

May 1987

Surround Sound
Super Bowl XXI
Page 64

Recording ENGINEER/PRODUCER

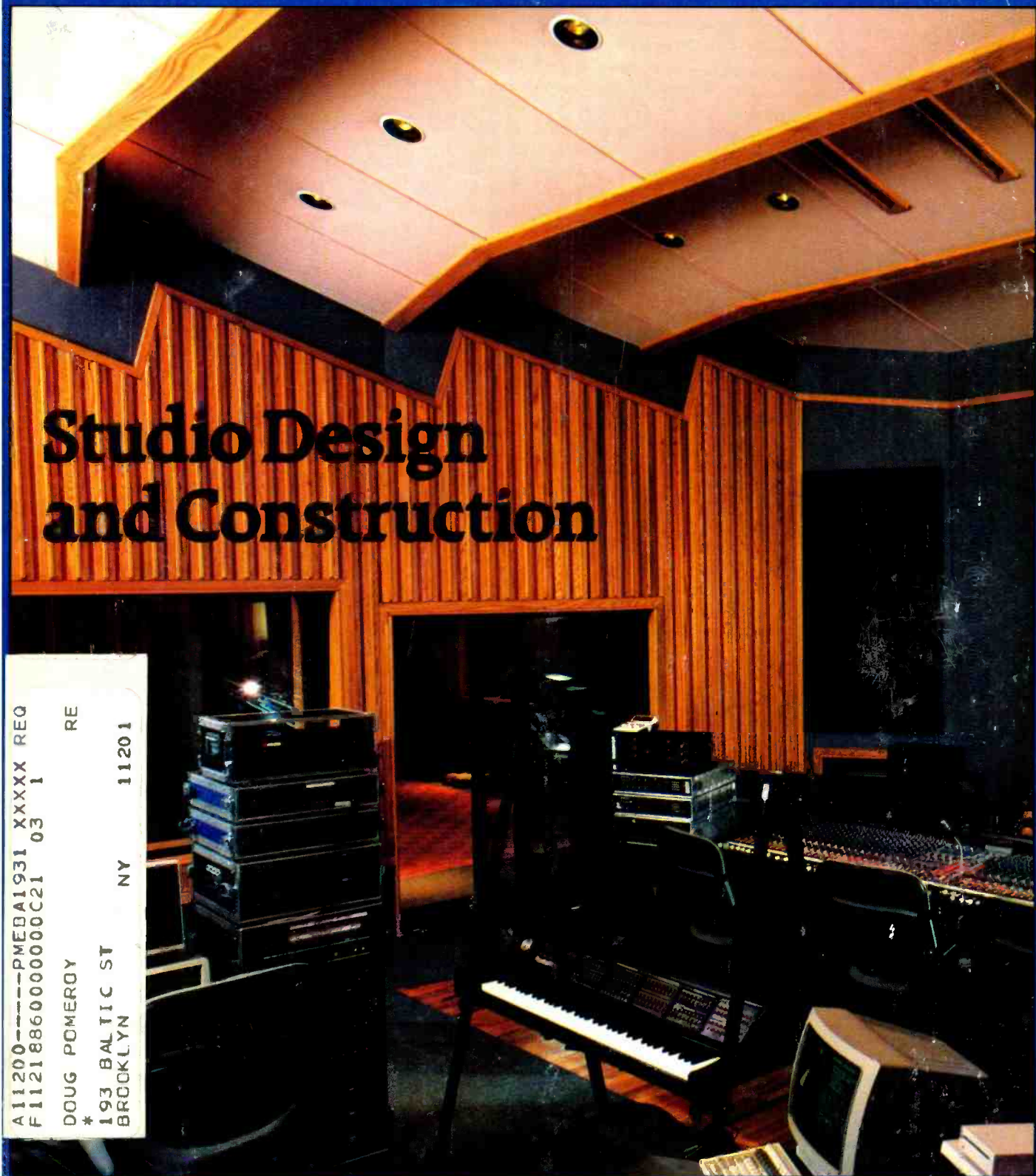
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Studio Design and Construction

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Why Leading from Tape

The Synclavier Tapeless Studio™ is available today!

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

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

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

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



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 No longer are you limited to storing and retrieving digital data on media designed for outdated tape technology.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

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

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"The Synclavier,[®] combined with the new Direct-to-Disk™ Multi-Track Recording System, provides us with the most compact, reliable, upgradeable, and high fidelity recording environment available today. For video-post, Foley, or music recording, it's a product which offers us tremendous benefits, both sonically and financially."

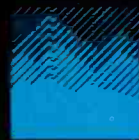
Murray Allen, President, Universal Recording Corporation

Using today's advanced computer technology, the Synclavier Tapeless Studio now offers more than just the ability to synthesize and create music. Now you can record "live audio"

simultaneously onto as many as 16 separate tracks. Dialogue, effects, vocals, and/or music tracks can be SMPTE synchronized and edited with word processing-like control at a single workstation.

The fidelity, speed, and flexibility of this system make the Synclavier Direct-to-Disk Multi-Track Recording System truly the most powerful digital audio system available today.

For a complete information package, including an audio cassette demonstrating the Synclavier and the Direct-to-Disk System, send \$5.00 to New England Digital Corporation, Box 546, White River Junction, Vermont 05001.



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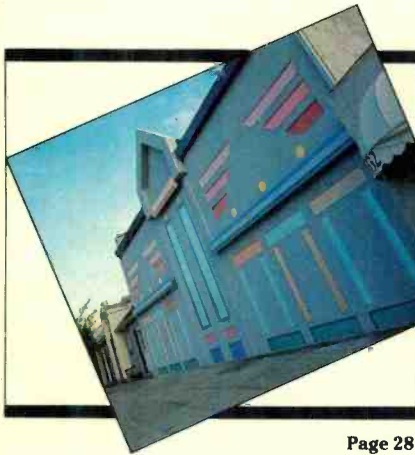
Contents

Recording ENGINEER/PRODUCER

May 1987

Volume 18

Number 5



Page 28

Studio Design and Construction

Given the ever changing requirements of recording and production facilities, RE/P examines the various aspects of designing, constructing and equipping a new or upgraded studio.

Acoustic Design and Monitoring Requirements for Recording and Production Facilities

Studio operators contemplating the construction of a new facility should be aware of the basic ground rules of acoustic design and monitoring systems.

By Chips Davis 22

Facility Spotlight:

Craig Huxley's The Enterprise Studio

A look at the primary decisions involved in the design, construction and equipping of a 3-room multi-function complex owned by leading musician/composer Craig Huxley.

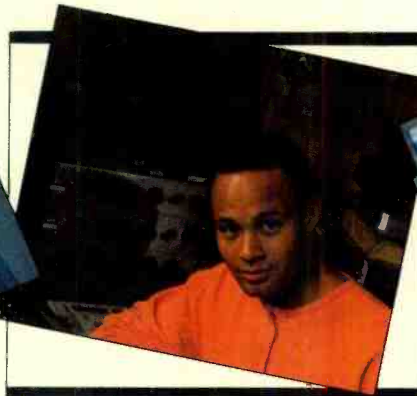
By Adrian Zarin 28

Designing Customized Facilities:

The Acoustician's Role

The use of an accredited acoustician, experienced in the analysis of sound isolation, noise control and acoustic treatment can save problems during the design and construction of a facility.

By John Eargle 40



Page 56

Electrical and Interface Systems for Recording and Production Studios

Noise and interference problems can easily be overcome by paying careful attention to the electrical wiring and equipment interconnect schemes.

By Larry Winter and Terry Pennington 46

Other Features

Property and Liability Insurance for Recording Studios

Insuring a recording or production studio and its operations involves choosing the right insurance agent/broker, and learning to manage risk.

By Robert J. Kribs 52

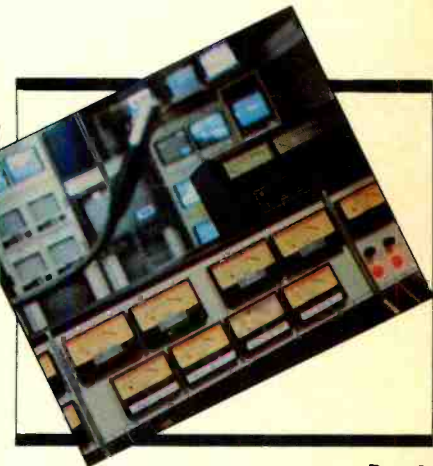
Production Viewpoint: Narada Michael Walden

This musician/producer shares some interesting recording and production techniques while working with Aretha Franklin and Whitney Houston, and details his new personal-use studio, Tarpan.

By Adrian Zarin 56

Surround Sound Production for Super Bowl XXI

The recent live television broadcast coverage of the Super Bowl XXI game with encoded surround sound information involved the



Page 64

use of some innovative preproduction and mixing techniques.

By Larry Blake 64

Hands On:

Akai MPX-820 MIDI-based 8-channel mixer.

By Paul D. Lehrman 72

Departments

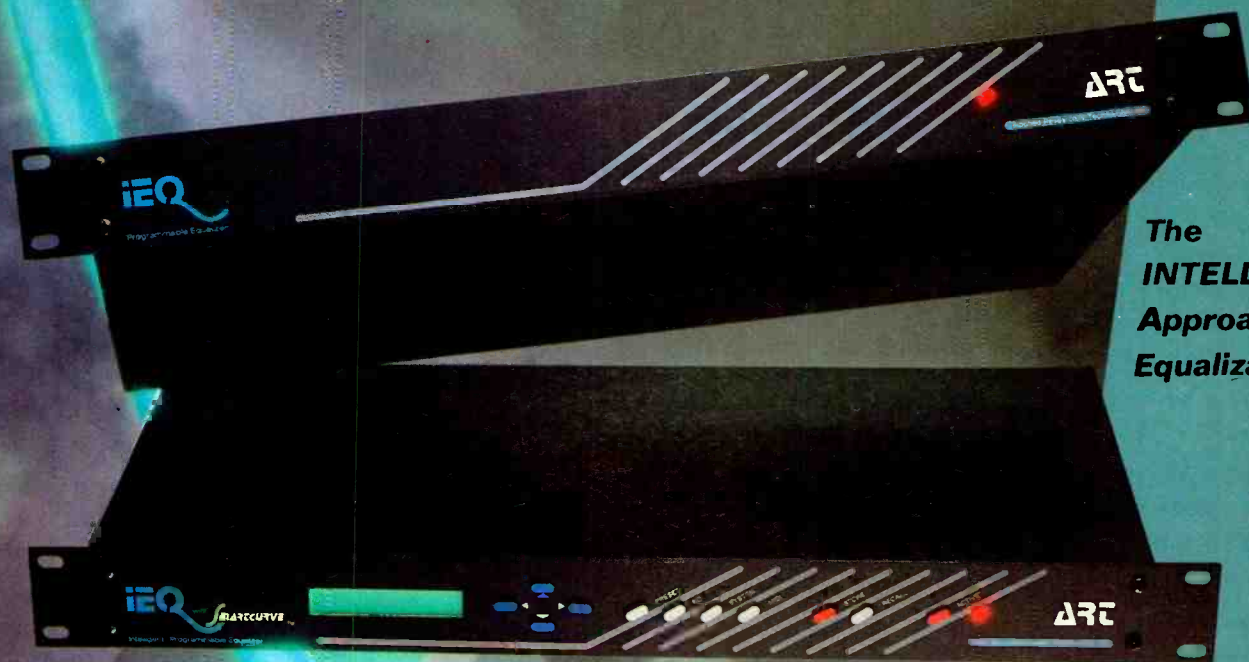
Editorial	4
News	6-8
Letters	10
Managing MIDI	12
Sound on the Road	14
Film Sound Today	16
Living with Technology	18
SPARS On-Line	20
Studio Update	76
New Products	80
Final Stage	87
Classified	86-87
Advertiser's Index	88

On the Cover:

Studio A at The Enterprise, Burbank, CA, owned by musician, producer and composer Craig Huxley, is based on an original design by acoustician Jeff Cooper. Photo by Michael LeRoy.

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News

Lyra Awards for Academy Award nominees

The five audio teams recently nominated for an Academy Award in the "Achievement in Sound" category were honored in late March with Lyra Awards from 3M.

Those receiving the Lyra Awards include: Roy Charman (production sound mixer), Graham Hartstone (chief dubbing dialogue mixer), Nicholas Le Messurier (music dubbing mixer) and Michael Carter (SFX dubbing mixer) for *Aliens*; William Nelson III (production sound mixer), Lester H. Fresholtz (supervisor, re-recording mixers), Vernon Poore (re-recording music mixer) and Richard L. Alexander (re-recording SFX mixer) for *Heartbreak Ridge*; Simon Kaye (production sound mixer), John K. Wilkson (supervisor, re-recording mixers), Charles Grenzbach (re-recording music mixer) and Richard Rogers (re-recording SFX mixer) for *Platoon*; Gene S. Cantamessa (production sound mixer), Terryl B. Porter (supervisor, re-recording

mixers), Mel M. Metcalfe Jr. (re-recording music mixer) and David Hudson (re-recording SFX mixer) for *Star Trek IV: The Voyage Home*; and William B. Kaplan (production sound mixer), Don Mitchell (supervisor, re-recording mixers), Richard C. Kline (re-recording music mixer) and Kevin O'Connell (re-recording SFX mixer) for *Top Gun*.

In addition, the original music scoring mixers receiving awards include Eric Tomlinson for *Aliens*, Bobby Fernandez for *Heartbreak Ridge*, Roger Monk for *Platoon*, Danny Wallin for *Star Trek IV: The Voyage Home* and Brian Reeves for *Top Gun*.

The list of industry veterans recognized as "pioneers in sound" includes Dick Olson, Fred Hynes, Leon Leon, Joe Wissman, Jim Stewart, Fred Wilson, Bill Edmandson, Joe Kelly, Al Green, Frank Pontiuies, John Frayne and Pete Vlahos.

WaveFrame formed to develop digital systems

The new company, based in Boulder,

CO, will be involved with the design, manufacture and marketing of digital audio workstations for the audio production industry.

The management team is headed by Glenn T. Edens, president and CEO. Previously, Edens was co-founder of portable computer maker Grid Systems and, prior to that, he was with Apple Corporation. Vice president of engineering is John L. Melanson. Steven M. Cunningham, who was previously director of sales and marketing for J.L. Cooper Electronics and 360 Systems, serves as vice-president of marketing. Senior engineers include Eric Lindemann, formerly of Linn Electronics and Mitsubishi/Quad-Eight; Carl Fravel, founder of Gentle Electric; Dana Massey, formerly of E-mu Systems; and Roger Powell, developer of the Texture MIDI sequencer and keyboardist with Todd Rundgren's Utopia.

The company's digital audio workstation will be modular in design to allow a wide variety of configurations. Third-party product development will be encour-

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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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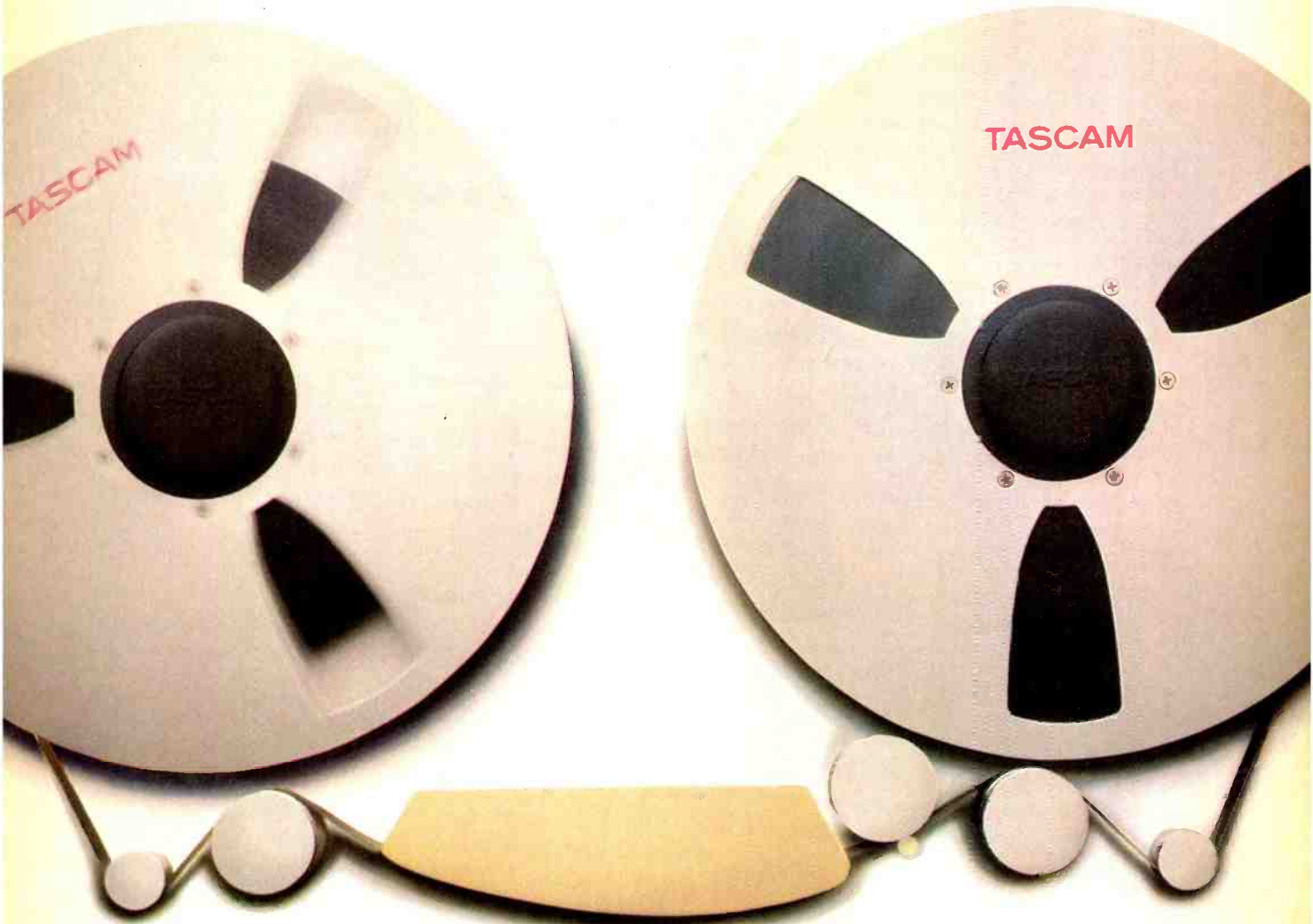
BPA Membership applied for December 1986

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Post production is a race against time.

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You can't afford to use equipment that slows you down. Because the competition is waiting to pass you by.

In a unique response to this situation, Tascam's engineers have created the first racing machine for audio post production.

Introducing the 24 Track Tascam ATR-80 Recorder/Reproducer.

The ATR-80 is built for speed. The kind of speed that makes post production a lot more productive.

For example, it shuttles at 380 IPS. Which means you'll never have to wait for the audiotape to catch up with the videotape.

It cuts editing time too, because unlike other audio recorders, the ATR-80 allows edit previews. In other words, it eliminates retakes and ruined tracks.

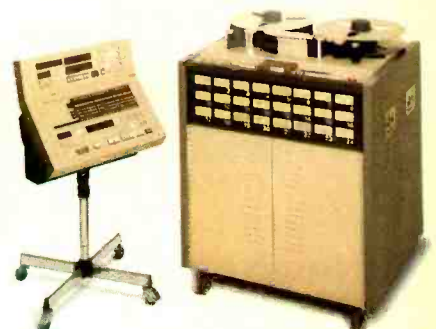
It also accelerates the overdubbing process. Tascam's unique heads are so uniformly responsive, you can make final EQing decisions right in the Sync Mode. Without ever having to rewind and check the sound from the Repro head.

The ATR-80 even accepts 14 inch reels—a complete, uninterrupted one hour show's worth of tape.

The fact is, no other post production recorder keeps you competitive in so many ways.

So take a look at an ATR-80 and put it through its paces.

It'll be first past the post.



TASCAM

A 32-Track version of the ATR-80 will be available soon. ©1986 TEAC Corporation, Tokyo.
Harman (Audio) UK, LTD. Mill St. Slough, Berks. SL2 5DD.

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aged by WaveFrame's open-system architecture, details of which will be announced in mid-1987. Delivery of production systems is scheduled for the latter part of 1987.

London-based Lansdowne Studios acquire CTS

In late February the CTS studio complex was acquired by Lansdowne Recording Studios. CTS, one of England's largest studios, has been the subject of industry-wide speculation in recent months. Various rumors suggested imminent closure or sale of the studios by owner Lee International.

Although both CTS and Lansdowne are best known for film and TV soundtrack recording, the new directors point out that their clients will be drawn from all aspects of music recording, not just the film world. The new board of directors, to include Adrian Kerridge, Chris Dibble and Johnny Pearson (Lansdowne), Peter Harris and Dick Lewzey (CTS), will be implementing a number of changes at CTS in the near future. Improvements will be made in facilities and services; the new management intends to shake off the somewhat "institution-like" atmosphere that characterizes CTS after 15 years of ownership by public companies.

The two facilities will operate autonomously, although bookings for either can be made at both locations. Five studios are available, including a 130-musician capacity room at CTS, studios accommodating up to 25, 35 and 40 musicians, and a fully equipped synthesizer suite.

- In a related move, the trial period of CTS' Neve all-digital DSP console, operated under the U.K.'s Department of Trade and Industry scheme to encourage new technology, is approaching its end. To incorporate technology advances that have taken place during the trial period, the preproduction DSP console is being withdrawn to enable Neve to carry out an extensive refurbishment program. It is anticipated that the DSP console could be re-installed in the CTS Music Centre, Wembley, North London, at a later date. The studio expects to replace the DSP with a V Series console fitted with NECAM 96 fader automation.

Recording textbook from Howard W. Sams

"Introduction to Professional Recording Techniques," by Bruce Bartlett, a new title in the Howard W. Sams Audio Library, introduces the John Woram

Audio Series. This new series is aimed at the professional, student and hobbyist.

The new title is said to provide special emphasis on microphones and mic techniques, as well as sampling, sequencing and MIDI control. Topics covered include the recording and reproduction chain; simple home recording; setting up the studio; microphones and microphone techniques; control-room techniques; on-location recording; and judging the recording.

Yamaha establishing new organization structure

Effective April 1, Yamaha International Corporation has changed its overall corporate structure. The name will be changed to Yamaha Corporation of America (YCA) and the corporation will have a new subsidiary company, Yamaha Corporation, USA.

YCA will coordinate and develop strategies for all Yamaha corporations in the United States, with the exception of Yamaha Motor Corporation, USA and its subsidiary companies. YCA will also coordinate Yamaha's operations in Canada, Mexico and Panama; again, with the exception of motorized products. Seiji (Sam) Kajimura will be president of YCA.

Kelly Quan Research appoints first dealer for synchronizer controller

Pro Media, San Francisco, has been named as Kelly Quan Research's exclusive Northern California dealer for the new SC610 synchronizer controller software. In addition to offering full CMX compatibility, the new series is said to feature selectable screen display and runs on any IBM PC.

Features include the setting up of ADR loops with only two keystrokes; insertion of sound effects with as little as three keystrokes; saving of EDLs to disk; performing auto assembly from EDLs; and control of a Cipher time code generator.

Obituary: Richard C. Heyser

Richard C. Heyser, president-elect of the AES, died on Saturday, March 14. His creativity in audio technology and his service and concern with the AES will be missed by the audio production industry.

Heyser was born in 1931 in Chicago, and attended the University of Arizona, where he received a BSEE degree in 1953. Awarded a Charles LeGeyt Fortescue Fellowship for advanced studies, he earned an MSEE degree from the California Institute of Technology in 1954, where he spent the next two years

doing post-graduate work.

From 1956 until the time of his death, Heyser was associated with CalTec's Jet Propulsion Laboratory, Pasadena, CA, where he was a member of the technical staff. His work involved communication and instrumentation design for all major space programs at JPL, beginning with the conceptual design of America's first satellite, Explorer I. Recently, he had been involved in the application of coherent spread spectrum techniques to improving underwater sound research and medical ultrasound imaging.

In addition to his work at JPL, Heyser maintained a personal laboratory where he conducted research on audio and acoustic measurement techniques. This effort resulted in a number of papers published in the *JAES* and elsewhere. He was awarded nine patents in the field of audio and communication techniques, including time-delay spectrometry.

A member of the IEEE, Heyser was a fellow of the AES, and the recipient of its Silver Medal Award in 1983.

A personal tribute from Don Eager, Techron division manager, Crown International:

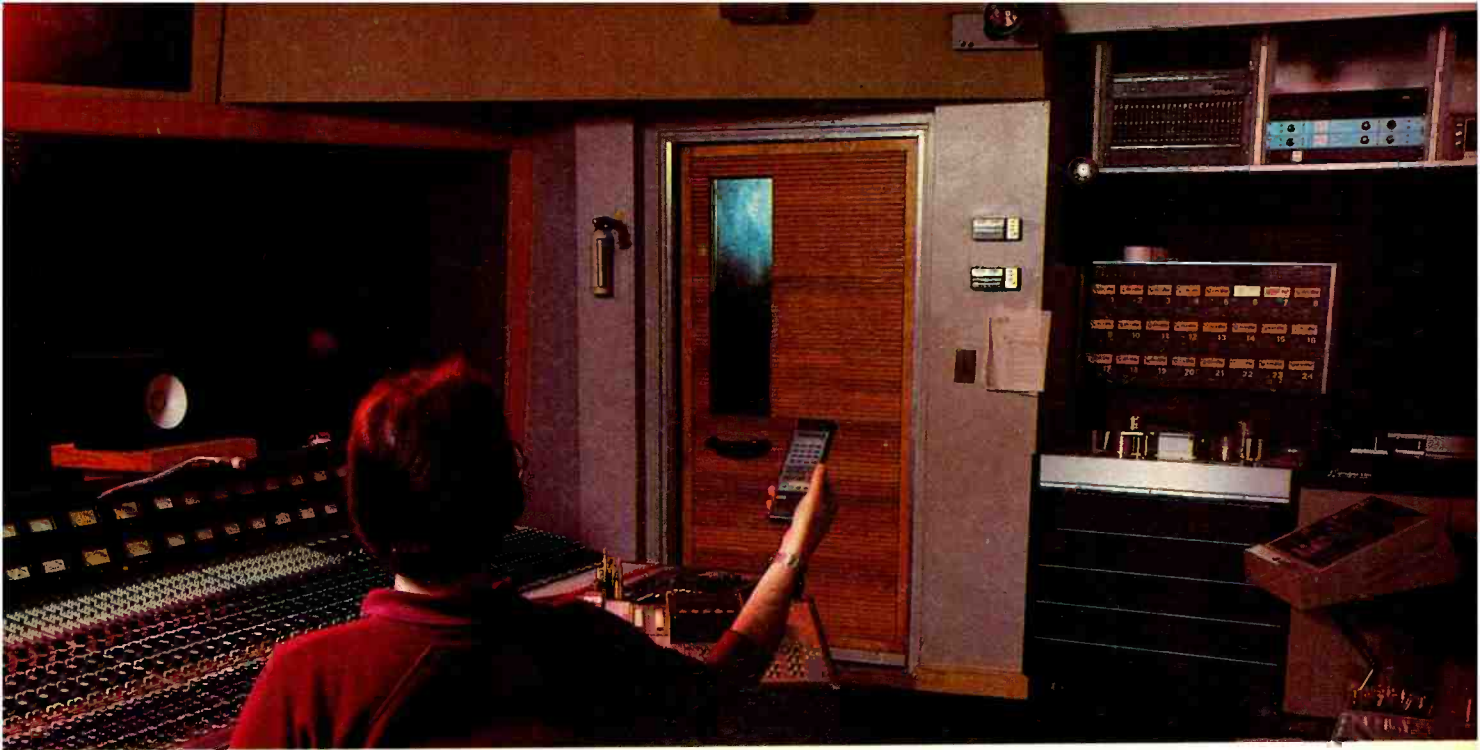
Richard Heyser, a man who shared what he possessed with others, once said at an AES meeting: "It pains me to give away in 10 minutes what it took me a lifetime to learn."

That pain must not have been too great because Dick always shared his knowledge with anyone who was genuinely interested in his work. So often he would sit for hours sharing his discoveries with people gathered around him. If anyone would indicate that Dick was talking above their understanding, he would smoothly and undetectably shift his conversation to a level that could be understood. Dick's way of life consisted of giving to others. He only took away the secret cloaks of nature to let the truths be known.

What boyish enthusiasm Dick possessed, and how he infected those who knew him with that enthusiasm and curiosity. Dick's discoveries about energy, measurement and multidimensional observation will not be fully understood for many years to come.

Richard Heyser, truly a man of great insight about our universe. His work was built on the philosophy "There is an *is*." Yes, he understood that this old universe has order, and there is a God that made that order.

REP



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Audio Performance

Many digital processors force a trade-off between enhancing the sound with effects and degrading it with noise or timbral coloration. Not this one. Frequency response of the *wet* signal is 20 Hz – 20 kHz and dynamic range at the effect output is 94 dB: specs equivalent to a CD. The DRV-3000's impeccable sound is made possible by a pair of high performance digital sampling processors (DSPs) operating at 44.1 kHz.

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Dual Reverb/Effects Combinations

Each DRV-3000 program is a combination of two effects in series or parallel. Choose from five Reverb spaces or plates, two Early Reflections, two Echoes, Auto Pan, Flange, Chorus, Ensemble, two Pitch Shifters and a Parametric EQ/Driver. Combine reverbs for rich and complex ambience, create dual effects that go beyond special to unique.

Assignable Control Jacks + MIDI

Real time response adds more effectiveness to any signal processor. The DRV-3000's two assignable footswitch jacks can step through programs, change reverb times or cancel the effect. The program memory and pitch shifter speak MIDI as well.

Stretch Your Imagination

Start exploring the possibilities at your Korg dealer. But be ready. The DRV-3000 is so powerful, so responsive, so quiet, it will expand your idea of what effects can be.

