and mono modes
MIDI Overflow Mode for slaving units
8 simultaneous polyphonic channels
MIDI remote programming
MIDI guitar controller compatible

If you're interested in a high-performance synth, it's time to test drive an Ensoniq ESQ-1 Digital Wave Synthesizer. It puts 120 sounds at your fingertips as fast as you can switch it on and plug in a cartridge. But that's only the beginning.

In addition to standard synthesizer waveforms, the ESQ-1 features complex multi-sampled waves for a total of 32 waveforms on board. Each of the ESQ-1's 8 voices uses 3 digital oscillators with the ability to assign a different waveform to each oscillator. That's thousands of distinct sonic possibilities.

The ESQ-1 is simple to program because it lets you see what's really going on inside. Its 80-character lighted display shows ten programs or parameters simultaneously. So you'll spend less time writing down numbers and more time laying down music.

A built-in 8-track polyphonic sequencer makes the ESQ-1 an ideal MIDI studio. Each track can play internal voices, external MIDI instruments, or a combination of both. And each track can be assigned a separate program and MIDI channel. Like any good studio, the ESQ-1 can auto-correct timing, auto-locate passages and balance individual tracks during mixdown.

You can build songs made up of 30 different sequences and store them internally, externally on tape or on 3.5" diskettes using the Mirage Sampling Keyboard or Multi-Sampler.

If controlling other MIDI instruments is on your list of priorities, the ESQ-1 puts you in the driver's seat. It supports poly, omni and mono modes along with Ensoniq's multi and overflow modes that extend the MIDI capability of the ESQ-1 far beyond ordinary synths. You won't ever have to leave the comfort of its 61-note weighted, velocity sensitive keyboard to play any MIDI instrument in your setup.

Comparable high performance digital waveform synthesizers and MIDI sequencers can easily exceed the legal limits of your cash on hand. But the good news is that the ESQ-1 comes from Ensoniq—at a sane price of just $1395. For a glimpse of technology that's earned the name "advanced", put an ESQ-1 through its paces at your authorized Ensoniq dealer today.

Although you should always fasten your seat belt when playing the ESQ-1, you don't have to wear a helmet or obey the 55mph speed limit. ESQ-1 and Mirage are trademarks of ENSONIQ Corp.

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Managing MIDI

By Paul D. Lehrman

One of the hottest topics of debate in synthesizer and sampler circles these days is: Who owns the sounds? Once upon a time, if an electronic music composer found his sounds being "quoted" by another composer, he'd be flattered. These days, he's more likely to sue.

The standardization of the way sounds are defined, which began with the first digitally controlled synthesizers and is now universal thanks to MIDI, has led to this problem. Because a modern synthesizer's sounds are determined by discrete numerical values, making a perfect copy of someone else's patch is a cinch. With the appearance of standard formats for sample storage (which can now not only handle sounds destined for one type of machine, but even transport them between devices of different manufacturers), the same can be said for samples.

The issue is not whether a patch or a sample can be copyrighted. I'm no lawyer, but common sense says that it is, just the way a song on paper or a collection of numbers on disc can be copyrighted. The real issue is whether such a copyright is enforceable, and how that enforcement should be handled by the recording community.

First off, let's consider who are the suppliers and consumers of sounds. Far to one side, there are those specialist programmers who maintain that programming has become such an esoteric art that the few who have mastered it should benefit greatly—at the expense of those who can't get the hang of it. Also on that side are those who think that if a celebrity has come up with a sound, it is a valuable commodity, and therefore not to be distributed carelessly.

At the opposite end of the spectrum, we have the faceless hordes of synth owners, professionals and amateurs, who are slavering for the latest hot patch and aren't particular about where or how they get it.

Squarely in the center are working musicians who do much of their own programming and like to sell off or swap some of their patches, at the same time collecting others. These folks are concerned with both ends of the pipeline, as are music stores, who like to build up large collections of sounds to help them sell hardware.

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

The fact of the matter is that, ultimately, the faceless hordes will win. They will get their sounds, and the people who produce them will stop getting paid for them. The reason for this is simply that, unlike software programs on disk, there is no way to copyright sounds, or even to tag them with any information as to their origin. Considering how much computer software is pirated, even among people who consider an overdue library book to be a mortal sin, it's foolish to think that we can keep track of a sound once it leaves its owner.

There must be a way to reward good programmers and recognize the value of good patches and samples.

Many of those who make money selling sounds, of course, deny this. One theory goes that if people are forced to pay for sounds, they will consider them valuable enough that they won't give them away to their friends. But this falls down when you consider that someone who pays $50 for a patch bank might sell it to two friends for $25 each, who then sell it to their friends for $10, and pretty soon you're down to the cost of a disk. When sounds are swapped, they lose value even faster than when they're paid for in cash. And, unlike computer "shareware," where at least the author's name and address follows the thing around, once a patch gets into the public domain (i.e., the first time it changes hands for free), it stays there.

Squarely in the center are working musicians who do much of their own programming and like to sell off or swap some of their patches.

Furthermore, sounds have a finite life span. As a sound becomes more widespread, it becomes cliched and less valuable. (How many patches do you have named "Jump," and when was the last time you used any of them?)

Even if a programmer finds someone doing something illegal with his sounds, what is there to be done about it? Certainly someone who steals your RAM cartridge, makes 100 copies of it, and then hawks it on the street with your name on the label is breaking the law. But most offenses are going to be less blatant than that.

Let's say you hear one of your sounds on a new hit album. How can you be sure it's yours? Okay, maybe you manage (by court order) to get the patch parameters from the offending party. If the patch name is different, is it your patch? If three parameters have been changed, does that make it different? How about seven parameters, or 50? Where do you draw the line?

And, if there really is an infringement, who's the guilty party? The artist, the producer, the session player, or the guy from the previous session who left his disk lying around the control room—or maybe swapped some sounds with the studio's resident synth programmer?

Surely there must be a way to reward good programmers, and to recognize the value of good patches and samples. It does take artistry, skill and patience to create sounds that are both useful and original. By discouraging blatant piracy, and by giving credit where it is due (maybe on the album cover), we in the professional recording field can both maintain our integrity and keep the art of programming alive.

Granted, if someone comes up to us with a free disk of the latest super sounds from Hollywood, we're not necessarily going to turn him into the police. But if we end up using them on a multi-platinum project, we should at least try to find out where they came from and acknowledge the source.

Good programmers know that selling sounds out of the back of a magazine can be a quick way to make a few bucks. If they're in it for the long term, however, session credits and the career continuity they provide are much more valuable. R·E/P

Sound on the Road

By David Scheirman

The concert sound business has undergone many changes within the past decade. The growing field of live sound to serve the entertainment industry and public events has helped cottage industries grow into international corporations; it has enabled certain fledgling manufacturing firms to experience rapid growth and turn small private partnerships into publicly held companies. And this unique blend of art and science that comprises live sound is now helping to develop its own new educational programs and job opportunities.

Any discussion of "live sound" usually contains a certain subjective element, for we all experience sound in different ways. There are certain parameters, however, that help to interpret or define what most people (whether performer, technician or audience member) call "good" live sound.

Good sound is typically a unique synthesis of many different things that come together for a performance: creative musicianship, technical excellence, proper choice and use of hardware, and advantageous acoustics.

Individuals and service companies that learn to properly combine these various aspects of live sound into a consistent product that pleases performers and audience members alike usually will find a ready market for their skills.

In future issues, this column will focus on those live sound mixers, concert sound companies and professional sound equipment manufacturers that are helping to advance live sound's state of the art. The concert sound community is a diverse one, and those individuals and companies that serve it do so for a variety of reasons. While seeking to improve live sound hardware and operating techniques may be a noble ambition, a sense of economic realism is also a fundamental characteristic of the concert-sound company able to survive the uncertain trends that have helped to shape the live sound industry during the last few years.

As rock concert touring evolved throughout the Sixties, early suppliers of live-sound systems were often commercial rental companies located in major cities, including companies that stocked vocal reinforcement systems for use at fairs and other public events. The early touring rock acts helped define a need for more powerful sound systems capable of providing full-bandwidth musical reinforcement.

Existing sound equipment manufacturers responded to this new market demand in different ways. Some companies felt that it was a passing fad, and made little effort to carry out research and development in what was then a new field. Some of today's major touring sound companies grew to take a lion's share of their own industry by being the first to realize potential solutions to the challenges presented by the touring musical groups.

Other equipment firms realized that the new market demands being placed on sound equipment, and the novel uses of some of their products by early touring sound companies, were helping to create a new industry altogether. Not quite sound recording or radio broadcasting or commercial PA, this new industry required products and operating techniques that did not exist at the time. Many parts of today's standard concert sound systems were created from scratch, often borrowing from the technology and parts industries. Contemporary multiple-mix stage monitoring systems, portable electrical power distribution set-ups and dedicated, portable mainframe mixing consoles were not in evidence 20 years ago.

Today's pro-sound equipment industry has come into being through the help of the touring sound business. As the hardware being used becomes more complex, many live sound service companies have chosen to focus their energies on concert sound itself, and have left the development and manufacturing of new products to established equipment companies.

Changes affecting touring sound companies will also affect the audio equipment firms that serve this industry, along with those persons who have chosen live sound as a career.

Those individuals and companies that are providing consistently good live sound, helping to improve their industry, and operating at a profit, have some valuable and interesting stories to share. "Sound On The Road" exists for this reason. R·E/P

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


PRO Verb
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Film Sound Today

By Larry Blake

In recent years, the process of using computer-based systems to edit sound for films and video has come to the forefront of sound engineering technology. One obvious fault of almost all of the systems that this writer has seen is the lack of a sensible database organization. Discussions with manufacturers reveal that this is something they will be taking a hard look at...eventually.

This omission is somewhat surprising. because sounds will be all but inaccessible without a well thought out user interface. In the old days (i.e., today), one could shuffle through whatever cross-referencing and organization the sound librarians have done on paper. If that failed, then the editor could simply thread up ¼-inch tapes or 35mm library masters until the right sound was found.

Even when a library is stored on conventional magnetic media, such as ¼-inch tape, a computer database is still a great help not only in finding sounds quickly, but also in organizing the massive amount of paperwork generated in film sound post-production. (I know of only two serious database programs that answer this need. Are there any others for sale out there?)

Today, usually the only person who comes in contact with the computer in film sound is the supervising sound editor, who will "spot" the show and create reel-by-reel lists of what effects the sound editors should cut in. Similar paperwork can be printed out to assist the transfer person.

The point to be made here is that the work is still the same: Transfers are made from ¼-inch to 35mm mag, and they are cut on Moviolas and film synchronizers. However, random-access digital editing will change this situation. The transfer stage as we know it today will be virtually eliminated, and will become more of a mastering process to transfer the unedited field tapes to whatever storage medium the editing system uses. Most importantly, sound editors will use the computer for the act of editing itself, and will have to be as comfortable with the system's database structure as they are today with mag film and Moviolas.

Thus there is the need for companies manufacturing and proposing digital editing systems to agree upon some standards concerning database organization. This standardization would, I hope, allow editors to use different systems as easily as MIDI allows musicians to link synthesizers. (Other examples of standardization include the Association of American Publishers manuscript project, designed to standardize typesetting codes, and the Library of Congress' recent guidelines for optical storage.)

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

Standardization would take place in two areas: the descriptions used in cross referencing sound effects, and the database field structure used to define what the sound is, where it came from and how it will be used. (The field structure has to take into account the dialogue and music elements that will be stored and edited.)

One obvious fault of computer-based editing systems is the lack of sensible database organization.

The first issue is a tricky one because no two people describe the same sound in the same way. To one it's a "car," to the other it's an "auto." Driving in the vehicle is described as a "steady" to one person, and a "constant" to another. Having worked in a sound library, I have no illusions that a set of cross references could be created that would please everyone. People will describe effects by whatever words they choose.

However, because there are so few digital editing systems in the field today, there is still time for the industry to create a *recommended* standard for cross-referencing descriptions. Probably the most sensible approach would be to recommend one term, for example "auto," while listing synonyms such as "automobile" and "car."

If the various digital sound editing systems also employ a similar database structure, then editors will not have to be "multilingual" in order to move back and forth between competing systems. In contrast to sound effects cross referencing, the method of defining the format of the recording at hand (mono, stereo; number of tracks; analog, digital), its function (original recording, cut element, pre-mix, printing master), the type of recording (production track, multitrack music master, effects recording), source (prefixes for both commercial and well-known private sound effects libraries), date and place of recording, etc., is a straightforward matter that *shouldn't* be too hard to agree upon.

Regardless of whether the guidelines are followed completely, they will give a head start to programmers writing the user interface, and to users starting the enormous task of entering a sound database into a computer.

Because the recommended standards would be defined by a group of experienced professionals, both the lexicon and database field layout should be close to what the sound editor would design, given the chance. All too frequently the software for such systems is written by computer people, whose knowledge of what is required comes from a talk or two with a working professional. We will be better off if we can head off these well-meaning computer experts at the pass.

Despite the grand wishes of the various manufacturers, no one system will automatically become the standard. This is one race that is going to be tough to call, and there will undoubtedly be many casualties. It should be clear that the standards would not in any way impede competition between different systems.

From this writer's point of view, the two biggest challenges faced by designers of digital editors are how to store the sound library (including dialogue and music); and how to get the edited sound elements to the re-recording stage. It is both a blessing and a curse that the choice of mass-storage options is so wide: hard disks, optical discs (CD, laserdisc, WORM, etc.), R-DAT, analog and digital multitracks, etc.

In any event, these challenges will be compounded by the variety of storage media and it will be a long time before one can edit on one system and mix on another on a hardware level.

However, a database is much more malleable (software, right?) and everybody—manufacturers and the public—will benefit if the user interface truly bridges people and machines in a sensible manner. The battle for the survival of the fittest will be fought on the field of hardware, I believe, and there's no reason not to make certain aspects of the software a DMZ.

We must seize this opportunity now, before everything gets to be locked down and *de facto* standards are arrived at by default. I would like to open up a dialogue between manufacturers and users. Anyone wishing to participate is invited to write to me directly at P.O. Box 288, Hollywood, CA 90078. R-E/P

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Living with Technology

By Stephen St. Croix

The way I see it (and I see it a lot, because I open up almost everything that comes onto the pro-audio market), most of the big boys in the digital audio game have come to the same conclusions about the best way to get in and out of their RAM. The same 16-bit DACs and low-pass filters show up in most digital toys we buy today—literally interchangeable parts in competing products!

The way I hear it (and I hear the stuff 12 hours a day, 4 days a week), they *don't* sound the same at all. I'm not talking about features, bandwidth or even software. I'm talking about the fact that these devices sound extremely different from each other, even when used as simple delay lines.

As 16-bit units with 20kHz bandwidth become more popular, the design of the sample-and-hold circuitry becomes more critical. Such things as settling time, buffer impedances and capacitor characteristics greatly influence the final sound. This has been taken seriously, and today's circuits show it.

Yes, digital technology actually works now. We are all impressed. But now that we are recovering from the shock of all this, we should be aware of the sonic shortcomings that are *not* part of the conversion process, but simply of the unit's analog sections. Don't forget that within every digital device there lurks two important areas of analog circuitry: input and output.

The analog IC designers have not been on vacation while the digital chip guys have been doubling our power each year. There are now better, cleaner, faster and higher bandwidth op-amps, some of which are amazingly close to theoretical optimum. Wonderful. Just like a car—more speed and more danger of losing control.

While it was impossible to get high performance out of the old 741, almost no one could design a really bad circuit with it; a circuit that had hidden quirks. With today's newest high-speed, high-impedance, high-bandwidth op-amps, however, extremely impressive designs are possible. On the other hand, almost every designer can now create a really bad circuit the first try.

A race is on to use the newest high-performance analog chips. It's often impossible for circuit designers to learn all the idiosyncrasies of each amp within the real working environment, and just about the worst environment that mankind can provide is inside a chassis where high-speed digital circuitry is slicing up the dc rails and broadcasting RF directly to every analog device within.

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

Ultra-high-speed op-amps love to become unstable. High speed means shorter wavelengths which means that you can't put a bypass capacitor two inches away and expect it to work like it did in 1979. It won't. Today, the cap has to be almost touching the chip, with the leads cut short; and then the cap itself may be too slow.

The new hyper-fast op-amps can definitely sound good, but only in the hands of well-versed design engineers.

The type of capacitors and resistor compounds, and the length, thickness, location and termination impedance of the traces, can all influence the sound of a circuit. In fact, there are so many soft variables involved in analog design that it qualifies as an *art*. Let's hope it won't go down in history as a lost art.

Digital hardware design is logical. It is sort of like building with Lego blocks; it's...well...sort of...digital. If you follow the rules in the spec book, hook up the stuff in a semi-logical fashion, watch timing and keep your rails clean, all the little ones and zeros will go just where you want them to. If they don't, the design is wrong, and you have to find the mistake and fix it. No real mystery.

Furthermore, there may be a dozen completely different ways to do a given thing digitally, like print the word "reverb" on your display. Any one of these approaches will work, and look exactly the same. The only differences would be in how much PCB real estate, power, CPU time or money it actually took to get that word on screen. In fact, of the 12 possible designs, only one would actually be right or truly optimum. The others would be wrong, but might make no difference to you as the user, only adding a bit to the unit's parts cost or size. The visual and audio results are the same.

Not so with analog. If the engineers haven't figured out the optimum circuit, you don't have the optimum sound.

The same digital chip made by two different manufacturers may have different propagation delays (speeds). This makes no difference as long as they fall within the timing window. If the same analog chip made by two different manufacturers have different speeds, however, they may sound *very* different in a high-performance audio circuit.

Different brands of any given op-amp may also have differing noise floors, power-supply noise rejection figures, and so on. Many of these specs change as the parts become warm, and each brand changes in a different way.

The new hyper-fast op-amps can definitely sound good, but only in the hands of well-versed design engineers. Console manufacturers are taking all this quite seriously, and are spending a lot of time learning how to get the most out of these devices without losing control (short duration instabilities, audio modulated by dc supply hash, rectification of RF to noise, and more).

Digital toys do not always demonstrate an understanding of this. I have seen high-frequency oscillations present in the analog circuitry within at least three of today's devices.

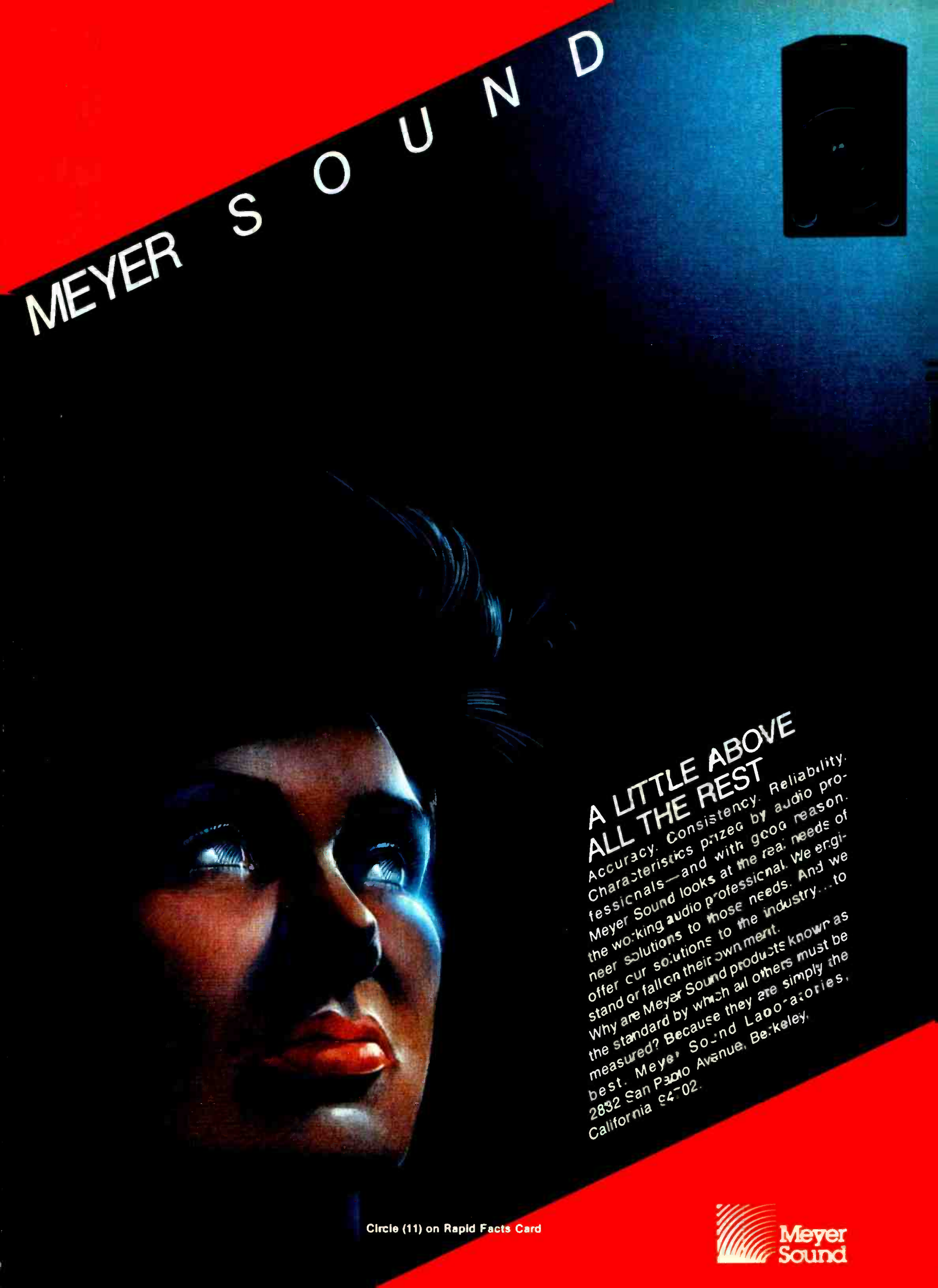
What does all this really mean to you, the user? Don't assume that the slightly edgy or muddy sound of your new 16-bit processor is inherent to digital technology. It may simply be instabilities in the audio path.

If you doubt that the answer to this problem can be this simple, listen to several CD players from various manufacturers using identical DACs and sampling schemes. Any audio differences will then, of course, be due largely to the design of the few analog components inside each machine.

A few years ago a friend said an interesting thing to me: "As digital resolution becomes higher and higher, it approaches the ultimate goal—analog." I wish I had said that.

I feel that 16-bit digitization is borderline, at best. Eighteen-bit is just around the corner, and there are several companies working feverishly on 20-bit DACs. I know of one significant company currently designing a very comprehensive 36-bit system.

Let's hope the analog circuit designs are ready when it happens. R·E/P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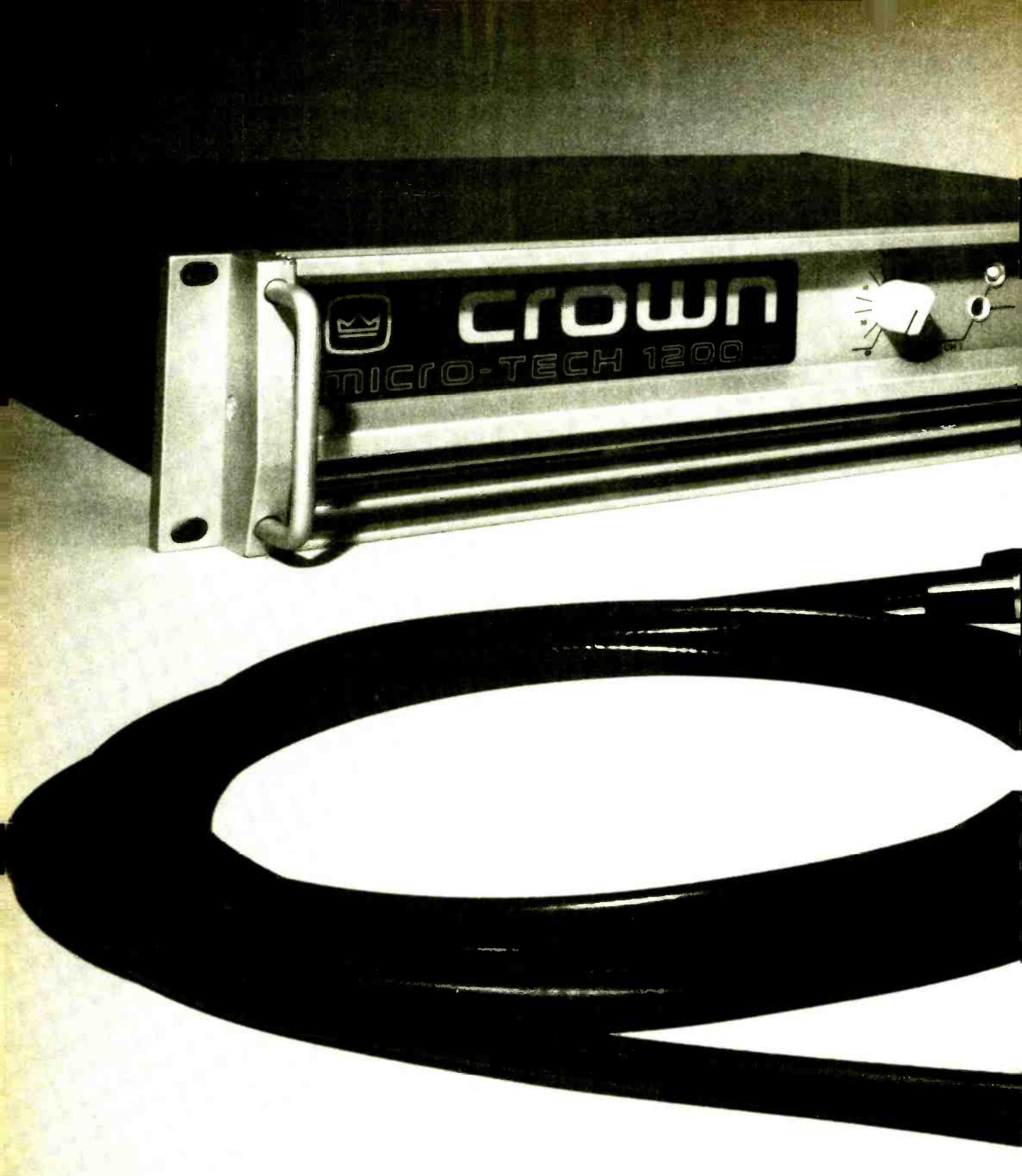

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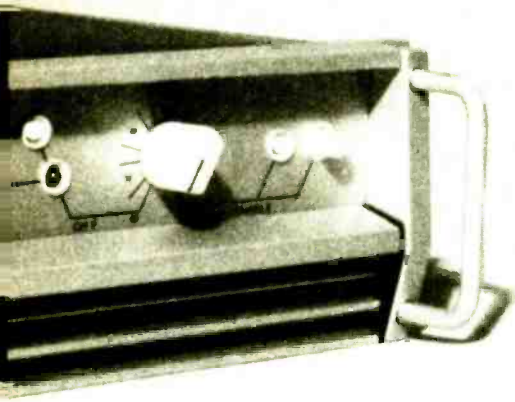

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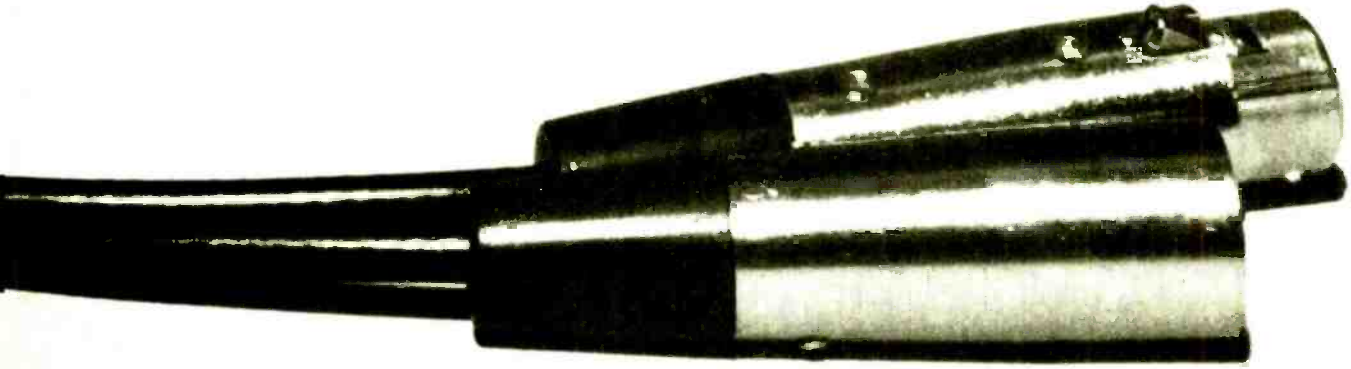

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SPARS On-Line

By Gary Helmers

Although the state of the U.S. economy is not a usual topic of conversation among studio owners and operators, it formed a dominant focus at the recent SPARS business conference in New York. Our economy is a crucial facet of the type of realities that studio management must analyze and comprehend in the ongoing quest for profitable operation. Such factors as changes in the tax laws, problems with the low value of the dollar against foreign currencies and sluggish advertising revenue at the television networks, have become part of our daily business lives.

At the SPARS conference, the economic and technical future of studio operations was analyzed by a panel of industry authorities. Our moderator was industry forecaster and SPARS consultant Martin Polon, of Polon Research International, who provided invaluable insights and illuminated all of us to the hard facts of 1987.

We focused on several new directions that will affect the bottom line for studios. What follows are some projections for the future:

Many feel that the "mothership" scenario will continue to develop, with some studios diversifying and acting as central switching points, while others will evolve as satellites with more specialized services. This direction will continue through the rest of the decade, but we foresee dramatic changes in the modes of profitability.

In the past, studios could achieve technological parity or dominance with the competition by "writing off" large equipment purchases against before-tax profits, thus shielding significant amounts of after-tax income. The old tax law allowed—virtually mandated—"negative profitability" via capital equipment purchases. The new tax code will not allow this. Loss of any kind is frowned upon as a deduction.

This does not mean that new equipment purchases under the new tax law will cease to be attractive for studio owners and operators. It is almost as if the federal government has codified the old saying, "It takes money to make money." If a studio owner needs equipment to enhance a line of business (*a la* the mothership scenario) and makes that purchase after Jan. 1, 1987, there will be far less of the old incentives.

The investment tax credit will be gone, accelerated depreciation will be replaced with a longer time frame, and some of the tax-saving schemes for financing the purchase will no longer be acceptable.

While at first glance the future appears darker, the profit engendered by the new purchases will be taxed at the rate of 34%, which is the new corporate ceiling. The investment will produce a greater after-tax profit. Under the old law, that profit would end up being taxed at nearly 50%.

The new law says, in effect: "It's OK to make a profit but not OK to make a loss.

By way of an example, a new post-production room coming on line in 1987, or later, that yields $1 million in gross proftablility will produce $660,000 in after-tax profits. That is for U.S. taxes, of course. Under the old law, there would have been only $500,000 available after taxes.

The new law says, in effect: "It's OK to make a profit, but not OK to make a loss."

Proper perception of the new law is the key to profitable survival. If the studios see this change as replacing incentives to purchase equipment with incentives to enjoy a greater profit, then we come out about even. But it may take some time for this understanding to reach the street.

The issue of foreign exchange has entered the relationship among studios, foreign equipment manufacturers and sales subsidiaries/dealers. Stabilization of this relationship is a function of the unmeasurable factor of perception.

It is important to remember that the dollar has dropped in 18 months from 265 yen to the dollar to about 150 yen. Although it has bounced back to about 160 yen at the time of writing (early December), it is still a change of about 57%. Yet the price adjustments for foreign studio equipment have not even begun to approach those levels. The same can be said for studio equipment from England, West Germany and Switzerland, where the dollar has also sagged in value.

Japanese manufacturers have been the most aggressive in absorbing some of the differential to retain market share. Their perception is that they can afford on an item-by-item basis to adjust the prices of their studio-equipment lines. This adjustment can be a very complex matter indeed, involving U.S. subsidiaries that do nothing but manage the flow of monies for domestic manufacturing facilities in the U.S. distribution and service facilities in the United States, and so on.

In effect, studios have been getting a break—and it should continue. It appears that a "discrete" agreement on United States/Japanese economic issues has been worked out, with dollar stabilization in the 160 to 165 yen range. We will keep our fingers crossed on that one, however. It would appear that 1987 will be a year with positive indications for the economics of studio operations, but also a year in which more attention will have to be given to economic factors than ever before.

The regular series of SPARS studio business conferences serves as a forum for examining business and economic issues that affect the day-to-day operation of a successful studio. Recent conference topics have included insurance, leasing, banking, employee benefits and public relations. The next business conference will cover using the business plan as a tool to evaluate the components of a successful studio operation. Various guest speakers will address new venture plans, continuing operation plans and plans for expansion.

As discussed in the December "SPARS On-line" column, a carefully constructed business plan is the first step toward a successful studio. For a step-by-step examination of several business plans, successful and not so successful, you might consider attending the next studio business conference, which will be held at UCLA on March 28 and 29. For more details, contact the SPARS national office, Box 11333, Beverly Hills, CA 90213; 818-999-0566. R·E/P

Gary Helmers is the executive director of SPARS.

