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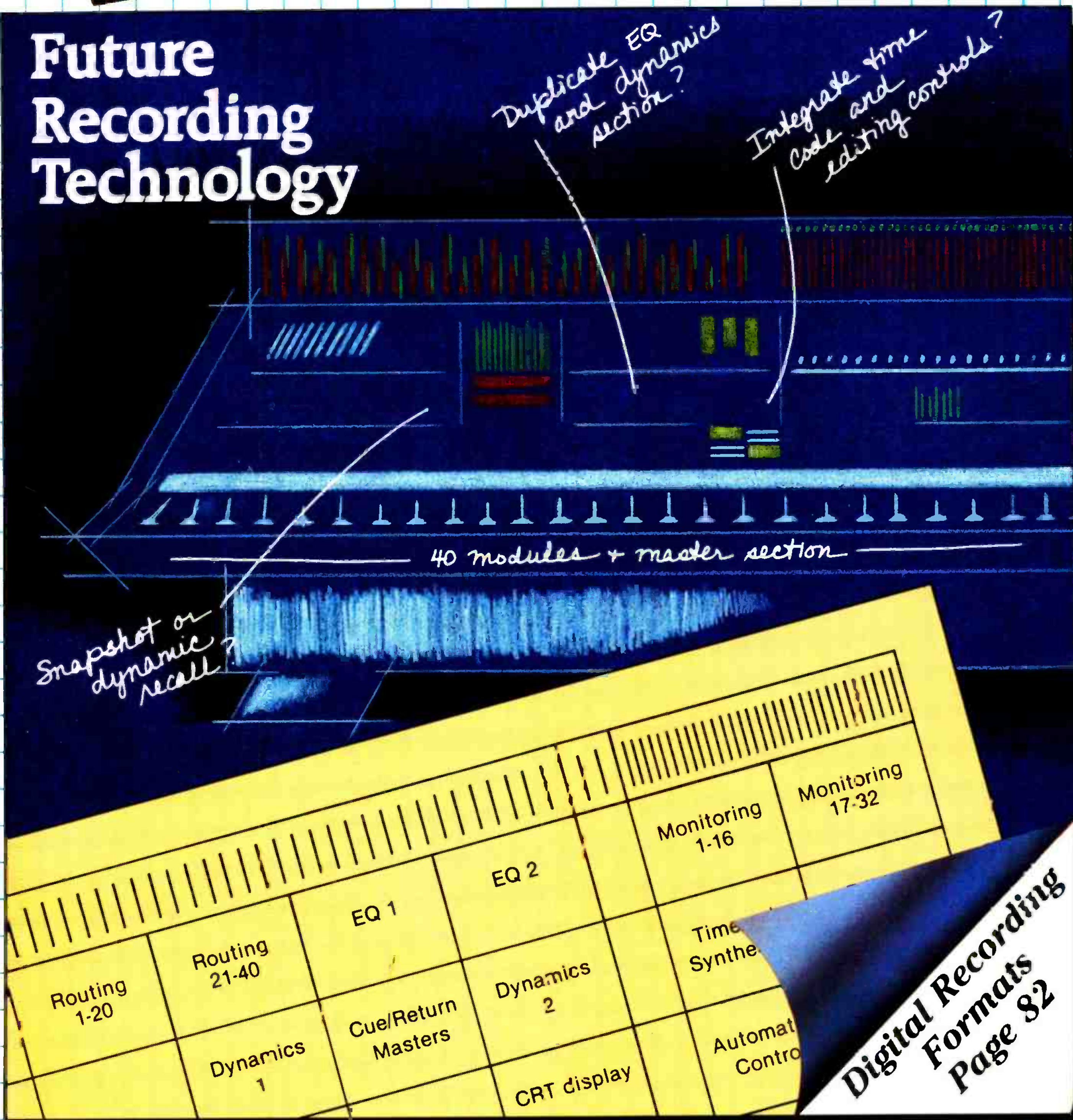

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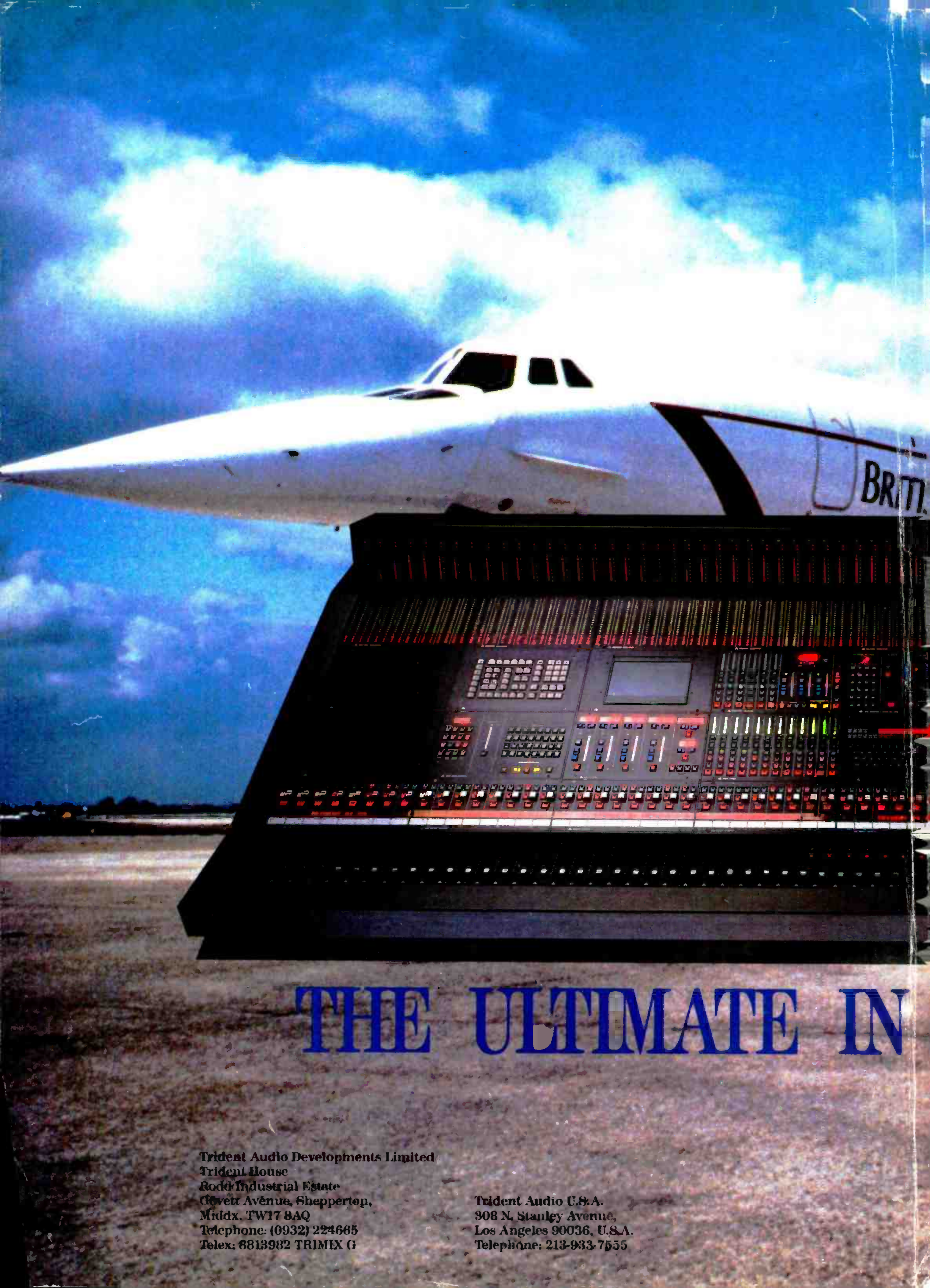

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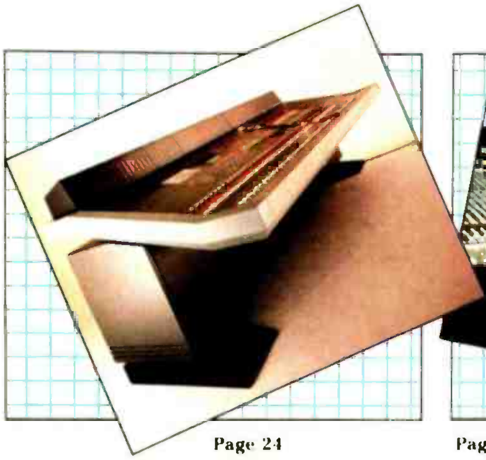

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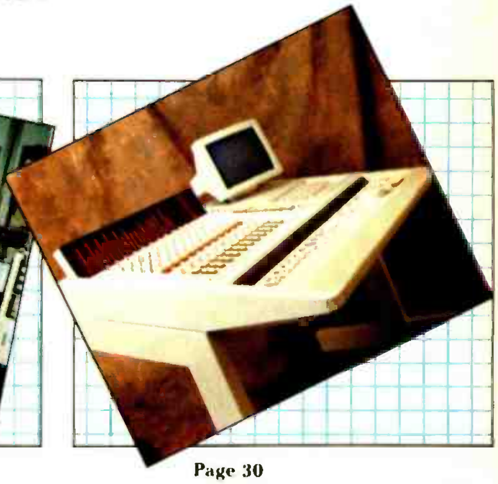

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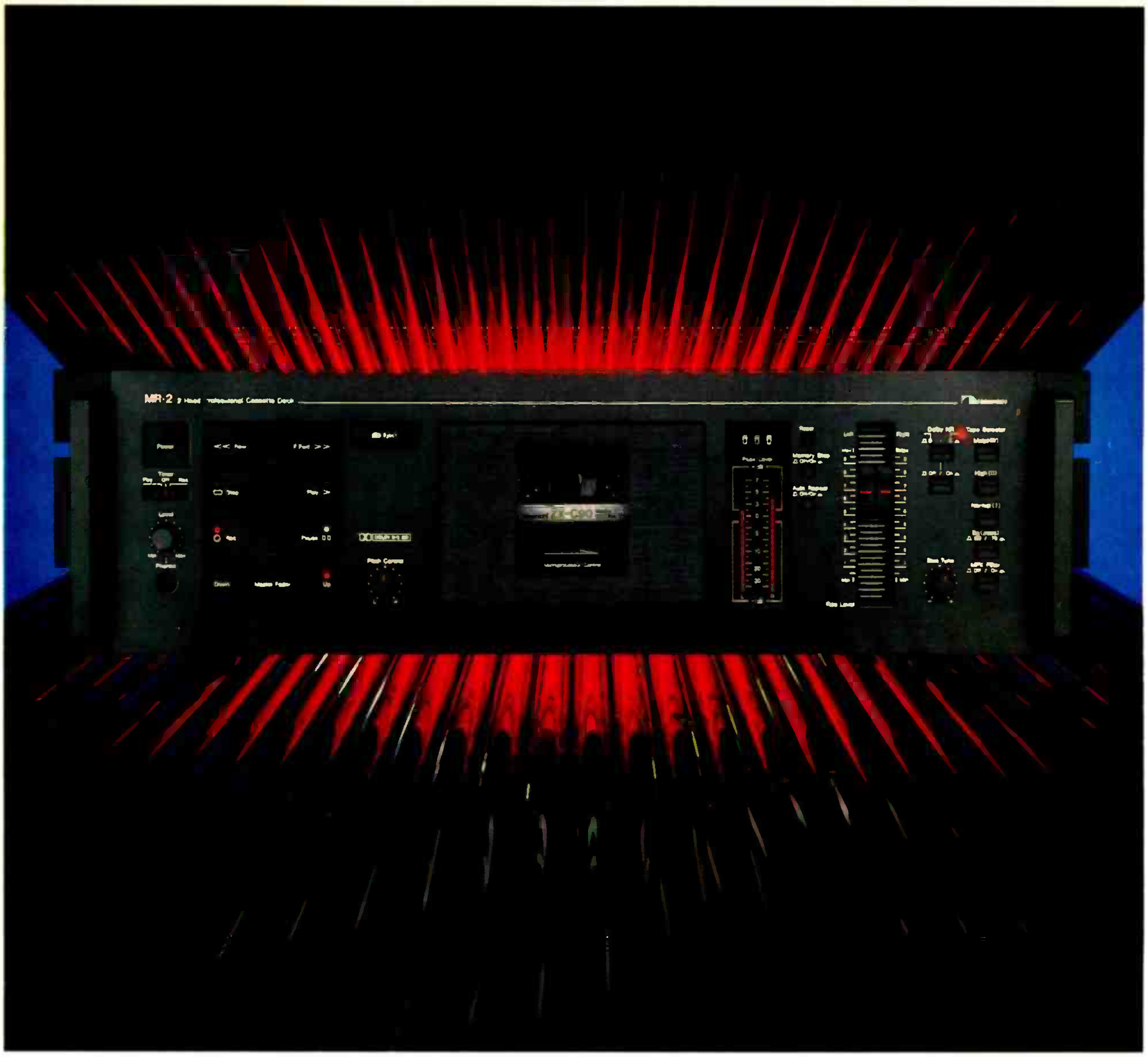

Editorial

Visuals for Audio

Continuing with my theme of console topographies (see: October Editorial), it occurs to me the pro audio industry can learn a great deal about display ergonomics from the type of editing, special effects and graphics systems currently being used by video production facilities.

Now that the availability of assignable or virtual audio production consoles is going to force us to rethink the display requirements of control surfaces, we would do well to study the ways in which other industries handle such problems.

With few exceptions, our industry has been somewhat reluctant to accept new technology and ways of working in the control room. Console automation represents a good example.

When the first systems appeared on the market, I recall several audio engineers and producers expressing extreme reluctance about using computer keyboards and VDUs to label and store mixes, control tape transports, and so on. These days, however, practically all audio professionals readily accept such tools as adding to their creativity in the studio. And now that we have access to recall systems that, by their very nature, require VDUs to display previous console control settings, it's easy to see that a similar design revolution is just around the corner.

Why do I stress the coming wave of assignable console designs with a potential revolution in display approaches? Simply because the topographies of future consoles will be able to centralize the display of current and previous control settings, rather than relying on the controlling element itself. As many of you are already aware, a rotary or linear potentiometer provides two distinct functions: it alters or modifies the associated electronic circuitry; and also indicates to what level the control is currently set.

With assignable designs, however, where one or several modules are remapped to control the appropriate analog or digital signal processing block, we need to devise alternative ways of presenting the operator with the information relating to the selected EQ section, auxiliary send levels, dynamic function, etc. Unless each control is equipped with a servo circuit that automatically resets the corresponding knob or fader to its previous value (a possible yet potentially costly approach), how best should this information be displayed?

One way might be to use LED ladders for level displays, or alphanumeric windows for discrete values such as EQ center frequency or compression ratio. The main drawback to both of these approaches, to my mind at least, is that you have to be pretty close to the console surface to see them, and most of us find it difficult (but not impossible) to relate numeric values with the corresponding audio effect.

A more flexible design philosophy would involve the use of a conventional video monitor or high-resolution (and backlit) liquid crystal display. Not only does this approach provide the opportunity to present data on a variety of screen sizes—and even duplicate VDUs for situations that involve multiple users, such as film rerecording consoles—the displays can also be enhanced as we adapt to the operational flexibility offered by assignable designs.

Consider, for example, the display of EQ information for a pair of channel inputs being used to control stereo drum tracks. First-generation systems might be set up to display the actual center frequency, cut/boost and bandwidth of four equalization bands. Later, as we become used to selecting which channel input(s) we want to interrogate—maybe using point-and-click-style software and a companion trackball, or even a VDU with softkeys arranged around its edge that are relabeled according to the selected application—it would not be difficult to revise the EQ display software to present graphic representations of individual or overlapping responses. In the same way, a dynamics section could be updated to visually display the threshold and slopes being used, rather than their corresponding values.

Sounds familiar? One or two currently available assignable and all-digital consoles do indeed incorporate such display flexibility. And several video editing systems utilize softkeys and adaptive software to allow a user to tailor the screen displays to reflect the mode of operation with which they are the most comfortable, and hence able to maximize their creative potential.

The point I am making here, however, is that without feedback from you, the user, the next generation of consoles may fall short on the type of display technologies and ergonomics that will streamline and maximinize your efficiency. **RE/P** is interested in hearing from you about both system integration and how console and system status should be displayed.

Mel Lambert
Editor

News

Le Mobile acquires new West Coast base

The North Hollywood rehearsal room facility, formerly operated by Leeds Musical Instrument Rentals, will complement the services offered by the Le Mobile remote truck, according to Guy Charbonneau, owner and chief engineer. "Le Mobile now has a home and office when off the road, and the Leeds rooms will have immediate access to an audio production facility."

Full cartage services along with a wide selection of instrument rentals will continue to be available through Leeds, and improvements in the rooms are planned for the near future.

Biamp management purchases company

Biamp Systems, for the past 15 months a wholly owned subsidiary of Leupold & Stevens, Portland, OR, has been acquired by the management team in a leveraged buyout.

"L&S has concluded it needs to focus its management and financial resources on the core business, since Biamp represented only a small portion of its total revenues," Jerry Payette, vice-president of finance, and one of the company's new owners, says.

According to Ralph Lockhart, president, "L&S' substantial investment has given Biamp the best equipped, most efficient manufacturing facility of its size in the industry; advanced computer-aided engineering and automated test equipment tools; an aggressive advertising and point-of-sale brochure program; and a number of successful new products."

A/T Scharff Rentals supplies digital 24-track for "Saturday Night Live"

In addition to a Sony PCM-3324 DASH-format digital multitrack for recording all audio material, the company has also supplied the *Saturday Night Live* sound-production crew with a Lexicon PCM-70, Yamaha SPX-90, Yamaha REV-7, and Lexicon Super Prime Time effects units, plus an AMS DMX15-80 digital delay line.

All *SNL* shows are now remixed from digital multitrack for sweetening prior to syndication. Also rented for the season are four Neve compressor/limiters, and a Sony APR-5003 analog 2-track with center-track time code.

By The Numbers offers optical-disk storage

Having successfully transferred a discrete 4-channel digital surround program to NTSC laserdisc format, the company now offers a compatible optical storage and retrieval system for its Colossus processor. A standard laserdisc player may be used to provide four channels of monaural, two stereo pairs or discrete 4-channel playback, in addition to the disc's conventional FM and/or digital audio tracks.

Audio replay is controlled via a laserdisc player's normal remote control or via an external computer. Location to a specific video frame number in the CAV (constant angular velocity) mode takes only a few seconds. The CLV mode,

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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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AUDIO LOGIC

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News

which stores up to an hour of audio per side (double that of CAV), requires a somewhat more extensive location effort but, according to the company, is reasonably efficient.

Potential applications include frame accurate random access to multichannel sound and music effects and double-system exhibition formats.

The Colossus audio processor enables storage of up to four discrete channels of 16-bit linear digital audio at a sampling frequency of 44.1kHz or 50kHz, using a standard VCR or VTR format. Frame or half-field editing is also included.

Audio Intervisual Design named Sony pro audio dealer

In Southern California, AID will be selling all of Sony digital products, as well as the company's analog consoles, tape machines and microphones. Previously, AID was handling only digital products, with sales topping $600,000 in July alone, according to Jim Pace, vice president of digital sales.

"Our new appointment gives us more tools to expand further into areas such as broadcast and video post-production," he says. "We're still a very high-end company though, and will continue to concentrate on those interested in leading-edge technology."

Recent sales include digital systems for Compact Disc production at MCA, A&M and Capitol Records.

"By next year we expect there will be seven Compact Disc manufacturing plants on-line in the United States," Pace continues. "More and more audio production facilities are moving into CD mastering, and teleproduction facilities see the need for digital audio. Business has been very healthy for the past 18 months and many companies are in an expansion phase."

AID also has been named one of four dealers in the United States to handle the new line of Sanken microphones manufactured in Japan.

SMPL customized editing software now available for license

According to John Simonton, designer of Synchronous Technologies' SMPL system and SMPL Lock, his proprietary time code software can be supplied on a custom license arrangement.

"The software will enable manufacturers to obtain time code-based features, customized for their products, without the lengthy development process that normally accompanies any new technology," he says.

"We have spent several years developing products that synchronize audio, video and MIDI-based equipment. Our developed expertise can save other companies a great deal of time and expense."

Allen and Heath Brenell recently introduced CMPTE, a Commodore 64 interface for controlling its CMC series consoles, that incorporates several SMPL features, including a time code generator and reader, MIDI Song Pointer information, external triggers and punch-in/punch-out.

Otari to use Ampex tape for quality testing

The company's entire range of analog and digital machines will now be quality tested exclusively with Ampex recording tape. In addition, all Otari analog and digital recorders to be delivered in the United States, Mexico and Central and South America will be packed with Grand Master 456 analog or 467 digital mastering tape.

UCLA offers courses in film-sound recording and design

The Extension Department of the Arts at UCLA will offer two classes during the winter quarter in sound for motion pictures. The first course, to be taught by Brent Keast, manager of Cinesound Corporation, is titled "Sound Recording for Motion Pictures." The 12-session class begins Jan. 7 and will feature lectures, demonstrations, discussions with guest professionals and hands-on participation.

Topics will include basic physics of magnetic and optical recording, studio and location recording, mixing and automated dialogue replacement. Keast holds a master of fine arts degree from the USC Cinema Department.

"Sound Design of Special Effects for Motion Pictures" begins Feb. 14, and will be taught by sound designer/composer Frank Serafine, whose credits include *Short Circuit, Poltergeist II, Star Trek I* and *III*, and *Tron*.

Compact Video wins Emmy awards for sound mixing

The four awards were presented at the Academy of Television Arts and Sciences' annual Emmy Awards presentation, in early September at the Pasadena Civic Auditorium.

David Fluhr won an Emmy for the tape-sound mixing of the PBS music special *Mr. Previn Comes to Town*, and John Reitz, David Campbell and Gregg Rudloff were each awarded Emmys for their film-sound mixing of NBC drama special *An Early Frost*.

News Notes

Dolby Laboratories has appointed **Syntec International Pty. Ltd.** as its distributor for studio products in Australia, including noise reduction and the new SR system; 60 Gibbes Street, Chatswood, NSW 2067, Australia; 02-406-4700; Fax 02-406-6136.

The **New York field office** of **Studer Revox America** has moved to 161 Avenue of the Americas, Suite 901, immediately across Spring Street from its previous location; the telephone number, 212-255-4462, remains unchanged.

JRF/Magnetic Sciences has joined with **Globe Precision Products PTE Ltd.** to form **Globe Magnetic Sciences PTE Ltd.** The company is the first to sell and service audio magnetic heads in Asia outside of Japan.

John and Cookie French, principals of JRF/Magnetic and **Arthur Ngiam,** managing director of Globe Precision Products, have hired **Jim Meng** as manager of the Singapore facility.

People

Alfred J. (A.J.) Menozzi has been appointed vice president, marketing and sales, at **dbx/ADC.** Previously, Menozzi was general manager, northeast region, at Toshiba America, and has held sales executive positions with Magnavox.

Stop Press

SPARS to hold business plan conference

The Society of Professional Audio Recording Studios will be holding a 2-day conference centered around the theme of developing a suitable business plan for audio-related facilities and studios.

Scheduled to be at UCLA, Westwood, CA, during the second week in March, the conference will address the requirements of audio-for-video, tape duplication, commercials and jingle production, film and video scoring, record production and industrial audio/video projects.

For further details, contact co-organizers Guy Costa, Mowtown Studios, Hollywood, and Nick Colleran, Alpha Recording Studios, Richmond, VA, or the SPARS national office: 818-999-0566.

R-E/P

Forget the exchange rates. We're rolling back the price of a Studer 24-track one more time.

The decline of the dollar forced us to raise the price of the A80VU MKIV a few months ago. But now it's going back down!

The US dollar hit the skids. The yen, the mark, and the Swiss franc went up. And the prices of imported audio gear went up right in step.

At Studer we scrimped and squeezed to keep price hikes to a minimum. But finally we had to raise the price of our A80VU MKIV 24-track package – at an all time low for two years – back above $30,000.

Unfortunately, that left some smaller studios in a tough spot. They wanted Studer quality and reliability. But their budget considerations forced them to look at lesser machines – which had also jumped up in price!

At Studer Revox America, we decided we did not want to lose our growing market share among smaller studios. First we placed a special quantity order from the Swiss factory. Then we shaved our own margin to the bone. Our accounting department winced, sighed, and reluctantly consented.

So now the price for a complete A80VU MKIV 24-track with autolocator and channel remote is back under $30,000.

Remember, this is the legendary hit-making machine still used by dozens of world-class studios. Plus, it now incorporates the latest transport logic updates as well as transformerless input and output cards. Nothing has been downgraded or compromised.

Needless to say this is a limited time offer and subject to withdrawal without prior notice. So if you expect to buy a 24-track anytime in the next couple years, it would be prudent to call your nearest Studer office today.

If the dollar dives again, tomorrow could be too late.

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STUDER REVOX

Letters

MIDI anomalies

From: Scott Spain, Seattle, WA

I am writing in reference to Paul Lehrman's column, "Managing MIDI." As chief of operations at a major 24-track room that is MIDI-compatible, I would like to share some of my experiences in the world of MIDI.

Synchronization of parts seems to be one of the biggies. In some MIDI Thru systems, you will end up with a 3ms delay in the slave (Thru) synths: a synth slaved from a Thru port of a Thru port with a fixed bpm of 120 would be around 1/512th a beat behind the master.

As meager as that delay may sound, it becomes very troublesome in a tune with a tempo that cannot be allowed to drift 0.1 bpm This holds true for digital delay, and also early reflection settings of digital reverbs.

I like having my electronic drums running live during mixdown, but it can be a big pain if you want amenities such as search-to-cue, or addressable SMPTE time code start/stop times. Also, waiting for a drum machine to sync up while doing overdubs is a drag.

I'm all for personal-use MIDI studios. They allow a performer to handle pre-production outside of a commercial facility, which can save the artist thousands of dollars in session time. Lately, there have been Top 10 records produced at home by some very talented people but who had great songs to begin with.

Since the early Seventies my peers have been telling me, "Scotti, you can record a hit song in your bathroom, and it will still be a hit." However, getting someone (a record company executive, for example) to listen to a poor recording requires a certain amount of trickery.

Our role in the professional-studio business is to turn an artist's music into a 3-D picture that the listener can "see" through. To inspire a listener to be attentive to a point of interest without the sense of sight is a real trick; one that takes years of training, not to mention hundreds of thousands of dollars in electronic equipment and acoustic design.

There are no dark secrets to hooking up MIDI systems, or to sampling; engineers have been putting outs to ins, and ins to outs for a long time. Also, we have been sampling just as long, only our storage medium was tape instead of RAM chips. The formula remains the same: garbage in equals garbage out. Getting the most out of an instrument's sound—recording or sampling it—is a job that engineers and commercial studios have been doing for years.

Building a world-class record is, and hopefully will remain, a team effort.

I have a love/hate relationship with MIDI systems. In the right hands it is magnificent; in the wrong ones, it's a mess.

Paul Lehrman's regular columns can help all of us get the best out of our MIDI systems, and use them creatively in the studio. We need all the help we can get.

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PITCH CHANGE D F.B. GAIN= 10 %	ADR-NOISE GATE TRG. MSK= 5ms	SYMPHONIC MOD. DEPTH= 50 %
STEREO PHASING MOD. DLY= 3.0ms	CHORUS A DM DEPTH= 50 %	CHORUS B AM DEPTH= 10 %
REV 1 HALL REV TIME= 2.6s	REV 2 ROOM DELAY = 20.0ms	REV 3 VOCAL LPF =8.0 kHz
REV 4 PLATE HIGH = 0.7	EARLY REF. 1 TYPE = RANDOM	EARLY REF. 2 ROOM SIZE = 2.0
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The SPX90 offers MIDI-compatibility including the ability to make program changes during live performance via MIDI. Some effects can

Without the expensive side effect.

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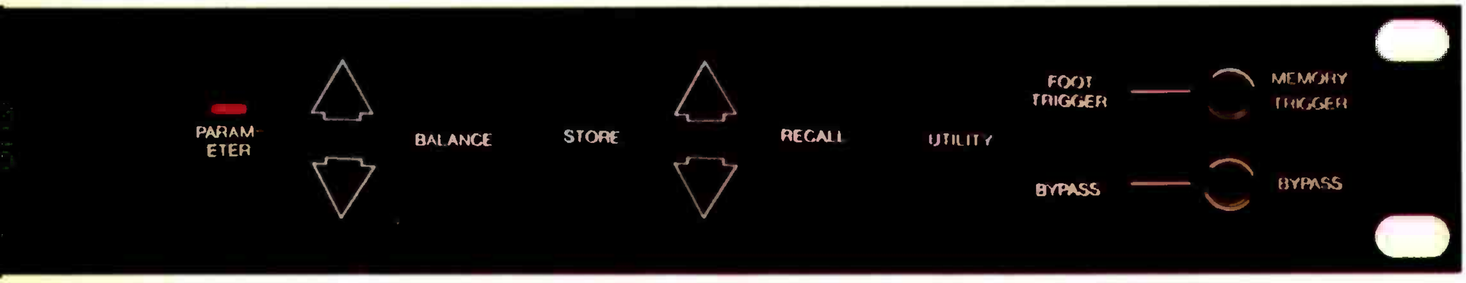

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Managing MIDI

By Paul D. Lehrman

Rule No. 1: Music is communication between human beings.

Rule No. 2: Machines can either help that communication, or get in the way.

Humans are not perfect beings; one of the key elements that makes music such an effective medium of communication is the imperfection and unpredictability of a given performance.

Machines are inherently the opposite: they are perfect (if they are working correctly) and flawlessly interpret the instructions given to them.

This dichotomy leads to one of the oldest and most common complaints about computer generated music: that it lacks "the human touch." When music is composed by typing numbers and letters into a terminal or by entering notes one at a time into a sequencer, it will by design, lack the little inaccuracies in pitch and rhythm that we perceive as the mark of a human performer. The result might be spectacular, or fun to dance to, but it quickly becomes very boring to listen to.

A partial solution to the problem is to introduce elements of randomness into the process. Rhythms and tempos can be "shuffled" slightly, pitches can be periodically set slightly "off," or timbres can be occasionally distorted. But these effects are only approximations of what a human being singing, playing a saxophone, or banging a drum really does. When a good player deviates from a mathematically perfect performance, it is not a random occurrence—it is a deliberate act for the sake of heightening expression. What a musician plays cannot be accurately described by a set of hard-and-fast rules, no matter how complex or random.

Therefore, the best way for machines to make music that sounds human is to somehow communicate to them the "physicality" of a performance. MIDI, with its plethora of real-time controller signals, is capable of dealing (at least on the theoretical level) with physical expression well enough to bring the sense across. Hardware that can translate a wide variety of physical movements into MIDI data and synthesizers that can make musical use of such data are still in the embryonic stage.

As Canadian composer Bill Buxton said at the Digicon '83 conference at the dawn of MIDI, "If we want to expand the range of musical expression, we have to use new gestures like blowing, sucking, squeezing, kicking and caressing."

More than three years later, we have still to see on the market a "squeeze" or "caress" controller. In synthesizer design, the emphasis is still on patch programming: what happens when you hit a key. But more thought must be given to what happens *afterward*: how a note (or passage) can be modified in real time under the control of a performer while it is playing.

A partial solution to the problem is to introduce elements of randomness into the process.

Many MIDI keyboards incorporate velocity, aftertouch sensing, and pitch and modulation wheels. Although these controls expand the range available to keyboard players, they don't do much for folks familiar with the type of expression available from, say, a violin. (Foot pedals and breath controllers can help, but few players actually use such devices as these.)

Imagine what a violinist does to change the sound. The right hand, through the bow, controls volume, articulation, and timbre by changing the position or angle of the bow. The left hand, through the strings, controls pitch in infinite gradations to vibrato depth, vibrato speed, and also articulation.

Now imagine the most sophisticated MIDI keyboard, and try to think how it could be used to control all those parameters with the ease and precision of a violinist. Not easy, is it?

Even a drummer has far more sound control than a MIDI drum pad can allow: a pad will only tell how hard it is hit, whereas an acoustic drum will also respond to where it is struck, what part of the stick is used, how long it remains in contact with the head and what happens after the initial hit.

Guitar controllers inevitably involve compromises that lessen, rather than enhance, a guitarist's expressive capabilities. Controllers are fine for expanding the catalog of sounds available to the player, but those sounds cannot be reproduced in a "guitarish" manner. There is no such thing, as far as I know, as polyphonic pitchbend, nor are there any synthesizers that can respond *simultaneously* to string-bending (i.e. continuous pitchbend) and left-hand movement over the frets under a sustained note (i.e. chromatic pitchbend). And forget trying to replicate the timbral and pitch nuances of left-hand hammering and pull-offs.

Pitch-to-MIDI converters, which theoretically give any musician access to MIDI, aren't really much better in this regard. Although some of them track quite fast, none can read more than one note at a time. With the exception of one well-known device, none can discriminate between articulated and legato passages—every new pitch means a new attack, which is not the way people sing, or play wind and string instruments. (Even the device in question, when used with most synths, is limited to legato passages that don't go more than an octave above or below the starting note.) And none can do much in terms of detecting timbral changes or vibrato.

Other types of controllers are coming to market, which is good news for players of those traditional instruments that they resemble. There are two MIDI "vibes" available, a MIDI violin and various types of percussion devices that provide a wider variety of control than simple pads.

But these are still a long way off from Bill Buxton's dream. What's needed is a whole catalog of physical devices that strap onto feet, hands, mouth, scalp, elbows, or whatever, whose output can be translated, in real time, into MIDI control codes.

The makers of the synthesizers themselves also have to consider real-time control more seriously. If a synthesizer receives a system-exclusive instruction, it should change the timbre immediately, not shut down the sound and wait for the next "note-on" before the change is effected.

And if we are going to use them in music production, such physical gestures have to be easily recorded and edited, which is the job of the software developers. The editing part is the tough one: Designing a human interface that is both comprehensive and understandable can be a real challenge. Graphics, color, and the ability to view multiple parameters on the screen and change viewing scale and direction are just some of the tools that should be employed so that the editing capabilities can match in intuitiveness the physical processes used to create the data.

After all, editing music one instruction at a time makes almost as little sense as writing it that way. R·E/P

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

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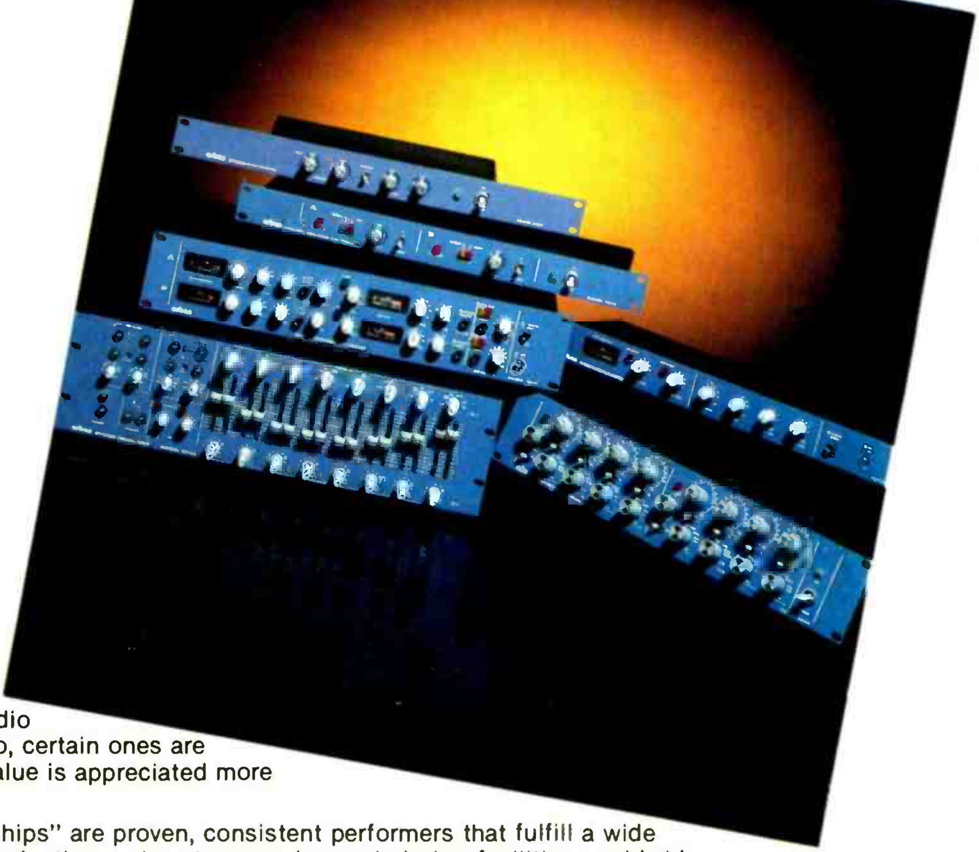

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Circle (9) on Rapid Facts Card

Film sound today

By Larry Blake

In the October issue I discussed the potential of the proposed R-DAT stereo digital recording format in production sound. Now, I'll look at its possibilities in the broad field of film and video post-production.