For a free catalog of Korg products, send your name and address, plus \$1.00 for postage and handling, to: Korg U.S.A., 89 Frost St., Westbury, NY 11590, Korg U.S.A. West, 7886 Deering Ave., Canoga Park, CA 91304 Exclusively distributed in Canada by: Erikson Music, 378 Isabay Street, St. Laurent, Quebec H4T 1W1

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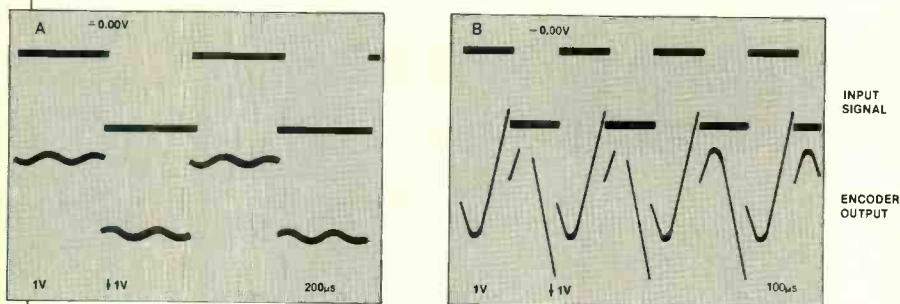
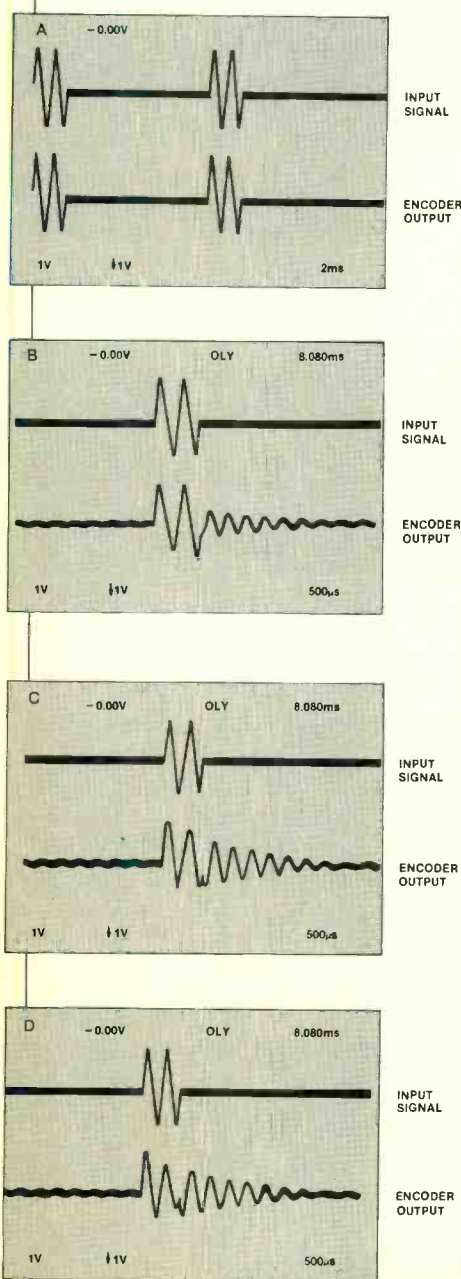


Figure 1. [Above] Square wave response of copy-code encoder with -80dB notch attenuation at a) 1kHz and b) 3.7kHz .



Anti-copying measures

From: Bart Locanthi, president, BNL Research Associates, Pasadena, CA.

A draft bill (S.506), now being considered by the United States Senate, requires that a copy-code scanner be incorporated into all digital audio recording devices. A similar bill (HR1384) is also before the House. The aim is to prevent unauthorized copying of recordings.

In order for the simple-minded copy-code scanner described in both bills to operate, all recordings that are to be encoded must be processed through a band elimination (notch) filter to remove frequencies between 3.7kHz and 3.9kHz . Various test signals were passed through a notch filter having the characteristics of the proposed encoder. The effects on the signal are shown in Figures 1 and 2.

While we are all trying to capture, with the highest fidelity, every nuance of the musician's performance, it is ludicrous to consider eliminating from the recording a part of the audio spectrum that may contain music. We seem to be facing a possible law that mandates that the consumer pay a premium for hardware that reproduces inferior recordings.

At the request of the record companies, the design of the consumer digital audio tape recording machines included two characteristics aimed at preventing direct digital-to-digital copying of Compact Discs (CDs) and prerecorded digital audio tapes. These are:

1) the use of 32kHz and 48kHz sampling frequencies, neither of which is compatible with the 44.1kHz sampling frequency used for CDs and prerecorded digital audio tapes; and

2) a device that stops the digital audio tape recorder when it detects write-protect flags which are incorporated into the digital audio data stream for CDs and

Figure 2. [Left] Tone burst response of copy-code encoder with -80dB notch attenuation at a) 1kHz b) 3kHz c) 3.4kHz d) 3.6kHz .

prerecorded digital audio tapes.

In the light of these measures that are now in effect, it seems unreasonable to require an additional device that would impair, forever, the quality of the original recording.

Following are lists of the Senators and Congressmen proposing the law:

For S.506: Senators Gore, Wilson, Cranston, Danforth and Kerry.

For HR1384: Congressmen Waxman, Cooper, Moorhead, Berman Sundquist and Morrison.

High performance digital processing

From: Gil Griffith, sales manager, Eventide, Inc. Little Ferry, NJ.

I read with great interest the column entitled "Living with Technology" in the March issue of *RE/P*.

In it, writer Stephen St. Croix discussed a powerful, new "high performance digital processor and reverb unit" that had recently been introduced. He went on to describe its unique on-board compiler enabling the user to design DSP effects on a computer and then upload them into the processor.

While we share in Mr. St. Croix's interest in this new reverb/effects processor, we would like to point out that the unit he has outlined is not new; in fact it has been available for three years now.

The device he is describing is the Eventide SP2016 digital reverb/effects processor, combined with the Eventide SPUD (Signal Processing User Development) system, which was developed in 1984.

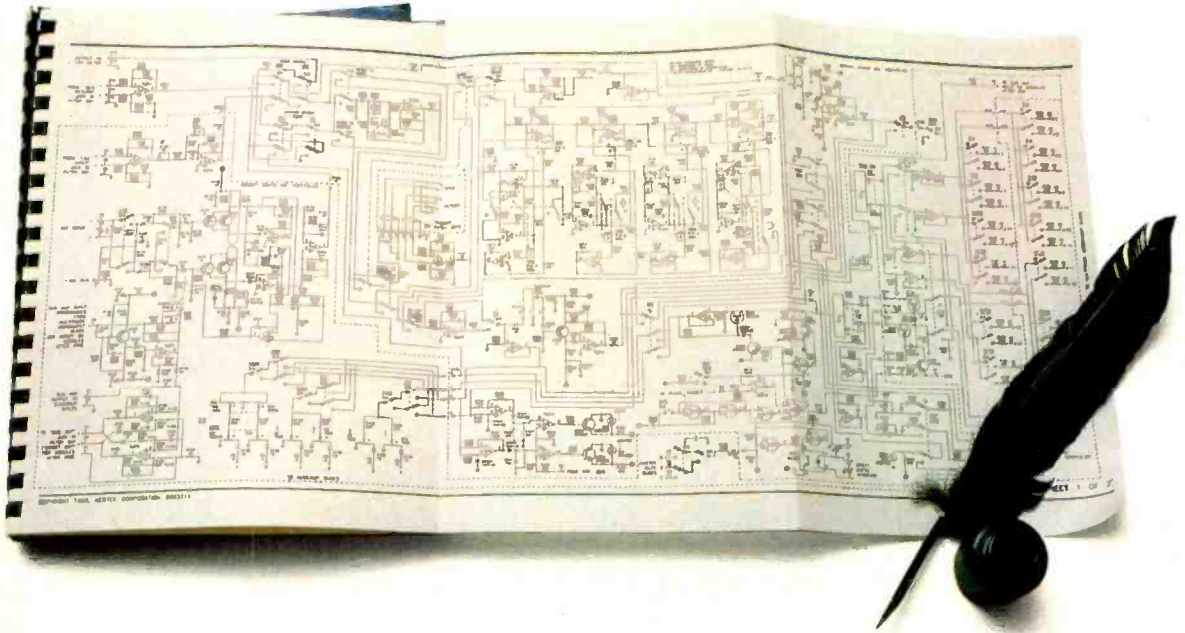
Comprised of an SP2016 equipped with an IEEE-488 interface, custom Eventide software and an IBM PC, compatible, or HP computer, it becomes a powerful, versatile development system for the creation of a world of effects, limited only by the creativity of the developer.

We don't know if Steve was referring to our unit specifically, or that of some other manufacturer, but we wanted you (and the world) to know that we have it, and we had it first.

Reply from Stephen St. Croix, *RE/P's* technology developments consulting editor:

No, I was not describing the SP2016. As I remember it, the low level of penetration of HP hardware into the recording studio community may have had something to do with your system's acceptance. Since you brought it up, I was talking about the new Quantec XL. It won't be available for another few weeks. **REP**

To us, it's sheet music.



It goes without saying that great performance begins with great design. And that mixing console designs are judged by the most critical performance standard: Great sound.

Still, you may not care that Neotek designed transformerless consoles years ahead of everyone else. Or that we introduced state variable equalizers. Or that circuits that others would say are revolutionary are pretty much old hat with us. Our new hybrid amps, for example.

But you do care about sonic quality. That's how your own work is judged. It's how you judge ours.

Neotek's reputation for performance is built on outstanding circuit design. It's the reason no other console sound compares.

Our sound is the reason Abbey Road bought their first American made console. The reason you find Neotek credits on the finest classical CDs. Why top mic and synthesizer manufacturers demonstrate on Neotek consoles.

You can put our designs to work in your studio. Whether you need to capture a perfect vocal, get every nuance of a drum kit, or input a hot sample that makes your synth sound its best,

your Neotek sound will be remarkably brilliant, clear, and musical.

That's why we work so hard to put our console designs at the leading edge of technology. It puts your sound ahead of all the muddy and fuzzy alternatives.

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Circle (8) on Rapid Facts Card

May 1987 *Recording Engineer/Producer* 11

Managing MIDI

By Paul D. Lehrman

Before computers, MIDI and digital gear of all stripes invaded the recording studio, the house maintenance technician occupied a singular position. It was he who could strip down anything in the place and reassemble it—usually so it was working better—in half an hour. When there was custom work to be done—maybe some doodad that a client needed to produce some wild effect—the maintenance tech could be relied upon to dive into the project head first.

Today, however, maintenance work is as much sending circuit boards and EPROMs back to the manufacturer as it is tracking down noisy capacitors. With computerized consoles, editors and sequencers, custom design is now largely the duty of computer hackers, and hence many large studios have taken on a full-time programmer to handle the task.

But smaller studios, and especially artist-owned studios, cannot afford the luxury, and so they have to rely on commercial software suppliers. Although much commercial software is terrific, invariably a studio will need a custom application or modification that the commercial purveyors aren't interested in writing. When that happens, even the most devoted owner of a multimegabyte number-cruncher will look at its smiling CRT and curse loudly.

A few years ago, when Apple IIs and Commodore 64s were the state of the personal computer art, this wasn't such a big problem. You didn't have to be a brilliant programmer to do nifty little things with these machines. For example, I wrote my first working music software on an Apple II clone in the pre-MIDI days. It was a quick and dirty utility for a then-state-of-the-art music program that took a 256-byte wavetable and subjected it to various forms of filtering. Although it was the first computer program I had written in over a decade—my previous experience consisting of a college course in PL/1 programming on an IBM 360 mainframe—it took me only two days.

Over the next year, I wrote dozens of programs. The Apple II was fun to use, had BASIC in ROM, and the available music hardware was relatively simple and extremely well-documented. I found lots of other folks, many of them self-taught programmers, who were doing the same thing.

Then MIDI came out, and it all stopped dead. It wasn't MIDI's fault; the possibili-

ties of data combinations were more complex than anything seen before, but that could be dealt with. The blame lay more in the interfaces, and later in the computers themselves. The introductory booklet for a well-known MIDI card for the Apple II and Commodore 64 contains 30 pages of timer-, register-, interrupt-, sync-, and data-control codes, which you have to know about if you're going to write a program. The booklet doesn't even mention MIDI—if you want to know about that, you've got to get a second book!

***With the coming of MIDI
and the 68000, the
programming priesthood—
whose power had been
briefly diluted—once again
held the only keys
to the kingdom.***

And all of a sudden, BASIC wasn't fast enough. If I wanted to work with MIDI, I was going to have to expend a significant amount of energy learning assembler, or buy (and learn) some high-level language development system. If I were to keep up with the rest of the world and leave the safety of the Apple II for a more advanced machine like the Macintosh, there was a new, complex operating system to master.

So, I stopped writing code, and relied on other people to develop the tools I wanted. With the coming of MIDI and the 68000, the programming priesthood—whose power had been briefly diluted—once again held the only keys to the kingdom. It was as if a little window has opened, allowing a little sunshine to come through, and then shut again.

Fortunately in the world of computers, nothing stands still, and the vacuum of good music-development tools for non-professionals was too big to last. Again, a democratic movement in music software is afoot, and I, for one, am overjoyed.

I recently came across a newly released commercial program that comprises a set of machine-language routines for use in conjunction with a standard BASIC language, either MS-Basic from Microsoft or ZBasic from Zedcor. The commands provide a programmer with the ability to input and output MIDI events or strings, selectively filter data and set up variable-sized input and output buffers. It also works with any MIDI interface for the Macintosh, allowing the user to specify clock speed and port.

Although the current version could not support a full-blown multichannel sequencer (only because it lacks certain time-stamping functions—and those are promised for the next version), it can, and will, be used to create custom applications that previously would have required a thorough grounding in T or some other high-level language. And now that there are true BASIC compilers for the Mac, it is possible to create stand-alone applications using the language—you no longer have to include a "run-time interpreter" with every program.

The package comes with several examples of simple application programs in both run-time and compiled versions. One is a program that prints incoming MIDI data on the screen. Another memorizes a pitchwheel move and plays it back every time you hit a note. There's also a Casio CZ librarian, and OTIS (for "One-Track Itsybitsy Sequencer") which lets you record, edit (line by line, like an ancient word processor), play back and print out a single track.

The system's creators have carefully balanced sophistication and simplicity. The instruction set is small (only 10 commands), but more than adequate. BASIC, after all, was designed to let novices program computers, and the package complicates things not at all. The pitchbend program, for example, takes up only 20 lines of code. Here are a few applications off the top of my head.

- *A MIDI echo device.* Anything you play in, it repeats a preset interval later. It can transpose, change channels or even invert note order or pitches at the same time.

- *Patch librarians* for "orphan" synths; MIDI instruments that have been ignored by major software manufacturers.

- *A setup librarian.* With one command, it loads all your synths and processors with the patches and performance data for the day's session.

- Here's my favorite, inspired by watching a Broadway pit musician struggle through a 100-page score with an average of three patch changes per change: a *multichannel patch "stepper,"* which will let you define all the patch changes and controller definitions for a score, in order, for several synths. Hit a foot pedal, and the program advances to the next cue. Hit a button (or combination) on the master keyboard, and the program jumps to a specified event. Simple, but think of the headaches it will avoid.

By the time you read this, I'll probably have written this program.

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

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Sound on the Road

By David Scheirman

When a new sound system is being assembled for touring use by a sound company or musical group, or an existing road package is being upgraded, a host of choices confront the system assembler. If the rack-mounted signal processing devices are to be purchased as a long-term capital investment, what is the life expectancy and potential resale value of various items? Will they interface with the existing electronics racks' ac provisions and wiring harnesses without additional modifications? If audio quality is truly a governing factor over and above cost, will the units under consideration improve or degrade the overall fidelity of the sound system?

When examining signal processing devices in a portable concert sound system, certain categories will become apparent, including individual input signal control, tools for creative mixing effects and devices for altering overall system fidelity. While the types, brand names and packaging of each will vary from system to system, some directional trends are taking place.

• **Individual input controls.** We take for granted today's sophisticated input EQ sections, group output assignments and effects buses. In the future, we may very well take for granted onboard compressor-limiters and noise gates for each input module. Already available on the most expensive recording and broadcast production desks, these input signal processing functions can help live soundmixers get one step closer to achieving total control of the multiple microphone and DI inputs at their command.

One custom-console designer recently prototyped a new touring console that offers just such features; each input module has integral signal processing, doing away with the multiple channel-insertable devices currently found in touring sound equipment racks. As these signal processing functions start to move onto the console surface itself (and if they do), will consoles become larger and heavier, to compensate for the outboard electronics racks shrinking in size?

Today's studio and broadcast desks offering such functions are not only heavy, but also costly. Simpler, sophisticated circuitry developments and lighter materials may take live-console design in this direction.

• **Creative mixing effects.** Tape-loop echo devices and vacuum-tube effects

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

units have given way to digital technology; today a veritable smorgasbord of affordable, controllable reverb and special effects are available from many manufacturers. While the "flavor-of-the-month" syndrome still holds forth on occasion, this side of touring sound equipment is starting to open up considerably. A wider variety of programmable reverb and delay devices than ever before can be seen on the road.

Several manufacturers are just scratching the surface of MIDI technology for both consoles and effects. As this trend

Will the special effects units under consideration improve or degrade the overall fidelity of the concert sound system?

grows, we can perhaps look forward to dedicated control sections on the mixing console that interface directly with specific rack-mounted effects devices, incorporating both original, performer-generated input signal as well as the soundmixer's own adjustments.

While steps have already been taken in the direction of creating and storing custom reverb, slapback and delay programs with today's more complex effects tools, some soundmixers have found these devices intimidating; we can look for new devices that offer great programming flexibility with improved user interface functions.

• **Utility delays.** Although not strictly within the realm of "creative mixing effects," what we can call utility delays are none the less becoming a more common part of large touring sound systems. As the more advanced sound systems start to feature both rear-delay flown clusters and multiple-delay outputs for performing array time/phase alignments, such devices are finding a ready market on the road.

Outdoor major festival setups with delay towers, hanging rock arena systems with mid/high-frequency augmentation clusters at the back of the hall, and high-quality multizoned distributed ceiling systems for industrial theater events all make use of such devices. Many sound companies are currently using their in-stock special effects delays while shopping for just the right utility delay units.

• **Fidelity alteration devices.** "Fidelity alteration" is perhaps an odd expression,

but how else do you describe the interesting trend toward signal processing units whose apparent design objective is to change the overall tonal quality or "fidelity" of an entire live mix? Several manufacturers now offer dynamic-range expanders or program compensators boosting lows and highs to present an averaged or optimum tonal character.

Also available in this category are "enhancers," devices that selectively stress the higher frequencies to enhance intelligibility. Often used as special effects on selected inputs—to boost articulation of such items as piano and acoustic guitar—enhancers are also used on entire program mixes, particularly in highly reverberant rooms.

Bass sub-harmonic synthesizers, for use with inputs such as kick drum, bass guitar and synthesizer, and for boosting the apparent low-frequency level of an entire sound system, also are appearing on the road more frequently. A single rack-mounted device can be more cost-effective than half a truck of subwoofers. Knowledgeable sound system operators have found that this concept will work acceptably if the original sound system's bass speaker components can handle the additional input load.

• **What's next?** Smaller, more compact, more versatile and more affordable signal processing devices will become available as the cost of the circuit parts continues to drop; the ideal combination of price, packaging and performance is never quite attained. Something more ideal and more tantalizing always seems to appear (often right after the year's major purchases have been shipped in).

The "wish list" of many touring sound system operators might include a single-rack space, professional unit with eight or 10 interchangeable modules that could include gates, expanders enhancers, limiters and even crossovers, thereby providing the soundmixer with the ability to channel-insert without consuming valuable rack space.

Currently, modular packages like this offer up to eight or nine cards in three to four rack spaces; the units are relatively costly and bulky. Several manufacturers are working on prototypes of units that offer more modules in less space; one company has fielded an interesting 8-channel compressor/limiter in a single-space package.

Make a thorough survey of the marketplace before making purchasing commitments; the signal processing field is changing rapidly.

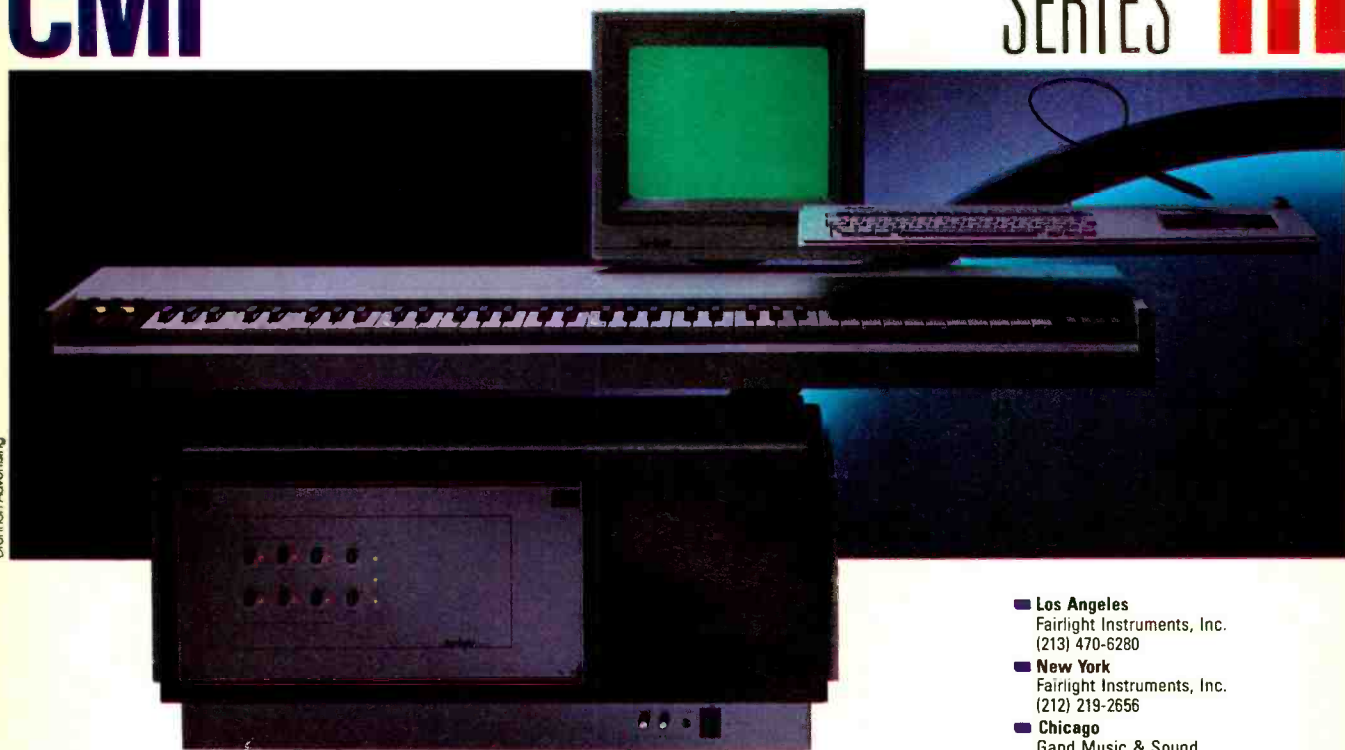
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May 1987 *Recording Engineer/Producer* 15

Film Sound Today

By Larry Blake

Stereo production recorders: can't live with them, can't live without them. The use of stereo or, more precisely, 2-track tape machines during production has come into favor during recent years.

The discussion below applies equally to any stereo production recorder: Nagra, Stellavox, Uher, PCM-FI or R-DAT. Also note that stereo means 2-track mono in almost every instance, with separate information on each track.

As stated in last month's column, clear log sheets are an important part of a production mixer's job; recording with a stereo machine places an even greater demand that they be concise and "it-wasn't-clear"-proof. On a recent shoot, I found it all too easy to become lazy while working with two tracks; put the boom mic on one track and either the wireless or the plant (hidden) mics on the other.

This technique spares the production mixer the pressure of having to make a permanent blend on the set as the scene is shot. The rationale is that, by separating the different mics across two tracks, the sound editors and re-recording mixers will be better able to blend the two during post-production.

I think this reasoning is specious on two counts. First, sound editors like to edit on 35mm single-stripe mag film, with sibilance and offending noises easily removed by scraping with a razor blade or sanding with emory cloth. *Producers* like single-stripe mono transfers because they are cheaper (than fullcoat mag). However, even if editors retained their ability to razor-blade tracks on 2-track material by using 3-stripe mag, or separate transfers of each track onto single-stripe, there would still be a problem.

Editors and mixers never, and I mean *never*, have the time to audition each track separately to determine what is recorded on them and how to handle the separate information. They have a tough enough time deciphering what is on the different takes that have to be intercut to produce a smooth scene, without having to listen to two separate tracks per shot.

Another problem arises in that the sound editor doesn't know how the picture editor's worktrack was transferred: left-track only, right-track-only, 50/50 or what? Because the worktrack is the guide for "mod matching" the fresh transfer ("reprint") of production tracks that will go to the dub, it's hard to make a comparison if you don't know what you are comparing.

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

Stereo production recorders are also Pandora's boxes, even when recording a single channel across the two tracks in mono mode. I have two problems with this technique, compared to recording the single channel on just the left track. While it is true that signal-to-noise ratio will suffer slightly with single-track mono, I can't imagine that anyone would be able to hear the increase in tape hiss with dialogue recorded on a soundstage.

The quality could suffer noticeably, however, if a 2-track mono recording was transferred on a machine whose

The resultant phasing might slip past the editor, only to rear its ugly head on the dubbing stage.

azimuth differed greatly from that of the original field recorder; the chances are that phase-cancellation problems might not be noticeable.

A worst-case scenario, however, would have the resultant phasing slip past the editor listening on a sync block and Moviola, only to rear its ugly head on the dubbing stage.

Also, single-track recording makes things much simpler for the transfer person; only one needle is moving on the playback 1/4-inch machine's meters. There's no confusion here, simply patch it into the input of the single-track recorder. No summing amps, no mixing board and no interpretation on the part of the transfer person is needed to combine the tracks. The chances of polarity or phase problems screwing up the transfer are virtually eliminated.

All of the above is yet another reason why the production team should also reduce the number of times they record separate information across two tracks. The extra trouble that production crews take in delivering one good track will pay off in post-production.

Having said all this, I still believe that stereo machines are useful tools on production and I wouldn't do a film without one. First, and most obviously, the machine is there if the opportunity arises to capture stereo sound effects, such as crowd walla, that are difficult to pull out of a library or record at a later date. Also, there are times during multicamera shoots when more than one track is almost necessary because it is impossible to "mix" for camera perspective.

Two tracks can also come in handy when you are working in an environment with a consistently loud background, such as by the ocean or in traffic. The second track can be used to create a continuous segue that otherwise would have been impossible with the background and dialogue tied together. The technique of recording an ambient mic on a separate track doesn't usually work, I am told, if the scene is shot on a relatively quiet set and the extra mic is intended to capture room tone "fill" throughout each take.

Think about it: room tone has to match the character of the production dialogue track. To secure a "minus-dialogue" recording during the take means that the ambient mic has to point in a *different* direction. The production sound team would do better to cajole the crew to sit still for 10 seconds immediately after the scene. (Remember to note the existence of end-of-take room tone in the sound log!)

Because these 10 seconds of silence "take time" to record and can't be done after every shot, the production mixer can give the sound editors a few seconds of fill at the beginning of each and every take, simply by recording the vocal slates before the cameras roll, and *not* between the time the Nagra starts and the assistant cameraperson yells "marker" before hitting the clap stick. These precious few seconds are often very important to sound editors.

All of these techniques are the subject of endless debates. The real point to be emphasized here is that, however a production sound team secures a useable track, the effort is worth the trouble. Post-production dialogue replacement (looping) costs money and, in the hands of unskilling ADR editors and dialogue re-recording mixers, can make the actor look downright silly. As a result, there are some directors, such as Woody Allen, who simply refuse to loop.

In recent years, production and re-recording mixers have found themselves caught up in (no pun intended) a loop: if the directors and producers don't care about taking the time to capture a good production track, the dubbing mixer has to work extra hard to breath life into ADR tracks while matching them with location tracks filled with camera noise. When the re-recording mixer makes it all work, the producer is even less inclined to give the production mixer on his next film a chance. No microphone, recorder or mixer can solve this problem. **RE/P**

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Living with Technology

By Stephen St. Croix

Something very interesting happened while I was waiting for Godot.

For years now I have been waiting for technology to advance to the point of putting 18- or 20-bit audio conversion on the street. I mean, 20 bits ought to just about do it, don't you think? Almost analog—almost. Good Old Digital Oversampled Technology. But it is taking forever. Gee, it seems that each bit is twice as hard as all those before it. I wonder why that is?

Remember the first time you played with an 8-bit sampled drum machine? It seemed the perfect way to use 8 bits; how much can you mess up a snare drum? How long ago was that, 20 years?

Although I accepted and used those 8-bit drum machines, I did so only after much modification. I usually resampled new drum sounds at two or three times the machine's original clock rate, burned my own ROMs (of course, two or three times the memory is needed) and even opened up the machine's low-pass filter sweep ranges.

After all of this, I actually got very usable results. For cymbals I often went with four times the original sample rate, and four times the length. This means that I had to make a free standing card (with 16 times the original memory) that was simply triggered by the drum machine's original cymbal trigger, and left to run at its own speed. A real pain, but a real cymbal; sort of.

Then came the 8-bit samplers and synthesizers. I use them. We all do. If the skip algorithms are very good, and the voice filters are good and properly programmed, and we are careful with exactly how they are used, we can get them on the air without the record label getting nasty calls from customers.

We have lived with this 8-bit conversion technology for so long because the A-to-Ds and DACs were very easy to make (and control), 8-bit address and bus technology was simply there already, RAM was so expensive, and so little of it was in each IC. (Don't forget that for any given sampling frequency, changing the sample resolution from 8- to 16-bit resolution doubles the amount of memory needed for each sample which, in turn, doubles the amount of RAM needed for each second of audio stored.)

Non-linear (companded) 8-bit, along with several other clever schemes to get

the most out of those pitiful few bits, have added to the life of the 1-byte word, but enough is enough!

Okay. Twelve- and 16-bit synthesizers and drum machines are finally on the scene. That's pretty good, since adding only one bit doubles the resolution and dynamic range of the system.

This gets me to the point: I have never thought that 16-bit resolution was truly enough to simulate analog. I figured that oversampled 20-bit might get the job done, but that by the time it does, my ears will happily roll off at 4kHz.

For the first time the pro-audio community can perform digital processing of high enough quality to approach sounding like analog itself.

Because of the convergence of customer awareness and technology, our target market right now is that sweet 16: 16-bit linear for CD and R-DAT. For some time the consumer has had access to some very sophisticated 16-bit technology. They have been living with ultra-linear, dual D-to-A conversion, and even oversampling in their CD players. They have been *listening*, however, to recordings made with 8- and 12-bit instruments recorded with simple 1x, shared 16-bit conversion!

In fairness to our own manufacturing industry, the truth of the matter is that 16-bit digital-to-analog conversion is relatively easy, while 16-bit A-to-D conversion is not. In fact, it is very difficult to do well.

To begin with, when using a sample frequency of 44 and change (44.1 or 48kHz) some serious filters are needed to deliver 20kHz audio without aliasing. Filters this severe are always audible and nobody I know argues that this sound is an improvement. Distortion, noise and group-delay problems from these brick-wall filters, sample and hold settling and aperture distortion products, digital interference within the chassis and actual circuit board layout are only some of the factors that dictate the performance of A-to-D conversion.

Simply moving a capacitor a 1/4-inch on a PCB can determine if you are going to get 14 or 15 bits out of the system. In fact, the potential dynamic range of 16 bits is so high that it puts the last two bits right in the noise floor of conventional

analog design, even without a bunch of 20MHz clock garbage flying around.