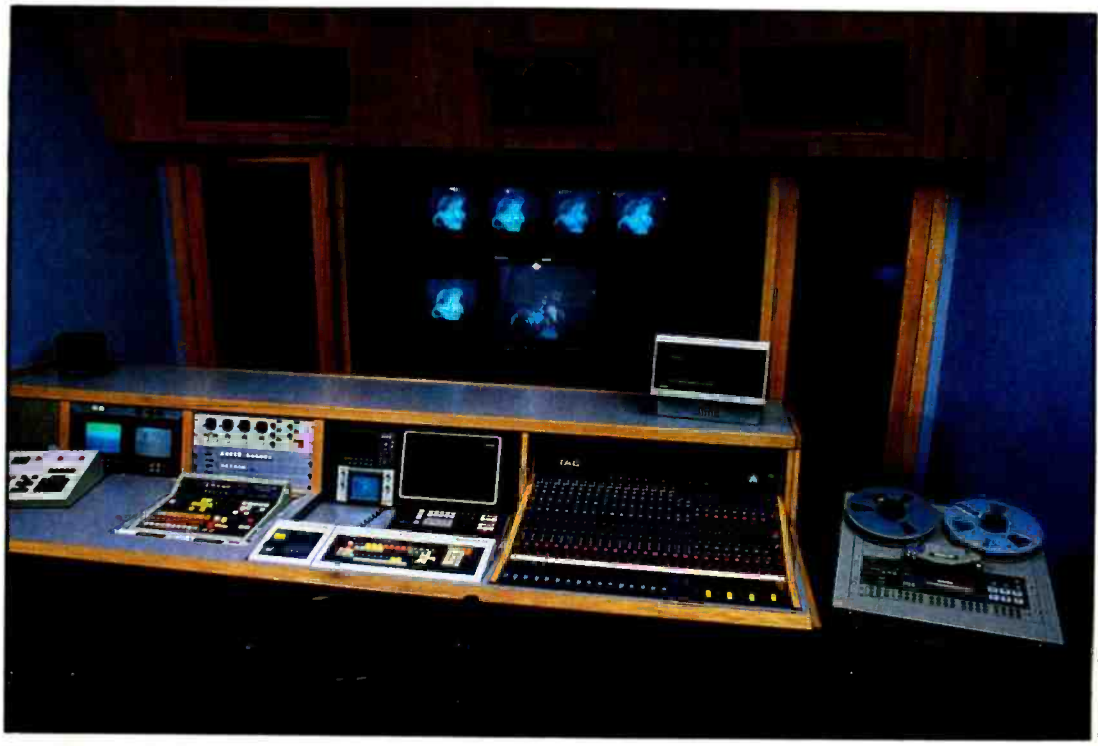
Photo by Elizabeth Annas/Photosensations

Visual Eyes audio-for video post-production room.

Narrow-Gauge Multitrack Applications

By Adrian Zarin

The use of cost-effective narrow-gauge multitracks has matured from using ¼-, ½- and 1-inch semi-pro applications to a viable alternative to conventional track formats.

Historically, narrow-gauge multitracks have long been considered the province of hobbyists and musicians with personal-use studios in their basements or garages. Today, however, such technology has matured to a stage where it is playing an increasingly vital role in many full-scale professional applications.

In the area of electronic music production—whether for the visual media or album release—sophisticated MIDI synchronizers have taken over much of the tracking functions once handled by tape machines. Consequently, 1/2- or 1/4-inch 8-tracks and 1/2- or 1-inch 16-tracks can provide a cost-effective means of recording the few non-electronic sound sources (such as vocals or acoustic instruments) that may be integrated into a synthesizer-based production.

And in the post-production environment, many facilities have found that a narrow-gauge multitrack represents an ideal format for assembling music, dialogue and effects sources in a way that satisfies both their own operating budgets and the new "audio consciousness" in the visual media.

Narrow-gauge formats first emerged during the Seventies, in the form of 1/4-inch 4-tracks; the doubling and quadrupling of track formats brought about the advent of 1/2-inch 8-track and 1-inch 16-track. Concurrently with the increasing number of tracks being of-

Adrian Zarin is a Los Angeles-based electronic synthesis, composer and free lance writer, and a regular contributor to RE/P.

fered by narrow-gauge transports, the marketplace has also witnessed a progressive reduction in scale, giving rise to 1/4-inch 8-track, 1/2-inch 16-track and, perhaps even more dramatically, multitrack cassette formats.

The most formidable, and obvious, design problem confronting developers of narrow-gauge multitracks is the fact that a smaller tape track produces reduced frequency response, lower signal-to-noise ratios and greater crosstalk. The twin liabilities of higher noise and limited frequency response mean that some form of noise reduction is practically essential, along with higher tape speed. Consequently, the optimal use of narrow-gauge technology begins with selecting a design that best suits the particular applications at hand.

A good way of coming to terms with the potential of narrow-gauge techniques is to examine the operational procedures of commercial facilities that use such technology on a day-to-day basis. Accordingly, the remainder of this article will consider three different types of narrow-gauge facilities: a combination audio post-production and video editing room; a 16-track electronic scoring studio; and an 8-track electronic scoring studio operating in tandem with a voice-over post-production facility.

Narrow-gauge video post-production

Alan Kozlowski of Visual Eyes Productions, Santa Monica, CA, describes his studio as a crossover room. "It's a video editing suite, but the audio facilities were designed for full-scale music projects."

This broad-based design is reflected in Visual Eyes' varied client roster. Recently, the facility provided audio post and sweetening for Lionel Ritchie's HBO Special, *The Making of Dancing on the Ceiling*, and it regularly performs the same services for broadcast television's NFL Sports Trivia Show. The studio has also handled audio post-production on several Richard Simmons home exercise video tapes, industrials for La Costa skin care products and a number of TV commercials, including political ads for Los Angeles mayor Tom Bradley.

A Grass Valley model 100 component video switcher and model 41 edit controlling system coordinate the room's complement of VCRs and VTRs, which include Sony BVW-10, BVW-15 and BVW-40 Betacam machines, a Sony BVU-800 U-matic and a BVH-2000 C-format 1-inch mastering machine.

The video editor also controls the studio's miltitrack machine, a Tascam MS-16 1-inch 16-track, which is interfaced with the editing system via a Calloway Electronics synchronizer.

Photo by Elizabeth Annas/Photosensations

The three co-owners of Visual Eyes Productions comprise (left-to-right): Sandra Hay, director, Alan Kozlowski, technical director and Doug Rosen, financial director.

"To the editor's computers, the MS-16 looks like just another videotape machine," Kozlowski says. "It locks up in a 5s pre-roll with all the other machines. We can roll it along with the video machines and record the audio tracks directly onto the 16-track while we're recording the video on the 1-inch mastering machine. In other words, we can build all the audio tracks while we're doing the video editing, sweeten and mix them, and then lay down the final mix back onto the video master."

Acoustic design was handled by Brett Thoeny of BoTo Design, and includes Westlake monitors, Crown power amps and an iso booth for voice-overs and single instrument overdubs. The audio console is a 16-in/8-bus TAC Scorpion. According to Kozlowski, a lot of thought went into finding the right cost-effective console to interface with the facility's 1-inch 16-track.

"We definitely needed 16-track monitoring. But, because we're never really laying down more than a couple of tracks at a time, we didn't really need more than eight output buses. We can just repatch and reassign to different tape tracks. For mixdown, a 16-input board is more than adequate; it provides all the effects returns and everything else we need."

The need to interface the 16-track with Visual Eyes' video editing operations imposed yet another set of console requirements.

"For video editing," the engineer continues, "you need to be able to switch audio tracks between the preview switcher and various other functions. The TAC board really worked out well, because it has direct outputs on every input module, which we send to our audio-follow-video switcher.

"Instead of going directly out of the stereo bus, we have the outputs of different machines come into the console's input modules and then straight through those direct outs to the switches. These signals come back into the board through the echo returns, are routed to the stereo output bus and then sent to the mastering machine.

"As a result, we have some flexibility depending on the mode we're operating in—we can send the source machine directly into the stereo bus, or route them through the audio-follow switches."

Visual Eyes is also equipped with a Sony PCM-F1 processor, which encodes digital signals onto the facility's BVU-800 U-matic.

"A lot of times, we record in the field with the F1," Kozlowski says, "and transfer those digitally recorded tracks onto the MS-16. On a lot of projects, we're also working with MIDI scores from music composers. These too are usually recorded directly from the synthesizers onto videotape via an F1 at the scoring studio.

"We receive an F1 music master with time code, which we can transfer onto the 16-track along with the dialogue and effects. We build the whole show and then go back and sort out just the audio edits. We have a program that will take an entire edit decision list and pull out just the audio edits, which we then rebuild directly onto the 16-track from the F1 tapes (in a series of transfers). We add music, sound effects and other audio sources, then do the final mix to picture."

While transferring audio sources (whether digital or analog) to the MS-16, Kozlowski tends to be sparing in his use of noise reduction.

"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."

As an alternative, he has developed a technique for improving signal-to-noise ratio on the multitrack without recourse to noise reduction.

"On shows where I have plenty of tracks to spare, I'll turn off the noise reduction and just record the same signal on multiple tracks. This way, you double your track information, and pick up an extra 3dB signal-to-noise ratio. It doesn't sound quite as 'mushy.' On high-quality music projects, I can get a little cleaner front-end transients with this technique."

As the above comments indicate, the

Photo by Elizabeth Annas/Photosensations

A Tascam MS-16 1-inch 16-track is available for preassembly of music, effects and dialogue tracks, prior to remix to picture.

Photo by Elizabeth Annas/Photosensations

An outboard rack located behind the video and audio mixing position houses a collection of Sony BVW-10, -15 and -40 Betacam videocassette decks, plus a BVU-800 ¾-inch U-matic VCR. Audio processors (left) include a dbx series 900 rack and a Lexicon PCM-60. To the right is a Sony PCM-F1 digital audio processor for field recording and replay of digitally encoded music masters.

schemes: "On *Lies*, for example, I went in and looked at the picture on a flat bed (editing table) and used click loops on the mag film to put against the picture. So there I was dealing with the reality of sprockets against sprockets.

"To maintain sync once I was in the studio, I used a UREI digital metronome and a 60Hz sync tone for playback purposes, which worked really well. The film has some long, 5-minute chase cues in it, and things were syncing up very nicely."

Apart from the occasional flute, sax, guitar or bass part, (all of which are usually played by Donahue) most of the sounds on the composer's scores are generated by his collection of synthesizers. An Oberheim Matrix 12 acts as master MIDI keyboard controller, driving an E-mu Systems Emulator II, PPG Wave 2.2, Roland MKS-30 module, Yamaha DX-7 and a TX-7 rack. A LinnDrum modified to output 24 ppq sync code triggers the arpeggiator of a Roland Jupiter 8 [a non-MIDI synth]. In contrast to what is rapidly becoming standard practice, the conservatory-trained composer and musician prefers not to use a sequencer.

"I really use MIDI quite basically," he says, "I can obtain sound textures by combining the output from several synths, which means I'm doing everything in real time onto tape. I'm often using all my synths for each pass, rather than using one or two synths for each part and having a sequencer play it all."

Because Donahue's work involves a large number of simultaneous audio outputs from his various synthesizers, he needs a console with substantial input capabilities. After some careful research, he settled on a 20-input/8-bus Panasonic Ramsa WRT820 console. Apart from plenty of inputs, Donahue also needed a board that would suit his one-man style of working.

"The Ramsa is good for this," he says, "because you can leave your input attenuation where it is and have a second trim control for tape playback levels. You don't have to upset your input levels just to hear a quick playback of a multitrack take."

Donahue has evolved a more-or-less standard configuration of input channel assignments for his electronic equipment. Input channels 1 through 4 are devoted to the LinnDrum: separate outputs for kick and snare, plus a composite stereo output for all the other drum voices. Channels 5 and 6 accommodate the Wave 2.2's stereo outputs; channels 7 and 8 the Jupiter 8's stereo outs; channels 9 and 10 the DX-7 and TX-7; channels 11 and 12 the MKS'30's stereo unit; and channels 13 and 14 the Matrix 12's stereo outs. Channel 15 is left open, channel 16 handles the EII, while channels 17 through 20 are used as effects returns for Donahue's Roland SRV-2000 and Yamaha REV7 digital reverbs.

For each part that goes down to tape, the relevant synthesizer outputs (as triggered by the Matrix 12's keyboard) are mixed to stereo and printed on two tracks of the B-16. But even though he uses two tracks for each part, Donahue generally finds he has enough tracks to record his entire score without having to bounce down tracks. The B-16's built-in Dolby C noise reduction would make track bouncing a problem anyway, he observes.

"Dolby C doesn't really like to be used more than once. If you double it (by bouncing two Dolby-encoded tracks onto a single track), you can get breathing problems. It's most noticeable on rhythmic parts, such a hi-hat or other percussion."

Donahue's principal mixdown machine is a Fostex model 20 2-track with center-track time code.

Not all of Donahue's projects call for stereo mixdown, but when they do he likes to provide two different 2-channel mixes. One is standard left-right stereo mix, whereas the other has all of the melodic parts on one channel and all of the percussive elements on the other.

"When they do the final mix," he explains, " the main musical elements they might want to bring down are those that get in the way of dialogue. Because, more often than not that means percussive sounds, I provide those on a separate channel. This way, if anything

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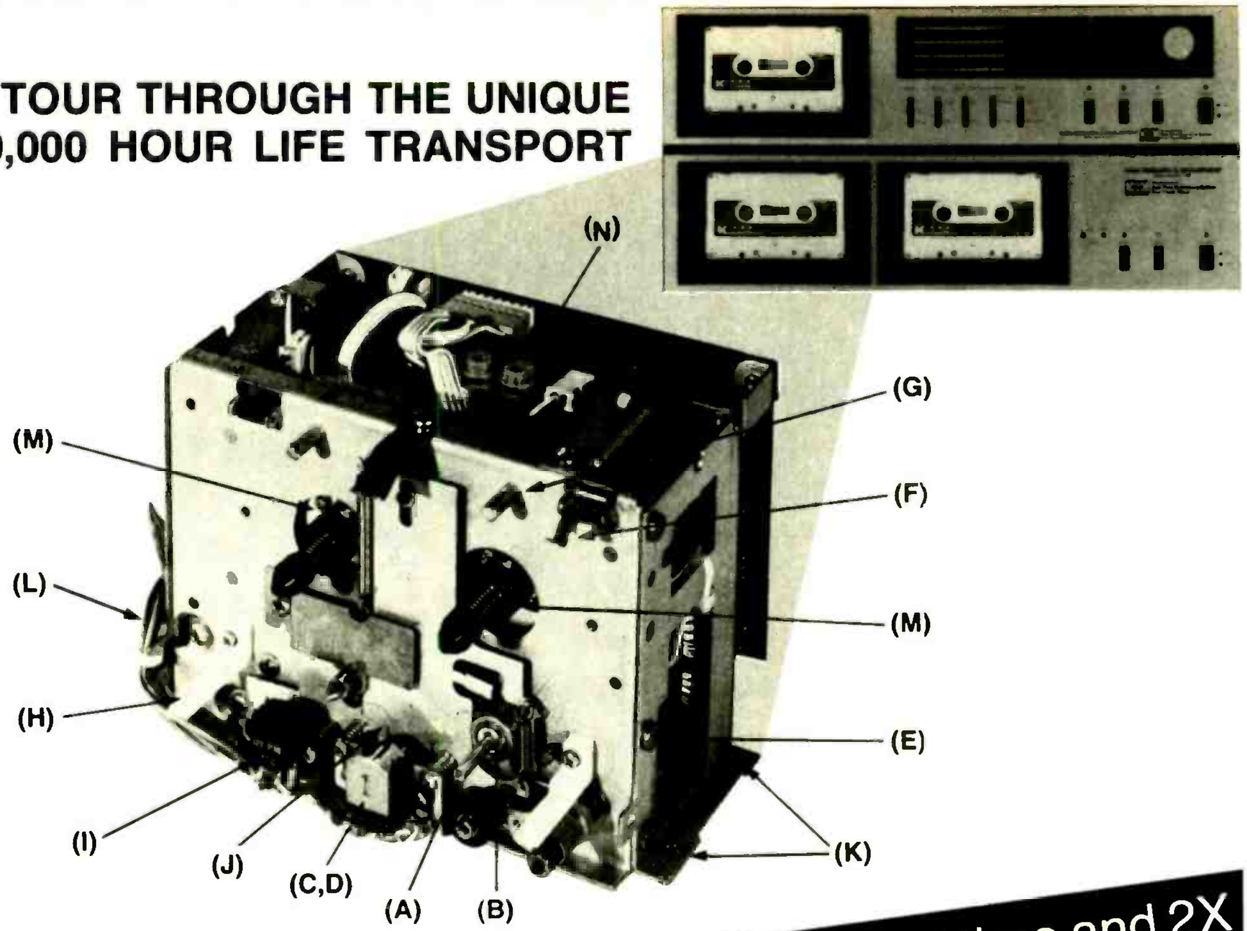

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Circle (15) on Rapid Facts Card

Donahue's production equipment includes (from left-to-right): A PPG Wave 2.2, Oberheim Matrix 12, LinnDrum, Yamaha DX-7 TX-7 rack, Roland Jupiter 8 and an MKS-20 synth module; 20-input Panasonic RAMSA WRT820 console; E-mu Systems Emulator II sampling synthesizer; and a Fostex B-16 ½-inch 16-track with built-in Dolby C-type noise reduction.

Composer/synthesist Mark Donahue in his personal-use, 16-track narrow-gauge facility.

Jeffrey Hedquist, of Hedquist Productions (left) and Paul Fauerso of Amber Waves Music Productions. Studio hardware includes a Soundcraft series 400B console, Calibration Instruments MDM TA2 monitors, an Otari MX-5050 MkIII ½-inch 8-track, Yamaha KX-88, DX-7 and DX-21 synths, Roland Jupiter 8 with MIDI interface, Yamaha SPX-90 and Roland SRV-2000 digital effects. Music sequences are controlled from an Apple Mac+ running Southworth Total Music MIDI software.

comes down, it's likely to be the drums, rather than something more critical, such as the level of the strings in relation to the brass."

Amber Waves

Located in Fairfield, IA, Amber Waves is a music production company that services local clients as well as clients from the East and West Coasts. Proprietor Paul Fauerso composes music for a variety of outlets—including television specials for the Broadcast Group in Washington, DC, as well as TV and radio jingles for a number of Connecticut agencies such as Decker, Guertin, Cheyne and Clement, McCabe. He recently completed scoring for a film project, *Knights In White Satin.*

Much of Fauerso's ad work is done in conjunction with another producer, Jeffrey Hedquist, who has a 4-track Otari and Nedtek-equipped voice-over studio in Fairfield. Fauerso's own studio is based on an Otari MX-5050 MK III 1/2-inch 8-track and an MX-5050B 2-track mastering machine, linked to a 16-inch/4-bus Soundcraft series 400B console.

The composer has a wide range of synthesizers at his disposal, including a Roland Jupiter 8, Yamaha KX-88 master MIDI keyboard with DX-7, DX-21 and TX-7 synths, a Roland JX10, Prophet 2000 and a LinnDrum. Apart from adding MIDI capability, the LinnDrum has been further modified by the installation of several DrumWar voice chips.

Outboard gear includes a Yamaha SPX-90, Roland SRV-2000, DeltaLab Effectron and Lexicon PCM-60. The studio recenty acquired an Apple Macintosh Plus with a 20Mbyte hard disk on which Fauerso runs Southworth Total Music sequencing software.

For the *Knights In White Satin* film score, which was Fauerso's first project with the Mac, the MIDI sequencer's "tracks" took over the role usually performed by the 8-track. Outputs from the various synthesizers, when triggered by the Total Music sequencer, were sent through the console and mixed directly down to the MX-5050 2-track. On future projects, however, Fauerso anticipates synchronizing MIDI tracks with acoustic instrument parts recorded onto the 8-track.

"Now that I'm starting to move into more sophisticated MIDI sequencing applications," he says, "I'm starting to feel the need for more channels on my console. What I plan to do in the future is to get a larger board, or else a second board to supplement the one I have.

"I'd like to be able to sync the computer with the multitrack and have enough inputs on the board so that I could do separate EQ and processing on all the drum and synth parts.

"Although I'd like to have another 16 console inputs, I don't really need more than eight tape tracks. More and more I'm going to be running parts live during mixdown, using Total Music and MIDI controls."

Like many synthesists, Fauerso's introduction to this kind of "virtual tracking" came in the form of retriggering LinnDrum sequences during mixdown, using a prerecorded sync code track on tape. It's a technique the composer has long been using during jingle work, in order to minimize generation loss and other tape-based signal degradation.

"I usually print the Linn sync tone on track 8 of the MX-5050. I'll always print the drum part itself on another track—just to use a reference while I'm overdubbing. It's a lot more covenient than having to start from the top on every pass, as you have to do when you're syncing the LinnDrum to tape."

Toward the end of the project, this reference drum track is usually replaced with a permanent part. To avoid adjacent track bleed either from or onto the Linn sync track, Fauerso tries to leave track 7 blank whenever possible. Unlike the 16-track narrow-gauge studios discussed previously tape tracks are usually at a premium at Amber Waves, and often track 7 must be used.

"When that happens, I try to put something on track 7 that's not going to be at a very high level in the final mix. Or maybe something like a high-pitched bell part. This way, I can EQ out the bottom and lower midrange, which is where that sync tone tends to dwell.

"Also, if you put a rhythmic part on track 7 that is very much in contrast to the beat of the Linn, you shouldn't record it at too high a level. Because if you do, it can disturb the sync tone track and all of a sudden your drum part will be off."

Judicious track management enters into all aspects of work at a narrow-gauge 8-track facility such as Amber Waves. Fauerso carefully plans where and when each track will be recorded.

"I try not to put any important parts on outside tracks 1 or 8, and I leave myself room to bounce tracks down. For example, if I'm going to double or triple a vocal, I'll put those vocal tracks on before all my tracks get filled up. I might record a drum program, a bass part and one keyboard part, and then bounce the tracks that need to be combined. Then I'll add strings, horns or any other sweetening I need.

"I'm not using noise reduction, so I try to run everything at as hot a level as I can, in order to achieve the best signal-to-noise ratio. But, at the same time, I listen very carefully to make sure I'm not losing any high-end because of oversaturating the tape. For the most part, noise isn't really a problem because there's usually fairly active program material on most of the jingles—there's lots of signal on the tape. When I mix down, I just leader the tape right up to where the cue starts. This way, the client doesn't sit around listening to hiss before the cue starts."

Depending on the requirements of a particular project, Fauerso will make either a mono or stereo mix of his tracks. In many cases, this is then brought over to Jeffrey Hedquist's facility, where voice-overs and other audio elements can be added.

In facilities such as Amber Waves, sophisticated MIDI sequencers have usurped some of the cost-effective multitracking functions traditionally supplied by narrow-gauge tape machines. But as long as the human voice, acoustic instruments and natural sounds continue to figure in audio production for all media, narrow-gauge equipment seems to have an important role to perform.

R·E/P

Owner Jeff Siegel working at Musitech, a particularly well-equipped narrow-gauge, MIDI-capable facility.

The Evolution of MIDI-Based Narrow-Gauge Studios

By Rob Tuffly

Jeff Siegel's Musitech personal-use facility.

Although only a year old, Jeff Siegel's 8-track studio has evolved into more than a demo outpost. Through the benefits of MIDI, the studio can offer advantages over strictly tape-based studios.

"With my equipment, I can turn out a good demo—one where I create, write and record a song—in a maximum of three days," says Siegel, musician/engineer and owner of Musitech, Culver City, CA. "That means working a few lengthy sessions, like the film score I'm working on now."

Musitech is primarily a narrow-gauge facility for record-demo production, pre-production and composition work, film scoring and jingle production. It's also Siegel's personal-use studio, where he can play out his musical ideas.

In the past, the majority of 8-track facilities were tape-based studios, where inexpensive demo productions were produced. However, with MIDI-capable drum machines, sequencers and keyboards, another type of 8-track studio—geared for electronic-music production—is quickly becoming a viable option.

A studio offering electronic and tape capabilities has advantages over traditional tape-based studios, including:

—pre-production and composition capabilities;

—speed and accuracy in obtaining a completed sound and/or demo;

—the capability to replay, manipulate and

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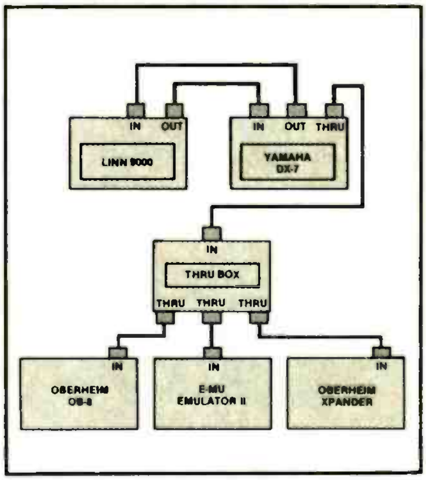


Figure 1. Schematic of Musitech's MIDI-interconnect scheme between a Linn 9000, Yamaha DX-7, E-mu Systems Emulator II, and Oberheim OB-8 and Xpander, in which the DX-7 functions as a master keyboard controller.

modify live, first-generation sequencer tracks, resulting in a high-quality sound at low cost; and
—the capability to merge the musician, engineer and producer into one technically proficient role.

Because the idea of owning a studio appealed to Siegel as a musician, Musitech was constructed around his array of synthesizers, which includes a Yamaha DX-7, an E-mu Systems Emulator II with 17.6 seconds of digital sampling, plus an Oberheim OB-8 and Xpander. All of these units are MIDI-capable, and are linked together via a Linn 9000 32-track drum machine/sequencer, with the DX-7 serving as master keyboard controller.

The console is a 24-input Soundcraft Series 200 working with an Otari MX-5050B 8-track with autolocator, and a second MX-5050B half-track mastering machine. Siegel's outboard gear array would impress any narrow-gauge studio owner: a Yamaha REV-7 digital reverb and signal processor; a dbx model 160 compressor/limiter; an Aphex Aural Exciter; a Roland SDE-1000 digital delay; and a Yamaha model 1000 digital reverb. Studio monitoring is provided by Electro-Voice Sentry 100As, with a pair of ubiquitous Yamaha NS-10Ms resting on the console meterbridge.

Hardware vs. software

As is well known, MIDI-capable sequencers are microprocessor-based units that record, sequence and playback MIDI data. In addition to overdubbing, mode/track selection, tempo changes and copy/paste functions, MIDI sequencers can store note on/off data, start/stop/continue commands, MIDI-clock and chase-lock data. In short, a sophisticated sequencer forms the heart of many MIDI-capable facilities.

Currently, two forms of sequencers exist: hardware- and software-oriented devices. The first type, which includes the Linn 9000 and the Yamaha QX-1, comprises a dedicated sequencer that houses custom software (usually on an EPROM chip). Using an integral storage unit—such as a 3.5-inch floppy disk drive—an integrated sequencer offers MIDI-In, -Out and -Thru connectors, plus sync options via click-track inputs and outputs.

The second type A sequencer, a software-oriented unit, requires more explanation. Unlike the Linn 9000, for example, a software-based system includes a full-scale computer with disk drive, video monitor and, usually, a special MIDI-interface box. (Should the master computer be equipped with built-in MIDI ports, such as the Atari ST520 or Commodore Amiga, this latter device can be omitted from the MIDI network.) For controlling MIDI-equipped synthesizers and drum machines, a variety of software programs are available from third-party vendors. [MIDI data recorders and sequencer software programs are discussed in the October 1985 issue of **RE/P**—*Editor.*]

Before purchasing either type of controlling sequencer, however, users should reference their existing MIDI-capable equipment to decide what is specifically required to equip a studio with MIDI functions.

The advantages and disadvantages of the two sequencer types can be summarized as follows:

Hardware-based:

- With a dedicated sequencer, portability is greatly increased: just unplug your cables and power cord and you're off to another gig.
- Because the unit requires no external programming, it can be argued that hardware-based sequencers are geared more toward the musician than the studio engineer. Through the use of dedicated keys and simplified visual readouts, such a unit is easy to learn and often becomes more like a musical instrument that its software-based counterpart. By using fewer commands and less technology, the user is writing music rather than programming MIDI. Consequently, user-friendliness is high and learning time greatly decreased.
- On some units the sequencer doubles as a synthesizer or drum machine, allowing users to add to their instrument arrays, as well as controling existing MIDI equipment.
- It is often said that purchasing this type of equipment represents a relatively inexpensive way of introducing the user to the computer world. From here, decisions can be made to continue exploring more complex computer-based equipment, or to progress at a rate that might be more related to a user's ability to assimilate the technology.

Software-based:

- After the initial investment, a computer-based MIDI system (with appropriate controlling software) has the capability not only to control synthesizers and drum machines, but also to perform account financing, word processing, digital sampling and conventional computer capabilities unavailable with dedicated hardware sequencers.
- Although such a system is aimed at the computer-literate musician and engineer, its MIDI editing, routing, storage and other functions are limited primarily by software intelligence and on-board memory or disk capacity. To update a software-based system, the user simply obtains a revised sequencer program, thus reconfiguring the computer with enhanced options.
- Screen display is important. Although a user can become accustomed to working with, for example, a small LED window, the ease of operation offered by a full-size, 40- or 80-character monitor screen is greatly enhanced.

Learning to specialize

Because Siegel uses a Linn 9000 as a drum machine and sequencer, Musitech represents a good example of a hardware-oriented facility.

"Right now, for the money I had to spend, I've got what sequencing gear I need," he says. "I don't need computer printouts of scores, or to change a lot of synth patch points. Because of the EII and the Linn, I am basically computer literate. If I did have an Apple Macintosh, I'd store my EII samples, plus patches for my DX-7, and also do my accounts on it. But I'm just not in a dire need for one. I'd rather purchase more outboard gear, like

Mixing and recording hardware at Musitech includes a Soundcraft series 200 console, Otari MX-5050B 8- and 2-tracks, and Electro-Voice Sentry 100A monitors.

parametric EQs, so that I can get a better sound with what equipment I have."

Siegel forsees basically two ways in which an eight-track can survive in the professional recording field. One way is to go 16-track as soon as possible, providing clients with more versatility. The second avenue is to implement electronic tracking, allowing synthesizer tracks to be built and MIDI data stored in a computer for subsequent processing and remix.

Moreover, with a narrow-gauge studio and a limited budget, to remain technically proficient, Siegel argues that certain economic values must be kept in proper perspective.

"Since I constructed this studio for myself, to pay for the gear and to keep money coming in, I opened my doors to outside clients. When I think of purchasing any gear now, I have to consider: Do I need a new piece of processing equipment to bring in more clients, or would it be nice to have for myself? Or would it be just real neat to have?"

Via an in-house synthesizer array, MIDI-based facilities allow more adventurous musical textures to be obtained. Because of limited tracks, however, this option was previously unavailable for conventional tape-based eight-track studios. By implementing an Emulator II's digital sampling function, a floppy-disk sound library, and/or various DX-7 ROM sound cartridges, Musitech clients can build multi-instrument tracks on a sequencer that sound as if a studio full of musicians had played on the production.

In addition, because Siegel also makes his equipment available to outside sessions, he can use past musical ideas and newfound engineering/producing techniques during client productions. This benefit promotes a more confident, comfortable and direct form of communication within his studio. Vocalists rarely have a strong musical idea in mind when they enter his facility, he says, but they can benefit especially from this type of studio.

Siegel's synthesizer array comprises of (clockwise from top left): an E-mu Systems Emulator II, Oberheim OB-8, Linndrum 9000 MIDI sequencer/drum machine, Yamaha DX-7 and Oberheim Xpander.

His synthesizer configuration also allows a musician to access up to 26 different synth voices, all playing different lines simultaneously. With such a capability, Siegel considers that booking expensive time to a 24-track studio is unecessary for demo productions.

"I don't think producers really care if they have album-caliber material for a demo tape. They just want to have a real good sound and find out exactly where the song is going, along with a good taste of the musician. You can do that on eight-track.

"However, if I had 16 tracks, I could certainly think of things to do with them. The way I see it, you work to the capacity of your studio. It's sort of like your salary; if you suddenly double your take-home pay, you'll find ways to spend it pretty quickly."

From a specialization standpoint, Siegel forsees his studio maintaining both an electronic-music and pre-production direction. Musitech can benefit a musician or group that eventually will record at a larger facility.

"Basically, if I were a hot band, I would do all my sequencing at a place like mine—live to eight-track with the Linn sync tone striped on one track, and keep that as a reference tape. Then move either my entire keyboard setup, or just the disks and cartridges, to a 24-track studio. There, I'd play it all back live, and put every sound on its own fader for real versatility during mixdown."

A typical Musitech session

For Siegel, the key to using a MIDI-capable studio lies in the speed and accuracy with which he can obtain a client's desired sound.

"If I want to try something, I can try it fast. Because the 9000 records and plays back looped sequences, you don't have to hear the whole track for just one or two bars; and you can insert or copy bars anywhere within a sequence." The Linn 9000 also enables rehearsal of synthesizer lead lines and drum patterns.

Within the control room, Siegel beings by laying a basic drum pattern from the Linn. Then, switching to the unit's synth option, he uses the DX-7 as a controller to start laying keyboard tracks. As can be seen from the block diagram, in Figure 1, the 9000 and DX-7 are linked together via MIDI, as are the Oberheim OB-8, Xpander and Emulator II. This consistent setup provides a dependable keyboard configuration for Siegel, which in turn promotes speed during the recording and mixing stages.

After the various drum and keyboard parts are layed into the 9000, Siegel replays the sound via a Soundcraft Series 200 console for equalization and signal processing. Although the Series 200 has 16 available inputs, with a bit of perserverence he can use the board's tape returns for additional effects loops.

"Before I can mix the drums," Siegel says, "I first have to figure out how many tracks will be taken up by synthesizers. I then dedicate the remaining channels to drums and percussion. Basically, I mix backwards compared to most engineers. Because there are so many drums outputs on the 9000, I would need 20 inputs to mix them on a complex tune with multiple drum patches.

"When I begin a drum mix, normally I like to isolate the 9000's snare and kick to separate inputs on the board. If I have enough inputs free, I also isolate conga-high and -low, cowbell and the crash cymbal. Then, the last two channels on the board will have the inputs from the Linn's main stereo outs—whatever drums aren't assigned to their individual channel will be summed as left and right stereo on the board. This configuration works well because the Linn lets you pan and adjust volume on individual drum sounds from its front-panel sliders.

"When I get the drum/synth sound that I want, I lay it in stereo on track No. 1 and No. 2 of the eight-track. Then I go back later and start overdubbing guitars, vocals, bass or whatever else the client wants."