For the first time, one recording medium offers the potential to answer virtually all professional and consumer audio needs. Production recording, sound effects storage and retrieval, multitrack recording, sound editing and re-recording—all of these applications will be able to use R-DAT technology. The only format not included is theatrical motion-picture playback, although that too could be accommodated with interlock projection of three or four R-DAT machines.

Consumers will be able to use R-DAT as replacements for both cassettes and Compact Discs at home and in the car. I hope the existence of consumer R-DAT machines will bring down the price of both transports and blank tape for the pro audio industry.

The downside of the consumer connection is that competition with Compact Disc—both from pre-recorded R-DAT cassettes and from its potential as the "ultimate" home-taping format—might delay the introduction of R-DAT.

Probably the most interesting application of this new technology will be the use of multiple, time code-interlocked 2-track R-DAT machines as an "a la carte" multitrack. One imagines that R-DAT "multitracks" will be high-tech melting pots, with transports chosen for their "solid low-end" being placed in a rack next to another manufacturer's model noted for its "crisp highs."

Many have visualized this idea as implying that a transport will be assigned to every console I/O module. My hunch is that, for the time being, the current practice of separate recorders and mixers will continue for now, because consoles are replaced with greater frequency than recorders and there are many advantages to locating recorders outside the control room. Many studios are adopting the central machine room concept to share access to expensive tape machines between several production areas.

One scenario for these "multi 2-tracks" is to make them available in groups of four transports. The "low-end" 8-track machine could be upgraded easily to a top-of-the-line, full-featured 32-track with the addition of 12 transports and three 8-track controller cards. Presumably, a master-control unit will be rack-mounted with the transports.

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

The costs involved here are quite enticing. If a record/play R-DAT transport sells for $1,500, 32 digital tracks will cost about $35,000 including master control units and remotes. This price disparity

For the format to make sense in post-production, R-DAT should be used in editing and re-recording.

compared to current PD and DASH digital multitracks will not prevent R-DAT "multi 2-tracks" from becoming a reality. Even if manufacturers of DASH and PD transports continue to sell you their *very* expensive reel-to-reel, stationary-head digital multitracks, there are more than 80 other companies that can avail to the R-DAT technology.

Equally appealing is the possibility of using multiple playback-only transports both in film re-recording to replace 35mm mag dubbers, and for sound effects storage and retrieval. Traditional recording studios could also employ playback transports during mixdown. So a 2-studio facility might have 32 record/play channels (16 transports) for each studio, with another 32-channels available to either studio during mixdowns. The result is that you would never need to record simultaneously on more than 32-tracks.

For the format to make sense in film sound post-production, R-DAT should be used in both editing and re-recording. In contrast to the music recording world, where what is being replaced (multitracks) is a known commodity, the field of computerized, random-access sound editing is still young and malleable. Using only one recording format for post-production sound would simplify introduction of the new technology.

As many have speculated, sound effects could be stored either on multiple players or on a few "jukebox" changers. Effects being auditioned would be either downloaded to a local hard disk or even on-board RAM, considering the ability of the coming generation of microprocessors to address *gigabytes* of RAM. However, the sounds first have to be transferred from the R-DAT sound library. Although R-DAT players would not provide time random-access capabilities, as would be the case with a disk-based system, various rumors endow R-DAT with the ability to cue to any point on a 2-hour tape within six to 15 seconds.

There is one advantage that a tape based system will have in the storage and retrieval of sounds: the costly and time-consuming tape to optical-disk mastering step will be avoided. Also, there is the possibility of re-using R-DAT cassettes.

Continuing with the process, instead of the current practice of sending 100 edited 1,000-foot 35mm units to the re-recording stage for mixing a busy reel, the edit decision list created during random-access sound editing is replayed and the cut effects and dialogue tracks are recorded as "dubbing units" on multiple R-DAT transports. To allow slipping of individual effects without bouncing tracks, two mono sound effects probably will not be combined on one R-DAT cassette.

Today, if there is a change in the picture edit, all sound units have to be passed back to the sound editors, and each 35mm elements shortened or lengthened to accommodate the change. With computerized digital editing, the picture editor just tells the sound department of the new footages and, after a short time on the computer, the sound editor will simply record the sound for the reel again on another set of R-DAT cassettes.

I must hasten to add that most of the ideas presented here didn't spring whole from my brain. Everybody in professional audio is trying to solve the same practical problems while looking at the same potential solutions. Let's hope whatever way R-DAT technology evolves, standards for nomenclature, handling and system layout can be agreed upon, thereby creating a new method of recording, editing, and re-recording motion pictures that will be sonically and operationally transparent.

In the January issue, I will discuss the database and standardization problems facing manufacturers and users of the coming generations of digital sound editing devices. R·E/P

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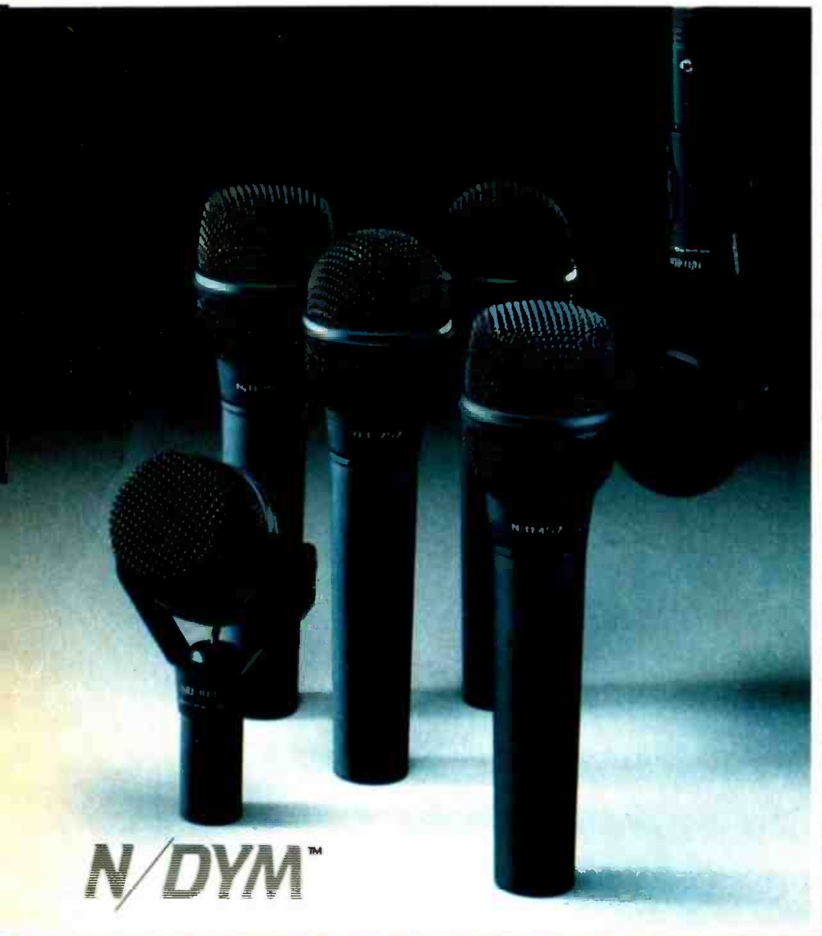

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Technology

By Stephen St. Croix

Revision — **n. 1:** an act of revising. **2:** a revised version. **3:** the promotion of confusion and anxiety amongst computer and software users.

Ah, to Rev, or not to Rev, that is the question: Whether 'tis nobler to ship the device with the software unfinished, thereby causing the customer to suffer the slings and arrows of outrageous operation, or to take arms against a sea of crashes and, by opposing, end them. To ship complete, risking delivery of a fatal shock to the buyer.

The same 16-bit, A-to-D and D-to-A conversion chip sets are now showing up in the majority of today's digital processors, and each month more and more of the other internal hardware is being shared by competitive products. All of which means that as we shop today, the actual hardware (whether or not you're aware of it) is becoming less of an issue. It is the software that we are really comparing while exploring a device's features, power, convenience and even sound. The front panel hardware usually can be remoted, and soon, graphic-computer interfaces will force even this to common ground.

In an earlier column I touched upon the then rare practice of shipping products containing software that was *not yet implemented*, in one or more key areas. Well, things have gotten worse. Now, not only is the practice of shipping software with a revision number smaller than 1.0 a more common occurrence, it is becoming all too popular.

There is unbelievable pressure on a manufacturer to develop, show and then ship new products as rapidly as possible. The reason for this is pretty simple: The other guy is doing the same thing. The most obvious advantage to you, the user, is that you get this month's monster technology, in your studio right now, or close to it. The biggest disadvantage, however, is that you may find yourself fighting with this monster in the middle of a session.

It is a generally accepted concept that software with a revision number smaller than 1.0 still contains bugs and dead ends; it is simply *not ready yet* and belongs in the hands of those brave beta testers. But you now stand an alarmingly good chance of actually paying for some rev 0.86b package and unwittingly becoming one of those same beta boys.

Some manufacturers have figured out that an end user might not be attracted to the dangers of some new toy with unfinished V0.7 software inside. So they have come up with an ingenious answer: Change the number. Which means not only that the device may be dangerous, but that it might not work properly in the real world application it was designed for. A Rev. 1.0 might work, but we are beginning to learn that something more on the order of Rev 2.15 is probably the real thing.

There is unbelievable pressure on a manufacturer to develop, show and then ship new products.

We are learning that the digit to the left of the decimal usually means that the big changes we've been waiting for have actually happened (and we will probably have to pay for them). The first place to the right seems to mean that several small but annoying weirdnesses have been repaired, while the second place to the right means that the five guys out there with Hyper-Speed Mark II computers can finally stop losing sleep over that 1-frame offset in the first bar when they are chasing SMPTE time code with MIDI pointers.

Wouldn't a real standard be wonderful? Let's say, just for fun, that V1.0 means that the product actually *works*. Really. All of the features advertised would be there. It won't take your hours of hard work on a one-way trip to Mars if you happen to go to a rarely used function a little differently than they thought you might. If V1.0 would actual *mean* this, we could buy software and software-intensive products with a lot more confidence when they first come out. In the long run the entire industry would benefit.

A little more communication during development between the manufacturer and well-chosen beta engineers or musicians would go a long way toward achieving this goal, as would customer pressure on the manufacturer to not sell it until it flies. We all need our new toys now, but if your new toy crashes every time you try that one thing you wanted to do, and the factory says the next Rev will fix that and that this one is only 90 days away....

Several companies do play it straight. If their product says V1.0 or above, it flies. The problem is that if the situation in general doesn't improve soon, we won't even trust the good guys, because we will be used to the fact that the designation V1.0 doesn't mean much on most of our software.

This scenario also applies to a free-standing software, which in a way presents even worse problems. With a conventional QWERTY keyboard used as the input device, there are a lot more chances for you to come across keystroke combinations that may crash the program than there are with a piece of gear that has 10 front panel buttons.

In the near future we will see a powerful new trend in our toys. If this problem isn't resolved by then, lots of people are going to be very unhappy.

For $12.00 or so, you will soon be able to buy a very nice, very fast, complete microprocessor that is low power (CMOS) and contains on-board RAM, ROM and even EEPROM. If that isn't enough to raise your interest, consider this: The same computer will have eight channels of A-to-D conversion, along with real interrupt features. (The option of battery operation will also be there just for fun.)

You may already own a few of these new micros or equivalent systems from other manufacturers. They will be hidden in the newest generation of synchronizers, remote controls, computer-to-SMPTE time-code interfaces and free standing editing or automation systems. It might even be in that hot new guitar synthesizer you pick up next year.

Such products will become much more intelligent and can obviously become faster and more powerful, while losing weight and costing less. Today, electronic design engineers can throw one of these computers at a design problem as easily as they could a simple gate only a few years ago. (Let's hope the designers assign a bit of this new-found intelligence to interfacing.)

All of this will be great *if* the software permanently burned into these new chips is completely debugged. As inexpensive as these chips are, they still cost 10 times as much as the ROM used today. A software update might mean not only new ROM, but a new engine as well.

Because of the race for your bucks in this small industry, we will continue to see new products put on the street in unfinished versions for as long as certain manufacturers feel they can get away with it. It's up to you to force them to provide dealers with completely working products. If you don't, they won't.

Ay, there's the rub. R·E/P

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

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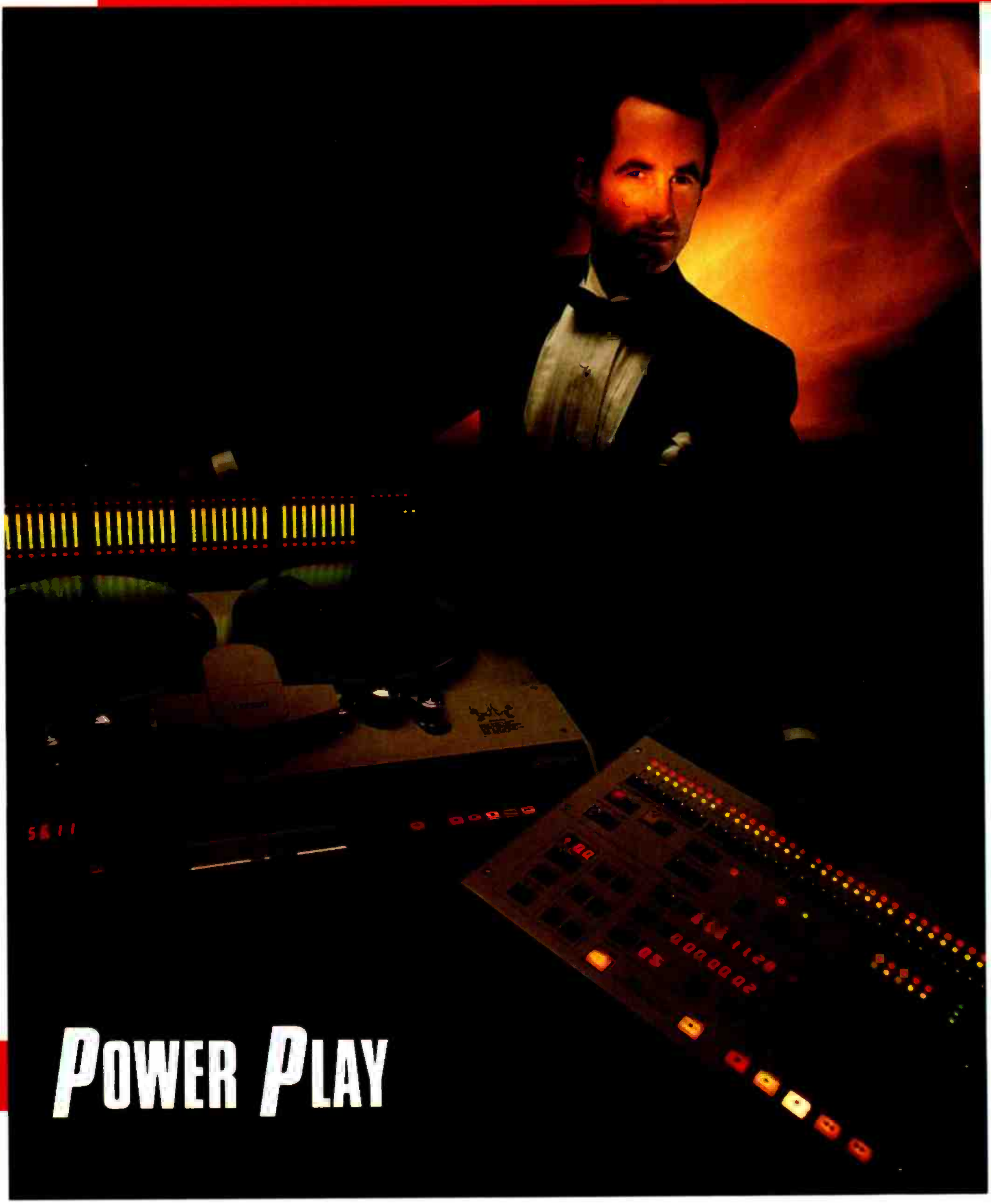

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SPARS On-line

By Gary Helmers

SPARS is a non-profit organization. We planned it that way. Production and recording studios can also turn themselves into non-profit organizations. No one ever plans it that way; often it happens because there is no plan at all.

Taking the time to commit a business plan to paper is easily ignored in the day-to-day bustle of running a facility. Without that plan, however, it is unlikely that the business will ever reach its full potential.

A business plan forms a blueprint for the construction of a business, in much the same way that architectural drawings serve as the blueprint for a studio's physical construction.

Developing a business plan has two primary objectives. First and foremost, it organizes your individual approach to the studio business. Secondly, it is a presentation to bankers, creditors and potential investors of what you are going to do, when you are going to do it, and what financial and people resources you will utilize.

A well-constructed business plan is a work of art. Like any creative endeavor, completing all the details can be frustrating but the final product is worth the effort. Just the process of putting into words your ideas, goals and aspirations can make the difference between success and failure. A business plan is an essential task for the business novice and the veteran; for the start-up studio and the continuing operation.

Following are the basic elements of a business plan.

• *Executive Summary.* This is just a fancy term for an introduction. You, the executive, should write the summary when you have finished all other phases of the business plan.

• *Statement of Purpose and Concept.* What will make your studio business unique and necessary in the world of existing facilities? Try to state in one or two sentences exactly what business you are in and why. Briefly state what you believe to be your strengths and weaknesses. You must be able to define for yourself, and for bankers, creditors and potential partners, just what your business will be.

• *Definition of Objectives.* Start with personal objectives, which will determine how you structure many aspects of your business. Is your personal objective increased net worth? A steady income? Independence? Guaranteed availability of the creative tools you and your studio will need for your projects?

Then, define your long-term objectives. Do you intend to build the business and sell it, build to some target level and maintain, or expand the scope of services you offer? The definition of long-term objectives is especially important in a partnership. Too often business partners find two or three years down the line that one partner wants to plow profits back into the business and expand, while the other partner wants to take his share of profits and vacation in the Bahamas.

A business plan forms a blueprint for the construction of a business.

Finally, define the business objectives. What are your monthly and yearly sales objectives? What are your profit goals?

Objectives can reduce confusion about what you are really trying to do. Additionally they give you a target. As Thoreau said: "In the long run, men hit only what they aim at."

• *Market Analysis.* The purpose of the market analysis is to completely identify the target group of people that will become your clients. Include the following factors:

–Geographical size of your market. Do you consider your market to be your city, state or the whole world?

–Types of markets you will pursue; jingles, records, demos, scoring, tape duplication, among others.

–Potential client profiles. What exactly is your client's job? Who does the client work for? What hours do they like to work? What are their hobbies? If you are in more than one type of market, you may describe separate kinds of clients.

–Influences on clients. What kinds of things will make a client use your facility? The kind of equipment you own? Its location? Advertising? Your own skill and expertise?

–Competition. Describe the strengths and weaknesses of each of your competitors and assess your own strengths and weaknesses.

• *Description of The Working Facility.* Detail the physical space and equipment necessary for your operation, and the staff required to run it.

• *Marketing Plan.* Marketing is the entire process of enticing the target group you defined in the market analysis into giving you money for your services. Consider the following:

–The methods you will use to sell.

–The kinds of advertising you will use.

–The features and services you will emphasize.

–How you will differentiate yourself from the competition, how the competition will respond and how you will deal with their response.

–How credit approval will be administered for clients.

–What you will do if your business grows faster (or slower) than you had planned for.

• *People.* The studio business is extremely people-orientated; all employees have contact with the client. It is very important to describe in detail the qualifications for each staff position listed in your description of the working facility.

Describe how responsibilities will be divided among the staff and who will report to who. As your staff grows, this will take the form of an organizational chart. Create a process for salary increases, a plan for staff growth, and a compensation plan that will reward the key people to keep your studio successful.

• *Budget Projections.* Objectives, market analysis, facility plans, the marketing plan and staff requirements all have financial implications that must be summarized in projected budgets. The construction of budget projections is a process that has been written about in numerous textbooks, pamphlets and articles. There are even software programs available to assist you. Find an approach that you feel comfortable with and give it a try.

Make sure your projected budgets include:

–Capital expenditures budget.

–Weekly, monthly and yearly cash flow budgets.

–Projected operating statement and balance sheet.

Now go back, review what you have written and write the executive summary. You should have a much more coherent picture of the business you are trying to build; the strengths and weaknesses of your plan; the problem areas; and the critical building blocks of your enterprise.

Don't hesitate to consult with your colleagues in the studio business. Most studio owners are very willing to share their personal experience, their mistakes and successes. R·E/P

Gary Helmers is the executive director of SPARS

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Developments in Assignable Console Designs

By Malcolm Toft

The new generation of assignable consoles may incorporate the most significant and fundamental ergonomic changes that the recording industry has ever seen.

The ever increasing sophistication of outboard equipment, coupled with digital multitrack recording becoming more commonplace, has put increasing pressure on console design and ergonomics. The use of extra outboard gear demands more auxiliary sends, while 32-track digital requires a greater number of console inputs.

It is rapidly becoming apparent that conventional analog console design is reaching a stage where it can no longer cope adequately with these demands. Although running out of inputs might be a major annoyance on a session, it can be accommodated by adding a submixer or, if the facility has been prudent, adding modules in a prewired frame. However, a shortage of auxiliary sends or other facilities cannot be so easily overcome, if they can be overcome at all.

Console manufacturers must be aware of these new requirements and develop acceptable solutions to problems set by the end-user. "Acceptable" is perhaps most important within this context, for it will become necessary that the operational aspects of consoles undergo major changes: to the extent, possibly, that the conventional console as we know it, consisting of duplicated modules, may all but disappear. The new generation of all-digital consoles may incorporate the most significant and fundamental ergonomic and electronic changes that the recording industry has ever seen.

In order to address the problem, it becomes clear that any solution can only be achieved by fundamentally reappraising the operational aspect of a recording console. Only then can certain decisions be made that will effect the technical implementation of such solutions.

Malcolm Toft joined Trident Studios, London, in 1968 and worked on sessions with such artists as the Beatles, David Bowie, James Taylor and Joe Cocker. In 1972 he started Trident Audio Development, a company that specializes in the manufacture of consoles for multitrack studios, sound reinforcement, video post-production and film.

Art and technology

Because music recording represents a marriage of art and technology, equipment manufacturers often have to rely on feedback from non-technical musicians, producers and artists; it is often at this stage in dialogue that misinterpretation becomes common.

For example, posing the question, "What is the most desirable feature you'd like to see incorporated in a mixing console?" For most end users the reply would be: "total automation."

Translating this request literally, the manufacturer then supplies a console with all functions automated via VCAs, only to find that the wonderful new answer to everyone's problems is unacceptable because there is a historic dislike of VCAs in the signal path and the system becomes far too complex from an operating standpoint.

Manufacturers attempting to present a radical change in console operation has to have a wide and preferably working knowledge of recording studio techniques, so that they can best determine what would or would not be acceptable in working conditions.

After an analysis of the use of an automated equalizer, as an example in the direct relationship to current working conditions, it becomes apparent that once the equalization controls have been adjusted they are, in the majority of cases, left static. Dynamic automation of EQ functions is therefore not required. However, a static form of automation (an instant reset) of all controls is an ideal solution, because it is simple to operate, cost-effective and fulfills that majority of operational criteria.

On further analysis, it also becomes apparent that, apart from faders, static automation can be applied to most other console functions. Faders most definitely require dynamic or real-time automation, because the recording or production engineer is not only constantly readjusting input levels, but wants to record the movement from one session to another session.

Realistically, a recording or production engineer can only accurately set the controls on one channel or module at a time. On the control surface of a conventional console equipped with 40 modules, an engineer is only using 2.5% of the controls at any one time, so 97.5% of the console is wasted space.

Control assignment

It makes sense to design a console whereby only one set of controls is used and assigned to the particular channel an engineer needs to work on. The idea of assignability is appealing for several reasons: you are no longer constrained by the physical size of a module, because it is no longer duplicated (the 2.5% becomes 100%); operational handspan is reduced; and the number of mechanical controls that can fail is reduced, in the above example, by a factor of 40:1.

There are pitfalls, however. It is easy to get carried away with the concept of assignability, and believe that *all* functions can be assigned. If followed through, you would be left with a console consisting of one set of controls, and a fader that are assignable to their appropriate functions.

This topography represents a far from practical solution. A practical assignable console would have only carefully chosen controls, and the remainder repeated as on a conventional console. Because, as a basis, we must have the conventional number of faders, the console length is not reduced dramatically, if at all. Operator reach can be significantly shortened, however, because the equivalent of a conventional module only contains a few controls.

The next major decision to make is whether the audio signal itself remains analog, or should be converted to digital.

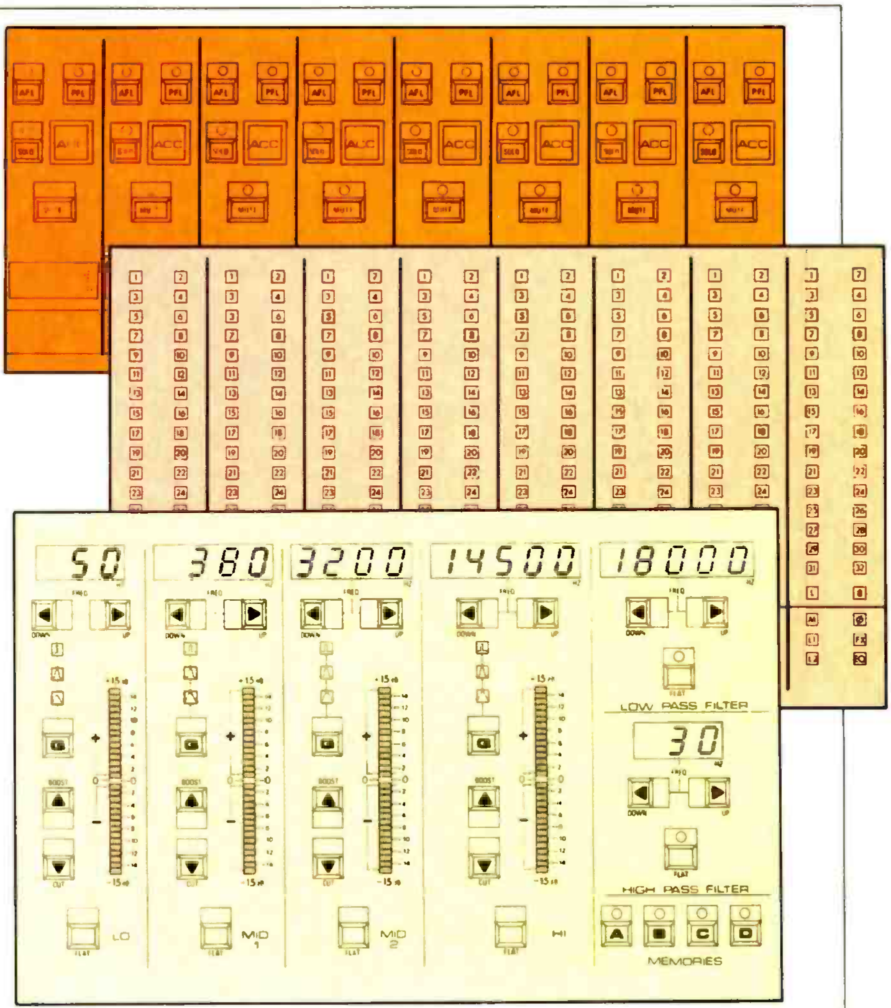

Figure 1. *(Top) An 8-input access panel with channel access, mono AFL, PFL, stereo solo, mute and automute controls, plus alphanumeric display per input.*

Figure 2. *(Center) Routing indicator panels display group assignments, input selection (mic/line), phase reverse, EQ and dynamics insert status.*

Figure 3. *(Bottom) A single assignable equalization module features four 16-frequency parametric EQ section, plus sweep high- and low-pass filters.*

At this point a commercial decision has to be made, because it's a well-known fact that, with present microprocessor-based technology, a fully digital console would cost at least three times that of its analog counterpart.

Furthermore, all-digital signal processing in console terms is still very much in its infancy. I strongly believe that in order to produce a fully digital console for production in, for example, three year's time, a commitment in hardware terms must be taken now. Because in the world of computers and related technology, three years represents a long time, it's likely that when the console is offered on the market, faster, cheaper and more advanced processing devices will be available.

But if this argument was always applied, no one would ever develop new products. However, there is always a right time and a wrong time to become involved in leading-edge technology. Because the development of an all-digital console embodies two areas of radical development—that of digital signal processing and a major ergonomic change—it would appear sensible to develop each one a step at a time.

By first developing a digitally controlled analog console, the operational and ergonomic advantages of digital manipulation can be fully explored, while allowing digital signal-processing techniques to mature.

It would be a further bonus if the development of a digitally controlled analog console could be structured to be theoretically feasible, when the price/performance ratio becomes a commercial reality, to replace the digitally controlled analog audio cards with all-digital audio signal processing.

There is no reason why a digitally controlled analog console designed accord-

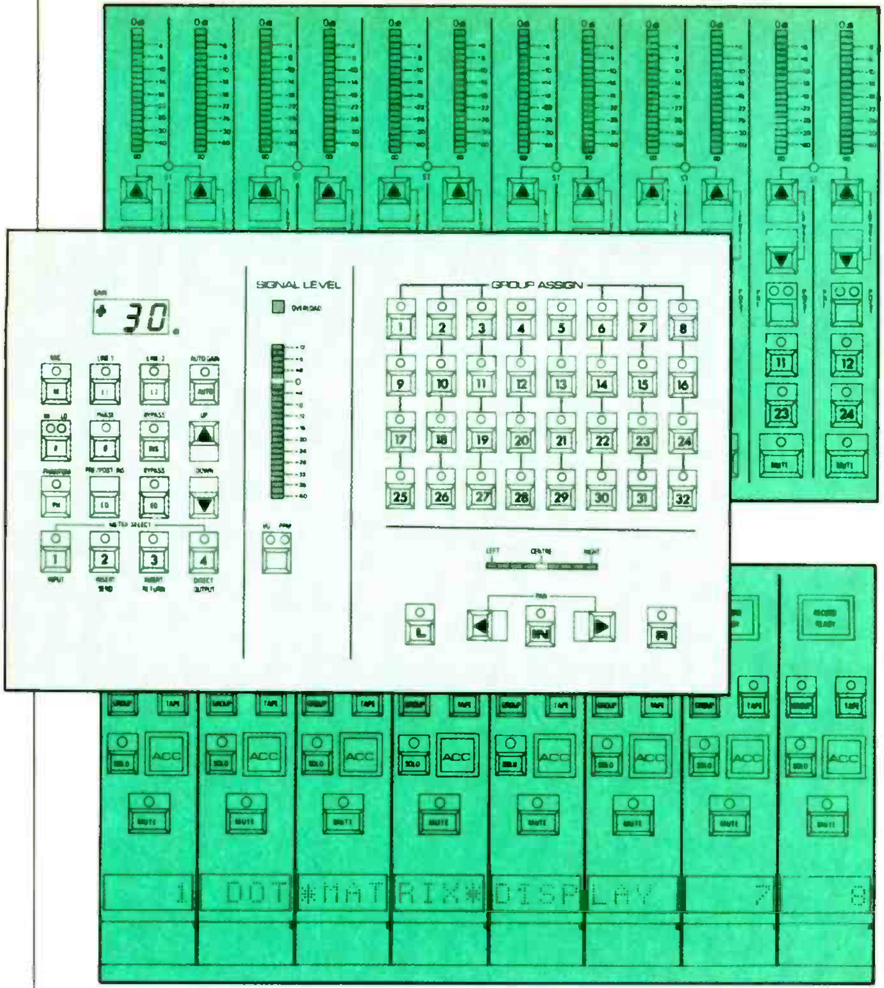

Figure 4. *(Top) The auxiliary send panel comprises 12 identical sections that can be ganged for 12 stereo pairs.*

Figure 5. *(Center) The assignable mic/line routing section enables selection between mic and 2-line level inputs, as well as assignment and pan to 32 output groups.*

Figure 6. *(Bottom) Individual access panels are provided for each monitor channel, with stereo solo, mute, group/tape switching and record/ready selection.*

ing to the above criteria should not be priced very favorably against conventional, high-end analog consoles.

Practical realization

Apart from the channel, group and monitor faders, a proposed design of a digitally controlled assignable console would provide static memory of all levels, routing, equalization, auxiliary sends, panning, mutes and solos. An onboard random-access memory of 1Mbyte would allow all settings to be stored and archived to floppy disk. Because the static-memory system is linked to SMPTE time code, console settings can be reset to an accuracy of one video frame which is 33ms.

A change can comprise just one setting or every control in the console, referred to as an "event." It is possible to have up to 128 different events that can be repeated to 512 times during a mix.

Our console design does not contain any input or output modules in the accepted sense. Instead, the console allows the user to access a number of central control panels via an illuminated access button situated above each fader. These access buttons, together with mute, AFL (after- or post-fade listen), PFL (pre-fade listen), stereo solo and channel-status indicator LEDs, are the only controls duplicated for each input.

The input section comprises three central control panels about the size of an 8"x11" sheet of paper and separately consists of equalization, auxiliary sends (24) and input selection, level, routing and panning. Directly above the faders are the access panels (Figure 1) that cater for eight inputs and provide individual channel access, mono AFL, PFL, stereo solo, mute and automute.

A 4-character alphanumeric display per input enables information relating to that input to be memorized and loaded from disk. A conventional writing strip is also provided below the display for additional information that does not need to be memorized.

Above the central control panels and farthest from the operator are the routing indicator panels (Figure 2). Again, these are laid out in sections of eight and consist entirely of LEDs that indicate to which group the input is routed; which input is selected (mic or one of two line inputs); phase reverse; equalization and dynamics insertion (an optional noise gate/limiter per channel).

The input equalizer (Figure 3) consists of six individual sections. The first four sections are almost identical and comprise a 16-frequency parametric equalizer with digital display of center frequency, boost and cut buttons with bargraph display, a 3-position 'Q' (bandwidth) control and individual bypass keys. The last two sections are sweep high- and low-pass filters, which, again, have 16 frequencies with digital display in each range, a slope rate of 12dB/octave and individual bypass keys. Completing the equalizer section are four memory keys that enable up to four different EQ settings per console input to be stored for recall.

The auxiliary send panel (Figure 4) is divided into 12 identical sections providing level control, pre/post fader selection, mute and the ability to re-route the auxiliary sends to provide up to 24 sends from each input. Aux sends can also be ganged together to form 12 stereo pairs, allowing the stereo balance to be set by adjusting the adjacent level buttons for correct perspective. When ganged, the aux-send level controls will track together, thereby maintaining a stereo image. Similarly, either mute button will affect both sections. An LED between each pair indicates when the stereo mode is selected.

The final panel in the input section is the mic/line and channel routing section (Figure 5), the left-hand section of which provides selection between three input sources: mic, line 1 or line 2. Line 1 would normally be the appropriate multitrack return, while line 2 could be used for a synthesizer or any other line-level input.

Below the mic input key is a high- or low-impedance select button; set to the high mode, it allows for the direct injection of a high-impedance musical instrument, such as a guitar. A digital display indicates the level set for the appropriate input, and can be set or adjusted via the up/down keys.