A-to-D conversion to 16 bits is still an art, not a science. Even when the problems mentioned above are resolved to the point where the target of a 92dB-96dB dynamic range is hit, other things that you don't often hear about, such as sample-and-hold complex distortion factors, monotonicity, linearity, DLE (differential linearity error; watch for this, because you are going to start hearing about it) and zero-cross characteristics can keep things from sounding right.

So, with all this in mind, and still in search of Godot, I went to the AES Convention in London this past March, and spent 10 days visiting European manufacturers to catch up on various technologies. It was during these travels that I came upon something that has actually changed my thoughts on the absolute potential of 16-bit audio.

One company out there has designed a 16-bit conversion system for its newest signal processor that is revolutionary. I have never heard anything like it. The system offers all of the advantages of the best high-end consumer CD players to come along later this year, and more.

This new unit has dual 16-bit, 4x oversampling DACs with filters so good that square waves make it through the entire A-to-D and D-to-A loop essentially as...square waves. It is the first device to offer 2x oversampled *inputs*. The company decided that no commercial A-to-D system available offered the desired input performance, and that the trick of using a good DAC in conjunction with a successive approximation register didn't cut it either.

A radical new system was designed that uses flash converters in a way that is ingenious, and the theoretical possibility of those gross errors that plague conventional A-to-D converters simply aren't there. This unit's sample-and-hold design (and all the other magic) is so good that I actually listened to 4-bit music through the device and found it quite acceptable. Even 2-bit music was impressive! The last bit is fully under control.

I don't write about this device to promote it; I highlight it because for the first time the professional audio community can perform digital processing that's not only as high in quality as the best CD players that will eventually play it back, but also of high enough quality to approach sounding like analog itself.

A journey of 65,536 steps begins with a single bit.

REP

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

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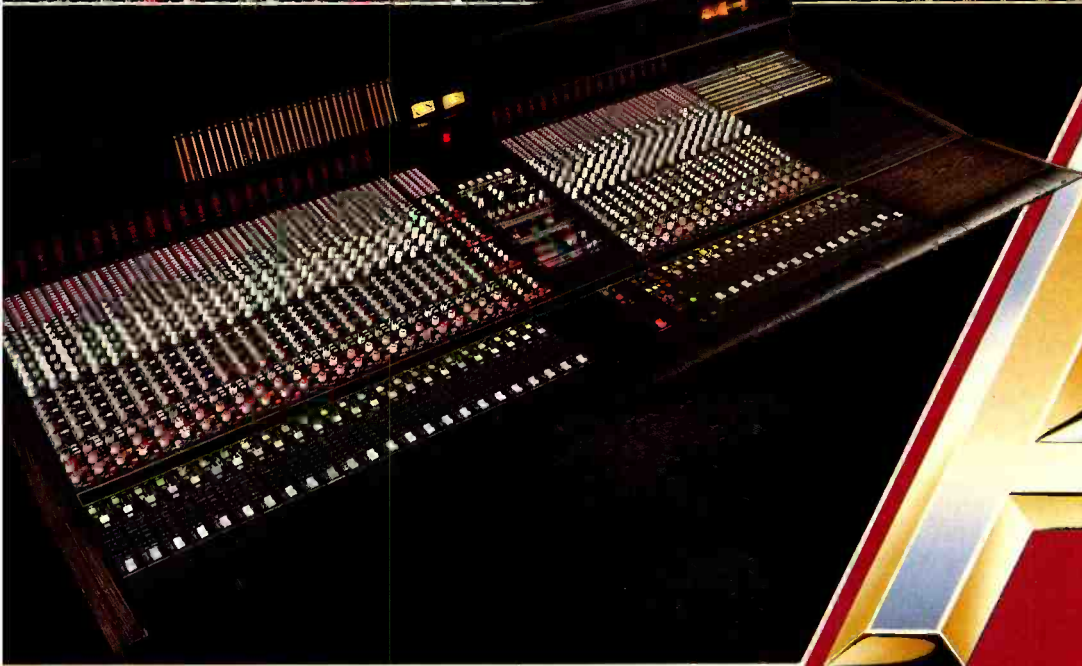
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GEORGE MASSENBURG LABS

SPARS On-Line

By Gary Helmers

In this month's column, I'd like to discuss studio construction with a focus on choice of locale, marketing and some of the potential pitfalls.

Within most communities large enough to support more than one recording facility, there is a trend for studios, video post-production houses and related services to locate in one section of town. In some instances, a successful operation may deviate from the norm in an attempt to service a particular market. For example, if your intended customers are ad agencies, how do you determine the ideal location? In many cities the ad agencies are located near the city's center, some distance from most music studios and video production houses, in an area where rents are high and parking is impossible.

Why locate a studio there? Because when an advertising client parks and settles in for the day, the idea of going across town for a 2-hour session is not appealing. The game plan is to be as near to as many clients as possible.

Get a map of the city and identify prospective clients, competitors and related service facilities. Look for the area with the highest concentration of potential business. Walk the neighborhood, get a feel for the parking situation and look for freeway access, restaurants and other amenities.

Contact a real-estate broker and carefully evaluate all possible locations for the new facility. Examine square footage, ceiling heights, air conditioning, location of support pillars, elevator access, stairwells, proximity to restrooms, thickness of foundations slabs, electrical fixtures and the area's appearance.

Once the space is selected, negotiate a lease that includes free rent during construction, and an escalating rent that is favorable for the first two years of operation. Also, negotiate a non-disturbance clause. If the building is sold during your lease period, the new owner must honor all conditions of the contract.

The design phase should be enjoyable. If you've been in business a while, you will already know how you want the space laid out from an ergonomic viewpoint, but an acoustical consultant is advisable. Not only will you avoid costly errors, but your new facility will have some added credibility.

Allow a reasonable length of time for final drawings. Under close scrutiny,

some very exciting ideas may not hold up. Blueprints are reasonably inexpensive; walls are not.

Your biggest obstacle will be the city permit/planning office. Try to show the minimum amount of detail in your drawing plans, especially regarding the finish work. The reasoning here is obvious: The more exotic the drawings, the more questions will be raised by the building department. (An example: Instead of designating the areas as "control room," "studio," etc., just refer to room No. 1 or room No. 2.)

Make sure that your building contractor knows how to talk to inspectors and fully understands your ultimate requirements of mechanical isolation.

Studio construction is very specialized, especially in the mechanical phase. Most cities have an energy compliance office which will first ask: "How do you justify double the air conditioning of a typical office space?" Next comes: "Why do you need independent air handlers for each room?" These are tough questions. You may have to contract a mechanical engineering company to make the calculations necessary to account for your additional needs.

Another problem, especially where floated floors and raised platforms are employed, is handicapped access. Ramps require a large amount of space and regulations can be very strict.

Sprinkler systems are another source of aggravation. Not only do they look bad but, because studios have so many irregular shapes and odd angles, the sprinkler system cannot be arranged in an economical grid pattern. At a cost of approximately \$120-\$175 per installed head (assuming the basic supply system already exists), the extra heads represent a significant cost. Remember, those speaker soffits that looked great on paper will require additional sprinkler heads. In many cities, enclosed cavities of a certain size—such as speaker enclosures, spaces between double glass and trap areas—will require a separate sprinkler head.

The air space above the ceilings may also require sprinklers. If you build with metal studs and the building has concrete floors, this may not be an issue. But, if the building has exposed wood in any cavity, you'll need sprinklers. Don't

be surprised by this major source of frustration. Fire codes are very demanding, especially in high-rise buildings.

Earlier, it was advised to keep the drawings as simple as possible to avoid complications with the planning and permit people. Often it is easier to get a variance from the job site inspector when something exotic is planned. Some examples might be the interior trap areas; the use of flex instead of rigid conduit between wall systems; and cable runs through fire walls.

Make sure that your contractor knows how to talk to inspectors and fully understands your ultimate requirements of mechanical isolation. A good relationship between the contractor and the inspector is essential. Many studio construction techniques are in the gray area of the building codes. An inspector's interpretation of the code will impact the cost and ultimate success of your project.

During construction, take the opportunity to premarket your new facility. Size up potential clients and even consider tours of the job site. Start a direct mail campaign, but be careful about your opening date. Construction delays are common and can be a source of great embarrassment. Premarketing can also include calling existing clients and involving them with the project. You are responsible for creating excitement in the client community.

Document and photograph each step of the construction process. This will be valuable for published articles about your facility. Trade magazines are supported by hardware advertisers and you might strike a deal for a co-op ad showing some new equipment being installed. Magazines enjoy editorial about expansion and new facilities, so don't be shy in publicizing your efforts.

The most important issue is the market. Is there a real demand for your services? Many studio owners make the mistake of building and *then* marketing. Scope out the scene, zero in on what is in demand and then create the facility. Don't build a studio that is beyond the budgets of your prospective clientele. Do some market analysis and poll clients regarding their needs and their budgets. Purchase equipment accordingly and keep the business in proper perspective.

There are countless horror stories about studio construction. As a SPARS member, there is valuable information available regarding the hazards. Have fun and don't forget to double your construction and budget estimates! **REP**

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



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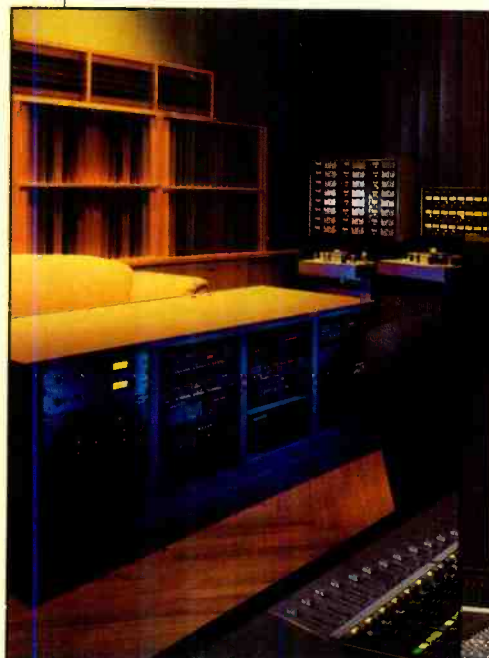
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Acoustic Design and Monitoring Requirements

for Recording and Production Facilities



The control room of Granny's House Recording Studios, Reno, NV, features a Live-End/Dead-End design implemented by the author, Chips Davis. Recording equipment includes a 48-input Solid State Logic SL6000E console with Total Recall and Studio Computer automation, Studer A800 MkIII and A80 analog 24-tracks, Studer A-820 analog 2-tracks (with interchangeable 1/4- and 1/2-inch headblocks), UREI model 813 Time Align monitors powered by Crown Delta-Omega amplification Yamaha NS-10M and Auratone 4C Sound Cube close-field monitors.



By Chips Davis

Studio operators contemplating the construction of a new facility, or the redesign of an existing environment, should be aware of the basic ground rules of acoustic design and monitoring systems. Of particular importance is the "room/monitor" interface.

Have you ever heard a broadcast commercial sound better than the programming that surrounds it? Live music on television that made the 3-inch speaker come alive? A stereo mix that seemed to have front-to-back, up-and-down dimensionality, as well as left and right? Can you detect absolute polarity? Have you ever heard a tape or disc that seemed to sound fantastic everywhere you played it, regardless of the price or vintage of the replay system?

Did that commercial *really* sound that much better? Can that 3-inch speaker really put out decent sound? What is absolute polarity, and can you hear it? Can you use absolute polarity as an effect? Can you mix left/right, front/back and up/down, with some surround effect, in stereo on just two speakers? Do some mixes really translate?

If you're planning to build or rebuild a studio or control room, the answers to these questions will dictate the form, function and eventual value and utility of your new facility.

New technologies and understandings of the physics of recording and reproducing sound are forcing us to rethink almost everything we've come to believe about audio, acoustics and studio/control room design.

New technologies make new demands. We're at the dawn of the digital era. Digital sounds good now; it'll sound even better in the future.

Video is now a major part of the creative and revenue base of a rapidly growing number of studios, both commercial and personal-use. There is no mastering engineer to get you out of trouble in stereo television and video post session—it's home delivery!

The decisions regarding what form your new studio will take are choices in bricks and mortar—choices you'll have to live with for a very long time. Nowhere is the edict "form follows function" more true than in the rapidly changing field of studio/control room acoustic design.

Critical environments

The better the technical quality of the hardware you choose, the more critical the acoustic environment in which that hardware must live. The quieter the electronics, the wider the dynamic range; the more sensitive the microphone, the greater the need for a quiet, competent, non-fatiguing acoustic environment.

New methods of sound generation, recording, storage and signal processing are pushing us toward mathematical

Chips Davis is president of Chips Davis, LEDE Designs, based in Las Vegas, NV and San Rafael, CA.

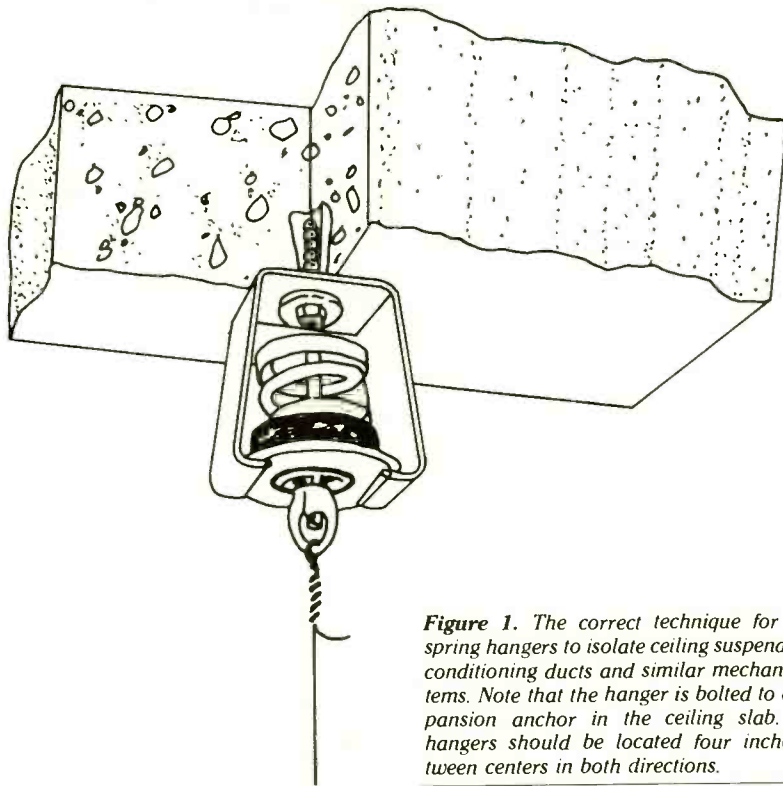


Figure 1. The correct technique for using spring hangers to isolate ceiling suspended air conditioning ducts and similar mechanic systems. Note that the hanger is bolted to an expansion anchor in the ceiling slab. Such hangers should be located four inches between centers in both directions.

limits of dynamic range and signal-to-noise ratios. In a few years, mastering to a dynamic range exceeding 110dB will be commonplace.

Clients demand that we studio owners change equipment almost as often as we change socks; physical facilities, bricks and mortar, however, are much more permanent. You share a responsibility with the studio designer to make sure that your new "dream room" will not become a nightmare the day you run the first session.

It's up to you, the studio owner and/or user, to be your own consumer protection agency. Generally speaking the business of studio design has been more a product of imagination than of science. Before you invest your money it's essential that you invest your time, if for no other reason than to separate the facts from the fantasies.

To survive in the studio market you need to evaluate honestly, objectively and realistically your real budget and actual needs. You must examine your ego, your market and your money.

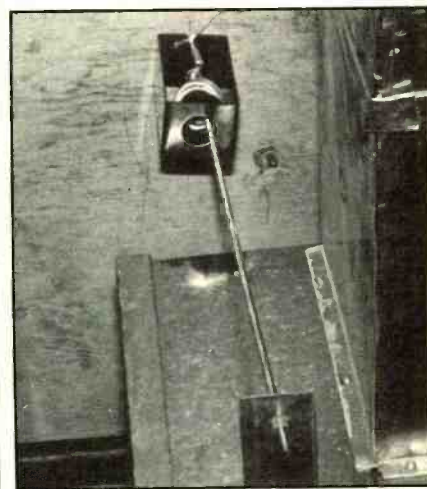
Market needs

You've got to determine whether you really need to try for the absolute perfection of a brand new "world class" (what a misused term) complex built from the ground up. Can, and more importantly will, your market pay for that kind of quality? Are your needs better served rebuilding, or just retrofitting an existing

facility. What are your *real* needs? How are you going to pay for this?

Does it make sense to install a few hundred thousand dollars worth of consoles and recorders in a control room designed to be no more than an equipment box? (I'm sure that you've seen that happen more than you care to recall.)

Figure 2. How *not* to mount spring hangers. Note that the suspension rod connected to the suspended air-conditioning ductwork is tilted from the perpendicular, with the result that it now touches the ceiling-mounted hanger. Consider the additional expense of having to rehang the entire HVAC system, as happened to this studio, to provide adequate sound isolation.



Once you've ascertained the market needs, you'll need to access one or more of the excellent books on basic acoustics and studio design. (A recommended reading list has been included at the end of this article.)

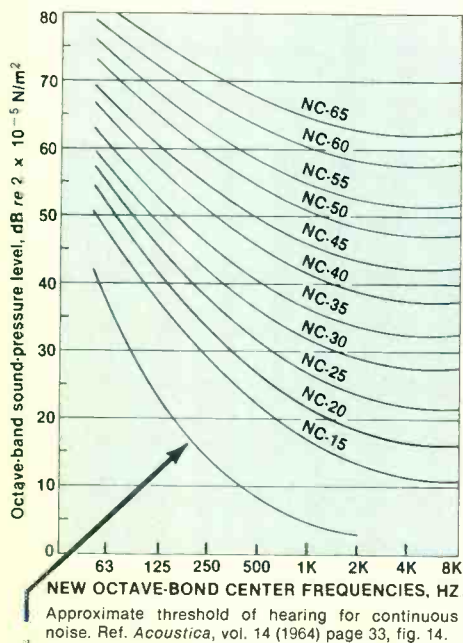
You need to establish a list of criteria and stick to it.

Taking as gospel the advice of the guy down the block who built his own studio is suicide, unless he is a reputable acoustical engineer or designer with a proven track record of success.

You have to examine your own attitudes about sound and recording before you move forward. Even though you make your living from achieving a great sound from just about any room, you've got to temper the "I can get a good sound..." attitude and start demanding better acoustic technology. Sure, you can get a good sound out of almost any room, but is the cost in effort, uncertainty and ear fatigue worth it? Wouldn't you like to take the words "trust me" out of your vocabulary?

It costs no more for a studio designer to do it right the first time. As a matter of fact, doing it right the first time usually costs less. A well-conceived design, whether for ground-up construction or a weekend redo, can save far more than the design fees in construction change orders alone. There is little room for experimentation with a crew of union craftsmen on the job site. Doing it right just takes a little more effort on your part.

Figure 3. Standard Noise Curves from NC-15 through NC-65.



Noise considerations

A quiet recording environment requires that you, the designer and the construction crews pay infinite attention to detail. An improperly isolated conduit, a coupled wall system, an air lead or a shorted HVAC isolator can negate a couple hundred thousand dollars worth of construction in a minute (Figure 1 and 2).

The simple rule is: Where air goes, sound follows; studios and control rooms must be airtight. I've measured major control rooms where the subaudible air handler noise alone was over 100dB at 15Hz. Although you can't hear it, you sure know the noise is there. Ear fatigue and physical stress in such a room is debilitating. Blowing the isolation ballgame for lack of attention to detail can ruin your whole day.

On the subject of isolation and noise, it's important to understand that people like to respond to very complex questions with very simple answers. Be aware that single-number answers can be very misleading and consequently very costly. Be cautious of the fast answer, the uncalculated, off-the-cuff response. Acoustics is a science, not magic.

Place a 100dB sound source on one side of a wall system with a Sound Transmission Coefficient (STC) of 56, and you'll measure an attenuation of approximately 60dB at 1kHz on the other side.

Shift the frequency of that same 100dB signal down to 125Hz, and your attenuation drops to 35dB; at 63Hz the attenuation through the same wall will be less than 20dB.

Locating a recording space designed for a Noise Curve (NC) of 15 (very quiet room) next to a control room wall with an STC of 56, will result in an actual NC (the ambient noise leaking from one space to the other) of 65 at 63Hz. At 1kHz the NC becomes 40—which will be very noisy. Figure 3 provides NC curves from NC-15 through NC-65.

As if all of this wasn't bad enough, you've also got to consider the consequences of the methods used to achieve adequate sound isolation. The only effective way to achieve efficient isolation is through the use of mass and airspace. Mass and airspace contain sound without the uncertain and sometimes negative side effects of less efficient methods of isolation.

Lightweight wall and window systems absorb sound by diaphragmatic action: they resonate at frequencies complementary to their mass, panel size, depth and composition. Resonating panels can act as very narrow-band, acoustic filters, subtracting their complementary frequencies from the audible signal within the room.

If you place a lightweight wall or window system near a loudspeaker—in a soffit, ceiling or side wall, for example—the speaker can drive that wall, causing both attenuation at the wall or window's resonant frequency and a serious potential for a sustained ringing at both sides of the resonance frequency.

Frequency response anomalies

The end result is a wall system that has the unfortunate potential of acting both as panel absorber and re-resonator—a disastrous situation if uniform frequency response is your goal.

What happens if you physically attach or couple the monitor loudspeakers to a resonant, lightweight wall? The speaker drives the wall and introduces still yet another negative effect called "Early, Early Sound," which turns the entire coupled shell into a gigantic sound board.

Some designs actually use this effect, believing the result to be "bottom-end punch." The coupling creates an artificial, inaccurate and exaggerated bass response in the room, thereby making mix decisions very difficult, at best. Designers then compound the error with negative peak or compression ceilings, doubling the bass wave back upon itself. This reinforces the bass wave at some frequencies while cancelling it at others. The net result is a booming, inaccurate bottom-end response.

Low-frequency response

In small-room acoustics—control rooms are considered "small" acoustically—the first design consideration for bass is modal spacing: the natural resonance of a space is defined by its height, width and length ratios. (Just as the length and diameter of an organ pipe, for example, determines its natural resonant frequency, the height, width and length ratio of a room determines its resonance modes.)

Remember the last time you sang in the shower? Remember how some notes seemed to fill the space with rich, long-lasting tone while others notes wimped out and evaporated? The notes that made you sound like a basso profundo were those in natural resonance with the room modes. They were on the nodes, and were reinforced, while the notes that corresponded to the nulls are diminished. These natural resonances or modes are referred to as Eigentones.

Room dimensions that are multiples of one another cause Eigentones to become additive. This creates a dominant resonance in the room that sustains complementary bass notes at the expense of all others.

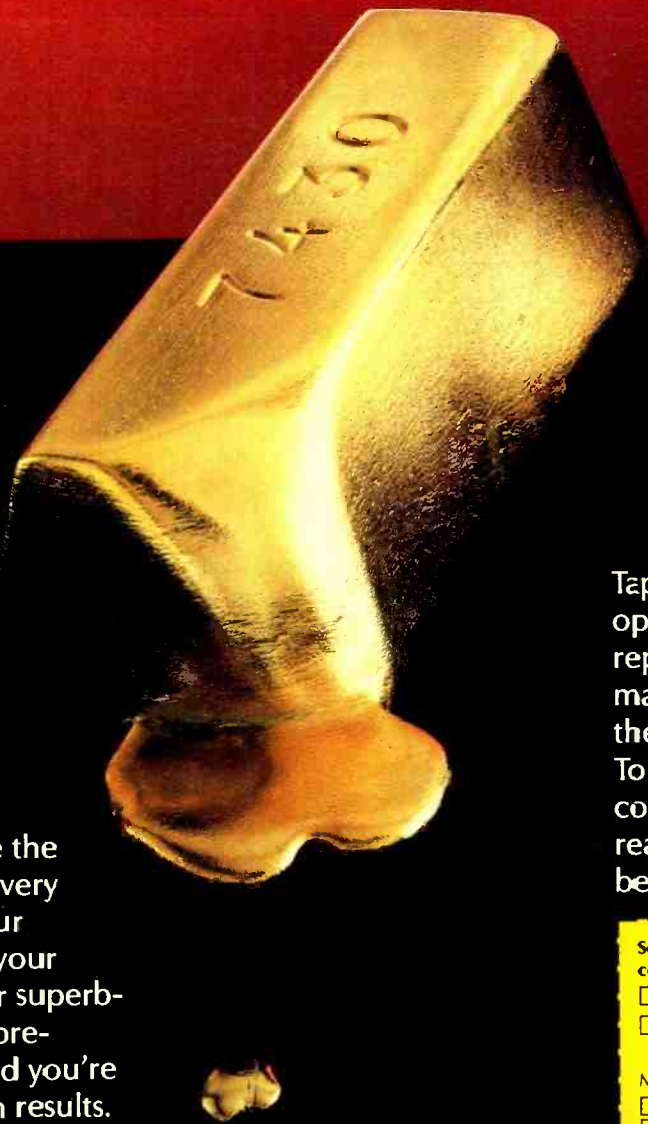
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Facility Spotlight: Craig Huxley's The Enterprise Studio

By Adrian Zarin

In these days of an increasing emphasis being placed on electronic music scoring and production for film, television and conventional sessions, what were the primary decisions involved in the design, construction and equipping of a 3-room multifunction complex owned by leading musician/composer Craig Huxley?



In the late Eighties, electronic instrumentation is the common denominator that links music production for film and television with the production of album and single releases. Meeting the demands of both music genres on an equal footing would seem essential for any new recording facility.

Craig Huxley is a synthesist/composer with equal experience on both sides of the production fence. He has played on albums by Michael Jackson, Neil Diamond, Roberta Flack and Frank Sinatra, among others, and has provided music and sound effects for such films as *Star Trek I, II and III*, *Short Circuit*, *2010: A Space Odyssey*, *Cat People*, *Blue Lagoon* and *Firefox*. During the planning stages of his new multiroom studio, The Enterprise, Huxley evolved a number of design ideas for a facility that would accom-

Studio A's 4-tiered control room features a 56-input customized Amek model 3500 console with Audio Kinetics MasterMix VCA-based automation, Otari MTR-90 24- and MTR-12 analog 2-tracks (a Mitsubishi PD-format digital 32-track is also available for use throughout the 4-room complex), Quedest Q412 and UREI 813 Time Align monitors. Also seen here are a selection of the facility's large collection of synthesizers and sequencers, including a New England Digital Synclavier DMS PolyTower system. To the right of the control room is Studio A's live recording area.

Adrian Zarin is a Los Angeles-based electronic synthesist, composer and free-lance writer, and a regular contributor to **REP**.

moderate both record session work and audio production for film/television, yet without compromises in either area. A pastel-colored building tucked into an otherwise drab Burbank thoroughfare, the studio represents the realization of Huxley's concepts.

"Throughout the Eighties," he recalls, "I was traveling around doing sessions—often at the original Westlake facility [Hollywood] with Quincy Jones—where we would have a stack of modular synthesizers and outboard gear jammed all over the control room. I began to dream of building a studio that would incorporate the spaciousness and high-ceiling feeling of a film re-recording stage, with all the features of a rock recording studio.

"I envisioned a large, open control room, where the [synthesizer] programmer and producer would be in the same stereo or surround-sound field, as well as the engineer and second engineer working at the board. This led to the design philosophy behind *The Enterprise*."

When Huxley opened *The Enterprise* in early 1986, it was essentially a 1-studio facility. The original studio (renamed Studio C in the present *Enterprise* complex) featured a 27' x 34' recording area and a large (20' x 25') control room, laid out in three descending tiers (from back to front). It is a design, he concedes, that deliberately resembles that of a theater.

Another important element of the original and present facility is a synthesizer programming/pre-production room, Studio S, equipped with a 24-input Amek Scorpion console, Tannoy Little Red Visonic 9000 and JBL model 4311 monitors, a Yamaha TX-816 rack, Roland super Jupiter and Opcode Systems sequencing/librarian software for the Apple MacIntosh Plus.

A 64-voice New England Digital Synclavier DMS PolyTower digital sampling/sequencing system is also available for use in either Studio C or Studio S, depending on the needs of specific projects.

The *Enterprise* complex, decorated in Italy's chic "Memphis" style by designer Cameron Ashton, also houses accounting offices, an audio equipment rental/sales operation, IA film scoring company, Alivity Productions, and the headquarters of Huxley's record label, Sonic Atmospheres.

Recently, Huxley completed construction on two additional studios at *The Enterprise*. Studio A features a slightly larger (25' x 30') control room than Studio C, linked to a 25' x 27' recording area. Construction and adaptation of architect Jeff Cooper's original design for Studio C fell to contractor Ernie Ramirez and Michael Stocker, *The Enterprise*'s current director of engineering. Rounding out the facility is another new studio,

Studio B, which comprises a smaller (18' x 25') two-tiered version of the design principles embodied in the Studio A control room that was completed in March of this year.

Each of the new studios is equipped with an NED Synclavier PolyTower, identical to the system used in Studio C, and a selection of rack-mount synthesizer modules. This arrangement enables each of the studios to operate as a self-sufficient electronic-music command center. At the same time, however, each studio has its own unique characteristics and functions.

"Studio A," Huxley observes, "has the biggest recording room, and is certainly the most suitable for live recording. We've already had the entire orchestra from *Dallas* in there. Studio B will be a special mix room suitable for overdubs as well. And Studio C is ideal for smaller bands, plus mixing and overdub work."

All of the three main studios are also

linked via audio and video tie lines. Huxley places great emphasis on the capability of each of the facility's rooms to work together in a synergistic manner.

"Clients can work—as some are already doing—in two different rooms at once," he elaborates. "We have many different phases of the recording process going on simultaneously. The pre-production room [Studio S] has been very helpful to clients as well, allowing them to get into the flow of being here and working with some of the equipment—it's useful in helping them clarify their ideas."

Physical design: Studio A

As mentioned, one of Huxley's chief goals in planning each of the studio's three main control rooms was to create a spacious area suitable for synthesizer-based sessions, and to provide an acoustically consistent environment. The design of Studio A's control room is based, to a large extent, on that of Studio

Composer and *The Enterprise* owner Craig Huxley in Studio A's live recording area. He is pictured here with his patented invention Blaster Beam, an "instrument" used to produce sound effects for film scores.





Studio C's 3-tiered control room features a 60-input customized Amek model 2500 console with Audio Kinetics MasterMix automation, a custom Keith Klawitter monitoring system, Electro-Voice Sentry and Yamaha NS-10M close-field monitors, an Otari MTR-90 24-track, an overhead video projection system and screen, plus various synthesizers and effects. To the right can be seen the entrance to a dedicated vocal and instrumental overdub booth.