Other mixing and recording options are available to the producer via the Linn 9000's sync output.

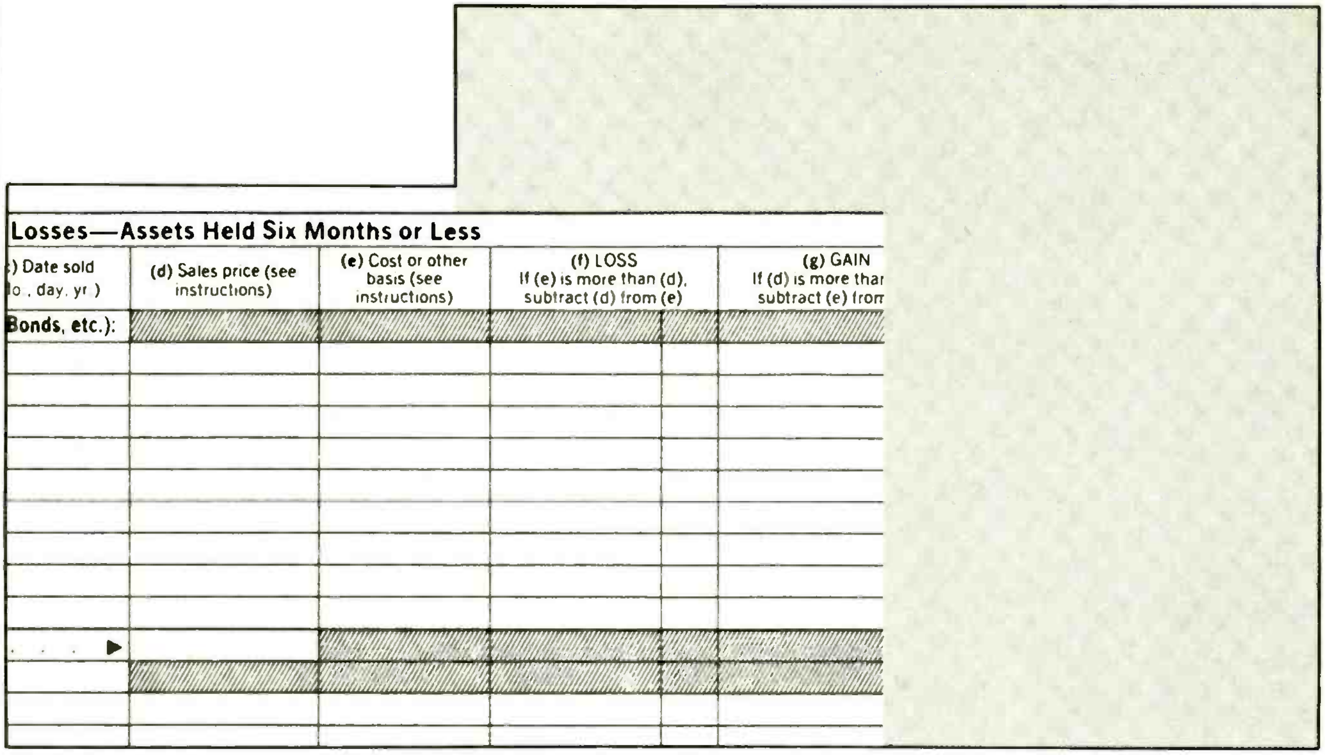
"I usually lay a sync tone onto the eight-track before I start recording," he says. "Then, I play back the tone through a 50ms delay on its way to the 9000, and record the completed sequences onto tape. Having the song offest 50ms behind the sync tone allows me to move [future] overdubbed sequences around the original beat, by changing the delay time on the sequence tracks. This is very handy for obtaining a human feel, or for fixing a keyboard part with a slow attack."

But what of the Emulator II's timecode-generating option?

"SMPTE is great for film or video when you have to punch-in at a certain frame, or if you're going to sync to another machine. The Linn doesn't recognize timecode yet and, basically, it isn't that necessary for the tunes I do here."

According to Siegel, Musitech will continue to focus on its current pre- and demo-production direction.

"Opening my studio, and moving more toward the engineer and producer roles, has introduced me to a lot of people I might not have met if I were still performing live shows. Studio work is a different kind of life for me; I miss the live gigs. But, for right now, I'm open to anything." R·E/P



Losses—Assets Held Six Months or Less

) Date sold lo., day, yr.)	(d) Sales price (see instructions)	(e) Cost or other basis (see instructions)	(f) LOSS If (e) is more than (d), subtract (d) from (e)	(g) GAIN If (d) is more thar subtract (e) from
Bonds, etc.):				
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Understanding the New Tax Laws

By Kathleen White

How will the new provisions of TRA 86 affect the pro-audio industry—and what can recording and production facilities do to minimize its effects?

After a tortuous passage through Congress, the Tax Reform Act of 1986 (TRA 86) made its way to the White House to be signed into law on October 22, 1986. The bill represents the most extensive overhaul of the federal tax code since the early Forties, when tax rates shot up to fund World War II activities. It comes in the wake of 12 major tax bills enacted over the past 25 years. The most striking feature of this latest effort is that it will affect all Americans, regardless of their income level.

In general, TRA 86 provides tax cuts for both individuals and businesses, and simplifies the existing tax-rate structure. It also repeals many deductions, exemptions and tax preference items, more commonly known as loopholes. If all goes as planned, many Americans will see their tax bills cut an average of 6.1%.

Meanwhile, corporations will pick up the slack to the tune of $120 billion. For the first time in tax history, the corporate tax rate will be higher than the individual rate.

As accountants pore over copies of the 925-page code, intense speculation and debate has already begun: What will this "revolution in taxes" do for (or to) individuals, businesses and the economy? And how will the pro-audio industry fare under the new system?

Kathleen White is a Boston-based free lance writer, musician and composer with a background in business management. She is a regular contributor to RE/P.

Checklist of Taxation Changes

If the issues discussed in the accompanying article cause some confusion or raise serious questions about your tax situation in particular, you're not alone. The following checklist highlights the main changes you should be aware of:

- *The elimination of the investment tax credit (ITC) and most types of tax shelters.*
- *Changes in the depreciation laws and the capital gains tax.*
- *Modification of the R&D tax credit.*
- *Increased restrictions on business expense deductions.*
- *A revamped tax rate structure for both individuals and businesses.*

Keep in mind that TRA 86 is a very individualized tax system. In other words, its effects on businesses can only be determined within the context of how each company operates and copes with taxes. Whatever your situation, we strongly recommend that you consult with an accountant or tax attorney for a complete explanation of the new code and its effects.

It would be impossible to interpret the new code in its entirety and predict its effects at this time; such a task will keep accountants and lawyers busy for years. Nonetheless, certain provisions in the new code have already caught the attention of facility owners, equipment manufacturers and rental companies. These provisions, described here in a general way and outside the overall tax picture, could change the ways in which our industry does business.

Investment tax credit repeal

It is generally agreed that the repeal of the investment tax credit (ITC) will have an immediate, industry-wide impact. In fact, the impact is already being felt, because ITC was repealed retroactive to January 1, 1986. The ITC provision allowed companies to deduct 10% of the cost of any capital investment (i.e., any personal property used in a business, excluding real estate) from their taxes. If a studio bought a $300,000 console, for example, it received a $30,000 tax credit.

The repeal of ITC is not good news for capital-intensive industries, such as the pro-audio business; since 1962 it has served as an incentive for businesses to invest in equipment. Companies that have relied heavily on this credit in making purchasing decisions will be most affected, and manufacturers may find a much changed marketplace for their products, particularly in the area of high-end equipment.

According to Peter Scharff of A/T Scharff Rentals, New York, "The bottom line is that it will be more expensive to buy equipment without the ITC—and our business is to buy equipment."

Scharff hopes that studios will decide to rent rather than purchase, and that the increased business for his company will compensate for the loss.

Chris Stone, president of the Record Plant, Los Angeles, feels that high-end studios in particular will be affected.

"We buy all the latest, state-of-the-art equipment," he says. "TRA 86 is taking away some of the best incentives for small, capital-intensive businesses like studios. It will really impact manufacturers' high-end equipment business. Mid-level studios attempting to upgrade will suffer, too."

Marsh Williams of Morningstar, a division of Discovery Systems, says he knew last year that ITC would be repealed.

"Although it is a big loss, we make our purchase decisions from several standpoints, only *one* of which is taxes. I'm not too worried about it; we have to go ahead with our upgrades."

Murray Allen, president of Universal Recording, Chicago, echoes Williams' view: "You buy hardware for *business* reasons, not just tax reasons. One thing we know for sure—we must make our equipment purchases more carefully."

Most equipment manufacturers are still unclear how the loss of the ITC will affect their business. As Harvey Schein of Lexicon explains: "It's too early to tell what the effect will be on our high-end equipment sales. It may possibly give more credence to leasing as a financing tool, especially with the interest rates coming down. We're in the process of analyzing the effects. We could bear the cost by possibly reducing our prices."

To add a note of confusion to the whole issue, rumors are already afoot that the ITC will return in some form within the next year or two.

Depreciation schedules

TRA 86 also changes the depreciation rules. Under the old system, a certain percentage of the cost of capital investments (including real estate) could be written off over a set period of years. One depreciation method, the accelerated cost recovery system (ACRS), allowed more depreciation to be taken in the first couple of years following a capital purchase.

The new tax code extends the recovery period to seven, 10 or 15 years, depending on the item; for commercial real estate, the period is extended from 19 to 31½ years; it also modifies the ACRS in ways too complicated to go into here. The new rules apply to any property placed in service after the end of 1986. Happily, companies can elect to use straight-line, double-declining balance, or 150% declining balance methods.

Opinion is mixed regarding the effect of such changes, primarily because it's too early to predict how depreciation schedules will balance out relative to other factors.

"We'll want to purchase longer lasting equipment," Allen says, "and it may be cheaper to lease equipment. We're going to wait until it all shakes out."

"We've been using double-declining 7-year life method all along," Stone explains, "so there's no big change for us. Also, when we purchase major pieces of equipment, we set up lease-purchase agreements, and we can write off the cost of the lease."

Tax shelters

A big change under TRA 86 is the elimination of most kinds of tax shelters. Under the old system, investments could be made in various kinds of projects, and any losses from these projects could then be written off against regular income. Under the new code, "passive" losses, such as those from limited partnerships, can no longer be written off against "active" income, such as salaries, earnings and portfolio income. If you are involved in a limited partnership, we suggest that you consult with your accountant, pronto.

Clearly, this provision was intended to

encourage people to invest in economically sound projects, rather than ones designed to lose money for tax purposes. Although many private investors may be unhappy about this provision, it may have a positive effect within the pro-audio industry. Steve Parris of Morris J. Cohen & Co., CPAs for Sigma Sound Studios, New York, explains: "People will be looking for passive investments that are making money; recording studios making a profit could be *very* attractive.

"For instance, a limited partnership could be set up; the shareholders would be able to use any passive income generated against any other passive losses they may have."

Capital gains

Yet another TRA 86 provision eliminates any benefit from long-term capital gains. Previously, capital gains tax was applied to the sale of any asset—equipment, stocks and bonds, real estate, for instance—owned for longer than six months. Sixty percent of any gain made on the sale of an asset was exempt from taxes, and the remaining 40% was taxed at a top rate of 20%.

Under the new system, however, long-term capital gains are taxed at the same rate as ordinary income, although capital losses still fully offset capital gains. This is unlikely to affect the pro-audio industry in a big way, because companies are not heavily involved in selling off assets. However, for those planning to sell an asset—a business, equipment or property—taxes will be higher on the gain from the sale.

R&D credit

On a more positive note, TRA 86 revives the research and development tax credit, which expired at the end of 1985. The credit has been modified and extended to the end of 1988.

The new code allows 20% (down from 25%) of R&D costs to be written off. It also restricts the credit to R&D of a technological nature that will result in a new item for sale or use within a business. All of which is certainly good news for any organization involved in R&D.

According to Lexicon's Harvey Schein, "This will have a very positive effect for companies such as ours. The audio industry is technology-driven—the technology is constantly evolving. As an engineering company, all expenses we incur from R&D are subject to this credit. And, although the credit has been reduced, we're happy that it was extended—20% is still a lot better than 25% of nothing!"

Business expenses

Other deductions affected by TRA 86 include business expenses. There are new limitations on deductions for expenses relating to the business use of a home. In brief, deductions are now limited to the net income from the business, rather than the gross income. In other words, you can only use deductions if you make a profit. In addition, the "hobby-loss" rules have been tightened. For people with side-line incomes, this means that the income will be taxable if a profit is shown in three years out of five, as opposed to two years out of five under the old system.

Travel and entertainment deductions have also been curtailed. Deductions are now limited to 80% of the actual cost of expenses such as business meals and night-club admissions. Employees with unreimbursed business expenses will be especially hard hit. The new code requires that several conditions be met before deductions can be used. The employee must itemize expenses, and this requires that expenses exceed a certain limit—about $3,000 for single taxpayers. Beyond this, expenses must exceed 2% of the employee's adjusted gross income for the year; otherwise, no deduction can be made.

Performing artists do get a break on this, but again only if certain conditions are met. One, that the artist works for at least two employers in a year; two, that expenses exceed 10% of wages from performing; and, three, that the artist's adjusted gross income is not more than $16,000. This provision is best dubbed the "starving artist" tax break.

The bottom line

Having summarized here just a few of the new code's provisions, the big question is whether or not the loss of credits and deductions will balance out overall. TRA 86 does replace the 5-step graduated corporate rate structure, which ranged from 15 to 46%, with a 3-step structure of 15, 25 and 34%. However, this new tax rate structure will be phased in over two years, with 1987 being a transition year in which, as of January 1, 1987, there may be a temporary tax hike for some people this year.

In drafting TRA 86, legislators hoped to dispense with the existing labyrinthian code and create a simpler, fairer system. Implicit in the code's provision is the recognition that too many people, for too long, have been able to cheat the system. Also, legislators would like to see taxes become a much less important consideration in business decisions.

Whether or not the new code will divert Americans from their favorite pastime of cheating the feds remains to be seen. Ironically, the bill's passage initially spurred a veritable frenzy of takeovers, mergers, liquidations and restructurings as corporate America rushed to take advantage of soon-to-be-extinct tax breaks. Individuals, too, have been treated to a host of year-end tax-saving strategies.

Obviously, we are all going to have to wait and see what happens. It's worth noting that tax bills are created to be amended; there may be many changes made in the code over the next few years. There is also the question of the massive budget deficit, and how the government plans to reduce that. At this point, a few murmurs are being heard within the pro-audio industry.

As Peter Scharff points out, "There're so many nuances to the code, no one has really read them. It was supposed to be tax simplifications, but it isn't."

Bobby Nathan, owner of Unique Recording Studios, New York, comments that, "In general, we'll have to see how it all balances out and we can't do that yet—the best time to ask about this will be on April 15, 1987."

The tax rate cut may not mean much for those who have paid very low taxes courtesy of the old system's many loopholes.

"We will probably have to pay more because so many loopholes have been closed," says Record Plant owner, Chris Stone. "It will cause us to reach deeper into our pockets to pay our tax bill."

Some businesses might find it advantageous to restructure under the new system.

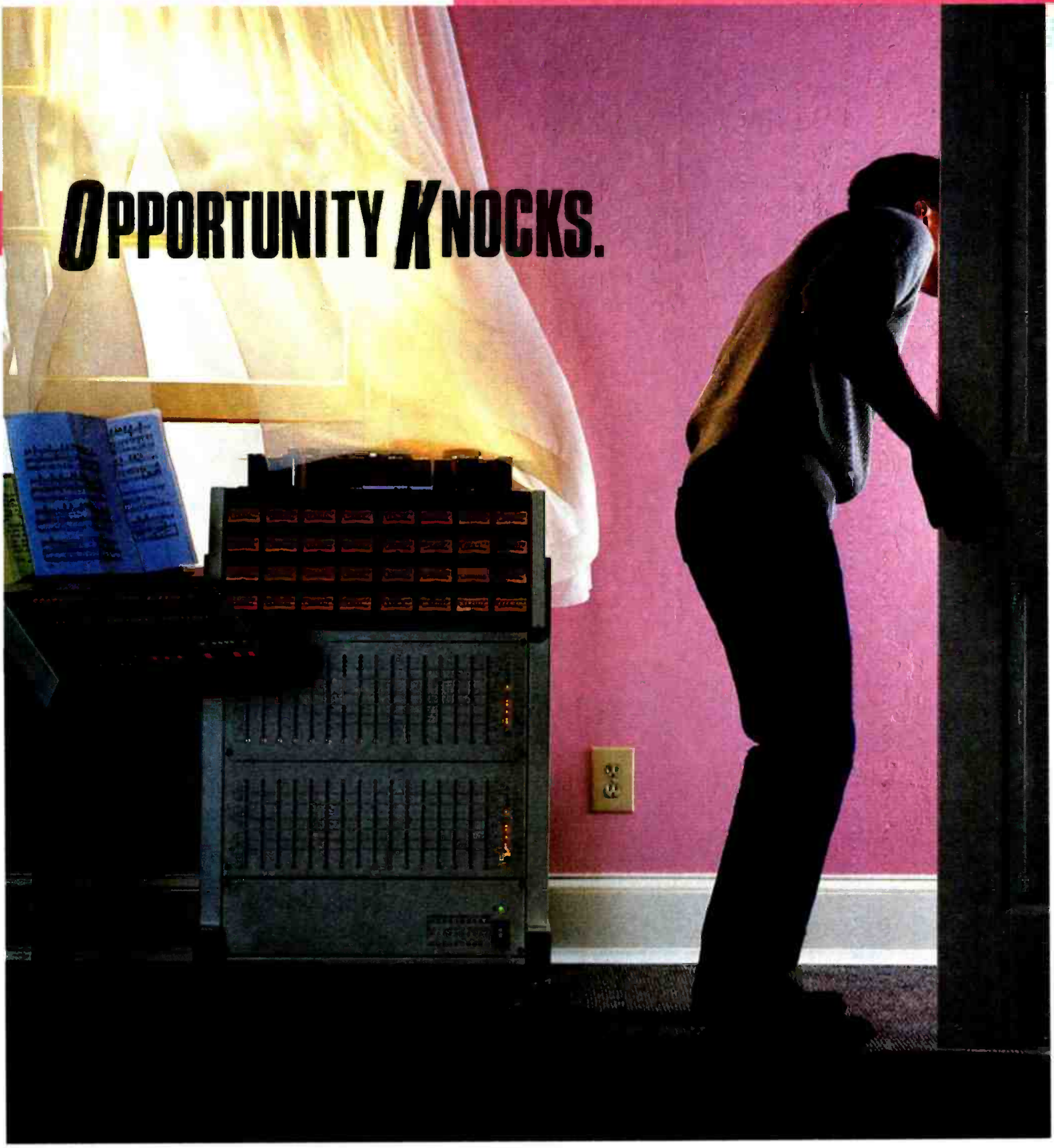
"TRA 86 may encourage a lot of studios to convert to an S corporation, in which the shareholders are taxed at the lower individual tax rate, rather than the corporate rate," says Joseph Ariana of Morris J. Cohen and Co., CPAs.

Doug Dickey, communications director for Solid State Logic says, "The U.S. pro-audio market is very buoyant, and I think the positive aspects of the bill will outweigh any negatives."

Marsh Williams responds in the same vein: "We're in a great position—our cash flow, net operating income, and return on investment have been great this past year.

"And we have the advantage of an excellent management staff in making business decisions. TRA 86 will have the biggest impact in companies that are less-well situated."

Murray Allen, owner of Universal Recording, offers perhaps the best appraisal of how TRA 86 will impact the industry. "I'm going to live with it and try to manipulate my life around it; the big purpose is *not* to pay taxes. Besides, the audio business is strictly a religious experience, and shouldn't be taxed at all." R-E/P



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Sound for Liberty Weekend

By Rosanne Soifer

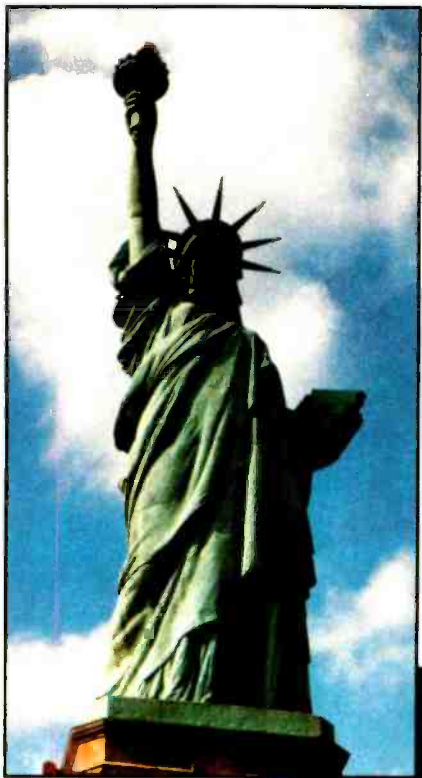


Photo by Rick Hope


Photo by Kooster McAllister


Photo courtesy of FM Productions, Brisbane, CA

Providing live and pre-recorded audio for this 4-day multi-media event was not without its own set of unique problems.

All large scale events normally require an enormous amount of planning. But for the Liberty Weekend, which celebrated the 100th anniversary of the dedication of the Statue of Liberty, the technical scope was unprecedented.

Consider the following factors: four days of events, many occurring simultaneously; 17 staging areas in two states; a worldwide TV broadcast and a radio simulcast; planning between officials from Liberty Weekend, the states of New York and New Jersey and such government agencies as the Coast Guard and the Secret Service.

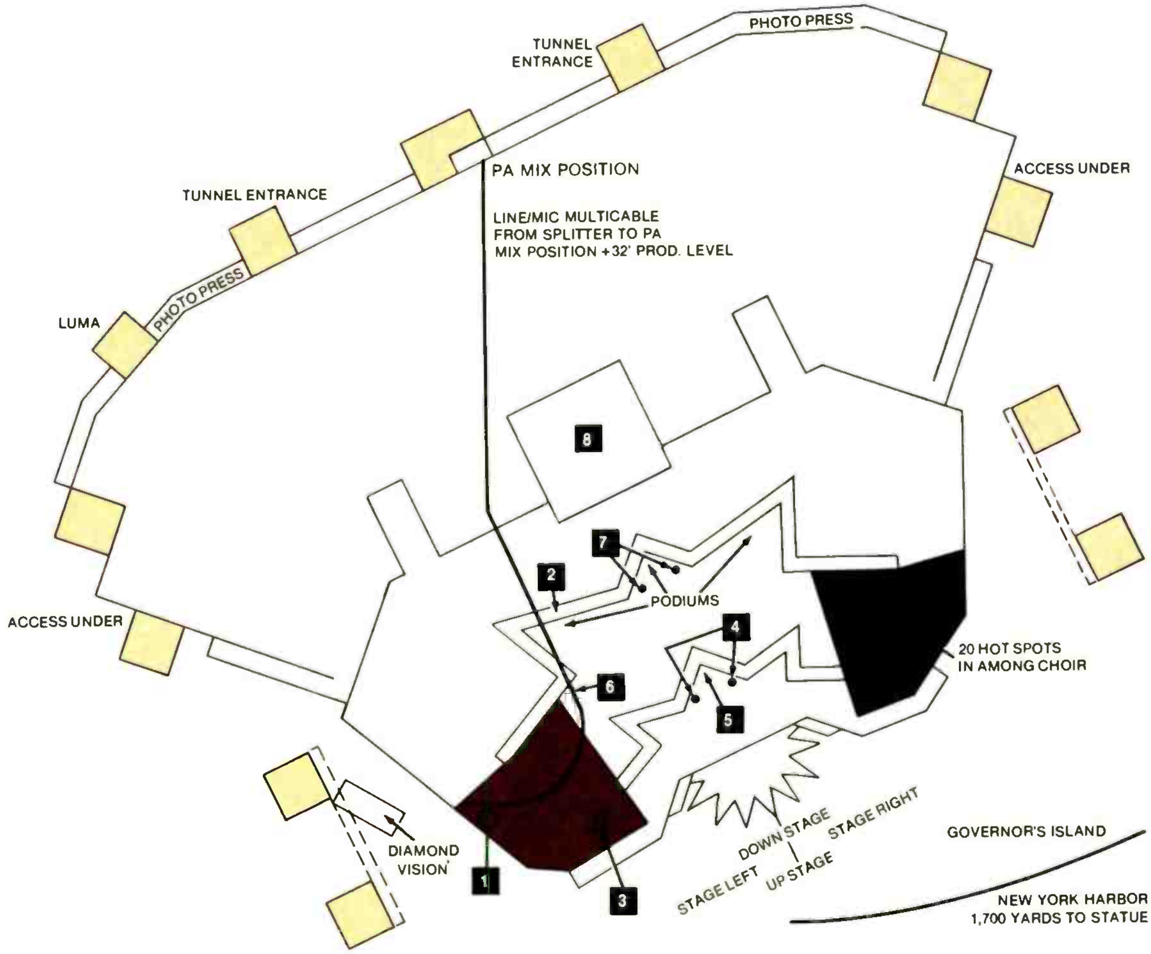
Despite the potential for what could have proved to be a logistical nightmare, the 4-day extravaganza occurred without major technical problems.

Planning for the 4-day Liberty Weekend, which took place from July 3rd through July 6th, began during the summer of 1985. In late October 1985, Statue of Liberty/Ellis Island Foundation president Lee Iacocca announced plans for Liberty Weekend, with David L. Wolper as executive producer. Wolper, who won an Emmy for "*Roots*", had also produced the opening and closing ceremonies of the 1984 Olympics.

The weekend's events were planned on a self-financing basis, using television rights, ticket sales and proceeds from the July 3 fund-raising opening ceremonies. The Statue's restoration funds, however, were kept separate from the proceeds of the weekend events.

Up to 17 different performance and staging areas were involved in the 4-day extravaganza, according to Laurence H. Estrin, who was in charge of coordinating and interfacing all the diverse production elements. Estrin is no stranger to large-scale events, having served as audio director or consultant for the Los Angeles Olympics, the 1984 Democratic Convention, Superbowl,

Rosanne Soifer is a New York-based free lance technical writer.



16° + 50° sound towers include: 2 TL 5050 LF cabinets and 3 HR 4020 horns with DH 2 drivers stacked and splayed.

Choir area

Orchestra area: 80 headphones

1. *Monitor mix position in orchestra*
2. *Four subwoofers built under the stage*
3. *Audio electrical distribution panel*
4. *Two pop-up floor monitors*
5. *Seven cabinets built under stairs and aimed at mid-point of bleachers*
6. *Multicable splitter to house-mix position*
7. *Pop-up monitors with grilles*
8. *Cameras 1 and 2*

Figure 1. *Sketch of Governor's Island layout showing major PA system components.*

various Emmy Award ceremonies, and more than 150 Hollywood film premiers.

Pre-event planning began in November 1985, Estrin recalls. His responsibilities involved not only audio and video for the live shows, but setting up the complex telephone and intercom systems needed for rehearsal, production and press sites, as well as audio press feeds. As Liberty Weekend drew closer, Jim Seiter, documentation chief, continually updated the audio, video and communications blueprints for each venue.

Opening ceremonies

The July 3rd Opening Ceremonies on Governor's Island almost didn't take place at that venue.

"The show was originally supposed to happen aboard the aircraft carrier USS Kennedy, and planning got underway for that eventuality," says Greg Watkins, telecommunications director for Opening Ceremonies and Op-Sail. "But then someone realized the carrier might be needed for a national emergency, so the venue was changed to Governor's Island."

The location change presented the planners with built-in problems, namely high winds and bad weather and their possible effects on audio quality.

"Money wasn't allocated for such items as plexiglass for wind screens or a more expensive sound system," Estrin explains. "We really took our chances that night. Because of the way the stage was situated—right near the water—if the usual 20-mph wind gusts worsened, the sea could have engulfed part of the stage. To some extent the TV air show suffered because wind was blowing the sound the wrong way."

The concerns for wind and weather affecting sound coverage also applied to the live musical performances. According to Bob Wolff, one of the many sound engineers involved on the project, "The orchestra played live to a pre-recorded tape as planned, with everyone wearing headphones. If the wind's effect on the

Interior of the Record Plant mobile during the four day Liberty Weekend, showing numerous outboard audio and video equipment necessary to handle this multi-media event.

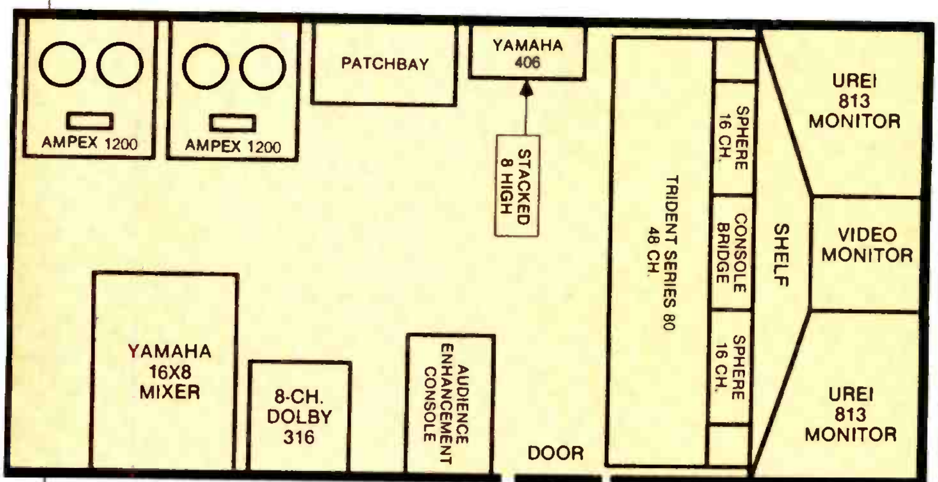


***Figure 2.** Equipment layout in the Record Plant truck.*

mics became too bad, just the tape would have been used. The majority of the music was taped, Frank Sinatra's selection being an exception.

"The orchestras' play-ons and play-offs were also scheduled to be played live. However, if it had rained that night, these also would have been played out from tape, using a pre-recorded 4-track master."

Almost all the music cues were pre-recorded on 24-track at RCA Studio A, New York, initially for weather insurance. Dennis Ferrant engineered the music tracks used during the July 4th fireworks display, while Rick Rowe and Frank Rodriques tracked the music for the Opening and Closing Ceremonies. According to Fred Chapin, a New York-based musician who played guitar, mandolin, banjo and miscellaneous percussion in the Liberty Orchestra for Opening Ceremonies, "Most of the time we were sidelining—the show for us went fine. We had a bit of a hard time understanding French President Mitterand, and coordinating the final live drum roll with the end of his speech.

"Conductor Ian Frases wore a headset so he could receive audio cues; he was the only one who could hear the pre-recorded click track.

"Larry Gatlin was the only act whose vocals, as well as his instrumental accompaniment were pre-recorded."

All audio mixing for Opening Ceremonies, including the live announcements, was handled by the New York Record Plant mobile truck.

"We had no problems or sudden script changes during the Ceremonies," says Kooster McAllister, Record Plant's director of remote operations. "We received our audio tape playback cues from the director (Dwight Hemion, of Smith/Hemion Productions) who was in the Unitel video truck parked behind us near the bleachers.

"We also supplied feeds for the on-stage mix, and fed a mono primary, a back-up mono and a stereo mix to the Unitel truck. The mono mix was recorded on a 1-inch videotape machine for viewing three hours later in the Pacific time zone. A separate stereo feed also went to ABC Radio for their coverage."

"The 'master clock' for us was really ABC Television," Estrin adds. "The time standard was TV-1, and everything had to coincide with ABC's on-air time."

"The bulk of the production work was done in California," McAllister recalls, "and Record Plant only knew 10 days before the Opening Ceremonies what the audio requirements would be, other than the fact that we'd need 150 mic and live inputs.

"This event was more complicated logistically than Live Aid, where Record Plant's truck served as the main production area for mixing live feeds, because of the numerous venues and transportation problems.

"The only major concern for us, particularly with the intercom system, was wind noise and interference. During the Opening Ceremony we had to put windscreens on the Sennheiser shotgun mics, because of the heavy winds."

Live-sound mixing

The live-sound console for the Opening Ceremonies on Governor's Island was located on top of the grandstand with a monitor console being sited at the back of the orchestra's top section. Podium and production mics were split beneath the stage to supply both the Record Plant truck monitor and house consoles.

Orchestra mics went directly to the truck where they were submixed and bussed out to the PA console. Submixes and production mics, in turn, were combined at the Trident console located in the truck. Outputs from a pair of Ampex MM-1200 24-tracks (Dolby A-encoded master and backup reels) went to a Yamaha console in the back of the truck.

Micing President Reagan during the Opening Ceremonies posed some creative audio challenges to engineer Doug Nelson, audio stage manager J.B.

Matteotti, and technical enginee , Ed Ciletti, all of whom formed part of the Record Plant crew. To help reduce feedback problems, the podium designed for the show used AKG C452 mics with VR-1 extension tubes and CK-3 capsules. In contrast to the main podium, which was lightweight and constructed of clear plexiglass, the President's personal podium, weighed approximately 385 pounds and was lined with armor. A trio of microphones covered the President's speech: one connected to his personal tape recorder; one as a backup; and the third, which also contained a kill button, fed the Record Plant truck.

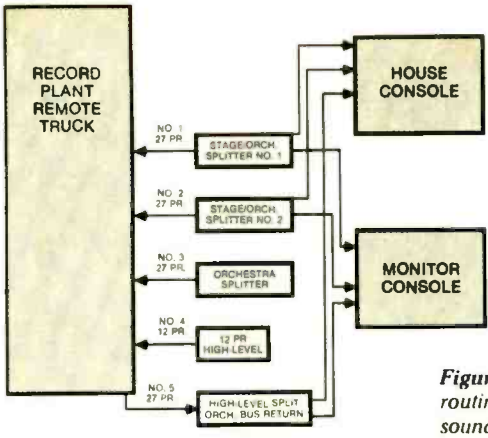


Figure 3. *Schematic flow chart of audio routing between the remote truck and live sound consoles.*

Pre-recorded music and cues

The Record Plant truck also functioned as a sound source for the July 4 Fireworks Ceremony. A pre-recorded 24-track tape contained all the music, announcements and fireworks cues, which were mixed live in the truck and distributed to ABC Television and radio station WPLJ for simulcast throughout the Tri-state area.