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By depressing an auto-gain key, the input gain will be set by sampling the incoming signal for a predetermined period and then automatically adjusting the gain of the input amplifier so that the level is set 5dB below peak amplitude. A master auto-gain control located in the console makes it possible to adjust all input levels simultaneously.

Other facilities in this section include phase reverse (on all inputs); bypass of the insert point; selection of the equalizer pre or post the insert point; phantom power on/off; and the ability to meter the input level at any one of four different places in conjunction with a 100-segment bargraph display situated on the meter overbridge in-line with each fader.

To the right is a level indicator that reads in tandem with the appropriate bargraph meter, thereby saving the engineer from having to look away from the panel in order to check input levels. This central meter can also be switched between VU or PPM ballistics; above it is an input overload indicator that sums all four of the input level monitoring points.

To the extreme right of this panel is a 32-way keypad for selecting an input to any combination of output groups (panning being between odd and even pairs). Separate keys select the input to the master monitor/remix outputs. A 9-position pan control with LED indication of stereo position also is provided.

The proposed console is equipped with a separate 32-way monitor section that is, in many ways, similar to the input section. Thirty-two long-throw faders are provided (these normally controlling the monitor levels) together with an access panel (Figure 6) that provides monitor access, stereo solo, mute, group/tape monitor switching plus a large, illuminated record-ready button. In conjunction with the recorder's master record button, the record-ready button will select any individual track-to-record mode. A programmable dot-matrix display of four alphanumeric characters per track completes the monitor access panel.

One master panel (Figure 7) provides all of the necessary monitor facilities. Six auxiliary sends are featured (and which can be rerouted to become 12) with pre/post selection and mute. Also provided is a 3-band equalizer offering high and low shelving with selectable turnover points in each range, plus a midrange section with digital frequency display from 700Hz to 7.2kHz, and three

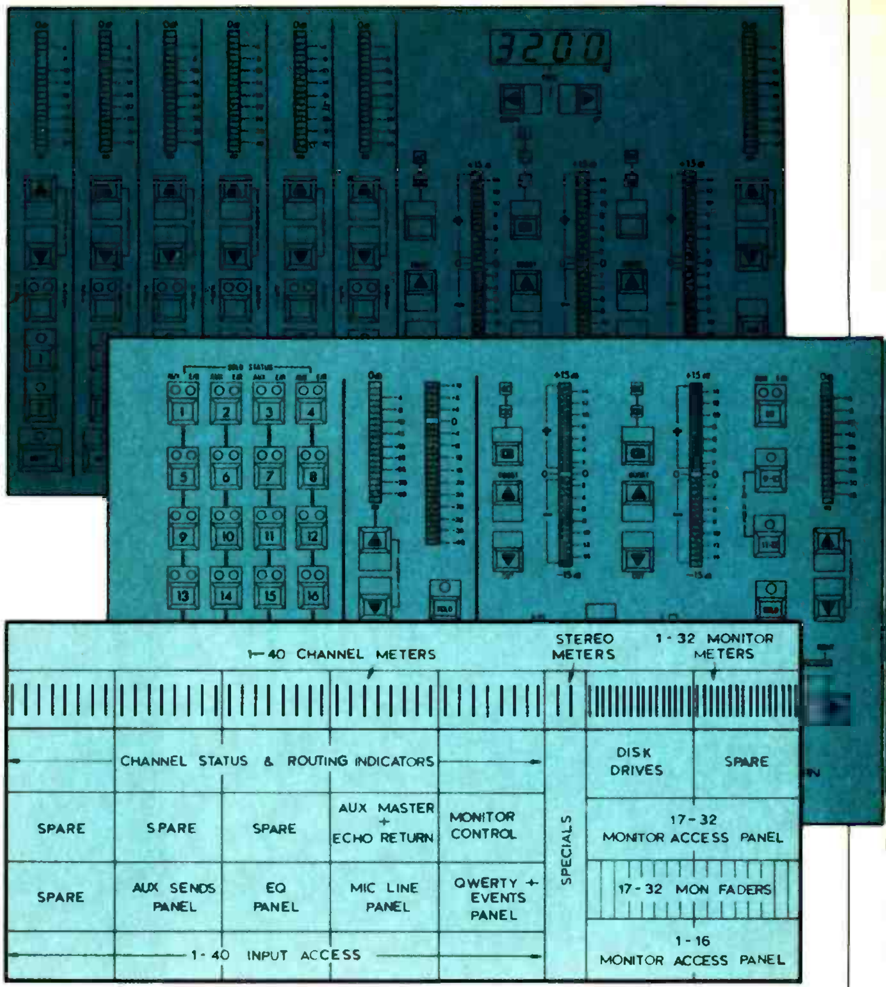

Figure 7. *(Top) A master monitor panel features controls for six auxiliary sends including 3-band EQ controls.*

Figure 8. *(Center) A single panel provides selection and assignment of aux sends and echo returns, with a dedicated EQ section.*

Figure 9. *(Bottom) Front panel layout of a 40/32 assignable console, showing location of relevant control panels and overall dimensions.*

bandwidth settings with overall bypass and four memories. Monitor panning is provided by means of up/down keys combined with a bargraph display of level and mute.

Because monitor levels usually are adjusted by the long-throw faders, an automated monitor mix becomes possible. Similarly, when the monitor section is used to provide additional line level inputs during mixdown, these can also be automated. (A fader-reverse button per monitor section allows the group-output level to be adjusted by means of the long-throw faders).

The final central control panel handles aux-master/echo returns (Figure 8). Simultaneous access to the appropriate aux master and echo return is provided via a 24-way keypad, a digital display indicating which one has been accessed. Above the display are up/down keys to set the aux-master level, with LED bargraphs to indicate level adjustment and actual levels. A high and low shelving equalizer can be selected to either the aux-master send or echo return; like all other equalizers in the system, it is also provided with a 4-memory capability. The echo-return section has a level and pan control, solo and the ability to route the return signal to aux sends nine through 12 for echo-to-foldback (cue).

Additional software will also provide display of data, such as track-sheet information on an external monitor.

As can be seen from the console's overall layout (Figure 9), it is of similar proportions to a conventional mixing desk. However, the operational hand span is considerably reduced which, we feel, is significant considering the amount of additional facilities offered. R-E/P

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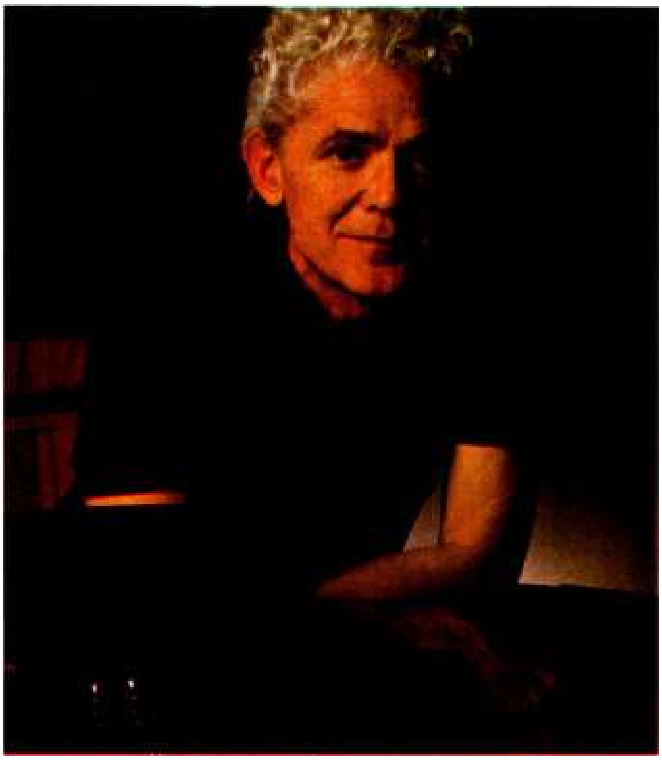

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Design Considerations for a Digital Recording and Editing System

The next generation of random-access editing systems will have to pay close attention to front panel ergonomics and layout.

By Charles Bagnaschi

The increased dynamic range and audio accuracy of Compact Discs, PCM tape and other digital-storage media are by now well established. The wide acceptance of CD as a high quality domestic format has raised audience expectations for various types of audio reproduction, which in turn has spurred the use of digital recording in film and video soundtrack production.

At the moment, however, the pro audio industry must sacrifice both productivity and creative capability for the higher quality of digitally stored audio.

Editing, for example, is a major problem when using current tape-based digital storage methods. Dual-deck editing and assembly systems are cumbersome and slow compared to analog tape. Although some systems emulate the conventional analog technique of "rocking" or "scrubbing" tape to locate cue or edit points, their low dynamic range and restricted bandwidth in this mode of operation make precise location rather difficult.

Other digital formats use an analog cue or auxiliary track to preview edits, with a replay quality that varies from one system to another, even from one edit to another. Certain systems allow razor-blade editing of the digitally encoded tape, but achieving consistent, accurate results demands carefully controlled operating conditions.

It has been apparent for some time that these problems could be eliminated by combining digital recording and storage with random-access retrieval. Recent developments in digital technology have brought random-access digital audio to a level of speed and cost effectiveness that merits serious consideration by the film and video production industries.

Essentially, random access is the ability to specify any piece of stored data for virtually instantaneous retrieval. The length of the "virtually instantaneous" retrieval interval is defined by the task to be performed. A word-processing system, for example, provides random access with a longer retrieval time than would be acceptable in an audio system.

Random access is dependent, to some extent, on the storage medium used. Although various media are possible, at the present time, the one most suited to low-cost, high data-rate transfer of large

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amounts of digital information is the hard or Winchester disk.

From concept to reality

The following discussion will make clear that it's a long way from the concept of random-access digital audio, or even from the development of cost-efficient, reliable and sufficiently rapid storage and retrieval methods, to a practical, realistic system. For random access to become more than a technological buzzword, it must facilitate the unrestricted manipulation of digitally encoded audio within integrated systems. Such systems should combine the enhanced audio quality of digital storage with the flexibility and creative freedom of highly evolved modern analog methods.

Consider, for instance, the impact of random-access techniques on the editing process. Random access allows digitally recorded audio to be organized in discrete segments that could be defined by beginning and end time code points. The segment would then be assigned a name by the system or by the user. A segment could be assembled from other segments, or one segment split into several smaller segments.

Random access allows the added flexibility to define a "reel" as a group consisting of any number of segments. In addition, material could be moved between reels or even between "jobs."

One advantage of organizing digital data as a hierarchical system of jobs, reels and segments is that this type of nomenclature already exists in the analog world, and is familiar to post-production engineers.

To maximize the efficiency of data storage and minimize the need for re-recording, editing would then be implemented as a series of pointers and instructions to replay specified segments. In this way, the integrity of the original material is always preserved during the random-access editing process.

Because the original digital data can be destroyed only by user command, multiple edits can be performed on the same material, generating as many edit decision lists (EDLs) as desired. These could then be stored at the end of the work session, transported on a floppy disk, for instance, and revised at will. Thus random-access editing allows a producer to postpone the final edit until the material is transferred to a storage medium (such as analog tape, Compact Disc or mag film) outside the system.

Again, the key to realizing this potential of random-access technology is the editing function. Unless edit points can be located and marked quickly and accurately, the usefulness of EDL and non-destructive editing capabilities are bound to be severely compromised. Editing on analog tape is a highly evolved, efficient and effective process, and any practical random-access audio production system should respond to the ear/hand coordination already developed by experienced audio editors.

The key to this is full-bandwidth "scrubbing" of the actual digitized audio so that clicks, pops, or "ess" sounds can be located with precision, and edits previewed with complete reliability. This represents a significant refinement of random-access technology, since information retrieved at slower rates from the magnetic disks requires some very clever digital processing to make it suitable for D/A conversion.

Although central to the usefulness of a random-access audio system, non-destructive editing is by no means the only function such a system could perform. Most signal-processing functions are now, or will soon be, possible in the digital domain, eliminating a 3dB increase in noise and added distortion incurred with each A/D or D/A conversion.

Recording, mixing, non-destructive editing, looping, panning, overdubbing, reverb, time compression/expansion, time alignment, equalization and dynamics processing are currently available in systems or in discrete digital processors. Other functions, including noise processing, restoration and specialized synthesis, must wait for additional applied research.

Hardware considerations

Having delineated some key functions of an integrated storage/editing system, an engineering team must decide how to generate the required computational power. A single large CPU such as a minicomputer represents an obvious solution, provided the designer is willing to work within the constraints imposed by its architecture. A less restrictive alternative is created by the distributed processing approach, using a number of smaller CPUs. This design approach allows greater freedom to optimize each individual processor for its intended function (signal processing, digital filtering, mixing, reverb, etc.). The inherent ability of a distributed processing network to simultaneously perform multiple tasks makes it much easier to obtain the processing power and speed necessary for various computation-intensive, digital signal-processing functions.

In expanding the range of tasks that users expect a random-access system to perform, we are moving beyond simple storage and retrieval into the area of a fully integrated digital audio production system, which uses random-access technology to streamline and facilitate conventional tasks, as well as to perform new ones.

Distributed processing is particularly useful in integrated systems because the individual processors can be networked using a high-speed data bus specially designed for passing information packets. Within that network, any operational node could be patched to any other node or group of nodes, lending a further degree of flexibility to the system. And, as additional processors are developed, the system can be expanded or updated without being entirely revised.

Figure 1. *Simplified block diagram of a random-access digital recording, editing and processing system.*

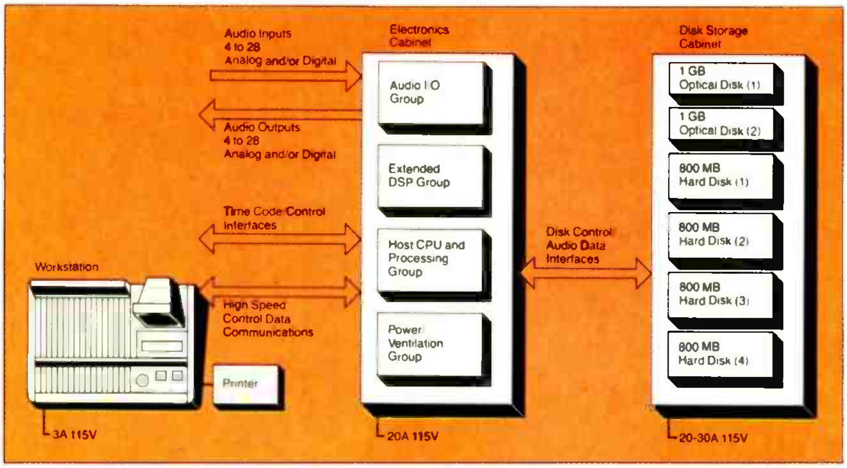


Figure 2. The proposed audio production workstation incorporates elements that control audio levels and editing commands. (See Figure 3 for a close-up view of a typical "channel strip," and Figure 4 for details of the editing controller section pictured lower right.)

Because a system based on distributed processing allows each processing node or group of nodes to be designed more or less independently, it would be useful before fully configuring the electronics to examine the likely applications for such an integrated system. These could include sound effects library storage, assembly and pre-lay, sweetening and pre-dubbing. A two-person mono dialogue track might be split into two independent tracks for separate processing or replacement. Dialogue replacement could be enhanced with powerful preview capabilities, instant access to all replacement takes and a virtually unlimited number of tracks.

Obviously, the entire post-production process, including routine operations such as editing, slip syncing, crossfades, background loops and other tasks, could benefit from the speed and flexibility of an integrated production system using random-access techniques. Digital masters also could be assembled in the digital domain as an EDL without the need for copying the original material.

A double engineering challenge is presented by this "form follows function" design approach. A range of digital signal processors must be developed and linked. In addition, they must be interfaced to the end user via control capabilities that facilitate the tasks.

Figure 1 shows a simplified block diagram of one possible random-access system design. Separating the disk-storage, electronics and control workstation provides a number of advantages. First, it allows the user to enlarge or reduce storage capacity and processing power independently of one another, thereby enabling each system to be optimized for a specific application or range of applications.

Overall system expandability is provided both by the modular structure of the hardware, and by well-defined bus interface standards. Well-structured, portable software is also important in providing for future system revisions and upgrades. (These approaches are not unlike the upgrade paths defined by computer manufacturers when introducing new system generations.)

Human interface

Of course, the key to making a realistic and practical random-access audio production system is the design of its control surface. The control design must strike a balance between the new possibilities offered by random-access digital technology and familiar control methods that maintain, as much as possible, the intuitive link between ear and hand.

One option is to mimic the layout of a conventional recording console. Although such a design would be a completely familiar and instantly usable, it offers no advantage over existing work methods because it is restricted to the functions it duplicates.

Since the design concept involves the use of a sophisticated, specialized computer, an obvious second alternative is to configure the control surface around a CRT display screen and alphanumeric keyboard. This approach allows the implementation of new control functions, but at the price of eliminating all existing ones, including those that may be the optimum methods for performing certain tasks. Thus it unavoidably changes the nature of the editing process.

The ear/hand link is replaced by an eye/mind information loop in which the operator receives primary information from a CRT screen and must remember a large catalog of complex keystroke sequences to access appropriate control functions. Such an approach would be logical for computer operators and data-processing engineers, but inappropriate if it requires a total reorientation for sound editors and producers.

As a first step toward designing a truly effective control interface, it would be useful to study end users job performance and monitor existing technqiues for efficiency and possible improvements. Our product development team spent a considerable amount of time observing and interacting with audio editors at work. We also presented some of our early ideas in technology demonstrations of full-bandwidth audio scrubbing and other basic control functions involved in audio editing.

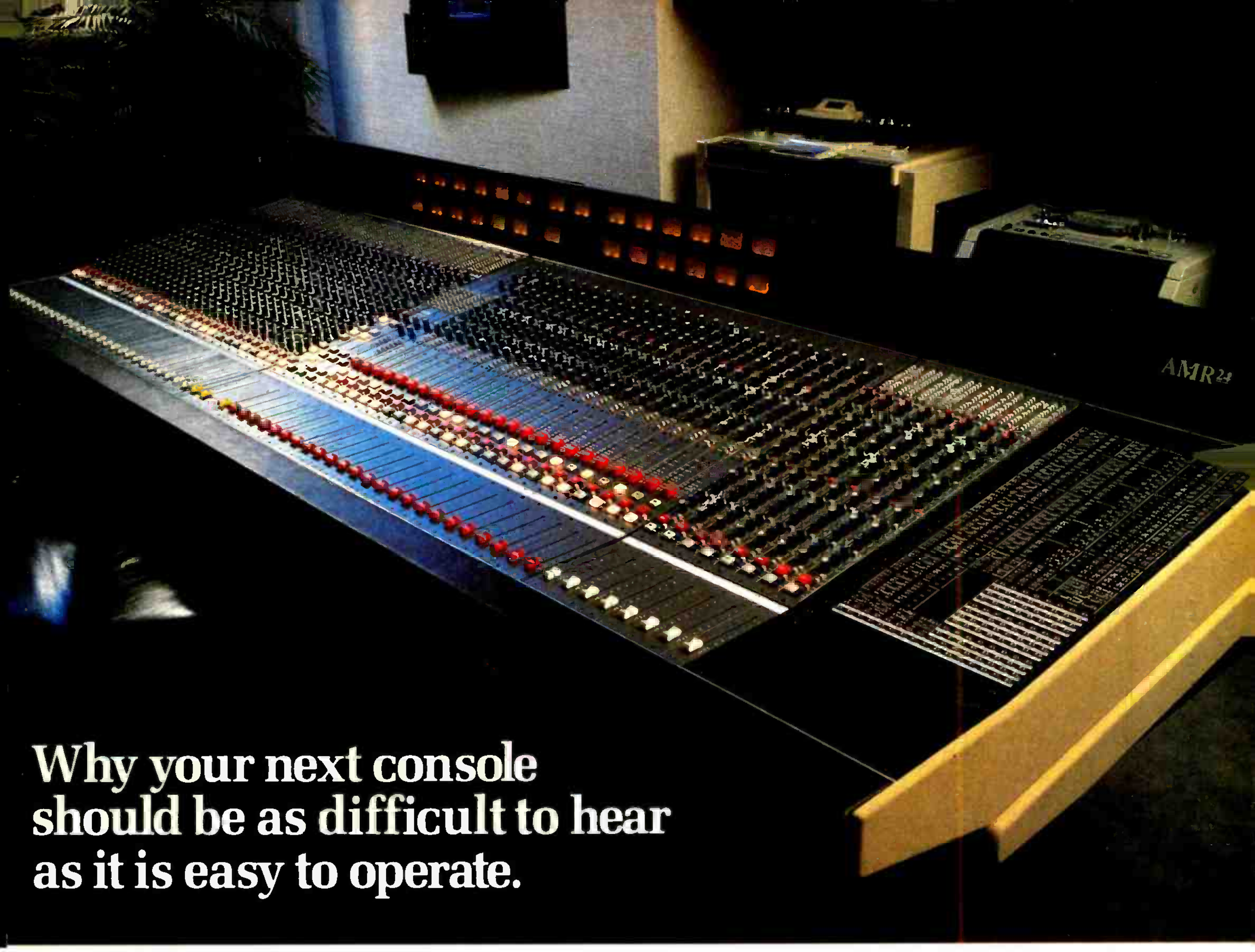
While individuals offered different suggestions, there was universal agreement at these sessions regarding the concept of marrying certain elements and functions of the conventional mixing console/recorder pair with a CRT screen and keyboard. This approach allows the incorporation of appropriate features and functions from audio mixing consoles, video editors, digital audio editors, CAD/CAM terminals, digital video-effects systems and computer science/information display technology.

Control-surface topography

The design of a control interface or workstation should allow signal handling to be optimized for both control flexibility and audio quality. These criteria are met by a "dry" control surface that acts strictly as a multifaceted remote control, and processes no audio at all. Such a configuration enables unconstrained optimization of information flow from the user to the system, and vice versa.

Although familiar manual controls such as faders, buttons and knobs are used extensively in the design shown in Figure 2, they operate only as remote controls for processing computers located in the equipment racks, not as switches or faders for the actual audio signal. This configuration also allows the most effective isolation of audio signals from digital-control elements.

One example of optimized information flow in the workstation is the random-access storage/retrieval process itself. This can be driven by conventional "tape-recorder" buttons labeled play, record, stop, fast forward and rewind (Figure 3). In addition, during the entry of EDL data, the same information can be sent to the digital processors via a



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:

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numeric keypad or set of function keys. These methods facilitate direct time code entry for dialogue replacement, stepped movement or cue-point location.

Retaining the ear/hand editing process is a worthwhile design goal for control surfaces, as well as for the processing electronics. This example uses a large knob that mimics "rocking the reels" of a conventional analog tape machine. In jog mode, the knob scrubs the digitized audio at a speed and direction proportional to the knob movement with full bandwidth and dynamic range; in shuttle mode, it acts as a variable speed fast forward and rewind control. Four buttons are located above the knob control marking and preview functions; they are "soft labelled" and change function according to the operational context.

The workstation illustrated here offers different ways to input the same information, allowing the user to select the fastest and most comfortable method of inputting. Information flow from the system to the user is also a primary design consideration. In this example, plasma meters for both signal levels and automation settings, along with "soft" control-function labels, are used with conventional lights and status indicators similar to a conventional console.

However, a normal console does not provide all forms of necessary information and signal control in a smooth, intuitive fashion. A CRT screen is used in the representative design to fill in these information gaps. To eliminate "paging" through software modules, and waiting for corresponding CRT displays, the screens could be kept to a minimum and made context sensitive. Depending on where the operator is in the system and what the operator is doing, the screen would automatically display appropriate information for the user.

Since most editors are unaccustomed to working with a computer screen, it would be unadvisable to require them to do their work on screen all the time. However, the listing of events by name, time code location, or both, is part of virtually every post-production job, a function that can be done on a CRT screen, not scraps of paper.

A standard QWERTY (not a "speak and spell") keyboard is used in the proposed design primarily for labeling and log-on information, and not as the sole means of entering command sequences. A further advantage to this process is that once the list has been entered, operations such as insertion of replacement dialogue can be performed automatically. The overall design goal should be to implement a natural command syntax using appropriate combinations of hard- and soft-labelled command keys, in conjunction with control elements like those found in conventional consoles (Figure 4).

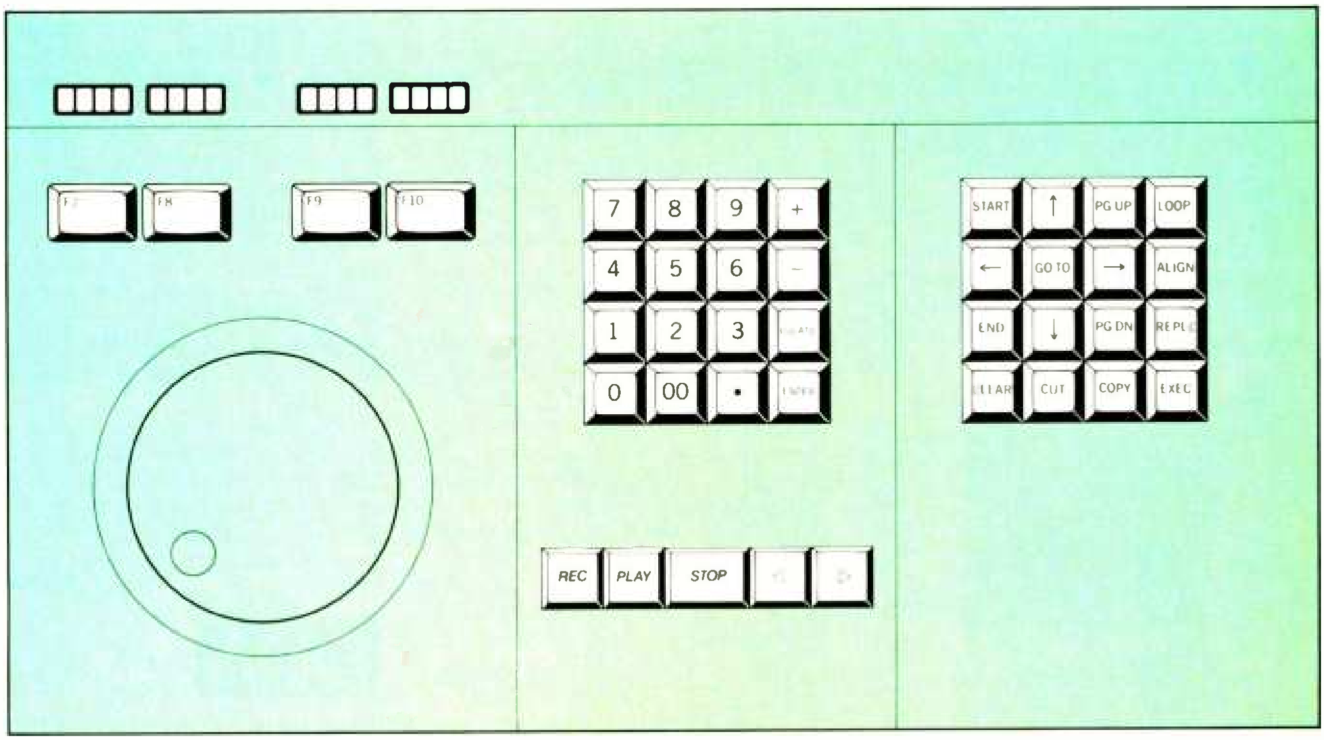
By no means the least significant advantage of an all-digital system is its inherent capability for 100% automation. This could take the form of both static automation (storing and resetting the console configuration, signal routing, crossfade parameters and other controls for each job or user) or dynamic automation (recording of all operator interactions, including all level adjustments and switch actuations).

Digital-to-digital interface

For today's working environment, connection to external analog and digital audio signals is an important factor in the design of any production system. Conversion between analog and digital domains has been refined over the course of several years, with contemporary thinking being to use 16-bit or greater PCM devices capable of operating at all standard sampling frequencies. Oversampling schemes are useful for minimizing the effects of sharp cutoff filters.

Direct-digital interfacing is of particular interest with a random-access system, because it allows the use of extertemal digital effects processors without

Figure 3. An editing controller would provide conventional "tape-machine" transport control buttons, as well as a keypad and dedicated keys for entering EDL and command instructions.



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Circle (17) on Rapid Facts Card

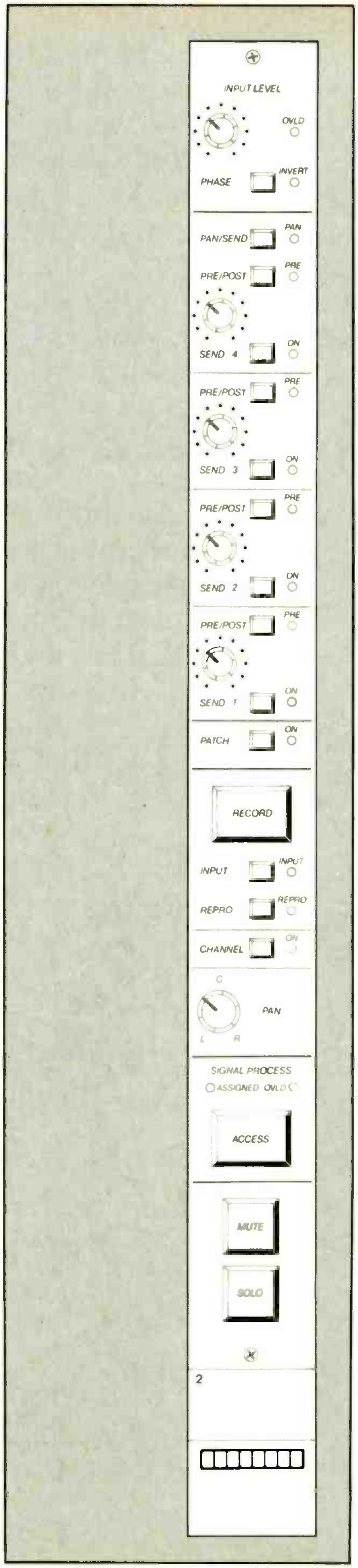

Figure 4. *Although front panel features, including faders, switches and knobs, would be familiar to uses of conventional analog consoles and editing systems, they would comprise control elements for digital processing electronics located in separate equipment racks.*

A/D or D/A conversion, or consequent signal degradation. Such connections could also be used to load and off-load digital program material. By installing a number of interface modules, various digital formats can be accommodated.

Storage schemes

To return to our starting point, all the benefits of random-access capabilities depend on an appropriate storage medium. Magnetic hard disks have been developed that store up to 120 track-minutes of audio per disk. Write-once optical disks can add considerably to the utility and time saving potential of a random-access system. Possibilities include off loading newly recorded audio data and edit commands in the background while the operator is working.

Optical disks can also be used on their own for off-line transfer functions. While Job A is in progress, the source soundtrack for Job B could be recorded on a second optical system, for later downloading into the main system. Archives of material such as sound effects libraries could be stored on optical disk and loaded into the system as needed.

It is already possible to retain audio in a digital format throughout the entire chain from recording, through production, mastering and distribution to the consumer, a process that has been achieved without enforcing radical changes in the way audio engineers exercise their craft. Random-access editing of digital audio is certain to become the preferred approach, not only for productivity and creative freedom but for preservation of the source material's audio quality.

Some of the random-access digital audio systems now becoming available blend traditional audio production methods with the best of current man/machine interfaces. This ongoing dialogue between users and developers of such systems will clearly result in further evolution. R-E/P

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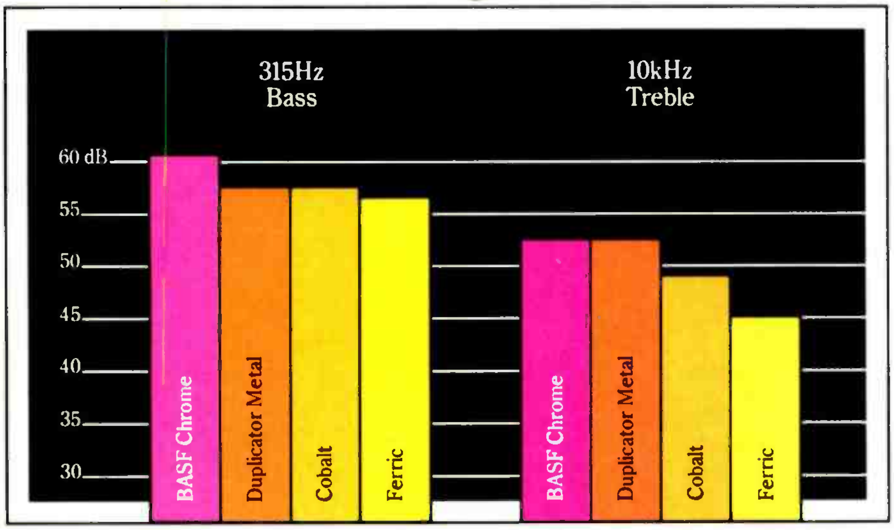
Bullets. Targets. And Dynamic Range.



DYNAMIC RANGE is the spread between maximum output level (MOL) and noise (tape hiss). It is a major criterion of tape quality because it shows the true capacity for music. Tapes with high output and high levels of tape hiss are really no better than low output tapes with low noise. It's the difference between output and noise that matters.

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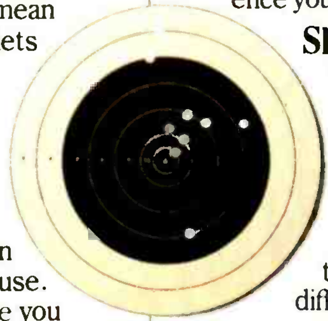
But talent and hard work alone won't get you that bullet. Because in the end, you're only going to sound as good on cassette as the tape you use. So reach for the best tape you can lay your hands on.



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Chrome on the range.

The chart shows the dynamic capability of tapes at critical frequencies in the musical spectrum. Dynamic range is the room available for music between the limits of tape distortion and hiss. The more room the better. And over the full musical range, BASF Chrome is obviously—and audibly—superior to even the most highly acclaimed alternatives. BASF Chrome tape comes closest to the original studio master.

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BASF

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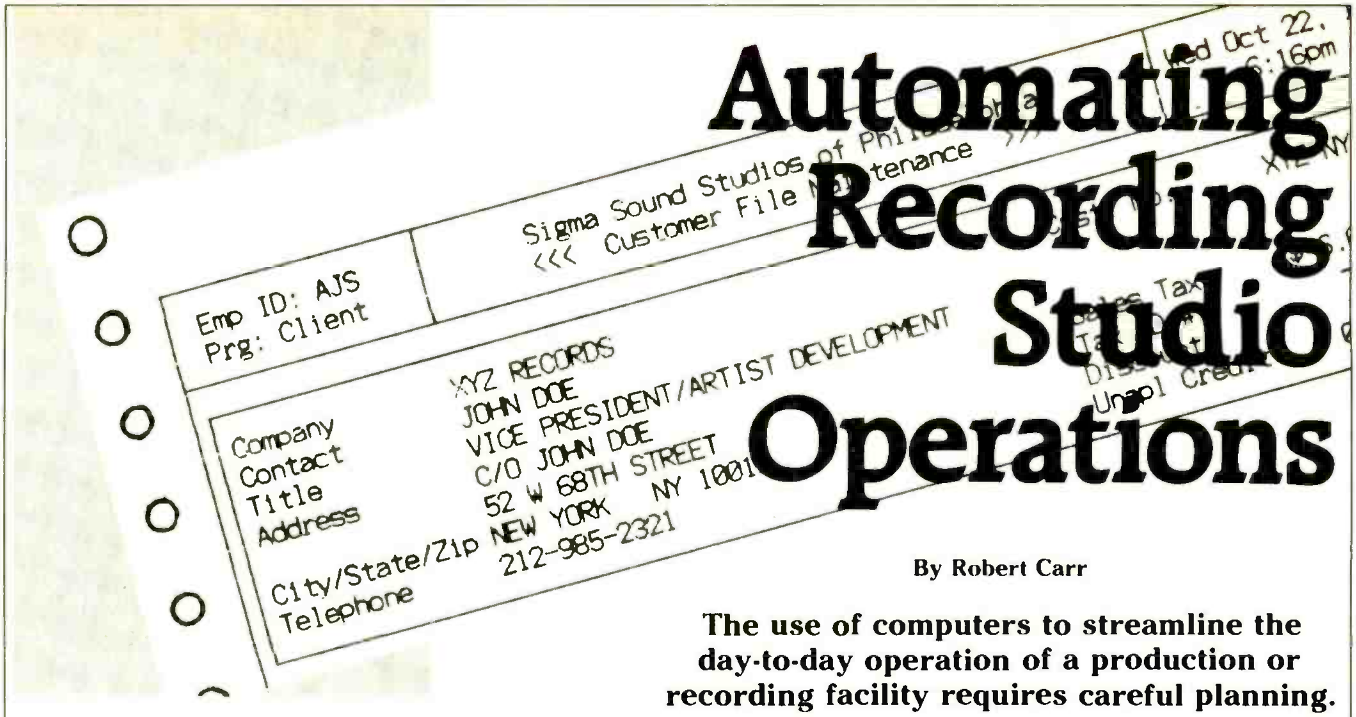
Automating Recording Studio Operations



By Robert Carr

The use of computers to streamline the day-to-day operation of a production or recording facility requires careful planning.