C. In implementing Huxley and Cooper's multitiered design in Studio A, there were a number of acoustical problems to be worked out. According to Michael Stocker, the facility's director of engineering, one of these problems concerned the control room's side walls.

"The walls are parallel, which created some initial [standing wave] difficulties. At one point, we were discussing slanting the tops of the walls 7° on each side. But, instead, we solved the problem by installing a pair of high-mass, low-end diffusers on either wall. As a result, we now have a lot of dispersion in the room."

The side walls are constructed of triple-layered drywall, covered by a layer of Owens-Corning 703 absorptive surfacing. A wood frame separates this layer from the wall's outermost, hardwood surface. The front wall is constructed of triple-layered drywall with an outer layer of 703.

The rear wall, however, is divided horizontally into two areas. The upper half is constructed of triple-layered drywall, a layer of 703 and a hardwood surface built onto a wood frame, an identical sandwich to that used on the side walls. The control room's tape-machine soffits are built into the lower-half of the rear wall. This section is constructed of triple-layered drywall with 703 on the outside, thus minimizing machine noise as much as possible.

The control room's tiered floor imposed another set of acoustical considerations. The lowest surface is a large hardwood floor that occupies the front

end of the room (the end nearest the control room window, which looks out onto Studio A's recording area). The console rests on this hardwood surface, and carpeted tiers begin to rise behind the mix position.

"In the first room we built [Studio C], we learned the importance of dampening the tiers with bass absorptive material," Huxley says of the design.

Accordingly, the front floor and the first carpeted tier are both filled with absorptive polyurethane foam.

"The main monitor speakers are not tightly coupled to the structure," Stocker explains. "Because they're only sitting on a single rubber mat, in the whole front area all vibration from about 15Hz and up is dampened. This way, we don't get a lot of dissipation before the sound actually hits the listening position.

"In the rear, the floors are suspended, and they have several layers of R-11 insulation inside. It's fairly well-damped back there and there are no resonances as such."

The control room ceiling in Studio A is also tiered. Ceiling tiers are baffled and constructed of drywall and plywood, with an outer layer of Owens-Corning 703.

"In the initial blueprint, the ceiling tiers corresponded exactly with the floor tiers," Stocker says. "I was a little worried about that, because I've had a lot of problems with similar areas—balconies, for example—in sound reinforcement applications. We ended up offsetting the ceiling tiers, moving them a little forward of the floor tiers, so as to ran-

domize any of the wavefronts that would be created by that situation."

Studios A and C are both outfitted with their own machine rooms, located to the rear of each control room. In each studio, a large window provides visual contact between the dedicated machine room and the corresponding control room. Each machine room has its own discrete air-conditioning system; main power for all the audio equipment located in each room comes from a separate source, derived from The Enterprise's own 2.4kV transformer.

The machine rooms house the studio's multitrack machines and mainframe components of the Synclavier PolyTower. The system's master keyboard, main terminal and floppy disc drive reside in the control room.

"Having a machine room creates a sense of spaciousness in the control room," Huxley observes. "But often, people like to have their 2-track and 4-track machines in the control room, so they can do tape editing. We therefore have set things up so that the client can locate those machines in either the control room or the machine room."

As mentioned, Studio A features the facility's largest recording area. In Huxley's view, the 25' x 27' room provides a very serviceable backup to the electronic, direct-inject tracking facilities that comprise The Enterprise's main emphasis.

"The Studio A live room is not immense," he concedes. "But, because of its 20-foot ceiling, the volume of the room is significant. Anyhow, live recording tends to be done with smaller ensembles these days—when it is done at all!"

The room features a hardwood floor and ceiling cap. The walls are surfaced with 703 (atop two layers of drywall). On two of the walls, the 703 is covered with ¾-inch louvered oak baffles.

"The room is oblong in shape," notes Stocker. "One wall runs at about a 12° angle, which breaks up the possibility of standing waves."

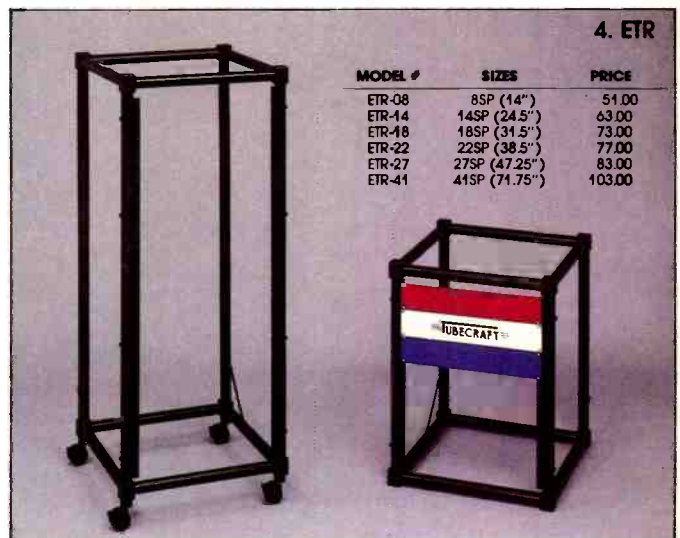
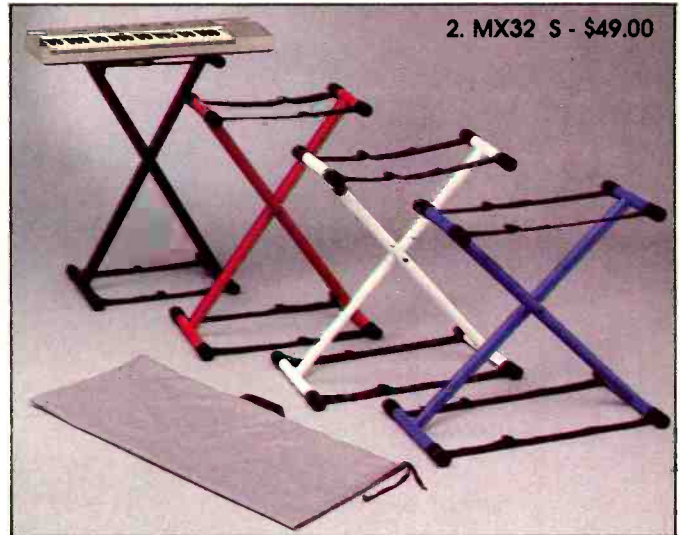
Physical design: Studio B

In many respects, Studio B's control room is a scaled-down version of the Studio A control room. Measuring 18' x 25', it is a 2-tiered rather than a 4-tiered room. The acoustic surfacing and treatment, however, is virtually identical to that of Studio A. A small, Foley-oriented iso booth is located in the rear of the room, to the right of a person seated in the mix box position. A full length, sliding glass door provides full visual contact with the control room and its video projection screen.

The control room's machine soffit is located to the left of the iso booth.

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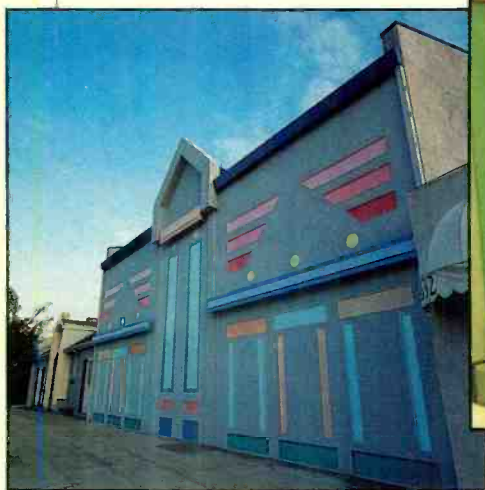
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Exterior of Craig Huxley's The Enterprise recording and post-production studio.



Entrance hall and reception area of The Enterprise studio complex, decorated in Italy's chic "Memphis" style by designer Cameron Ashton.



Off the main entrance lobby are located several offices for Huxley's affiliated companies, also decorated in the Memphis style.

patched into the Studio A control room," Stocker says, "it does not have a machine room of its own. As a result, it was important to have the machine soffit in Studio B absorb as much noise as possible. I put absorptive baffling inside the soffit, above the machines, and I'll probably put some sort of gobo in front of the machines by the time the room is completed."

On the right wall of the control room, another sliding glass door provides access to a 16'x12'x12' hardwood-surfaced, multipurpose recording area. The room, Stocker explains, can function as a

dedicated overdub booth for mixing and overdub projects undertaken in Studio B.

The facility's system of audio and video tie lines, however, makes it just as feasible for the room to act as a drum booth or secondary iso booth for a tracking date in Studios A or C. Conversely, Studio B's control room can also be used for tracking dates in one of the larger recording rooms.

Monitoring systems

Studio C is equipped with a Keith Klawitter monitoring system powered by Audire, Otez and Forte amplifiers. A total of five enclosures—handling left, right and two surround channels—each consist of six 12-inch drivers, four 3-inch dome midrange components and four ¾-inch dome tweeters. The absence of horn-coupled components was one aspect that attracted Huxley to the Klawitter system when he built Studio C a year ago.

"I never liked the throat distortion that comes from a horn," he offers, "nor do a lot of my clients. In many cases a lot of them just end up listening on [Yamaha] NS-10Ms. But here, many of them find themselves staying with the large monitors."

Because he was pleased with this type of dome-based monitor system, Huxley went with a similar, non-horn system when the time came to build Studios A and B.

For the two newer control rooms, he selected a Qusted system, consisting of Q412 left, center and right boxes, and two UREI model 813 surround enclosures. Each Qusted Q412 enclosure contains four 12-inch low-frequency components, a 3-inch midrange dome and a 1-inch tweeter. The system is tri-amped, each enclosure being powered by Yamaha model 5002, model 2002 and model 1001 amps.

Each control room also includes Yam-

aha NS-10Ms as close-field reference monitors.

"But, a lot of our clients carry their own reference monitors," Stocker adds, "so I have two amplifiers mounted under each console. They're available specifically for two alternate monitor sources."

Recording chain

The Enterprise is equipped with customized Amek consoles throughout the facility. Studio A features a 56-in/32-bus model 3500, Studio B has a 48/48 model 2500 and Studio C features a 60-in/48-out model 2500.

"The first console I started out with was an Amek," Huxley notes. "I was always pleased with the support I received from the company and with the sound quality. Once I started to build The Enterprise, I decided to stick with Amek, as long as we could customize them."

Among the custom features Huxley had added to the Amek consoles are quad monitoring, enabling the board to accommodate center and surround tracks, and eight effects sends (as opposed to the standard six). In addition, Studio C's model 2500 is equipped with stereo input modules on channels 37 through 48. All consoles are equipped with Audio Kinetics MasterMix disk-based fader automation for channel and master send faders. Studio A's console is also slated to receive a MasterMix automation package for its eight group buses.

The console configuration is said to enable each studio to handle a variety of multi-channel mixdown formats for film and television projects.

"We do a lot of film and TV projects that are mixed down directly to one of our [Otari MTR-90 MkII 1-inch] 8-tracks," Huxley says. "We record six tracks [plus time code]: three stereo pairs containing music, dialogue and effects. Music for *Crime Story*, *Dallas* and *Knots' Landing* is all done that way."

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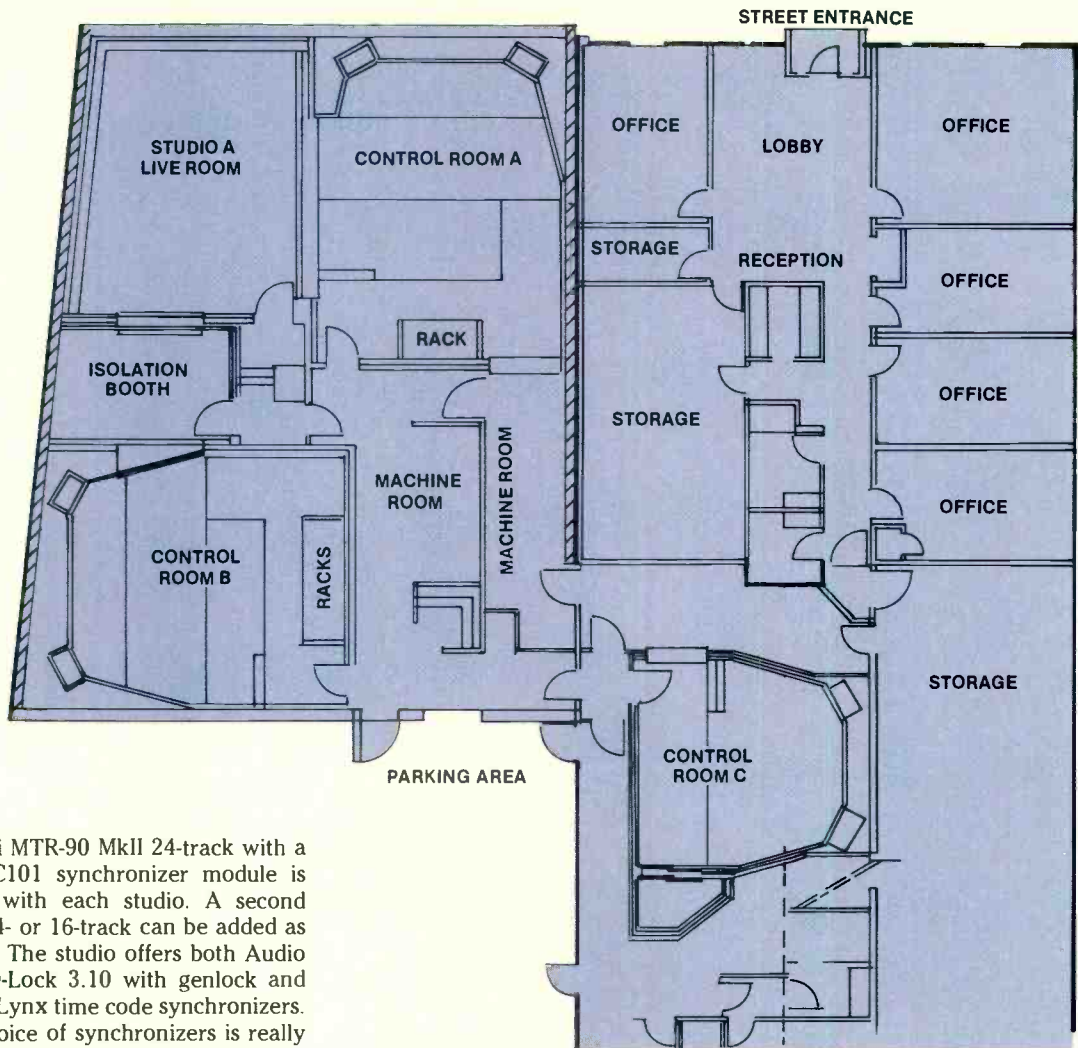
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The floor plan of The Enterprise showing the three major control rooms.

An Otari MTR-90 MkII 24-track with a built-in EC101 synchronizer module is furnished with each studio. A second MTR-90 24- or 16-track can be added as an option. The studio offers both Audio Kinetics Q-Lock 3.10 with genlock and TimeLine Lynx time code synchronizers.

"The choice of synchronizers is really up to the clients," Huxley notes. "They can use whatever they're most comfortable with. Sometimes we have as many as five machines synchronized on a given project, so we have a large number of synchronizers on hand."

As an alternative multitrack option, clients can also rent one of the studio's Mitsubishi PD-format X-850 digital 32-tracks. The studio offers a similar range of analog and digital choices for mixdown. An Otari MTR-12 or MTR-20 2-track or 4-track is furnished with each room. In addition, the studio is also awaiting delivery on three Mitsubishi X86 digital 2-tracks.

When it came to stocking The Enterprise with outboard gear, Huxley says that his thinking was shaped by his own musical activities.

"There's one thing in my experience with New Age and experimental music that predates what has become more and more common in the pop world and even the film world. And that's the simultaneous use of multiple ambiances: the phenomenon of creating a *distinct* placement for each group of instruments—one in a closet, one in a hall and another in a canyon, for example. These

more subtle and diffuse approaches to creating ambiances require a variety of delays and reverb units."

Accordingly, each studio is equipped with a large selection of outboard signal processors, including a choice of AMS RMX-16 or Lexicon 224XL digital reverbs, a Roland SRV-2000 reverb and SDE-3000 delay, Yamaha SPX-90, Aphex II Aural Exciter, a dbx series 900 rack and a 4-channel rack of Valley People Kepex II noise gates. These units can be supplemented from a pool of optional equipment that includes a Publison Infernal Machine 90, EMT 250, AMS DMX-1580S, Yamaha REV-7, Quantec QRS, Lexicon 480L, Drawmer noise gates, Pultec and API EQ, UREI LA2A, Fairchild 670 and Neve limiters, and a Dolby SR noise-reduction rack.

Video chain

All audio work that involves synchronization to picture is handled on ¾-inch U-matic.

"I've been doing it that way since

1979," says Huxley of his own experiences as a TV and film composer and synthesist. "Now days, most TV shows are shot on film and transferred to 1-inch video [from which a ¾-inch work print dub can be made for audio sweetening sessions].

"For me, one of the biggest advantages of working to video is that you can revise very quickly. You get the latest cut instantly; and it's in color, which is nicer than working to a black-and-white film dupe."

Each studio is equipped with a JVC model 8250 U-matic VCR. A Cinebeam projection system with RGB inputs projects a picture with a 10'×6' aspect ratio onto a large screen located at the front of each control room. A separate video monitor can be provided for conductors working in one of the live rooms.

"Again, while you're working in the control room, you have the sense of being in a theater," Huxley comments. "I used the studio for the first time when I was working on the score for *2010*. And

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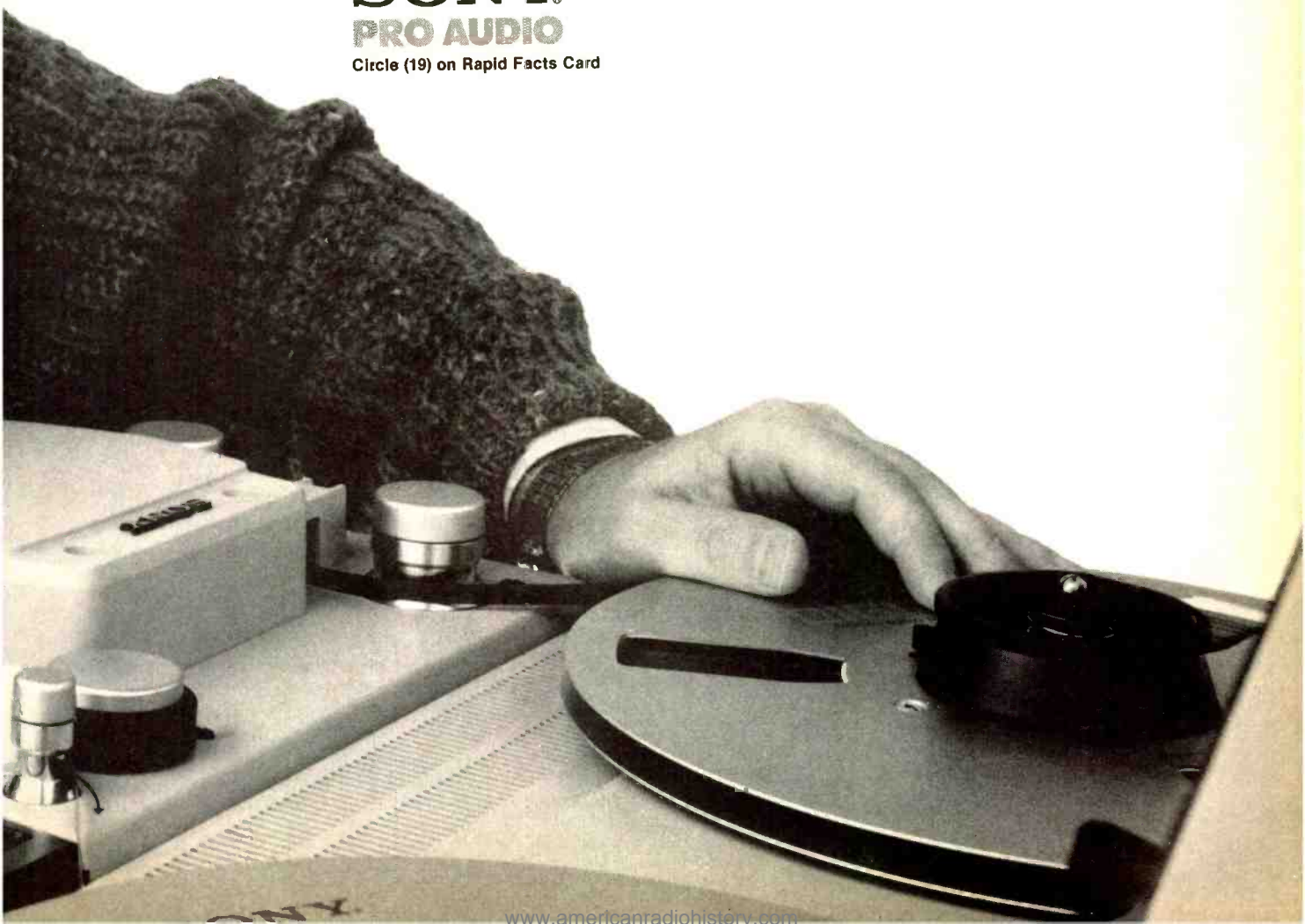
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Designing Customized Facilities: The Acoustician's Role

By John Eargle

The use of an accredited acoustician, experienced in the analysis of sound isolation, noise control and acoustic treatment, can save problems during the design and construction of a variety of recording and production studio areas.

All too often, studio owners contemplating modifications to an existing facility do not avail themselves to the full range of acoustical advice needed to ensure a first-rate result. Assuming that there are valid economic incentives to build or remodel the studio in the first place, it only makes sense to go the whole route and hire a competent acoustical consultant.

This article will provide an overview of what is required, in an attempt to convince the prospective studio builder that, while the job is not impossible, it is not simple enough to do alone.

There are two major areas of acoustics involved in the layout of any studio. First, there is the isolation of sound from the studio's surroundings, and noise control in general. The second area is concerned with the use of specific acoustical treatment in the studio or control room to match its intended purpose.

First, I'll describe briefly the scope of noise control and treatment, outlining some of the points that will be addressed by an acoustical consultant, and then move on to consider the acoustical goals for specific customized designs.

Isolation and noise control

In the areas of isolation and noise control, there are four major areas of concern:

- Airborne noise transmission into (or out of) the studio.
- The effect of impact noise (motors, plumbing and footfalls) carried by the structure itself.
- Locally generated air-handling noises.
- Undesirable coupling between adjacent operational areas.

Because of the vast database that supports the subject of airborne noise

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



The Designer's Checklist

- Look for uncaulked openings in a masonry wall, or openings in a wall for electrical outlets; both can provide effective sound leakage paths.
- Specify a sound lock door with a measured transmission loss equal to that of the surrounding wall.
- Watch for construction debris within a double wall between adjacent studios, and which material provides an effective sound transmission path.
- Reroute water pipes and fixtures as far from the studio area as possible to prevent plumbing noises.
- Air-conditioning ducts should be of large diameter to prevent excessive air turbulence.
- Watch for sound transmission through ac ducts if a single cooling plant is serving two recording areas.
- Use office, storage and maintenance shops to provide sound locks between recording areas.

transmission, this is about the only area that yields to prior analysis and specifications. (Airborne noise transmission is discussed in an accompanying sidebar.) What the acoustician is always looking for are the little things, many of them shortcuts in construction, that can subvert his design. For example, slight, uncaulked openings around a masonry wall, or openings in the wall for electrical outlets, often defeat the wall's purpose by providing minute, but effective, sound leakage paths.

Similarly, a competent consultant would not specify a sound lock door hav-

ing a measured transmission loss that was not at least as good as the wall into which it is placed.

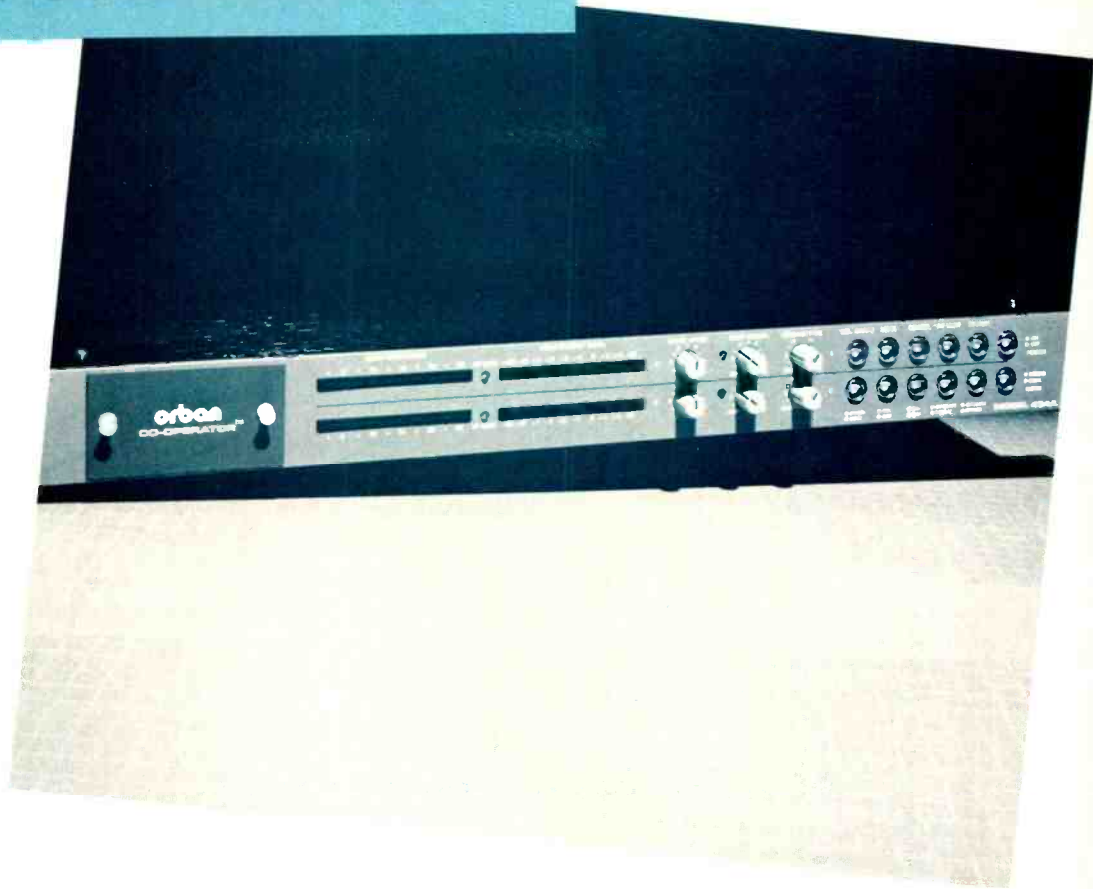
Stated generally, as the major sound leakage paths are blocked, then the secondary paths become significant; they *all* have to be kept under control. There were instances where adjacent studios were located on the same concrete slab. After all due attention had been paid to airborne sound transmission between the two rooms, coupling through the slab itself—as improbable as that may seem—became the limiting factor in isolation, necessitating the removal of a section of the slab.

In other cases, construction debris has piled up within a double-wall construction between adjacent remix rooms. Such material forms an effective transmission path between the two walls, virtually defeating their intended purpose. An experienced acoustical consultant has usually run into these or similar problems before, and will constantly be on the lookout for them again. Competent consultants will take nobody's word that something has been done, but will check for themselves.

Impact noise is a generic term that refers to noise coupled into the studio through the structure itself. High-pressure plumbing is usually the worst offender, and water pipes and fixtures should be routed and placed a couple of walls away from the studio. Air conditioning plants located on the roof are often troublesome, but the motors can be isolated using shock mounts. Elevators can pose problems as well. Obviously, bare floors above can be carpeted to kill the effects of footfalls. Any kind of construction work or hammering, often as far away as three or four floors, may be transmitted into the studio.

If the studio complex is to be located

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high, the studio builder may be willing to relax the NC requirements somewhat. The same kind of analysis is then made by the acoustician, and the new values of STC costed for comparison.

So far, we have assumed that the only requirements for noise isolation were from outside in. What about inside out? Assume for a moment that the studio will be used for rock and roll recording, and that there will be quite loud acoustical levels in the studio. Assume further that there are professional offices in an adjacent space and that a certain level of quiet must be maintained for the proper execution of their business.

The acoustician will then perform another set of calculations, assuming a maximum noise spectrum inside the studio and an acceptable NC requirement in the professional offices. He will subtract the two, arriving at new isolation requirements, which are then compared with the earlier requirements.

The highest value will be taken from the two sets of requirements.

These are the ones that will have to be specified to satisfy noise requirements in the professional office area.

Typical NC requirements for studio and post-production facilities are provided in Table 2.

The procedure outlined here is only one of the many disciplines of the acoustician. We have ignored matters of air handling and impact noise isolation; they are traditionally among the most difficult to estimate beforehand and to rectify once they have become apparent. Another difficult problem is the specification of multiple studios within the same basic structure. The prospective studio builder should always avail himself of experienced acoustical consultants, because such experts are likely to have encountered all these problem areas and have solved them.

Those readers interested in further coverage of this subject are referred to *Building a Recording Studio*, by Jeff Cooper: Recording Institute of America, New York, 1978.

Figure 2. Contours of Sound Transmission Class (STC). The dashed line represents the specific example quoted in the main article.

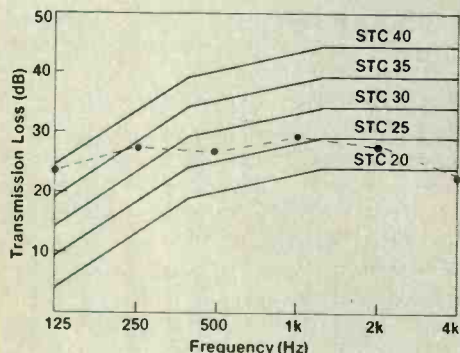


Table 2. Noise Criteria (NC) ranges for various activities.

Type of Facility	NC range
Recording and broadcast studios	15-20
Concert halls	20-25
Motion picture dubbing theaters	25-30

Sound Transmission Class 25 or 30 would be required if parallel activities are going on in the studio. (See accompanying sidebar on page 42-44 for further details of STC calculations.) However, many studio complexes have purpose-built overdub rooms, and these must be accorded the same degree of isolation given a studio. Again, the deader the better, with lots of bass trapping.

• *Announcer's booth.*

These booths are found in profusion in broadcast production studios, and in recording complexes that handle advertising voice-over work.

Many times, an announcer is recording

a voice-over to a previously recorded music track, and he will wear headphones. Other times, he is speaking without headphones. For this latter application, it is best if the room's acoustics are not extremely dead. There should be good damping of room modes to avoid coloration, but the walls should have sufficient reflection so that the announcer does not have the feeling that he is in an anechoic space, totally devoid of reflections.