"The last 15 seconds of President Reagan's speech, broadcast live from the USS Kennedy, functioned as the countdown for the fireworks tape," says engineer Bob Wolff. Word to start the fireworks portion came from the ABC truck on Governor's Island to Tommy Walker, pyrotechnical director, and the remote truck.

"There was no problem with synchronization because all the audio was on tape; it started and just kept rolling."

The fireworks team actually began work after the Americana Music concert in Liberty State Park, NJ, as the stage was being dismantled following a concert by the Boston Pops Esplanade Orchestra. A special computer program synchronized the detonation of fireworks on 42 barges surrounding lower Manhattan and the Statue.

In the Record Plant truck, which by now had moved across Governor's Island to the Castle Williams location, Estrin and his associates were able to follow the music sequences of specially arranged American and ethnic selections that corresponded to the fireworks cues. The music had been pre-recorded by the Marine Band, which, in 1886, played live at the original fireworks honoring Ms. Liberty, led by John Philip Sousa.

Musical presentations

Selected to represent youthful musical talents of both America and France were the Boy's Choir of Harlem and the Paris Boy's Choir, accompanied by the 55-piece Young Artist's Philharmonic Orchestra of Stamford, CT. The choirs and orchestra joined forces for "*Give Me Your Tired, Your Poor,*" from Emma Lazarus' *The New Colossus*, as used in the 1949 Irving Berlin show *Miss Liberty.*

In contrast to the majority of the Weekend's musical selections, which, with the exception of the July 5 classical music in Central Park, were pre-recorded at RCA Studio A, the choir and orchestra session took place on the second floor of the New York Coliseum. A temporary recording and rehearsal studio house a Yamaha M512 console, Otari MTR-10 2-track, nine Electro-Voice RE-18 mics and JBL monitor speakers.

At first the two choirs were asked to stand about 6 feet apart. After three takes, however, they were intermingled at the request of their respective music directors.

According to Wolff, who engineered these sessions, "We had two interesting noise problems: first, the Coliseum's fluorescent lights buzzed, but without them the orchestra couldn't see its chart , and secondly, noise from the ceiling fans was also picked up by the mics. The fans had to stay on, however, because of the high humidity." (The Coliseum's second floor is windowless and without central air conditioning.)

"Therefore, the tapes needed a lot of EQ work to remove this unwanted noise."

All of the session tapes were eventually mixed and transferred to NAB cartridge. The National Emblem March was used as a taped filler for 5s intervals during the July 5th ribbon cutting ceremony on Liberty Island.

Outdoor sound reinforcement

Maryland Sound Industries was awarded contracts to provide sound reinforcement systems for shows that took place at Giant Stadium, Central Park and Liberty State Park. A total of 24 sound engineers and auxiliary crew were employed, with stage construction at Giant Stadium beginning in mid-June.

Providing adequate coverage for an event the size of the Closing Ceremonies at Giant Stadium was complicated by the fact that much of the running order and stage presentations continued to vary.

According to Dave Reynolds, MSI house mixer, "This was the first time that a proscenium show and a stage show have been done in the same arena at the same time. As a result, we had to come up with some unusual sound solutions at Giant Stadium.

"Because of the two dancing waterfalls designed by FM Productions, the usual stacks of equipment we'd normally use in this sort of venue had to be hidden. Producer David Wolper told us that 'good sound should be heard and not seen' and in this case it's especially true."

During the construction period, it was decided to erect a central speaker cluster, on-stage cabinets and 22 speakers located around the stadium to provide foldback for performers.

As with any outdoor show, rain was always a potential threat. During set up, special waterproof coverings were made for all of the large-scale equipment, rather than being forced to move everything inside the stadium at night should it rain.

The use of several vehicles posed one or two problems prior to the July 6 Closing Ceremonies. These included micing the "lead" Elvis Presley motorcycle and the "oldies" group, comprised of Bobby Rydell, Fabian and Frankie Avalon, who rode to the performance area on the centerfield stage in three antique cars. According to Reynolds, "We were worried that the vehicles' ignitions would cause interference with the TV sound, but the radio mics worked well."

A ground level monitoring system provided the performers with audio cues. The dark areas on stage that were slightly visible to the live audience (but which were undetectable to the TV audience) were comprised of built-in monitors covered by grills substantial enough for performers to stand on them.

Again, most of the music for the Closing Ceremonies was performed live to a

Audio Notes from Liberty Weekend's Closing Ceremonies

By Edward Greene

The closing ceremonies of the 4-day Liberty Weekend celebrations were an entertainment spectacular rivaling the memorable shows from the 1984 Olympics, with preparation starting almost a year in advance.

The first technical meeting took place in Los Angeles in the early fall of 1985 to discuss the general parameters of the proposed show and to try to develop the equipment and crew requirements to handle the event. The show was to take place in the Giants football stadium in East Rutherford, NJ, and would be attended by approximately 75,000 people. My assignment was to direct and operate the television audio feed; Larry Estrin of Best Audio was responsible for stadium sound and communications. Maryland Sound provided the house sound system.

ABC Television leased the Greene/Crowe mobile audio/video truck to serve as the site's master control, and I was hired as free lance audio mixer.

Immediately after that first meeting I placed key personnel and equipment on hold for the projected weeks of rehearsal. Six key field audio engineers and 12 RF diversity wireless mic systems were among the items of priority. With all the shows projected for Liberty Weekend, there were likely to be shortages of people and equipment for anyone waiting too long to book them.

It was soon decided that, because of the problems of weather and rehearsal time, the show orchestra would be entirely pre-recorded. Elliot Lawrence was chosen as music director, and we mutually decided on Regent Sound, New York, to be the pre-record studio. The choir of 1,200 would sing live, but would have limited rehearsal time. As a result, 20 studio singers were overdubbed five times to provide a solid base for the choir.

Almost all of the principal vocals were performed live using wireless hand mics. More than 37 frequencies for the various vocal and communications channels were used in or near the Meadowlands venue.

The main area for wireless mic receivers would be just behind the conductor's podium on center stage. A smaller sub-stage traveled from the front of the mainstage to two marks close to center field. Six receivers traveled under the sub-stage, which the crew nicknamed "The Death Star" (because at night it looked like a Lucasfilm creation).

In case of a power or audio-cable failure, six additional wireless receivers on the same frequency served as backup on the main stage. To the credit of the engineering and operational people involved, no failures or problems occurred.

To cover the event's live aspects—marching bands, horse guards, trumpeters, bell ringers, choir, fireworks, podiums, and audience—36 mics, mostly in pairs, were scattered throughout the stadium. Although using three dozen mics in a studio is not uncommon, covering a stadium this size took a truckload of cable and many long arduous hours to set up and test.

Pre-recording sessions took almost a month to finish at Regent Sound on multiple 24-tracks running at 30ips. Additional recording took place in other studios around the country, to accommodate the artists' busy schedules. All sessions were mixed down to 4-track, ½-inch masters, which were played out from the remote truck.

The entire show was recorded and mixed in stereo, but for radio audiences only. Because ABC Television declined to broadcast the event in stereo, great care had to be taken that the mono television feed was fully compatible. All in all, no major technical problems occurred either during rehearsal or the live telecast.

Probably the most serious problem during rehearsal resulted from a performer putting a radio mic down and leaving the area. Fortunately, the mic was still on, so one of the field engineers traveled the entire stage shouting until we could hear him close to the mic, and finally located it.

My intention in mixing the show for broadcast was to provide the right combination of pre-recorded and live sound; to make the event sound real both for the stadium audience and the listeners at home. No attempt was made to compromise either audience. Ambience mics were grouped by location in the stadium, so that appropriate mics would be operating for different scenes.

The technical "secret" in this kind of multimedia production lies in preparation and proper personnel, along with sufficient redundancies to cover anticipated failures.

Edward Greene is a TV sound mixer, with Greene/Crowe an audio/video mobile truck company, based in North Hollywood, CA.

pre-recorded tape being replayed from the Greene/Crowe mobile truck parked inside the stadium.

As with the other large Weekend venues, Giant Stadium was provided with a complex system of intercoms and an in-stadium telephone system.

"The Greene/Crowe staff helped make a job this size easier than it might have been," says Bob Tourkow, production communications designer for the Giant Stadium, "because they have one of the most sophisticated intercom systems of any TV mobile unit around. The skill of their audio communications engineer, Keith Hall, makes it easy to integrate their system with the massive external system I was responsible for."

Tourkow's communication setup, based upon RTS and Clear-Com units, not only had to interface with the audio and video production areas, but also with the movable stage, dancing waters display and lighting crews.

"The magnitude of the Closing Ceremonies was such that it was difficult to confirm anything until all the people involved were on site," Tourkow says, "since communications had to be designed with junction points everywhere."

"Even though the July 4th Americana Music concert in Liberty State Park and the Giant Stadium Closing Ceremonies included some rock music," continues Larry Estrin, production coordinator, "it just couldn't sound like a rock concert, even though we used 'rock-concert' amounts and types of audio equipment. The concerts at Liberty State Park usually have two speaker stacks—we had five.

"We had to use electric delay to approximate the sound of a symphony orchestra, and also lower the overall acoustic level. For this reason the stage used diamond-shaped acoustic panels to accommodate a symphony 'band-shell' sound, like the one in Central Park.

"An average concert-goer at any rock concert might be about 500-feet from a speaker tower. Yet here, they were only 150- to 200-feet away. Conversely, at Giant Stadium, most rock acts don't use a rolling stage, although Patti LaBelle did, with a sound system located on and inside it. Patti performed the stage moved out to center field."

Even though planning for Liberty Weekend began in the fall of 1985, a number of technical components weren't finalized up until, in some cases, the very last minute. There were absolutely no major catastrophes during the 4-day celebration, which, for an event this size, says a lot for the audio professionals involved. R·E/P

The author thanks Henry Neiger of the Statue of Liberty/Ellis Island Foundation, and Kathy Stahlman of FM Productions for their help in obtaining photos.

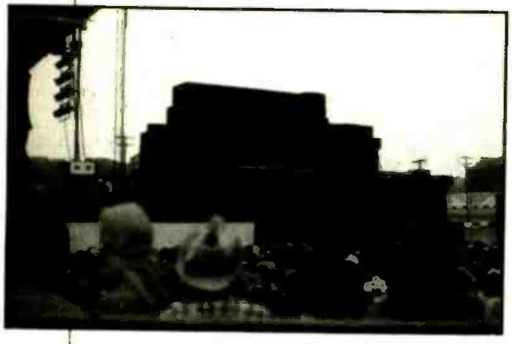


Figure 1: Early rock concert events were served by component speaker systems using available hardware.

Concert Sound System Development

By David Scheirman

The continuing emphasis on high-quality sound and rugged construction will result in some radical reorientation for both sound system rental companies and equipment manufacturers.

Professional concert sound reinforcement systems for touring and permanent installations have undergone noticeable changes during the past five years. These changes have involved not only the equipment used in such systems, but the companies that design and operate them and the events where they are used.

Today's live performance events are served by sound systems that feature multiple-mix consoles, digital signal processing units, compact amplifiers and modular loudspeaker systems hung from the rafters of large arenas. Commercial manufacturers are taking more of an active part in the design of such systems which formerly were the province of touring sound companies.

Contemporary concert events, whether an indoor heavy-metal rock concert or outdoor ethnic music festival, feature better quality audio with a more even audience coverage than ever before. Trying to examine the systems that serve such events can be an exhaustive study. Trying to predict the future direction that the format of such systems will take is risky. Nevertheless, let's take a look at some current trends and possible future directions. Then we'll examine a typical regional sound firm serving a large public event and see how our predicted trends match up to reality.

Establishing the requirement for quality sound

Truly effective, full-bandwidth portable sound systems were once usually seen only at rock concerts. Rock music was a "counter-cultural" phenomenon, appealing to the younger generations of the Fifties and Sixties. Sound system design grew up with rock and roll and its audiences (Figure 1).

Today's decision-makers in civic and political groups, as well as administrators on campuses and in public buildings, grew up with what is still today's popular musical style. Perhaps remembering the poor sound system used at their own high-school prom, many of these people are now the ones authorizing extensive production budgets for fairs and major civic events. Which means that the sound companies serving such events are expected to provide *quality* sound.

Rock concerts are still with us; many major sport arenas and performing art facilities report that popular music events are actually contributing more revenue to facility operations than the dance, symphonic or sporting events for which the buildings were originally designed. Amplified music has been spilling over into mainstream cultural activities as well; for example, the Liberty Weekend celebrations in New York this past summer showcased nationally known rock acts along with other types of talent.

The wide range of performance events available today may lead to the following industry development patterns:

- More regional civic events requiring concert sound systems.
- An increasing number of "World Class" media events involving massive

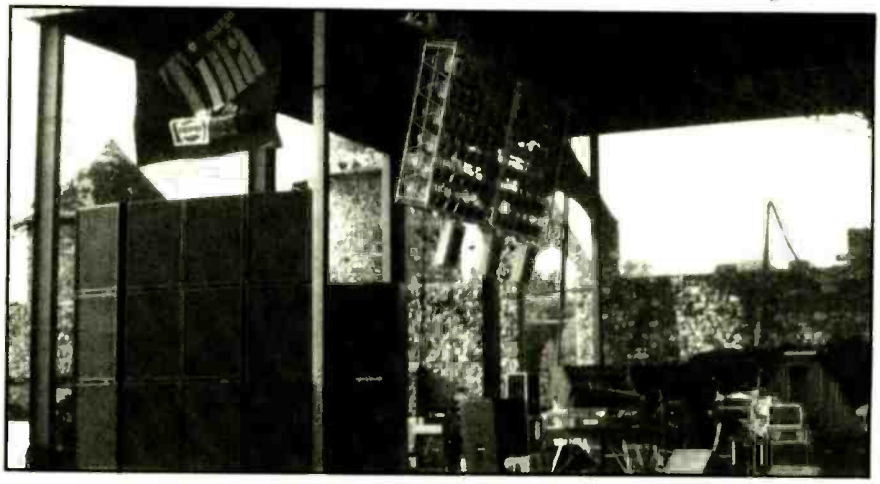


Figure 2: Large 1-box, full-range enclosures (Clair C-4), as supplied to Tina Turner's 1985 North American tour.

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

Figure 3: Trapezoidal loudspeaker enclosures allow the assembly of compact, curved arrays. Shown here: Sundown Sound's CDI 2-box system supplied to the Coliseum, Portand, OR.

sound systems along with broadcast interface.

- New sound companies dedicated to serving such events.
- Commercially available specialized hardware, providing efficient assembly of large systems.

Consoles, signal processing and microphones

Mixing console design is evolving rapidly. While analog-to-digital technology and computer automation has invaded the recording studio domain, the vast majority of mixing consoles used today for live performance are traditional analog devices that have followed a line of development set out over the past decade by several manufacturers.

As features such as solo/mute functions, auxiliary output sends and submaster/matrix mix buses become universally recognized, the list of commercially available consoles has expanded. While 16 or 24 inputs used to be standard, today's configurations are 32, 40 or more.

Today's electronics racks house a wide variety of components, centered mainly on system-drive units, channel-insertable audio controllers and digital special-effects devices. Sophisticated test and measuring devices, including real-time analyzers microprocessor controlled fast Fourier transform (FFT) units, are becoming more common.

Although units such as compressor-limiters, graphic equalizers and delay units, manufactured by early sound companies for concert use, were once seen only in the recording studio or high-fidelity realms. However, complete major systems are now put together from off-the-shelf commercial products.

Certain microphones have achieved classic status on concert stages; many sound companies offer rental microphone kits that resemble similar kits in systems across the country. Perhaps the most interesting developments in stage microphone technology are in wireless systems and the new breed of miniature condenser mics available for special-application instrument use.

A look into the future of consoles, processing and microphones might show us the following:

- Forty inputs will be the standard input configuration for concert sound systems. Additional consoles for mixing opening acts and keyboard/percussion submixes will be carried.
- Advanced manufacturers will develop a virtual console for live performance use, digitizing every control on the console surface. Console automation and information control devices now in recording studios also will become available.
- American, British and Japanese signal-processing device manufacturers will make an increasingly better use of digital technology to produce cheaper and flexible effects devices.
- Input devices available for the concert stage will include direct-input units with active electronics and greater signal control, and smaller higher fidelity microphones.

Crossovers and amplifiers

The system drive and power components available today are becoming more integrated. As manufacturers become aware of what professional sound companies require, the burden of research, development and prototype buildup is shifted away from the touring sound companies. Some of the newer equipment manufacturing companies got their start in that very business.

New crossover technology includes the availability of commercial units that offer integral limiter circuits for loudspeaker protection and output phase adjustment. Some units now make a provision for time-compensation adjustments with variable delays or preset delay circuits for each bandpass tailored to a specific loudspeaker system.

Some companies are beginning to integrate the crossover and power amplifier into a single package. Other firms have developed dedicated processors (including crossover, limiter and thermal-protection devices) to accompany their proprietary speaker systems with the processors designed to be located in the power-amplifier racks.

Amplifiers are becoming increasingly smaller; yet more wattage is also required. The challenge here is to combine compact packaging and reliable circuitry design with accurate sonic reproduction capabilities. Some manufacturers are examining different technology applications, while others concentrate on conventional design methods optimized for the touring-sound industry.

A few patterns emerging in crossover/amplifier design and use:

- Commercially available frequency-dividing networks are evolving into sophisticated system control units.
- As improved processors are developed, more sound system operators will feel comfortable having the crossover functions located in the power amplifier racks.
- Amplifier manufacturers, recognizing the need for quality crossovers, will begin to develop their own control products.
- By 1991, sombody somewhere will

Figure 4: Visual appearance of loudspeaker arrays is becoming increasingly important. Shown here: Audio Techniques' 2-box speaker system featuring natural wood finish, as supplied to the group Chicago.

Figure 5: An aluminum-truss hanging system developed by Speedo Sound for use with EAW KF-550 enclosures.

have perfected a power amplifier the size of a cigar box with integral speaker protection systems that develops in excess of 2,500W rms.

Speaker systems

Modular loudspeaker systems dominate today's professional concert sound rigs. Having first appeared approximately 15 years ago, through the early work of such companies as Clair Bros. Audio, McCune Sound Systems, Northwest Sound and Tychobrahe, some of the first "1-box" systems are still with us. The practical size limits of "1-box" modular systems have probably been reached (Figure 2).

Today's speaker systems are often trapezoidal or "wedge-cut" in shape, allowing the construction of tighter, more compact modular arrays. Frequently, loosely knit "networks" of companies that use the same type of speaker enclosure develop to help ease logistical problems, having nothing in common with each other except the type of hardware purchased (Figure 3).

As the visual aspects of the hardware used for large performance events becomes more important, sound companies are paying more attention to speaker system cosmetics. Today's systems are visually pleasing, and many refuse to play second fiddle to elegant lighting hardware (Figure 4).

The hanging and rigging of speaker systems is becoming increasingly complex. At one time only a handful of major touring firms could suspend speaker systems in the air; today several specialized rigging companies offer their services for hire. Regional sound companies are developing their own unique proprietary hanging hardware using traditional chain-motor hoists (Figure 5).

Figure 6: Custom floor slant monitor speaker enclosure developed by M.D. Systems for Steppenwolf.

One-box modular systems have led to the development of 2-box systems, and dedicated subwoofer enclosures are now available from several manufacturers for use with 1-box systems. Enclosures housing drive units from 15 to 24 inches in diameter offer supplemental low-frequency information to both small and large concert systems.

Present directions might indicate some of the following future development trends for concert sound loudspeaker systems:

- Commercial manufacturers will continue to offer speaker systems that are acceptable even to veteran touring firms.
- Both rectangular and trapezoidal enclosures will be popular. The increasing use of subwoofers with 1-box modular systems will give concert-goers a true extended-range listening experience.
- Loudspeaker components handling frequencies below 100Hz will remain on the ground; components handling frequencies above 100Hz will be flown above the stage area whenever possible.
- Enclosures will become lighter and stronger due to improved transducers and the use of composite building materials.

Figure 7: One of the country's largest civic events held on an annual basis, the Sternwheel Regatta in Charleston, WV, features rock concerts and paddlewheel steamboat races.

Stage monitors

Monitoring systems for on-stage performers are becoming increasingly powerful and complex, although certain standard design trends are evident. Although major manufacturers are beginning to discover what is required on the concert stage, many concert sound companies are still developing interesting proprietary monitor enclosures for specific applications (Figure 6).

Stage monitor consoles are becoming increasingly versatile. Output configurations of 12 to 16 mixes are becoming common, with many monitor desks housing the same number of input modules as a given system's house desk. Some consoles feature onboard parametric EQ for each output mix, in an attempt to simplify the signal path by doing away with outboard equalization.

Major sound companies have worked together with console manufacturers to develop specialized monitor desks. Some smaller companies, including Compact Monitor Systems and Modular Sound Reinforcement, have been specializing in the rental of stage monitor systems for new acts on the showcase club circuit, major tour opening acts and rock video production.

These directions will continue into the future, as onstage sound becomes more critical:

- The monitor mix position will continue to evolve into "control central," with full patching facilities, full communications and production monitoring.
- Console manufacturers will reduce the price of complex products that feature VCA grouping, programmable muting and output matrix switching.
- Off-the-shelf, high-fidelity monitor speaker enclosures will be available, perhaps even in designer colors.

Reality meets prediction

Stage Audio & Lighting Productions, Charleston, WV, is a relatively new regional sound-system rental company. Stage Audio caters to a wide variety of touring musical groups and single-

Figure 8: Stage Audio's house mixing position for the Charleston Sternwheel Regatta, featuring a Soundcraft 800B 40-input console.

performance events. Stage Audio recently supplied complete production services for one of the nation's largest annual civic events, the Charleston Sternwheel Regatta (Figure 7).

Crowds in excess of 100,000 a night were attracted to the city's downtown area to witness paddlewheel steamboat races and to enjoy concerts by such acts as the Commodores, Quarterflash and Tanya Tucker.

"We have seen an increase in the production budgets available for sound and lighting at civic events in the past few years," says Phil Kovacevich, Stage Audio owner. "That translates into better sound for more people. Based in part on the increased activity in this part of our business, we decided to put together a new sound system this year."

Kovacevich, a former professional musician who studied at Berkelee College of Music, Boston, began to examine available hardware before making major purchases.

"Putting together a large concert system from scratch doesn't make sense in today's economy," he says. "In the old days, sound companies used to build their own boxes and crossover. Reliable, road-tested hardware is available from stock that can be put together into excellent packaged systems."

Stage Audio chose a 40-input Soundcraft 800B console for the main deck (Figure 8).

"We felt it was important to offer consoles that visiting soundmixers would be familiar with," Kovacevich explains.

The system's graphic equalizers are Klark-Teknik model DN300s, and the new Brooke-Siren FDS-360 crossovers were chosen.

"It's the handiest crossover we've seen yet," he says. "The phase-trim feature allows the user to optimize the phase relationship of the lows, mids and highs; it gives you something else to fine-tune."

The firm selected Eastern Acoustic Works' KF-550 modular loudspeaker enclosures. A total of 30 boxes were used for the 52-foot wide stage at the Sternwheel Regatta, with five rows per side stacked three high (Figure 9). Additional rear-area delay tower stacks were used to reach the far-flung urban audience.

Figure 10: Subwoofers supplied by Stage Audio comprised EAW SB550L enclosures, each housing four RCF L15/P200 drivers.

Subwoofer enclosures used to enhance the system's bass presence comprised the new EAW SB550L, each of which housed four 15-inch RCF L16/P200 loudspeakers, offering up to 9mm peak-to-peak maximum linear displacement (Figure 10).

Crest model 4000 power amplifiers drove the loudspeaker system (Figure 11). Each 15-inch loudspeaker in the system was set up on a separate 13-gauge speaker wire, with four 15-inch drivers to a channel (700W to 2Ω, stereo). The amplifiers were bridged to mono for subwoofer operation. Crest model 3000 amplifiers drove the midrange and high-frequency components (12-inch RCF loudspeakers and JBL model 2445 2-inch compression drivers on an EAW horn).

Stage Audio's monitor system comprised a dozen of the new EAW SM-222 floor slants, powered by Crest model 3000 amplifiers with Klark-Teknik DN360 graphic equalizers and Brooke-Siren Systems FDS-320 2-channel bi-amplified crossovers (Figure 12).

While the main speaker system used at this event was stacked at stage level, each EAW KF-550 enclosure was equipped with Aeroquip track-type hanging hardware. A rear compartment for cable storage and black steel front grilles, along with handles and castors, made each box road-ready.

Figure 12: Singer/saxophonist Rindy Ross of Quarterflash on stage at the Charleston Sternwheel Regatta. Stage Audio supplied EAW SM-222 monitor wedges.

"We've tried to assemble this system in such a manner that it can easily do both large and small shown with ease," Kovacevich says. "We need to spend our available time getting out and doing shows. It's been good to find components for this system that had already been road-proven, and were developed by reputable manufacturers. It gives us a good selling point for the new system."

Regional companies to grow

Stage Audio & Lighting Productions has assembled, in a relatively short time, a roadworthy concert sound system that was acceptable to national touring groups performing at one of the country's largest annual civic events.

Sound-system components from a variety of manufacturers were used to offer a system that would please touring soundmixers for national groups. The system can be stacked or hung in venues of all sizes.

This particular sound system, and the event for which it was used, is typical of many recently built concert systems that have appeared in public across the nation in the past several years.

The major concert sound firms had a busy 1986 season and many relied on proprietary electronics and speaker systems. Not all sound companies are aimed toward the highly competitive national touring market. Many more are coming into existence, concentrating on providing quality service to their region of the country.

These are the companies that perhaps represent the major portion of market share for many pro-sound manufacturers. Major touring sound companies may be the racing teams, but regional sound companies are the dependable transportation fleet.

There will be more growth in this field within the next five years as development trends in concert sound system will inspire these companies as commercial manufacturers fine tune their product lines to meet market demands. R·E/P

Figure 9: The Sternwheel Regatta's main stage used 30 EAW KF-550 full range enclosures.

Figure 11: Stage Audio uses Crest Model 3000 power amplifiers to drive midrange and high-frequency components.

Photos by David Schelrman

Part 1

Test and Measurement Equipment

By Richard C. Cabot, Ph.D.

What are the important audio measurements you need to make on recording and production equipment and what type of test equipment should a facility look for?

Let's begin by reviewing the principal types of audio measurements and how they are performed. The most common audio measurements are frequency response, level gain/loss, harmonic distortion, SMPTE intermodulation distortion, noise level, signal-to-noise ratio, phase, wow and flutter, and transient response. This article explains some of the finer points of making such measurements and the type of features to watch for when selecting equipment to perform them in the studio.

The measurement of level is fundamental to most audio specifications. Level can be measured either in absolute or in relative terms. Power output is an example of an absolute level measurement; it does not require any reference. Signal-to-noise ratio and gain/loss are examples of relative measurements; the result is expressed as a ratio of two measurements. Frequency response is also a relative measurement, expressing the gain of the device under test as a function of frequency relative to the midband gain (usually at 1kHz).

Distortion measurements are a way of specifying the amount of unwanted anomalies added to a signal by a piece of equipment. The most common technique is total harmonic distortion (THD), but others exist. Distortion is also a level ratio measurement, expressing the amount of unwanted signal relative to the desired signal.

Measurements are performed as a way of verifying specified performance, or as a way of comparing several pieces of equipment. When these measurements are performed periodically on studio equipment, they are a good way to identify equipment in need of adjustment or repair. Whatever the application, audio measurements are an important part of a well-maintained studio.

Richard Cabot is vice president and principal engineer at Audio Precision, Beaverton, OR.

Level measurements

The simplest description of a level measurement is the ac voltage at a particular point in a system or signal path, and can be made with any one of several different types of meters. Let's look at some of their features and requirements.

The most obvious difference between various audio voltmeters is the type of display, analog or digital. Each has its advantages. Analog meters are not easy to use for exact measurements, because of their multiple scales and the need to interpolate numbers from the printed scale. In contrast, a digital meter provides a direct readout of the value to more digits of precision than could ever be read from an analog meter scale.

However, nothing is perfect; a digital meter is only suited for measuring relatively stable signals. While monitoring program level to determine system operating levels under actual use, it becomes extremely difficult to extract a single number from the mass of flashing LED numerals. An analog meter can handle this job with ease.

Another application for analog meters is monitoring the results of an adjustment for a peak or a null. Some manufacturers have put both analog and digital displays on the same instrument to offer the best of both worlds. Typically, the analog scale does not have very fine graduations, and is intended only for approximate measurements of rapidly changing signals.

Most meters of a few years ago were of the average responding/rms calibrated-type, in which the ac signal was rectified and averaged. The reading was adjusted to make the display give the rms value for a sinewave input. For other signals, the response is somewhat hard to define. Newer meters, however, actually measure the rms value of the waveform using special integrated circuits, which allow accurate measurements of voltage for all signals, not just sinewaves.

Rms voltage measurements accurately reflect the heating power of the waveform in a resistor or loudspeaker, a measurement critical to the correct specification of power in amplifiers.

Many noise specifications, however, were developed in the days of average-responding meters and to verify such measurements requires an average-responding unit. Some new devices allow selection between these two responses, giving compatibility with old and new techniques.

A voltmeter's bandwidth can have a significant effect on the accuracy of a reading. Consider a distorted sinewave being measured by two meters with different bandwidths. The meter with the narrower bandwidth does not respond to all of the harmonics, and provides a lower reading. The severity of this effect varies with the frequency being measured and the meter's bandwidth, which can be especially severe when measuring wideband noise. Most audio requirements are adequately served by a meter with a 300kHz bandwidth, which allows accurate measurement of signals to about 100kHz.

Accuracy is a measure of how well a meter measures a signal at a midband frequency, usually 1kHz. This sets a basic limit on the meter's performance in establishing the absolute amplitude of a signal. It is also important to look at the flatness specification to see how well this performance is maintained with changes in frequency, a parameter that describes how well the measurements at any other frequency will track those at 1kHz. If a meter has an accuracy of 1% at 1kHz, and a flatness of 1dB (10%) from 20Hz to 20kHz, the accuracy can be as bad as 11% at 20kHz.

Meters often have a specification on accuracy for each voltage range, being most accurate only in the range in which they were calibrated. A meter with a 1% accuracy on the 2V range, and a 1% per step accuracy, would be 3% accurate on the 200V scale. Using the flatness specification given earlier, the overall accuracy for a 100V 20kHz sinewave is 13%. In many meters an additional accuracy deration is given for readings as a percentage of full scale, making readings below full scale less accurate.

The accuracy specification is not normally as important as the flatness. When performing frequency-response or gain measurements, the results are relative and not affected by the absolute voltage used. When measuring gain, however, the instrument's attenuator accuracy is a direct error source.

Similar comments apply to the accuracy and flatness specifications for signal generators. Most are specified in the same manner as voltmeters, with the inaccuracies adding in much the same way. *Caveat emptor* is the byline when interpreting most of these specs.

Decibel measurements

Audio measurements are often expressed in decibels, which may be defined as the logarithmic ratio of two power measurements or (if the impedances are equal) as the ratio of two voltages. The defining equations are given below for both power and voltage measurements:

dB = 20 log (Voltage #1/Voltage #2)
= 10 log (Power #1/Power #2)

There is no difference between dB values from power measurements and dB values from voltage measurements, if the circuit is loaded in the reference impedance.

The reference for decibel measurements may be predefined, as in the case of dBm (reference: 1mW), or it may be the result of another measurement, as in gain or frequency response. When measuring dBm the reference impedance *must* be specified; both 600Ω and 150Ω are common reference impedances in audio applications.

The equations assume that the circuit being measured is terminated in the reference impedance used in dB calculation. However, most voltmeters are high-impedance devices and are calibrated in decibels relative to the voltage required to reach 1mW in the reference impedance. This voltage is 0.775V in the 600Ω case. Termination of the line with a 600Ω load is left up to the user. If this is not done, however, it is incorrect to speak of a dBm measurement.

The case of a decibel in an unloaded line is referred to as dBu (or sometimes dBv) to denote that it is referenced to a 0.775V level. This should not be confused with dBV, which is an unterminated voltage-based dB measurement referenced to 1V.

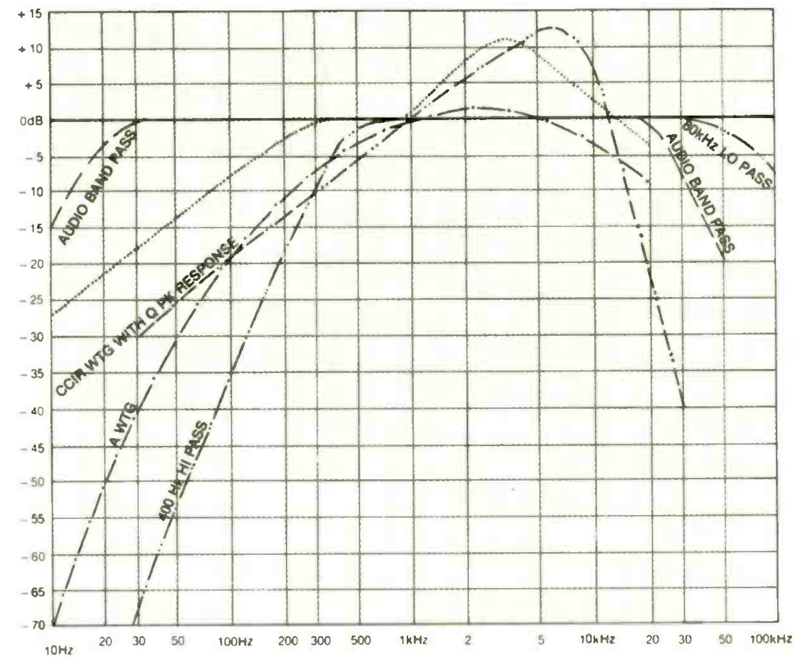


***Figure 1.** Frequency response of various weighting filters commonly used to make audio noise measurements.*

Noise measurements

Noise measurements are really a special case of voltage measurements. It has long been recognized that the ear's sensitivity to low-level signals varies with frequency, an effect that was studied in detail by Fletcher and Munson, and others. The ear is most sensitive in the 2kHz region, with rolloffs above and below this value. To predict how noisy something will sound it is necessary to use a filter that duplicates electrically this non-linear behavior.