Virtually every aspect of modern society has felt the impact of the personal computer and the pro audio industry is no exception. Besides the use of specialist software packages to control MIDI-equipped synthesizers and console automation systems, many production facilities are also investing healthy sums of money in office automation to streamline their financial, marketing and inventory functions.

Personal computers are signaling an end to the conventional ways of running a business. Yet for the studio on a budget, or for owners/operators who are not computer-literate, computerizing office functions does not have to be threatening or financially out of reach. Prices of hardware and software continue to fall and with the right planning and a list of realistic expectations, a facility can be on its way to successful business automation faster than one might think.

Analysis of needs

Planning is the essential foundation for any projected move into office automation. You probably didn't build your studio and control room without a blueprint of some kind, so don't expect to put together an automation system without developing a step-by-step plan. Although the plan doesn't have to be complicated, it should be thorough.

First, take a serious look at your business situation. Is your local marketplace relatively dynamic and full of opportunity, or is it capable of supporting only limited growth for the next five to 10 years? In comparison, how much do you expect your facility to grow in the next one, two, five and 10 years? How will client traffic and cash flow change over that same period of time?

Answers to basic market analysis questions such as these can justify or destroy your decision to automate. They also form the guidelines that determine the storage capacity and processing capability needs to support your business now and in the future.

Once you get past this first test, you're ready to look at a whole new set of questions. For example: What tasks use the most time in your studio? Retyping correspondence? Preparing periodic financial reports? Tracking various inventories or vendor bills? Scheduling studio time? Generating invoices and balancing your books?

All of these operations are ideally suited to the natural abilities of a personal computer, regardless of the size of your operation. Making a complete list of everything you expect from a computer system will give you an idea about the type of hardware and software you need to buy.

If you find that your budget is not sufficient to take care of all the items on your list right now, use the list to direct yourself to the one or two packages that can automate the largest percentage or the most important aspects of your workload. (See the accompanying sidebar for general software categories and their applications.)

Hardware considerations

Unlike audio hardware, computer systems are secondary to the software's capabilities. Once you've determined the major tasks to be tackled, find software packages that fulfill these tasks in the way that's right for you. Only after you've found the right software package can you make an intelligent choice

Robert Carr is a special-projects consultant with software publisher Ashron-Tate, and specializes in the development of office automation systems. He was a professional musician for 17 years and has been a free lance writer since 1979.

General Categories of Software and their Applications

***Word processing** programs generate and manipulate text material such as letters, reports, invoices and instruction booklets for new techs.*

***Spreadsheet** programs calculate and manipulate numbers for financial reports or projections.*

***Database** programs provide filing and organizing of client files, equipment inventories, vendor lists, studio rate charts, parts lists and work schedules. If the database program supports arithmetic functions and has its own programming language, it can be molded into a customized accounting package or tracking system for calculating studio hours, scheduling, generating invoices, and tracking tape stock inventory.*

***Telecommunication** programs facilitate the transfer of data between a computer and remote sites. Such software provides access to databases for confirming client credit, arranging airline reservations, communicating via electronic mail, and searching for the most recent research on topics like acoustics, audio, management, music and computers. You can even send or receive MIDI information.*

REINFORCEMENT: *THE NEXT GENERATION*

When the Music Store Mixer Won't Cut It

The simple fact is, all the other PA consoles available today lack the processing, monitoring, and routing capabilities that today's touring acts have grown to expect in the studio. The WHEATSTONE MTX-1080 is the reinforcement console that PA mixers have been asking for. It's loaded with features, like programmable muting; 8 effects send controls (each with pre, post and off functions, programmable to pre-fader or pre-EQ); four-band sweepable equalization with switchable Q and peak/shelf modes; tunable HPF; separate electronically balanced mic and line inputs (transformer balanced option available); XLR direct channel outputs; and channel, subgroup and main output insert points. Of course, the console also has eight 11x1 input matrix mixes (up to 16 are available using optional matrix expander modules). Mainframe size, module complement, group placement and aux zone control modules are configured per client specifications.

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regarding what hardware (computer, monitor, disk drives, printer, plug-in peripherals) will best support and complement the software.

However, one hardware decision that you must make very early in the planning process is whether to adopt a single- or multi-user approach. A single-user system can be as simple as one computer doing all the work. Or you can have several stand-alone computers, with each computer station dedicated to only one or two types of tasks. By purchasing several of the same model of computer, you can exchange information among the various stations by simply transporting the data on a floppy disk.

If you have different kinds of computers for performing different tasks, then data can be transferred from one system to the other using a telecommunications software package. For small facilities, the single-user approach may be the most cost-effective.

If you want the convenience of having all the data available to all users at all times, then you must invest in a multi-user system, such as a local area network (LAN) or terminals connected to a centralized mainframe or mini-computer. Although all the same criteria apply to both systems, the complexity increases substantially in a multi-user environment.

Other important considerations when purchasing hardware include processing speed, storage capacity, vendor reputation and expandability.

Current computer designs are usually replaced by new technology quickly, sometimes in a matter of months, so expect that whatever you buy to probably be obsolete in 18 months. However, the machine you buy should serve you well for at least three to five years without any major problems. Your local computer dealer can provide details of what's presently available, and the costs involved to get you up and running.

Software options

A typical personal computer is capable of running a wide range of software packages in each of four categories: word processing, spreadsheets, database programs and telecommunications. Some software even comes with its own programming language so you can customize the existing package to do anything you want.

Generally, a successful automation system is considered to be one that speeds up the processing of information and makes the business' existing procedures more cost- and time-efficient. If the software does the intended job in a sophisticated way, but forces the users to relearn new or unusual ways of doing a task they already know, the program is probably a poor choice for that particular application. Programs that appear to operate in a "non-intuitive" way usually cause frustration and a lot of anxiety. Not only do users have to learn how to run a computer, they also have to relearn the way they're supposed to do their job.

The best advice is to involve the prospective users during the planning stages: Ask them what features they want, and what capabilities will help them do their jobs better and easier. If they feel as though they're an integral part of the shift toward office automation, they'll be more inclined to support the system once it has been installed.

Finding solutions

There are three ways that software can solve your specific business requirements: you can use the package just the way it is, customize the package, or write a program from scratch.

Off-the-shelf software: This approach is probably more than adequate for most 1- and 2-room studios, especially for tasks such as word processing and spreadsheets. Most packages can handle just about anything a small business might need. However, more complex applications, like accounting and database functions, may require one of the following approaches for your business.

Customizing off-the shelf software: Even if you cannot find the perfect software, you may be able to purchase a package that's close to meeting all of your needs, and then customize it to meet your specific requirements. Some programs utilize a proprietary programming language, which give the user virtually unlimited potential. In most cases, however, you first have to learn how to operate the basics of the program. Once you have a firm grasp of the underlying concept and structure, you can then proceed to learn enough about the program's native language to make the necessary changes.

If you choose to customize, check two factors: First, that you have the legal right to alter the program, and that you have access to the source code or programming language. Current software agreements state that you are only buying a license to use the program. You don't own it. To alter the program, you

Office Automation Software for Recording Studios

*Although the market for office automation software for studios is reasonably young, included are details of three representative examples of packages specifically for the recording industry. Caveat emptor: the information presented here was extracted directly from each company's promotional literature; none of these programs have been tested by **RE/P** staff.*

• ***Dataline's Automated Studio Management System*** *runs on IBM PC and AT or 100% compatibles equipped with 512K RAM and a 30Mbyte hard disk. The system handles the various kinds of information needed to schedule and bill recording sessions by centralizing business data. Areas covered include client data and project management; studio booking and usage; price quotations; invoicing and accounting; inventory and pricing; and tape library. The system integrates these functions to track a session from booking to final billing.*

Circle (103) on Rapid Facts Card

• ***Pristine's Recording Studio Management System*** *runs on MS-DOS-compatible PCs with at least 256K RAM. A hard disk is recommended for facilities running three or more rooms. The package is said to allow a studio to be managed from the initial phone call for a booking, through the billing process and to the financial statements. It provides reports of studio utilization; unbilled work orders; customer sales analysis; accounts receivable aging and trial balance; inventory status; tape library; accounts payable aging and trial balance; and financial statements. Each module is available separately or as a total integration solution.*

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• ***Studio Master System*** *runs on an Apple Macintosh with 512K RAM and is designed to provide an "electronic snapshot" of console faders, panning, equalization and send levels. The package also provides documentation of all studio usage: billable, non-billable, maintenance and downtime conditions. The invoicing system calculates session time to the minute, and bills can include all materials, extra equipment, miscellaneous out-of-pocket expenses and applicable sales taxes.*

Circle (105) on Rapid Facts Card

may need written authorization from the publisher. Secondly, the program may not allow access to the source code, which means you wouldn't be able to alter it regardless of the legal rights you had.

Writing a program from scratch: In the event that there is no software on the market that meets your needs, your only option may be to write your own. Although you have a good chance of getting exactly what features you want, you also open up a new set of problems.

Keep in mind that system design is not something you pick up overnight; it's as much of a craft as audio engineering or disc cutting. Who will design the system to ensure maximum efficiency and lowest chance of failure? What programming language is optimum for the applications you're developing? (Each language has strengths and weaknesses, such as speed, ease of use and memory requirements, that will influence this decision.) What are the hardware requirements for the language you choose? Do you want a compiled program or one that can be altered while it's being used?

Should you prototype the projected program, and how much of the program should be prototyped prior to actual coding of the finished program? Should your prototype and finished program be written in the same or different languages? Who will be responsible for initially debugging the program? Who will document the original code and all the eventual changes in the program?

Who will produce training materials and reference manuals for the users? What kind of documentation and manuals should be produced? How will your staff be trained? Who will service the program when hidden "bugs" turn up one or two years from now, and who's financially responsible? Who will answer user's questions when the answers are not in documentation?

These are just a few of the questions that need to be dealt with before you decide to write your own software.

If you don't feel comfortable with sorting out all the particulars by yourself (and most people don't), you can always have a professional programmer to design and write the programs for you. But check out the programmer's track record, and try to assess their business stability. In other words, will they be around to support you in a year or two? If your system goes down during business hours, you're going to want someone to fix it fast.

Even with flawless documentation, a second programmer will need time to become familiar with the overall concept and operation of your system, not to mention the time required for them to decipher the original programmer's logic and style of coding.

Fortunately, another alternative is available: Hire a computer consultant that has extensive professional contacts to provide all these services to you. A reputable consultant should be able to efficiently direct your needs analysis and accurately formulate an RFP (Request for Proposal) to solicit system bids. They can also provide a link between you and the people that will keep you up and running over the long term, regardless of the migration of individual programmers and maintenance staff.

Keep in mind, however, that the consultant will be doing all your talking for you, and should have a good understanding of your business and its needs.

Remember, when automating your production facility, that there's no need to proceed quickly. Choosing an automation system is probably one of the most important business decisions you'll make. As your staff becomes used to the system, and it becomes integrated within the daily routine, you'll find that eventually it will become the business side of your studio. With that much responsibility in one place, the choice should be correct. And when it is, you'll wonder how you ever got along without it. R-E/P

Microprocessor control, amorphous metal core heads, and superior performance . . . technology that's typically AEG.

Chances are that the M-21 Professional Audio Tape Recorder from AEG will outperform whatever 2-Track you're currently using or considering for future purchase. No other machine is built to such exacting standards, no other machine handles tape as gently yet rapidly, and no other machine is presently available with Amorphous Metal Butterfly Core Heads. (Ours are standard equipment; ask about our exclusive head warranty.)

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AEG

Production Viewpoint: Russ Titelman

By Ralph Jones

As a staff producer and A&R executive at Warner Brothers Records, this busy songwriter/producer has seen success with a wide range of recording projects, including Steve Winwood's *Back In The High Life.*

Russ Titelman has spent virtually his entire adult life in the music industry, forging a career that has touched the creative efforts of a host of major artists.

At the tender age of 16, Titelman was already working with Phil Spector, playing guitar and singing on song demos and album projects. The young musician turned his hand to songwriting as well, and promptly landed a contract, through Don Kirschner, with Screen Gems/Columbia Music. While under contract, he co-wrote songs with Carole King, Gerry Goffin, Barry Mann and Cynthia Weill. Concurrently, Titelman's activities as a studio musician resulted in a long association with composer/producer Jack Nitzsche, with whom he worked on the film *Performance*.

In 1970, session colleague Lowell George formed Little Feat; Titelman brought them to Warner Brothers Records, where they were immediately signed. Co-producing the band's self-titled debut album with Warner producer Lenny Waronker, Titelman earned a position as staff producer for the label, where he has remained to this day.

In his 16 years with Warners, Titelman has assembled an impressive discography, including projects with Randy Newman (*Randy Newman Live, Sail Away, Little Criminals, Born Again* and *Trouble in Paradise*, James Taylor (*Gorilla* and *In the Pocket*), George Harrison (*George Harrison*), Rickie Lee Jones (*Rickie Lee Jones* and *Pirates*), Paul Simon (*Hearts and Bones*), George Benson (*20/20*) and Christine McVie (*Christine McVie*).

When **RE/P** caught up with the busy producer/A&R executive, Titelman had just completed production of *Back In The High Life*—Steve Winwood's first album in four years.

Ralph Jones is a free-lance producer/engineer and a regular contributor to **RE/P**.

RE/P (Ralph Jones): One of the distinguishing characteristics of your discography is that most of the people you've worked with are strong, highly individual talents: James Taylor, Randy Newman and Paul Simon, for example. Usually, such recording artists have firm ideas about arrangements and production when they begin a project. How do you deal with that as a producer?

Russ Titelman: I think that a producer's job under those circumstances is to try to get the most of the artist's musical personality on record. So, you just try to encourage as much input as you can.

In a way, a strong involvement with arrangement is part of production. Some producers are arrangers as well—David Foster, for example. David's thrust is a musical one, rather than engineering, and I think he provides a lot more musical input on his records than many other producers. With someone like Randy Newman, though, you don't need an outside string or horn arranger, because he does that so brilliantly.

All of the artists I've worked with have their own approach and personality. Generally, it's a collaboration and, if the chemistry is good, inevitably something interesting comes out of it.

RE/P: With most of these artists, is it safe to assume that you wouldn't necessarily be aiming for a sound that's currently popular? It's not a case of taking a raw talent and molding it to hit the Top Five.

RT: Yes, I see what you're saying, but because you *always* want to have a hit, you try to make something you think will be accessible. You also have to stay out of the way, however, if an artist *is* trying to hit the charts. James Taylor and Paul Simon have had hits throughout their careers. They're simply great songwriters and great interpreters of their own material.

RE/P: Is it a matter of personal choice that you've worked with these particular artists, or is it due to Warners' A&R philosophy?

RT: The label has a lot to do with it. Warner A&R is known for its unusual taste: the label favors artists who have a lot of respect in the industry—for what they do, and not necessarily for big sales. Mo Ostin and Lenny Waronker have a philosophy of staying with people who are great, even if they don't have big hits.

I think that this philosophy also may be a factor in drawing other artists to work with us: James Taylor, for instance, loved Randy Newman and Ry Cooder's records; Rickie Lee Jones wanted the same producers who worked with Randy Newman.

RE/P: Would you describe yourself a "technical" producer?

RT: No, I don't think so. I don't have a broad knowledge of technical equipment. I dabble in engineering. Sometimes, if I hear something that I feel isn't quite right, I'll change the EQ or ask for a certain kind of echo, that kind of thing. I have very *strong* ideas about how things should sound. Usually, however, I'll let the engineer do what he wants and then come in and change it around myself, if need be.

RE/P: Do you work with a regular session engineer?

RT: Actually, no. I've worked with a lot of different people, and I get new ideas that way. The earlier records that Lenny Waronker and I did were all engineered by Lee Herschberg. The Rufus record [*Live/Stompin' At The Savoy*] was engineered by Mark Linett, who was a Warner Brothers employee; it was mixed by Elliot Scheiner, who worked with Steely Dan. Christine McVie's album, *Christine McVie*, was engineered by David Richards, who was the engineer at Mountain Studios Montreux [Switzerland] and mixed by Eliot.

"Steve made a conscious effort to break away from what he'd been doing previously."

The new Steve Winwood album, *Back In The High Life*, was engineered mainly by Tom Lord Alge, a young engineer who works at Unique Recording, New York. He's a kid from New Jersey—23 or 24 years old, and just an exploding talent. He made a beautiful-sounding record. Tom has a lot of ideas; he comes out of dance mixes, and you have to rein him in a little now and then so that he doesn't go too far.

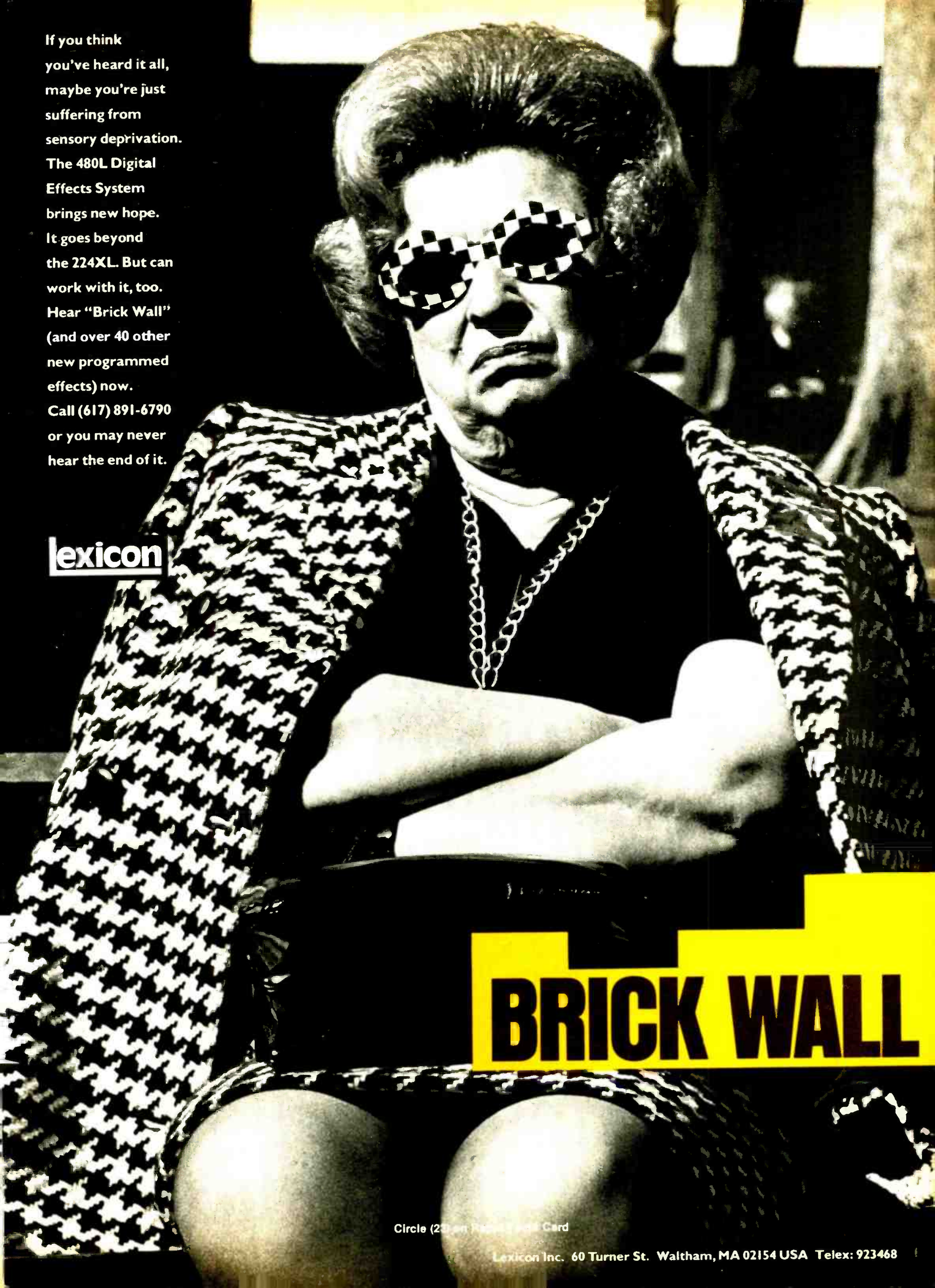
Steve and I were working at Right Track and Power Station with Jason Corsaro, but then Jason had to leave to work with Fleetwood Mac. We went to another little studio for a couple of weeks, but Robby Kilgore kept encouraging us to go over to Unique. He took me there one afternoon, and I was a little surprised; it's in a pretty funky building. I was going, "Robby, how could you have brought me to a place like this? I paid my dues already!" But we looked around and the studio was interesting.

As it turns out, it's really great there: things work, and the engineers are hip. The first day we went in, we worked with Chris Lord Alge, Tom's older brother. All this happened so fast. We were flying vocals from tape to tape, and things that often take a half an hour took all of two minutes. Then, we started working with Tom and we loved him. So, we just stayed.

RE/P: Tom has a strong background with dance remixes. Do you think that, because of his engineering influence, there was more use of effects on this project than on Winwood's previous albums?

RT: Not more; the effects are just used *differently*. I was listening to one of Steve's earlier solo albums, and he had multiple repeat on his voice, that kind of thing. But on this project, during mixing we would say: "Let's have a breakdown here." And Tommy would take, for example, the sound of the conga and key a sound from a keyboard off it. All of a sudden, there's all this sound. I mean, *we* would never have done that, but it sounded great!

RE/P: This latest project represents a real departure for Steve Winwood, on a number of levels. He's an artist who can, and has, produced himself. Yet, here you are producing him. He can play virtually any instrument with great proficiency, yet on this new album he's using a number of session players.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Russ Titelman with Steve Winwood during the recording of Back In The High Life *at Unique Recording Studios, New York.*

"The session was a wonderful, creative experience. I guess that I was, in part, an editor, a sounding board...I tried to make it easier, so that he wouldn't have to worry about the sound."

RT: Steve made a conscious effort to break away from what he'd been doing previously. I think he wanted to be able to step back a little bit, and let the responsibility go somewhere else. So Steve and I co-produced the album. Although we used a lot of other musicians, he still played almost everything. For example, he either sequenced or played all the bass parts and a lot of the keyboards; he played the Hammond B-3 on a few of the tracks; and he did an amazing guitar solo on one cut.

He also did more inventive arranging. I have a feeling that something happens when there's another musical force in the studio. It gives you different ideas, and spurs you on to do things that you otherwise wouldn't have done. I think that comes across on the record; there's a tremendous "spirit" there.

RE/P: Were there tracks on which other musicians played—and Winwood perhaps was influenced by—that didn't make it into the final mix?

RT: Yes. On one of the songs, T-Bone Wolk, Daryl Hall's bass player, came in and played. The part that he gave us just wasn't quite right, but there were a lot of good ideas on it, so Steve played it on a [Yamaha] DX-7 using a Macintosh sequencer. He used some of T-Bone's ideas, and worked on the part with him, so his influence is there. It's a very distinctive-sounding bass part.

RE/P: Can you describe your relationship in the studio with Steve Winwood?

RT: Great. The session was a wonderful, creative experience. I guess that I was, in part, an editor, a sounding board. For a lot of the time, I took the responsibility off of him—helped him with the vocals, that sort of thing. I tried to make it easier, so that he wouldn't have to worry about the sound. I think my contribution also was like a casting director: securing drummers like John Robinson and Mickey Curry, for example.

Steve's a little shy, sometimes. For instance, on his demo, he had indicated a rhythm guitar part on a couple of the songs, but he really didn't want to play it himself. So, I said, "Well, gee, that sounds like Nile Rodgers. Let's get him to play it." Nile came in and played this amazing part [on "Wake Me Up On Judgment Day"]. He took the idea that Steve had, and went another place with it.

RE/P: Have you had any experience with digital recording, either multitrack or mastering?

RT: Yes, a lot. We started very early on with digital. Lenny and I produced Gordon Lightfoot's *Dream Street Rose*, which was all-digital. We used the 3M Digital Multitrack System; Warner favored the 3M DMS, and Lee Herschberg liked it a lot. I feel that it's probably the best-sounding of all the digital multitracks.

Around the same time that we did *Dream Street Rose*, Ry Cooder made a digital album called *Bop 'Til You Drop*. Ry's record is often said to be the first digital pop album, but we actually started recording before he did—although I don't know if we finished first. We mixed from the 3M 32-track to a 3M 4-track. After that, we used DMS on Randy Newman's *Trouble In Paradise* and, as I recall, some of Rickie Lee Jones' record *Pirates* also was done on the 3M, linked up with an analog 24-track.

On the Winwood record, we recorded analog and mixed to digital; same thing with the McVie record. On the Rufus/Chaka Khan record, the live part was analog mixed to digital, and the studio part was digital throughout.

RE/P: Do you prefer tracking in analog and then mixing to 2-track digital?

RT: Well, it works; I like it; and it's less expensive. After all, it's the *artist's* money. You have to try to be careful about the budget, and digital is very expensive. But, on the Winwood record, we made all our safeties on digital at the Power Station, New York. I hear something different after you transfer to digital; I think that you gain something sonically. We used the Mitsubishi X-80A 2-track to mix Steve's record, and that machine is just fabulous. The bottom is so beautiful, and the top is great.

Steve had never worked in digital before. We had started to mix analog, but the 2-track broke down right when we were in the middle of mixing an important cut on the record. I said, "Let's just get the X-80 and keep going." As soon as I heard the result, I said, "This is staying." When I was out of the room, Steve did an A/B test against the analog 2-track. Afterward, when he came out of the studio, his eyes were just lit up!

RE/P: Do you have a preference of working with one particular brand of console?

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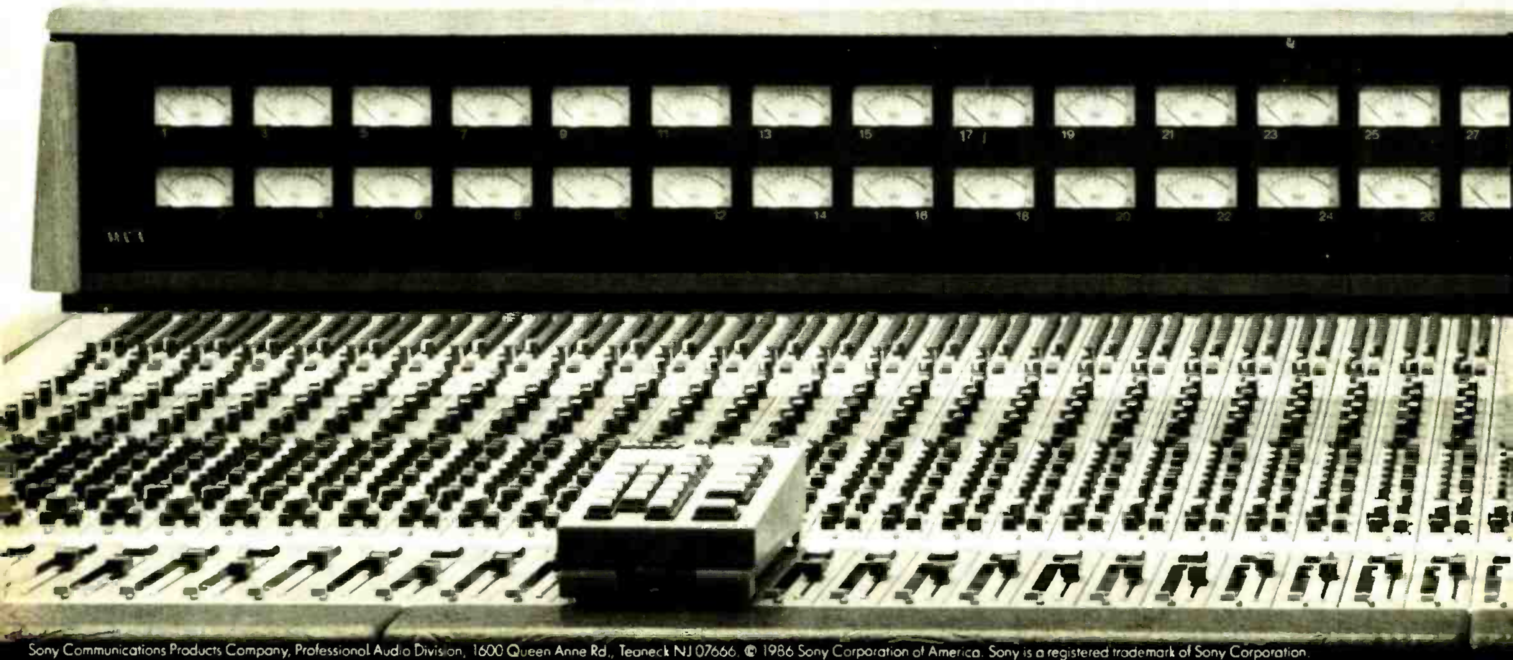

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RT: Yes: Solid State Logic, with its Total Recall automation. I've been working with SSLs a lot lately. If you know how to operate that board intelligently, you can do things *very* quickly. Tom Lord Alge uses the SSL like a player uses an instrument, and Winwood's real good on it, too. I think I prefer a Neve board for tracking, but look: You *can* make a beautiful-sounding record on an SSL. You just have to make use of their strengths.

RE/P: What about your choice of monitor loudspeakers?

RT: I like UREI monitors—the big ones, 813s. For close-fields, I use E-V Sentry 100Ns; I found them through Arif Mardin. You can play them very loud, and it doesn't cause ear fatigue. I was using David Visonic 9000s for a while, too; they're hyped tremendously on the top end, but the bottom and middle are beautiful. That's the thing, though: if you like the speakers, you *have* to get used to what they do—so that you don't pull down the highs, for instance, and end up with nothing on the tape.

RE/P: Have you had much experience with sound sampling?

RT: When we tracked the Winwood album, we started out using a drum machine, and then we put live drums on later. In certain cases, when the drummer wasn't quite "in the pocket" on a track, we'd sample his kick and snare—so we had his sound—then program it on a Linndrum and lay it down. But we'd keep the real tom fills and the hi-hat, so it sounded like a real drummer. And we did take samples from other places, to add to the snare drum sound.

RE/P: Do you prefer the sound of a live drummer to that of a machine?

RT: Yes, I do. Of course, when the drum machine is the right thing, then I prefer to use that, but I love working with musicians. A lot depends on how the machine is programmed. For example, I heard some new Miles Davis tracks that Tommy LiPuma's producing, including this unbelievable piece that Marcus Miller wrote for Miles. It has real complicated changes, and Marcus programmed it using Wendell, the drum machine developed by Roger Nichols. It's astounding: it sounds like a live drummer!

RE/P: You have been working in the Warner A&R department, as well as handling production duties. Have you signed any new artists?

RT: I signed an artist named Jude Cole, with whom I'm going to work. He's a good songwriter, a great guitar player and a great little record-maker. He makes demos with wonderful drum-machine sounds. But I don't usually get that involved in the signings; I'm always in the studio. I'd like to do more A&R.

RE/P: What do you look for in a new artist? What makes you say, "This one needs to be pursued"?

RT: It's just something that strikes you. It's a tough question . . . it's emotional. If you're in an A&R position, you have a certain amount of credibility and knowledge. You can spot good writing: that's one thing that we do. But there's something *beyond* that, because there's a lot of good writing out there. I look for something "magical" that you can't describe; something *extra* that hits me emotionally.

RE/P: Have you ever encountered an artist who you thought should be signed to the label, yet you knew that you were not *the person to produce them?*

RT: Oh, sure. When Van Halen was signed, Ted Templeton, Mo Ostin and I went to see them play here in Los Angeles. There was hardly anybody in the club. Out they came: David Lee Roth and Eddie Van Halen. The band played "You Really Got Me," and it was quite obvious. Mo leaned over and said, "What do you think?" I said, "Sounds like the real thing to me. Sounds like *it*."

Well, that type of music is not what I do, it's not where my interests lie. I can appreciate it, but it's really Teddy's thing. He's great with that stuff—just the best—but I have no business doing that kind of music.

There are a lot of different kinds of artists, and certain artists appeal to certain labels. For example, there's a specific thrust at Arista—artists like Dionne Warwick and Whitney Houston. That might not be the cup of tea at Warner or Columbia.

"I look for something magical that you can't describe; something extra that hits me emotionally."

Russ Titelman

RE/P: Warner seems to look more for the combination writer/artist.

RT: Yes, I think that's basically right, but we have Patti Austin and Chaka Khan: they are two of the best singers there are, and neither is known as a songwriter. Chaka writes, but not prolifically; Patti, I think, just dabbles a little bit.

RE/P: Working with artists like that, who don't write, do you guide the artist in a particular direction?

RT: Yes, but usually with their consent. The A&R staff at Warner are a big help with that: Michael Ostin is a good sounding board, and we have very open communication. Michael was executive producer on the Patti Austin record, so he was involved in overseeing the selection of material.

RE/P: Do you think that your songwriting background has been important for you as a producer?

RT: It's helped me tremendously, but I don't think that it's a prerequisite. There are people in this business who don't have a musical background, or any particular musical talent, yet they make great producers. Take Lenny Waronker; he's had no musical training at all, but he's been around music all his life. His father was a fiddle player with the 20th Century Fox orchestra, so he's known Randy Newman since he was a little boy. Lenny used to go to all the scoring sessions, and I think that's reflected in the kind of records he's made; they're very visual. And our work together has been very visual, like orchestrated movie music.

I listen to a lot of classical music and, lately, to a lot of opera. Once you do that for a while, you get to learn what a great

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Circle (26) on Rapid Facts Card

Tom Lord Alge, session engineer (left), Russ Titelman and Steve Winwood.

"I suppose that I have a way of working, but the records probably should speak to that question. If there is a 'Russ Titelman' style, it'll be there on the records."

performance is. I love Puccini; he was the best songwriter that ever lived. And Pavarotti . . . [sighs] Opera is the most emotional material you'll ever hear, and I find that it's a very good genre to study for production. It is *the* high art: theatre, acting, music, storytelling. I think that opera has influenced by work on an intuitive level and teaches you how to achieve something emotionally and musically.

RE/P: I'm struck by your remarks about melody. Many producers talk less about melody than about lyrics, "feel," and so on. But it's actually not so much of a surprise coming from you, given the caliber of the artists you've worked with.

RT: Well, yeah: Paul Simon, Randy Newman and James Taylor are great melody writers. Some of those Newman songs are real plaintive: Texas girl at the funeral of her father; Germany before the war . . . Simple, strong melodic statements, and very moving, very emotional.

RE/P: One last question: Do you think that you have a particular production style?

RT: I think you could answer that better than me! I suppose that I have a way of working, but the records probably should speak to that question. If there is a "Russ Titelman" style, it'll be there on the records. R·E/P

Photos by Karen Petersen.

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CONSEQUENCES: When pushed hard, most tweeters simply fail. Transient detail blurs, and the material itself deforms and breaks down. Other materials can't take the stress, and crack under pressure.