• *Coupled studios.*

Some modern studio complexes have successfully arranged for adjacent studios to be linked for larger recording

sessions. Generally, strings would be located in one studio, with woodwinds, brass and percussion in the other. One simple way of handling larger sessions such as these is to use tie-lines and closed circuit video links between two completely separate studios. This is not truly satisfactory, however, in that video links do not provide the desired 2-way eye contact.

The use of double sets of double-glazed sliding doors between adjacent studios is not without its problems. However, triple- or quadruple-fixed glazing can approach conventional wall constructions in isolation properties, if properly implemented. The approach is expensive, no matter how it may be implemented. It is questionable, however, whether enough isolation is realistically possible, considering the expanse of glass required for proper eye contact.

Changing the ground rules slightly, a synthesizer pre-production or programming studio can be located next to a large multipurpose studio. Because the synthesizers are usually recorded direct-inject, any acoustical leakage into that space from the larger studio would not be a fundamental problem. The space could then be used as an adjunct to the main studio for such groups as softer string ensembles and vocal choruses.

• *Film dubbing studios.*

Most dubbing or re-recording studios resemble small (350 seat average) motion picture theaters. Because the aim is to generate a final multichannel film mix that will satisfy all requirements in the field, the acoustics are made to resemble an "ideal" motion-picture theater.

Reverberation time is kept short, normally less than 1s, and the console is generally located about two-thirds back in the house, where the mixer can hear the surround channel in proper relation with the behind-the-screen channels.

Ironically, the normally low ambient noise level in a good dubbing theater is often degraded electrically by adding a controlled noise spectrum to the room. This is done to simulate the poor noise conditions—the so-called "popcorn" noise—actually existing in commercial movie houses.

Otherwise, it is important for a good dubbing room to have excellent isolation from other working areas, and to have low noise levels when they are called for.

As will be readily appreciated, designing a studio for any purpose requires attention to details of noise isolation and prevention, as well as the actual acoustical treatment in the studio itself. Both considerations are of equal importance if a studio is to function as a successful commercial facility.

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Circle (22) on Rapid Facts Card

Electrical and Interface Systems for Recording and Production Studios

By Larry Winter and Terry Pennington

Noise and interference problems can easily be overcome by paying careful attention to the electrical wiring and equipment interconnect schemes.

Today's studio systems involve complex combinations of sophisticated signal processing and amplification equipment. Although each individual product may have superlative specifications, various interconnection methods can result in serious degradation of the overall system's noise and distortion performance.

The intent of this article is to provide a concise presentation of basic equipment interconnection guidelines, which will help to minimize ground-loop and airborne interference problems, and maximize interequipment compatibility. Note our use of the expression "guideline," instead of the word "rule." When it comes to grounding and interference, due to the complexity and unknown values of actual ground paths, there are precious few hard-and-fast rules that can be counted on to work in every situation.

Interactive vs. induced problems

For the purpose of clarity, let's divide hum, noise and interference phenomena into two categories according to their cause:

- *Interactive* hum and noise, caused by ground loops or gain mismatching between two or more pieces of equipment that are interacting with one another;
- *Induced* distortion, which comprises all of the airborne garbage (transformer hum, radio signals, dimmer buzz, etc.)

Larry Winter is vice president of marketing and Terry Pennington is director of technical marketing at Rane Corporation, Mountlake Terrace, WA.

that is induced into cabling or electronics through a number of means to be discussed later.

The number one cause of interactive hum and buzz problems is the notorious ground loop; that invisible gremlin that seems to defy all logic, evade all attempts of discovery and to tempt one to seriously consider exorcism. To understand the concept of a ground loop, one must first know the true nature of "ground." There's really no such thing (nothing is sacred anymore—first Santa Claus, now "ground").

Those schematic symbols you thought meant "ground" really mean: "This connects to a piece of conducting material of unknown length and impedance which, combined with some unknown number of other similar conductors, might eventually be linked in some fashion to 'earth ground,' which is not a perfect ground either."

The fact is that we have not yet developed the perfect conductor. Therefore, *all ground connection points have some amount of impedance or resistance between them.* Even though these resistances are very small, current flowing through them creates different voltages at each grounding point, none of them being zero.

A ground loop is a situation where two actual ground paths exist between two or more pieces of equipment, creating a circuit or "loop" through which current can flow between different potentials in the

ground system.

The result is that power-supply ripple (60Hz, 120Hz and/or 180Hz) and shield interference can bleed into the audio. In unbalanced systems, where shield ground is also signal ground, voltages developed through ground loops automatically become part of the signal. What makes ground loops occur so frequently and be so difficult to locate is the variety of grounding schemes that various manufacturers and electricians can use, and the possibilities of either redundant or missing ground paths when all this equipment is hooked together.

Single-point grounding

Because we now know the true resistive nature of grounding systems, we can identify some basic principles to help preclude the chances of creating loops in the grounds which, in turn, can cause noise. The primary guideline that pertains to both electrical *and* electronic systems is to connect all grounds to a *single* point whenever possible. This is otherwise known as the *Star* grounding configuration; it reduces the likelihood of redundant ground paths which, in reality, are loops.

In the studio, this means making sure that all electrical outlets have their neutral and grounding wires ultimately connected to the *same* ground bus in a single electrical panel. The "neutral" wire is the white-coated (by code) half of the ac line that returns current to the

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more ATM33R microphones in a demo studio, at no more investment than one expensive condenser.

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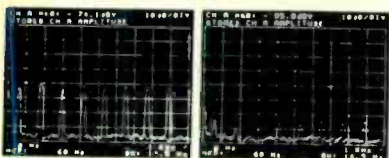
Cotton filler acts as strain relief and also reduces handling noise by preventing changes in stray capacitance.

Very high density, braided shield blocks most static and other noise.

Tough yet very flexible jacket can be unpacked from extreme cold and used immediately. The brittle point is -56°F (-49°C).

Available with a satin finish in 10 attractive colors to aid channel identification and/or to complement visual appearance. Fits standard XLR connectors.

You don't hear the fluorescent lights, motors, SCR dimmers, static, buzz, hum and handling noise with Canare L-4E6S. This shielded professional microphone cable is immune to electro magnetic noise due to its unique Star-Quad configuration. Compared to the leading 2-conductor microphone cable, Canare L-4E6S offers 10 times more rejection of the worst source of EMI...impulse noise from SCR dimmers. This cable blocks the noise, but not the program. Its low series resistance and low capacitance give L-4E6S extended frequency response, in mic runs of over 300 feet the 3 dB downpoint is at 50 kHz.



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electrical panel—it is one of the two current-carrying ac lines.

The "grounding" wire is the bare copper line that is *not* current-carrying, but rather the safety line that grounds the outlet and the appliance casing (those that use 3-prong line cords) to the elec-

The primary guideline that pertains to both electrical and electronic systems is to connect all grounds to a single point.

trical panel. In conduit systems, the conduit itself is often the grounding line.

The grounding wire is primarily a safety back-up, so that any defect in the appliance that could connect 120Vac to the case is actually shorted to ground, causing the panel breaker, rather than the operator, to blow.

In electronic equipment this wire serves as the ground path for chassis and shielding, which is often separate from signal ground.

The goal is to be sure that all equipment is grounded to a single bus in only one power outlet panel. And even in single-panel systems, it's a good idea to float cable shields in the studio and connect them at the console or effects rack. The reason for this is that the shielding ground path goes through the chassis, into the grounding pin on the line cord and finally to the electrical panel ground bus. If the shield is connected to the chassis of *both* units, you have yourself a ground loop from shield to chassis No. 1, to panel to chassis No. 2, to shield to chassis No. 1.

This resultant loop may or may not cause hum; it depends on the actual impedances and resultant voltages in these particular grounding lines. Lifting shield ground at the source end and connecting it to chassis on the receiving end eliminates the loop, reducing the likelihood of noise.

If you're adding on, remodeling or building a new facility, be sure that grounding wires terminate properly at the panel bus. Keep lighting circuits separate from wall outlets, and run grounding wires or conduit in each circuit separately all the way to the panel, making sure these are connected *only* at the panel and nowhere else in the walls.

To maximize the effectiveness of chassis and cable shielding, ensure that the panel has a good earth ground with

No. 2 stranded copper wire connection. Refer to the April 1985 issue of *RE/P*, where in his article entitled "Edward Van Halen's 5150 Studio," Howard Weiss describes the right way to tie into earth ground. Whether or not the studio is on the same circuit as the control room, it is advisable to float the signal cable shields at the studio end and connect them to the case at the patch bay or console end. This configuration avoids a ground loop between the two rooms: one leg of the loop through the electrical panel and the other through the signal cables.

Theory into practice

With the electrical system properly configured, attention now turns to equipment connection techniques. In the same fashion as the electrical system just described, audio equipment has two "ground" systems that are treated in a number of ways by different manufacturers. The signal ground ("negative" or "common") is a current-carrying ground, same as the "neutral" in the ac system. Chassis or shield ground is *not* current-carrying, but functions rather as a drain path to shunt most of the airborne interference to earth ground before it can be induced into the signal electronics.

In balanced systems, the chassis/shielding ground is *separate* from the signal lines; the troubles begin when deciding exactly where to connect chassis to shield, and how to avoid redundant ground connections that form loops.

The situation becomes even stickier when unbalanced equipment enters the picture; here the signal ground and shield ground are tied together, and there is regrettably no standard on how this is done in each piece of equipment. The recent trends toward floating and ground-lift configurations has helped compatibility greatly but in order to know when to lift ground or where to tie shield, you should still have a good idea of how the overall ground scheme should work in theory.

Tip No. 1: Cable shielding should be connected to the chassis only at the receiving end, and *not* at the sending end. Why? Because, to work correctly a shield need only be drained to ground at *one* point. Connecting it to the chassis of both units creates a ground path between them. If these units are grounded via some other path (such as the line cord grounding pin, or the front panels connected through a common rack rail) then the shield path forms a ground loop.

In balanced systems use the ground lift at the sending unit (if so equipped), and make sure the shield is connected to the chassis (via the ground link) at the receiving unit. Unless it's a permanent installa-

FROM: FLOATING OUTPUT TO: BALANCED PATCH BAY



FROM: UNBALANCED OUTPUT WITH UNGROUNDED AC CORD TO: BALANCED PATCH BAY OR SNAKE



FROM: UNBALANCED OUTPUT WITH UNGROUNDED AC CORD TO: UNBALANCED PATCH BAY OR SNAKE



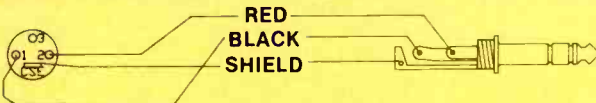
FROM: FLOATING OUTPUT TO: UNBALANCED PATCH BAY OR SNAKE



FROM: 3-PIN BALANCED TO: BALANCED PATCH BAY OR SNAKE



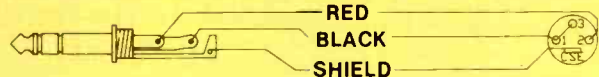
FROM: 3-PIN UNBALANCED TO: BALANCED PATCH BAY OR SNAKE



FROM: FLOATING OUTPUT WITH GROUNDED AC CORD TO: 3-PIN BALANCED INPUT



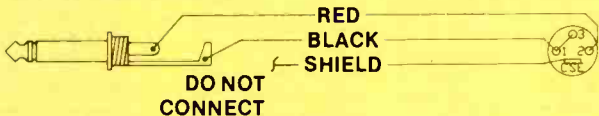
FROM: FLOATING OUTPUT TO: 3-PIN UNBALANCED



FROM: 3-PIN BALANCED TO: 3-PIN BALANCED



FROM: UNBALANCED TO: 3-PIN BALANCED



Interconnect schemes between various balanced, unbalanced and floating connectors used in patchbays and wiring snakes.

tion, avoid wiring special cables with shield connected only at one end. (Modify the sending unit instead if it isn't floating, or does not have a ground lift switch.)

How does this apply to unbalanced systems, one might wonder? Read on...

Tip No. 2: Shield and signal ground in cables should be *separate* wires, even in unbalanced systems. Use 2-conductor microphone cable with shield to make *all* signal connections. In unbalanced systems, leave the shield unconnected (floating) at the sending end of the cable, and connect it to the signal ground only at the receiving unit.

With the newer floating, unbalanced designs, the output connector is a 1/4-inch TRS (stereo) jack with signal ground connected to ring, and the sleeve floating (unconnected) from chassis. In this way, 1/4-inch stereo cables may be used without modifications inside the plug at one end.

Why keep these wires separate if they connect at one end anyway? Remember, shield and ground lines have resistance. Adding a separate shield line helps shunt induced interference current directly to chassis, but tying shield to ground at both ends creates a loop and nullifies its effectiveness.

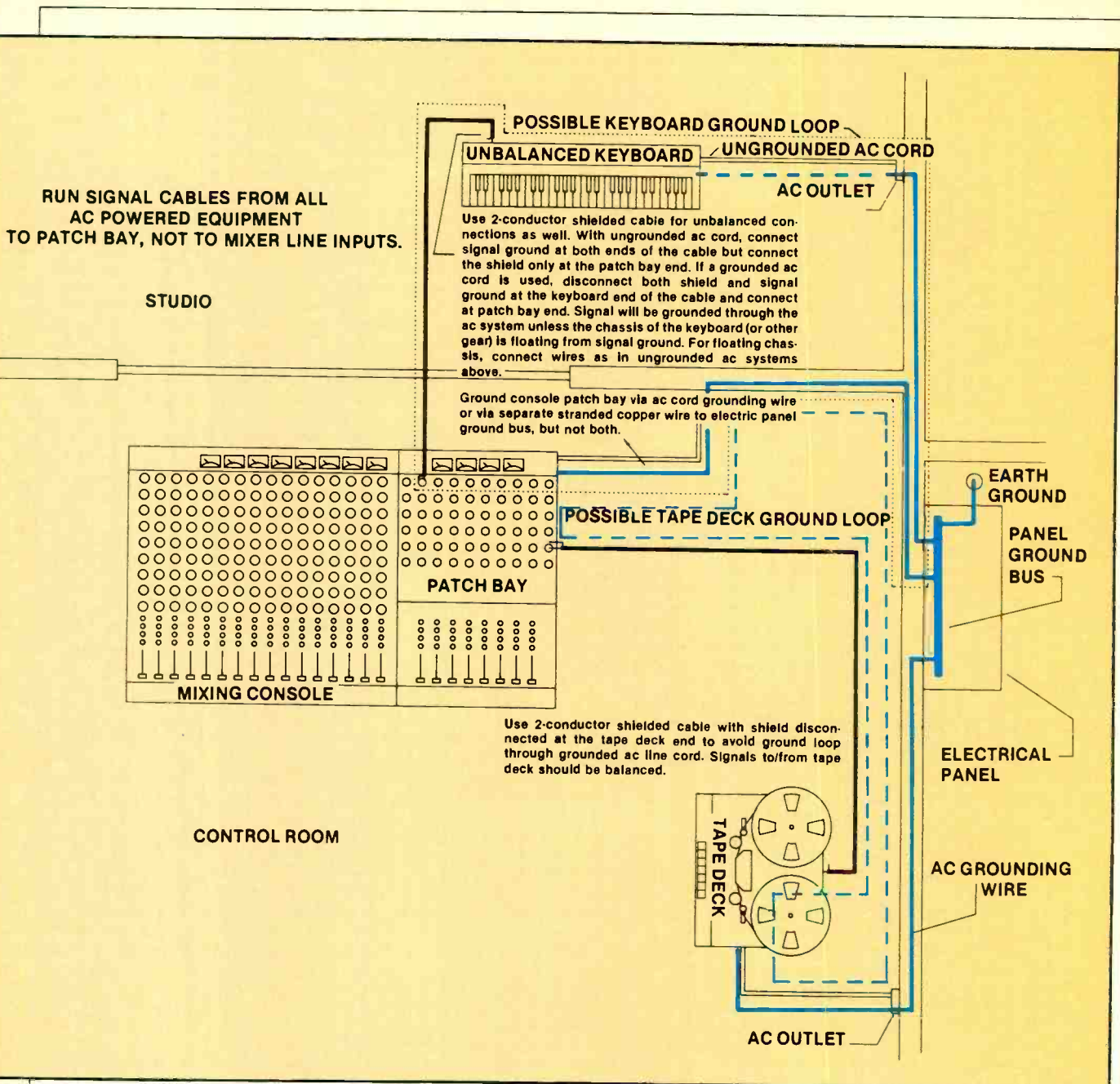
Tip No. 3: Determine a grounding scheme for your system and make sure that each and every piece of equipment complies with it. This means that you must know and verify *all* the grounding paths in each unit: shield connections, chassis connections, line cord grounding connections, and so on. Then tie each unit into the system to prevent redundant grounds, and to adhere to the star-grounding system as much as possible.

For example, if you decide to use the line-cord grounding pin as the chassis/shield ground path, then make sure that *all* units in the system ground the chassis through a 3-pin line cord. You'll also need to modify 2-pin models to comply. Otherwise chassis grounds will be mixed into the signal ground of the other units, and trouble will likely occur. Check that cable shielding in this situation is tied only at the receiving end (configure all ground-lift switches accordingly or verify floating outputs) to preclude looping with the electrical ground.

Pick a central point, such as the console or patch bay, into which you will run *all* connecting cables. Don't run some cables to the patch bay and some to the console, or ground loops may creep in.

Optimizing system signal-to-noise performance

An important, often overlooked, aspect of interconnecting equipment is



A typical wiring interconnect scheme for a small studio showing the typical locations at which ground loops and interface problems can result, and potential solutions.

maintaining optimum signal-to-noise ratio all the way through the system. If the signal level in any one piece of a chain of equipment drops below optimum (while the noise floor, of course, remains the same) the noise performance of the whole system is degraded, because adding gain after this point also amplifies the previous noise level.

Before the current proliferation of solid-state circuit components, to preserve frequency response and output levels, you had to pay attention to proper impedance matching of equipment. Today's op-amp technology has virtually eliminated the impedance problem, but now we must contend with level match-

ing (or gain matching).

It is all too common to come across a parametric equalizer installed across a mixer input's channel-insert loop with only a moderate signal level fed through it. By the time additional gain is added in the mixer's subgroup and master stages, the program material sounds like a leaky tire and, of course, the equalizer is blamed.

To overcome such problems, set the equalizer level control to unity gain and run the highest signal level through that the unit can take without overload. (Which means setting the mixer trim and pad settings as hot as possible.) Pull down the channel fader if you have to,

but keep a strong signal through the equalizer.

Not surprisingly, there is a definite relationship between induced noise and these interactive practices. Correctly implemented grounding techniques preserve the maximum shielding effectiveness of cables and equipment chassis. A ground loop can "push" various ground points away from virtual ground (or "virtually zero-voltage" ground) by allowing current to flow through resistive ground lines. Thus the induced distortion voltages on the shield lines are shunted to some point away from virtual ground, resulting in some portion of the interference voltage bleeding into the

signal lines. While breaking the loop may not change the resistance in the lines, it restores virtual ground to the shield connection points to drain these properly and reduce bleed-through.

Induced interference problems

Once the difficulties with ground circuits have been tamed, you often find that hum and buzz are minimized to a point where the system begins to show off other noise pick-up capabilities. The major contributor in the induced-noise category is RFI (radio frequency interference), which can be devilishly difficult to eliminate. RFI is caused by all radio-frequency transmissions, as well as devices such as light dimmers, motors and their control electronics, and a plethora of other radiation sources. Interconnect wires between audio components make very good antennas for this type of interference.

As pointed out earlier, cable shields are not perfect conductors, and therefore cannot function as perfect shields. The use of high quality cable is important, as is following reasonable practices in cable bundling and routing. RFI can be reduced by changing cable routing until these antennas are "de-tuned" and pick-up minimized. Because a certain amount of pick-up by the cables is unavoidable in stronger RF fields, you must then rely on electronic cancellation and filtering techniques to solve the problem.

Use balanced lines

The use of balanced signal lines between equipment can result in significant cancellation of RFI picked up by the cables. Unwanted signals that are induced equally onto the positive and negative cable wires are subtracted by the differential inputs—either a transformer or active op-amp circuit. The ability of the input stage to cancel frequencies that are common to both balanced lines is called the Common Mode Rejection Ratio (CMRR), which is expressed in decibels. Because the CMRR of audio equipment can vary greatly, if you want your balanced lines to be effective you should look for a CMRR spec of 60dB, or more. In active input circuits, the op-amp's CMRR can be seriously degraded if high tolerance resistors are not used. Look for a trim pot or 0.1% resistors in active input circuitry before you believe any impressive CMRR spec.

Built-in RF filters

It is becoming a more standard practice for manufacturers to install RF filters at the input stages, utilizing ferrite-head inductors and precision R-C filters. If your facility is located in a strong RF field—maybe near an AM or TV anten-

na—look for these design features in each piece of equipment before you purchase. In some cases, RF fields are so strong that interference is induced right through the chassis or front panel of a unit and into circuitry located after the input filtering and differential circuits.

If you encounter severe RFI problems, find the culprit(s) by bypassing one piece of gear at a time. Once located, unplug the input cable and listen for the interference: if you still hear it, the interference is being induced through the

chassis or front panel. If you don't hear it, then the interference is coming in through the cable and being detected, probably in the input circuitry.

These situations require different solutions, so try to determine this before you call for help. A reputable manufacturer should make an effort to eliminate the problem—find out from your equipment supplier which manufacturers are noted for their support, and consider this when you make your hardware purchases.

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Circle (25) on Rapid Facts Card

Property and Liability Insurance for Recording Studios

By Robert J. Kribs

Insuring a recording or production studio involves choosing the right insurance agent/broker, and learning to manage risk.

Your business is exposed to financial loss from all sorts of angles. Some of the losses are not insurable, such as having insufficient revenue to cover your exposures. You cannot buy insurance for that kind of risk.

But there are other aspects that are insurable. For example, there is the building that houses your operation. You also have a lot of money tied up in equipment. If your equipment is destroyed, you're out of business, at least for a while. You probably have a tape library, which we'll discuss in some detail later in this article.

There are other miscellaneous factors: your furniture and fixtures; other kinds of equipment; your accounts receivable

Robert Kribs is president of Oxford Insurance Management, an insurance brokerage firm headquartered in Los Angeles. He is a past president of the Independent Insurance Agents and Brokers Association, and handles insurance matters for a number of major production facilities.

and your future income stream.

Before buying insurance to take care of all these risks or exposures, you first should identify, then eliminate, as many risks as you can. This is the first step in designing any kind of risk management program. Your agent or broker can help you identify the many risks to which you and your business are exposed.

An extreme example of identifying your risks occurred during the OPEC oil crisis of the last decade. People were buying giant barrels, filling them up with gasoline and putting them in their garages. That's a heck of an exposure. What do you do? Get rid of the barrel.

The second step is to minimize or control risk in areas where you cannot eliminate it. Beef up security. Install an alarm system or hire a 24-hour guard for your facility.

To guard against fire, install a sprinkler or Halon system. The cost of your pro-

perty insurance will be reduced by as much as $\frac{2}{3}$ to $\frac{3}{4}$ if you install a sprinkler system. It's yet another way to minimize your exposure.

After you've eliminated some risks and minimized others, the next step is to decide between buying and not buying insurance to cover your risks. If you decide to buy insurance, what you are doing is technically transferring that risk to a company that's willing to take it over for you.

If you decide not to purchase insurance, you could say, "Well, we're going to self-insure this exposure." Technically, self-insurance requires that you set up a sinking fund into which you build a reserve to pay for losses. Most people don't do it, however. What you're really doing is not insuring at all.

Tape libraries

Let's examine the procedure of eliminating risk, minimizing it and then considering insurance if you need it, using the example of a studio's tape library.

Who owns the master tape? Most likely you own the tape stock, but the producer, musician or record company that commissioned your studio for the session owns what's recorded on it. If it's the result of a 3-week album project, you might have seen some money up front; if it's a 1-day jingle session, it might be months before you see any money.

The physical tape itself isn't worth much; it's the material recorded on the tape that you need to worry about. That's where the bucks are.

After a project is finished and the studio gets paid for a tape, can you get rid of it to eliminate your risk? Probably not. In a month or a year, the client may come back and will want to do a remix or something else with the masters. If the tape has disappeared, the client may sue you and say, "Hey, I had a use for that tape and you blew it."

Case Example: Insuring the Old and New Record Plant Studios, Los Angeles

A recent exposure that developed at the Record Plant is a good example of how studio owners should work with their insurance agent or broker. Chris Stone, Record Plant president, recently constructed a new building in Hollywood, and moved his operation from the previous studios. He purchased Business Interruption insurance, so that if the place burned down his lost income would be replaced, as would that of his key people.

But, as construction proceeded at the new building, and Chris was winding down at the old facility, we perceived an exposure that is rather rare. At the old building, his need for Business Interruption insurance was shrinking; if the location burned he would not re-

*place it. In other words, this insurance would replace his lost income **only** until he was due to move to the new facility.*

What would happen if Chris gets down to within one month of moving from the old building and one month of work outstanding at the new construction, and the new facility burns down? Sounds awful, doesn't it? Business Interruption insurance isn't going to help him in this case, because you cannot buy coverage with respect to a building you don't yet occupy. And it's going to take another nine months or so to rebuild the place.

We came up with a new type of insurance for the Record Plant, and called it delayed opening insurance. Starting with a business interruption

form of coverage, we bent it into the kind of shape we needed.

The coverage said that if the new building was destroyed or damaged, and it prevented Chris from moving in by such-and-such a date, he was covered. Now, if the delay was caused by slow construction work, that's another ballgame. But if it was some kind of insurable loss, we could take care of the income lost for the period of time it would take to rebuild the facility.

We sold the idea to the insurance company by explaining to them that the coverage was essentially like any other Business Interruption situation, except that the building wasn't yet occupied.



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WED THURS FRI

A way to minimize your risk is to have the client sign a release stating that he is leaving the tape at his own risk. If you take the time and money to have a release drawn up, and in the next 20 years the use of that release helps you to avoid just one lawsuit, it will have more than paid for itself.

Duplicating master tapes is another way of minimizing your exposure. Although copy masters won't eliminate the risk entirely, if you duplicate a tape and take the dupe off the premises, you're going a long way to reducing potential exposure. The odds against both copies being destroyed at the same time are extremely high.

Protecting your tape library is probably the toughest insurance issue faced by the recording industry. If you dupe the finished master, you've protected the entire work. But what about a work in progress?

If you do make a dupe after the end of a session, or at a certain point in the project, you've only eliminated risk up to that point. The client goes back into the studio and, all of a sudden, you're building up value again. If you've duped a third of the session, stored it away and you lose the next part, you haven't lost the whole two-thirds.

If you follow the outline above, finding insurance to cover the tape library should be relatively easy. It may not represent an ideal situation—you've got signal degradation and so forth in the copy master to worry about—but you will be a lot better off than if you had done nothing.

Property and liability insurance

Other than through poor management, there are two basic ways that a recording or production facility owner can lose their assets and future income stream. One of them is the obvious destruction or theft of assets—they burn up, are stolen, or explode. Protection against that sort

Educating Your Insurance Agent

It's essential for you to educate your insurance agent or broker so that he really understands your type of business. He's never going to understand it the way you do, and that's not the point of the exercise. But if he doesn't understand what your studio does, and how you do it, he's not going to recognize the risks inherent to your business. As a result, he's not going to advise you properly.

I would start the same way you pick an attorney or an accountant: you want someone who appears to be bright, not someone tripping over his own shoelaces.

Next, check out his reputation around town. If he's referred by someone in the recording business, that's even better. Ask him a lot of questions. And if he doesn't ask you questions, pass on the prospect, because he is not going to know your business well enough.

He's going to need to ask you a lot of questions about your business and how you do it, why you do it, and so on. If he doesn't bring up the question of a tape library, that doesn't mean that he's not a bright insurance agent; it may just mean that he doesn't know enough about your business. That's when you've got to start educating him so he can do his job properly.

of incident comes under the general heading of property insurance.

The other way a facility owner can lose assets is if he is forced to pay them over to a third party because of liability for injury or damage. That loss of assets falls under the general heading of liability insurance.

• **Property insurance.** There are essentially two ways you can buy property insurance. One is for cases where the perils that are being insured against, such as fire, explosion, riots and vandalism, are listed in the policy. The problem with this sort of coverage is that there are a lot of things that can happen, but which nobody thinks about in

advance.

There is another and better way of securing coverage for property loss. The insurance company provides for everything except the few perils that are uninsurable, such as normal wear and tear, or that require special handling, such as an earthquake or flood. This type of insurance is called the "all-risk" approach. The insurance company gives you the whole ball of wax and says, "We'll cover all risks of physical loss or damage, except as specifically hereinafter excluded." This is a much better coverage approach, because you don't have to worry about every possible peril against which you might need insurance.

If you've chosen this all-risk type of insurance, the next question is: "Do you insure to full replacement value, or to what the insurance companies refers to as the actual cash value of your property?"

The preferred solution is to insure everything on a replacement-cost basis. If the facility burns down, you can walk right out, buy all-new equipment again and be back in business.

• **Liability insurance** covers all sorts of areas not related to property insurance. Basically, it deals with third parties and how they interface with your business.

There are several areas in which you should have liability coverage. Check with your agent for the coverage that's appropriate for your facility.

If you have any parties at the studio, or if someone likes to unwind after a session, you need to protect yourself. You need Host Liquor Liability coverage. (This coverage is different from liquor liability insurance, which is appropriate to a bar or liquor store.)

If you hold a party at the studio and somebody gets drunk, hurts someone and the studio owner gets sued for serving him, you will be protected. Host Liquor Liability coverage is included in the broadform Comprehensive General Liability endorsement to be discussed later.

What to Look for In Insurance Coverage

Most of us are under the impression that the insurance requirements of the studio industry can be satisfied by the standard types of insurance policy provided for other industrial risks. Contrary to popular wisdom, however, selecting the right insurance cover can be one of your most important business decisions; taking out the wrong or inappropriate policy could cost you many thousands of dollars—at worst, it could cause an end to your business.

A well-written policy should provide coverage against the following:

• All risks of physical loss or damage (subject to certain standard exclusions) to recording equipment and all other contents, including sound proofing. It

should provide for settlement on a full new for old basis; meaning that no deduction will be made for the age of the property.

- Loss or damage to customer's musical instruments.
- Loss or damage to customer's recordings in storage at your premises.
- Loss or damage to customer's recordings as per recommended terms of hire, including accidental erasure.
- Loss of revenue.
- Loss of book debts.
- Loss of money.
- Loss or damage to employee's personal effects.
- Injury to employees as a result of actual or attempted robbery.

- Legal liabilities to employees.
- Legal liabilities to third parties.
- Legal liabilities for leased or rented premises.

Editor's note: The above brief description of coverage was abstracted from material provided by Northwood O'Neill, a London, England-based insurance company that offers special terms to members of the Association of Professional Recording Studios.

No implicit or implied endorsement from RE/P nor Intertec Publishing is to be attached to the inclusion of these details. It is intended to provide a general overview of typical services available to the recording and production studio industry.