Various attempts have been made to mimic the ear's response, resulting in several noise measurement standards. Some of these "weighting" filters are shown in Figure 1. The most common filter in the United States is the A-weighting curve, which is placed in front of a high-sensitivity voltmeter and

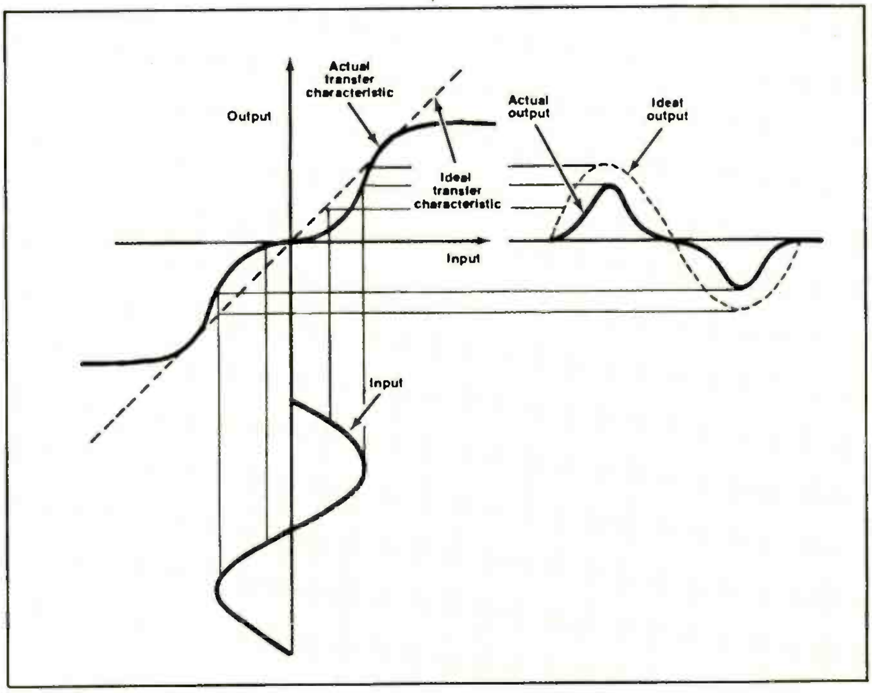


***Figure 2.** Total harmonic distortion test of transfer characteristics.*

the noise amplitude is measured. An average-responding meter is normally used for A-weighted noise measurements, although rms is beginning to be seen.

Some equipment manufacturers specify noise with a 20Hz to 20kHz bandwidth filter and a rms-responding meter. European equipment is usually specified with a CCIR filter, and a special ac/dc converter called a quasi-peak detector. This latter technique is supposed to correlate better with the subjective level of noise than A-weighted, average-response measurements.

About a dozen different weighting-filter/voltmeter combinations have been proposed for measuring noise. Each technique has its group of supporters; they are usually the people whose equipment look best with that particular measurement.

Noise may be expressed as an absolute level (usually in dBm or dBu) simply by measuring the weighted voltage assuming proper termination) at the desired point in the system. This is not very meaningful, however. A 1mV noise spec at the output of a power amplifier may be quite good; a 1mV noise spec at the output of a microphone, on the other hand, would render it useless for anything but recording jet planes.

A better way to express noise performance is the signal-to-noise ratio. S/N is a dB measure of the noise voltage using the signal level measured at the same point for a reference, which makes numbers at different points in the system directly comparable. A signal with a given S/N ratio can be amplified with a perfect amplifier, or attenuated with no change in the noise. Any degradation in S/N ratio at later points in the system is due to limitations of the equipment that follows.

Understanding distortion

Distortion is a measure of signal impurity, and is usually expressed as a percentage or dB ratio of the undesired components to the desired components of the signal. Distortion of a device is measured by feeding it one or more sinewaves of various amplitudes and frequencies. Any frequencies at the output that were not present at the input are termed distortion.

Let's first take the case of harmonic distortion measurement. The transfer characteristic of a typical device is shown in Figure 2, which represents the output voltage at any point in the signal waveform for a given input voltage—ideally this is a straight line. Because the actual transfer characteristic is non-linear, a distorted version of the input waveshape appears at the output.

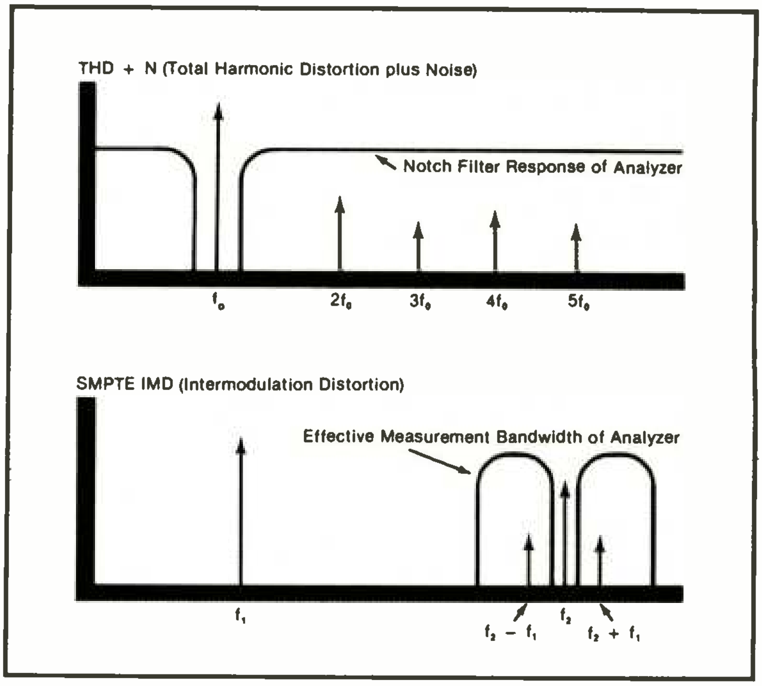


Figure 3. *Frequency spectra of THD and SMPTE IM measurement signals.*

Harmonic distortion measurements excite the device under test with a sinewave and measure the output spectrum. Due to the non-linearity of the transfer characteristic, the output is not sinusoidal. Using Fourier series analysis, it can be shown that the output waveform consists of the original input sinewave, plus sinewaves at integer multiples (harmonics) of the input frequency. The spectrum of the distorted signal is shown in Figure 3. The harmonic amplitudes are proportional to the amount of distortion in the device under test.

A distortion analyzer removes the fundamental of the signal to be investigated, and measures the remainder. A block diagram of a typical harmonic distortion analyzer is shown in Figure 4. The fundamental is removed with a notch filter, the output of which is then displayed on an ac voltmeter. Additional circuitry (not shown) is required to enable setting of the correct input level for calibrated measurements.

Because of the notch-filter response, any signal other than the fundamental will influence the results—not just harmonics. Any practical signal contains some hum and noise, which the distortion analyzer will include in the reading. The correct term for this measurement is then: "Total harmonic distortion and noise" (THD+N).

Additional filters are included on most distortion analyzers to reduce unwanted hum and noise. They usually consists of a 400Hz high-pass filter and a pair of low-

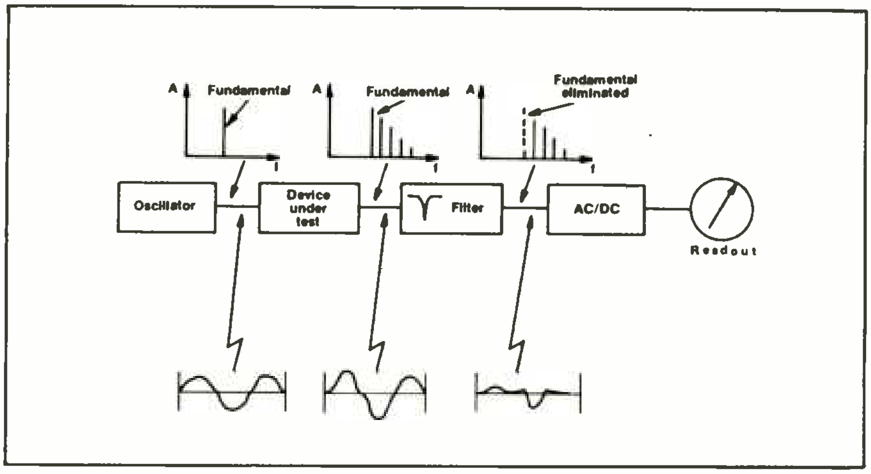


Figure 4. *Simplified block diagram of a typical total harmonic distortion analyzer.*

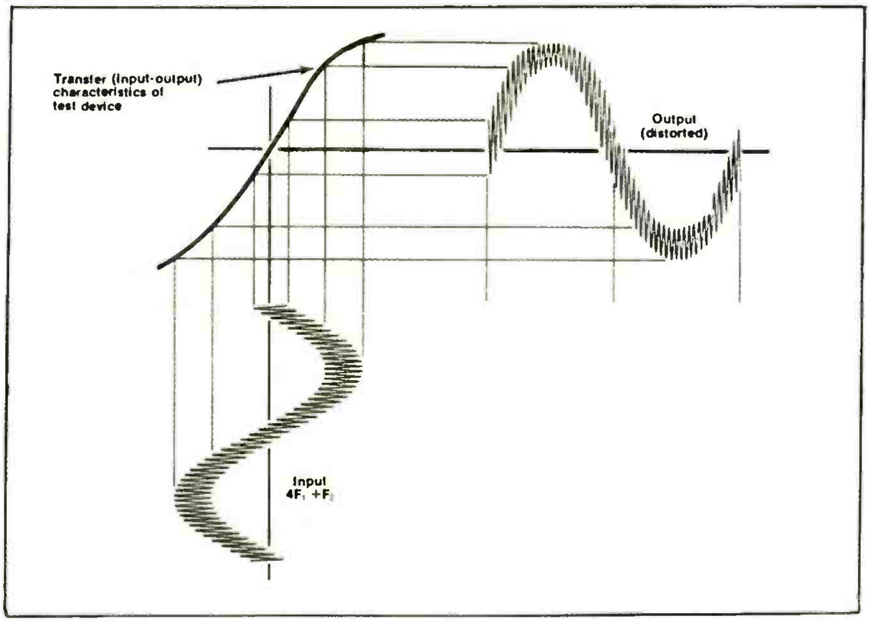


Figure 5. SMPTE IM test of transfer characteristiscs.

pass filters at 20kHz or 30kHz: Used in conjunction with a good differential input, such filters can solve most practical measurement problems.

Another distortion measurement is the interaction or intermodulation of two or more signals as they pass through a device simultaneously. Of the many tests that have been devised to measure this interaction, the most common is SMPTE IM, which uses a low-frequency tone (usually 60Hz) and a HF tone (usually 7kHz) mixed in a 4:1 amplitude ratio. The amount that the LF tone modulates the HF tone indicates the degree of nonlinearity.

As shown in Figure 5, when this composite signal is applied to the test device, the output waveform is distorted. As the HF tone is moved along the transfer characteristic by the LF tone, its amplitude changes. The result is an LF amplitude modulation of the HF tone, and is apparent in the frequency domain as sidebands around the HF component. The power in these sidebands represents the nonlinearity in the tested device.

The signal is measured by filtering off the LF tone, as shown in Figure 6. The HF tone is then demodulated as an AM radio signal and low-pass filtered to remove any remaining HF energy. The resulting demodulated low-frequency signal is the distortion, and is displayed as a percentage of the HF tone's amplitude.

To make the reading represent the true power in the distortion, measurements ahould be performed with an rms-responding meter in the distortion analyzer. Older instruments used average-responding meters, but these are rapidly becoming outdated. With most practical distortion measurements the rms response will read about 2dB higher than the average response.

Accuracy of most distortion analyzers is specified at 1dB, but this can be misleading. As was explained above for voltmeters, separate specifications are often put on the bandwidth and ranges.

A more important specification for distortion is the measurement system's residual distortion. Manufacturers of distortion analyzers often specify the oscillator and analyzer section separately. A device for which the oscillator and analyzer are each specified at 0.002% THD can have a system residual distortion of 0.004%.

If the noise of the analyzer and/or the oscillator is specified separately, this value must be added to the residual specification to find the system's residual THD+N. (It is not uncommon to find this limited system residuals at most input voltages.)

Many commercial units specify the residual distortion at only one input voltage, or at the full-scale of one range; the performance usually degrades by as much as 10dB when the signal is at the bottom of an input range. It is best to look for a spec on total system residual THD+N that includes *all* error sources combined, and holds for all input voltages and frequencies. Otherwise you have got a lot of calculator work ahead.

Manually operated distortion analyzers of just a few years ago required extensive adjustment of notch-filter frequency and depth to ensure adequate rejection of the fundamental. In addition, the 100% reference level had to be set to enable calibrated ratio measurements. Later units ganged together the oscillator and analyzer frequency controls, so that the user did not have to hunt for the correct frequency to make a distortion measurement. Such semi-automatic units are relatively easy to operate when the source can be sighted near the analyzer.

However, many measurements do not allow access to the signal source—putting the user right back to knob twisting and time wasting. Typical of these applications is tape-machine testing using a prerecorded test tape. The better distortion analyzers on the market today are fully automatic in both frequency tuning and amplitude setting. Such automation makes distortion measurements as easy as pressing buttons, whether the generator is two inches or a roll of tape away. R·E/P

Editor's note: Part 2 of this article, to be published in the February issue, will consider the important features of performing phase, frequency and wow and flutter measurements, along with a discussion of balanced and unbalanced connections to test and measurement equipment.

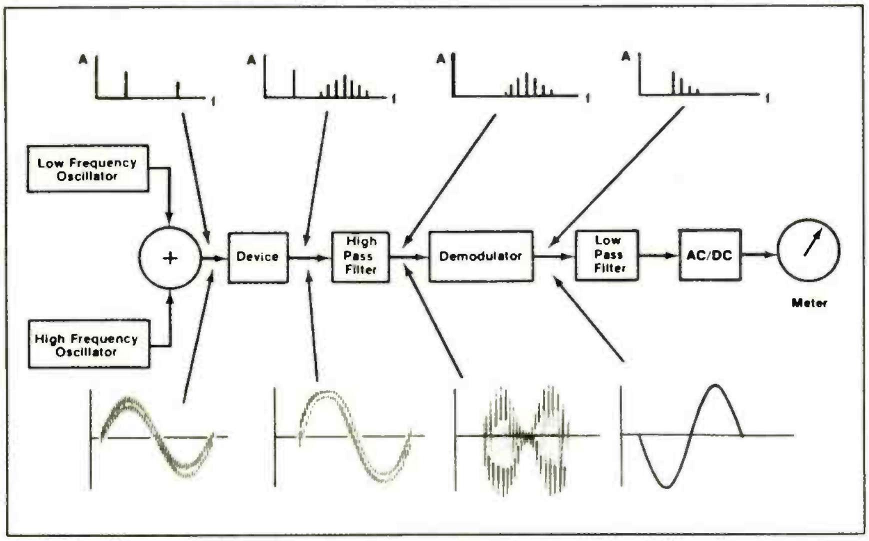
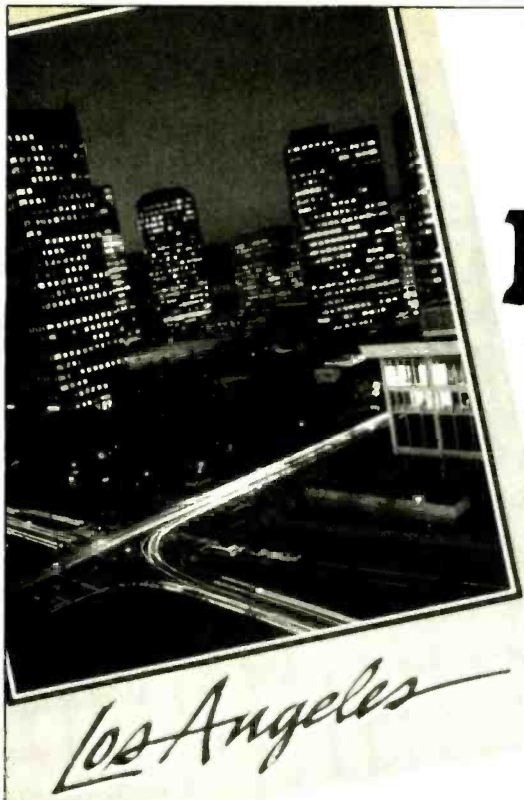


Figure 6. Simplified diagram of a typical SMPTE IM analyzer.



Digital Developments at the 81st AES Convention

By Mel Lambert, editor

Given the size and scope of the recent Audio Engineering Society Convention and Exhibition, at the Los Angeles Hilton Hotel and Convention, Nov. 12-16, it would be impossible for me to provide details of every new product and development announced by the 180-plus exhibitors.

Instead, as has become my customary policy for reporting AES conventions, I have decided to spotlight some of the more important developments in digital technology, including tape machines, outboard processors, editing systems and consoles.

Details of new hardware and techniques that fall into other categories of recording and production equipment can be found in this issue's "New Products" section, beginning on page 86, and in subsequent issues of **RE/P**.

Digital tape machines

DASH- and PD-format transports continue to penetrate just about every facet of the recording and post-production industries. On the PD front, **Otari** demonstrated the 32-track on 1-inch DTR-900, while **Mitsubishi** showed the 32-track X-850, 16-track X-400 and 2-track X-86. During a specially organized press conference, representatives from both companies gave a practical demonstration of format compatibility between their respective machines. Having replayed material originally recorded on an X-850 and a DTR-900-32—just to show that the tapes contained PCM-encoded digital audio—the reels were swapped with predictable results for replay of both digital data and auxiliary tracks.

John Carey, Otari's marketing director, tells me that four DTR-900-32 multitracks have now been sold to Nashville-based studios. In late August Masterfonics took delivery of the first machine delivered in the United States, while Ronnie Milsap's GroundStar studio will be installing a pair of machines in the near future. The name of the third customer was not made available during the convention. In addition, orders are currently pending from studios in Los Angeles and New York, as well as several European facilities.

Questioned as to the possible availability of an Otari PD-format 2-track, Carey says that "it is indeed under consideration, but in response to customer demand." In all likelihood, I understand, the machine would consist of a repackaged Mitsubishi X-86 transport bearing the Otari brand name. (As many readers may already be aware, the X-850 transport is manufactured on an OEM basis by Otari for Mitsubishi, which explains why the former's deck layout bears a more than passing resemblance to that used in the MTR-90 analog multitrack.)

"Whether or not we offer a PD-format 2-track," Carey says, "depends on what users look for us to provide in terms of customer support and our reputation in the marketplace."

According to Cary Fischer, Mitsubishi's director of U.S. sales and technical services, the X-86 prints 20 bits of digital audio to tape. Although currently available A/D converters are only capable of 16-bit resolution, he explains, the company's engineering staff is now evaluating an 18+2 converter that digitizes 18-bit audio and produces a pair of "high-frequency" ranging bits.

"By the end of March," he says, "we will have the capability of recording 18+2 digital audio on the X-86 at a sampling frequency of 96kHz using the new DAC chips."

Fischer also stated that four X-850s had been sold during the course of the convention, and that interest in the 16-track X-400 was coming in the main from broadcasters and post-production facilities, while the 32-track usually was being selected by recording studios.

New for the show was the XE-2 digital editor for electronic assembly and editing, using a pair of X-86 2-tracks, which will be made available in production quantities by next summer. The XE-2's integral bit-lock synchronizer will also be offered as an optional upgrade for the X-850, to provide single-cable time code interlock.

In the DASH corner, **Sony** announced delivery of the 300th PCM-3324 24-track, using the opportunity to present a commemorative plaque to the first U.S. purchaser, John Moran, president and chief engineer of Digital Services, Houston.

I understand that Sony also has plans to unveil a DASH-format 48-track in the not too distant future (PCM-3348?), once it has had the opportunity to evaluate a series of prototype thin-film heads currently under development in Japan.

New for the show was the 2-speed PCM-3404, which enables recording and replay in both 7.5ips DASH-S and 15ips Twin-DASH formats. Also included are built-in electronic editing features, including a 12s memory for locating and rehearsing edit points. Sampling frequency can be set to 44.1kHz or 48kHz, with 12.5% varispeed selectable from the front panel or remote ports. A time code generator/reader, chase synchronizer, AES/EBU and SDF-2 (PCM-1610/30-compatible) digital I/Os are also featured.

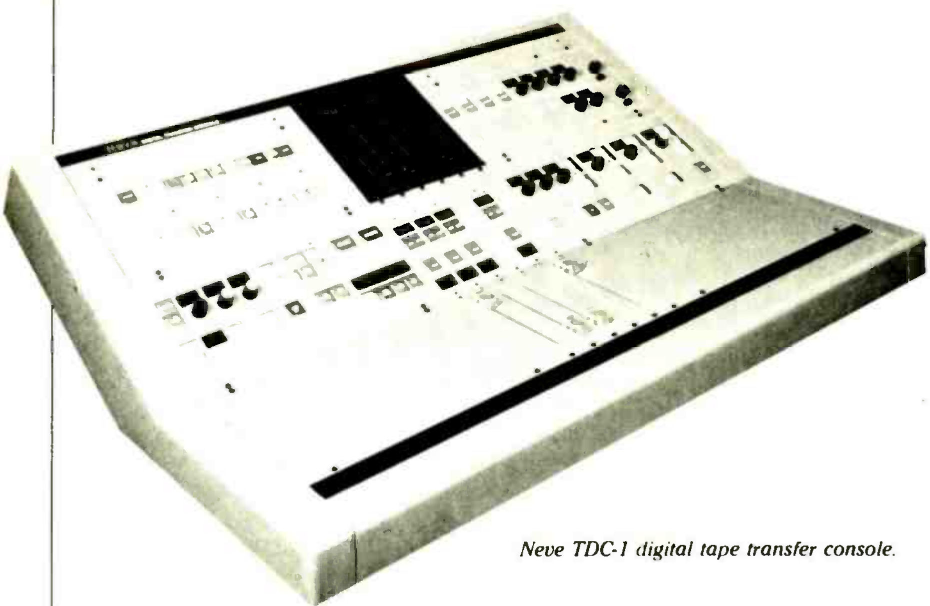
New accessories from Sony for digital post-production facilities include the DFX-2400 2-channel sampling frequency converter, which accepts 32kHz, 44.056kHz, 44.1kHz and 48kHz sampling frequencies via AES/EBU and 1610/30 digital formats. Also unveiled was the VSU-3310 Vari-Sync controller for the PCM-3324, which allows the multitrack's tape speed to be varied by up to 12.5% at a switchable sampling frequency of 44.056kHz, 44.1kHz or 48kHz.

Meanwhile, **Studer** was showing first production units of its Twin-DASH D820X 2-track, with deliveries now scheduled to begin by early spring; no word, however, on a planned DASH-format multitrack transport.

According to Dave Oren, **TEAC** director of marketing, the third DASH signatory plans to release a Tascam DASH-format multitrack by early 1988, with a stereo machine to follow soon.

As is well known, there has been a certain amount of controversy regarding the design of anti-aliasing filters for PCM digital recorders and processors. Having selected a viable sampling frequency (usually 44.056/44.1kHz, 48kHz or 50kHz), steep low-pass filters must be inserted ahead of the A/D encode circuits to remove all frequencies above half the sampling frequency. And, on replay, the reconstituted analog waveform must be filtered to remove residual sampling-frequency components. Designing such filters, however, is not without its own inherent problems, such as maintaining linear phase response over the passband and reducing square-wave ringing.

In an attempt to provide a possible alternative, **Apogee Electronics Corporation** unveiled a pair of low-pass filters for direct replacement of Murata



Neve TDC-1 digital tape transfer console.

and Soshin units used in present DASH- and PD-format recorders, and which are said to exhibit improved linear-phase and ripple response.

According to Bruce Jackson, Apogee's director of engineering, the model 944-G is an inverse Chebyshev input filter designed to provide a "gentle" rolloff between 20kHz and 30kHz.

"The assumption being," Jackson says, "that a recording engineer can make an intelligent assumption about the frequency content of the input signal, and attenuate at the console frequencies that could cause aliasing problems during PCM encoding. As a result, the rolloff curve can be less 'demanding,' with a resultant improvement in sonic quality."

The model 944-S elliptical design is intended for use as either an input or output low-pass filter, and exhibits a tighter rolloff response to the Nyquist Frequency ($F_c/2$).

Jackson says that both filters exhibit a ±5 degree deviation from linear phase response in the 20Hz to 20kHz band, with 0.005% distortion; stopband attenuation for the 994-S is a quoted -80dB, and -60dB for the 994-G.

Village Recorders, West Los Angeles, is reported to be the company's first customer, having replaced the 48 I/O filters in a PCM-3324 with Apogee units.

Professional R-DAT technology (rotary-head/digital audio tape) also made a major appearance at the AES Convention. Sony was demonstrating "show samples" (a quaint euphemism that I had not come across before) of its DAT-X2 portable R-DAT recorder. The hand-built prototypes on display featured a 2-hour record time at a 48kHz sampling frequency, AES/EBU digital I/O, balanced analog I/O and full time code capability. Also being shown were prototypes of various R-DAT duplication systems, error-rate counters, data generators and other peripherals used with Sony's designs and evaluations.

According to Larry Blake, **RE/P's** film sound consulting editor, (also see: "Letters," page 8) the proposed wish list for a "digital Nagra" described in his October "Film Sound Today" column bears a close likeness to the features incorporated in the DAT-X2. Production versions are expected to be made available by mid-spring, at prices described as being "competitive" with currently available analog portables.

Digital consoles

In the past I have reported on all-digital designs from Neve, Denon, Sony, JVC and Enertec-Schlumberger. New for the AES Convention were three products by Neve, Yamaha and Harmonia Mundi.

The **Neve** TDC-1 digital transfer console, intended for CD pre-mastering and related duties, has a pair of stereo digital inputs (AES/EBU or PCM-1610/30-compatible) and one stereo analog input. Manual or auto crossfades can be made to analog and digital outputs.

All digital EQ and dynamic-range control settings can be memorized and recalled under time code supervision. Servo-controlled faders (one for each input) handle left and right balance, stereo balance and stereo level. Systems have been ordered by Sterling Sound and Master Disk, New York, Disk Mastering, Nashville, and Precision Lacquer, Los Angeles.

The **Yamaha** DMP7 features eight analog inputs and two analog outputs, with the capability of ganging together a maximum of four units via a dedicated digital I/O port for a 32-by-2 configuration. All I/O digital processing is to 16-bit resolution at a sampling frequency of 44.1kHz.

Each channel offers a 3-band digital EQ section, three effects/cue sends, stereo pan and a motorized fader. Each of the three effects/cue buses has a dedicated DSP for generating special-effect programs. Buses 1 and 2 provide 15 digital effects each, while effects bus 3 offers five digital effects, an external effects send and stereo effects returns. The main output buses also feature digital dynamic-range control, external controllable levels, and balanced and unbalanced analog outputs.

All mix and processing settings can be memorized in a total of 32 "snapshots" stored in on-board memory. An additional 67 can be downloaded to an external RAM cartridge. Each snapshot is accessible in real time, and all mix and processing parameters can be controlled externally via a MIDI interface.

An LCD window and keypad section controls the recall of processor settings, and the increase and decrease of level and parameter values via sets of cursor control and nudge buttons. First shipments of the new DMP7 are expected to begin in May.

Of a modular design, the Digital Tonmeister from **Harmonia Mundi** now features digital EQ, dynamics control and level adjustment functions in a separate "control surface" that connects to the familiar bw102 rack-mounted units. A programming module provides recall of parameter settings and a MIDI interface, while the parametric EQ controller module features LCD windows and control knobs for LF, two MF and HF sections on third-octave centers.

A remote level controller can be used in several modes to provide manual and various degrees of automatic fade in/outs. Disc mastering preview can also be achieved via a dedicated module that provides up to 2.7s of delay, with optional 16-bit D/A conversion. A dynamics control section is expected to be made available by early spring.

Digital I/O can be selected for compatibility with AES/EBU, PCM-1610/1630, EIAJ, Mitsubishi PD and other formats. Sampling frequency conversion is available between 44.1kHz and 48kHz.

Although **Solid State Logic** has shown a natural reluctance to tell the pro-audio industry too much about its continuing development of an all-digital console, three technical papers presented by SSL staff provided an interesting insight into the general areas of R&D that the company is currently exploring. Reported to represent more than 30,000 hours of research, development and programming, the three papers confirm the existence at SSL of proprietary 24-bit linear digital circuitry capable of very high-speed computation and processing.

One paper in particular, "Digital Audio Processing on a Grand Scale," written by Peter Eastty, SSL's digital project leader, explains that the company has taken an almost entirely hardware-independent approach to the design process.

As he explained, "Software-controlled audio processing provides a level of flexibility and a capacity that simply cannot be achieved by consoles whose capabilities are primarily determined by hardware considerations."

Eastty also showed a prototype digital processing board said to be capable of executing 40 million instructions per second, with a 25ns cycle time for arithmetic and data storage being done to 24-bit precision, these parameters lead to a memory bandwidth of 240 Mbytes/second for program, and 360 Mbytes/second for data access.

During a second technical presentation, titled "An Automated Approach to Digital Console Software Design," SSL's William Kentish and David Bell described a method whereby individual consoles are defined solely by two large software programs and a control-surface layout.

Summarizing these ongoing developments, Colin Sanders, SSL chairman and managing director, said that a product announcement would be made sometime in 1987, and that the "large-scale SSL Digital Studio System we anticipate offering in 1989 or 1990 should cost roughly the same as assignable analog consoles that will be available [in 1987]."

Random-access editing systems

Both AMS and Lexicon were maintaining a high degree of visibility with their disk-based recording and random-access editing systems.

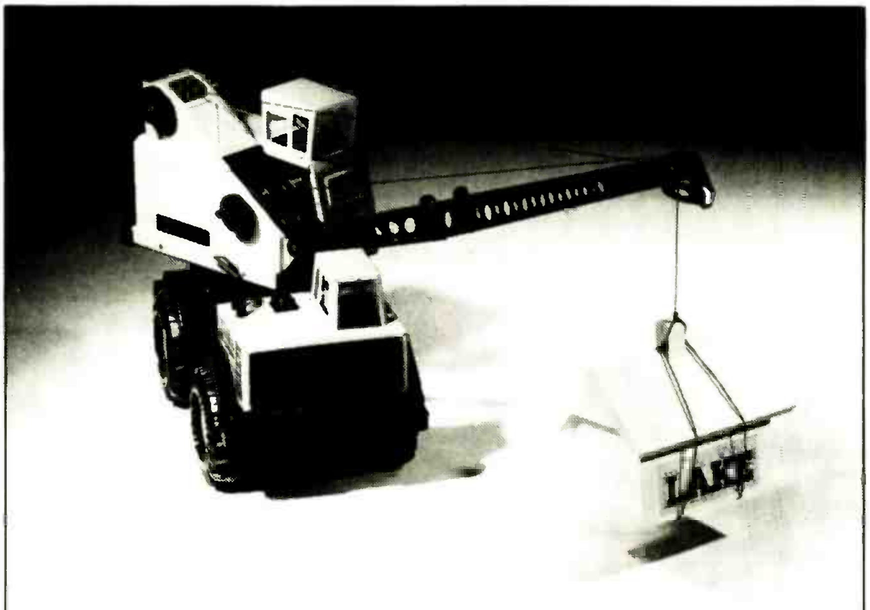
The **Advanced Music System** AudioFile now offers control of external audio- and videotape transports from the unit's front panel via RS-422 serial ports. For U-matic VCRs, the familiar jog, shuttle, fast-forward, reverse and play controllers have been provided, enabling still-frame time code locations to be captured via VITC.

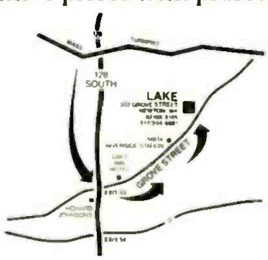
According to Harry Harris, president of Harris Sound, the U.S. distributors for AudioFile, the system's software has been enhanced to provide replay while in record mode, and simultaneous replay of five mono or three stereo channels (with simultaneous replay from all of the system's eight available channels to be made available in the near future). In addition, a "ripple" command now allows relative time code offsets to be made automatically to an exisiting audio decision/replay list, simply by marking the files to be advanced or retarded, and the number of time code frames required.

Several Los Angeles- and Chicago-based facilities are showing interest in the system, Harris says.

"By January [1987] we expect four or five systems to have been installed. We are now providing in-house demos at a client's location, which will let them have hands-on experience with the system in a real-world post-production environment. AMS is also considering the development of an all-digital mixer for AudioFile, and is evaluating customer requirements."

The **Lexicon** Opus integrated digital mixer, recorder and editing system was being demonstrated throughout the AES Convention. (A full description of the system's design appeared in the December issue.) Although the company





Yamaha DMP7 8-input digital mixer.

was the first to concede that not all of Opus' functions are yet fully operational, most of the critical capabilities were being demonstrated, including digital recording to hard disk, full bandwidth "rock-and-roll" edit-point location and non-destructive editing, mixing and panning, plus real-time connecton to external signal processors via digital I/O (in this case a model 480L digital effects unit).