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Circle (28) on Rapid Facts Card

Future Directions of Studio Monitor Designs

By John Eargle

The majority of currently available studio monitors use the traditional HF horn and compression driver combination. What other alternatives exist, and how might they influence the future of monitor designs?

Introduction:

For most of its 60 years, the majority of recording and production studios have used monitor loudspeakers based on the Western Electric traditions of high-frequency horn/compression-driver technology. The main reason for this tradition has been reliability. Such systems are relatively immune to abuse, accidental or otherwise, and can handle fairly high playback levels. Various manufacturers have dominated the studio monitor market, while custom builders have developed their own monitor designs—following the same Western Electric tradition.

Characteristics of traditional designs

Presently, the best of these monitors may be characterized by the following criteria:

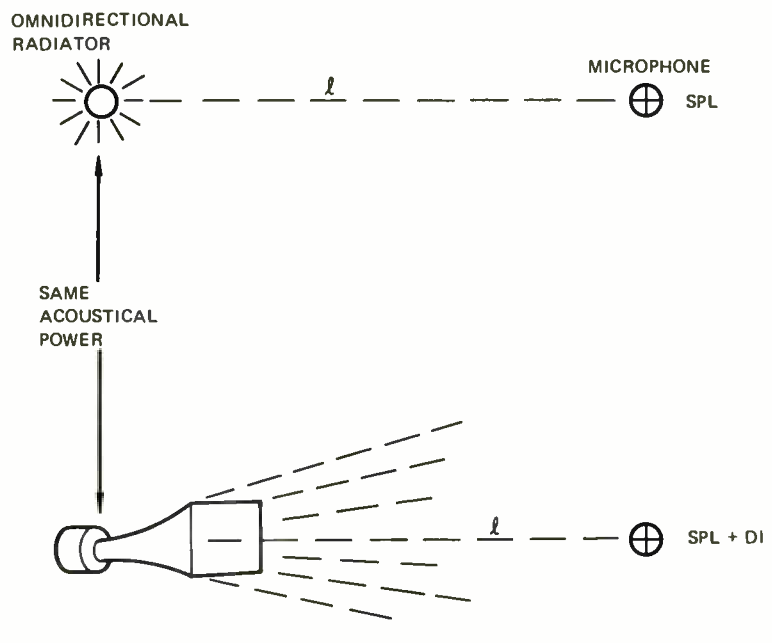
- Relatively high sensitivity (discussed further in the accompanying sidebar) 93dB to 97dB-SPL, 1W at 1m.
- Relatively high-power input capability and continuous input of 200W to 300W.
- High acoustical output capability. A 97dB sensitivity (1W at 1m) with a power input of 200W will produce a level of 120dB at 1m. In an average-size control room, a pair of such monitors may produce mid-band levels of 113dB to 115dB at the mixing position.
- Relative insensitivity to HF overload and burnout. The normally high efficiency of horn/driver systems requires considerable electrical padding to match its input with that of the low-frequency section, thus reducing the electrical signal actually seen by the compression driver.
- Relative acoustical simplicity. Most current systems are of two-way design, with a single crossover point in the 500Hz to 1kHz region.
- Generally low distortion at normal operating levels.

John Eargle is the president of JME Consulting Corporation, and has consulted for a variety of equipment manufacturers. He also manages to find time for regular jazz and classical sessions.

By the late Sixties, the audiophile movement was coming on strong, with an emphasis on simplified electronic and signal paths, direct-to-stereo recording, frequent use of bi-amplification and an emphasis on high-end consumer loudspeakers. As the Seventies progressed, the audiophile movement took on even stronger identity, with its own publications, its own dedicated suppliers of software and its almost universal suspicion of mainline recording practice, including the use of traditional monitors.

What such users were seeking in studio monitors could be found in a number of high-end, multi-element consumer models based on highly damped components. What they prized in their loudspeakers was the extreme smoothness that small diameter HF soft domes are often capable of producing, as well as generally low distortion throughout the range at low to moderate operating levels. (High playback levels were rarely required.) In fact, it can be shown that HF domes are capable of producing less distortion at moderate operating levels than typical compression driver/horn systems when both systems are producing the same acoustical output.

Figure 1. *Directivity Index, DI, is a measure of a loudspeaker's on axis "gain" compared to the same acoustical power radiated omnidirectionally.*



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Loudspeaker Efficiency, Sensitivity and Directivity Index

The efficiency of a loudspeaker is a measure of its ability to convert electrical power into acoustical power. It is usually expressed as a percentage, but you can just as easily speak in terms of a power conversion level, expressed in decibels, as shown in Table 1.

The relationship shown in Table 1 is given by the following equation relating conversion level, C, to efficiency, E:

$$C = 10 \log (E/100) \quad (1)$$

Sensitivity, S, is a measure of the loudspeaker's acoustical sound-pressure output level at a given distance, direction and power input. The universal standard specifies the measurement of SPL at a distance of 1m (or referred to 1m) along the loudspeaker's major axis with a nominal power input of 1W.

Directivity index (DI) may be thought of as the on-axis "gain" in decibels for a particular loudspeaker, compared to the same acoustical power radiated omnidirectionally. Because the concept may not be an obvious one, the data shown in Figure 1 (page 56) are offered for clarification.

A simple equation ties these three concepts together:

$$S = 109 + DI + C \quad (2)$$

As an example, suppose a given low-frequency transducer has a published sensitivity of 93dB, 1W at 1m, when it is soffit-mounted or mounted in a large flat baffle. Its DI is 3dB because of the "half-space" mounting, and its conversion level, C, can be determined by rearranging and solving Equation 2:

$$\begin{aligned} C &= S - 109 - DI \\ &= 93 - 109 - 3 = -19\text{dB} \end{aligned}$$

Table 1 does not include values down to −19, but you can solve Equation 1 for a value of 1.25% efficiency.

A 93dB woofer is typical for many monitors using a single, 15-inch LF driver, and the resulting efficiency probably comes as a surprise to many readers. A monitor with a dual low-frequency section with a 1W, 1m sensitivity of 99dB would have a conversion level of −13dB, indicating an efficiency of 5%.

At mid-frequencies, the efficiency is even less. For example, the conversion level of a monitor with a DI of 10dB and a sensitivity of 93dB at 1kHz can be calculated as follows:

$$C = 93 - 109 - 10 = -26\text{dB}$$

This value indicates efficiency of 0.25% for that frequency range, a result that is typical for monitors and fairly high in relation to most consumer systems.

While most loudspeaker systems feature reasonably low efficiencies, certain components, at least over part of their frequency range, can have high efficiencies. The maximum efficiency exhibited by a compression driver is 50%[1]; most commercial models do not make it beyond 30%, which is about 2dB lower.

In measuring compression drivers, you can ignore the DI effects of the mating horn by mounting the driver on what is called a plane wave tube. A simple calculation relates the pressure in the tube and its cross-sectional area to the actual acoustical power in the tube. The ratio between the acoustical power and input electrical power will provide us the efficiency of the driver.

A typical PWT curve is shown in Figure 2 (page 62). In this case, the cross-sectional area is 5cm². One milliwatt per square centimeter 1mW/cm² will produce a sound pressure level of 130dB. Hence, with an electrical input of 0.005mW a pressure in the tube of 130dB would indicate an efficiency of 100%. Obviously such a driver cannot be made, and the data shown in Figure 2 are what you commonly encounter with good drivers. Note that the vertical scale is marked in sound pressure level, efficiency and conversion level.

The compression driver's power output is flat only over a small portion of its frequency range. Above about 3kHz, it begins a steady HF rolloff of 6dB/octave, so that in the range between 8kHz and 16kHz its efficiency is between 5% and 1%.

If such a driver is to be used in a constant power response system, then selective equalization must be applied to the driver, internally in the dividing network, to cut down the excessive mid-range efficiency. If the driver is to be used on a horn with a rising DI above 3kHz, then only padding will be necessary to match the driver to the monitor's woofer section. The horn's DI will effectively equalize the driver's output—but only on axis. Off-axis, the rolloff will still be apparent.

Table 1: Power conversion level in decibels for a given loudspeaker efficiency.

Efficiency (%)	Conversion Level (dB)
100	0
50	−3
40	−4
31.5	−5
25	−6
20	−7
16	−8
12.5	−9
10	−10

Reference

1. JBL Publication, "Characteristics of High-frequency Compression Drivers;" Technical Note Volume 1, No. 8.

By the early Eighties, several high-end consumer models had become popular among classical recording engineers and producers. Today, it is safe to say that almost all classical recording relies on stock audiophile consumer loudspeakers for routine monitoring.

High-level domes in the recording studio?

Until recent years, nobody has given much thought to the cone/dome approach for use in conventional recording studios, the main reason being their extreme shortfall in HF output capability relative to compression-driver technology.

Here is an example of the shortfall: A typical high-quality, 1-inch dome tweeter designed for response above 4kHz may have a sensitivity (1W at 1m) in the range of 89dB to 91dB. Its electrical power input rating (P_F) may be about 25W to 30W. With this input, the acoustical output at 1m will be in the 103dB to 106dB range. (The calculation made here is simply summing the sensitivity rating and 10 logP_F.)

By comparison, an HF ring radiator—a compression device with integral horn designed for HF applications—may have a sensitivity (1W at 1m) of 105dB and P_F of 30W. Fully powered, the acoustical output at 1m will be 120dB. Essentially the same performance could be expected from an HF horn/driver combination operating in the 8Hz to 16Hz octave.

The 14dB difference is discouraging to system designers and has effectively ruled out the use of most dome tweeters for any high-level monitor application. In recent years, however, new high-power domes have been developed. These devices usually come in two sizes: 3-inch diameter model for mid-range use and 1-inch for the top of the spectrum.

Typical sensitivity and power handling characteristics for such devices are encouraging. For example, a 3-inch midrange dome may have a 1W, 1m sensitivity in the 95dB range and P_F of 80W to 100W. In some cases, ferrofluid cooling of the voice coil is integral to the design. Normally, such a device would cover the range from about 400Hz to 4kHz, and the maximum output at 1m would be about 115dB.

Obviously, we are getting up to where we would like to be, but are well behind a typical compression driver midrange system. Nevertheless, the output capabilities of such a mid-range dome will satisfy many producers and engineers.

An example of a newer 1-inch dome designed for high-level applications might have a 1W, 1m sensitivity of 94dB and P_F of 40W, producing an acoustical output at 1m of 110dB. Such a transducer can be integrated into a wideband system and, barring any extreme demand on the system's power bandwidth, it will match the mid-range and low-frequency elements on most material.

The role of dedicated electronics

A great many traditional monitors are still offered as passive systems, notwithstanding the fact that all monitors will sound better if bi- or tri-amped. By comparison, most of the newer breed of high-level cone/dome systems come with their own integral 3- or even 4-way dividing network/amplifier combinations. In laying out such systems, it is customary to match the output to the load; thus, there is certainly no need to provide a 400W amplifier for a 1-inch dome tweeter.

Other "smart" electronic options include program-directed shifting of high- and low-pass functions, which protect transducers by limiting voice-coil excursion. Some systems may include a degree of program limiting, further holding down peak-power inputs. Such functions may or may not be audible as such, and debate rages over their application in studio monitoring. There is no doubt, however, that slight audibility some of the time is preferable to a blown tweeter.

It may be significant that the first great success of cone/dome monitors has been in England. One company, in particular, uses dual woofers mounted in ported enclosures, just as do traditional monitors, with mid- and high-frequency domes on top.

In addition, a Scandinavian manufacturer produces a 4-way, tri-amplified system that is said to be capable of sustained output at 1m of 120dB. In one model, the 40W continuous power is available for the 21mm dome tweeter. Yet another company based in Canada makes a 4-way system based on a pair of 15-inch woofers, a 10-inch lower midrange cone, plus 3- and 1-inch domes on top. This 4-way system is driven by four power amplifiers.

Beware inadequate specifications

Because they are an outgrowth of mainline loudspeaker technology, conventional compression-driver monitors are normally completely specified with regard to sensitivity, power input requirements and directional data.

By comparison, many of the newer cone/dome systems carry questionable specifications, such as a statement of the continuous output level a pair of systems can deliver into a typical control room. What is a typical control room? And how far away from the loudspeakers was the measurement taken? In all likelihood, if these new monitors continue to be successful, a rigorous set of specifications will be developed for them, but that point has yet to be reached. Let the buyers beware—and let them insist on adequate performance data.

New directions in conventional monitors

Monitors based on compression-driver technology continue to evolve, primarily with the adaptation of exotic materials for diaphragms.

The concept of flat power response in a monitor's HF section was a significant development in the early Eighties. With broader dispersion horns, compression-driver monitors took on a new sound that was flatter and easier to equalize in the typical control room.

A well-known brand of phase-coherent monitors have been updated to include modern diaphragm materials and more

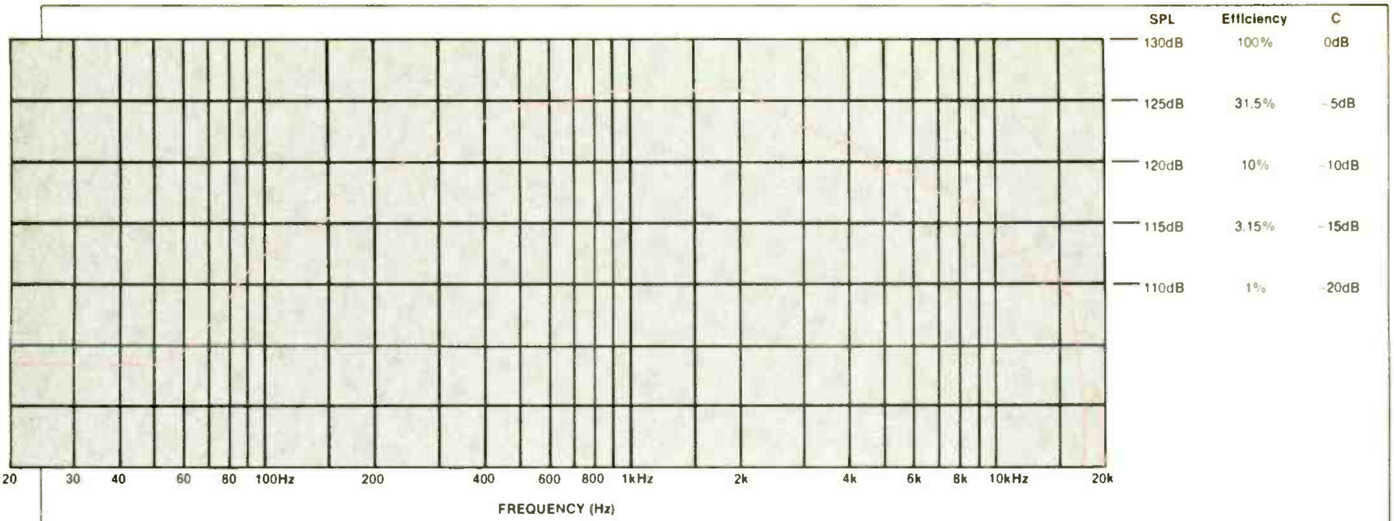

Figure 2. A plane wave tube curve for a typical compression driver with tube cross-sectional area of 5cm². The maximum efficiency between 1kHz and 2kHz is 40%.

linear woofers. In many ways, the company's line of products anticipated the demand for accurate time-domain response in loudspeakers, which has become standard today.

Components from a Japanese manufacturer have been used in a number of custom monitor designs. One of the company's most recent designs has a directivity index that closely matches the inverse of the HF driver's power response. What this means is that the combination horn and driver will produce flat on-axis response without the need for specific power-response equalization in the dividing network.

It goes without saying that all monitors, regardless of design philosophy, benefit from improvements in adhesive and magnet technology.

Although the requirements of high-level monitoring in the studio continue to demand the lion's share of the designer's time, it is that great proliferation of small bookshelf and close-field monitors that engineers use most of the time. In these areas, we have already seen the boundary between professional and consumer orientation removed completely. It is gratifying to note that the needs of the professional are now beginning to be successfully translated into benefits for the consumer. For too many recent years, the flow of technology has been the other way.

R·E/P

Surround Sound for Video

An ambisonic visual postcard.

By David Moore

After several years of intense behind-the-scenes activity, ambisonic surround-sound recording techniques have passed from the experimental stages into real-world applications and popularization. Despite the regrettable confusion with earlier attempts at 4-channel sound, engineers and producers are beginning to recognize the operational and psycho-acoustical advantages inherent in working with B-Format and UHJ-encoded ambisonic material.

Recent advances in consumer technology, including Compact Disc and stereo television, have come a long way in eliminating the distribution problems presented by earlier multichannel attempts at *surround sound.* Many users consider that ambisonically encoded material may find its way into mainstream records and television far faster than one might think possible.

Stereo television has already put a new emphasis on the quality of sound that accompanies the visuals, and UHJ-encoded ambisonics could very well open up a new dimension for home entertainment.

Recently, a new genre of home video was released by a Washington, DC-based production company, Nicholas Communications. The 30-minute video was designed to take advantage of the realism offered by ambisonic production techniques. Producer Stephen J. Nicholas conceived the idea of a video and audio postcard—not just a simple travel log, heavy with narration and stock-look film footage, but rather a carefully assembled series of images and music working to-

Scenes from the visual postcard showing some of the more memorable sights of Washington, D.C.

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together to evoke memories of experiences.

The goal was to project a sense of immediacy, a sense of being there in the imagery. According to Nicholas, the project seemed a natural marriage between visual images and realism of ambisonic techniques.

"Since my background is in the visual arts as well as in audio production," the producer says, "it seemed the natural thing to do; synchronizing musical imagery—colors, textures, hues and moods—and visual images to produce the sights and sounds of some of the great cities on our planet. We're presenting this form of information/entertainment in a long-form music video that takes the viewer on a thrilling journey."

His thinking was as follows: A venue would be selected for broad appeal, perhaps one that was already known as a tourist attraction, and one full of compelling, easily identifiable images. The sights and sounds of this location would then be recorded in the appropriate formats, always paying careful attention to the producer's requirements for high quality and containing as much ambisonically encoded information as possible.

Finally, these visual and aural images would be mixed over an original music score, which would be paced and timed for the shifting images.

Audio-video synergy

The first location chosen for the postcard treatment was Washington, DC; the eventual production spanned just about every form of visual and sound recording, as well as having to bring together elements from a variety of diverse media for final editing, mixing, duplication and sale.

Having a strong recording and musical background, Nicholas considered that the best way to break out with his new product would be to have a home video release that was strongly differentiated by an audibly superior soundtrack, matched with the highest quality visuals.

These requirements led him to digitally master the project on Mitsubishi X-800 multitrack and X-80 stereo machines at south Florida's Criteria Recording Studios and, more importantly, the use of ambisonic technology whenever and wherever possible.

Mac Emmerman, Criteria owner and past president of SPARS, was brought in early as the project's sound designer. His enthusiasm for the project, allied with the producer's quest for quality, quickly involved other high-tech aspects

A decision was made to offer the product only on Beta hi-fi and VHS hi-fi videocassettes, not only with an all-digitally mastered music soundtrack, but also with 2-channel ambisonic UHJ-Format surround sound. To this end, a Calrec Soundfield microphone was used to record as much of the audio material as possible, with special attention being given to its use in capturing ambient sounds on location.

Although the ambisonic system is capable of capturing a 3-dimensional *picture of sound* for later decoding and playback over a domestic surround-sound system, Nicholas thought that the technique's outstanding mono compatibility was also of value. Not only would a 2-channel UHJ soundtrack enable the critical

Ambisonic Surround-Sound Technology

Ambisonics is a total systems technology, aimed at capturing, recording and replaying surround-sound information to meet a wide range of professional and consumer needs.

In the studio, ambisonic information is recorded onto tape in the form of four B-Format signals, known as W, X, Y and Z. The W signal represents the omnidirectional soundfield arriving at the microphone position, while X, Y and Z signals are equivalent to bi-directional (figure-of-eight) microphones, pointing respectively forward to back, left to right, and up and down.

For horizontal surround soundfields, the Z signal is set to zero (or omitted), and only the W, X and Y signals used. (It should be noted that these four B-Format signals bear no relationship to the signals routed to a 4-speaker monitoring array and only serve as recording channels.)

The Calrec Soundfield microphone houses a tetrahedral array of four cardioid capsules. A companion control unit allows sounds to be panned and steered in horizontal and vertical planes either in real-time or in post-prodution from off-tape B-Format signals. Azimuth and elevation controls rotate and tilt the microphone's apparent spatial orientation, while dominance and zoom controls focus and emphasize sound on front and back planes.

To process ambisonic sound, three pieces of equipment are required:

- *A transcoder unit, which enables both encode (B-Format to 2-channel UHJ) and transcode (conventional 4-channel mixes to 2-channel UHJ) capabilities. With this unit, control of front and back stereo width is provided.*
- *A multitrack pan/rotate unit, which features sine/consine panpots and switchable controls that allow a 360° soundfield rotation. The panpots can be switched before or after the rotate control, and the unit can be cascaded via external B-Format inputs for recording or post-production control, to produce a suitable UHJ (2-, 3-, or 4-channel) master.*
- *A converter unit, allowing conventional console panpots to localize a soundifeld.*

In its simplest form, UHJ uses the same two recording or transmission channels as conventional stereo, using both amplitude and phase information to convey a horizontal surround-soundfield to the listener. Via a suitable UHJ decoding system, an ambisonics playback system can process information from multiple devices, including AM/FM transmissions, albums, cassettes and Compact Discs, allowing the listener to move around the listening area and face any direction without losing the sound image.

To fully use ambisonic technology, six or more monitor loudspeakers are necessary for full-sphere or periphonic reproduction. Otherwise, horizontal surround-sound monitoring (without Z-channel or height information) require only a decoder and four speakers, arranged around the listener.

Consumer systems are normally based on 2-channel ambisonic UHJ Format. This 2-channel signal, when properly decoded, provides four or more loudspeaker channels for surround-sound reproduction. And, in contrast to B-Format signals, the 2-channel UHJ signals can be monitored in either mono or stereo via conventional replay system.

Unlike conventional 4-channel quadrafonic sound—which uses variants of a 4-2-4 matrix—ambisonic technology allows the use of virtually any number of loudspeakers to recreate localized phantom images, even between side loudspeakers.

Editor's note: The information in this sidebar was extracted from "Ambisonics Surround-Sound Technology for Recording and Broadcast," published in the December 1983 issue of RE/P.

listener to recover location and music ambience contained within the original mix via a surround-sound decoder, the less well-equipped audio enthusiast could replay a perfectly acceptable monophonic sound. (Which, after all, represents the bulk of the current VCR market.)

With the selection of recording methods and venue decided, Emmerman and Nicholas linked up with Miami-based arranger/composer Mike Lewis, who was commissioned to write the original score. The producer's intention was to first record the score, shoot the visuals to fit, add location sound effects and ambience, and then edit, conform and synchronize all of the elements in a final mix.

Music scoring sessions

More than 30 of the Southeast's best session players were used at Criteria Studio E during the music tracking date. With only a rough idea of how the images were to go together, Lewis provided far more music than was required to accompany the finished product. The concept was to record a group of separate pieces, each related to a certain part of the nation's capital, and timed to the footage that would be shot on-location.

A total of 12 individual pieces appear in the final cut: "City Scapes," the opener; "Cherry Blossom;" "Tribute;" "Celebration;" "Zoo;" "Peace;" "Hirshorn Sculpture Garden" (shot at the Smithsonian Institution Garden); "Capitol;" "WDC Breakdance Capital USA;" "Pulse;" "Vietnam Memorial;" and "Solitude."

Some of these titles are self-explanatory; others make sense only to those who have visited these sites around the Capitol. (But, then again, you don't generally send a postcard from a place you haven't visited.) Together, all of these subjects form one producer's idea of what would be remembered about a visit to this historic city.

Two of the pieces presented technical problems for the producers of this ambitious project: "Vietnam Memorial" and "Breakdancing." Both segments required a great deal of preplanning—even then, as is often the case, there were massive saves made in the mix/final post process.

Location shoots

Both sequences were shot on location, using a Sony Betacam system. The memorial piece required an ambisonic sound-effects recording of helicopters in-flight. By special arrangement of the Air Force, Nicholas and Tom Gandy of Audio+Design/Calrec were granted access to Andrews Air Force Base. Their plan was to record the live sound of the president's Mission, a 6-chopper flight of helicopters designed to provide presidential transport in case of an emergency.

These sounds, properly mixed, equalized and brought in over the music, are intended to provide a haunting introduction. Using a Soundfield microphone and UHJ encoder. Nicholas and Gandy recorded the 2-channel sound onto a portable VHS hi-fi videocassette recorder, so that the tracks could be synchronized using the video sync reference.

The pair recorded about a half dozen passes of the helicopters—just enough to *layer up* in the studio and provide the effects required by the production.

The crew then headed downtown to capture the traffic noise that later accompanies "City Scapes," the piece used to introduce Nicholas' audio/video postcard. The use of a portable VHS hi-fi VCR to record the UHJ-encoded ambisonic ambience sounds was typical of the kind of decision a producer must make everyday. With a limited budget and the need to transport a lot of equipment to location, sometimes expediency must win out over perfectionism.

Because all of the material recorded on location needed some sort of reference for synchronization during post-production, using a VCR or other sync recorder for audio was mandatory. FM-encoded VHS hi-fi technology was in the right place at the right time, for the right price. (Later, we'll see that even a control-track reference may not help in post-production, unless some careful planning is done up-front.)

After making UHJ 2-track recordings of traffic sounds, the crew moved on to videotape the "WDC Breakdance Capitol USA" segment. Working with previously casted local talent, the high-energy piece captures the feeling of this urban art form, with a costumed Uncle Sam breaking away in a routine called "Style and Profile."

Sequences performed by the "Mighty Poppalots," a local dance group, were shot to a rough mix of the material recorded at Criteria, which in the case of the breakdance music contained no live sounds. Everything was built up of synthesizer and sequencer parts taken direct. In much the same way that rough tracks are used while producing a music video, the audio material was necessary to establish the duration of the piece, and to cue the performer's moves.

The video was shot without the customary practice of *dumping* the scratch audio onto the VTR's audio tracks. Instead, a time code generator resident in

Seen here during the mixdown for the "Washington DC" visual postcard project: (clockwise from top left) Stephen J. Nicholas (producer/director), Tom Gandy (Audio+Design/Calrec field engineer), Mike Lewis (composer/arranger) and Mack Emmerman (Criteria Studios president/sound designer).

Ambisonic Radio Drama

Late last year, KWMU-FM, the public radio station of the University of Missouri-St. Louis, broadcast Stanley Elkin's radio drama, The Coffee Room, *which was produced in the 2-channel UHJ ambisonic format. The play was the first ambisonic radio drama produced in the United States, and featured Elkin with 13 other St. Louis actors. It was directed by John Grassili and produced by KWMU's Lorin Cuoco and Sean Collins.*

In order to take full advantage of the opportunities offered by ambisonics, the play was recorded in a room thought to have the sound of a typical university coffee room, with the actors working 360° around a Calrec Soundfield microphone. The voice tracks produced from these sessions were edited and put onto an Otari 8-track along with sound effects recorded ambisonically on the Washington University campus.

An Audio+Design pan/rotate unit was employed to place effects and additional speech in the ambisonic soundfield. The final Dolby-A encoded UHJ ambisonic broadcast mix was done through a Harrison PRO-7 console by Collins onto a Studer A-810 2-track. Monitoring during mixdown was accomplished using four KEF 101s, two stereo amplifiers and a Minim AD10 ambisonic decoder.

KWMU premiered U.S. ambisonic broadcasting in May 1984, and has continually sought to integrate ambisonics into its program offerings.

the Betacam unit was used to provide a stripe for later use in editing.

Video post-production

The basic elements were now complete, but much more shooting and location work were needed to fill out the rest of the producer's concept. Following these two segments further, production moved to Editel/Chicago for video post.

By this time some of the other segments had been shot on other formats, including film. Other pieces took the form of film footage picked up from sources such as award-winning cinematographer Louis Schwartzberg, who provided the time-lapse and aerial sequences used so effectively in this postcard. Editel provided the equipment necessary to transfer all of this material to a common 1-inch C-Format for editing.

Off-line workprints were created, and the production team went to work editing the material to a rough mix of the audio. Editel's Montage editing system provided the ability to preview a large quantity of edits, and quickly assemble an off-line edit decision list.

Assisted by Larry Sexton, Nicholas finished the EDL and then moved into one of Editel's 1-inch, on-line suites for the final cut and addition of optical and special effects. Visual pacing of the production was assisted by extensive use of the facility's Ampex ADO system and various other digital visual effects, manipulated by editor Jerry Doskoczynsky under Nicholas' direction.

At this point, there was only rough conformance and synchronization of audio to video; the producer thought that the audio tracks could be massaged into place during the final mixdown process. A ¾-inch U-Matic workprint was dubbed from the 1-inch Edit Master, and the project moved back to Criteria for the final mix and synchronization.

Music and effects assembly

Parts of the soundtrack needed to be *flown-in* to the 32-track digital master on a Mitsubishi X-800 supplied by Digital Associates, Nashville, TN. Although control tracks have been provided on all of the audio material, it was discovered that the digital multitrack could not be slaved to the VHS-format videocasstte recorder.

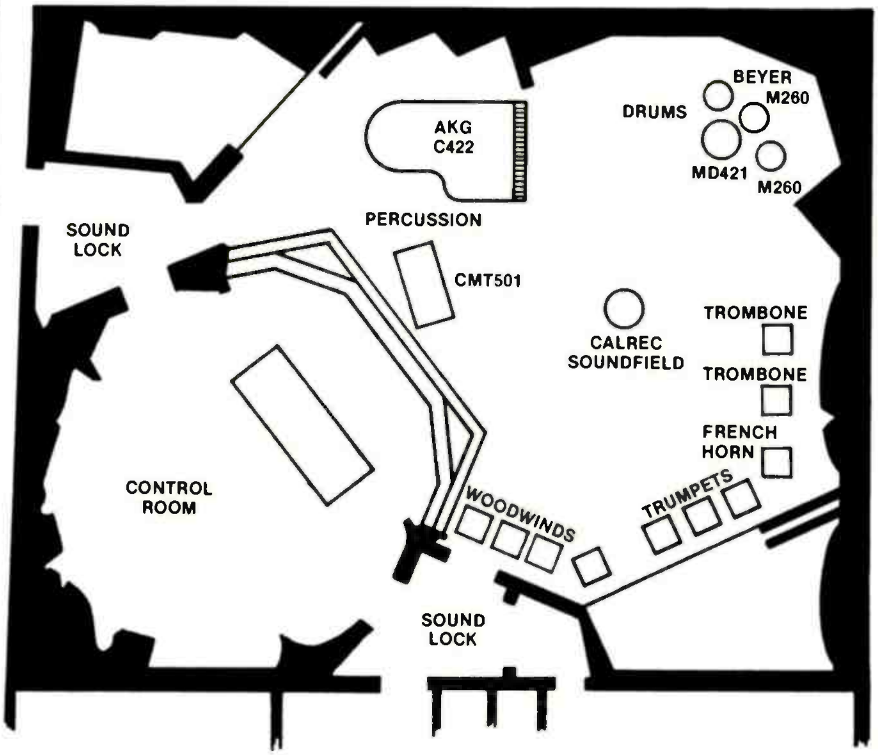
As many readers may already be aware, a digital recorder has its own control track, and generally prefers to be *the boss* in synchronization situations. (Try to servo-control the capstan of most digital recorders beyond a tight tolerance window and, with few exceptions, the audio mutes.)

During the original music-tracking dates, both ambisonic and spot microphones were used. Although this meant that the multitrack was pretty loaded up, by doing some bounce downs (no problem with digital material) a few tracks could be freed. Spare tracks were necessary because to provide creative control over the final ambisonic surround-sound mix, the UHJ-format material recorded in the field had to be decoded and re-recorded as four B-Format signals representing the horizontal W, X and Y plus vertical Z information.

Suitable monitoring equipment was set up in the control room, and the process of flying in the audio began. Fortunately, the combination of hits and rhythms that the producer wanted to develop did not contain any lip sync (there's none in the entire postcard), and so most timing could be worked out manually. Of course, the finished visuals had to look right, which meant many long hours of tries and retries before everyone was satisfied.

By now the project was moving forward, with a time code track holding the digital multitrack mix in sync with the video workprint by means of an Audio Kinetics Q.Lock 3.10 synchronizer. Following standard practice, the next step would have been to a final automated mix of the audio material, and then transfer this mix to a digital 2-track.

If the multitrack and 2-track time-code tracks were conformed, the audio would sync up to the picture perfectly.



Room and instrument layout during the recording of the "Vietnam Memorial" segment at East-wing (Studio E) Criteria Studios, Miami.

The orchestral music recording sessions at Criteria before the video shoot. The location visuals were shot to the musical tempos and edited during subsequent post-production.

Photo by Alan Jerram

Video post-production editing sessions at Editel/Chicago, with Nicholas (right) spotting time code locations during preparation of the edit-decision list.

and a complete mix would be ready to go out for duplication. But a Mitsubishi X-80 (now superceded by the X-86) only knows one way to record—the time code, control track and audio all at once—or what is often referred to as the *crash record* mode.

All of which meant that it would be impossible to produce a conformed Audio Edit Master, a format that could have represented a real time-saver during the process of readying the material for videocassette duplication. Again, one of Criteria's engineers, Richard Aker, had to work with the producer to fly-in each segment of the final mix onto the 2-track Edit Master, with the remixed ambisonic surround-sound information again encoded to 2-channel UHJ.

Once the audio material was completed, the final UHJ mix was transferred to dbx model 700 CPDM digital format before being sent out for videocassette duplication. CBS/Fox was chosen as production house for the first run of dupes, in both Beta hi-fi and VHS hi-fi formats, and specified CPDM-encoded material to ensure compatibility with its system. After a number of small glitches—like forgetting to run video sync to both the X-80 reel-to-reel digital recorder and the dbx/VTR recording system—CBS/Fox produced the dupes and readied them for market.

Since then, the project has been circulated among many in the audio community, and initial sales of the *Washington, DC* videocassette are said to be brisk. Demonstrations have proven that ambisonic information captured with the Calrec Soundfield microphone is sufficiently robust to withstand all of the different recording methods used during production, and still is capable of decoding and playback as convincing surround-sound information. Although not many people have access to a consumer UHJ decoder, Nicholas is content in the knowledge that the information is there for those who are interested in ambisonics and the equipment to match.

Final mixing of the project was done by Criteria's Mac Emmerman and Steve Johnson. According to Emmerman, although most of those involved in the project knew that the audio would largely be heard in mono, careful use of the ambisonic surround-sound information yielded the "best mono sum I've heard." When this writer heard the audio tracks for the first time, it was on a basic VHS videodeck. My initial impression was that there did indeed seem to be something very special about the sound quality, even when replayed over a 4-inch speaker.

Nicholas is already involved in preproduction of his next visual-postcard project. He has become a firm believer in the value of ambisonic recording to video and TV production. Interestingly enough, Nicholas would definitely use ambisonics technology in his next project, but does have reservations about video production.

"We all learned a lot during this first project," he says. "And I have developed a great respect for sprocket holes. If we have the budget, my next project will be done entirely on *film*, so we can avoid all of the interformat problems that can arise when a critical sound mix is married to pictures that are gathered from a lot of different sources."

Nicholas will continue his role as a proponent of ambisonic surround-sound, particularly for audio/video presentations. The technique is now beginning to be noticed, and recent business developments may make the producer a pivotal player in the technique gaining more widespread acceptance.