Everybody needs premises and operations coverage; it's part of doing business with the public. Contractual liability is another coverage you may need if you sign any kind of agreement that has a hold harmless clause, wherein you agree to indemnify, or hold harmless, the other party for whatever reason.

Although a studio isn't selling a product in the way that other businesses do, you should think about buying Product Liability coverage. Do you sometimes sell a used piece of equipment? Or maybe you do sell an actual product? Although you should have the coverage, it shouldn't cost too much because you really don't have much of an exposure. (After all, you're not in the business of manufacturing something that potentially can hurt somebody.)

If you have any construction work carried out by anybody outside of your studio staff, you need to have owner's protective insurance. It covers you for acts of contractors if they happen to spill into your lap, which they can well do.

Errors and omissions

Here's an example of something you need to consider: What happens if your tape operator or assistant engineer erases someone's tape, or takes five tracks off a project? It wasn't intentional; it was something that just happened. Could you be held liable for a mishap like that if, for example, it delayed the release of an album, or rendered unusable a commercial for the Super Bowl?

You might think about securing professional liability coverage for errors and omissions. I'm not going to go out on a limb and tell you that you *need* it, but it's something that you should at least consider.

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Cutting insurance cost

To cut costs, remember to look at all your exposures to determine what risks you can eliminate entirely and what risks you can minimize. Take advantage of services that insurance companies offer; they employ loss-prevention engineers who will come out to your facility and go through everything to help you prevent loss. Take advantage of their services.

Look at higher deductibles. Maybe you're used to a \$1,000 deductible; think about going to \$5,000—it might save your \$10,000 or more.

Consider installing a sprinkler or Halon system to protect the electronics; such a system can cut your property insurance premium by $\frac{2}{3}$ to $\frac{3}{4}$. Beef up security. Hire a guard, put bars on windows, install an alarm.

And, finally, the most important aspect of all: Find yourself the best insurance broker you can. **REP**

This article was adapted from a seminar presented at the 1986 SPARS Business Conference, Los Angeles.

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Production Viewpoint: Narada Michael Walden

By Adrian Zarin

This musician/producer shares some interesting recording and production techniques while working with Aretha Franklin and Whitney Houston, and also details his new personal-use studio, Tarpan.



Perhaps more than any other time in recent history, today's Top 100 belongs to the musician/producer. Few people exemplify this hybrid production skill as thoroughly as Narada Michael Walden.

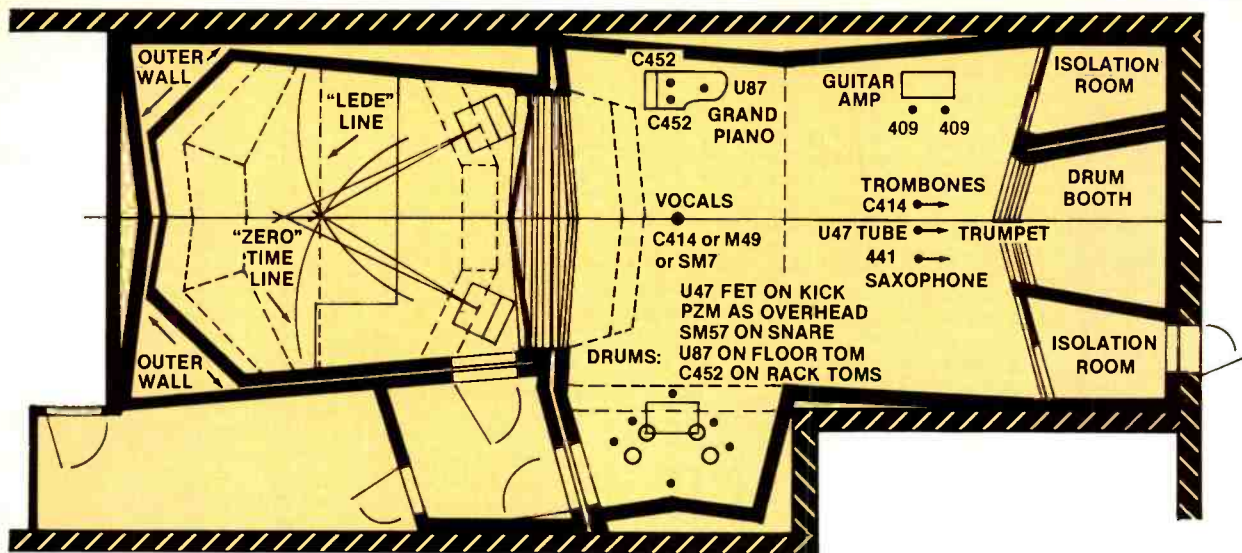
During the mid-Seventies, he rose from the ranks as a jazz-fusion drummer, locking in with such virtuoso talents as John McLaughlin and the Mahavishnu Orchestra (*Apocalypse*, *Visions of the Emerald Beyond* and *Inner Worlds* albums), Weather Report (*Black Market*), Jeff Beck (*Wired*), Roy Buchanan (*Loading Zone*), Chick Corea (*My Spanish Heart*), Robert Fripp (*Exposure*), Alan Holdsworth (*Velvet Darkness*), Jaco Pastorius (*Jaco Pastorius*) and Santana (*Oness*).

Walden's talents as a songwriter and arranger also come to the forefront on some of these sessions. In 1976 he released his first solo album, *Garden of Love Light*, which was followed by seven more albums, culminating with the 1985 release of *The Nature of Things*.

Another facet of Narada's musical identity emerged when he turned his hand to production. He scored substantial hits with Stacy Latisaw ("Love On a Two-Way Street"), Angela Bofill ("Too Tough"), Sister Sledge (*All American Girls*), Herbie Hancock and Patti Austin. 1985 was a particularly auspicious year for Walden, thanks largely to three records he produced. One was Whitney Houston's "How Will I Know." The other

Musician/Producer Narada Michael Walden at work on the Whitney Houston and Aretha Franklin albums.

Adrian Zarin is a Los Angeles-based electronic synthesist, composer and free-lance writer, and a regular contributor to REIP.



Room and microphone layout for recent sessions with Aretha Franklin and Whitney Houston at Tarpan Studios. Producer: Narada Michael Walden. Engineer: Dave Frazer. Note that a large collection of centrally located percussion instrumentation, which for reasons of space cannot be shown on this diagram, were covered with various C452, U47 FET and MD-409 microphones. Electronic keyboards and synthesizers located opposite the guitar amp received Countryman DI boxes.

two were songs Narada also had a hand in writing: "Freeway of Love" and "Who's Zoomin' Who?" from Aretha Franklin's much-heralded comeback album of the same name.

Riding the crest of these successes, Walden recently produced sessions for George Benson, Clarence Clemmons and Sheena Easton at Tarpan, his San Francisco personal-use studio.

RE/P recently caught up with Walden when he was in Los Angeles to deliver seven masters for a new Whitney Houston album. We found him in particularly good spirits, having just been named *Billboard's* "Producer of the Year" for 1986. Narada proved an expansive and enthusiastic commentator on the art and craft of making records. With two Walden-produced singles consecutively hitting No. 1 on the *Billboard* Hot 100 in April, the producer is a likely contender for 1987 honors.

The songs, "Nothing's Gonna Stop Us Now" by Starship and "I Knew You Were Waiting (For Me)" by Aretha Franklin, mark the first time in almost four years that a producer has been involved with back-to-back No. 1 singles. Walden is only the sixth producer in recent history to achieve such a status.

RE/P (Adrian Zarin): As compared with *Who's Zoomin' Who?*, the Aretha album has more of a ballad emphasis. Was that a deliberate plan that you and Aretha made in advance?

Narada Michael Walden: No. That's just the way it worked out. In fact, the duet with Larry Graham ["If You Need My Love Tonight"] was written about 2½ or three years ago, back when I was doing *Who's Zoomin' Who?* So was "Jimmy Lee," for that matter. On the other hand, "Do You Still Remember" was cut for her new.

But, those earlier songs I wrote for her were ones that she couldn't get behind at

the time, or that the record company couldn't get behind. It just worked out that some of them ended up on the Aretha album. So when you compile all the songs, you realize that it is a sweet kind of record.

RE/P: What do you look for when you're choosing an outside song for an artist?

NMW: A big hook...a big chorus...I'll tell you man, I'm a simple cat! A verse can be rewritten. Like on "How Will I Know," for example. As the song was originally written, there was no verse in it. So I wrote one. And I don't mind doing that...as long as that chorus is there.

RE/P: It would appear that many of your arrangements are aimed at maximizing those big choruses via key changes and similar techniques.

NMW: Oh yeah. You've got to grab the listener by the necktie and say, "Come here!"

RE/P: On "Freeway of Love," the chorus receives the same kind of buildup, thanks to the rhythmic accentuations that lead into it.

NMW: I don't like to mess around. I don't want it to slip by anyone. I demand your attention. You put a great voice on top of that and hopefully you've got a hit record. I do everything I can to make sure you know: "Hey, look out, here's a big hook coming right up to your chin."

RE/P: When you're working with an outside tune, what kind of demo do you like to have? Do you prefer the arrangement to be there on the demo, or do you prefer something more skeletal?

NMW: I'll tell you what happens. If I have a demo where the arrangement's all there, I'll end up using a lot of it, not relying so much on my own instinct. If I get a more "naked" demo, I tend to do

more myself, because I have to.

RE/P: And either way is all right with you?

NMW: It can be. I'm still a baby; I'm new enough, green enough, that I can work with a lot of different things.

RE/P: For a long time, you used The Automatt studios in San Francisco as your home base. But since David Rubison closed The Automatt, you've set up your own studio.

NMW: That's right. I bought Tres Virgos, changed the name and everything about it. It's now called Tarpan, which means "Satisfaction Unparalleled." It's a place that's now like a home to me. When The Automatt closed, I felt the need to have a place where I could set up my own working environment. I took over Tres Virgos, put down fresh carpets, fresh paint, fresh everything. And we actually brought over a console from The Automatt: the large Trident console [40-input TSM]. So far we've made a lot of great records on it. But, next year, we might go to the SSL formula.

RE/P: Have you had experience with Solid State Logic consoles?

NMW: I mixed "Freeway of Love" and "How Will I Know" on an SSL. It's a very quick board—especially when people

Re-equipping Tarpan Studios

The name of Narada Michael Walden's personal-use studio came from his spiritual master, Sri Chinmoy; but the more terrestrial task of equipping Tarpan and keeping it running on a daily basis is largely the responsibility of Dave Frazer, the facility's chief engineer. When Walden took over the former Tres Virgos recording studio 18 months ago, he and Frazer determined that some of the studio's equipment would have to be replaced before it could be reincarnated as Tarpan.

"Tres Virgos had a very small MCI board," Frazer explains. "As soon as we moved in, we needed to replace it with a console that could handle two multitrack machines. The 40-input Trident TSM from The Automatt was handy, so we went with that. We often run 46-track sessions, however, so lots of times we spill over into the monitor section [for additional multitrack returns]. We can get away with this because the TSM isn't an in-line design.

"The console is equipped with [Valley People] Fadex automation. Unfortunately," Frazer says, "it's a tape-based system, which means that we can usually only get away with between two and five passes before the time lag [between sync and tape replay of automation data] starts to become a problem. We mainly use the system just for overall muting.

"We've been discussing moving up a [Solid State Logic] SL-4000, probably in a 56-input configuration. We feel it would give us an advantage in terms of both sound and automation.

"Another disadvantage of tape-based automation," Frazer says, "is that it takes up tracks on your 24-track. And you can't merge mixes the way you can on a disk-based system."

Tarpan's present complement of 24-track machines consists of a Sony JH-24 and a Studer A80VU MkIV, synchronized via an Audio Kinetics Q-Lock 310.

"The JH-24 was here when we arrived, and Narada purchased the Studer. We basically went with the A80 because it works better with the JH-24 than an A800 would. The A80 is more compatible with the MCI in terms of speed, although there are still some discrepancies in chase and rewind times."

On some projects, Walden likes to install a 16-track head block on the JH-24 while cutting basics. The wider track width afforded by the 16-track format, the producer believes, makes for a "punchier" sound on rhythm tracks.

"It sounds quite a bit better," says Frazer, "so, once in a while, we'll put our basics on 16-track and do the over-

dubs on 24-track. Usually, we tend to need the 46-track track configuration—we've run as many as 44 or 45 tracks on a couple of mixes.

"Recently—on the new Whitney Houston record—we've also ended up making second and third slave reels. But we always bounce down tracks so that we only end up with two machines for the final mixdown."

Frazer likes to mix down to an Ampex ATR-102 ½-inch machine with SSI transformerless electronics. "We like the ½-inch format a lot. It gives you that extra 'warmth' and retains a smoother high-end than digital."

UREI Time Align 813Bs, Yamaha NS-10Ms, TAO ME265s and Auratone Sound Cubes provide the monitoring options at Tarpan. These control room monitors are powered by Crown D0-2000s with Delta Omega modules. Outboard gear used is a combination of in-house and rental items.

"For mixdown," explains Frazer, "we pretty much order the same outboard gear each time. We like to use a couple of the AMS RMX-16 reverbs, and we also love the DMX delay. Recently, we've been using the Quantec Room Simulator and the Eventide SP2016.

"We use the [Audio + Design] Scamp PanScan, and we always get Pultec EQs in for mixing. I have a Lang EQ that we use as well. And of course, there are plenty of Yamaha SPX-90s, which we use on a lot of tracks for keyboard effects. We try, however, to print the majority of keyboard effects while tracking—certainly any chorusing or delay that is an inherent part of the sound."

Although vocal mics in use at Tarpan vary from artist to artist, Frazer discloses that most of the lead vocals on Whitney Houston's new album were cut with an AKG C414. A Neumann M49 tube mic was selected for Sheena Easton; and Narada has long preferred a Shure SM57 for capturing Aretha's legendary voice.

"We sometimes use a Neve stereo compressor when we cut the vocals," the engineer adds. "Either that or the Aphex Compellor. We'll occasionally use the Neve to compress the overall mix as well—although not a lot. We tend to rely on the mastering facilities to add any additional compression that's needed."

Studio Acoustics

Tarpan's LEDE control room, where much of the tracking action takes place, is a rectangular space measuring 19'x18'. The height of the compression-design ceiling ranges between nine and 14 feet. The main studio area,

measuring 25'x35', also has a variable-height ceiling that is 14-feet high on the sides and 12 feet in the center.

Because Walden's drum kit is usually the only instrument in the main studio area during basic tracks, the room is more than adequate, Frazer says.

"If we do end up with, say, a couple of percussionists playing live out there, we'll just go for it all at once, quite frankly. A little leakage generally sounds nice anyhow. If there's a real problem, we can put somebody in one of the three iso booths—there are two small booths and one relatively large one. They're all a bit uncomfortable to work in though, which is why we tend to avoid them.

"We have also just acquired a rehearsal room next door, which we're going to have working as an additional recording area. It's a large, warehouse sort of space, with a high ceiling. We're running the tie lines now."

The Tarpan facility also includes a small songwriting/synthesizer pre-production room, which is equipped with a Tascam 80-8 8-track on ½-inch.

"Narada usually ends up writing songs and working with a couple of synthesists. One is Bongo Bob Smith; and the other Walter Afanasieff. Both have their own synth equipment, which basically lives here," Frazer says. "The main synths are an Oberheim Matrix 12, which we've used on just about every record, a Yamaha DX-7 and TX-816 tone rack, a Prophet 2002 sampler and a Roland Super Jupiter. It's something of an ever-changing setup, however."

A Roland MSQ-700 serves as the principle sequencer at Tarpan, although Narada and his associates are currently examining several sequencing software for personal computers. Sequencers and drum machines are synchronized to tape via a Roland SBX-80 Sync Box.

"But we generally print most of our sequencer tracks on tape," Frazer adds, "because the synth equipment is always doing double duty. It's generally needed up in the songwriting studio, so we can't tie it up for retriggering during a mixdown.

"We're in the process of really changing the sound we've been putting out of here," the engineer concludes. "The people who we have run across, and who have been incorporated into our system, are really designing sounds for us. In the past, we relied on more 'stock' sounds. Now, we'll have a guy go out into a room somewhere and sample a snare, just for us. This way, we get exactly the sound we want; a sound, moreover, that is unique to us."

want remixes with just a few small changes. I like the fact that each channel has its own compression. The board just has a nice "kick" that I really like.

But you have to know how to use an SSL. People think it doesn't have any bottom end, but the new series might change that. Most of the records you're hearing on the radio were recorded or mixed on an SSL. For my sound, it works really well.

RE/P: Are you using console automation in your current work?

NMW: In the mixing phases we are. What happens is that my engineer, Dave Frazer, will take time by himself to get a mix up. After he's comfortable, we'll make any minor changes we want. We'll write those in and fine tune from there. So we do use it—but not to the extent where spontaneity is no longer there, because it's very important to have that "human touch" prevail.

RE/P: Given the high technology that people are working with these days in the studio, do you think there's a danger that the human touch will not prevail?

NMW: It's not a danger so long as you know how to get down with it. It's like a

microwave oven; if you know how to work one, fine. If you don't, you're going to ruin dinner.

RE/P: When you took over *Tres Virgos*, did you alter the studio's acoustic design in any way?

NMW: No. I was very happy with the existing design. The control room is a Chips Davis' Live-End/Dead-End room and the large monitors [UREI Time Align 813Bs] are very accurate. George Benson enjoyed the studio thoroughly when he was there—and he has his own SSL studio. Recently we've had Sheena Easton, Whitney Houston and Aretha at Tarpan, and they were all very happy with it.

RE/P: Speaking of George Benson, what kind of direction did you envision for his new album?

NMW: We really wanted to get him back on the dance floor and back on black radio—back to the people, as it were. So there are R&B tracks like "Sure" on there, and songs like "Teaser," which is more of a dance R&B pop kind of track. Then there's "Too Many Times a Night," which is a nice pop kind of ballad, and "Kiss of Moonlight," which is a beautiful ballad that everyone can love.

That mixture of all kinds of music is what I'm really about. I'm not, for example, a producer like Larry Blackmon of Cameo, who does just one type of music very well. In my case, I enjoy the fact that I love to do all kinds of stuff.

I've always been influenced by a real mixture of black and white music. Look at me. I wasn't raised in a ghetto, *per se*, although I've been poor and gone through my struggles in life. But, while growing up, I listened to things by Johnny Mathis, and "Old Cape Cod" by Patti Page. I loved Jimmy Smith and Horace Silver, and all that jazz mess. I really feel blessed that I've had a chance to be exposed to all kinds of music.

RE/P: But the Motown influence is perhaps what comes through most forcibly on your recent productions. And I know that you're from that general area—Kalamazoo, MI.

NMW: Well yeah. Everyone says to me, "good songs, good songs;" and when you think of good songs that's Motown—they combined great songs with great tracks. James Jamerson slippin' and slidin' on the bottom. Benny Benjamin on drums. Then you had Holland-Dozier-Holland writing killer songs.

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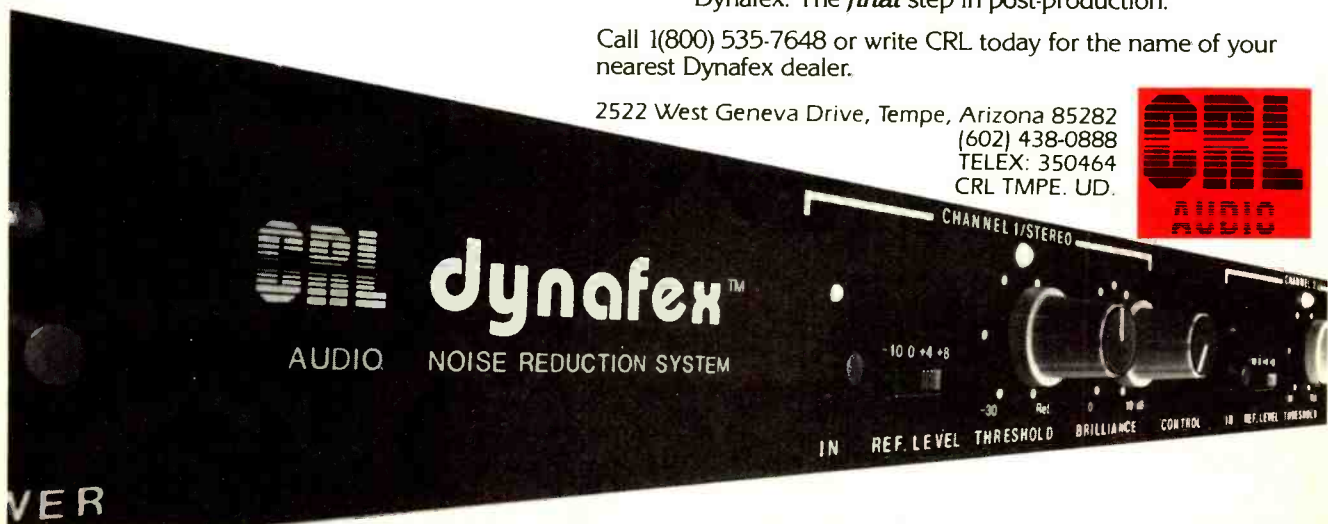
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It's hard to beat those kinds of combinations. To this day, when I want inspiration I'll go back to those records.

RE/P: *A lot of those arrangement elements we were talking about earlier—key changes, big lead-ins to choruses—come straight out of the Motown vocabulary.*

NMW: Yes, but not only Motown. Chicago too, and Philadelphia, New

“To this day, when I want inspiration I'll go back to those records.”

York. Man, we've been blessed to hear some great music in these times.

RE/P: *You usually play drums and the occasional piano part on your productions. Do you find it difficult to simultaneously play the roles of producer and musician?*

NMW: Yes. So what I do most of the time is start out by programming the drum grooves on a machine. I listen to it, get the feel right and decide whether or not I want to have live drums. If I do, I'll put the drums on, or go in and recut the basic tracks with the musicians.

But all that is after I've had a Poloroid of the arrangement, so to speak. Because if I go out fresh and just start cutting the drums, I haven't had a chance to scope out all the fine details of the arrangement—how the drums are working with all the other instruments. And that's very important.

RE/P: *If you don't recut the entire track, will you often replace selected drum-machine sounds with live drums? Might you, for example, pull the kick and snare out of your program and play them live, but keep the handclaps, shakers and other sounds from the drum machine?*

NMW: Oh yes, that's how I've been doing it for years. That's the old school right there...

RE/P: *What's the new school?*

NMW: We're getting away to a certain extent, from the machine kind of sounds. It's not so much the “formalized” sound of two years ago. There's more live percussion—tambourines, for example—so you get the whole Motown thing happening. Instead of all the [drum machine] handclaps, there's more of a giant “meaty” snare. It has that crack!

You'll hear it if you listen to the recent Cameo records or the tracks that Jimmy

Jam and Terry Lewis are producing—like the Janet Jackson songs.

RE/P: *Even on an older track such as “Who's Zoomin' Who?” the snare sound you got was hardly timid.*

NMW: But that was more of a lower-pitched snare sound with power; the new sound is higher pitched, and more of a serious crack.

RE/P: *When you want that crack on a record, how do you achieve it?*

NMW: Just by using different chips in the drum machine. They're just like having different drums available; each one has its own characteristics. If I'm looking for a serious crack, and I'm playing live drums, I like to use a Ludwig Black Beauty snare. It's got a metallic sound that'll just wipe you out. For ballads, I like using an old wooden Yamaha snare drum that I own. It's very deep and has a certain richness that I really like.

RE/P: *When you do use drum machines on a production; do you print those tracks on tape? Or do you print a sync or MIDI track and then retrigger the drum program live during mixdown?*

NMW: I just print the drum machine tracks on my multitrack. I know that my friend Peter Wolf, for example, re-syncs parts to keep them sounding more live [and save a tape generation]. But it isn't that big a deal to me.

I'd rather know that the groove is recorded in the pocket. When it's great, it goes on tape and I don't have to mess with it again. And, if you feel there's been some generation loss, add a little top-end EQ. It's no big deal, really.

Anyway, after I cut my basic tracks I put those tapes away, so that they stay pretty fresh. If I want to do a lot of vocalizing with an artist, I'll put up a vocal reel and set the tracks aside. I haven't really found retriggering to be all that necessary.

RE/P: *Typically, are there a lot of those subreels involved in your projects?*

NMW: I think I got into that the most on Whitney's new album. Usually the whole project is on two reels: 46 tracks. One reel will have all the basic instrumentation and the other will have vocals and all the spicy little overdubs. But, I got up to having two reels of vocals on some of these new songs with Whitney. I wanted to be able to keep things that were special, which is pretty unusual for me.

Usually, I'm the kind of cat that goes in there and nails it; I like to try to force myself to make decisions on the spot. I prefer that [way of working] to always leaving your homework undone. It makes for better records.

RE/P: *So for vocals, you'll bounce over a quick rhythm mix down to your slave reel?*

NMW: Yes. Usually the rhythm mix takes up no more than two to four tracks on the vocal slave reel.

RE/P: *For your recent sessions, you've assembled a solid stable of players around you, including bassist Randy Jackson, guitarist Corrado Rustici and*

“I'd rather know that the groove is recorded in the pocket. When it's great I don't have to mess with it again.”

keyboardist Walter Afansieff. Have you evolved certain special techniques for recording with these guys?

NMW: We all work together in the control room. We used to use headphones, but now we're cutting tracks to the large control monitors. I much prefer to cut a basic track as if we're playing live, because everyone can just groove and lock in with each other. You can always go back to punch-in and fine-tune things later on.

RE/P: *Do you set up the drum kit out in the studio?*

NMW: Yes. That's the only instrument that's recorded out in the room. Everything else goes direct.

Or, if we use amps, we have them off in iso booths. But all the cats are in the control room and they're jammin' loud.

RE/P: *Normally, you use the same studio and a lot of the same musicians for each record you produce. How do you tailor all those constants to each different artist, and prevent things from sounding too much the same on each date?*

NMW: I think the song takes care of a lot of that; for me the song is the real star. Everything else adjusts itself to the needs of the song. We also admire different sounds, so it's nice to take an excursion and try different things on each record. For that reason, it's never been a problem. I also think the singer's voice has a lot to do with shaping things as well.

RE/P: *At what stage in the recording process do you generally like to cut a final vocal?*

NMW: I like to get into vocals after I've pretty much got a basic track, without too many outside overdubs. I like to spend a day or two on vocals and then maybe do some background vocal work. And then I color it. What does it need?

Does it really need that string section? How about a horn part? Percussion? For me the thing is not to add things you *think* it needs without first hearing the vocals.

RE/P: To what extent do you consult with the artist in advance?

NMW: Just a case of: "Do you like this song or not? If you like it, is this key good for you?" If they say it's a little too low, I'll say, "good; because I was going to raise it anyway. I like when you have to scream for it. When it's too low, you sound too comfortable. You're boring me!"

RE/P: It sounds as though you like to push singers to their limits, both vocally and emotionally.

NMW: That's right. My attitude is more a matter of: "Don't come in here and waste my time. You have to be ready to sing, and get on out of here." It's like sprinting as opposed to a marathon. I think singles are like sprints. You've got three or four minutes to sell your message; actually, less than that.

If the mess isn't right in the first 30 seconds it's all over—either it hits you immediately or it doesn't. I realize that. And that's why I'm being called to do more and more hit producing—just one or two hit songs. It's not the big album session so much anymore. Just: "Can you give us a hit record?"

RE/P: How do you feel about being involved with an album project with four or five other producers?

NMW: I used to feel it was going against what I liked. I used to feel I had to stretch out with the album tracks. But I found that, hey, it's a great thing when you can come in with just one song—like Whitney's "How Will I Know"—and sell 12 million units. And get credit, almost as though you've done the whole record. Even though Michael Masser worked hard, Kashif, Jermaine Jackson...all the producers involved with the album worked hard.

I've learned through that experience that it's not a bad thing. Actually it's a good thing to do two or three songs, have a good relationship with the artist and just get out of there. As opposed to grinding through a whole album and not being very good friends afterward.

If you want to go on with the rest of the album, fine, but make sure you've got that big hit single first.

RE/P: Now that you've had the opportunity to do the better part of an album with Whitney Houston—as opposed to one song on her previous album—what kind of direction do you think you've



Session engineer Dave Frazer, Narada Michael Walden and Clarence Clemens of Bruce Springsteen's E-Street Band at Tarpan studios.

developed with her?

NMW: On the new album, we spent a long time getting really powerful performances vocally. Everything's in tune [laughs] and with lots of emotion—which is hard to get. Emotion is the hardest; and sometimes you feel, "If the emotion is happening, who cares whether it's in tune?" But I like to mess around with it long enough so that you can get both.

All the songs—except for maybe one—were selected by Olive Davis from outside sources. Initially, I felt that some were stronger than others; but ultimately I fell in love with them all. If I can really spend the time to tear a song apart and put it back together again, I can learn to fall in love with it. That's what happened here. And, so far, everyone's pretty happy with how it's turning out.

RE/P: As part of the pre-production process, do you spend a lot of time on your own taking those arrangements apart?

NMW: What I'll do is go to the piano and understand the chord changes—what's happening in the song, basically—in half an hour. Then I go in with my session players and teach them the song. I'll have them play it one way for me, take a certain part out, maybe have Corrado funk up one part, RJ [bassist Randy Jackson] lay out on another part...and it all comes together pretty quickly.

So much of what's great comes from trial and error. If you come into a session with a lot of preconceived notions, you can end up missing the best ideas. It's the same thing with songwriting. What sounds great on a Casio in a musician's bedroom, won't necessarily sound as good when you get it in the studio.

Vice versa, RJ and Corrado might put their hands on something I thought was insignificant and turn it right around. It

gives them a chance to add their creativity, and that becomes a smash. I've seen it happen time and time again, so that now I've purposely started to *underwrite*—to leave room for them on the date.

By the same token, people will say to me, "Wow man, how did you orchestrate Aretha's comeback?" And I'll tell you: We made sure we *didn't* clutter it up. Because Aretha Franklin is one of the greatest singers on the face of the planet, we left plenty room for her to *sing*. There is a lot of temptation these days to clutter things up, with 48, 60, 70 tracks of music, all kinds of synthesizers and sampling devices.

But what's *really* happening is when you have a tambourine, an acoustic piano, a backbeat, a funky chicken guitar and Aretha Franklin. That will last longer than all that other stuff. It's just the human element; you can't beat it.

RE/P: What plans do you have for the future?

NMW: I just finished up a song with the Starship. It's a tune from the *Mannequin* film soundtrack, called "Nothing's Gonna Stop Us Now." Also, finishing this Whitney record is a major priority for me right now.

Looking to the future, I'd like to work with Bowie, and I'd like to have time to do something with Earth, Wind & Fire, if I can. I feel they've done so much, and I'd like to be a part of that.

And I want to do an album of my own, probably early next year. I'm not the greatest vocalist in the world; I don't make any bones about it. But I do feel I can say and get away with things that other people can't. I'd like to at least put my hat in the ring and see what happens.

Photos by Pat Johnson.