Up to 480 "track minutes" of record time is available in a fully loaded system, with simultaneous replay from up to eight of the 28 available analog/digital outputs. Digital I/O is currently restricted to PCM-1610/30 format, although others will be made available in the future. Sampling frequencies of the 16-bit input A/D and four times oversampled 16-bit D/A output converters can be set to 44.056kHz, 44.1kHz or 48kHz.

For real-time control, the system will follow 24fps, 25fps, 30fps and drop-frame EBU and SMPTE time codes, NTSC and PAL video sync, or 60Hz pilot tone.

Installation of the first Opus system is expected to be made by next spring, according to Lance Korthals, director of marketing and sales.

Digital Synthesizers

Both NED and Fairlight announced enhanced features to their respective digital synthesizers and sampling systems.

The **New England Digital** Synclavier II system was demonstrated in three pro-audio applications: music recording, as the centerpiece of a "tapeless studio" with direct-to-disk recording, and in post-production.

The system now features up to 32Mbytes of on-board memory storage, 16-bit precision and a 100kHz sampling frequency. Up to 76 simultaneous sounds—either musical timbres or sound effects—can be split across the keyboard controller, with velocity and pressure sensitivity. Stereo sound sampling also is now available.

For post-production applications, the 76 sampled sounds can be panned and processed with pitch bend for Doppler effects, etc. Lock-up time to video is within 500μs, with sync being maintained down to 1/5th play speed for spotting visual cues. A new mouse-driven feature allows "rock-and-roll" editing over the selected sound sample, with the capability of deleting certain segments.

A separate direct-to-disk sampling option provides 16 tracks of data storage, with random access to any sound element and editing capability via the mouse interface. A total of 13.5 minutes per track of digital audio storage is now available at a sampling frequency of 50kHz. According to Bradley J. Naples, NED president, a free-standing direct-to-disk sampling system will be offered in the future; currently it must be used in conjunction with a Synclavier II.

Fairlight Instruments had also laid on a series of demonstrations during the AES Convention, featuring Jan Hammer, Chicago-based jingle/commercials producer Terry Fryer and Australian composer Mars Lasar. New features for the CMI Series III include time code chase/lock and full bandwidth stereo sampling. Engineer/producer Roger Nichols also presented the company with a CMI-compatible floppy disk containing digitally sampled drum and percussion sounds downloaded directly from Wendel, Nichols' sampling drum computer.

New England Digital Synclavier II direct-to-disk recording system.

Other developments

Other items of interest unveiled during the AES Convention included:

• The DAP-1 digital effects processor from **DCS Audio Products**, which features 5s of storage at the maximum sampling frequency (software variable between 12kHz and 48kHz); optional memory upgrades to provide almost 360s of full bandwidth storage; up to 1,000 preset effects; a basic configuration of four analog or digital (AES/EBU-format) I/Os, expandable to eight I/Os, with different processing on each of the eight channels; external MIDI control; and dedicated effects development software for an external Apple Macintosh.

• The **AKG Acoustics** DSP-610 Delta Stereo Processor, which comprises a multichannel programmable delay unit with integral signal processing for "enhanced localization, greater apparent depth of the sound stage, and improved spatial reproduction of sound sources" during live-performance applications. Up to 6-input channels are PCM-encoded to a 16+3-bit format for automatic routing to 10-output channels with different delay settings. External control of control parameters is achieved from an IBM-AT or compatible PC via and RS-232 serial port.

• The **360 Systems** Permanent Playback digital audio message system, which allows up to 84s of 10kHz bandwidth voice IDs, sound effects, laugh tracks, etc., recorded on internal EPROMs to be triggered to within a 1mS accuracy during post-production and sweetening. Units can be cascaded to provide longer sound sample times. A separate programming unit enables pre-recorded sounds to be transferred to EPROM for use in the system.

• The **Teldec** Direct Metal Mastering system for Compact Disc, which enables CD masters to be produced without the need of a clean room, using a conventional-style lathe and a custom-designed cutter head. Because the metal master can be used to produce mothers and pressing stampers, fewer production stages are required in CD manufacture, Teldec says, resulting in lower costs. A complete DMM-CD system is reported to cost less than one-third that of laser-based mastering systems.

Rapid Facts Card Numbers

For more information on products mentioned in this AES Convention report, circle the following numbers on the Rapid Facts Card to be found in the back of this month's issue:

Otari DTR-900 32-track 100
Mitsubishi X-850 32-track 101
Mitsubishi X-400 16-track 102
Mitsubishi X-86 2-track 103
Mitsubishi XE-2 digital editor 104
Sony PCM-3324 24-track 105
Sony PCM-3404 2-track 106
Sony DFX-2400 converter 107
Sony VSU-3310 Vari-Sync controller 108
Studer D820X 2-track 109
Apogee Electronics 944-series filters 110
Sony DAT-X2 portable R-DAT recorder 111
Neve DTC-1 transfer console 112
Yamaha DMP7 8-input mixer 113
Harmonia Mundi Digital Tonmeister 114
Solid State Logic digital articles 115
Advanced Music System AudioFile 116
Lexicon Opus 117
New England Digital Synclavier II 118
Fairlight CMI Series III 119
DCS Audio Products DAP-1 effects unit 120
AKG Acoustics DSP-610 stereo processor 121
360 Systems Permanent Playback system 122
Teldec DMM-CD system 123

R-E/P

Amplifier to Loudspeaker Connections

By Bob Hodas

A new design might be worth considering for studio and live sound applications.

Banana plugs, XLRs, terminal strips, phone jacks and other types of connectors continue to battle for the primary position as the pro-audio speaker cable connector. Various speaker manufacturers have their favorites, yet on many installations one may run into a variety of non-compatible connections. No standard has prevailed and until now there has been no method that appeared to offer a distinctly superior alternative.

ITT-Cannon of Australia recently introduced a connector that offers what appears to be a safe, roadworthy solution capable of meeting high-level specifications. I would like to take this opportunity to introduce this connector to RE/P readers, and solicit your response concerning its possible adoption as a standard.

The connectors, shown in Figure 1, are the AXR-PDN-11 and AXR-PDN-12 straight-cord plugs (a right-angle version will also be available), the AXR-PDN-31 (female) for mounting on the amplifier output and the AXR-PDN-32 (male) for mounting on the speaker. (Circular mounts also will be produced.)

Notice that the normal male-out/female-in configuration for low-level lines is reversed for safety reasons. With the large, high-wattage amplifiers available today, it is often possible to encounter 80V rails. For this reason, any connector that carries voltage will be female, preventing bridging with one's finger. Male connectors will be used at the receiving end (speaker cabinet).

The design is based on the familiar XLR-style shell, a configuration that has proven itself. Its use also will allow manufacturers to work with existing castings, cutting down on manufacturing and adaptation costs. Some users may question whether the connector pins are sufficiently large to carry enough amperage, but as will be shown, the configuration is actually overkill for our purposes.

The PDN part of the connector identification stands for positive, drain and negative, which represent the polarity configuration. (However, don't confuse the drain with "common," because this is meant to be a true energy drain; more on this later.)

A flat section on the connector housing provides a place for circuit identification, and 10 resistor-code boot colors are available. ITT-Cannon is proposing two different boot and insulator colors for channel identification: blue for right and white for left.

A large, D-shaped cable entry will pass circular cable up to 0.4 inches (10mm) in diameter, or figure eight of 0.3 (7mm) by 0.5 inches (12mm). This entry is large enough to handle even giant "audiophile" cables, and I had no problem using the extra large-sized Monster Cable Powerline 2. Two lengths of clamping screws are supplied to facilitate a variety

Bob Hodas is RE/P's evaluations and practices consulting editor.

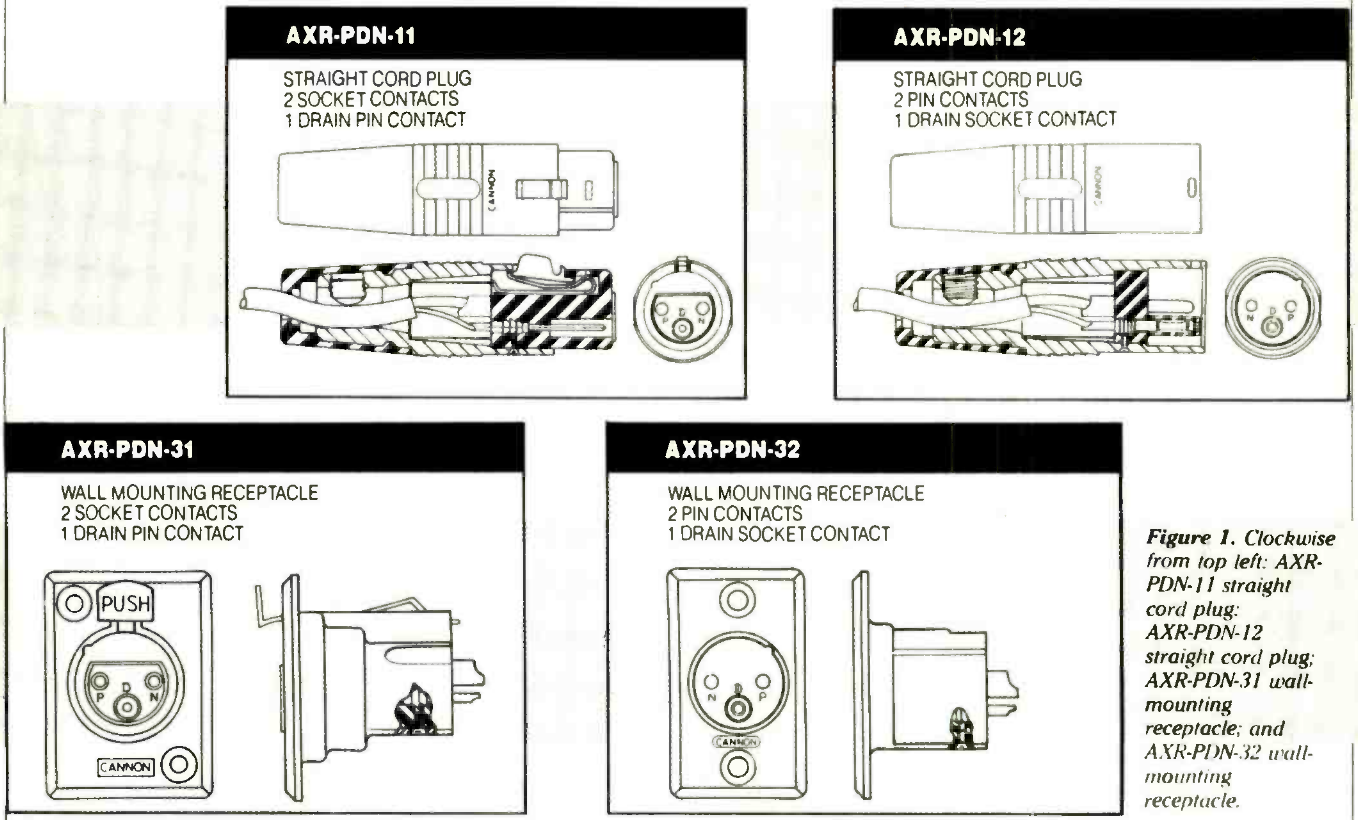


***Figure 1.** Clockwise from top left: AXR-PDN-11 straight cord plug; AXR-PDN-12 straight cord plug; AXR-PDN-31 wall-mounting receptacle; and AXR-PDN-32 wall-mounting receptacle.*

of cable diameters, for which Cannon claims provide maximum force with minimum cable damage. The entry is covered with an elastomer boot that may be trimmed to accommodate larger cables.

Power rating

The connector is designed for a continuous rating of 200V rms and 25A rms, a value that was established after surveying pro-audio needs. The values, as shown in Table 1, appear to be more than adequate. Using a local supply authority's specification to ensure compliance, the tested results are shown in Table 2. Voltage was applied between the specified contacts and, as can be seen, the connector far surpassed the requirements.

These stringent specifications must be met not only to ensure safety but to allow sufficient insulation between pins for soldering large wire. Pin buckets are large enough to hold 12AWG cable, and the insulation had to be strong enough to ensure that the cable could be soldered into place without loosening the contacts.

Basing part of the contact design on the experience gained from Cannon's line-voltage connector (LNE), the company is able to easily pass safety-finger probe tests as required in some specifications. To prevent confusion, the configuration is reversed from the LNE version (see Figure 1).

In addition, to fulfill the requirements sometimes needed when using shielded

Item	Material	Finish
Shell:	Diecast zinc alloy	Satin nickel
Contacts:	Copper alloy, machined	Bright tin
Insulator:	Socket: UL94V-0 polyester	Color: blue/white
	Pin: UL94V-0 polyester	Color: blue/white
Cable boot:	UL94-HB elastomer	Color: blue/white
Maximum ratings: 200 volts rms and 25 amps rms.		

Table 1. *Material specification for AXR connectors*

speaker cable, the drain contact is connected to the shell. (Drain contacts and shielded cables are necessary in situations such as munitions assembly, where static discharge could have disastrous results; the drain pin is always the first to make contact with this connector.)

Contacts themselves are machined out of high-conductive copper alloy and coated with bright tin plating. Gold contacts are also available for audiophile users. Remember that although gold is more resistant to corrosion it will slightly derate the current capability.

At the time of writing, only one U.S. manufacturer, Meyer Sound Laboratories, is using the connector. MSL has incorporated the AXR series into the new 500 series loudspeakers. Orders also have been placed by the U.K.-based Theater Technicians Group.

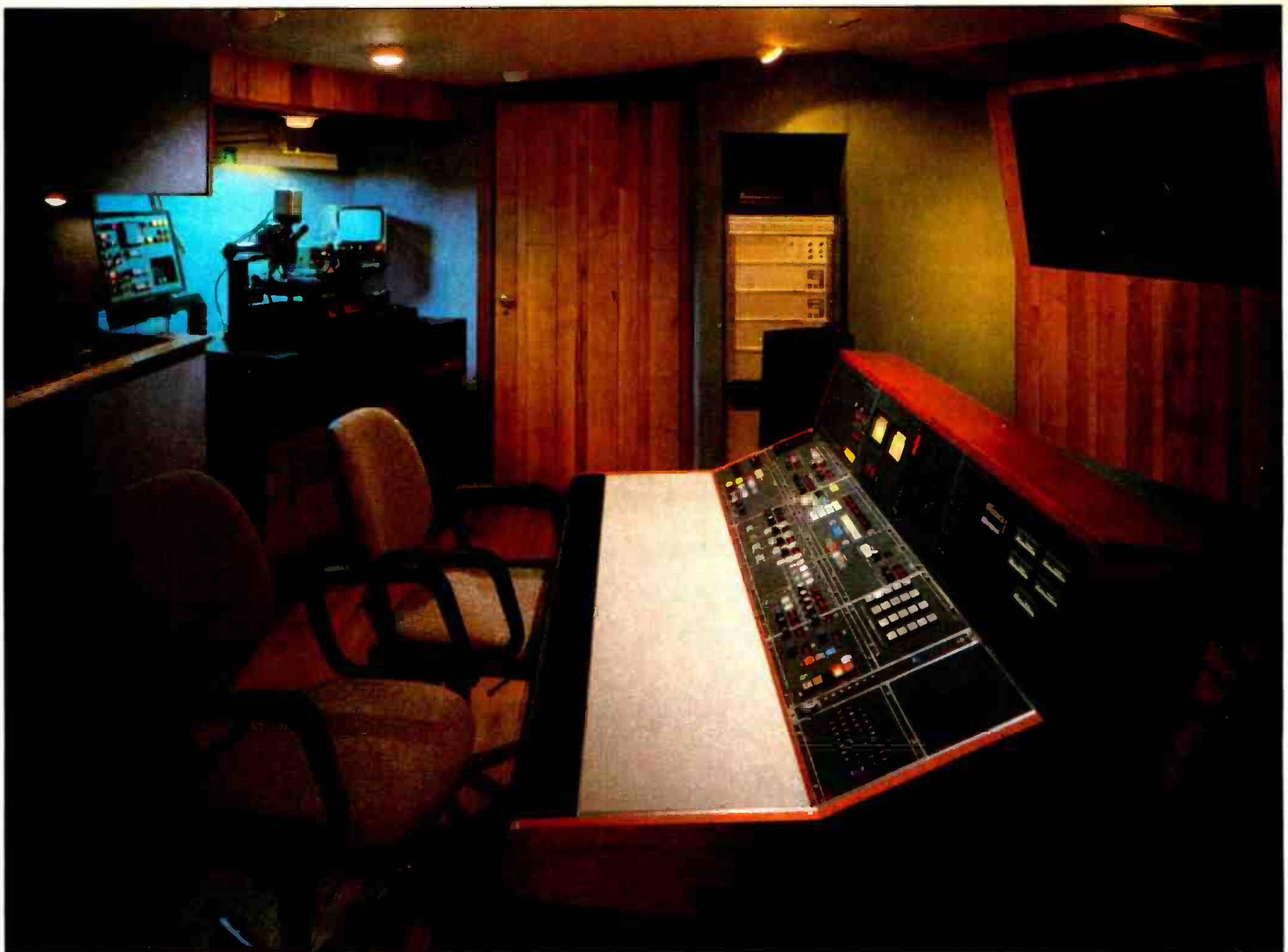
ITT-Cannon has begun introducing the new connector to companies and associations throughout Europe and the United States. It is reported that the British Broadcasting Corporation is interested in establishing a new standard, and support for the connector seems to be growing. R·E/P

Editor's note: RE/P is interested in your reaction to the possibility of standardizing a commercial loudspeaker connector, and would be happy to forward questions and comments to ITT-Cannon. Address correspondence to Bob Hodas, c/o **RE/P**, Suite 220, 1850 Whitley Ave., Hollywood, CA 90028, or through IMC E-mail to REP-US or Hodas-US.

This article is presented as an informational service to our readers. No endorsement of the technology or product is given or implied by **RE/P** or Intertec Publishing.

Contacts	Spec voltage applied for one minute	Actual voltage applied for one minute	Breakdown voltage
P-N	1,000V	4,000V	4,400V
P-N-D	1,000V	2,000V	3,400V
P-N shell	3,500V	4,000V	4,400V

Table 2. *Electrical performance of connectors*



Tape One's Disc 1 mastering room features a Neve DSP all-digital console.

Digital Audio Post-Production

By Bill Foster

The philosophy behind the creation of an all-digital DAPP facility for disc and CD mastering.

Perhaps I should begin by stating, surprisingly you may think, that I don't necessarily advocate digital recording as the *only* way to make records. Analog still has a long way to run and, with ½-inch stereo machines capable of respectable response to 30kHz and beyond, in many ways they outperform their digital counterparts.

Having said this, however, there is no question that ultimately all recording and production consoles will be digital. The movement toward digital technology since the launch of Compact Disc makes this inevitable. The real question is: When will the change take place?

A year ago I would have put my money on 1988 or 1989. Now, with the introduction of assignable analog consoles, enhanced noise-reduction systems and a new generation of analog tape

Bill Foster is co-owner of Tape One Studios, London, a DAPP facility equipped with a pair of all-digital consoles.

Closeup of the DSP disc-mastering console in Disc 1, which enables all front-panel settings to be stored and recalled for subsequent recuts from digital master tapes.

machines, the switch may be delayed until the early or even mid-Nineties. The diehard analog camp will continue to transfer direct-to-disc for years to come; and I for one hope they do. But for CD mastering, there will have to be an analog-to-digital converter somewhere, so why not early in the chain?

The obvious advantages of digital recording include freedom from tape noise, print-through and generation loss; the disadvantage, when using conventional analog desks, is the repeated use of A/D and D/A converters. Minimize the number of conversions in the system, and you greatly reduce the risk of compromising audio quality.

Digital options

Back in 1982, this sequence of reasoning prompted Tape One to look seriously at digital consoles and, in 1983, to order two all-digital audio post-production consoles. What we have learned from the experience is that today no new item of digital-based equipment is ever ready when the manufacturer estimates it to be. The delays we faced were largely due to the enormously complex software involved, and everyone's underestimation of how long it would take to eliminate the bugs.

Since being installed, our two systems—one intended primarily for CD master tape preparation, and the other for conventional disc mastering from digital master tapes—have proven to be incredibly reliable.

The first, and smaller, digital console was commissioned in May 1984, and has suffered only a couple of faults since then. By far the biggest problem we encountered was sporadic clicks on the audio, which we traced to the electronics operating temperature. Because the system is compact, the electronics are housed in a rack under the control surface. During sessions involving four or more people, as the temperature of the room climbed toward 70°F the clicks began. We solved the problem by moving the electronics rack out of the room. This proved to be a far from trivial task, because it required the building of an additional remote power supply for the control console.

Solving the problem also made us rethink the design of an air conditioning system planned a new area that was to house the larger disc-mastering console. A separate machine room had already been planned, but the air conditioning system was subsequently modified to ensure that the temperature in that room did not exceed 64°F.

It seems that all digital hardware needs to be kept cool, and even our PCM digital processors and editing system have been more reliable under these conditions.

Console topographies

Why is our disc-mastering console so large if there's little circuitry in it? This is where the man/machine interface comes into play. We didn't feel that mastering engineers could yet relate to miniaturization at the control surface. Our guys are good, but to ask them to change their entire approach to the operation of cutting a vinyl record, and then present them with a totally alien board would not be conducive to good mastering.

It is for the same reason that we have included conventional analog metering on the desk. An experienced mastering engineer will know the shape of a record groove by watching the response of moving-coil VU and/or PPM meters—which is not so easy to achieve with unfamiliar digital bargraph type displays.

A second CD preparation area, Digit 2, features an analog DDA console, a Harmonia Mundi bw102 digital processor, a Philips PQ code editor, a Sony DAE-1100 editor, PCM-701 digital audio processor and companion VCR recorder.

Detail of DAE 1100 and portion of DDA console area.

And it's not only the mastering engineers who we had in mind; many producers are also engineers, and they don't like working with equipment they can't relate to.

The next generation of consoles will very likely be smaller than current designs, a trend that has already started with some assignable analog desks. Students who are now taking their degrees on Macs will probably have no problem with handling a 48-track mixdown on a QWERTY keyboard. For us "geriatrics" from the Sixties and Seventies, however, it's got to be knobs and switches.

Perhaps one of the biggest difficulties facing Tape One back in 1983 was drafting a suitable specification for our two consoles. Technically, of course, we all know that digital equipment is flat from 20Hz to 20kHz, with a noise floor of 90dB or better. Well, as it turns out, the situation isn't quite like that.

There are such matters as headroom to be considered. If you want 12dB of headroom, you can only have a noise floor of around 78dB, and so on.

Fortunately, in post-production you don't need much headroom because you hope the master tape already has controlled peak levels. Digital is very unforgiving—peaks must be 100% modulation because, at 101%, there can be an *awful* lot of distortion.

Our smaller CD mastering and transfer console has very basic facilities. It was designed to operate at a fixed 44.1kHz sampling frequency, have PCM-1610/30 compatible inputs and outputs, and offer EQ, limiting/compression and gain control. The desk does this very effectively, and allows final correction and "tidying up" of digital masters, usually for CD production. The board has no fader automation or snapshot storage facilities, the operating software being restricted to just real-time signal processing tasks.

The larger disc-mastering console is a completely different animal. Heavily software-based, it is capable of up to 40 different control-surface set-ups, with all EQ, level, signal routing and lathe settings being stored on floppy disk. We didn't get the specification exactly right at the start, however, which is one of the key problems with digital consoles.

If you want to add, for example, an insertion point to an analog desk, you take a jack socket, some cable and a soldering

A custom-designed 2-channel DSP console in Digit 1 enables PCM-1610/30-encoded tapes to be re-equalized and processed prior to CD manufacture. Tape One engineers worked closely with Neve on the development of the currently available DTC-1 digital transfer console, which features two digital and one analog pairs of stereo channels for real-time crossfading; it is also compatible with 1610/30 and AES/EBU direct digital input and output formats.

iron, and hand wire the extras onto the PC board. Not so with a digital console. As well as the hardware involved, the software has to be modified, which may involve more than just a couple of lines of code and new master EPROMS.

Luckily, we got most of it right at the start and, after consultation with the manufacturers, have left the required modifications until now so they can all be done at the same time. The mods will mean taking the room out of service for a week, which, with the current booking level, is a pain in the neck!

Cost-effectiveness

On the subject of bookings, it would be predictable for me to say that the new Disc Mastering Suite is at capacity and they're cuing 3-deep at the door. In fact, although the new room is being heavily used, some analysis of our overall capacity and usage pattern has to be made to obtain a true picture of the digital console's values to our facility.

Before the new digital suite opened, we were already operating two conventional disc-mastering rooms. One housed a Neumann VMS-80 lathe with an SP-79 console; the other a VMS-70 and a custom-built mastering desk. By sharing Studer DAD-16 digital delay unit to provide preview outputs, both rooms had the facility to master from Sony PCM-1610 or -F1 encoded videocassettes and Mitsubishi X-80 format, but not simultaneously.

When the suite was commissioned, we closed one of our existing rooms, mated the VMS-80 with the all-digital console, and coupled the VMS-70 to the SP-79. Although our overall capacity has not increased as a result of these changes, both rooms can now handle digital master tapes at the same time, the digital desk having its own built-in preview system.

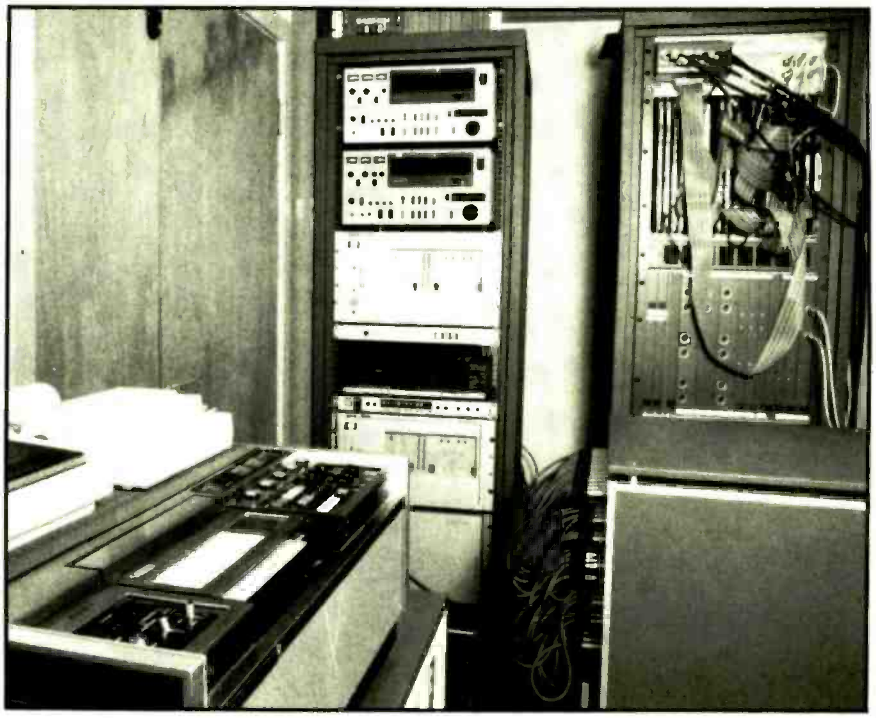
The digital console generates revenue in two ways. In addition to increasing our client base by attracting producers who have not used the facility before, we are also able to achieve higher earnings from selling the same quantity of studio time. At present, rates are 100% up on analog and approximately 40% above what we charge for digital mastering via an analog cutting desk.

One direct result of mastering via the digital console has been a substantial increase in revenue from tape copying. Almost all producers using Tape One will now order a PCM-1610 production copy to enable easier CD mastering. These production "masters" are then also used for export copying. Even if the end result is to be analog, because the source is digital an elevated rate is charged—usually 50% more.

None of the rates we charge in the United Kingdom are as high as those levied by our U.S. equivalents. This situation will be hard to change until more mastering rooms and studios become aware that the long term effect of charging too little for their services will be a lack of finances to invest in the increasingly sophisticated and expensive technology that will soon be mandatory for every studio wanting to survive into the next decade.

At Tape One we're fully aware of having effectively bought 1983 technology that's likely to be superseded within a fairly short time span. Because our new consoles are basic on a 24-bit bus architecture, we hope it will be possible to increase the resolution of the A/D converters when the technology is ready.

It is still likely, however, that another digital console will be required for the disc mastering suite before the cost of the



A central machine room houses the electronics racks for the two digital consoles (right), PCM processors and U-matic VCRs.

existing one has been written off. Fortunately, there are plenty of other production areas, such as copy rooms, where it can be "pensioned off." The console is, in effect, a standard 2-channel mastering desk; only the preview-signal delay would become redundant.

Commitment to digital

In the field of digital audio post-production we make the assumption that, by supplying a digital master tape, the producer or artist has made the commitment to digital. We would not normally convert an analog tape for processing if the end product is to be analog.

Digital signal processing offers some very significant advantages, the primary one being that if you dial up, say, a +6dB boost at 8kHz, you'll get exactly that, on both channels, in-phase and without any ringing or other side effects. Set the compressor section to 10dB of limiting and, depending on the selected attack time, that is what will happen—no overshoot, no pumping.

One area requiring a degree of caution is that of level changes. Many engineers who have used a well-known digital editing system claim that there's a change in sound when you deviate from unity gain. Of course, that unit uses 8-bit control of a 16-bit signal, whereas our digital consoles have a 24-bit operating system (up to 28-bit at critical places). Whatever the system being used, as soon as gain adjustments are made in the digital domain all kinds of things start happening: LSBs get truncated, digital dither may be applied and so on. Fading out a signal to infinite attenuation is, in fact, one of the most critical tests of any digital processing system.

But don't let these horror stories put you off. A/D converters, in my opinion, do far worse damage to the signal, and should be avoided at all cost. Unfortunately, no manufacturer has yet come up with an echo or reverb device equipped with AES/EBU digital inputs and outputs so, for the time being at least, the processed signal must pass into the analog domain.

Digital consoles have one great advantage over their analog counterparts: the ability to memorize and store all setup information on floppy disk or a similar storage medium. Because digital processors are effectively drift-free, it is possible to recreate a particular setup at any time in the future. In the mastering field this capability is invaluable.

The main reason why full automation wasn't included on our consoles is due to both cost considerations and because, with the exception of VCR-based systems, time code is not usually available on stereo master tapes to drive the automation system. However, with the additional data storage capacity of the new digital 2-track tape machines, second-generation consoles are likely to be fully automated.

Cost considerations

One of the questions I'm most frequently asked is: "How much did it all cost?" Well, I've never given a straight answer yet (and I don't intend to do so now!) but the formula goes a little like this:

A digital console will probably cost, at the outset, two and a half times that of its analog equivalent. Actually, "equivalent" is totally the wrong term to use here. Although a digital console will have all the regular features of a conventional desk, it will also offer a great deal more at no additional cost. Total assignability and recall come free with the

Hands-On

Aphex Studio Dominator

By Steve Keating

For the recording, audio production and broadcast industries, successfully controlling transient audio signals is a challenging task. Equipment designers and manufacturers have long been struggling to produce devices that would limit peak waveforms without audible alteration of the original program-material dynamics.

The approach taken by the Aphex Systems in its Studio Dominator is revolutionary, efficient and more than satisfactory. The unit is promoted as an "intelligent" 3-band peak processor with a proprietary circuit that varies the threshold for limiting, a technique that differs from traditional designs that are dormant past the input threshold level.

Steve Keating is chief engineer of KMET, Los Angeles.

Aphex claims that its Transient Enhancement Circuit (TEC) increases the perception of transients, while maintaining absolute peak limiting. This capability is most important for users who need to ensure that levels do not exceed a maximum level, including such applications as disc mastering, microwave links, tape duplication, digital recording and broadcast.

In addition to offering tri-band peak control, the Dominator can also be configured through installation of appropriate, plug-in option cards to serve as a primary peak limiter for FM and TV stations requiring pre- and de-emphasis protocol, and a special output card that produces a matrixed output for use in AM stereo.

Published performance specifications include a respectable maximum total harmonic distortion of 0.1% at 15dB of limiting of a 20Hz sinewave. Noise, unweighted 20Hz to 30kHz, is listed at 80dB below peak output, with a frequency response below threshold of 20Hz to 50kHz, ±0.2dB.

Compression ratio is specified as being "infinite above threshold." Wide-ranging input signals can be accommodated via the use of a 2-position internal slide switch that selects either high or low sensitivity. The overall input level range is a maximum of 20dB above a normal program level of nominal −10dBm (low), or a +4dBm (high).

Output level is front-panel adjustable in 1dB increments from −3dBu to +20dBu. Both input and output circuitry use differential-balanced, high-performance package, fader automation being the only extra. In fact, by the time you had added all the operational features provided by an all-digital design to an analog console, the price would be getting up around the same level.

One of the main drawbacks of a digital desk, especially a multichannel one, can be summarized in two words: sampling rate. It's all very well having a console that will accept a direct digital input from your 24- or 32-track, but what do you do if the tape to be remixed was recorded at 44.1kHz with pre-emphasis on some tracks only? The answer is either buy 24 channels of sampling rate conversion and emphasis switching, or go via analog through 24 D/A and 24 A/D converters.

To be realistic, it is almost impossible to distinguish between a signal that has gone directly from one digital device to another, with one that has been converted. But, repeat the exercise a few times via record, track-bounce, remix and mastering stages, and you may begin to hear strange things, especially during low-level passages.

To make the all-digital studio totally viable, we need one standard sampling rate—which needn't be 44.1kHz, because this conversion can be done at the CD mastering stage—and the number of bits should be increased, in my opinion, to at least 20. Having achieved this, with the increased signal-to-noise ratio thus gained, perhaps pre-emphasis can be done away with once and for all.

It is not my intention to dwell at length on digital standards, since this is a subject worthy of volumes in its own right. The plethora of different digital formats, sampling frequencies and, shortly I expect, bit rates, make the capital investment required to operate a good DAPP facility similar to that of a multiroom studio complex. It's no wonder, therefore, that it can often be more costly to book an hour at a DAPP room than at a 24-track studio.