Ambisonics may well be in that unique position as an idea whose time has come, and bears some careful watching during the next several years. R·E/P

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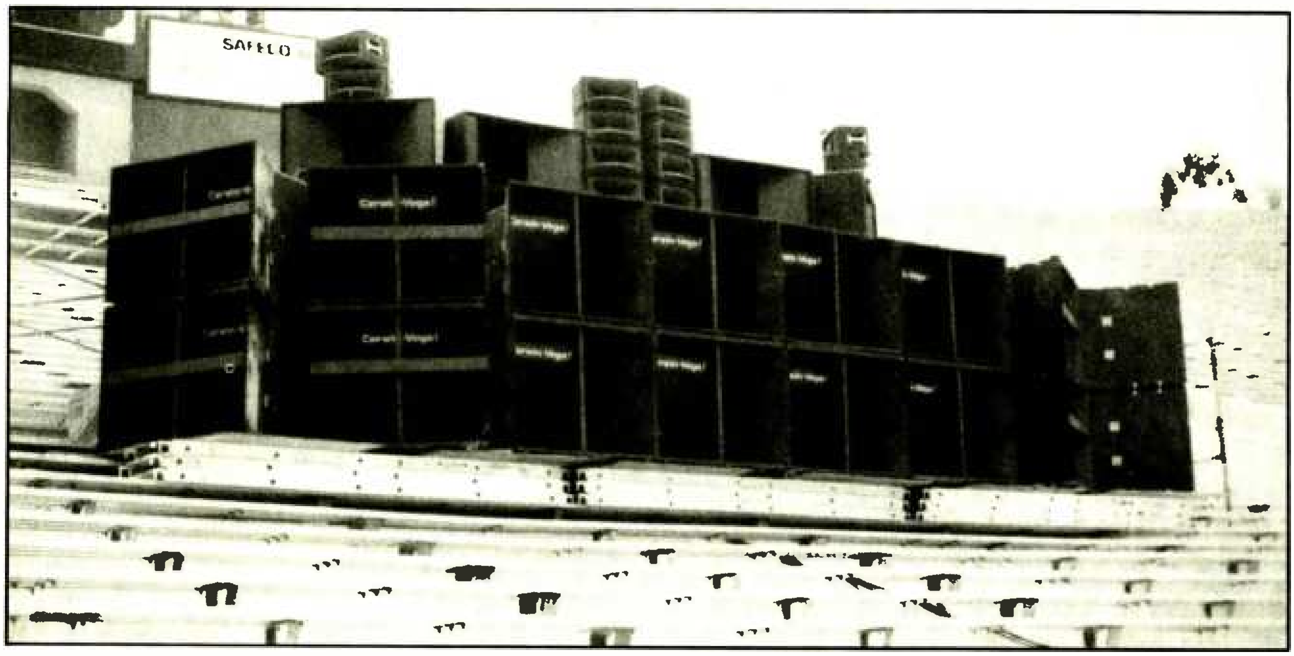

Photography by Ralph Jones.

Sound Design for Outdoor Arenas

By Ralph Jones

Prototype sound reinforcement system at the Pasadena Rose Bowl incorporates a unqiue quad-driver midrange horn.

Once every four years, the Church of Jesus Christ of Latter Day Saints holds its Southern California Area Dance Festival, an event that draws crowds of spectators who watch Mormon dance groups perform folk dances in traditional costume.

Begun in the Fifties, the Festival has been held at the Pasadena Rose Bowl since 1973. Last year's event, held in late July, attracted a capacity crowd of 110,000 people, with 13,700 dancers participating.

Glen Glancy, president of Continental Sound, a Canyon Country-based sound contracting firm, has been responsible for sound reinforcement ever since the festival was moved to the Rose Bowl. His task is not a simple one. In addition to voice amplification for the various announcers, the sound system is expected to provide high-quality music reproduction throughout the entire stadium—including the playing field where the dancers perform. To satisfy the requirement last year, Glancy, in his role as head of audio for the dance festival regional committee, teamed with Philadelphia P.A., a Riverside, CA-based touring sound company.

Glancy had enlisted PPA's services for the previous dance festival, for which company president Henry Austin had provided a large system modeled along the lines of the Clair Brothers S-4 cabinets. This time, however, Austin had different ideas. He had been approached by Cerwin-Vega, which was then developing a new, all-horn line of loudspeaker products designed specifically for use by the touring reinforcement market. Impressed by the ideas that Cerwin-Vega presented, Austin was eager to find an opportunity to work with the company.

Austin broached the festival project with Cerwin-Vega representatives, who determined that it would provide an ideal opportunity for field testing the basic principles and components of the new product line. The company agreed to provide, through PPA, a complete

Ralph Jones is a musician/producer and a regular RE/P contributor.

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reinforcement system for the event, with Hollywood Sound supplying the power service and playback equipment.

Cerwin-Vega design engineer Marshall Buck assumed responsibility for specifying the loudspeaker components and their placement. The result was a system that Rose Bowl executives referred to as "the best quality sound ever brought to the Bowl."

System description

"This system is our prototype of products made specifically for the sound reinforcement market," says Buck. For the event, there was a single, tight array of horn components stacked on risers at one end of the Bowl, placed approximately one-third of the way up in the seating area. Dwarfed by the scale of the venue, the array comprised 16 low-frequency horns stacked two high and eight wide; eight midrange horns stacked above the woofer units; and 16 tweeter horns in three stacks.

Buck describes the system as being 3-way, bi-amplified, with crossover points at 250Hz and 3.5kHz.

"We're using our CX-2 passive crossovers—which are 12dB per octave Butterworth filters—for the 250Hz crossover. They are cascaded in pairs, in a Linkwitz-Riley configuration, to give a 24dB per octave slope. For the 3.5kHz crossover, we are using a passive network to high-pass the tweeters, relying on the natural rolloff of the midrange units to complete the crossover."

Both the tweeter and the woofer components were standard Cerwin-Vega products, which Buck describes as "hot-rodded a bit" to suit the requirements of a large-scale reinforcement. The tweeter cabinet was a model RMH-3000, an exponential horn unit, fitted with the model JMH-1 driver (an 8Ω compression driver with a silicon elastomer surround).

Cerwin-Vega specifies the driver as being capable of handling 50W, with a rated sensitivity of 107dB (1W, 1 meter). In the dance festival system, tweeters were driven by model A-300 power amplifiers (300W per channel at 4Ω), with two tweeters per amplifier channel.

According to Buck, the tweeter cabinets have now been provided with new constant-coverage horns, and the company is working to extend the JMH-1's high-end response. One of the reasons for the change in horn design, he says, is to accommodate arraying—obviously a primary concern in large-scale reinforcement systems. The tweeters used in the Rose Bowl system were arrayed vertically to gain increased throw, and although the existing exponential horns were "never designed to be arrayed," their performance was nonetheless deemed adequate for the purposes of this test.

The woofer cabinet used at the Rose Bowl was the model L36PE, also a current product and referred to as a "junior earthquake horn." The enclosure is a folded-horn design with a 32Hz flare rate, driven by a single 18-inch low-frequency driver. Each cabinet was fitted with a mouth extender, which doubled the mouth area and increased the unit's response by 3dB at 30Hz. With the mouth extenders, the woofer unit has a rated sensitivity of 108dB (1W, 1 meter), and is specified to handle up to 400W RMS.

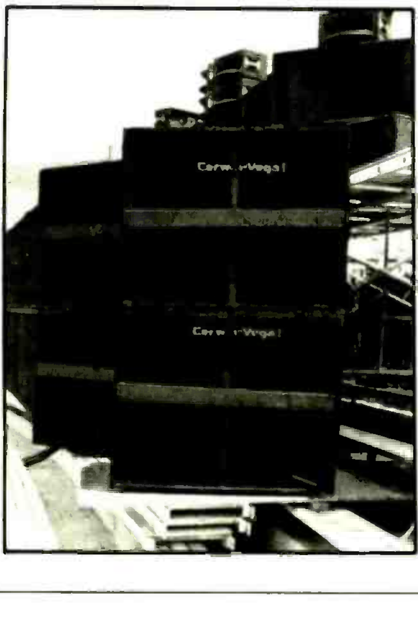
All but one of the woofer systems were driven with model LPA-600 power amplifiers; the remaining unit was powered by a prototype of a new switching amplifier, designated the SA-1000. Designed expressly for use at frequencies below 500Hz, the new unit weighs about seven pounds and is capable of delivering a claimed 1,200W RMS into a 16Ω load when powered with a 240Vac supply; connected to a 120V supply, the SA-1000 delivers 800W RMS into 16Ω.

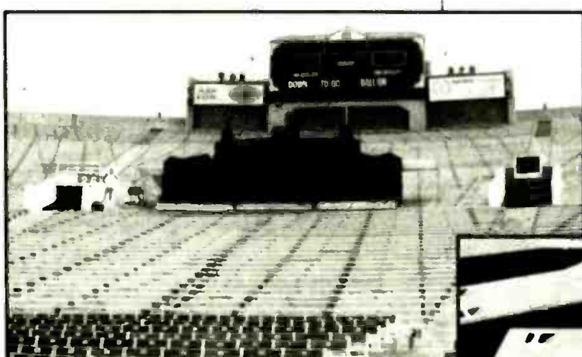
Because the amplifier does not use a power transformer, for safety reasons a ground-fault interrupter is incorporated on the ac line input, and the signal input is transformer isolated. The company plans to build the amplifier into its subwoofer cabinets, once final field tests like this one had proven the unit's reliability and effectiveness; the amplifier may also be offered as a separate component.

The Spruce Moose

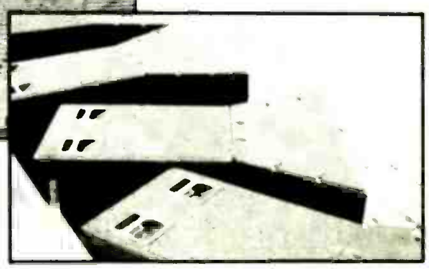
The real star of the dance festival system was a unique midrange horn that Cerwin-Vega president Gene Czerwinski has had under development for some time. Dubbed the Spruce Moose (the horn is made of wood), the unit is one of a family of three midrange horns whose design incorporates a single 90° fold. (The design was the subject of a paper that Buck presented at last October's

Cerwin-Vega model L36PE horn-loaded subwoofers, with mouth extenders, stacked one on top of the other.





Complete prototype sound system installed at the Rose Bowl for dance festival.



Top view of L36PE subwoofer showing technique of mounting the mouth extenders.

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AES Convention. Co-authored by Buck, E.J. Czerwinski and A. Duncan, the paper was somewhat whimsically entitled "Spruce Moose: A Slightly Bent Horn.")

The version used for the dance festival system was the largest in the projected family: It measures approximately 23 inches deep by 47 inches wide by 22 inches high, and weighs about 150 pounds. Coverage is specified as 40° vertical by 100° horizontal, and the horn is said to maintain good pattern control in the low end.

The cabinet houses four of the company's M-162 midrange compression drivers. Because each driver is rated at 150W RMS and 600W peak, the system has a quoted total power handling capability of 600W RMS, 2,400W peak. The system was designed to reproduce the entire mid-range region, and covers more than a decade: Cerwin-Vega specifies its frequency response from 250Hz to 3.5kHz, ±3dB, measured on a third-octave basis.

Although Cerwin-Vega has had the M-150 midrange driver in production for a couple of years now, the version used at the Rose Bowl was new. According to Buck, the major design change in the M-162 is the incorporation of a bored-out polepiece, backed with a chamber that is tuned lower than the previous version. This modification, undertaken concurrently with the development of the Spruce Moose, extends the driver's low-end range to 200Hz (vs. a lower limit of 300Hz for the earlier version).

The M-162 is designed with a relatively low (4:1) compression ratio. The unit's 4-inch diaphragm is fabricated from a phenolic resin cloth material; the voice coil is three inches in diameter and drives the diaphragm at a nodal point that Buck identified by observing the nodal breakup pattern of the dome. The polepiece is undercut for a symmetrical gap, and has an aluminum flux-stabilizing ring.

The combined effect of these design attributes is to reduce the measured third-harmonic distortion to a very low level—typically less than 0.3% at half the rated continuous power, according to Buck. Additionally, the phasing plug is relatively simple in design, with the result that the air overload distortion—a major contributor to the second harmonic component—remains approximately 0.5% at half power.

The four M-162 compression drivers are mounted within the box, two on each side of the center slot of the horn, above one another. Each driver's 2-inch throat is coupled—using a spacer that makes the mouth opening square—to an exponential flare that opens out to about eight by 1.5 inches.

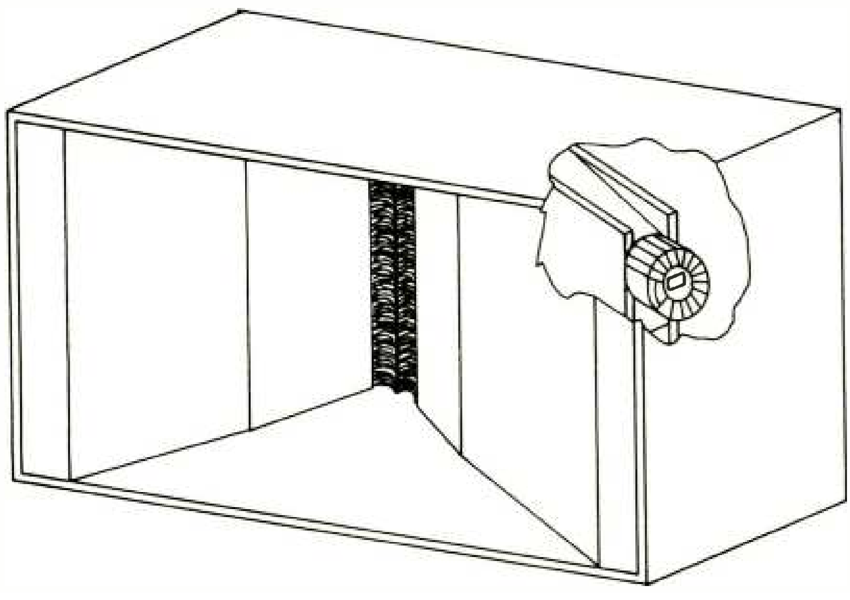
Each pair of drivers feeds one of two, 16-inch high, 1.5-inch wide slots at the center of the horn. The two slots stand side-by-side with a wedge-shaped spacer between them, and feed into the throat of the horn (which features several flares out to the mouth).

Each pair of drivers is wired in parallel in the cabinet and brought out to one of two connectors; because the M-162's nominal impedance is 16Ω, the resulting impedance at each connector is 8Ω. At the Rose Bowl, these two connections on each cabinet were wired in parallel, forming a 4Ω load.

Buck specifies the system's sensitivity as approximately 110dB, 1W, 1 meter. With 600W continuous power handling, the unit is thus capable of a maximum continuous output of 138dB SPL at 1 meter, or 128dB SPL at one-tenth power.

"This is a different approach than most people currently use for the midrange," Buck states. "It's funny. Cerwin-Vega was preaching cone midrange drivers 10 years ago — which was the last time we did any big outdoor stuff — because you had lower distortion; you weren't squeezing all the sound through a little slot. Now, the others are going toward cone midranges, and we're into compression drivers — but with a moderate compression ratio, so the distortion stays low."

Sectional view of Spruce Moose quad-driver midrange horn. Shown here is one of four drivers utilized in the design.



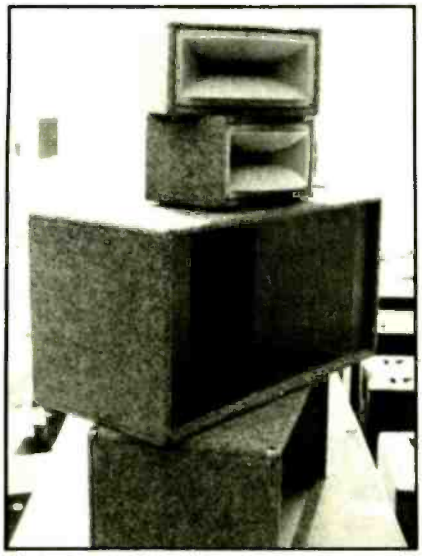
Testing and equalization

Two days before the Mormon Dance Festival began, Buck arrived at the Rose Bowl to test and equalize the completed sound system. His chosen test equipment comprised an Ivie third-octave analyzer and a Crown Tecron TEF-10 Time Delay Spectrometry (TDS) analyzer.

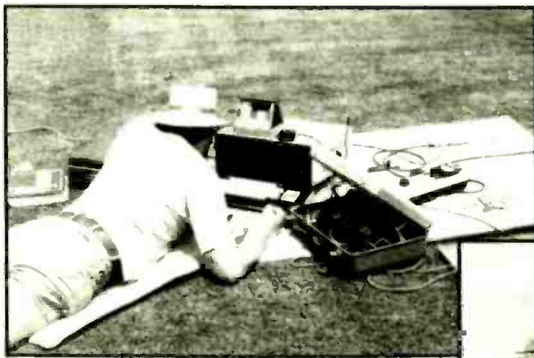
Buck selected a position on the playing field, about 250 feet away from the system, from which to take measurements. He used the microphone supplied with the Ivie analyzer as the measurement input to the TEF-10, and placed the mic on the ground near the measurement equipment.

After adjusting the delay of the TEF-10's tracking filter to compensate for propagation delay between the speaker system and measurement position, Buck began attempting to resolve the system's response. A substantial crosswind complicated the process, necessitating multiple passes to resolve the curve. Because the TEF generates a single sinewave sweep and then—after a brief delay—displays the measured response, the process took some time.

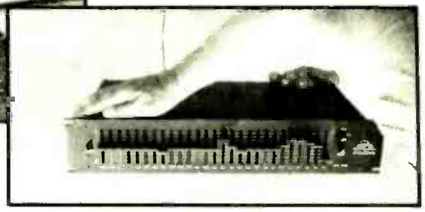
Finally, by widening the bandwidth of the tracking filter to smooth the curve and interpolating successive measurements, Buck was able to derive a usable curve.



Spruce Moose midrange horn with RMH-3000 tweeter horns stacked above.



During field-testing of the Rose Bowl prototype system, Cerwin-Vega design engineer Marshall Buck used a Crown TEF-10 analyzer to measure time and phase coherency.



Model TO-1 graphic equalizer with final frequency-response curve resulting from measurement session.

To equalize the system, Buck used a Cerwin-Vega TO-1 third-octave graphic equalizer patched at the analyzer's output. After further multiple passes and adjustments to the TO-1, he arrived at a reference equalization curve. The same TO-1 was then relocated to the amplifier rack.

Because the physical distance to the system resulted in a substantial rolloff above 10kHz, that region was not equalized. Buck later specified the final system response as being down about 8dB at 25Hz, with a slight rise in the low end, and ±3dB from 200Hz to 12kHz.

Following this process, Henry Austin and the assembled Philadelphia P.A. technicians played a number of music selections from Compact Disc. After some minor equalization changes made by ear to adjust the sound to taste, they pronouced the system ready.

The event

On the night of the festival, Glen Glancy operated the system from the announcer's booth, located high above the crowd near the lip of the Rose Bowl. Pre-assembled tapes, combining narration with musical selections, were replayed from a pair of Otari MX-5050 two-track machines. The sources were combined via a Yamaha M508 console, and the resultant mono mix was fed to a second console located at the amplifier rack behind the loudspeaker array. PPA technicians manned this position and monitored the system's performance.

When the Cerwin-Vega prototype system was tested and auditioned in the empty Rose Bowl, there was significant sound bounce-back from the seating areas, particularly those directly opposite the system. These multiple echoes tended, naturally, to reduce the system's overall intelligibility.

On the night of the dance festival, however, the Bowl was filled to capacity; even the playing field was for the most part, a sea of dancers. In this far more absorptive environment, the echoes were reduced to such low levels that they vanished beneath the direct sound of the system. As a result, the sound was extremely clear. Coverage was remarkably even, with every announcement audible and intelligible even at the uppermost regions of the stadium—no mean achievement, considering that the longest throw exceeded 750 feet.

The festival's purpose was, of course, to dance, and the program featured highly rhythmic popular as well as folk music. The finale, in particular, imbued with partriotic overtones, and featuring Neil Diamond's "America," demanded considerable energy from the Cerwin-Vega array. The projection capabilities of the all-horn system proved equal to the task; however, the pulse of the music could be felt high up in the stands, and the impact of the kick drum carried to the far reaches of the stadium. Judging from the crowd response, the music had the desired emotional impact—which testifies to the effectiveness of the system.

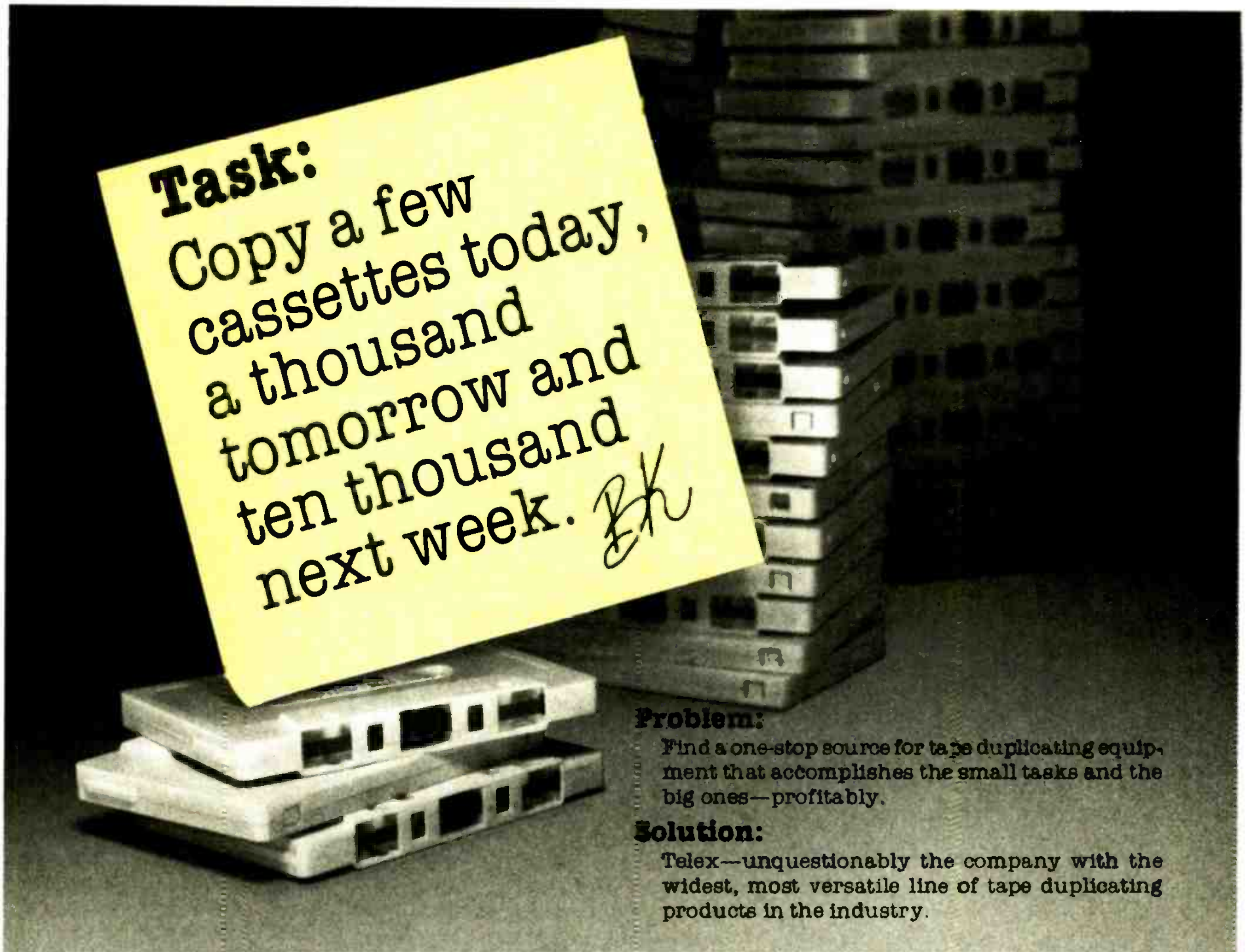
"Our purpose in coming out here," Buck says, "was not simply to provide music for this event, but also to get back into music playback and outdoor PA—to get involved with some sound-system companies, see what they need, and then provide it.

"The next step for us is more careful study of the interaction between units when you array them. Should you wish to add more units to get more level, you wouldn't want to go sideways here; instead, you'd want to build the system up vertically. But that's just not convenient in this circumstance, it being only a two-day event. Moreover, what we have here is more than sufficient for the job."

It would appear that Buck's last statement is justified. Rose Bowl officials were not the only people to express enthusiasm for the system's performance. Continental Sound's Glen Glancy pronounced the system to be "the best that we've ever had for this event." Philadelphia P.A. president Henry Austin—whose enthusiasm for Cerwin-Vega's R&D efforts made the field test possible—was clearly pleased.

"This system is a testimony to Gene Czerwinski, who is one of the best minds in the audio industry," Austin says. "His innovative approach is refreshing to see in an industry that has undergone only minimal changes in the last few years."

The performance of Cerwin-Vega's prototype belied its small physical size, delivering sound that approached moderate concert levels in a 110,000-seat stadium. Given that economic realities are today forcing sound-reinforcement professionals to seek ever more power from ever smaller systems, the development is promising.



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Digital Recording Formats

By John Monforte

A discussion of encoding and error-correction schemes for DASH and PD digital.

Although standards are not needed to make a product that will work well, they can help reduce the cost of production and service by using components available through multiple sources and manufactured under the principles of economy of scale. Also, there will be a greater acceptance of a product if the user feels it will be compatible with other similar products made by other manufacturers, as well as having upward compatibility with future, improved versions.

Although standardization becomes more acute when a drastically new device is proposed, it does not compare to the requirements of commonality needed to convert an entire system to a new technology. Just such a situation is facing the audio industry, as it begins the task of switching over the generation, manipulation, recording and dissemination of sound from analog to digital technology.

Within the audio industry, there have been many examples of good standards being selected. MIDI, for example, seems to be gaining rapid acceptance. Many of us have great expectations for the SMPTE/EBU interface bus, which should reduce dramatically the complexity and cost of transport synchronization.

John Monforte is a lecturer in the University of Miami's Music Engineering Program.

Our industry has had to form many standards to allow the adoption of digital technology for music recording and production; the Audio Engineering Society has undertaken the task to formulate many of them.

The AES committee has already drafted a few standards, with others soon to follow; proposals on such matters as sampling rates, clocking frequencies, I/O interfacing, data formats and measurement techniques are nearing completion.

Meanwhile, the pursuit to introduce digital audio continues in a healthy atmosphere of innovation and competition. Efforts by 3M, Denon, JVC, Soundstream, dbx, Sony, Mitsubishi, Studer and many other pioneering manufacturers, along with input and support of end users and many recording studios, have already produced and made obsolete several generations of digital audio equipment.

Now that the frontier is being tamed, we are presented with a narrower range of accepted options. Specifically, in the area of tape machines, there appears to be a wide acceptance of digital recorders for stereo mastering. Granted, there are a few different, incompatible systems available, including VCR-based processors and stationary-head designs. Some are data-compatible with Compact Disc mastering and some are not. Some are easy to edit with and others are less so.

Sony PCM-3000 Series 2-channel ¼-inch DASH-format transport, which is utilized in the PCM-3102 and PCM-3202 Twin-DASH recorders.

However, the generally low cost of such 2-track recorders has allowed them to proliferate to the point that the buzzwords "digitally mastered" are less astounding than they were only a few years ago.

Digital multitrack formats

Such is not the case with multitrack recorders. Because the market requirement for these machines will always be significantly smaller than that for stereo recorders, manufacturers find it more difficult to secure a return on their design and manufacturing investments. Sufficient multitrack machines will be used, however, to justify at least some measure of compatibility among them.

The first format to be developed was DASH, or Digital Audio Stationary Head, which has been agreed to by Sony, Studer, Matsushita and TEAC. The newer format is called Professional Digital, or PD, and is being used in machines from Mitsubishi, AEG and Otari.

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properties DASH and PD, it would be useful to consider the features required in any sort of multitrack recorder for both electronic and artistic reasons.

The design of any digital device will depend upon the word size and data rate of the signal it is designed to handle, because these factors determine the width of the data highway and the clock frequencies required. Already firmly established is a 16-bit linear, pulse code modulation (PCM) encoding technique, with a sample rate of 48kHz. (The Compact Disc rate of 44.1kHz still exists as a popular alternative and is likely to remain so.) These factors will govern design elements such as data density and structure of error-handling processors.

Although these parameters are instrumental in designing the analog-to-digital and digital-to-analog converters, the converters themselves are not really a function of a recorder. Ideally, they should be located in the signal flow next to the transducers. In the meantime, until digital mixing and processing devices come into widespread use, the converters will be housed in the recorder.

At this point it would be wise to consider features that are contained in the format used for CD mastering. For now and the foreseeable future, CD will be the primary digital audio format available to the consumer market. A multitrack machine could benefit from the use of components that can be produced at low cost.

Also, data-format similarities could reduce the need for more analog conversions (and therefore signal degradation) before the sound reaches the customer. For example, filters, D/A converters and oversampling technology can be used directly if the word size and data rate are similar.

On the other hand, it would not be logical to borrow the CD's data-encoding and error-correction methods, because CD error modes are unique to the medium and different from magnetic tape.

It would be sensible to frame the encoded data for the recorder at the same rate as that used for video signals. This approach would not only allow for simple video synchronization, but would also allow the machine to lock up into larger systems with the addition of a dedicated SMPTE/EBU time code track.

Both DASH and PD formats allow such synchronization with an analog track dedicated to recording time code. At first glance it would seem more economical to store the time code data, which is a digital bitstream to begin with, in the digital domain. This is not done because, in order for it to be decoded and error-corrected, digital data must be read near play speed. An analog time code track can be read in both directions and in fast-wind modes, allowing for accurate, rapid lockups.

Many 2-track digital machines comprise an audio processor that converts the analog signal to PCM digital and then transforms it into a video signal that can be recorded on a conventional ½-, ¾- or 1-inch videocassette or videotape recorder. Such devices have the principle virtue of allowing the storage of information at very high densities, typically more than an hour of 2-channel 16-bit/44.1kHz audio on a standard U-matic VCR.

When the number of tracks is multiplied by a factor of 16, however, the density requirement is increased proportionately. Rotary-head multitrack recording with VCRs and VTRs would demand either wider tape or higher writing speeds—in other words, the development of all-new hardware. Even so, there are some more obstacles. Because a single head assembly is used to record and replay all the digital-audio signals, re-recording of an individual audio track would require the re-writing of all other data.

This in turn would mean that two synchronized transports are required for a punch-in or overdub. Also, razor blade editing would not be possible; instead, editing would have to be done electronically, which is slower and more costly.

It appears that a stationary-head, reel-to-reel format would be best suited to multitrack applications. Fortunately, while early experiments in digital recording were underway, there was a parallel effort to develop thin-film heads for high-density data recording. A recording density also was defined to provide an acceptably low error rate, while allowing for real-world conditions of tape handling. As tape and head technologies mature, an extra measure of robustness can be expected.

All DASH-format transports, regardless of speed or track format (for reasons soon to be explained), use a density of 38.4Kbits per longitudinal inch, while the PD multitrack format is slightly more relaxed at 29.9Kbpi. PD also specifies a separate 2-channel format that allows the density to change from 20Kbpi at 15ips to 40Kbpi at 7.5ips. As a result of these densities, DASH should give the same error rate as read from the tape for all versions, while PD will tend to vary.

Depending on the error modes and the error-correction algorithms used by the recorder, these errors will be either corrected or concealed. It should be noted that although a format may allow for extensive error correction to occur, a manufacturer may elect to forego performing many of the complex calculations available to correct data and instead choose to conceal a greater number of the errors. In this way the manufacturer can build a machine that is less expensive than others in the same format.

In practice, an error rate of about one per second is reasonable and, if the tape appears to be deteriorating, it is advisable to copy the program while the error-correction circuitry is still able to determine the correct values. (This inconvenience should be relieved as tape and head technology develops.)

We now enter the areas where DASH and PD formats diverge. Each offers advantages over the other, but in most cases any advantage gained by one format in any particular parameter usually translates into a disadvantage when another parameter is considered. Table 1 shows a comparison betwen DASH and PD formats for different channel configurations.

Digital Audio Stationary Head

DASH was developed as a comprehensive fixed-head, open-reel format that can easily be extended over a wide range of tape speeds, channel numbers and tape widths[2]. Originally, ¼-inch 2-track and ½-inch 24-track versions were specified, but common elements can be extended easily to many other yet to be specified versions.

Sampling rates of 48kHz, 44.1kHz and 32kHz are supported, by varying the tape speed slightly so that the recording density (and thus the error rate) is held constant. Three tape speeds are possible at any given sampling rate. In fast-speed mode (30ips at 48kHz) one tape track is used per channel and contains the signal data along with all the other data used for error correction. Medium-speed mode is accomplished by running tape at half the fast speed (15ips at 48kHz), and sharing the data among two tracks. Of course, only half as many audio channels may be recorded on a given width of tape. Low-speed mode uses four tracks

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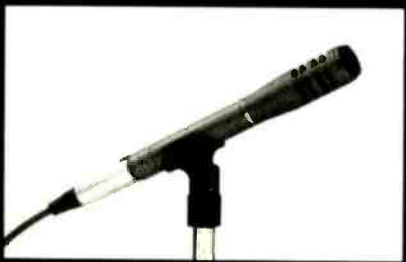

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per channel, and runs at half the medium speed (7½ips at 48kHz).

Using these techniques, much of the hardware can be shared between transports. Data from the converters can be multiplexed among the tracks as needed, and then encoded with check words prior to being recorded. It is easy to see how 4-, 16- or 48-track DASH-format machines could be achieved using this method, although such versions have not been formally adopted.

As thin-film head technology develops further, DASH will allow for recording at double the present data density, and will enable twice as many audio channels for a given tape width and speed. Provisions have been made in the DASH format to allow this adaptation to occur, while enabling a new machine to play older, single-density tapes.

Recently, DASH has provided a 2-channel format for use at 15ips. In the spirit of DASH, the data density remains the same, and the only difference is that the data are now repeated twice on the tape, thus cutting in half the error potential. There is also a delay for recording the second set of data, further reducing the possibility that a local defect on the tape will be uncorrectable. These changes are described as being simple and inexpensive to implement on a recorder[3].

DASH also requires each tape width format to include additional cue and auxiliary tracks, as shown in Figure 1. They are located at the edge of the tape in order to place data tracks nearer the center where there is less chance for damage. One track is used for time code, and another as a reference track that performs a function similar to the control track of a rotary-head VCR. The reference or Aux 2 track provides framing to the serial data streams, which is necessary to determine the function and significance of each bit. The capstan motor also monitors this signal to derive the correct speed and hence the proper data rate of the recording.

Aux 2 also contains sequential sector addresses that assist in the editing process, and can potentially allow a second recorder to synchronize to within an accuracy of one sample[4]. The track density of the auxiliary tracks is about 5% of that used for the audio data, and will be less sensitive to dropouts that could cause a loss of the capstan-servo lock. Also, by virtue of the track's low data density, it is possible to read it in fast-wind modes.

There are two other tracks on the opposite edge of a DASH-format tape that allow for either the recording of two analog stereo cue channels, or a mono channel plus an auxiliary data channel. The cue channels are essential for rock-and-roll of tape reels during razor blade editing.

Because of the PCM digital encoding, a user cannot expect the recorder to be able to deliver a signal when the tape speed is far outside its nominal play-speed range, or when in reverse mode. The analog auxiliary data channel can be used for a variety of functions not defined in the format—for example, console automation data, or synthesizer sync signals.

The four analog tracks may not necessarily be recorded by the same head as the one used for the digital signals. Even if they are, there will be a processing time delay from when data bits are read until the error-corrected 16-bit words arrive out the channel outputs.

The analog signals, on the other hand, can be retrieved instantaneously. In order to maintain correct sync between the digital audio and auxiliary tracks, delays (or head spacings) need to be defined. These details are based on the data formatting and error-correction requirements. Although by themselves they do not affect the performance of the standard, such parameters need to be defined so that tapes will be interchangeable between different machines.

DASH-format transports that are available, or soon to hit the pro-audio market, include the Studer D-820X Twin-DASH (15ips) 2-track; the Sony PCM-3102 and PCM-3202 2-tracks (the former using the standard 7.5ips tape speed and the latter 15ips plus Twin-DASH encoding); and the Sony PCM-3324 24-track ½-inch machine.