REP



Surround Sound Production for



Photo by Ron Brown/NFL Photos.

Super Bowl XXI

By Larry Blake

The recent live television broadcast coverage of the Super Bowl XXI game with encoded surround sound information involved the use of some innovative pre-production and mixing techniques.

Only in the past five years has hardware and software become available that allows surround sound from theatrical motion pictures to be experienced at home. The rear-channel signal that consumer decoders extract from laserdiscs, videocassettes and stereo television broadcasts is virtually the same information that professional theater units derive from 35mm stereo prints.

Almost all stereo television programs are mixed strictly for standard 2-speaker stereo reproduction. One of the first uses of surround sound in a made-for-broadcast program was the "Go to The Head of the Class" episode of *Amazing Stories*, broadcast in Dolby Surround on Nov. 21, 1986.

This half-hour episode was mixed in a manner similar to stereo theatrical features: three stereo stems (dialogue, music and sound effects) were folded down to a 2-track printing master. Center-channel

information on the Lt-Rt (Left-total-Right-total) composite mix is recorded equal (down 3dB from unity gain) and in-phase across the two tracks; playback over two speakers will result in a standard phantom-center channel. The matrix-encoded surround information is recorded out-of-phase across both tracks.

Currently, there are dozens of domestic decoders that will extract a relatively accurate signal; the units bearing the Dolby Surround trademark are licensed by Dolby Laboratories for this purpose. Lacking this investment, a simple left-minus-right box will provide the non-critical listener with a close approximation of the experience.

On Jan. 25, 1987, two months after the *Amazing Stories* broadcast, *CBS Sports'* presentation of Super Bowl XXI marked the first time that a *live* program contained matrixed Dolby Surround information as part of the MTS (Multichannel Television Sound) signal.

(A historical note: the sound crew for

the video presentation of the final concert by The Who on Dec. 17, 1982, had looked into broadcasting the show with Dolby Surround sound. They decided against using it at the last minute, however, because of the complications involved and the then small number of consumer decoders. The mix on the subsequently released videocassette version *did* contain encoded Dolby Surround information. Undoubtedly, there are many other home-video releases that have utilized matrixed rear channels of one form or the other.)

The germ for the idea of presenting the Super Bowl with encoded surround information arose at the May 1986 AES conference on stereo television techniques. At the Chicago seminar, Michael Rokosa, field technical manager for CBS Operations and Engineering, approached David Gray, chief engineer of Dolby Laboratories, film division, with the idea of using surround sound to take live sports programming beyond normal stereo.

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

Rokosa and Gray used the CBS production of the Michigan State-Notre Dame game on Sept. 20 to prove the viability of a live surround mix for sports broadcasts. The first event was the Super Bowl, which took place at the Rose Bowl in Pasadena, CA, in late January this year.

Audio preparations

Laying the miles of cable for Super Bowl XXI began 10 days before the game. Televised sports broadcast, being the major revenue-producer for the National Football League, is facilitated at all league stadiums by extensive video and audio wiring. Although the Rose Bowl has no NFL home team, it does host its titular college game every Jan. 1. It is perhaps surprising, therefore, that most of the wiring for this year's Super Bowl had to be brought in by CBS.

Audio runs utilized 12-pair snakes and Cannon FK-37 multipin connectors. The majority of the audio and video cabling from the stadium was sent to either the main transmission (TX) truck, for distribution throughout the CBS camp, or to mobile units 11A and 12A. The two 46-foot vehicles are virtually identical, and each is paired with a 40-foot support truck that carries all cabling, cameras, etc. Sound is accommodated by 36x12x4 Ward-Beck consoles located in a small (6' x8') mixing area immediately behind the master video control room.

The pre- and post-game *Super Bowl Today* show, halftime ceremonies and all post-game activities were the responsibility of the Los Angeles-based mobile unit 12A and mixer Mark Radulovich, assisted by Jennifer Spangler. Jeff James fed Radulovich a submix from the on-site *Super Bowl Today* set.

The game itself was mixed by mobile unit 11A's Bob Seiderman, who works out of New York. He was assisted by Jerry Jaick in the truck and by Steve Palecek, who supervised the RF setups on the field. CBS staff mixer Andy Bass handled tape playback for the national anthem and the halftime show, working out of a separate audio mobile provided by Best Audio, North Hollywood. Audio supervisor in the TX truck was Tom Jimenez of New York, who had mixed the Michigan State-Notre Dame game; helping him was Larry Gumpel.

Because the large number of potential audio signals far exceeds the 36-input capacity of the units' Ward-Beck consoles, the transmission truck uses a Datatek 100x50 routing switcher to distribute audio signals such as outputs from the 16 1-inch VTRs and remote-location feeds; for stereo broadcasts the switcher is used in a 50x25 mode. The video for all VTRs also appears on routing switchers throughout the camp.



Photo by Larry Blake.

View from the audio control room through to the front video control area aboard the CBS mobile unit 11A, which handled audio mixing duties for the Super Bowl game. Note the remote Dolby DS-4 monitor unit mounted atop the right hand side of the Ward-Beck production console's bank of main output VU meters.



Photo by Larry Blake.

Tom Jimenez, CBS audio supervisor, sets up audio routing assignments on a Datatek routing switcher located in the main transmission (TX) mobile truck. Beside him is David Gray, chief engineer of Dolby Laboratories' film division.

Each mixer's feed to TX consisted of a mono and a left-right stereo air mix (plus backups of each) and a surround track. Seiderman, working in unit 11A, also sent a stereo international feed, which was identical to the U.S. air mix except for the omission of the CBS announcers. Radulovich, working in 12A, provided no minus-announcer send because such a large percentage of his material was just that—an announcer talking to a camera.

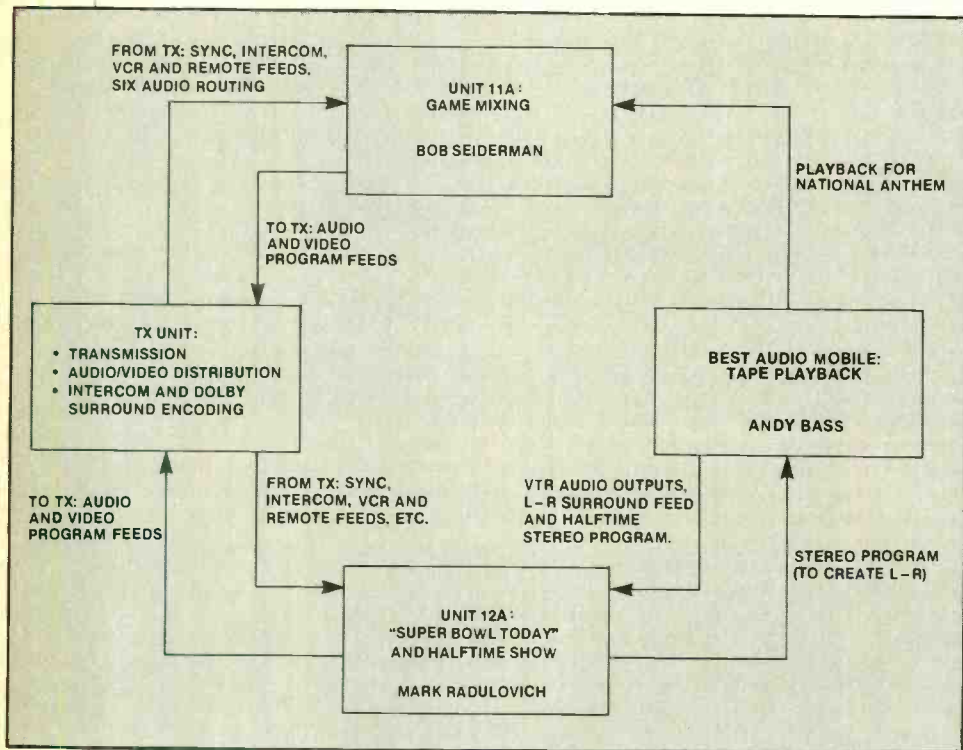
Primarily to keep tabs on what Radulovich was sending to the TX truck, the Best Audio mobile handling tape playback received a split of the stereo feed from 12A. This also allowed CBS staff mixer Andy Bass, working in the Best truck, to create an L-R signal for Radulovich (whose board was filled up) to send to TX as his surround feed. (The only time when Bass *didn't* send an L-R difference signal based on Radulovich's mix was during halftime, when it was taken from his own stereo feed.) As a result, there was surround information available for the whole 6-hour broadcast.

Mixing duties

The *Super Bowl Today* show uses in-studio host Brent Musburger, along with a combination of remote feeds from the United States and overseas, and pre-recorded features from four 1-inch VTRs. Because the video preview monitor on the wall in front of Radulovich doesn't necessarily reflect the upcoming shot, and because there is no audio-follow-video function on the CBS mobile units' production switchers, the 2-hour SBT show involved "seat of the pants mixing."

He was guided not only by the director on the intercom, but also by his close watch on the video monitors to note which VTR or remote feed looked like it might be on soon. Assistant engineer Jennifer Spangler's primary job was to dial up desired feeds on unit 12A's three local routing switchers.

Acoustic stereo micing came into play for Radulovich primarily during the post-game show in the locker room. In addition to an XY mic pair capturing an overall stereo sound field, he would often



Audio, video and sync interconnects between the four main mobile production units used during the live surround sound mixing of the Super Bowl XXI.

cross to an RF minicam that had a Shure stereo mic mounted on it.

Although there was a fairly clean delineation between the portions of the broadcast handled by mobile unit 11A and those by 12A, at any given moment the off-air truck became a remote for the air truck if it had cameras or signals that were not shared. In addition, Seiderman and Radulovich noted how they would "cover for each other, looking for good audio that the on-air mixer might not have access to." The change of command between units 12A and 11A at the beginning of the game occurred during a commercial break just prior to the announcement of the starting lineups.

For the coin toss, Seiderman ordinarily would use just a split from the referee's radio mic, which is provided by the NFL at all games. At this year's Super Bowl, he supplemented the ref's mic with a Crown PZM laid on the field near where the coin would hit. (Although Seiderman was hoping to capture the sound of the team captains talking, during rehearsals the referee thought he was supposed to hit the mic with the coin!)

Seiderman was responsible for providing headphone foldback to the two announcers: Pat Summerall working the play-by-play and John Madden providing color commentary. (Seiderman used Sennheiser HDM224 headsets with integral boom mic.) He simply gave the announcers a mono air mix to both ears, in-

cluding their voices, with one side on a Program Interrupt (or IFB, interruptible foldback) function so that he, the director, producer or associate director could advise them of commercial breaks and instant replays.

The announcers' voices were fed into an Orban model 245F stereo synthesizer to "bring them away from the screen and into the living room." Seiderman had initially planned on widening the sound image during the game and narrowing it when Summerall and Madden were seen on-screen. As it turned out, however, the whole game used the narrow position he set for an opening on-camera segment.

"One of the reasons I wanted a slight 'stereo' image on their voices," recalls Seiderman, "was to prevent stereo synthesizers at affiliates from kicking in when I was just sending [what would otherwise be] a mono announcer-only signal."

Radulovich and Seiderman spent some time matching the stereo widths on their respective synthesizers to prevent the image from jumping when switching between mobile trucks.

Crowd micing

Audience micing was handled by four Sennheiser 416s located on the far side (across the field from the press box) at ground level, pointed back at the near side, and not the immediate crowd. Seiderman uses this positioning as a mat-

ter of course to guard against what can be politely paraphrased as the "obnoxious guy with the cowbell" factor.

The four mics were evenly spaced as two pairs on the 20- and 40-yard lines; the sums of the two pairs created most of the main left-right crowd feed. Added to them were two other 416s located at the 30-yard lines on the near side, also facing high up into the stands.

Outputs from the near-side mics were also bled slightly into the surround channel, joining an AKG C451 facing down from the press box. Note that this orientation follows the stereo image as seen by the primary cameras located in the press box.

Parabolic dishes containing Sennheiser MKE-2 lavalier mics were used to capture on-field sounds, including the quarterback calling the snap and shoulder pad contacts. Because crowded sidelines can be expected to interfere with cable runs at a Super Bowl, two dishes were located on each side of the field, both on wireless links.

The combined output from all four parabolic mics was fed into a limiter (to guard against a bull's-eye while the referee was blowing his whistle) and then into a second Orban stereo synthesizer. Seiderman emphasizes that the intention was "to add depth to work the parabolics into the mix better; we're not really looking for stereo separation."

Dolby's David Gray notes that "in the first quarter we came to the realization that when Bob [Seiderman] reached for the parabolics, we lost too much of the crowd in the surrounds because he obviously pulled down the master crowd feed. It turns out that there is quite a bit of crowd in the parabolics.

"So, to help with continuity," Gray says, "Bobby started feeding some of his parabolic submix to the surround channel. Whenever he brought up the parabolics, some of it went into the surrounds, giving a nice, natural transition."

Halftime show

The 13-minute halftime show celebrating Hollywood's 100th anniversary, as heard by the TV audience, was derived totally from a 24-track tape sent to CBS from Walt Disney World, Florida. (Walt Disney Productions produced the show for the NFL.) The multitrack tape originally contained two copies—one encoded with dbx Type I noise reduction, the other non-encoded—of a stereo mix that the singers and musicians would perform to on the field.

CBS mixer Andy Bass selected the non-encoded mix and bounced it onto two open tracks, "lifting the level, re-equalizing and heavily limiting it with an Apex Dominator."

Audience microphone layout for Super Bowl surround sound broadcast.

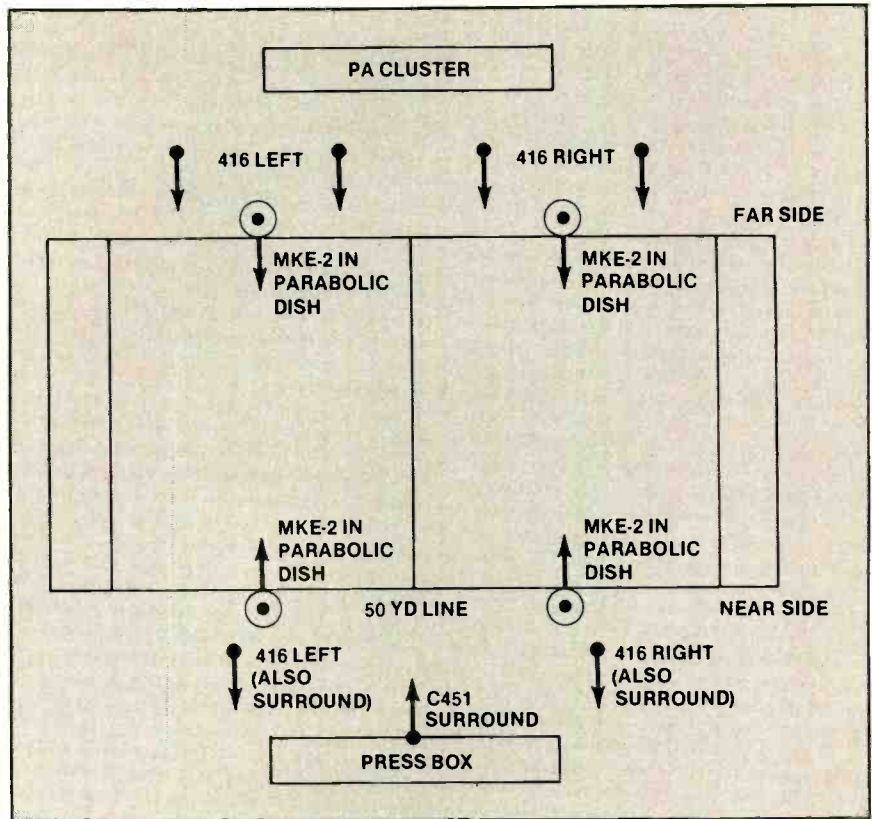
The original mix had far too much—15dB-20dB—dynamic range on material that had to play in the clear. This is *too* much for television; it just won't work," Bass says. "I *hate* that it won't work, but it won't."

In addition, he later recorded some overdubs of glockenspiel "glisses" and some clapping to give sound to on-screen performers at a certain point in the song.

Once all of the tracks were settled, Bass erased the original dbx-encoded stereo mix and a voice-count track which, he was assured, would not be used. This latter track was wiped because it had been printed hot, causing it to leak onto adjacent channels.

"Because I had tracks to burn, I tried to leave guard tracks around things that I felt needed protection," Bass says.

The 24-track tape contained drop-frame time code to synchronize with a 1-inch videotape containing an opening segment featuring George Burns, in addition to "Busby Berkeley" overhead shots of the dancers. Recorded on track one was 60Hz/30fps, non-drop time code to



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Photo by Larry Blake.

Bob Seiderman pictured at the 36-input Ward Beck console in mobile unit 11A.

help the fireworks crew's IBM PC synchronize the effects to music. As it turned out, for whatever reason, this code was not usable, and the crew ended up starting the program at the beginning of the music, with a manual override ability if necessary.

Also recorded on the playback tape was a cue track to be replayed over the PA, instructing the stadium crowd when to lift their seat cushions. The presence of a cue track in the PA feed obviously precluded the use of open mics to capture the live Super Bowl crowd reacting to the show.

To fill this void for the TV audience, Bass "pre-sweetened" crowd sounds recorded during the Rose Bowl game three weeks earlier. Two tracks contained a bed of crowd murmur and another pair had cheers in the appropriate places. At the end of the number Bass left a 15-second pad of taped crowd sounds to help Radulovich cross into the live mics in the stadium.

"The trick was blending the recorded tracks to the real crowd," says Bass. "What we had going for us was that Larry Estrin [president of Best Audio] had recorded the crowd sounds in the same stadium. That was the good news. The bad news was that his mic'ing—a single MS pair—was totally different from what Bob Seiderman used.

"During the first half I had to listen to the crowd sounds and, with the processing equipment available, try to match our tapes with his crowd. I couldn't make it perfect, but I think I got it close enough that it didn't bother anyone."

Prior to the air date, all music and crowd tracks were left relatively dry, or at least "too dry for the picture" for Bass' taste, resulting in the need for real-time processing.

Keeping in sync

Bass had the deceptively tough problem of keeping the sound coming back from the playback tape in sync for the TV audience. During Neil Diamond's pre-game rendition of the national anthem, Bass placed a 362ms delay on the taped music-send to Seiderman's truck, to put them in sync with the live vocal. A pair of Klark-Teknik DN-701 DDLs provided the needed delay.

Diamond's singing was late because he was standing approximately 362 feet from the PA cluster located on the far side of the field. So that the singer was hearing time-coincident sound from all sources, the PA house mixer had to delay the send to the floor slant monitors sited at Diamond's feet by 362ms.

Therefore, Diamond was singing in sync not only with the playback track as viewed on television, but also with the PA. For natural slap echo, Seiderman could only use the stadium mics located on the near side of the field; the four on the far side were hearing the music from the PA about 150ms before Diamond did!

The over-the-air delay for the halftime show was slightly less, 329ms, because the average performing area was closer to mid-field and the speaker cluster. The situation was further complicated by the fact that sound for the videotaped segment featuring George Burn's on-camera dialogue was coming from the 24-track, and therefore was subject to the 329ms air-feed delay. Putting him back in sync with the videotape required *advancing* the multitrack tape 10 frames for the whole halftime show, simply by setting an offset in the Adams-Smith time code synchronizer.

One possible solution to this problem—not delaying the air feed of the Burns segment—was precluded by the presence of a music intro to the live in-stadium show. Switching in the delay lines would have instantly created a 329ms gap.

Emergency back-up was provided by two independent systems comprised of Sony BVH-2000 1-inch VTRs, Otari MTR-90 series II multitracks and Adams-Smith System 2600 synchronizers. (All replay equipment was located in the Best Audio truck.) In addition to functioning as the master in their respective systems, the VTRs were locked to one another; pressing play on one video transport therefore started the whole operation.

The setup worked flawlessly during the Super Bowl broadcast. The contingency plan, if neither system had lock within the 10-second roll cue, was to change the speed reference of the 24-track decks from external to a line-locked 15ips. (During most rehearsals the systems locked within 3s.)

The only problem at this point would have been the critical sync on the George Burns introduction. Because Radulovich received stereo audio feeds from the pair of 1-inch machines, in addition to Bass's main stereo mix of the elements on the multitrack and an L-R surround send, in-sync audio could have been taken from the VTRs. As a precaution, Bass recorded onto the VTRs a composite mix of everything that would have gone over the air, including the music mix, crowd sounds and overdubs.

Using the videotape tracks, however, would have required two tough crossfades for Radulovich. The first would have occurred at the beginning, when Bass would have told him on the intercom that there was no lock, and that he would have to go to a VTR; and the second after the George Burns segment, when the multitrack audio would be brought back in.

Again, because the multitrack mix had to be delayed to keep the dancers in sync for the TV audience, these fades would have resulted in third-second audio jumps. The only problem later in the show would have been a tougher crossfade between live and playback dancing for CBS halftime director Duke Struck.

It was therefore decided that the production crew would stay with the multitrack tape, even if the MTR-90s were not locked to the VTRs. The excellent speed stability of all machines would have probably resulted in near-perfect sync; drift would not be a problem since the only critical point occurred at the beginning.

In a worst-case situation, with both 24-tracks failing to operate during the show, the very last backup would have been to use the two 1-inch videotapes. This solution had many problems of its own: lack of the voice cue track for the stadium crowd; the VTR output was not going through the DDL; and canned crowd sounds would be in the music being fed over the PA!

Dolby Surround sound

Dolby Surround matrix encoding—creating an Lt-Rt from the stereo and surround feeds—took place in TX using two Dolby DS-4 monitoring/recording units, one assigned to unit 11A's U.S. feeds and the other to 12A's. (The international stereo feed from 11A was distributed to the broadcast "pool" with no encoded surround information, because this would have necessitated the use of yet one more DS-4.)

Changeover, between which Lt-Rt went to the CBS air feed, was made on a set of two 20x1 Datatek AFV routing switchers, located in the TX truck. One

unit switched main video and stereo audio feeds. The other was devoted to backup feeds.

Although there were trim potentiometers on the left-right-surround sends to the DS-4s, the level in the TX truck was kept constant, and was determined by the surround feeds from mobile units 11A and 12A. Surround monitoring was provided only in Bob Seiderman's truck and in TX.

In addition, CBS and Dolby Laboratories set up a trailer where Rokosa and Gray could listen to the mix through a Shure HTS-5000 consumer surround decoder, in an environment somewhat resembling John Q. Public's living room. Also, of course, there was simply no room for Rokosa and Gray (or other interested parties) in Seiderman and Radulovich's cramped quarters.

There was no surround monitoring in Radulovich's truck, because there were the other listening environments (the monitoring trailer, TX and Seiderman's booth) functioning as his "surround-feed ears." Additionally, unit 12A's surround feed was static—Bass' L-R send—so there was very little that he could change.

Monitoring in all the trunks was set, coincidentally, to standard Dolby Stereo

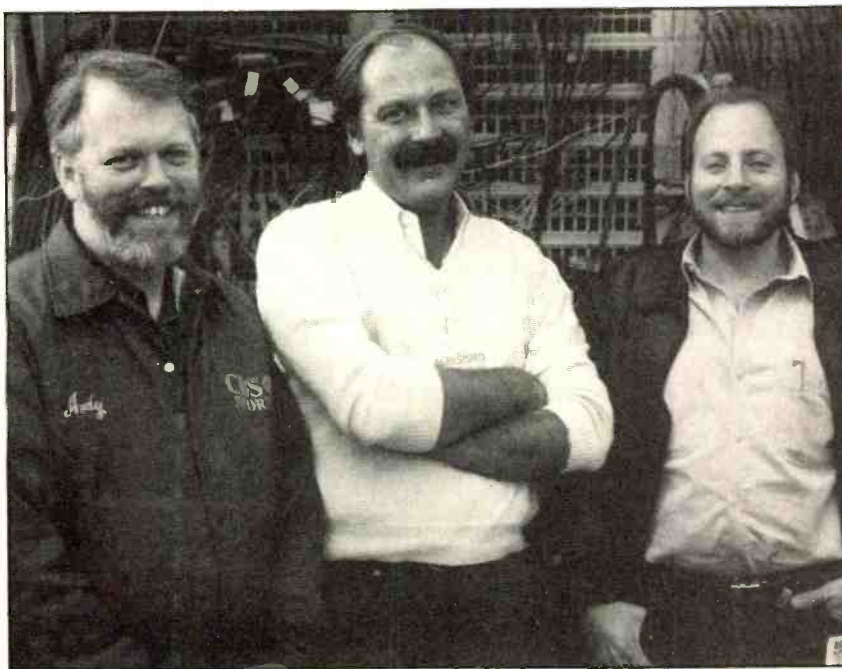


Photo by Larry Blake

CBS staff mixing engineers for the Super Bowl XXI broadcast include (from left to right): Andy Bass, who handled tape playback for the national anthem and halftime show aboard the Best Audio mobile; Mark Radulovich, who mixed the pre- and post-game Super Bowl Today show, halftime ceremonies and all post-game activities from mobile unit 12A; and Bob Seiderman, who mixed the game audio from aboard unit 11A.

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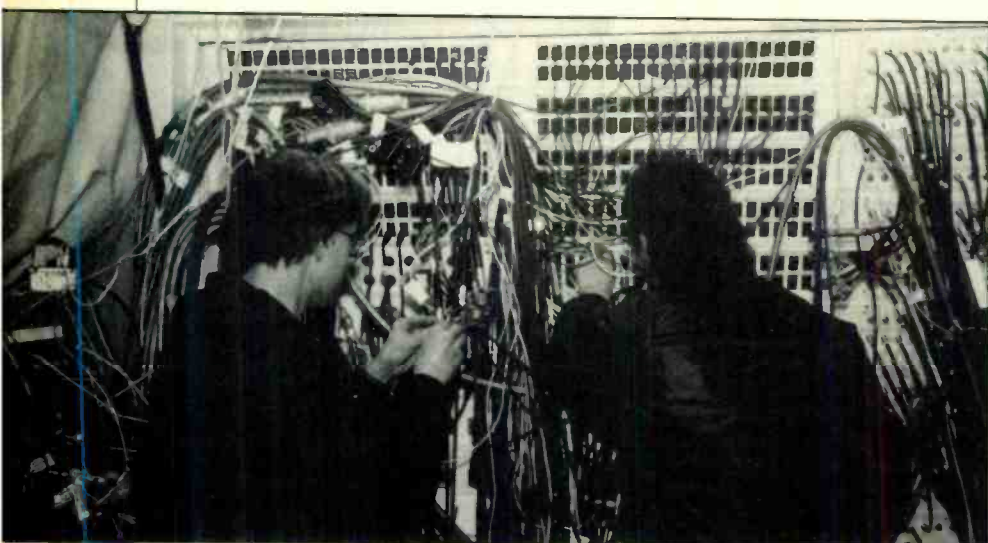


Photo by Larry Blake.

The rear of the TX truck served as a focal point for audio and video routing between the three mobile trucks handling mixing duties for the Super Bowl broadcast.

level of 85dB/c, slow, with pink noise at nominal zero-VU bus level. They arrived at this figure by playing music in Bob Seiderman's truck at a level he was most accustomed to; this turned out to be almost exactly 85dB.

Speakers provided in all CBS trucks were JBL model 4301s. Gray notes that "from our DS-424 [monitoring matrix] experience [in recording studios] we have found that the JBL 4300 series speakers relate quite well to the X [wide-range film monitoring] curve, if you don't have the speaker's presence knobs turned up too much."

Seiderman, who had a meter bridge and control head for 11A's DS-4 located in his booth, would often switch between stereo and surround stereo, to keep tabs on his surround feed. Another primary concern was phase and its affect on mono compatibility. The mono checks that Seiderman made were always from the stereo mode.

"It's very tough to go to mono from surround. There's such a drastic difference that the only way you would hear anything would be to have a level boost on the mono," Seiderman says. "Psycho-acoustically, you have no chance of knowing what you have."

"Once I knew I had good stereo," he continues, "I was confident of decent mono because of my experience with previous stereo broadcasts. As long as I checked the scope or the mono meter often; eyeballing them is something I can always afford to do. Of course, occasionally I *would* punch down to mono."

Regarding compression and limiting, both mixers used only slight peak limiting on the announcer submix. Seiderman had limiters on both the parabolic and crowd mic submixes; the thresholds

were set so high, he recalls, as to have them doing "nothing."

Sound and video transmission

There were at least four paths from the BNC video and XLR audio connectors at the rear of the TX truck to the CBS Broadcast Center in New York, the network's master control. The primary audio and video feeds (main and backup of each) were sent to Mount Wilson a few miles away from the Rose Bowl, Pasadena, on two microwave links. From there they were sent via diplexed telco lines to the Pacific Bell routing hub in Hollywood, and then on to CBS Television City on Fairfax Blvd.

At Television City the signal was encoded with closed-captioned subtitles placed in the video vertical interval. (You guessed it: using a court stenographer's keypad, someone is entering *very fast* a good portion of what the play-by-play and color announcers are saying.) The main satellite feed to New York was uplinked at TV City to Telstar 301 and downlinked at the AT&T CORAM station on Long Island, which sent it via land lines to the New York Broadcast Center.

The first-level transmission contingency plan, if the TV City uplink or Telstar 301 failed, was to send the signal to the AT&T TOC (Television Operations Center) for uplinking to Telstar 302. This feed was received at Group W in Stamford, CT, and microwaved to the Broadcast Center. Telstar 302 would also be used if TV City fell into a hole in the ground, with Pac Bell sending its feed from Mount Wilson directly to AT&T. The only difference in this case would be the loss of closed captioning.

The final, if-all-else-fails, backup was an on-site Ku-band satellite uplink that

would beam the signal directly to the RCA KI-11 satellite and on to the downlink station in New Jersey. A dish at TV City also saw this signal, allowing a caption-encoded uplink from there in the event of failure at either Mount Wilson or the Pac Bell hub.

Upon arrival at the CBS Broadcast Center, New York, all incoming video passes through a frame store to synchronize it with house sync. (At the Rose Bowl, house sync—composite video with burst and 7% video black—was distributed by the TX truck. An automatic changeover device received outputs from two generators, with instantaneous switching if one unit went down.) A delay usually does not have to be added to the audio if the signal, from the point of diplexing, has only passed through a single frame synchronizer. The resulting audio delay would be one frame or less, and probably would not be noticeable.

National commercials were inserted at two "co-ord" studios in New York, with a third studio used to insert regional/national spots. (The classic example is that a tire company would not advertise snow tires in Florida.)

The broadcast was then sent back to the local affiliates via CBS' normal 12 network satellite routes. These routes all come in handy when a network sports broadcast requires many different regional feeds. For the Super Bowl, only two main feeds were necessary: video plus mono audio and video plus stereo audio. (It should be noted that the mono feed was a composite of Lt and Rt, which were the only audio signals to leave Pasadena.)