Format standardization

It is unlikely that we will see any serious moves toward standardization before the end of this decade. In fact, I'm not sure that it's even desirable. Standardization is often only a euphemism for stagnation. There is no way that I for one, want to see the recording industry locked into 44.1kHz, 16-bit for the next 20 years just because the CD manufacturers find it cheap to build their players that way.

OK, one or two studio equipment manufacturers may go to the wall, and with them some studios who pick the wrong standard. So be it.

With a little bit of care in purchasing and the charging of realistic rates, thereby writing off a piece of equipment before advancing technology writes it off anyway, most studios should be able to survive the transition to digital.

Of course, not every studio will be able to afford each new piece of equipment that appears on the market, and this is where the DAPP facilities will come into their own, by offering conversions between PD, DASH, PCM video-based and any other fancy format that the backroom boys dream up. Such facilities are already available for 2-track recordings, and multitrack standards conversion must surely follow. Switching digital formats will soon be as easy as changing NTSC video to PAL or SECAM, but without the quality loss problems of the latter.

All things considered, the future for DAPP is beginning to look very promising indeed. R·E/P

Additional User Comments

*Because of the unit's potential application on a wide variety of audio production sessions—specifically where absolute peak limiting is necessary—**RE/P** gathered additional comments from users working in music recording, commercial production, tape duplication film sound re-recording, audio-for-video and disc mastering.*

• Russ Terrana, *chief recording engineer for Hitsville Recording, West Hollywood, CA: "Because I can't really hear it in the signal path, the Dominator is unique for mixing sessions. It is a 'transparent' piece of equipment, and doesn't make the resulting signal sound processed or squashed.*

"Because it can control the amount of limiting and compression used on low frequencies, the unit adds a lot of 'punch' to the kick drum; previously, I had to punch in a lot of level to get that same kind of effect. Because I can adjust the limiter functions to limit a signal to an exact point, and let the remaining signals pass unaltered, I can eliminate any pumping effects."

• Andy Morris, *chief engineer of Buzzy's Recording Services, Hollywood, CA, and whose credits include 10 years experience in commercial and audio post-production: "With the Dominator, it looks as if you could optimize headroom per storage format. Because there are different rise constants and speeds for various formats like audio cassette, 35mm single-stripe mag, or 30 ips ¼-inch, these formats have unique high-frequency limitations before the program material sounds 'splashy.'*

"For an audio cassette master, you have to consider transients that can saturate the tape and cause distortion. With the Dominator you can maintain headroom and *sound quality, and whatever coloration that will occur. And, because control of release time is afforded, I can perform frequency-discriminating limiting.*

"In our post-production transfer room, the unit doesn't induce a lack of 'sparkle' or 'openess' nor does it dull the signal. Compared to the other limiters, it offers lower distortion, better signal-to-noise ratio, and more 'openess,' which means less smearing from distortion and excessive phase shift."

• Richard Portman, *independent re-recording mixer: "I use two units and split the stereo pairs into four mono channels for use during final layback onto a 4-track recorder. At Disney Studios I recently mixed a show using the Dominator—a rap music film from Jamaica, with music mixer Rob Fraboni. Rob likes things loud! The problem with loud program material is that I'm always asked to exceed what I know I can lay back onto mag film. Consequently, I am always looking for a limiter that will let me have the signal loud, yet 'crunch' it so that it goes down on film without sounding 'muddy' or 'trashy,' as if heavy limiting was used, thus losing sound clarity.*

"In my opinion, the Dominator accomplishes this: It crunches the sound without it appearing to be crunched, even at 20dB to 30dB [gain reduction]. It can knock the hell out of overall level and hold it within a limit, yet not really alter the signal's dynamics.

"The unit also allows me to keep overall recording energies within a predefined spectrum—levels I know I can get through an optical camera [to print mono Academy or Dolby Stereo optical soundtracks]—particularly in that peak area that shows up on really fast-acting meters, but which the ear can't hear.

"In essence, the optical-camera to film relationship is directly analogous to a disc-cutting lathe/lacquer relationship. Sometimes, there is so much signal range that you can't print or cut that signal onto a respective format. But the Dominator enables me to make the program material as loud as the producer wants, without being hampered by the camera."

• Brian Gardner, *a disc-cutting engineer at Bernie Grundman Mastering, Hollywood, CA, and whose recent credits include Janet Jackson's* Control, *Stevie Wonder's* In Square Circle, *and Barry Manilow's* Searching For Love: *"The Dominator doesn't destroy transients, yet at the same time it still gives a compressed, solid sound.*

"I especially like the crossover feature. With selections of certain limit ranges, many times all that is required is limiting an out-of-control vocal—not so much the low- or midrange frequencies.

"Currently, we place the unit between the tape machine and the console. A lot of times, I'm reluctant to use any limiting at all on an out-of-control vocal. The Dominator allows you to stop the signal against a brick wall.

"It seems to work best when used subtlely. I use it mostly to compress a frequency into a range that vinyl can handle. Plus, it sometimes tightens the drums, and better places bass frequencies in a phantom center."

• Jerry Clemens, *Modern Videofilm's manager of sound services, Hollywood, CA, and whose recent sessions included audio-for-video sweetening for a Patti Labelle TV special, a Ford Theatre HBO special* and Solid Gold. *"I work on soundtrack preparation for home videos and in-flight movies. For the latter, because of the high ambient noise levels in aircraft, I need to reduce the dynamic range on a given track. But I have to accomplish this without a pumping effect, or flattening the sound too much.*

"With the Dominator and [Aphex] Compellor combination, I can bring the track down to 8dB or 9dB of dynamic range, which is all you can hear on an aircraft. Plus, the noise floor doesn't fluctuate. Sure it flattens out some sounds, but that's to be expected from such high limiting and compression ratios.

"My only objection about the unit, and this applies to both the Dominator and the Compellor, is the lack of contrast between the lettering and its background color. The legends are hard to read in lower light levels."

Technical Specifications

Frequency response: *below threshold: 20Hz to 50kHz, ±0.2dB; above threshold: 20Hz to 50kHz, ±0.5dB.*
Total harmonic distortion: *below threshold: typically 0.1% or less, 20Hz to 20kHz; above threshold (15dB limiting): 20Hz, 0.1%, 1kHz, 0.04%; 15kHz, 0.017%.*
Noise *(unweighted, 20Hz to 30kHz): 80dB below peak output level.*
Maximum output level: *20dB above normalized input setting of −10 or +4.*
Peak output level: *adjustable in 1dB steps from −3dBu to +20dBu.*
Compression ratio: *infinite above threshold.*

Circle (164) on Rapid Facts Card

operational amplifiers. Balanced bridging input impedance is 160kΩ, and balanced output source impedance is 20Ω; the latter is capable of driving almost any type of conventional load.

Primary power is through a standard IEC 3-prong receptacle with accompanying fuse holder. Voltage range is either 100V to 120V ac, or 220V to 240V ac; power consumption is expectedly minimal. A note of caution: Due to the unit's 1¾-inch, single-rack height package, a fair amount of heat is generated during operation. As a result, sufficient ventilation around the top and bottom within a rack is recommended.

Front-panel controls are minimal, but effective in shaping the final product. The input *drive* control adjusts the degree to which limiting will occur for a given input level. Increasing the control increases the amount of limiting, but doesn't alter the peak output level. Most limiters use input threshold controls to set the degree of limiting activity, and subsequently decrease the overall output amplitude when greater amounts of limiting are desired.

The LF EQ control varies the amount of low-frequency drive to the low-band limiter. The LF crossover switch determines the highest frequency this circuit will act on: one setting is 80Hz, the other is 160Hz.

The HF EQ control provides the same function as its low-frequency counterpart, except the HF crossover switch determines the lowest frequency at which this circuit will operate: 1.7kHz or 4.5kHz.

The mid-band frequencies are automatically controlled by the content of the input program material.

Switching on-line the unit's limiting function is done with the *process* switch. Setting the process to its off position does not disable the unit's equalization sections, only the amplitude control circuitry. A 6dB boost or cut of the low or high bands is possible regardless of the process switch position, so long as the program material level is below the threshold of limiting; above threshold, and EQ boost will be reduced as each band approaches maximum limiting.

Much like conventional limiters, the *release time* control sets the release time of the limiting section from moderate to extremely fast.

A *stereo tracking* switch links together the control sections of the left and right audio channels, resulting in equal gain reduction for both channels regardless of the channel with a greater signal level present.

The transient enhancement circuit is claimed to "restore articulation to the audio by manipulating the below-threshold audio envelope, without changing the average power content or peak amplitude of the audio waveform. It can be used to bring out more 'punch' in an audio source, or to improve the dimension and imaging qualities of the audio. A noticeable increase in the perceived loudness of the audio is an additional benefit of the TEC process. Depending on audio source type and quality, the effect of the TEC may be subtle or quite noticeable."

Evaluation procedure

After receipt of a stock unit for evaluation, I bench-tested the unit for basic performance. Frequency response was found to be well within the published parameters, as was harmonic distortion and noise floor. However, most processing units that control the amplitude of energy contained within the audio spectrum exhibit different performance characteristics when dynamically altering complex waveforms generated within the bandwidth of 20Hz to 20kHz.

It is difficult to accurately measure the performance parameters of processing amplifiers when they are acting on program material, especially music containing extended fundamental and harmonic waveforms that may be symmetrical, or quite asymmetrical, as well as substantially dynamic in energy content and duration. A high quality monitoring system and a good pair of ears are the best final reference.

Feeding contemporary music into the Dominator with control settings recommended in the instruction manual's condensed set-up procedure produced noticeable results when the input and output signals were compared from a directly connected monitoring system.

The density expected from the action of the unit was evident, but the often audible manipulation of the signal amplitude was minimal, even with the controls adjusted for above-average processing levels.

Installation of the unit into a primary on-air chain to evaluate its performance in the real world was simplified through the provision of a second processing loop, identical to the primary chain with regard to incoming signal and output loading. After careful adjustment of the operating controls, the Dominator was put on-line and evaluated as an active element of the total processing scheme by listening for extended periods on a variety of receiver types.

Minor control adjustments were made throughout the 24-hour evaluation period, and several individuals with keen aural sensitivity around the station were asked their opinion of the overall sound. All were favorably impressed with the "texture," "density," and substantial loudness of the air sound, with few comments regarding the presence of any undesirable "foot-prints," or processing "by-products".

The front-panel capability of switching the limiting activity on- and off-line allowed for instantaneous comparison between the effect the unit had on the program material versus the unprocessed signal. Apparent loudness without any "graininess" or "coarseness" was primarily noticeable, as well as the maintenance of consistently high-quality reproduction of program dynamics.

I would recommend the Dominator in any application requiring consistent, high-quality peak-level control of wide-range audio program material. R·E/P

NEW

Vega PRO wireless at an affordable price.

Cetec Vega's famous high-quality "PRO" wireless microphones are now available in new, highly affordable versions. Enhance your sound installation with the new PRO 1-B and PRO 1-H systems. Ideal for all of your professional audio applications, these systems are great for broadcasting and entertainment applications, as well as for use in audio-visual systems for industrial seminars, church and school sound systems, etc.

The PRO 1-B system consists of the Model T-37 bodypack transmitter and Model R-31A receiver. The PRO 1-H system consists of the Model T-36 hand-held transmitter and R-31A receiver.

Both systems feature DYNEX®II, Cetec Vega's advanced audio processor for the highest signal-to-noise ratio, widest dynamic range, and most natural sound.

R-31A PRO Receiver

The R-31A PRO receiver features two easy-to-read LED bargraph displays—one for RF signal level and the other for audio level. The receiver is extremely sensitive, highly selective, and very stable. Either line-level or mic-level outputs may be selected through the rear-panel XLR connector. Line-level output is also available from the rear-panel terminal strip. The receiver can be powered by either AC or external DC.

T-37 PRO Bodypack Transmitter

The T-37 PRO bodypack transmitter accepts all positive-biased and most negative-biased electret lavalier mics via an easy-to-use mini 4-pin XLR connector. Conventional panel-mounted mic on/off and recessed power on/off toggle switches are featured. The ultra-rugged case stands up to hard use.

T-36 PRO Hand-Held Transmitter

The T-36 PRO hand-held transmitter uses the popular Electro-Voice BK-1 ("Black Knight") condenser element with an attractive black windscreen. Housed in an attractively contoured black case, it has Cetec Vega's patented internal dipole antenna. An audio gain control and power and audio on/off switches are conveniently located on the bottom.

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Equipping an Electronic Music Production Facility

By Paul D. Lehrman

With electronic music production assuming greater importance, what features and functions should a recording or production studio look for in MIDI-equipped synthesizers, sequencers and outboard effects equipment?

It's been only three years since MIDI was introduced, but already one would be hard-pressed to find a commercial recording or production studio anywhere in the world that doesn't use the standard in one form or another.

MIDI sequencers offer greater accuracy and control than multitrack tape, and MIDI hardware is beginning to assume many of the mixing and processing chores traditionally handled by discrete equipment. All of these functions allow studios to work faster, more efficiently and more cheaply.

How involved a facility becomes with MIDI depends primarily on the type of music it produces. A studio that concentrates on classical or acoustic music will have minimal use for MIDI equipment, while one that works primarily in music for visuals or dance rock can take great advantage of MIDI computers and synthesizers. Determining how it can help your studio requires a hard look at your methods and clientele, and deciding what the investment of time, money and energy required to get involved with MIDI is worth.

Even when a studio has gotten past the question of whether to get involved with MIDI, there is still plenty of confusion as to how best to go about it. This writer gets daily calls from large and small studios that are wondering about the best synthesizer, the best computer, the best software and how the heck are they going to learn all of this stuff fast enough and well enough to make it pay off.

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

Finding the space

The first concern a studio should have is where it's going to put everything. Installing as many of the MIDI toys—keyboards and computers—into the control room as possible will considerably improve communication and speed. The engineer's mix will be the same as the musicians', and headphones, intercoms and talkback problems can become a thing of the past.

If your control room is small, a more radical solution may be called for. Some studios find they need to reverse their floor plan completely, by putting all of the electronic gear in the big room, where the mics and the Steinway used to be, and leaving what was the control room empty for recording vocals and those occasional acoustic instruments.

Of course, live recording areas and control rooms are designed for very different purposes. If the acoustics of the larger room cannot easily be toned down, this idea may end up being more expensive than it's worth. Knocking out the walls to the control room to make it bigger can seem even less attractive, but there is an alternative.

For most sessions, MIDI programming can be handled by one or two people. Another room, which can be quite small (like your current tape library), can be dedicated to nothing but MIDI equipment, with audio and MIDI tie lines connecting it to the control room. A large console is not necessary—synth programmers generally have little use for 6-band parametric EQ, or four separate cue mixes—nor are tape machines and long-throw monitors. Because most MIDI equipment is now rack-mountable, it's possible to squeeze a lot of equipment into a minimum amount of space.

An added bonus to this plan is that the small room can be rented out by itself as a "pre-programming" room. The cost to the client can be significantly lower than that of the main room, and if it's set up correctly, you can handle two separate sessions simultaneously—just like building a second studio, but at a fraction of the expense.

Wiring and routing

In the past, a typical studio keyboard setup would include a stand or two laden

Control Room A at Dallas Sound Lab, Irving, TX, features (from left to right): a host of MIDI-capable keyboards from Kurzweil, Yamaha, Oberheim and Sequential: a 48-voice NED Synclavier with time code/MIDI interface; Linn 9000 and Akai S612 MIDI sampler; and Apple II personal computer running Yamaha DX-Pro software. All MIDI equipment is centered around a fully automated Solid State Logic SI6056 console, linked to a Sony DASH-format PCM-3324 digital multitrack on an Otari MTR-90, with time code interlock provided by an Audio Kinetics Q.Lock 3.10 synchronizer.

with instruments, connected by a rat's nest of cabels and direct boxes. This was useful when keyboards were optional (or extra-cost) on a session, or when they were constantly going in and out of the studio. When you're using MIDI, however, keyboards are the mainstays. You wouldn't let your console wiring hang all over your control room floor; keyboards should be afforded the same respect.

Permanently wiring and mounting DI boxes is one step. When keyboards have to be taken in and out (which will still happen), use short cables to connect them to the permanent fixtures. Likewise with MIDI cables: fixed MIDI receptacles on the wall will do much to alleviate clutter and possible disaster. Also be sure to keep plenty of slots open for client's equipment.

MIDI cables should be kept relatively short. They are essentially unbalanced and run at high transfer speeds (of the order of 31kHz). At the same time, however, because MIDI lines carry digital data they are relatively resistant to noise and hum, meaning that runs of 50 feet are possible in a pinch. For longer runs, Thru Boxes can be employed at regular intervals.

The layout of the wiring should allow for as few MIDI delays as possible. Thru Boxes, because they reprocess the data, can introduce delays, as can chaining MIDI lines through one synth to another. The best solution is a star network, in which the path to every MIDI device is not necessarily as short as possible, but instead is the same effective length. On the other hand, MIDI paths going *to* a recording device should be kept delay-free, to avoid timing problems while laying down tracks.

MIDI patch panels are an idea whose time has not come, possibly because those 5-pin DIN connectors do not tolerate constant pulling in and out as well as ¼-inch jacks.

MIDI patch panels are an idea whose time has not yet come, possibly because those 5-pin DIN connectors do not tolerate constant pulling in and out as well as ¼-inch jacks. But having some kind of MIDI routing network is essential. Clients will want to be able to use various devices as master controllers, and even though some synths may never be used as controllers, if you want to load patches into them, you will need a 2-way (MIDI In and Out) path. You can build your own patch panel, using either DIN connectors or something more robust (just remember that you cannot mult a MIDI line simply by wiring a few jacks in parallel), or you can purchase one of the several switch boxes available.

Having the right controllers

Musicians, as you already know, have very individual tastes. Some will be perfectly happy playing everything on "old faithful," while others simply *must* have a weighted keyboard, or a true piano action, or will even like the miniature keys of certain models. You should have as many of these devices available as you can afford, along with as many controller toys (foot pedals, breath controllers) as possible. For clients who will only touch a real acoustic piano, you will have to invest in either a MIDI-modified piano, or once of the new ones with built-in MIDI sensors.

Other types of MIDI controllers are now becoming common (read: not too expensive). If your clients are not all keyboard whizzes, a set of MIDI drum pads, a pitch-to-MIDI converter, a universal guitar converter, or even one of the new MIDI mallet controllers can be worthwhile.

Also useful are some of the new real-time MIDI processors and "mappers," which can add to a player's arsenal of controllable functions, or change one type of function into another, like using a foot pedal to control pitchbend. (The more elaborate, dedicated master keyboards have many of these functions built-in.)

Programming MIDI tracks does not always have to be a solo activity—some sequencers are now capable of recording simultaneous input from several MIDI sources. If you have one of these devices, a MIDI mixer (not to be confused with a MIDI-controlled audio mixer) can allow you to record a complete ensemble of players in one pass.

Synthesizers and processors

Popularity, price, usefulness and flexibility are the determining factors in which synthesizers a studio will choose. The sword of popularity, however, cuts two ways. Synthesizers that are extremely well-known to clients can be a good choice, but so can those that are not as common.

Although the latter may be beyond the price and complexity level that most clients can afford for their own collections, there's no reason why your studio cannot support them. A few exotic, high-end synths in a studio's arsenal (if you or your staff know their way around them) can attract clients just as effectively as a set of rare tube mics, or the latest digital reverb.

Your synth collection must be versatile: analog, FM, sampling and other types of synthesis should all be well represented. Synthesizers are sounding better and more realistic all the time—with a little thought, you can assemble a synth collection that can replace 99% of what formerly had to be recorded acoustically. Because few of them are very programmable, the more drum synthesizers you can supply the better; clients will also want as large a pallette of drum sounds to choose from as possible.

Whichever types of synthesizer you select, a wide variety of sounds should be on hand for all of them, in the form of RAM cartridges, disks or computer systems with librarian software. Clients don't want to spend all their time (and money) programming your studio's synths, so if they don't have the sound at hand that they want, *you* should. If they have it on a disk, you should have the software that will let them load it. Of course, it's common to be in a situation where no pre-programmed sound seems to fit. If your library is broad enough, however, a client can find something close, and it can then be tweaked to make it work.

Remember also that synthesizers, unlike some studio equipment, are eminently portable. If clients need a particular synth that you don't have (and don't want to buy), they can bring it in themselves, or you can rent it for the session. You may encounter a client who is inseparable from a pre-MIDI synth. Devices that convert MIDI signals to a form that the antique can deal with (and vice versa) can help out in such situations.

MIDI-controlled processors, including delays, reverbs and mixers, can help speed up production by automating the mixing process to a certain degree. If a song demands a gated reverb for two notes in bar 82, it is far easier to program that information into a sequencer (as a MIDI program change) than to worry about getting it right each time you mix. MIDI-controlled audio mixers are still in their infancy, but even automating the channel mutes can save a lot of time further down the road. For programmable processing gear, again, having access to a variety of effects on cartridge or disc will make your clients very happy.

Synchronizers

Unless you're producing totally instrumental music, and your synthesizers can produce any sound you may possibly want, somewhere along the way you will have to marry your MIDI sequences to tape. There are a number of synchronization methods now in use, none of them perfect (see my "Managing MIDI" column on page 10 in this issue) but all useful in the electronic production studio.

A facility whose work is primarily in-house projects can usually get away with using just one method of sync. An 8-track commercials production house, for example, can record FSK sync to drive a sequencer (which handles all the instruments and processing) on track 1, leaving six tracks (don't forget the guard track) for vocals, acoustic guitars, etc. However, if clients bring in their own pre-striped tapes, or want to use sequencers or drum machines that respond to other forms of sync, it's important to have devices on hand that can read and translate many forms of time code. These devices are getting cheaper almost by the week; a good selection should not be beyond the means of any studio.

Of course, these days everyone wants time code synchronization. Properly used, time code can significantly improve the efficiency of a MIDI studio. It can provide instant autolocation and allow multiple devices, both mechanical and digital, to be locked together and triggered with unerring accuracy. Time code readers and generators, like other sync devices, are rapidly coming down in price, but they vary widely in quality and flexibility. Software-based time code devices will soon be coming to market, and offer the best hope. They can be made more user-friendly and flexible than hardware-based units, and are far easier to upgrade.

In the meantime, look carefully before selecting any time code equipment to ensure that it does what you need it to and is compatible with your other equipment. Pay particular attention to the quality of the code being generated; how tolerant the device is of less-than-ideal code; the processing speed (sync delays produced by some units can make life very difficult); and, if you do a lot of tempo changes, how easy those are to program.

Sequencers, computers and software

The sequencer is at the heart of a MIDI-equipped studio. It functions as multitrack tape recorder, editor, orchestrator and even mixer. Although performing musicians continue to debate the merits of computer-based sequencer vs. dedicated machines, for a studio (where portability and roadworthiness is not a concern) the decision is easy. Computers can display large amounts of data on the screen at one time, as opposed to the typical 2-line LCD in a dedicated sequencer, making editing far easier and more intuitive. Because one computer can be used for many different types of programs, having on hand several different sequencers, as well as patch editors and librarians, click-track calculators, and even word processors and spreadsheets can be accomplished with a minimum of investment.

Also useful are some of the new real-time MIDI processors and "mappers," which can add to a player's arsenal of controllable functions...

Your choice of software will be dependent on the type of music you produce. If 90% of your work is dance tracks, a simple recording program with limited editing will do fine. If, on the other hand, you're doing orchestral film scores with multiple patch and tempo changes, loops and transposed repeats, you might want something more sophisticated. Should clients want to program tracks and then print them out for live musicians to play, then music-notation software is called for.

However, none of the notation software currently on the market is very fast; you should examine closely what's available to make sure that it will really save more time than it uses up. Finally, if half of your clients already have a copy of a particular program, you'll be doing them a service by having one too.

If you already have a computer, and don't want to invest in another one, then your choice of which machine to use is already made in large part. But if you're buying your first computer, follow the cardinal rule of computer purchasing: Check out the software first, and when you find a program you like, buy a computer that will run it. The Commodore C64 and C128 and the Apple II family are relatively cheap and very well supported in terms of sequencers and patch editors. Because of their comparatively small memories and slow processing speed, however, they are not appropriate for very large projects, and tend to be a little cumbersome to work with.

The Apple Macintosh and the IBM PC family (plus its many clones) are the big boys in the field, and offer features such as advanced graphics, better interactivity, larger memories (which not only mean that more notes can be crammed into them, but also that editing functions are handled much faster), and higher speed. If you're heavily involved with sound sampling, the Macintosh has the edge these days, because most of the companies developing sample editors are putting them on the Mac. (It's still too early to tell about the Commodore Amiga and the Atari ST1040, because very little software is available for them yet.)

Be careful, when comparing features and prices, of hidden costs. With some computers you can just hook up a MIDI interface (the Atari ST520/1040 even has one built in), put in your disc and fly. Others require both internal and external hardware, graphics cards, extra memory boards, etc.

Installing a MIDI studio is more than just putting in all the right equipment—it's also a matter of finding the right people to run it. Unless there's someone on every session who knows how to use all that technology to save the client's time,

Installing a MIDI studio is more than just putting in all the right equipment—it's also a matter of finding the right people to run it.

it will just make things more complicated and slower. A sequencer is fast only if the person running it knows it well, and can make it jump through all its hoops without having to dig into the manual every five minutes. A huge library of synthesizer patches is only as useful as the person choosing from it—what patches are available, and how appropriate each of them may be to a given situation. Samplers can do great things, but only if whoever is loading and editing the samples knows what he is doing.

Likewise, the equipment limitations must be well-known. Lush string pads are hard to produce on an FM synthesizer; the programmer who knows that will not waste time trying to coax one out, but will instead use an analog synth. If a sync track on a tape is damaged, an engineer should know whether it's recoverable and, if not, then the best way to get around the problem.

Good MIDI programmers are made, not born. To be an effective programmer, one must be intimate with all forms of music, from folk to funk; have a conservatory-level background in classical theory; be able to recite MIDI codes as fast as you can your children's names; and have the patience of an elementary-school band director. This combination of skills does not come easy, and the ability to make neat sounds from a synthesizer does not necessarily make for an effective studio programmer.

But there are ways that you and your

staff can learn about MIDI. If it's just a question of getting used to a new piece of hardware, taking it home for a couple of days can help. Talking to other users of the same device can also help you pick up hint, tricks and potential pitfalls. Studying manuals can sometimes work, but they vary dramatically in quality—the best ones can teach you a lot, even if you haven't got the unit in front of you, while the worst ones can send you screaming into the night.

If more help is needed, many music schools and studios give instruction in the care and feeding of MIDI, and some dealers and manufacturers provide clinics. Some of these clinics are designed solely to sell equipment (especially if they're free), but others are intended for more advanced users that are dying to know how they can get more bang out of what they've already got.

If you need personalized help, there are a few good consultants who will come into your facility and show you all the neat tricks they've learned. Sometimes it's a good idea to hire a consultant to handle *all* your programming for the first couple of weeks or so, while you and your staff watch and learn how it's done. (You may find that clients are flocking to your door because of the consultant, in which case you might do well to work out a more long-term arrangement.)

Support is also available from other sources. Because of the complexity of MIDI hardware and software, and the pressure on manufacturers to get product out the door before their competitors do, most—if not all—of the devices out there have some bugs that cannot be fixed with a little judicious application of a soldering iron. Invariably, the worst of these bugs will show up just as you're finishing a 2-month project when, all of a sudden, everything locks up. For this reason, many manufacturers offer computer bulletin boards, which users can call with questions or problems. Public database services like IMC-Easi, CompuServe, PAN and The Source can put a studio in touch with hundreds of other users who are more than willing to share their experiences and opinions. Furthermore, you can help yourself by buying equipment from a local dealer with a good reputation for support; a few bucks saved buying from a no-frills discount mail-order house may end up turning into many expensive hours trying to get help.

The most important thing is to keep ahead of your clients. Some of them will know more about a specific piece of gear than you do.

The most important thing is to keep ahead of your clients. Some of them will know more about a specific piece of gear than you do. But, because it's your studio, you should know best how things work *together*, and the best way to get your studio as a whole to respond to clients' needs.

Like any new technology, MIDI has to be respected, learned and mastered. Once you've done that, your studio will be a part of a very brave, new world.

R·E/P

Recording ENGINEER/PRODUCER

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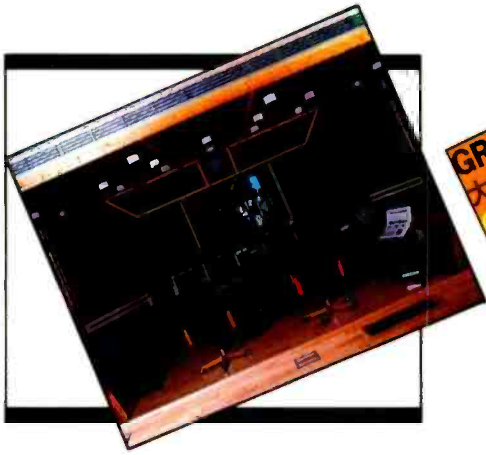
Compiled by Irma Allread



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Studio Update

Northeast

Giant Sound (New York) has added a **New England Digital** Synclavier digital synthesizer, for use in a dedicated preproduction room, the main control room, or at a client's own facility.

Currently available with 8Mbytes of memory, sampling capabilities range from 20.9s at a 100kHz sampling frequency, to 94.9s at 44.1kHz.

The Synclavier also features a 75,000 note, 200-track sequencer, a MIDI patch bay and time code lockup from half to double speed in both forward and backward directions. *1776 Broadway, New York, NY 10019; 212-247-1160.*

PhotoMag Recording Studios (New York) has added Dominick Tavella to its staff as recording engineer.

Tavella started in the industry as a transfer engineer at New York's Du-Art Film Laboratories. During his 10-year tenure there, he designed and built his own studio, and expanded the company's audio business from recording student films to mixing theatrical features. *300 E. 34th St., New York, NY 10016; 212-683-9672.*

Highland Studio (Delmont, PA) has added the following equipment to its 16-track facility: **UREI 809** studio monitors, **Lexicon 200** digital reverberation, **Neumann U-89** microphone, **Neumann KM-84** microphones, **Sennheiser MD-441** microphones, **Aphex Aural Exciter Type C** and **Nakamichi** cassette duplication system. *5 W. Pittsburgh St., Delmont, PA 15626; 412-468-6661.*

Sheffield Audio-Video Productions (Phoenix, MD) recently took delivery of its new **Sony PCM-3202** twin-DASH digital 2-track. Sheffield is one of the first 10 studios in the country to receive the unit. *13816 Sunnybrook Road, Phoenix, MD 21131; 301-628-7260.*

Normandy Sound (Warren, RI) has installed a **Solid State Logic 4000E series** mixing console with Total Recall automation.

"This update was essential," says Ogden Fell, general manager. "The Boston and New England record companies, producers and artists are now seeking out the ability to record and mix on the computerized consoles." *25 Market St., Warren, RI 02885; 401-247-0218.*

Forge Recording Studios (Malvern, PA) has opened a new 24-track audio facility, featuring a **Sony PCM-3324 DASH format** 24-track digital machine, **Neotek Elite** console and **Audio Kinetics Master Mix** automation. *119 Great Valley Parkway, Malvern, PA 19355; 215-935-1422.*

Forge Recording Studio.

Sigma Sound Studios (Philadelphia) has named **Corey Kissinger** as manager of its technical services department. His responsibilities will include studio remodeling, maintenance of all studio equipment and the supervision of two support staff members. Previously, Kissinger was technical manager at Alpha International Recording Studios, Philadelphia, and chief engineer at AV Studios, Reading, PA. *212 N. 12th St., Philadelphia, PA 19107; 215-561-3660.*

Southeast

Sixteenth Avenue Sound (Nashville) has taken delivery of the **Solid State Logic** SL4000-E 48-input console, a **Mitsubishi** X-850 PD-format 32-track and X-86 2-track. Also features are **Lexicon** 480L digital effects, an **AMS** DMX-15 digital delay and a **Studer** A820 2-track. *3524 West End Ave., Nashville, TN 37205; 615-269-5296.*

Hayes Recording Studio (Tampa, FL) has installed an **Audio-Kinetics** Q.Lock 3-10 time code interface system, a **Studer** A-80RC MkII 30 ips ½-inch 2-track, and a Lexicon model 200 digital reverb. *2406 S. MacDill Ave., Tampa, FL 33609; 813-837-6384.*

South Central

Omega Audio & Productions (Dallas) has taken delivery of the **CMX** Cass I Audio post-production editing system.

The Cass I allows up to 5-machine audio editing, including an on-line access to two different analog 24-tracks as well as a Mitsubishi X-80 digital recorder.

All standard CMX functions, such as preview, list management and dedicated key strokes, are available. *8036 Aviation Place, Dallas, TX 75235; 214-350-9066.*

Goodnight Dallas (Dallas), recently installed the following equipment: **Yamaha** REV-7 and SPX-90 digital signal processors, two **Nakamichi** MR-1 cassette decks, a Sony PCM-501 digital processor, **Neumann** TLM-170s, **Sennheiser** MKH-40s, and **Beyer** M160s mics. *11260 Goodnight Lane, Dallas, TX 75229; 214-241-5182.*

Midwest

ARS Recording Studio (Alsip, IL) has completed a major rebuild of its main studio. The room now includes RPG Diffusors and angled mirrors in the rear, and rough-cut cedar with removable baf-

Studio Update

fles that are absorbent on one side and reflective on the other. *11628 S. Pulaski. Alsip. IL 60658; 312-371-8424.*

Southern California

Cannon Films (Hollywood) has added four **Lexicon PCM-70** digital effects processors, three model 224XL digital reverbs, two model 97 Super Prime Time programmable digital delays, a model 200 digital reverb and two model 480L digital effects systems.