Table 1a. Summary of DASH 2- and 24-channel format.

Format	DASH 2-channel	DASH 24-channel
Tape speed	7.5 & 15ips	30ips
Tape width	¼-inch	½-inch
Sampling frequency	48, 44.1 & 32kHz	48, 44.1 & 32kHz
Quantization	16-bit linear PCM	16-bit linear PCM
Number of tracks	12	28
Tracks per channel	4	1
Bit density	38.4Kbpi	38.4Kbpi
Moduation	2/4 M	2/4 M
Error correction	RSC & CRC	RSC & CRC

Table 1b. Summary of various 2-, 16- and 32-channel Prodigital formats.

Format	PD 2-channel high rate	PD 2-channel low speed	PD 2-channel high speed	PD 16-channel	PD 32-channel
Tape speed	15ips	7.5ips	15ips	30ips	30ips
Tape width	¼-inch	¼-inch	¼-inch	½-inch	1-inch
Sampling frequency	96kHz	48 & 44.1kHz	48 & 44.1kHz	48 & 44.1kHz	48 & 44.1kHz
Quantization	16- or 20-bit linear PCM	16- or 20-bit linear PCM	16- or 20-bit linear PCM	16--bit linear PCM	16-bit linear PCM
Number of tracks	12	12	12	45	25
Tracks per channel	4	4	4	1.25	1.25
Bit density	40Kbpi	40Kbpi	20Kbpi	29.9Kbpi	29.9Kbpi
Modulation	2/4 M	2/4 M	2/4 M	4/6 M	4/6 M
Error correction	RSC-IV & CRC	RSC-IV & CRC	RSC & CRC	RSC & CRC	RSC & CRC

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TEAC plans to unveil a range of Tascam stereo and multitrack DASH machines by the middle of 1987. [At press time, Sony announced a switchable 7.5/15 ips machine, the PCM-3402, which will also feature comprehensive digital editing and synchronization capabilities not offered on its existing 2-track transports — Editor.]

Professional Digital

PD is a set of formats that evolved from early development work by Mitsubishi. It consists of a 2-channel format and a similar, yet distinct, multichannel format. While on one hand some

Figure 1. DASH standard-density, 2- and 24-channel formats make use of eight and 24 data tracks at respective speeds of 7.5 and 30 ips, on ¼- and ½-inch tape.

DASH 2-CHANNEL FORMAT

```
Track 12 ***** CUE L (L+R) ****************
Track 11 ***** CUE R (Aux) ****************
Track 10 ///// CH 2 ///////////////////////
Track  9 \\\\\ CH 1 \\\\\\\\\\\\\\\\\\\\\\\
Track  8 ///// CH 2 ///////////////////////
Track  7 \\\\\ CH 1 \\\\\\\\\\\\\\\\\\\\\\\
Track  6 ///// CH 2 ///////////////////////
Track  5 \\\\\ CH 1 \\\\\\\\\\\\\\\\\\\\\\\
Track  4 ///// CH 2 ///////////////////////
Track  3 \\\\\ CH 1 \\\\\\\\\\\\\\\\\\\\\\\
Track  2 +++++ Aux 2 (Control Reference) ++
Track  1 +++++ Aux 1 (Time Code) ++++++++++
```

DASH 24-CHANNEL FORMAT

```
Track 28 ***** CUE L (L+R) ****************
Track 27 ***** CUE R (Aux) ****************
Track 26 ///// CH 24 //////////////////////
Track 25 ///// CH 23 //////////////////////
Track 24 ///// CH 22 //////////////////////
Track 23 ///// CH 21 //////////////////////
Track 22 ///// CH 20 //////////////////////
Track 21 ///// CH 19 //////////////////////
Track 20 ///// CH 18 //////////////////////
Track 19 ///// CH 17 //////////////////////
Track 18 ///// CH 16 //////////////////////
Track 17 ///// CH 15 //////////////////////
Track 16 ///// CH 14 //////////////////////
Track 15 ///// CH 13 //////////////////////
Track 14 ///// CH 12 //////////////////////
Track 13 ///// CH 11 //////////////////////
Track 12 ///// CH 10 //////////////////////
Track 11 ///// CH  9 //////////////////////
Track 10 ///// CH  8 //////////////////////
Track  9 ///// CH  7 //////////////////////
Track  8 ///// CH  6 //////////////////////
Track  7 ///// CH  5 //////////////////////
Track  6 ///// CH  4 //////////////////////
Track  5 ///// CH  3 //////////////////////
Track  4 ///// CH  2 //////////////////////
Track  3 ///// CH  1 //////////////////////
Track  2 +++++ Aux 2 (Control Reference) ++
Track  1 +++++ Time Code ++++++++++++++++++
```

Figure 2. PD 2- and 32-channel formats utilize ¼- and 1-inch tape running at 7.5/15 and 30 ips respectively.

PD 2-CHANNEL FORMAT

```
Track 12 ***** Aux Analog 2 *************
Track 11 ***** Aux Analog 1 *************
Track 10 ///// Ch 1 and 2 ///////////////
Track  9 ///// Ch 1 and 2 ///////////////
Track  8 ///// Ch 1 and 2 ///////////////
Track  7 ///// Ch 1 and 2 ///////////////
Track  6 ///// Ch 1 and 2 ///////////////
Track  5 ///// Ch 1 and 2 ///////////////
Track  4 ///// Ch 1 and 2 ///////////////
Track  3 ///// Ch 1 and 2 ///////////////
Track  2 ##### Aux Digital ##############
Track  1 +++++ Time Code ++++++++++++++++
```

PD 32-CHANNEL FORMAT

```
Track 45 ##### Aux Digital 2 ###########
Track 44 ***** Aux Analog 2 ************
Track 43 ***** Aux Analog 1 ************
Track 42 ///// Ch 32 Group D ///////////
Track 41 ///// Ch 24 Group D ///////////
Track 40 ///// Ch 16 Group D ///////////
Track 39 ///// Ch  8 Group D ///////////
Track 38 ///// Ch 31 Group D ///////////
Track 37 ///// Ch 23 Group D ///////////
Track 36 ///// Ch 15 Group D ///////////
Track 35 ///// Ch  7 Group D ///////////
Track 34 ||||| Ch 30 Group C |||||||||||
Track 33 ||||| Ch 22 Group C |||||||||||
Track 32 ||||| Ch 14 Group C |||||||||||
Track 31 ||||| Ch  6 Group C |||||||||||
Track 30 ||||| Ch 29 Group C |||||||||||
Track 29 ||||| Ch 21 Group C |||||||||||
Track 28 ||||| Ch 13 Group C |||||||||||
Track 27 ||||| Ch  5 Group C |||||||||||
Track 26 ///// Q-Parity Group D ////////
Track 25 ||||| Q-Parity Group C ||||||||
Track 24 \\\\\ Q-Parity Group B \\\\\\\\
Track 23 !!!!! Q-Parity Group A !!!!!!!!
Track 22 \\\\\ Ch 28 Group B \\\\\\\\\\\
Track 21 \\\\\ Ch 20 Group B \\\\\\\\\\\
Track 20 \\\\\ Ch 12 Group B \\\\\\\\\\\
Track 19 \\\\\ Ch  4 Group B \\\\\\\\\\\
Track 18 \\\\\ Ch 27 Group B \\\\\\\\\\\
Track 17 \\\\\ Ch 19 Group B \\\\\\\\\\\
Track 16 \\\\\ Ch 11 Group B \\\\\\\\\\\
Track 15 \\\\\ Ch  3 Group B \\\\\\\\\\\
Track 14 !!!!! Ch 26 Group A !!!!!!!!!!!
Track 13 !!!!! Ch 18 Group A !!!!!!!!!!!
Track 12 !!!!! Ch 10 Group A !!!!!!!!!!!
Track 11 !!!!! Ch  2 Group A !!!!!!!!!!!
Track 10 !!!!! Ch 25 Group A !!!!!!!!!!!
Track  9 !!!!! Ch 17 Group A !!!!!!!!!!!
Track  8 !!!!! Ch  9 Group A !!!!!!!!!!!
Track  7 !!!!! Ch  1 Group A !!!!!!!!!!!
Track  6 ///// P-Parity Group D ////////
Track  5 ||||| P-Parity Group C ||||||||
Track  4 \\\\\ P-Parity Group B \\\\\\\\
Track  3 !!!!! P-Parity Group A !!!!!!!!
Track  2 +++++ Time Code +++++++++++++++
Track  1 ##### Aux Digital 1 ###########
```

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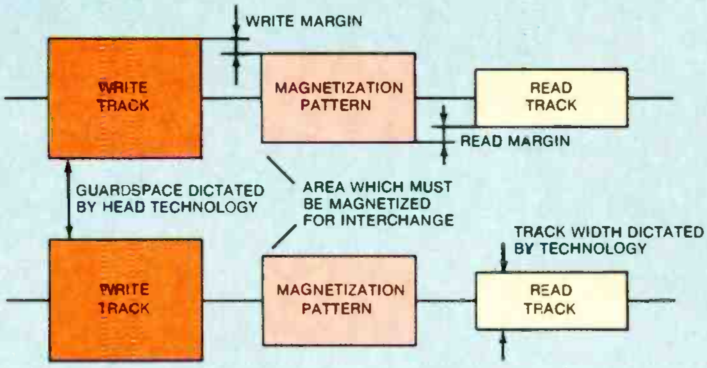
Recording Digital Audio with Stationary Head Transports

By Roger Lagadec, Ph.D.



The choice of a track geometry.

Digital audio recorders must, first of all, accommodate the physical properties of tape. Quarter-, ½- and 1-inch video-grade tape is commonly used for digital reel-to-reel recorders. With a thickness between 26 microns and 29 microns, videotape is considerably thinner than analog audio tape, making it possible to achieve longer playing times with the same reel size while keeping the familiar tape speeds of 30ips, 15ips and 7.5ips.

However, digital audio requires high bit rates and in turn a large bandwidth. One channel of 16-bit/48kHz digital audio, without any redundancy, requires 768,000 bits per second, or approximately 400kHz of bandwidth—much more than the bandwidth offered by analog recorders at 30ips.

Because it is unrealistic to increase the tape speed, and therefore tape consumption, the necessary solution is to operate with a high recording density; in other words, to push recording technology by roughly a factor of 10 to the domain of microns.

Recording data bits is possible with signal-to-noise ratios much lower than in analog, because bit recognition, rather than precise analog waveform reproduction, is now the basic discipline. Therefore, much narrower tracks than in analog recording can be used, with their width limited by mechanical considerations (tracking and tape guidance) rather than by signal-to-noise ratio.

In addition, bits and words originating from one channel can be distributed over several tracks, thus reducing the bit rate and density per track. This method, however, is not yet realistic with multichannel recorders, because it would lead to an unrealistic number of very narrow tracks. Finally, the recording bandwidth can be reduced by almost 33% by coding the bit streams prior to recording.

Digital recording with stationary heads always results in a tape with many narrow tracks and high-density recording. Critical mechanical tolerances are unavoidable. Consider 2-channel digital recording at 7.5ips with the geometry proposed by several IEC working groups as shown in the figure above. The minimum wavelength for recording is typically 2 microns. In ferrite technology, the width of a read track is typically 160 microns, and a write track is 340 microns. The tape can be cut to 20-micron width tolerance.

For reliable recording and playback, a tape guide system with an overall accuracy of ±30 microns or greater is needed, and an azimuth error below ±0.05 degrees. Both figures are difficult to achieve without careful adjustments; high-precision mechanical engineering is even more important with digital than with analog recordings.

Several head technologies find application in digital audio. With ferrite and similar material, miniature mechanical engineering is used in manufacturing. Mechanical accuracy, tooling and material brittleness yield a rather large guard space between write tracks (typically 200 microns), and a minimum read-track width (typically 160 microns). With thin-film head technology, the accuracy of the head itself improves, and the guard space can be reduced. Thus, for the same magnetization pattern, wider tolerances can be accepted in the tape guide system.

Bandwidth is also a limitation in digital recording. The amplitude and phase response (as well as azimuth errors) distort the data signals. As a result, it is necessary to use equalizers that reshape the date signals, or to use digital processing for the same purpose. When the data equalizers are not adjusted optimally, the signal from a track can easily get lost. In analog recording, the result is a far less severe, gradual degradation of sound quality. Today, exchanging digital tapes is still critical than with analog, although there has been steady improvement over the past few years.

All digital audio formats have several techniques in common. First, redundant data are computed and added to the original data. This allows the correction of detected errors up to some level. (One must insist that, in the case of successful error correction, the output signal is as true to the original as if errors had not occurred in the first place.)

Second, interleaving—or the shuffling of data prior to recording, compensated by their de-shuffling at playback—is used because such a technique reduces the effects of dropouts, as shown in the figure above. Third, the data are formatted into blocks, beginning with a sync work and ending with a CRC; this technique allows the detection of blocks with errors.

Although the basic techniques of error protection are well-known and universal, there are several possible strategies for dealing with dropouts, fingerprints, track loss and splicing.

It is also worth remembering that mechanical quality and the "smartness" of signal processing are at least as important as theory; expressions such as "one click every one million years" is only true on paper.

Roger Lagadec, Ph.D., is general manager of technology management, communications products group Sony Corporation and works at the Atsugi facility in Japan.

Sony PCM-3324 ½-inch DASH-format 24-track.

engineers and producers may be annoyed by the changes of direction (the 2-channel format is not upwardly compatible with the old one), the new standards contain improvements based on a great deal of field experience with the older versions, and even with DASH itself. Developers of the PD format are confident that its virtues will allow rapid acceptance, despite PD's relative newness to the recording market.

Perhaps the most fundamental difference with the PD format is the lack of a dedicated control track. Just as with DASH, there needs to be a way of unambiguously framing a continuous serial bitstream, to indicate which bits are data, which are for error detection and what position a given bit will hold in a data word.

Because PD includes framing information on each data track, any track can be used to derive the capstan speed and determine output clocking requirements, even if one track is badly damaged. (It must be said that a manufacturer may elect to rely on only one of the tracks to handle all the formatting and never opt to switch if there are problems, thereby making the machine format-compatible but cheaper.)

The PD multitrack format allows for 32 audio channels to be recorded onto 1-inch tape, as shown in Figure 2. A 24-track version is specified, which simply omits recording on any tracks devoted to the unused channels. A 16-track version uses ½-inch tape, and is otherwise identical to the 32-track. A separate 2-channel on ¼-inch format also is specified, but it departs significantly from that of the multitrack one[5].

The 2-channel version is capable of supporting word lengths of up to 20 bits; if 16-bit converters are used, the four least significant bits (LSBs) are assigned to zero. Two-channel machines can take three forms: a low-speed/high-density version, a high-speed/low-density version, or a high-speed/high-density version with a doubled sampling rate.

All versions include four extra channels: one for time code, another for auxiliary data and two audio "scratch" channels for editing. The multitrack versions also contain a second auxiliary data track. Like DASH-format machines, provisions are made for both 48kHz and 44.1kHz sampling rates.

The four auxiliary tracks are not actually recorded in an analog fashion. Instead of trying to linearize digital recording tape through the addition of bias, PD designers opted to record these signals using pulse width modulation (PWM) techniques. Like PCM recording, PWM takes advantage of the saturation recording properties of digital tape. It differs substantially from PCM in many aspects, however the amplitude of the analog waveform determines the duty cycle of a pulse train. At rest, the output signal is a symmetrical square wave.

As the analog signal goes positive, the output persists in the "high" state longer than in the "low" state, and vice versa. Although the fidelity of this method is not optimum for critical sound recording, it uses simple converters, requires no error correction and can be reproduced over a wide speed range, even in reverse.

Tape speed of the 2-track PD format can be either 15ips or 7.5ips; the slower speed has identical block structures and merely doubles the data density. This important feature of 2-track PD can be seen as either an asset or a frivolity, depending on your viewpoint. One one hand, it is easier and cheaper to make a 2-speed recorder in the PD format, because no special data-formatting changes are required. High-speed machines will have half the errors of a low-speed transport, and offer a much lower data density than DASH, making it more robust.

However, there is every reason to expect that tape and heads will improve dramatically. Assuming that the higher data densities are adequate for current use, the high-speed format may fall into complete disuse in the future, unless razor blade editing is needed. (It should not be forgotten that a narrow, analog cue track recorded at 7.5ips may be difficult to use during rock-and-roll editing; doubling the record speed to 15ips will not only improve replay quality of the cue tracks, but also enable more precise edit location.)

The unique feature of 2-track PD is the adoption of a 96kHz sampling-rate version that retains the packing density of the low-speed version but runs tape at the higher speed.

In the 2-track format, data from both channels and their error-check words share all of the eight available data tracks. The re-recording of any channel requires a rewrite of all the data. Apparently, developers of the PD format decided that the 2-channel machine is to be used for stereo recording and therefore does not require individual channel punch-in. This and other considerations that make the process of mixdown different from multitrack recording allowed the option of developing error-correction and data-formatting methods that are optimized for the application. Hence the difference between PD multitrack and 2-channel recorders.

For PD-format multitracks, a data-formatting method is used that is quite unlike DASH. While each audio channel is encoded onto one tape track, each group of eight channels is associated with a pair of tracks that contain the P and Q Reed-Solomon check words, respectively. By splitting data among tracks, as shown in Figure 2, the chance for serious error to a single signal is reduced. Of course, if a punch-in is performed, all three tracks (two parity plus data) must be rewritten, the parity data of unchanged tracks also being re-recorded at that time.

PD-format transports that are available, or soon to be released, include the Mitsubishi X-86 2-track, 15ips machine with switchable 48 and 44.1kHz

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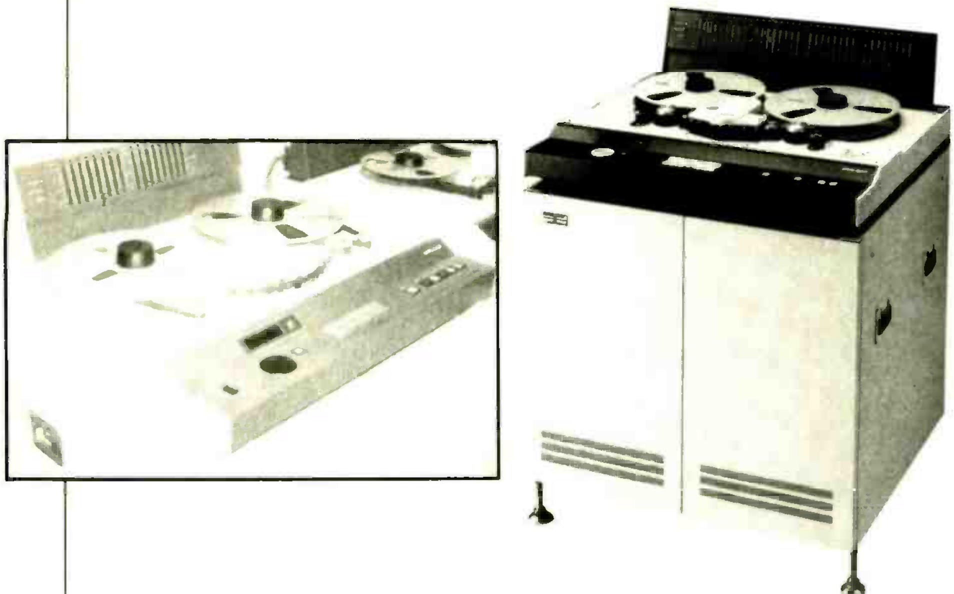

Otari's PD-format digital multitracks include the DTR-900 32-track 1-inch, with accompanying autolocator (right); and DTR-400 ½-inch 16-track machine.

sampling frequencies; the X-850 32-track on 1-inch machine; and the Otari DTR-900 32-track on 1-inch (also available in a 24-track version, prewired for 32-track operation).

The Mitsubishi X-86HF 2-track with a 96kHz sampling frequency will be made available in December 1986, while the X-86LT 7.5ips 2-track is scheduled for delivery in March 1987. The Mitsubishi X-400 16-track on ½-inch machine was unveiled earlier this fall. Otari's 16-track on ½-inch transport, designated DTR-400, also is scheduled for delivery in the late fall, while its 2-track machine, the DTR-200/100, is expected in the first quarter of 1987.

AEG, the third company supporting the PD format, is rumored to be developing its own transports. Meanwhile, it will continue to market Mitsubishi X-86 and X-850 units in Europe under similar model designations.

Two-track tapes made on earlier Mitsubishi X-80 machines are incompatible with the PD-format X-86, although the company says that a playback option for replaying X-80 tapes on the X-86 will be made available at the end of 1986. Tapes recorded on the earlier X-800 32-track are fully compatible with the newer PD-format X-850 digital multitrack.

Error correction

Both DASH and PD formats use error-correction schemes that are well-suited for the tape medium, including the techniques of redundancy, interleaving, cyclic-redundancy checking and Reed-Solomon parity checking; they only vary in the form of execution. Such technical topics have been widely discussed in the literature. If you are interested in learning further details and are mathematically inclined, the *Journal of the AES* is an excellent source. A less technical description without any of those dreaded polynomials is also available [6].

Sources of data errors include badly written data, tape defects and dropouts, contamination of heads and tape, punch-ins and splice edits. Both formats have allowed for all those sources, and provide mechanisms whereby the damaged data can be either reconstructed or, if that is not possible, to be concealed.

It is not easy to scan the DASH and PD specifications and determine which format will perform better under which circumstances. For instance, you might think that the more error-correction words that are included, the safer the data will be. But it should be remembered that now there are more words to be recorded on a given space of tape, providing a greater chance that something may be lost.

Interleaving is considered desirable, because adjacent data are scattered up and down the length of tape. When a dropout occurs, only a small amount of any given digital sound sample is damaged, making the restoration process easier. On the other hand, if a splice edit is encountered, more words are corrupted.

Digital technology provides the possibility of electronic editing, which can result in undetectable edits. As with video editing, two machines and a fairly elaborate controller are needed.

Unlike videotape, however, the designers of both formats have made provisions for razor-blade editing, which requires the tape to be placed on a rough textured block of metal, sawn in half with no regard to data framing, and then joined back together with another similarly mutilated piece of tape using a patch that contains gooey adhesives. This wreaks havoc on the error handlers.

Also, one should consider the implications of using this method on a medium that relies on time code. Discontinuous code is complicated to handle, and the process of restriping allows for countless opportunities to add to the confusion.

On balance, it appears that the audio industry, after several years of pioneering work, now has been provided with two effective open-reel digital standards. This discussion should assist you in making an informed decision on how to enter the market for digital mastering. By now, most of the major recorder manufacturers have elected to compete in the marketplace with one format or the other. That they have lined up behind one of the standards is both an endorsement of the practicality of the standard and a virtual guarantee there will be resistance to any capricious changes.

There is one more important factor to consider: the two formats are not completely incompatible. With the soon-to-be adopted AES/EBU digital data-bus format, data can be copied from one format to the other without any signal alteration. It is possible, therefore, to begin a project in one format and complete it in another. R·E/P

Acknowledgement: Thanks to Al Simons of Sony, Peter Germanson of Digital Entertainment Corporation, U.S. distributors of Mitsubishi products, and Jeff Phillips of Otari for the time and effort they spent in answering my many questions and reviewing my work. Although these individuals aren't the engineers who designed the formats, their roles as liaisons between the designers and the end users ultimately will prove crucial in industry's acceptance of the medium.

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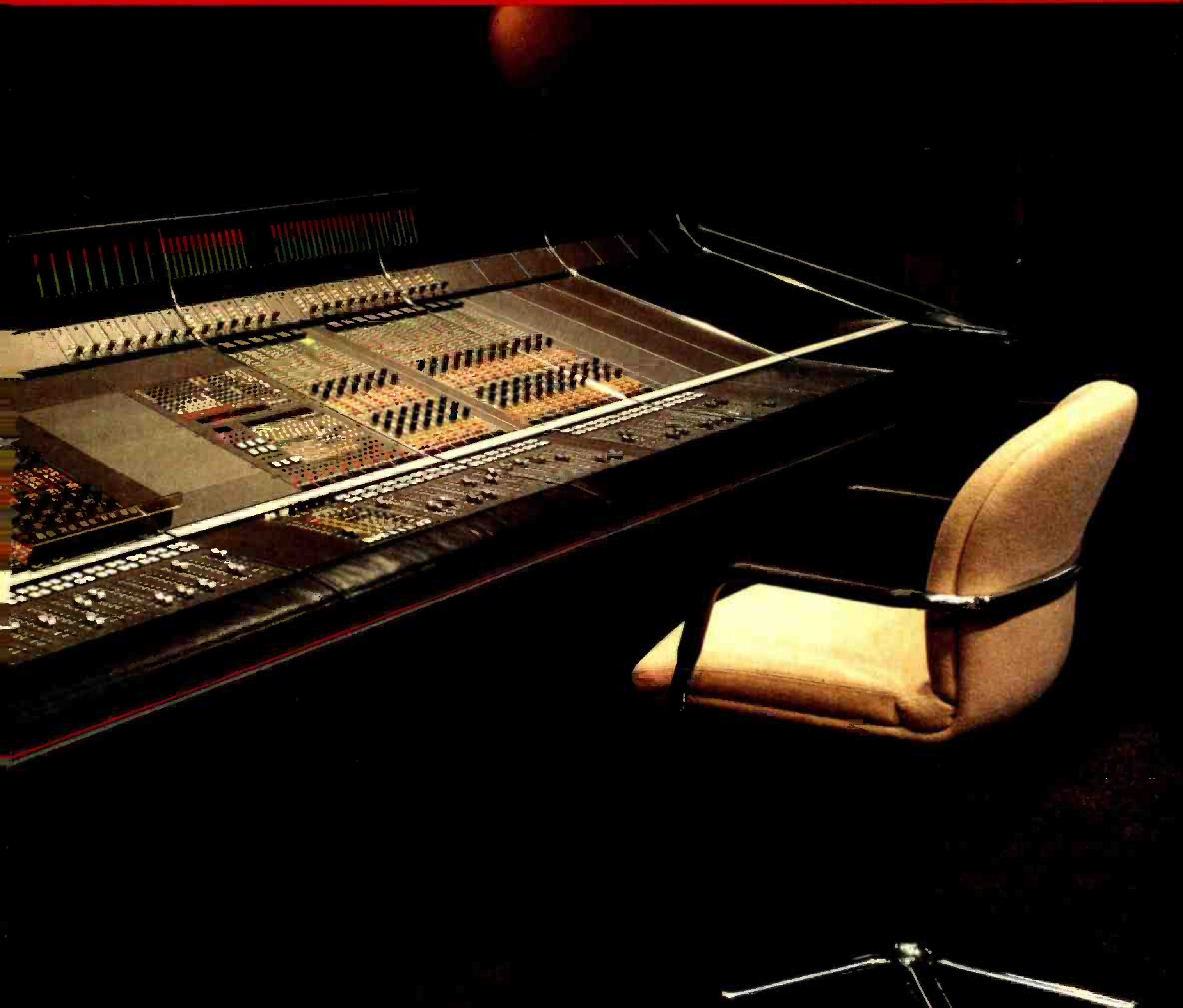


CompuSonics DSP-2002 Recording and Editing System

By Carl Kaller

While researching the availability of digital editing systems, International Recording concluded that it needed a system to provide a large sampling time and immediate recall of information, and would lessen the number of tape generations. In edit mode, we wanted rehearse capabilities and the ability to trigger special effects from time code to picture.

In short, we wanted to perform all the manipulations possible with coventional mag film—such as butt edits, sibilance removals and crossfades—but in the digital domain.

Carl Kaller is a film sound editor and engineer at International Recording, North Hollywood, CA.

The DSP-2002 random-access editing system allows us to record and play back music, dialogue and sound effects using the unit's integral hard disk drive. The sequenced audio signals can then be laid back to multitrack, videotape or 35mm mag film for subsequent audio sweeten-

The system functions basically like a 2-track tape recorder. Audio can be recorded onto the system from analog or digital sources using predetermined edit points and time code references. The digitized audio can then be replayed in sync with time code cues.

Basically, the system is the same size as a personal computer workstation, and comprises an alphanumeric keyboard with keypad, a 12-inch video-display terminal and two rack-mountable audio-control modules.

The first module houses the 16-bit A/D and D/A converters running at a 50kHz sampling frequency. A pair of XLR-type input and output connectors enable interface with conventional audio sources and tape machines. The second module incorporates a pair of RS-232 serial interfaces, a SCSI [small computer system interface] port, CSX data-reduction circuitry plus hard and floppy disk drives.

Without the use of data reduction, a standard 2002 (at a 50kHz sampling rate) stores 10 minutes of stereo or 20 minutes of mono audio on its internal 143Mbyte hard disk. By implementing CSX processing, CompuSonics claims a 1.5:1 to 3:1 data reduction, yielding 20 minutes of stereo and 40 minutes of mono sampling. [CSX data reduction is in Larry Blake's article, "Digital Sound for Motion Pictures," published in the October 1985 issue of **RE/P**—*Editor.]*

Digital audio is stored on hard disks as a soundfile with "pointers" or "flags" (edit positions). Cue and search speeds are limited only by access time of the hard disk and data-transfer rates. Although soundfiles are manipulated by means of edits, they always remain in their original form on the disk. Except by deleting or trimming, the original soundfile is never corrupted, and can be used to create multiple sound effects according to the designated edit points.

Furthermore, by using high-speed SCSI ports to transfer data to and from the hard disk, the system's 1Mbyte of internal RAM is utilized only as a buffer space, from which soundfiles are recorded and played back in real time.

Because of its ability to instantly access a specific sound, most 2002s used in this country are finding applications at film and audio-for-video post-production houses.

Our 2002 is equipped with different operational features than previously described, in that it is software selectable for either 44.1kHz or 50kHz sampling frequencies, offering a type of varispeed playback capability. We use five 143Mbyte hard disk drives (providing more than 100 minutes of stereo audio) plus a 500Mbyte backup cartridge system to hold a sound-effects library.

System software

Key to the 2002's operation is its menu-driven software: six subprograms arranged as a branch system, whereby one utility links to another. In this way, an operator can step through the software package one utility at a time.

When initialized, the first utility to appear is the directory screen, which is split to show soundfile and edit list directories. Depending on the length of a soundfile, all files are numbered sequentially from 1 to 999, or until storage space is depleted. The same criteria applies to the edit list directory.

Below the directories is a 14-cell grid map—a standard layout on all screens—that defines different functions for the numeric keypad, depending on which utility you've selected. Functions such as deleting a soundfile, switching back and forth between edit of soundfile direc-

Circle (53) on Rapid Facts Card

Circle (54) on Rapis Facts Card

tories, creating a new edit list, playing a selected soundfile and moving to other screens can be selected at a keystroke.

Hitting the *rec* key activates the Record utility, which operates only with soundfiles. From this screen you can determine the time remaining on the hard disk. This information is displayed in an 8-digit time code-style format, as are all system-status clocks, in hours, minutes, seconds and frames.

Within this utility, reassigned keys have functions for stereo or mono selection; you can record two channels simultaneously in stereo, or one mono channel at a time. When working with mono information, only the left channel is active. For mono playback, however, both channels output the same signal. Although you cannot record two mono tracks separately, or split stereo channels, this limitation has never posed a problem so far.

Currently, there are two versions of the Record utility. One version is a standard record screen that allows naming of soundfiles, input selection between stereo or mono information, plus monitoring, playback and so on. On the augmented Record screen (which doubles as a Dialogue Record screen—see below) there are functions not found on a standard display, including *vidin* and *vidout* windows for marking time code locations from videotape; save; reset for resetting video-in and -out times; and source selection from digital BNC connectors or analog XLRs.

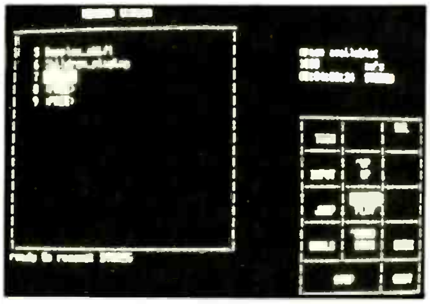
To provide direct digital connection to a Sony PCM-1610 or -1630 digital processor, seven BNCs are wired to an accompanying board within the second control module. For compatibility with a 1610/30, these BNCs are labeled as two sets of data-ins and -outs, two external syncs and word-clock.

Currently, one-software problem exists in our Record utility: the hard disk time-remaining clock doesn't count down as samples are being recorded. Instead, the clock only adjusts its time *after* a recording is complete. To work around this, I have to know ahead of time how long a sample will be, and subtract it from the time-remaining clock; otherwise, I can easily run out of disk space during a recording session.

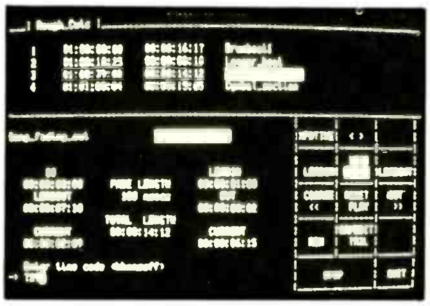
The Dialogue Record update is basically an augmented Record utility, and enables the sampling of production dialogue tracks into a 2002. Videotape is used to provide visual cues and time code locations for actors and engineers, with all recording handled by the 2002.

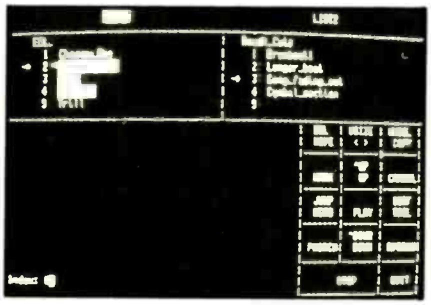
Its operation is straightforward: when an actor completes a sample line, the 2002 asks if you want to keep this digitized take or not. If you input yes, you then name the soundfile by scene number and actor. From there, the computer automatically updates the take numbers sequentially.

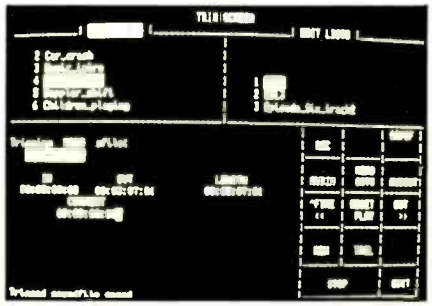
The director can then rehearse individual takes in or out of sync with the video workprint, and decide which take he wants. To obtain that take, I hit one button labeled *buy*, thus transferring it verbatim to the edit list.



Currently available branch-system software for the DSP-2002 includes: (clockwise from top left) Directory, Record, Editor, Trim, Edit List and soon-to-be implemented Crossfade utilities.







But, if the director decides that one take should have started one frame earlier, for example, I can slide it out of its original sync position. By simply changing the video-in time code by whatever offset is required, I can move the entire take in reference to picture, sliding back and forth in real-time, without corrupting the original soundfile. Working with 24-track, on the other hand, you would have to lay off the effects to a striped 2- or 4-track, or make that move in the mix while constructing dialogue tracks.

Editor screen

In the strictest sense of the word, actual edits are not performed within the Editor utility. Instead the playback sequences—better labeled as "events-decision lists"—are constructed from predetermined pointers and flags, with audio remaining unaltered in its original soundfile. In fact, the Trim utility (see below) is the only program within this branch system that performs editing.

International Recording's editor/engineer Carl Kaller (left) and computer technician David Marvitt. The DSP-2002 system is seen here with 96-input custom dubbing console, located in the facility's main dubbing theater.

There are four scroll keys used to move through an edit or soundfile, labeled as very fine, fine, very coarse and coarse. When the 2002 is powered up, the very-fine designation defaults to a 0.4-video frame increment (13.3ms), with all subsequent scroll keys increasing by a factor of 10. Very fine can be reduced, however, to as small as 0.1 video frames (3ms). Although CompuSonics could configure the cursor keys to operate at a finer rate, 3ms is all we need.