According to Corey Bailey, head of Cannon's sound department, "We didn't have room for live chambers or plates, because we built three studios in a very small area. So we're relying on three products for all of our ambience, as well as for external effects processing with the PCM-70s and the Super Prime Times." *640 S. San Vicente, Los Angeles, CA 90048; 213-658-2100.*

Master Control (Burbank) has recently added a pair of **JBL 4406** speakers to its selection of monitor loudspeakers. A **Hafler P500** power amp has been acquired for use with the facility's **Yamaha NS-10M** close-field speakers. In addition, a **Lexicon 224XL** reverb has been added to the outboard rack, and a **Studer A725** CD player in the control room. *3401 W. Burbank Blvd., Burbank, CA 91505; 818-842-0800.*

Marvin's Place (Hollywood) has announced the re-opening of the personal-use studio formerly owned by the late Marvin Gaye. The studio is managed by chief engineer, **Richard Barcellona**, administrator, **Jeanette Acosta-Hunziker**, and engineer **Michael Monarch**. Equipment includes a **Neve** 8108 52-input console, **MCI HM-24** 24-track, **Studer** A-80 ½-inch 2-track, **JBL/Augsburger** monitors, and **Sony PCM-1610** digital processor. *6553 Sunset Blvd., Los Angeles, CA 90028; 213-462-5818.*

Magnetic Media Productions (Los Angeles) has added a **Tascam MS-16** 1-inch 16-track, a **Sony PCM-501** digital processor, **Lexicon PCM-70** and PCM-42 effects, plus **Studio Master** 16/B/2 console. *7250 Hillside Ave., Los Angeles, CA 90046; 213-850-5268.*

Cantrax Recorders (Long Beach) has added a **Studer A820** 2-track, Valley People stereo dynamite compressors, and **JBL Biradial** monitors. *2119 Fidler Ave., Long Beach, CA 90815; 213-498-6492.*

Conway Recording Studio (Hollywood) has opened Studio B, which now features **Mitsubishi X-850** digital 32-tracks PD-format, and a 56-input **Neve** V-series desk. *655 N St. Andrews Place, Hollywood, CA 90004; 213-463-2175.*

Image Recording, (Hollywood) has installed a **Solid State Logic SI4056-E** console in Studio A. The console features a 60-input mainframe, computer automation with Total Recall, an integrated video switcher and plasma metering with spectral display.

The control room was rewired with **Neumann** cable, and a **Sony** video monitor also was added. *1020 N. Sycamore Ave., Hollywood, CA 90038; 213-850-1030.*

Northern California

Music Annex (San Francisco) has appointed Randy Bobo as chief engineer of the new San Francisco division. "What David [Porter, studio president] wanted to accomplish in developing the new San Francisco facility is exactly what I had in

Michael Smith, director/producer of KPIX Creative Services working with Randy Bobo, chief engineer, in Music Annex's Studio One.

mind," Bobo says. "We selected state-of-the-art equipment for the studios, but we were always aware that it's not just the gear that makes a good facility: it's the *people* who insure it's a success."

The studio features an Amek 2500 28x24x28 console, MCI JM-16/24 multitrack, Otari MTR-12 2-track with time code, Audio Kinetics Q.Lock synchronizer, and a Sony BVH-100 1-inch VTR with TBC.

"I think this is one of the most modern and sophisticated sound studios in town," said Jon Kroll, of the band Grey Direct, "and they've outfitted the studio with up-to-date equipment that helped me do my job." *69 Green St., San Francisco, CA 94111; 415-421-6622.*

Sound Recording Organization (San Francisco) has added **Jan Cohn** to its staff as business manager. Cohn comes from Broadway Video, where she worked in the scheduling department. *1338 Mission St., San Francisco, CA 94103; 415-863-0400.*

Russian Hill Recording (San Francisco) has added a Solid State Logic SL4000-E console in Studio A.

The console will incorporate modifications developed by technicians at SSL, Russian Hill Recording and Lucasfilm, so enable the facility to re-orientate its direction from music sessions to film work. Studio A's control room will also undergo remodeling to accommodate the new console and its peripheral equipment.

The studio is also adding a new synthesizer/media room that will house a 16-track, E-mu Systems Emulator II, Yamaha DX-7 and an Apple MacIntosh. *1520 Pacific Ave., San Francisco, CA 94109; 415-474-4520.*

England

Air Studios (London) has opened Studio 5 to serve as the MIDI programming facility.

The new room is based around a Fairlight CMI Series 3 digital synthesizer and an Apple Macintosh, providing a total of 216 MIDI tracks controlling a variety of synthesizers and samplers. In addition to independent operation in Studio 5, the entire system can be used under remote control from any of the four existing control rooms.

Also, Air Monserrate has added a 60-channel solid state logic SL4000-E that features 48 SSL modules and 12 Focusrite modules. Also available are two Mitsubishi X-850 digital multitracks. *214 Oxford St., London; W1N 9DF; 01-637-2758.*

Black Barn Studios (Ripley/Surrey, England) has a new 24-track studio that features 48-channel Soundcraft TS24 in-line console with Audio Kinetics Master-Mix fader automation, a Soundcraft Saturn 24-track with Total Remote, two Soundcraft Series 20 stereo tape machines and four Soundcraft SA Series power amplifiers. *3 The Green, Dunsborough Cottages, Ripley, Surrey, England GU23 6AL; 0483-222-600.*

AES New Products

Studer A820 multichannel recorder

The A820, with integrated Dolby SR, accepts 14-inch reels and is convertible between 1-and 2-inch tape widths.

The capstan motor has its own dedicated microprocessor control with three standard tape speeds. All transport operating keys are user-programmable, with a choice of more than 40 functions assignable from an internal software library.

Audio alignment parameters may be set for all 24 channels simultaneously and automatically. Digital memories store alignment parameters for two tape formulations as well as for .8-, 16- and 24-track headblocks.

To optimize erase current on each track, metal heads, Dolby HX Pro, phase compensation circuits and special D/A converters are included. Noise reduction cards (Dolby SR, Dolby A or Telcom) may be integrated into the A820, with NR alignment levels set and stored digitally with the other internal alignment parameters.

All communication between the overbridge and the transport deck is via serial data exchange, which allows remote placement of the overbridge display and functions up to 300 feet away from the deck with connection by a single 4-conductor cable.

Circle (124) on Rapid Facts Card

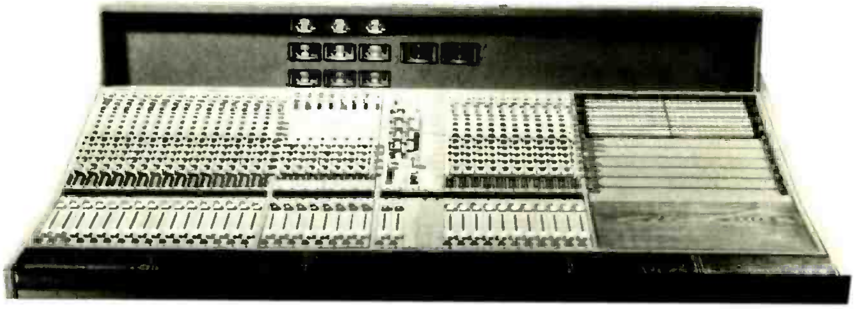
Amek announces Classic and G2520 consoles

Available in 32-, 48- or 64-input configurations, Classic features 12 group/meter modules and a standard 19-inch jackfield. Also provided are eight mono buses, two stereo buses, eight auxiliary sends and a 4-band sweepable equalizer per channel.

This production console features two fader options that can be used in conjunction with digital grouping and fader automation systems.

The G2520 multitrack console is available with 40 or 56 input channels. Additional features include bargraph metering, separate input fader blocks allowing options for fader automation, digital grouping and dual-input capability allowing each module to have two inputs during mixdown.

Circle (158) on Rapid Facts Card



Audio-Technica features AT 4462 ENG mixer

The portable mixer offers two monaural inputs which may be panned left or right, and two standard stereo inputs. All inputs and the two 600Ω outputs are transformer coupled and operated at microphone or line level.

Pre-fader "cue" is provided for each input. A special circuit allows program audio or any external source, such as a satellite feed, to be delivered as a monitor feed to the talent over conventional microphone cables connected to input channels 1 and 2, with no interference presented to the microphone signal. An audible tone can also be sent to the mixer headphones whenever peaking or limiting occurs.

Others features include 12V phantom power with provision for A-B power modification, VU meters, stereo/mono output switching, slate tone and internal slate mike, a 3—frequency oscillator, 2-color LEDs to indicate potential overload or limiter operation and a stereo limiter that may be switched to provide separate channel limiting.

Circle (131) on Rapid Facts Card

Yamaha PD2500 power amplifier

The dual-channel unit is said to deliver up to 500W rms per channel into 2Ω, or 1kW in bridged mono operation into a 4Ω load. A special power-supply design reduces the weight to 25.5 pounds; it occupies 3½ inches of 19-inch rack space.

The forced-air cooling system reduces the required heat sink areas and the air flow system is designed so that close rack-mounting of several amplifiers will not affect cooling efficiency.

Circle (153) on Rapid Facts Card

Tascam ATR-80 analog 24-track

Multi-microprocessor technology provides a "rehearse" feature that enables an engineer to preview drop-in edits without affecting the master. A 4-bit microprocessor and 8-bit D/A converter control system are said to provide seamless punch in/out and accurate edits.

The unit accommodates 14-inch reels and offers fast wind speeds of up to 375 ips. Both the reel motors and the PLL capstan motors utilize Samarium Cobalt to reduce mass, yet produce higher torque.

Also featured are contourless synch and repro heads.

Circle (128) on Rapid Facts Card

UREI model 7922 delay line

The unit features two independently adjustable outputs that enable alignment of acoustic centers of separate drivers within a single loudspeaker array.

In high-resolution mode, audio delay can be controlled in 10μS steps equivalent to about .125 inch resolution. Maximum delay is 327μS.

Circle (161) on Rapid Facts Card

New Products

Soundcraft Series 8000 live-sound console

Designed for concert-sound applications, the series 8000 features a 4-band parametric EQ section per input, eight individual auxiliary sends and individual 8-bus routing with separate LED indicators.

The console is available in both front-of-house and stage monitor configurations, in 24-, 32- and 40-input frame sizes.

Circle (162) on Rapid Facts Card

UREI introduces C series monitors

All three models in the series feature a claimed frequency response envelop to beyond 17.5kHz. And uses a 801C coaxial loudspeaker combined with a titanium-diaphragm compression driver to provide a single-point sound source.

The Time Align feature UREI says solves time smear by considering driver placement and adjusting crossover group delay parameters to achieve simultaneous arrival of sound from the voice coils of the two transducers.

Circle (130) on Rapid Facts Card

Yamaha MV802 mount mixer

The 8x2 unit includes channels for mic-level input, two independent auxiliary submix sends with stereo returns for compatibility with effects units, and VCA control of master levels via an optional foot control.

Other features include a clip level indicator, two aux sends, two aux returns, stereo master control, stereo level meter, and head phone output.

Circle (160) on Rapid Facts Card

Klark Teknik introduces JADE loudspeaker system

Based on a compact 2-way bass reflex acoustic system with integrated electronic filtering and amplification, the JADE 1 is said to provide 110dB peak output at 1m, and 105dB continuous output at 1m.

With a quoted frequency response of 55Hz to 17kHz, ±3dB, the electronically balanced system has an input impedance of 20kΩ balanced and 10kΩ unbalanced. Input sensitivity level for maximum output is +4dBu.

Circle (157) on Rapid Facts Card

New Products

Shure FP42
Stereo field production Mixer

Designed for remote broadcast and field production applications, the FP42 provides stereo outputs and four input channels, all switchable for mic- or line-level operation. Each input channel includes a level control, center-detented stereo pan-pot, and a pull-pot cuing feature for cuing or checking each input via headphones. The unit also features a stereo master level control.

Other features include a tone oscillator for line and level checks, a direct mix bus/output and phantom power for condenser microphone operation. Also included are dual VU meters, which are calibrated for +4dBm and +8dBm. Built-in stereo peak limiters are equipped with LED overload indicators.

Circle (127) on Rapid Facts Card

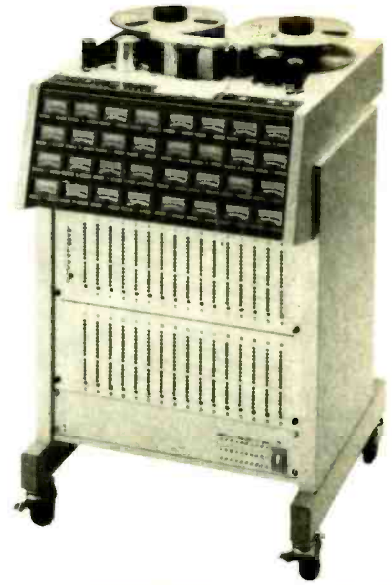
Otari MX-80
analog 24/32 track

Featuring a microprocessor controlled, constant tension tape transport with dc servoed reel motors, the unit is user-convertible between 30/15 and 15/7.5 ips speed pairs.

A built-in mini-autolocator with three cue-point memories, repeat function and return to zero is provided. Other features include tape speed referenced seamless punch-in/gapless punch-out, Dolby HX-Pro bias optimization as standard and a full-function remote session controller.

The unit is available in a 2-inch, 32-channel, 24-channel and a 24-channel prewired for 32.

Circle (129) on Rapid Facts Card



Digital Creations
ARMS-II console
automation system

The second-generation system, now available with moving faders, provides complete tape-based automated mix-down facilities plus mix-data storage on hard or floppy disks. Also provided are complete off-line editing, including splice and merge features. One track of the multitrack tape is required for time code.

Also featured is an independent solo/mute system and VCA fader modules incorporating the Valley International TA-101 gain cell.

Circle (156) on Rapid Facts Card

Studer
A807 analog 2-track

Built on a rigid, die-cast aluminum alloy chassis, the new unit is said to be suited to applications in remote recording, studio recording, broadcast production and industrial audio/video.

The two AC spooling motors are driven by 3-phase switched motor drive amplifiers for low heat dissipation, and optimum torque.

Tape-locator functions include locate-to-zero address locate, locate start and loop play. Other features include backspace, library wind, reverse play, and variable speed.

Circle (155) on Rapid Facts Card

Panasonic WS-A240
sub-woofer system

Designed to operate over the bottom two octaves for sound reinforcement and playback applications that require precise, low-frequency reproduction, the modular system delivers output from 30Hz to 80Hz.

It has a power capacity of 200W, a nominal impedance to 8Ω and offers a quoted amplitude response of ±3dB, 35Hz to 100Hz and a sensitivity of 92dB at 1w/1m.

The WS-A240 sub-woofer measures 16"x22"x11" and features a molded enclosure with external interlocking ribs to permit stacking of multiple units.

Circle (125) on Rapid Facts Card

Soundcarft TS12
in-line console

The console features 12 group buses with separate input faders configured for 24-track recording and mixing and six stereo subgroups.

The input/output module incorporates 4-band equalization including two switchable frequencies for high and low.

The two mid-bands are swept with switchable bandwidth. Also provided are two sets of Mute Group facilities.

During remix, the 12 main mix buses can be used as additional auxiliary sends. When the fader and bounce features are used together during remix, the result is four stereo groups plus ten auxiliary sends—four of which can be used as two stereo pairs.

Circle (126) on Rapid Facts Card

Yamaha
PM-1800 console

Available in four configurations of 16-, 24-, 32- or 40-input channels, the console also features four stereo auxiliary returns, eight group mixing buses, master stereo bus, an 8x4 mix matrix configuration and eight master mute groups. All auxiliary and group buses may be operated independently, resulting in a total of 14 discrete output mix buses.

Each input channel includes a 4-band sweep EQ section with in/out switch. A 12dB per octave high-pass filter on each input channel has its own in/out switch and can be swept in cutoff frequency from 20Hz to 400Hz.

Circle (159) on Rapid Facts Card

New Products

Neve V Series analog console

The V series is a 48-bus multitrack analog music recording console available in 36- 48- or 60-channel frame sizes, which can be fitted with NECAM 96, computer assisted mixdown automation system.

The console has 8 mono/4 stereo auxiliaries to give more effects paths for mixdown.

A centrally positioned monitor path status indication enables rapid console status checks.

Other features include: a choice of metering options, an independently assignable patch section allowing flexibility of insertion points, and a modular design enabling simple breakdown.

The desk includes integrated 12-channel sections, an 8-channel monitor unit, Master Status controls for input/output and an integral bantam patch field. The central monitor panel includes cue mix controls and up to four optional dedicated stereo effects returns.

Circle (132) on Rapid Facts Card

Sony MXP-29 production mixer

Offering eight channels with +4dB balanced inputs, the mixer incorporates mic and line switching as well as 3-band EQ, two aux sends, pan and PFL on each input. Individually selected 48V phantom power supply can be controlled from the front panel.

The mixer's output section features bargraph stereo LED metering, master and aux outputs faders, monitoring via headphones and talkback to master and aux outputs. Additional features include a pair of sub inputs with panning and level, monitor inputs and an oscillator.

Circle (139) on Rapid Facts Card

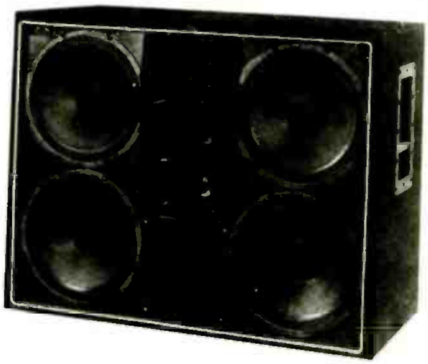
New Products

Community Light & Sound CS70 loudspeaker

Using four, 12-inch low-frequency drivers, a pair of midrange drivers with 2-inch throats, and a focused array high-frequency section, the speaker is said to deliver a response from 45Hz to 18kHz. With a broad-band sensitivity of 105dB at 1W, 1m, the CS70 is capable of producing continuous SPL levels in excess of 130dB at 1m and peak levels above 140dB, the company says.

The 3-way passively crossed system will handle input levels of 600rms and 1.5kW program.

Circle (146) on Rapid Facts Card



Furman Sound announces QN-4A quad noise gate

The 4-channel noise gate features three wide range controls on each channel: a threshold control (adjustable from -80 to +20 dBv), release rate control (adjustable from 5mS to 5s) and a depth control allowing the user to adjust the action of the gate, from gentle level reduction to total off.

The unit also features front panel calibrations (in dBV) for adjustments, threshold indicators and key inputs (for special effects such as "ducking") on each channel and a ground lift switch to eliminate ground loops.

Circle (133) on Rapid Facts Card

Rane GE 30 graphic equalizer

The ⅓-octave equalizer, features accurate bandpass control of frequencies on and between the iso points.

Level adjustments of any two adjacent control sliders allow alignment of the filter center frequency with that of a feedback or absorption node.

Circle (137) on Rapid Facts Card

Otari CTM-10 NAB cart machine

The unit features record-phase compensation and Dolby HX-Pro bias circuits. It can handle both A and AA-size NAB cartridges. Replay and record/replay versions can be operated standalone or interconnected.

Also featured are transformerless, balanced inputs and outputs, an LED tape timer with auto-reset and a built-in headphone amplifier.

The HX-Pro circuit alters the amount of bias current applied during recording by monitoring the amount of high-frequency information contained within the audio signal. This action counteracts the "self-biasing" effect of high-level signals with HF content.

Circle (151) on Rapid Facts Card

D&R Electronica multi-gate

The unit incorporates a dual-band, sweepable key filter, and conventional control parameters. The multigate attack delay and hold controls allow the user to create and modify the entire amplitude envelope of any signal independent of its dynamic structure.

In addition, the multigate's attach circuit allows tracking of extreme transients such as drums and piano without distortion, the company claims.

Circle (147) on Rapid Facts Card

Samson Concert TD wireless mic system

The TD System consists of the rack-mount, crystal-controlled CR-2 receiver (up to 10 channels can be used simultaneously) and the CH-2 handheld transmitter.

Optional mic capsules are available, including the Electro-Voice N/D 757.

The CT-2 belt pack transmitter can be used with a wide range of electret condenser lavalier mics.

Circle (152) on Rapid Facts Card

Ultimate Support Apache/Comanche keyboard stands

The Apache, a multiple keyboard stand, and the Comanche, a single keyboard stand, offer the following options: number of tiers, tier lengths, tier height, individual support bar tilt, number of support bars per tier, and leg angle adjustment.

Pivot fittings at the top of the framework allow lower braces to make adjustments for leg angles and the ability to straddle objects when necessary.

Each frame can be equipped with another tier up to 60 inches in length and are infinitely adjustable along the height of the leg system.

Circle (148) on Rapid Facts Card

Beyer introduces percussion mics

Including four moving coil dynamic mics, the M 420, M 422, M 380, M 201, and an MC 713 condenser, the new series provides for the integration of electronics into the drum set.

The frequency response of the M 422 is tailored for reproduction of snare drums and hi hats, with low-end rolloff that isolates unwanted frequencies produced by the bass drum or floor tom.

The low-end rolloff and hypercardioid polar pattern of the M 420 are optimized for reproduction of rack toms. The small size allows close placement without restricting the drummer's technique.

The M 380 is said to withstand SPLs of up to 140dB without overload or distortion, and provides a frequency response of 15Hz to 20kHz.

For floor tom micing, The M 201 low-mass moving coil transducer produces a "wide uncolored frequency response and transient response."

The MC 713 is a high-sensitivity (-39 dB) condenser microphone which delivers "detailed sound" for overhead micing in live and studio applications.

Circle (136) on Rapid Facts Card

New Products

First Order Effects releases software for Eventide SP2016

Included in the release are a variety of split programs that allow the SP2016 to simultaneously execute two independent mono or stereo effects. Available programs include various combinations of hall and plate reverbs, as well as reverbs paired with chorusers, stereo panners, delay lines and digital samplers. Some combinations feature internal effects chaining.

As well as general-purpose programs, the new release also contains a number of special effects including Stereo Shimmer, Random Ambience, Psycho-Panner, Dynamic Reverb and Sub-Base Synth.

In addition, a new program ROM, Studio Toolkit, includes a pink/white noise source, sweepable sine wave oscillator filter/reshaper, phase meter, and mono to stereo synthesizer. All programs are compatible with MIDI interfacing; upcoming programs include a MIDI pitch-tracker split effects combination and special effects.

Circle (312) on Rapid Facts Card

Shure SM89 shotgun mic

Designed for location film/TV production, and theater sound reinforcement, the mic features a quoted on-axis frequency range of 60Hz to 20kHz.

Below 60Hz, a low-frequency rolloff is employed for minimum pickup of wind, mechanical vibration, ambient noise and rumble without affecting voice frequencies, the company says. A built-in switch allows selection of a 60Hz or 160Hz rolloff frequency.

Other features include phantom voltage range of 11V to 52V and, a built-in windscreen.

The optional A89SM shock mount allows the mic to be attached to a boom, fishpole, stand, or other fitting with a 5/8", 3/8", or 5/16" external thread.

Circle (150) on Rapid Facts Card

Jamo loudspeaker series

The three-model series features a bass-reflex design and maximum sound pressure levels of between 116dB - 121dB A treble array of multiple horn-mounted tweeters enables amplification loads of as much as 400W continuous without breakup in the soundfield the manufacturer claims.

Each system has standard XLR connectors sockets and screwed terminals to accommodate cables as large as 6mm.

Circle (142) on Rapid Facts Card

Akai introduces MG14D rack mount recorder

The time code rack-mountable 14-track recorder features a GX glass ferrite head. The 12 segment LED bargraphs display signal level on each recording track.

The recorder section includes a sync-track and an internal time control track with 12 tracks devoted to audio.

A 10-key program pad allows tape locations to be entered into several memories.

Circle (135) on Rapid Facts Card

Circle (37) on Rapid Facts Card

New Products

Software for Techron TEF systems

The Polar software provides polar-response plots for loudspeakers and microphones measured with the company's TEF systems 10 and 12 analyzers.

From the data collected, the program can generate a polar display for up to 400 different frequencies. Output SPL can be plotted at various angles on- and off-axis, measured at one frequency or a band.

Displays can be viewed singly, in multiple curves one at a time, or in a range of curves in frequency sequence, to enable review of performance trends and polar response at different frequencies.

Circle (321) on Rapid Facts Card

Fostex model 160 multitrack mixer

The unit is a multiple input/output 4x2x2 mixer with an integral 4-track/stereo cassette recorder with 3¾ ips tape speed and Dolby C noise reduction. Multi-purpose inputs for the 160 accessory send/receive processing are included, as is a dedicated sync input to track 4 for use in MIDI applications.

Other features include a dedicated stereo bus with metering, pitch control, independent headphone level control and optional foot switch for remote puch-in/-out control.

Circle (134) on Rapid Facts Card

Apogee Sound AE-6 loudspeaker/ A-5 floor monitor

The AE-6 loundspeaker and A-5 dual-channel signal processor provide a 2-way, biamped stage monitor speaker system.

The cabinet houses a 12-inch high power cone driver and a 1-inch throat high frequency horn/driver combination. The design incorporates ferrofluid for low distortion short term impedance handling.

The A-5 dual-channel processor circuitry includes two channels of active balanced inputs (transformer inputs optional), active crossovers, time-domain alignment circuitry, low- and high-frequency corrective equalization, and separate low-frequency and high-frequency limiters.

Circle (145) on Rapid Facts Card

SONUS releases MIDI interface for Macintosh

For use with the Apple Mac and any MIDI-equipped synthesizer, the MacFace interface offers two MIDI Ins and six Out/Thrus. It can also be used as a conventional Thru box.

The interface attaches to either the Mac's modem or printer port and can switch between MIDI and peripheral connect without unplugging. The MIDI Out/Thru function also enables the user to play instruments attached to the Macintosh when MIDI sequencing software is not being used.

Circle (149) on Rapid Facts Card

AKG ADR 68K digital reverb

The new unit developed by Ursa Major and now being marketed by AKG, includes plate, chamber, room, and hall reverb algorithms, two split programs (plate/hall and room/room splits), as well as reverse reverb.

The software-based device offers eight seconds of 16-bit, 15kHz sampling, which can be broken up into four pieces of 2 seconds each. A pair of 2-second samples can be played back into one of the reverb programs. Samples may be edited for start and stop points, then triggered by the audio inputs, by impulse trigger jacks located on the remote, or by pressing the manual play buttons.

Circle (138) on Rapid Facts Card

Recording ENGINEER/PRODUCER

Coming in February:

Reverb and Delay Effects

- An Engineer's Guide to Digital Delay and Reverb Processors
- MIDI Control of Outboard Delay and Reverb Units

Additional feature articles will spotlight:

- Production Techniques for Dance-Single Remixes
- Facility Profile of Master Sound Astoria Studios
- Functional Equations for Recording and Production Engineers
- Developments in Disc and CD Mastering Techniques
- Production Viewpoint: Roy Thomas Baker
- Hands-On Review of Dolby Spectral Recording Modules

New Products

CMX Cass 1E audio editing system

The unit simultaneously controls up to six audio tape recorders or VTRs plus 14 general-purpose interfaces, permitting mixing of an entire soundtrack.

Designed for use in TV post-production facilities that offer audio for video sweetening and post-production, the system provides both hard disk and floppy disk storage as well as an additional CMX-compatible 8-inch floppy option.

All models will accept a CMX video editing list (EDL) for use as a starting point in audio post-production.

Circle (144) on Rapid Facts Card

Logitek Crossfire automated crossfader

Designed to be controlled directly from contact closure outputs on A/B roll edit systems, the unit provides audio fades between two sets of dual-channel inputs, using any of three fade styles.

Audio can be "faded-to-off", or can be faded down to background levels for voice-overs. A front-panel thumbwheel switch sets the fade duration anywhere from an instant cut to 9.9s in 0.1s increments.

Circle (143) on Rapid Facts Card

Fane Colossus 24E bass speaker

Designed for stage, concert hall and theater applications, the new LF driver may be used in horn loaded, tuned reflex, or infinite baffle systems.

The cellular foam cone uses a fatigue-resistant, double roll suspension and can reproduce frequencies from 80Hz down to 20Hz Fane says. It also has a quoted power rating of 400W, a 23-pound magnet assembly, a 4-inch voice coil diameter, and a ¾-inch usable excursion. Sensitivity is 101dB at 1W at 1m.

Circle (141) on Rapid Facts Card

Ensoniq Visual Editing for Mirage and IBM PC

The new Vision software features eight different screens, including animate function, which allows viewing of current wavesample tuning and volume problems as it plays in the Mirage.

Other features include single-key commands to move through and around the screens, a 3-D graphic wavesample display, a Mirage diskette copy function, a user interface which enables parameter modification, 4 note PC keyboard sound ability, PC disk sound storage ability.

The system requires a Mirage Digital Sampling keyboard or Digital Multi-Sampler; an IBM PC, XT, AT or compatible with a minimum of 320K memory, one disk drive, DOS 2.0 or greater, an IBM Color Graphic Adapter, Hercules Graphics Adapter (or other popular graphics adapters); and a Mirage Advanced Sampler's Guide (optional).

Circle (140) on Rapid Facts Card

Final Stage

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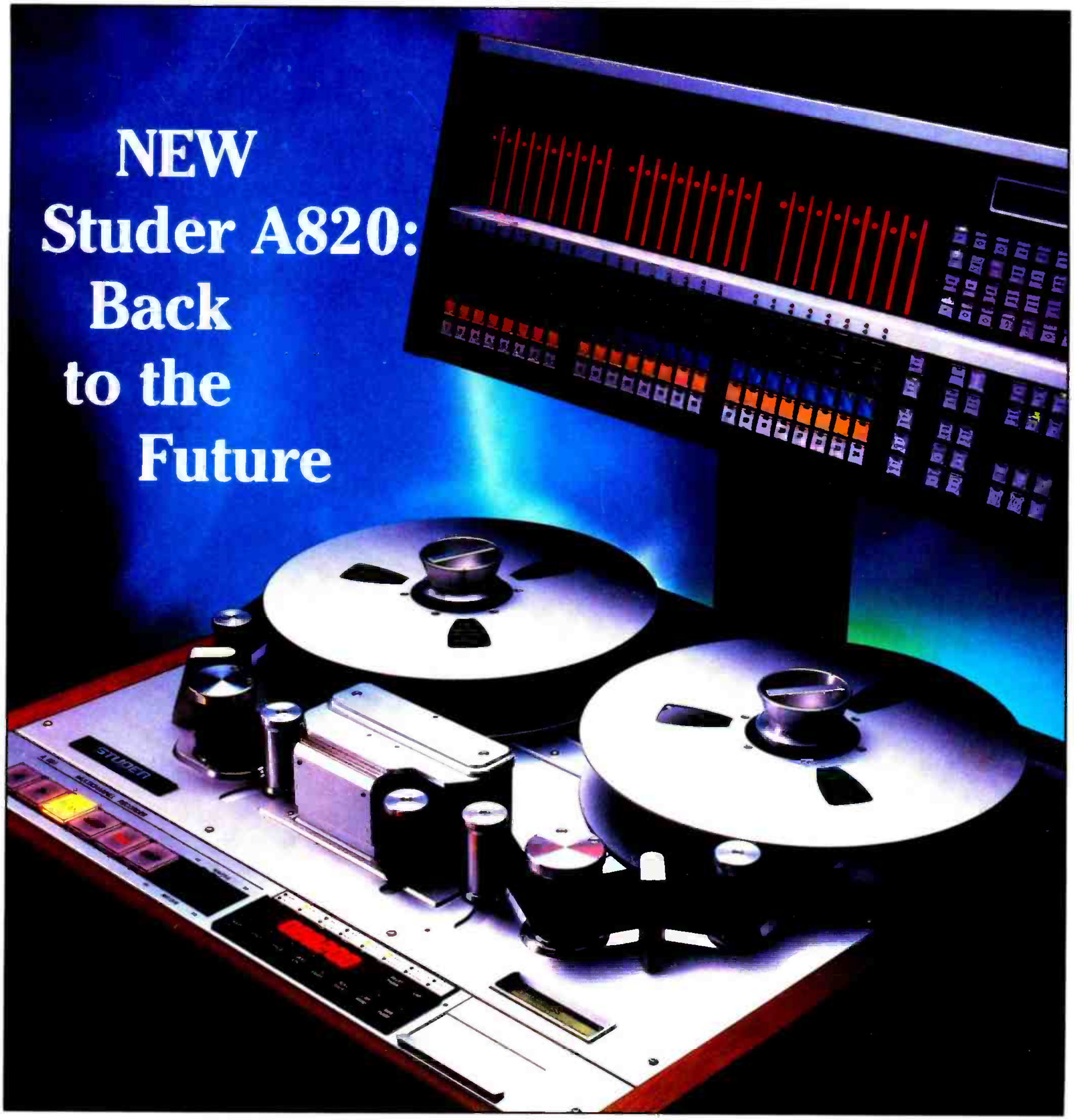

The future of multi-track mastering was commonly assumed to be 100% digital. But now Studer has built a multi-track for the future…by going back to thoroughly refine and update analog technology.

For the best possible combination of reliability, production capabilities, format compatibility, and sonic performance, the Studer A820 challenges *all* competitors. Analog *and* digital. No matter the price.

First, the A820 is fast and flexible. With total microprocessor control, it starts smoother, locks quicker, locates faster, and shuttles tape better than anything the competition has yet to offer. The tougher the job and the tighter the deadline, the more you'll love the A820.

The A820 is also fully user programmable. An extensive software menu lets you choose the operating features you want, and audio alignment is automatic with all parameters (including NR levels) set and stored digitally.

Finally, the A820 shakes the "sound assumptions." With new amorphous metal heads, advanced phase compensation circuits, and fully integrated Dolby™ SR as an option, the A820 boldly challenges the costliest digital machines for overall sonic performance. Let your ears be the judge.

Some other manufacturers apparently assumed analog could not get significantly better. With the arrival of the A820, that's now a questionable assumption. Call your nearest Studer office for detailed directions back to the future. Studer Revox America, Inc., 1425 Elm Hill Pike, Nashville, TN 37210, (615) 254-5651

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