The editor utility can perform simple tape-editing techniques, such as looping. Suppose, for example, that I want to create a loop in a sequence. I first scroll to the point where a loop will begin and enter that location—referenced to the current position in a specific soundfile—as a loop-in point. Then I scroll to the end of the sequence and enter a loop-out point in the same way. After choosing the number of loops desired (up to 999), I can rehearse this sequence, play an individual sequence, or play the en-

Time Code Synchronization Applications

By David Gregg

Since last January, Transcom Media has been using two 2002 systems. One features five, 143Mbyte hard drives and the other uses nine drives. Both systems have modules housed in a centrally located machine room and connect to a cartridge backup tape system.

The 5-drive system offers 100 minutes of stereo storage. The 9-drive system, located in our master mix room, offers more than three hours of 16kHz stereo. It also provides variable 32kHz, 44.1kHz or 50kHz sampling frequencies for use during sweetening of our TV animation series.

Because the 2002 is incapable of following time code in its base form, an external reader must be connected to the system via an RS-232 serial port. To provide interface with a variety of time code readers, CompuSonics can supply custom software for a specific device to enable the 2002 to function as an event-relay slave.

Dialogue replacement is the most obvious place where time code is needed to ensure perfect sync. If I want to synchronize to video time code locations when building a specific events list, I toggle to the Editor screen and hit the vidin, key which snatches time code locations from our time code reader and enters them on the 2002's display. Or, if the sequence will be offset to a specific time code location, I work with my syncoff function (also located in the Editor utility), which basically starts an effect at an audio-in point, and synchronizes a predetermined point to a time code location in the middle of that event. For layback to videotape, the 2002 automatically outputs audio to both the analog XLR and BNC connectors.

Excluding butt-splice events, we currently run about 50 events per list, with three lists per 20-minute episode. As long as the time code points don't overlap, we can perform 150 to 200 consecutive events. If the events do overlap, the computer sends an error message because, as yet, the 2002 cannot perform any multitasking functions. After 50 consecutive butt-splice events, however, it takes the computer only six seconds to clear its buffer and load the next sequences.

Where we do want overlaps, we use a second events list, thereby offsetting two effects. We can't play them back simultaneously from the 2002, but can build those offsets separately onto 24-track. For precise playback onto videotape, our time code reader can read valid code at 1/50th real-time playback speed from a VCR. Using the jog wheel on a U-matic VCR, and the 2002's capability of laying effects one frame from each other, we obtain highly accurate audio layback capabilities.

David Gregg is vice-president of post-production at Transcom Media, New York.

tire events list from top to bottom.

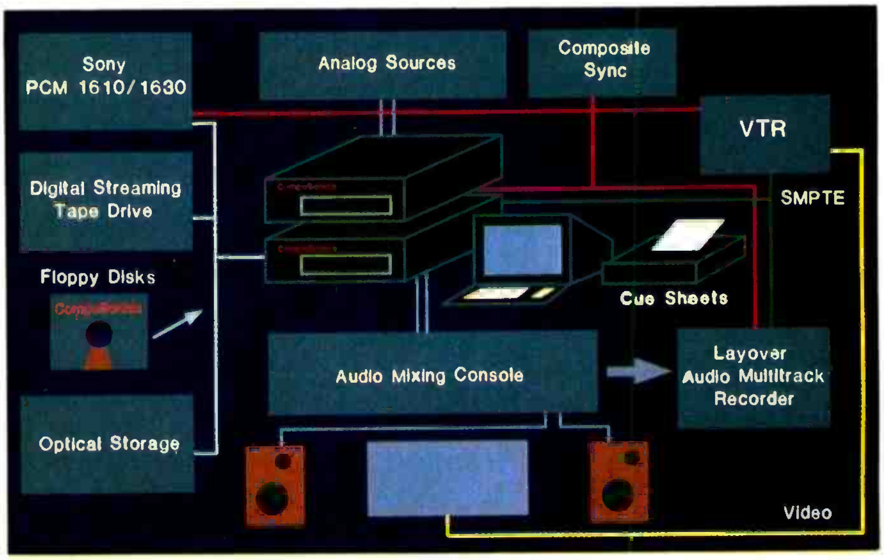
In terms of the music editing, the looping functions are really tight—a continuous 4-frame loop can be performed seamlessly. However, if an event becomes too confusing, and I need to start again from scratch, a *reset* key resets loop-in and -out, plus audio-in and -out locations.

Sometime soon, I would like to see more time code features implemented in the Editor utility. We will soon do all our work referenced to time code and, unfortunately, only one of the utilities, the Dialogue Record screen, is supplied with a video-out time. It would be better if we had a display for time code-in and -out, and the duration of the take. Split stereo editing, I feel, would also be an advantage.

To perform a permanent soundfile edit, I must use the Trim utility. Basically, this screen trims unwanted information from a soundfile, deleting pops, clicks or entire passages. By using audio-in and -out keys, I can hard-edit together the remaining information as a butt splice.

The final screens, Edit List, List Editing and Crossfade, are utilities that display events lists stored on floppy and hard disk. Within these utilities, as with other screens, events and soundfiles can be rehearsed both on- and off-line, then synchronized and laid to tape. Sequences can also be moved between two preselected events lists, with each list being appended as desired.

According to CompuSonics, a Crossfade utility still under development will allow crossfading from one audio event to another. Variables will include lead-in and -out times, fade length (in milliseconds), current times for events and total length for a newly created crossfade. The utility will also provide true overlapping crossfades, using two split-mono or -stereo outputs.



Block diagram of DSP-2002 being utilized in a video post-production environment.

Technical Specifications

Record and replay format: 2-channel stereo, or dual mono.
Analog inputs: two XLR-type, +4dBm line level.
Analog outputs: two XLR-type, +4dBm line level.
Sampling rate: 50kHz (with other options available, including 16kHz, 32kHz and 44.1kHz).
Signal processor: 16-bit PCM A/D and D/A converters.
Signal-to-noise ratio: greater than 92dB.
Dynamic Range: greater than 92dB.
Frequency range: 20Hz to 20kHz.
Wow and flutter: "none."
Third-harmonic distortion (1kHz, 10V peak-to-peak): less than 0.01%.
Channel separation: 92dB.
Internal RAM: 1Mbyte of random-access memory.
Floppy-disk drive: storage capacity of 3.3Mbytes.
Hard-disk drive: storage capacity of 143Mbytes.
Telecommunications interface: AT&T Accunet modem (with optional 56kbaud transmission rate).
Manufacturer: CompuSonics Corporation.

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Final comments

One thing I like about the system is the speed with which I can obtain a soundfile. When the system is initialized, the entry at the top of the soundfile directory is cued and ready for playback. Because less studio time is required to access sound effects, we can save money on an average session by using the random-access capabilities.

For example, on an average-length film, a production company might use up to 10 sound-effects editors, taking up to a month to complete the sound editing process. At International we are hoping that, in terms of transfer time and sound-effects building and logging to multitrack, the 2002 will cut that job down to two weeks for one person.

The system also works well in certain situations where you have established or recurring sound effects. All those key effects can be loaded into the 2002, readily accessible with a push of a button.

On the other side of the coin, I would like to see a hardware mod or upgrade

Transcom engineer Danny Mundhank (left) and CompuSonic software engineer Peter Roos operating the facility's 2002 in conjunction with a 36-input Solid State Logic SL6000E console.

that would allow shuttling control over the audio—maybe like the jog wheel fitted to a U-matic VCR to provide rock-and-roll video cuing—instead of keyboard scrolling controls. With the cursor keys, sometimes you would hold the key down too long, and a buffer repeats the scroll command; it can go right past your event point. In that respect, the hands-on feel of a conventional tape deck is better for editing and cuing.

When we bought the system, we knew that the 2002 had still not fully matured, but we were willing to work with it. We just wanted to make sure that we had one of these devices installed in our facility. And now, we have already outgrown our backup tape system, and are currently awaiting delivery of a 12-inch WORM (write-once/read-many) optical-disc system capable of storing 2Gbytes of data. At a 44.1kHz sampling rate, the optical drive will enable on-line storage of more than 6 hours of stereo 16-bit audio. R·E/P

Studio update

Northeast

Magnetic Sound Studio (New York), a 9,000-square foot facility, has been redesigned and acoustically upgraded. The 4-room complex offers audio-for-video mixing, audio sweetening, transfers and audio relays, as well as film mixing and re-recording services. Beverly Dichter, whose family founded Photo Mag two decades ago, is director of operations.

According to Walter Tennenbaum, studio director, "With the advent of the music-video revolution, stereo TV, SAP, digital audio and other important audio advances, we recognized that quality audio is no longer a luxury to producers but a necessity."

Recently installed hardware includes **Neve V-series** post-production consoles with NECAM 96 automation. The customized boards, which will be used for audio-for-video and film post-production, provide 8- and 4-track monitoring, with stereo music, dialogue and effects plus laugh track or SAP. Also installed are **Adam-Smith series 2600** time code synchronizers. The studio is equipped with adjoining narration booths, **Otari MTR-90** MkII 24-tracks, BVH-2000 U-matics and B&W model 808 speakers. "The 808s have a range of the output SPL from a symphony orchestra to the sound of a pin dropping," Rotta says. *222 E. 44 St., New York, NY 10017; 212-687-9030.*

Aura Sonic Ltd. (Flushing, NY) has updated its ASL Mobile Audio/Video truck with a **Harrison MR-4** automated console, an **Otari MTR-90** 24-track and **MTR-10** 2-track, **Nakamichi DMP-100** digital processors with two **Sony SL-HF-900** Super Beta videocassette machines and a pair of **UREI 813** monitors. *P.O. Box 791, Flushing, NY 11352; 718-886-6500.*

West 55th Street Studio (New York), has added an **Otari MTR-90** 24-track, a **Studer** 2-track, a **Sony PCM-701** digital 2-track, a **Harrison** console and **Westlake** studio monitors. Effects units now include **AMS RMX-16** reverb and two **DMX-15-80** digital delays, a **Lexicon PCM 70** and two **PCM-24s**. The studio's synths include the **E-mu systems Emulator II+, Akai S9000** sampler, **Yamaha DX-7, Ensoniq Mirage** rack and **Roland's Super Jupiter** and **Jupiter 8.** *240 W. 55th St., New York, NY 10019; 212-757-7185.*

Greene Street Recording (New York) is the first studio in the United States to receive the **AMEK APC-1000** assignable console. The console includes 80 inputs and features full recall facilities, synchronous reset and dynamic reset systems and the **GML Moving Fader Systems**. According to Steve Loeb, owner of the studio, "With our purchase of the George Massenburg-automated **AMEK APC-1000**, we're on the cutting edge.

"We're excited to be the first in the United States to offer the recording industry an opportunity to work on this extraordinary console." *112 Greene St., New York, NY 10012; 212-226-4278.*

Studio update

Unique Recording Studio: Studio B.

Unique Recording (New York) recently upgraded with the installation of four **Studer A800** 24-tracks. Co-owner **Bobby Nathan** anticipates purchase of two more A800 24-tracks to replace existing units in the facility's "MIDI City" pre-production room.

The Studer machines are locked together for 48-track operation using **Adams-Smith** time code synchronizers. *701 7th Avenue, 8th and 10th floors, New York, NY 10036; 212-921-1711.*

Southeast

Strawberry Jamm and Higher Skys Studios (West Columbia, SC) have merged to become **Strawberry Skys,** a 24-track, fully automated and computer-assisted facility. The new complex will have the ability to interface with any MIDI demand and will be equipped to handle all basic video capability, according to Bob Curlee of Strawberry Jamm and Gary Bolton of Higher Skys.

"The big feature we offer the national client is a 24-track, automated recording service for about one-third the nationally advertised rates," Curlee says. *1706 Platt Springs Road, West Columbia, SC; 803-794-9300.*

Reflection Sound Studios (Charlotte, NC) has added a **Sony APR-5000** 2-track, a **Lexicon PCM-70** effects processor and two **Valley People 440 limiters** to the equipment in Studio A, all rewired with **Monster Cable series One** and **Mogami cable.** *1018 Central Ave., Charlotte, NC 28204; 704-377-4596.*

Southern California

Harmony Gold (Los Angeles) has purchased **Intersound,** an audio and video post-production facility located in both Los Angeles and Rome, from Lorimar-Telepictures. The facility includes ADR rooms, Foley stages and digital Telecine equipment music and effect editing services. A new on-line video editing room

Studio update

will contain five 1-inch machines, a **CMX** controller and a **Quanta** character generator. A new off-line editing room includes a **Chyron** character generator and a **Sony** special effects unit. *8746 Sunset Blvd., Los Angeles, CA 90069; 213-652-3741.*

Fred Jones Recording Services (Hollywood) has added a **Sequential Prophet 2000** digital sampling keyboard to its music and effects room. The 5-room media facility has also acquired a **Linn 9000** sampling drum machine and sequencer. To enhance the **Cipher Digital** and **Editron** synchronization, the studio offers 35mm mag transfers. *6565 Sunset Blvd., Suite 211, Hollywood, CA 90028-8521; 213-467-4122.*

Holywood Studio (Los Angeles), a private studio owned by Bruce Sudano and Donna Summer, features a **Yamaha DX-7** and **Rhodes Chroma** synthesizers. Also included in the room are a **Linn LM-1** and sequencer. Recording equipment includes an **Amek TAC Matchless** 26-channel board. **Sony** JH-24-track and 2-track tape machines; **Lexicon 224** digital reverb; **UREI LA-2A** limiter; 4-band **Neve** EQ modules, a **Studer** cassette and assorted **dbx** limiters and compressors. *Lakeside Associates: 4 Alegria, Irvine, CA 92720; 714-730-1333.*

Northern California

Aapex Tape Duplication (Santa Rosa) has installed more than 100 **Nakamichi MR-2** cassette decks for real-time tape duplications.

Warren Dennis, Aapex general manager, says that the MR2x were chosen because "I've been frustrated by duplications that compromise audio quality. Now my reproductions sound as good as the original." *350 E. Todd Road, Suite A, Santa Rosa, CA 95407; 707-585-1132.*

Different Fur Recording (San Francisco) has announced its opening as the first Solid State Logic studio in the San Francisco/Northern California region. Hardware will include the **Solid State Logic 4056 E-series** Master Studio System with Total Recall and **Sony PCM-3324** DASH-format digital multitrack. A **Studer A80** analog multitrack will also be featured. *3470 19th St., San Francisco, CA 94110; 415-864-1967.*

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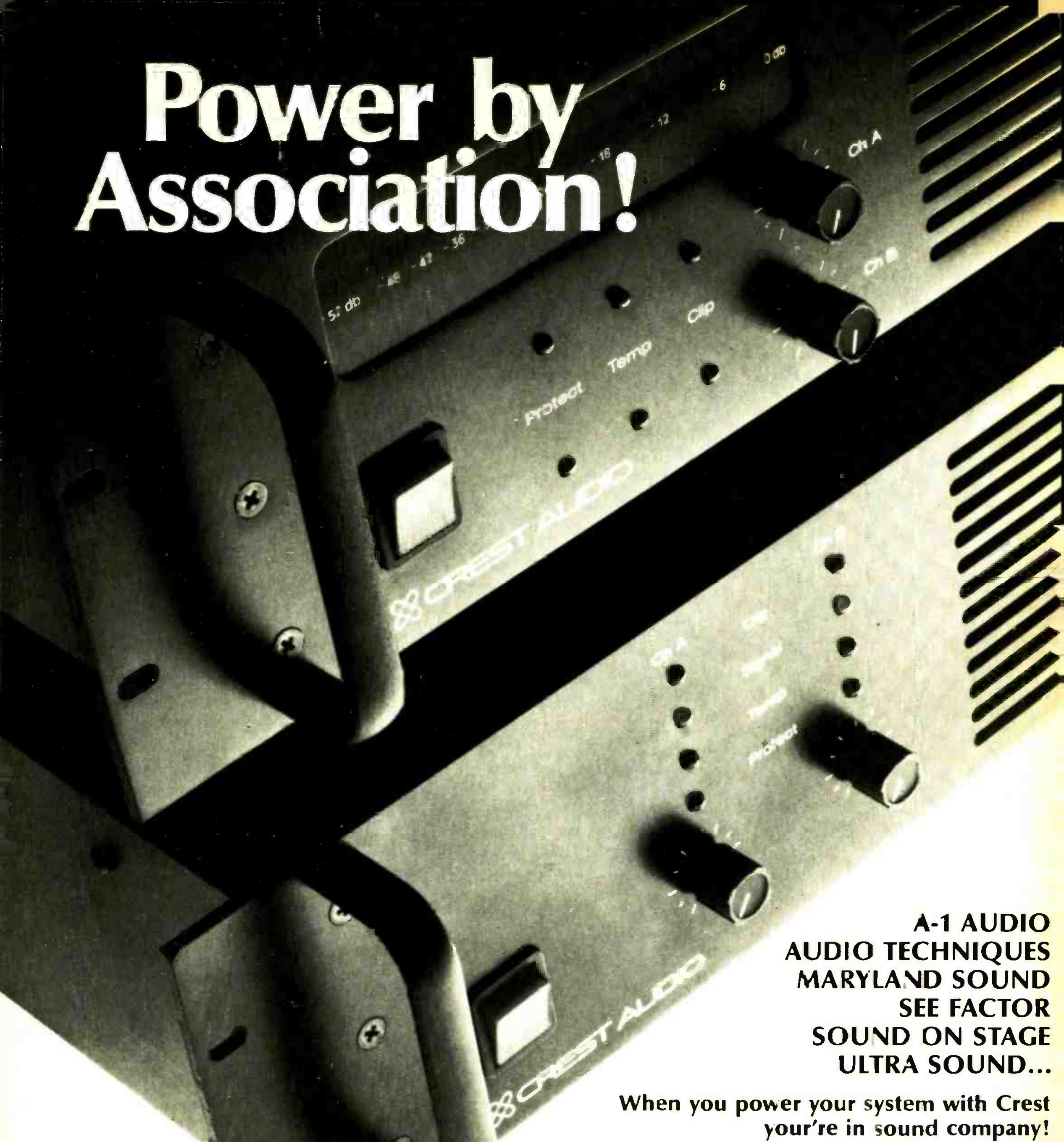

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New products

Wendel JNR Labs digital sound replacement unit

The device can be used to automatically replace prerecorded, live or synthesized percussion sound with digital samples. Sounds contained on ROM cartridges are inserted into one of two slots on Wendel Jr. and then triggered by audio signal commands.

Roger Nichols, company president, has transferred 50 of his personally recorded percussion sounds onto Wendel Carts and plans to expand the library as customers require. The company will also offer a special service to provide custom carts using an artist's prerecorded drum sound.

Currently under development is a unit that will feature real time dynamic tracking and, when used with two Wendel Jr. units, will faithfully reproduce virtually any drum sound identically, the company claims. Also on the drawing board is a PC-based digital audio sampling device that will allow users to create their own sound samples on floppy disk.

Circle (106) on Rapid Facts Card

Cerwin Vega CS70 loudspeaker system

Using four, 12-inch low-frequency drivers, a pair of midrange drivers with 2-inch throats, and a focused array high-frequency section, the new 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 (107) on Rapid Facts Card

APL dome coaxial control room monitor

APL introduces a 3-way loudspeaker monitor design that incorporates coaxially mounted cone and dome direct radiator drivers.

The monitor combines 15-inch woofers, a 10-inch midrange cone driver, and a coaxially mounted 1-inch soft dome tweeter in a 3-way active system. The system's electronic crossover employs 18dB/octave, phase accurate filter characteristics.

Circle (108) on Rapid Facts Card

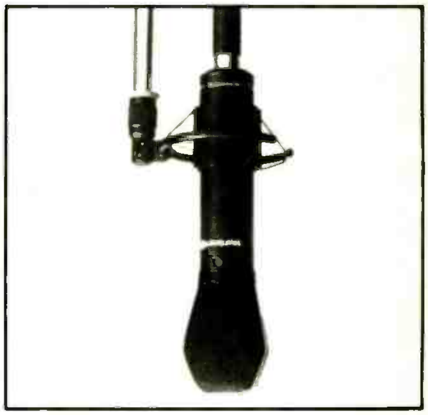
New products

Beyer MC740 condenser mic

The double-membrane microphone is switch-selectable between five polar patterns: omnidirectional, wide cardioid, cardioid, hypercardioid and figure-eight. Patterns can also be selected via an optional power supply.

With its 10dB attenuation pad switched in, the MC740's elastic suspension system is said to enable it to withstand SPLs of up to 144dB without distortion. A selectable highpass filter operates at 80Hz or 160Hz.

Circle (109) on Rapid Facts Card



Lexicon introduces 480L digital effects system

The stereo unit uses the same tool designed for both current and LARC remote console as the 224XL digital reverb, and contains seven banks of programs, each with user-variable parameters. It will produce reverb sounds and time delay-based effects, as well as sampling. Setups and sounds can be stored, transported and repeated with non-volatile memory cartridges.

The 480L's digital I/O interface is compatible with Sony PCM-1610/1630 processors, allowing the unit to be integrated into digital mastering and production systems. The 480L's architecture will also support interfaces with the emerging generation of random-access digital audio recording/editing and storage devices, according to Lexicon.

The first software update will feature Dynamic MIDI, which allows any effects or reverb parameter to be assigned to any variable MIDI controller. Up to 20 parameters can be simultaneously controlled from sequencers, drum machines, pitch or mod wheels, keyboard velocity or aftertouch and any other MIDI controller. MIDI program change and system exclusive send/receive will also be supported.

Current software includes banks of halls, rooms, wild spaces, plates and effects as well as a sampling bank. Reverb programs contain 24 parameters, including two called Shape and Spread. An input delay of up to 254ms can be applied to the dry signal, producing sounds in which the effect precedes the source signal.

Circle (110) on Rapid Facts Card

Hardy M-1 Mic Preamp

Featuring the 990 discrete op-amp, Jensen JE-16-B mic input transformer, and DC servo circuitry, the M-1 is available with one to four mic preamp cards installed. Each card has a dual range gain control and range switch for exact gain settings, a Phantom supply on/off switch, and a phase reverse switch. Other features include XLR inputs and outputs, toroidal power transformer, and internally selectable mains voltage.

Circle (111) on Rapid Facts Card

New products

800 series audio mastering tapes from 3M

The new generation of audio mastering tapes is designed for location recording, dialogue and jingle applications. Developed using the same family of oxides as 226 mastering tape, the 806, 807, 808 and 809 will replace 3M's existing Scotch 206, 207, 208 and 209 products.

The 806 formulation, according to 3M, is designed to eliminate the traditional trade-off between performance and print specifications. As a result, the company explains, 806 provides flexibility in applications requiring a balance between signal-to-noise and signal-to-print ratios. It incorporates a 1½-mil base and combines the high output capability (+8dB) of Scotch 206, with improved print-through characteristics that reach the level of Scotch 208.

Circle (112) on Rapid Facts Card

JBL Control 1 monitor loudspeaker

Designed for a studio or facility requiring a small-sized monitor with a forward sound character, the monitor's response is described as smooth and extended but with a broad rise in the upper midrange, making its sound character similar to the model 4311.

Because of its size, the Control 1 can be mounted on top of a console. The unit's molded enclosure will accommodate an optional mounting adaptor for permanent attachment to any rigid surface; a clamp system is also available for semi-permanent mounting.

The speaker can be used in close proximity to a video monitor because the high- and low-frequency transducers are magnetically shielded.

The 2-way, 4Ω system incorporates a ¾-inch tweeter, a 5-inch woofer and a dividing network.

Circle (113) on Rapid Facts Card

New products

Monster Cable M series

Consisting of the M1 speaker and M1000 interconnect cables, the M series MicroFiber dielectric technology and winding techniques are said to allow signals to travel faster and cleaner, eliminate background noise and increase transient response.

In addition to custom lengths, M1 speaker cable will be made available in 15- and 25-foot pairs terminated with Monster Cable's X-Terminators.

Comprised of a 3-wire multiple-gauge network for each conductor, M1's bandwidth-balanced design is claimed to provide coherency of frequency and phase response over the entire musical spectrum.

Available in a variety of precut lengths, M1000 will be terminated with a new RCA gold-plated connector and locking outer ring for better contact and pull-proof reliability.

Circle (114) on Rapid Facts Card

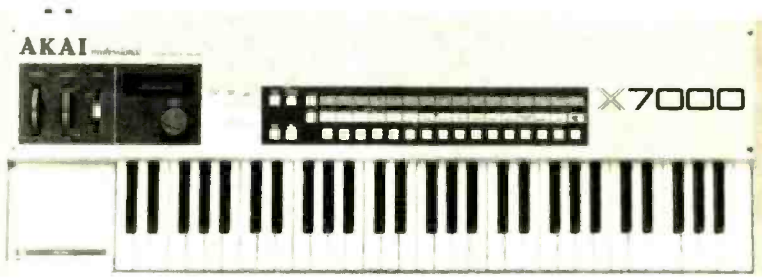
Akai X7000 digital sampling keyboard

Utilizing technology similar to the S900 Super Sampler, the X7000 also features 12-bit linear technology, an adjustable sample frequency from 4kHz to 40kHz, and six individual sample points, each assignable to different key groups and MIDI channels.

Editing features include 1-shot looping, alternating LFO pitch modulation with adjustable depth and delay and adjustable release rate.

The 61-key velocity-sensitive keyboard can be set up for more than six octaves of true voice range and a built-in 2.8-inch diskdrive allows compatability with all of S612 format disks, giving access to a library of digitally sampled sounds.

Circle (115) on Rapid Facts Card



RCF SCD6000 studio monitor from EAW

Distributed in the United States through Eastern Acoustic Works, the RCF model SCD6000 close-field studio monitor incorporates a single magnet structure that contains both a 2-inch midrange and a 1-inch high-frequency, soft-dome driver. This configuration is said to enable both drivers to be located within 2½-inches center to center, for optimal combining at crossover.

The use of special high-temperature magnetic damping fluid in both dome drivers provides higher damping, thereby reducing distortion and eliminating high frequency cavity resonances typical of small diameter drivers, the company claims.

The low-frequency comprises a 12-inch driver incorporating a polylaminated cone and butyl rubber surround. The 3-inch voice coil and magnetic gap have been optimized for maximum linear excursion, while the use of a large magnetic circuit provides high efficiency. The system is rated at 91dB SPL 1W/1m sensitivity.

The front baffle features a minimum-diffraction design that can be rotated for ideal orientation for both horizontal and vertical mounting.

Technical specifications include a quoted frequency response of 50Hz to 20kHz, ±2.5dB, and an off-axis response of 45Hz to 15kHz ±3dB, over a 120° angle

Circle (116) on Rapid Facts Card

New products

Biamp introduces rack-mount mixer

The RackMax unit is a 16-input stereo mixer with performance specifications and features including a 48V phantom power switch on each channel, 100mm faders, a complete solo system, three jumperable auxiliary sends per channel and LED metering in a standard rack width. The unit is 3-inches deep.

Circle (117) on Rapid Facts Card

Aries audio mixing consoles

The Aries line features two models, a 16/8/16 and a 24/8/16. Each console includes: 3-band input equalizer with sweepable midrange, four aux sends switchable pre/post, eight bus assignments, mute, and PFL and peak overload indicators. Also included are 16 monitor/FX returns, eight with 2-band equalization.

Circle (118) on Rapid Facts Card

Denecke introduces DCODE TS-1 time code slate

Featuring a high intensity 1-inch LED readout, the unit displays time code, user bits and drop frame status with switchable high and low brightness levels.

A clap-activated function displays the time code when it is in the open position and freezes for three frames when closed. The display then switches to user bits for one second before turning off. The lightweight TS-1 offers single-hand operation and an optional wireless receiver.

Circle (119) on Rapid Facts Card

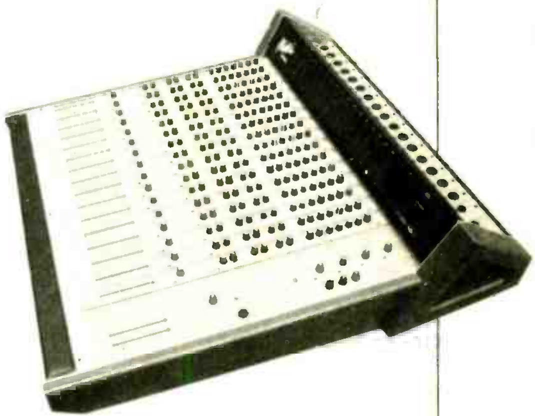
Soundtracs MIDI series consoles

Designed primarily for keyboard applications, the console series is of an in-line format with either a 16- or 24-input/output module mainframe incorporating 16-track monitoring and 16 sub groups.

Also featured are 32 or 48 MIDI-controlled inputs with additional MIDI control of four auxiliary sends and eight optional MIDI-controlled effects returns.

A built-in microprocessor enables programming of console status, which may be designated to any of the 16 MIDI channels currently available.

Circle (120) on Rapid Facts Card



New products

Barcus-Berry BBE 802 processor

Designed for use in recording studios, live concerts, TV and radio broadcasting and other commercial sound applications, the BBE 802 is a multiband, program controlled signal processor.

The unit uses high-speed dynamic gain-control circuitry to audibly improve the reproduction of program transients, adding brightness and presence without introducing stridency.

In the bypass mode, swept frequency response of the system is essentially flat from 20Hz to 20kHz and in the process mode, high-frequency response is controlled by program. Amplitude changes are developed only in direct response to application of a spectrally diverse program signal.

Also available for smaller studios and club sound systems, the BBE 402 maxie a multi-band program controlled processor, can add brightness and presence without introducing the underside stridency, which so often is characteristic of equalized sound, especially at peak levels, the company says. The unit increases voice intelligibility by eliminating frequency band masking when important sibilant and consonant elements are represented in the program signal.

Circle (121) on Rapid Facts Card

Otari SMPTE/EBU time code reader

Designed to be used as an accessory for an audio or video tape recorder, the EC-201 time code reader features 1/20 to 60X play-speed reading, 40 hour continuous use on battery power and reshaping circuitry on the loop output.

This unit also offers a full hexidecimal user bits display with a hold button for edit logging; a –10dBV to +1dBV input range; balanced XLR inputs/outputs, and includes an ac adapter, belt clip and batteries.

Circle (122) on Rapid Facts Card

New products

Digital synthesis software from digidesign

Using software-based synthesis, sounds can be designed with Softsynth's graphic programming screens and then synthesized by the Macintosh and transferred to a digital sampling keyboard for playback.

Each of the program's 32 oscillators/harmonics has a 40-stage amplitude envelope, 15-stage pitch envelope, selectable wave type and variable frequency. The time slice mode provides a single master envelope for an entire sound, and up to 40 different timbre events can be positioned in the sound.

After the sound has been designed, Softsynth generates a 16-bit digital sample that can be transferred to any Softsynth compatible sampler using a Macintosh MIDI interface.

Circle (123) on Rapid Facts Card

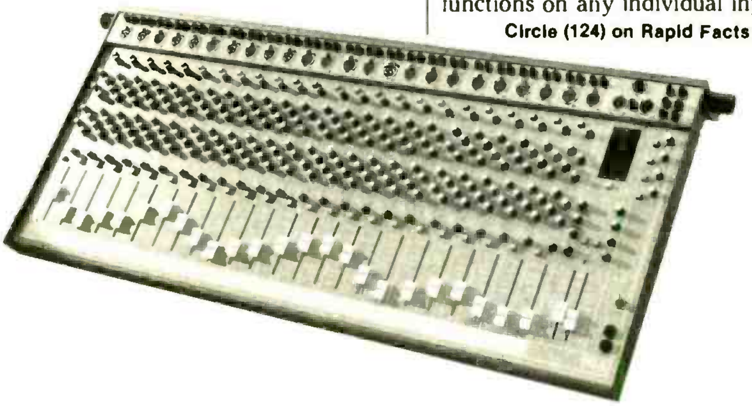
Connectronics Seck 242 stereo console

The 24-input mixer features a mic/line input switch on each channel that selects low or high impedance. Both inputs are electronically balanced and together cover a range from –55dBm to +10dBm with 25dBm overload margin.

A 3-band equalization is featured, with sweep midrange of 330Hz to 6.5kHz. The high EQ is set at 11kHz and the low at 45kHz.

Each channel has two pre-fade auxiliary sends for monitoring, foldback or special effects and two post-fade sends for feeding effects or for specialized house P.A. or recording. In addition, each channel has a pre-EQ insert point to cater for outboard limiting and/or delay functions on any individual input.

Circle (124) on Rapid Facts Card



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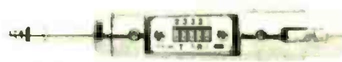

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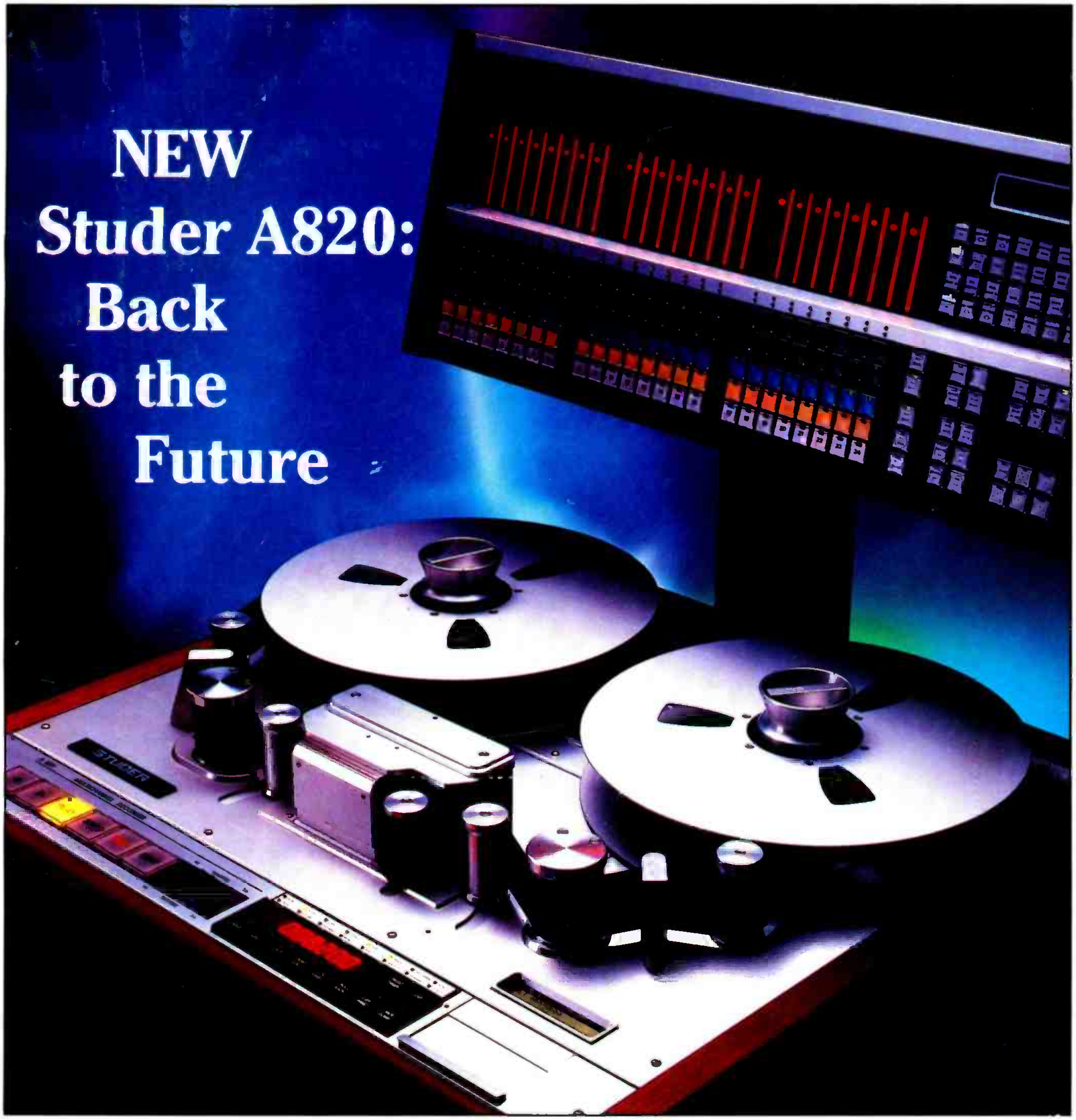

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