Complete testing of all audio and video satellite feeds was done on the day before the game, in addition to spot tests completed a few hours before the pre-game show. For level setting, Tom Jimenez sent, at zero-VU, 1kHz left and right; 1kHz left-only and right-only for channel identification; 10kHz both channels as a check not only for frequency response, but also for phase alignment of both tracks throughout the broadcasting chain; and 15kHz on both channels for frequency response.

Jimenez next sent a series of educational tones: 1kHz from left-center-right-surround, which formed a circle on the oscilloscopes. This test tone sequence also checked that the level between channels was exact, because the Lt signal would be equal to the Rt. The same check was done with a 7kHz tone to not only see the affect of phase differences in the transmission path, but also because that is the frequency at which the filter in the surround channel starts to come into effect.



Photo by Andrew Waterman, Estrin Associates.

The audio control and tape machine area of the Best Audio mobile, where CBS engineer Andy Bass handled multitrack playback and mixing for the national anthem and half-time show. Equipment housed in the mobile include a Yamaha PM-2000-24 console, Sony BVH-2000 1-inch VTRs, an Adams Smith System 2600 time code synchronization system, Electro-Voice Sentry 100 monitor loudspeakers, an Ampex ATR-104 4-track and two Otari MTR-90 24-tracks.

Finally, pink noise was sent left-only, right-only, left-right and left-right with surround fading in and out. This signal would again indicate correct system polarity, in addition to watching the

oscilloscope Lissajous pattern go from a diagonal, straight line, to an out-of-phase ball and back again.

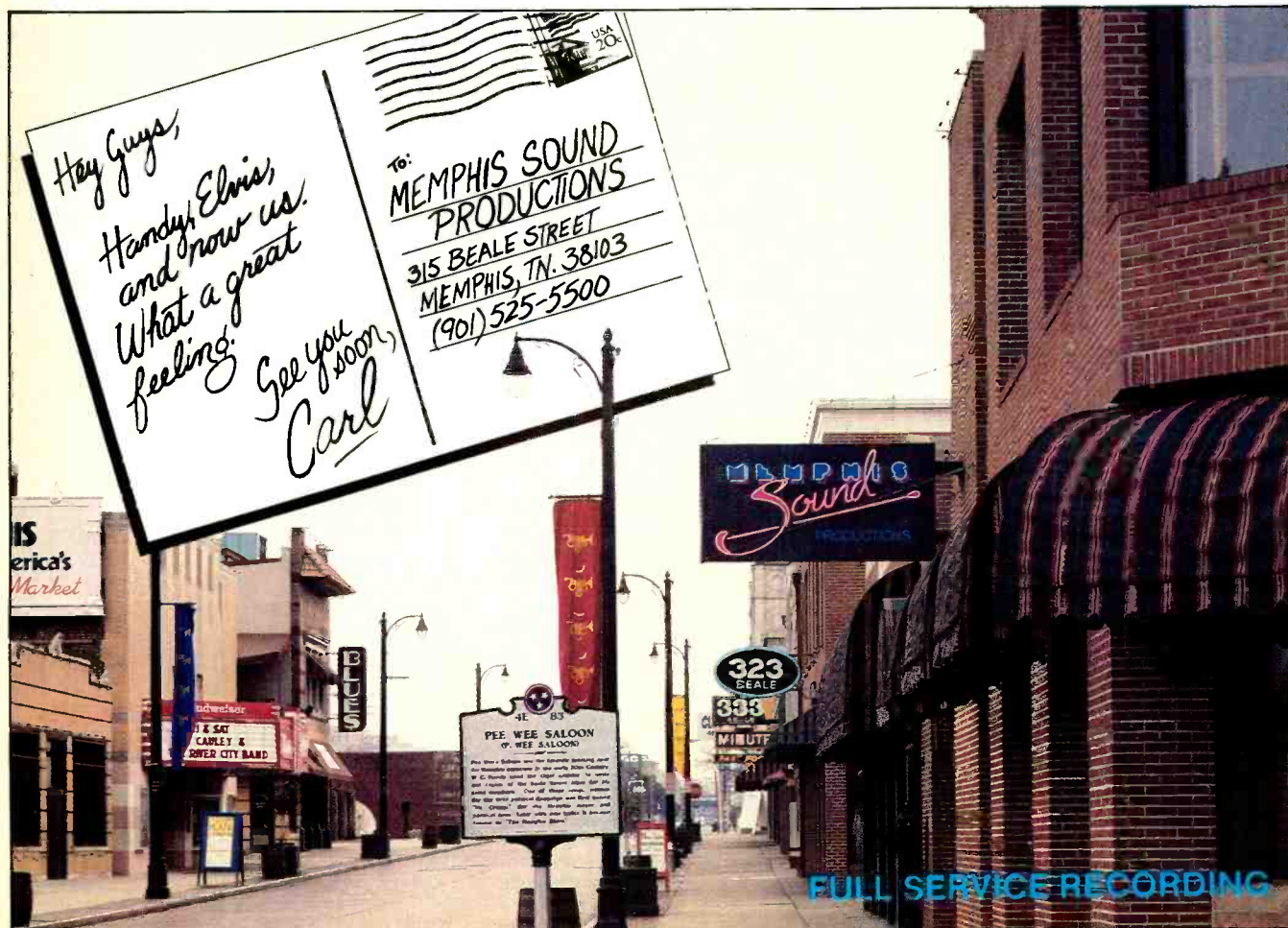
During most of the test procedure, the principal parties were linked via a

transcontinental conference call: TX truck at the site, Pac Bell hub in Hollywood, AT&T TOC in North Hollywood, TX at CBS Television City in Los Angeles, satellite downlink in Connecticut and finally the New York TX. In addition, there were people in the New York and Los Angeles CBS studios to ensure that all points of entrance and audio routing within the facilities were still at 0vu, in phase, etc.

By game time, 39 CBS affiliates had converted their transmitters to MTS, some for that day only. The sad irony here is the CBS owned and operated stations in the three largest markets, New York, Los Angeles and Chicago, broadcast Super Bowl XXI in *mono*.

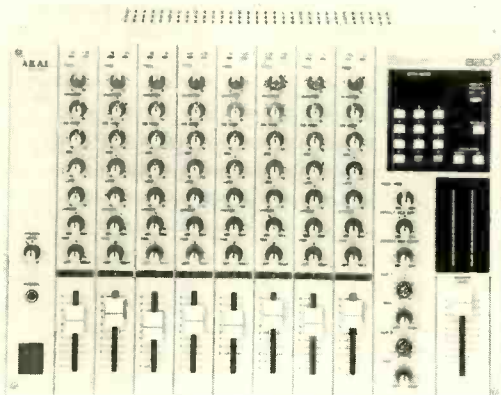
The good news, however, is that reportedly these stations will be installing MTS equipment in the transmitters and plants by the end of the year; the fourth CBS owned and operated station in Philadelphia has been broadcasting in stereo for the past year. In addition, the Dolby Surround aspect of the broadcast was considered such a success that CBS used surround and coding for the NCAA Final Four basketball playoffs.

R/E/P



Circle (32) on Rapid Facts Card

Akai MPX-820



MIDI-based 8-channel Mixer

By Paul D. Lehrman

The number of tasks MIDI is being called upon to perform these days seems to be increasing exponentially. Time-domain processing, equalization, signal muting and routing, and even lighting displays, are all becoming MIDI-controllable, as the hardware—and the ways that designers and users think about it—become more sophisticated.

One important aspect of such developments that has been slow in developing, however, is MIDI-controlled audio mixing. But now Akai has released the first of what promises to be a new generation of automated consoles.

The MPX-820 is a rack-mountable 8-input stereo mixer. Actually, MIDI is just one aspect—albeit probably the most significant—of the mixer's operation. In several ways, the MPX-820 represents a new level of low-priced studio automation. Like many more expensive automated mixing consoles, the device takes digital "snapshots" of all the control positions, one point in time, and then records the data into one of 99 on-board storage registers.

These registers can be edited, and then

recalled in any order by an on-board numeric keypad or by MIDI program changes; alternatively, they can be incremented (up or down) by foot switches or (up only) by pulses recorded on a special tape sync track. The contents of the registers can also be downloaded to a standard data cassette, and will soon be addressable through a computer, using software currently being developed by a number of third parties (see accompanying sidebar on page 74).

It is the completeness of data storage and conformity with the MIDI specification that makes the MPX-820 unique. Other mixers use MIDI to control channel on/offs and effects routings, and still others use their own proprietary codes—sometimes based on MIDI, but not totally compatible with it—to control individual channel levels. The MPX-820, however, is the first mixer to include channel and master levels, pans, send and receive levels and even EQ in its internal registers, and to make them accessible by standard MIDI commands.

MIDI applications

The advantages of such a system are not limited to the MIDI studio. In live performance, the mixer can be configured instantaneously from any MIDI key-

board. In a conventional small studio, the unit's tape-sync function can be used for time-based automation, provided there is an extra tape track available for the sync tone. The mixer would seem to be of most use, however, in conjunction with a sequencer in a MIDI-based studio to provide a degree of precision and flexibility in automated production not previously possible.

Normally, automated mixing consoles work off some kind of clock; they execute preprogrammed changes at specified times. The moves themselves are stored either in RAM or on a medium such as floppy disk, and are then triggered by a timing pulse, usually derived from a recorded track, like an FSK or time code stripe. Editing or changing the timing of the moves can be a cumbersome job, often requiring the user to modify *all* of the events in a song, or even restripe the sync track, just to change one move.

When the MPX-820 is used with a MIDI sequencer, however, moving or deleting an event is as simple as altering the position of a MIDI program change, and that change will have no effect on any other events. If the tempo of a tune needs to be altered, there's no need to retime each mixer move. A sequencer with cut-and-paste or loop functions will manipulate the mixer's instructions in the same way as musical data, making the process of reprogramming the mixer essentially invisible to the user.

Panel layout

Except for the numeric keypad and large, 2-digit LED display in the upper right corner, the MPX-820 looks like any other small 8x2 mixer. It fits into a standard 19-inch rack, and is 7U (12¼ inches) high. The eight channel inputs on the rear panel feature both low-impedance balanced and high-impedance unbalanced connectors, with pad switches for each. Direct input and output jacks are provided, and there is a trimpot for each channel on the front panel, along with green (-10) and red (+10) peak-indicating lights.

Each channel has a monitor send (pre-fader), effects send (post-fader), panpot and three EQ controls labelled High, Mid and Low. The EQ controls are marked +10 and -10 at their extreme positions. Neither they nor the pan controls have center detents. Channel levels are controlled by 100mm slide faders, calibrated from 0 to 10, with extra markings between 7 and 8, suggesting that the manufacturer considers this to be the optimum

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operating range.

The master audio section has a single fader on both channels, left and right effects-receive controls, and two aux input knobs, each with its own pan control (but again no center detents). There are no effects- or monitor-send masters. Two vertical, 12-segment LED bargraphs display left and right master levels, marked from -20 to +8. There is an uncalibrated knob labeled fade time, and a headphone jack with accompanying level control.

Rear-panel jacks are provided for the main stereo outputs, effects send (mono) and receive (stereo), monitor send and aux inputs (with pad switches), all unbalanced; footswitches to increment program changes up and down; tape sync input and output; and MIDI In, Out and Thru connectors.

The aux inputs are ostensibly for gang-ing together multiple units, but one could imagine many other uses for them, such as multiple effects returns, or even as two extra instrument inputs. (The literature available from Akai's U.S. distributor states that up to eight units can be ganged together; there seems to be no reason, however, why that number cannot be higher—and the manual makes no

mention of a limit.)

The programming section, Data/Mode, includes the previously mentioned LED display, 10 numbered buttons, A and B buttons, a memory-protect switch, programming-incrementing buttons (down and up) and a Manual switch that defeats whatever settings are in the memory and restores manual control to the mixer.

Electrical characteristics

Although this looks like an ordinary mixer, inside it quite clearly is not. With the exception of the input trims and headphone output knob, none of the front-panel controls actually control audio signals. Instead, they produce dc voltages that are sent to a digital processor at the rear of the unit. This processor then controls the audio circuits. It's an elegant system but one with some potential drawbacks.

One such drawback is that the resolution of each of the controls—unlike those in an analog board—is, of necessity, finite. Fortunately, this resolution is relatively small. All of the knobs—EQ, sends, receives and pan—have a resolution on the order of 0.1dB, although for the EQ and pan controls this gets broader as you turn them further from

the center.

The faders have a fairly consistent 0.5dB resolution at the top and middle of their travel, widening to about 1dB at the bottom. Fader travel is smooth and is laid out well: the change in level between calibration markings is about 6dB near the top of its travel, increasing to 9dB toward the bottom. At the extreme ends, however, there is no level change—either between 9 and 10 at the top, or 0 and 1 at the bottom—meaning that 20% of the travel is wasted.

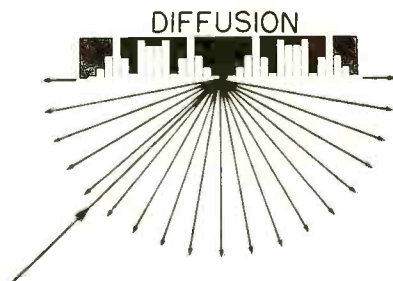
Except for the fader extremes, the mixer behaves so smoothly that it's easy to forget that the controls are not acting directly on the audio signal. Occasionally, there seems to be a slight sluggishness in the action of one control or another, probably caused by processing delay. It's only a minor annoyance, though.

The EQ and pan controls have their own eccentricities. The equalizers feature Baxandall-type circuits, with a bandwidth that increases as one moves a knob further away from its center position. (The effect is much more pronounced in the bass than the treble.) Panel markings indicate their range as +10dB, but the manual says they are ±15dB. In the case of the High control,

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APPLICATIONS

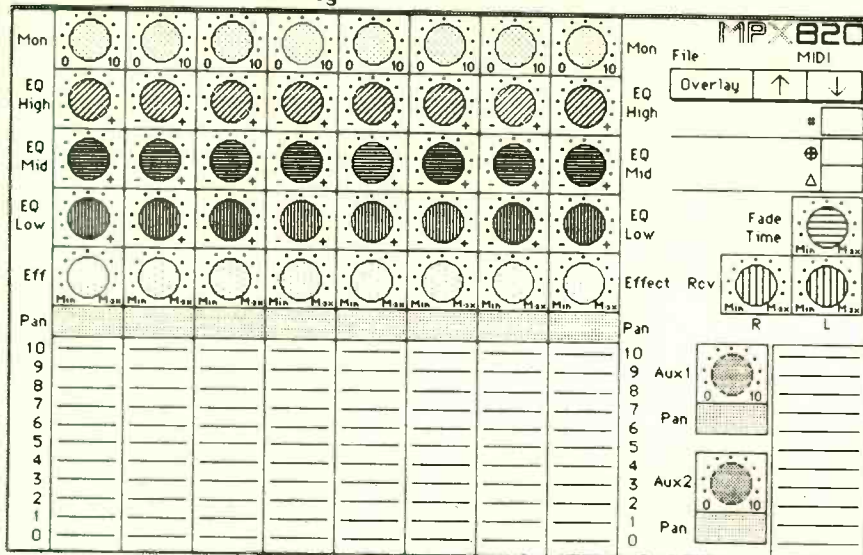
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Patent Pending

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External Control and Editing Software

Two software companies, Opcode Systems and Hybrid Arts, have announced editing programs for the MPX-820 that will allow control settings to be modified in an external computer and stored on disk. These settings can then be reloaded into the mixer's internal memory, just as one can do now with many synthesizers. Opcode's program runs on the Apple Macintosh, while Hybrid Arts' product is designed for the Atari ST series.

As can be seen from the accompanying screen dump taken from a preliminary version of the Opcode software, programs are stored in banks of 99, and any program can be singled out for editing, at which time a simulation of the mixer's control surface appears on the Mac screen.

the manual is correct about the levels, but incorrect about the corner frequency: it quotes 10kHz while in reality it is 8kHz. The Low control, on the other hand is +18/-16dB, and the corner frequency at the extreme settings is 50Hz.

The action of both of these controls is so broad at full settings, that both of them have a +5dB effect at 1kHz. The effect of the treble control is visible, in fact, down to 150Hz, while the bass control operates all the way up to 2kHz. The action of the Mid knob is centered at 1.4kHz (the manual says 1.5kHz—close enough), +12/-15dB, with ± 5 dB points at 400Hz and 4kHz.

An EQ circuit offering this much equalization on such a large portion of

the audio spectrum can play havoc with levels. Because the built-in level indicators only monitor the post-EQ signal, if you were to turn the midrange control all the way off, it would be all too easy to overload the input pre-amp without the +10 light being illuminated.

Worse, however, is the lack of center detents in these controls. The nature of the circuit is such that the unit is relatively forgiving of small changes at the center of the controls' ranges, but it makes the job of taking the EQ out of the circuit very difficult. This is an important deficiency in what is supposed to be, after all, an electronic-music mixer, in that many synthesists prefer to work without any EQ at all, making the appropriate tonal changes on the synthesizers themselves.

The pan controls, because they are less forgiving near their center positions, suffer even more from the lack of detents. They also appear slightly off—in the review unit the center position gave an output that was consistently 1dB to 2dB too high on the left side (but this is likely the fault of the summing amplifiers themselves). Unfortunately, there is no way to balance the left and right outputs, except by using the panpots.

In most respects, the mixer behaves the way it's supposed to. Effective input noise measured -118dB at the mic inputs and -90dB at the line inputs (unweighted), and frequency response (once you manage to null out the EQ) is quite flat from 20Hz, with a 2dB rolloff at 20kHz. Crosstalk is better than 70dB, both between the left and right outputs, and across the inputs.

Maximum output level is +20dBm, but some of the other choices for levels and

gain structures are a little strange. For example, the effects returns are 20dB less sensitive than the line outputs, which means that an effects unit needs to produce a reasonable output signal. The individual channel outputs are heavily attenuated before they reach the main mix bus, and it is impossible to drive a single channel so hard that it clips the main output. In itself, this is not bad, but it means that under most conditions the optimum operating range of the channel faders is fairly small, and on the high side.

Setting up programs

Programming the MPX-820 is a very straightforward procedure, as follows:

1. Turn off the memory-protect switch;
2. Press the manual button (or the zero key twice—register 00 is the manual register);
3. Set all the controls where you want them; and
4. Press A followed by the two numbers of the register in which you want to store the current setup.

The state of every control on the board, except the input trims and the headphone level, is recorded in an on-board register. To recall a setup, press the appropriate pair of numbers on the keypad, use the program-increment functions (either from the front panel, footswitches or a recorded sync track), or send the device a MIDI program change.

To modify a called program, move whichever controls need changing and store the program again (in the same or a different register); only the settings of controls that have been moved will be changed in memory.

Also stored with each program is the position of the Fade Time knob, a feature that allows the user to determine how long a called program will take to execute. It is a smart feature, in that it will do smooth fades to the desired console state from any previous state. The manual says that the knob (which is marked only min and max) can be set from 40ms to 30s, but the actual range of times is about 15ms to 20s. The automatic fades produced by the mixer are very smooth, regardless of length or complexity.

The mixer can be set to respond to data on any MIDI channel, and is always in Omni Off mode. It reads true MIDI program numbers—for example, what most synthesizers and sequencers send out as 1 is received as 0. This is a little confusing because you have to remember to subtract one every time you want to call a program externally. And if you send patch #1, it doesn't call a program: it puts the mixer in manual mode!

Editing is also straightforward, although there are some serious peculiari-

ties. The question of how to display the recalled programs on automated mixers has plagued their makers since day one. Some of the solutions have included moving faders, up and down nulling lights, or external video or LCD displays; there are probably many others most of us haven't heard about. Akai's solution is to do as little as possible.

To interrogate the virtual setting of a control in a particular program, you must physically move it. When you move past the correct point, the LED display blinks and an extra dot appears, showing that you have edited the program. If you move the control back, the display blinks again. The dot will stay on, even if you have reset the control perfectly, until you store the preset or call up another one.

Finding the correct null point of any control in this manner can be an exercise in frustration. The display only tells you when you've moved *past* the setting, not when you are *on* it. Then, like Zeno's paradox, while you can get close it seems as if you can never get it exactly right.

To make things worse, the unit's designers have decided that a physical control will not start to work at all until you move it past the null point—which means that a major change in a control position may have no effect whatsoever, if you happen to be moving in the wrong direction. You have to go back past the virtual position, and then reverse the motion again. This could be disastrous in a live situation where a level has to be brought down quickly, because it first has to be brought up!

In-use assessment

In-use within a MIDI studio, these drawbacks are not quite as serious as they might first appear. Generally speaking, editing a pre-programmed mix is often just a matter of moving one or two controls slightly; at that level of complexity, the system is reasonably usable.

In general, however, using the mixer left this reviewer feeling a little uneasy. The idea of a mix being a series of snapshots seems restrictive and detracts from the feeling that, like a piece of music, a mix should be a living, breathing entity. The programmable fade time helps quite a bit, but it makes the assumption that when you do a crossfade, you want all of the controls to move at the same relative rate. The maximum of 99 snapshots for any one piece of music also seems limiting, and precludes any great degree of subtlety in a mix—first, because there isn't room to store a large number of moves and second because storing each snapshot pretty much requires that you stop the music.

For quick-and-dirty production chores, however, the MPX-820 is fast and easy to set up, and very easy to learn. A typical short instrumental track, lasting about three minutes and involving seven synths and a drum machine, can sound pretty good with only 15 or 20 moves. The design does require you to think a little differently from the way you might be used to. Occasionally, you may find that doing a certain kind of move is essentially impossible, so you'll have to figure out a way to fake it.

The MPX-820 has one hidden feature which, unless you are a student of MIDI

implementation charts, you would have no way of knowing about; no where else in the manual, besides the chart, is it mentioned. The unit responds to MIDI controller #7, or MIDI Volume, by changing the master volume in real time. The only apparent advantage to this function is that initial and final fades can be handled conceivably by controllers instead of program changes, leaving two extra internal registers free for something else.

Overall evaluation

That the MX-820's EQ controls have no detents is annoying, but that the pan controls have none is criminal. The bandwidth of the EQ controls is far too wide to be effective. The single effects send is limiting, especially considering that low-cost, true stereo reverbs are now available. The monitor send could be used conceivably as a second effects send but, because it is pre-fader (and the effects send is post-fader), it would not be easy. The lack of master effects- or monitor-send controls means that setting those levels involves much guesswork. Unless your effects unit has a calibrated input-level indicator (and most don't), you have absolutely no way of knowing if you're overdriving the bus.

Since the input trim controls are not automated, precisely reproducing a mix (especially an old one) can be far more of a chore than necessary and, in some ways, this defeats the whole purpose of the mixer. The gain structure is quite restrictive, and could benefit from higher levels throughout. The fade-time control desperately needs calibration.

The editing system is clumsy and slow. Having external software will help enormously, and I hope that someone will eventually develop an editor/librarian for all of the many computers in general use by studios and musicians.

On the plus side, the unit receives high marks for being very easy to understand and to use, once the concepts involved are understood. The automation functions, as far as they go, work remarkably well. The sound quality is good, and the low noise figures are particularly impressive.

The MPX-820 is an important first step toward MIDI-based studio automation. However—and I'll be happy to listen to anyone who disagrees with me on this—I believe that the snapshot approach is the wrong way to go, especially considering that MIDI has literally hundreds of potential discrete controllers available, which if used properly, could do a credible job of running a complex mixer in real time.

R/E/P

Author's note: My special thanks to Walter Lenk for technical assistance with this review.

Technical Specifications

Frequency response: (Mic/Inst/Line): 20Hz to 20kHz, +0.5/-0.7dB.

THD: 0.04% at 20Hz-20kHz, +20dBm out.

EIN: -128dBm maximum.

Output Noise: 1 millivolt all faders down, master fader full up, all pans centered.

Max gain: Mic -70dBm; Inst -35dBm; Line -20dBm.

Crosstalk: 70dB at 1kHz.

Channel EQ: Hi 10kHz shelving, ± 15 dB; Mid 1.5kHz peaking, $Q=0.5$, ± 15 dB; Low 100Hz shelving, ± 15 dB.

Channel inputs: XLR Mic -55dBm to -20dBm 2.5k Ω ; Inst -35dBm to -6dBm 100k Ω ; Line -10dBm to +10dBm 10k Ω .

Aux inputs: Mic -50dBm to -15dBm 2.7k Ω ; Inst -30dBm to -1dBm 47k Ω ; Line -5dBm +15dBm 10k Ω .

Effects return: +15dBm (max input) 50k Ω .

Outputs: Main Left and Right +20dBm 150k Ω . Monitor (post EQ and fader) +20dBm 150 Ω ; Effects (post fader) +20dBm 150 Ω .

Programmable features: All levels, sends, pans, aux: 0.03dB resolution; EQ, (Hi, Mid, Low): 0.12dB resolution; fade time: 40ms-30s.

Memory: 99 sets of front panel settings; battery backup 10 years; time for tape backup for 2.2 minutes.

Dimensions: 482.6(W) \times 310(H) \times 203(D) mm (EIA Rack mount/7U).

Weight: 10.5kg.

Circle (97) on Rapid Facts Card

Studio Update

Northeast

Unique Recording Studios (New York) has installed two **Lexicon 480L** digital effects and reverb systems in Studios A and B. A **Lexicon 224X** was also installed at MIDI City, the facility's electronic music area. Two **Studer A-80RC** 1/2-inch 2 tracks were also installed in the studio's editing room. 701 7th Ave., 8th Fl., New York, NY 10036; 212-921-1711.

Omega Studios (Rockville, MD) is upgrading Studio C with an **Auditronics 24x16** console, two **Technics SL-P1200** CD players, a 1/2-inch 8-track, two **Yamaha SPX-90s** and a **Kurzweil 250** digital synth with four sound block modules. 5609 Fishers Lane, Rockville, MD 20852; 301-946-4686.

Production Masters (Pittsburgh) has upgraded its 24-track recording facility with the addition of an **Alpha Audio the Boss** computerized audio editing sys-

tem. 321 First Ave., Pittsburgh, PA 15222; 412-281-8500.

Cove City Sound Studios (Glen Cove, NY) has added a complete MIDI room to Studio B. Hardware includes two **Yamaha DX-7** synthesizers, a **Yamaha QX-1** sequencer, **E-mu Systems Emulator 11+**, **Minimoog**, **E-mu SP-12** drum machine, **Roland Juno 106** and a **Juno 60**. 7 Pratt Blvd., Glen Cove, NY 11542; 516-759-9110.

Power Play Records (Newark, NJ) has upgraded with a **Soundcraft 2400** console, **Nakamichi DMP-100** EIAJ-format digital processor, **Sony VO-5850** 1/2-inch VCR, **3M D-5000** character generator with 140-font library, **LinnDrum LM1**, **Simmons** drum set and **Yamaha DX-27** synth. Also added were a **Studer A-80 Mark IV** 24-track, **dbx model 166** gate/limiters, two **Roland SRV-2000** digital effects, **Eventide H-949 Harmonizer**, **ITC/3M Delta IV** stereo cart ma-

chines and three **dbx model 140A Type II** stereo noise reduction units for the studio's 3/4-inch editing decks. 198 Bloomfield Ave., Newark, NJ 07104; 201-481-0972.

Mark Custom Recording Studios (Clarence, NY) has purchased 50 **Nakamichi** cassette decks for in-house duplicating. 10815 Bodine Road, Clarence, NY 14031; 716-759-2600.

Sterling Sound (New York) has added a **Neve DTC-1** digital tape transfer console for CD pre-mastering.

"No other company has been able to create a board of this quality in the digital domain," says chief engineer **Ted Jensen**. "Neve has always been receptive to our suggestions."

By keeping all signal processing in the digital domain, the DTC-1 eliminates two conversions and maintains the original sound, Jensen added. 1790 Broadway, New York, NY 10019; 212-757-8519.

Southeast

Emerald Sound Studio (Nashville) has merged with other entertainment groups to form **Emerald Entertainment Group**. The studio has upgraded to include a 56-input **Solid State Logic SL4064** with Total Recall automation, a **Mitsubishi X850** PD-format digital 32-track and a **Studer A710** cassette deck. A new foldback system includes a cue mixer with eight channels and pan-pots, and overall low and high EQ with overall volume control.

Outboard equipment includes a **Publison Internal Machine 90**, **Lexicon 224XL** digital reverb and a **Lexicon 480L** digital effects system. Mics include an **AKG D-112** and **Bruel & Kjaer 4007**.

Other companies included in the merger are Robert Porter Management, Moore Publishing Company and the Missoula and Montana Music Club. 1033 16th Ave. South, Nashville, TN 37212; 615-321-0511.

Ron Rose Productions (Tampa, FL) has added a **Sony BVH-2000** 1-inch C-format VTR, a software update for its **Cipher Digital** time code synchronizer

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Send Studio Update announcements to: Sarah Stephenson, Recording Engineer/Producer, Intertec Publishing, 9221 Quivira Road, Overland Park, KS 66215.

Studio Update

Windmill Lane's new computer music studio—Studio 3, Dublin, Ireland.



controls on the console to be stored in non-volatile memory. *Bristol, England.*

Ireland

Windmill Lane Studios (Dublin) has opened a new computer music studio—Studio 3, designed by acoustician **Andy Munro** of Munro Associates. The new facility features the **Fairlight Series III**, PPG with **Waveterm**, **Oberheim Expander** and **Matrix 6 Super Jupiter** and **E-mu SP12** drum machine.

The studio is tie-lined to Studio 1 and to the facility's broadcast-standard video suites.

Studio 2 also has upgraded with the installation of a **Soundcraft TS224** console, a **Studio A800** 24-track and complete interior renovation. *4 Windmill Lane, Dublin 2. 01-713444.* **R·E·P**

Send Studio Update announcements to: Sarah Stephenson, Recording Engineer/Producer, Intertec Publishing, 9221 Quivira Road, Overland Park, KS 66215.



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Circle (36) on Rapid Facts Card

May 1987 *Recording Engineer/Producer* 79

New Products

3M HCDA 3000 digital recording system

The new 16-bit linear system provides up to 20 minutes of stereo recording on a data-style cartridge without companding. The system is capable of recording digital audio at either 48kHz or 44.1kHz sampling rate.

The single length tape cartridge is capable of holding up to 31 separate cuts. When recording time remains, an addi-

tional cut can be added without re-dubbing.

Front control panel features include an LED display featuring feedback on keypad instructions, cut number, track information, time, signal status, error status and overlevel conditions.

Input level is a quoted -19dBm to +18dBm and balanced output is +22dBm.

Circle (75) on Rapid Facts Card



Amek G2520 console

For multitrack recording and video post-production, the console is manually operated within signal paths set up from master status switches.

The console features VCA faders and is ready for computer-assisted mixing. It also features a standard microprocessor-controlled subgrouping.

Available in 24- and 48-track formats, with balanced bussing throughout, the console is available in two frame sizes to accommodate 40 or 56 inputs with on-board or external jackfields.

Circle (76) on Rapid Facts Card

Sycologic PSP-Percussion signal processor

In addition to conventional pad-to-MIDI and MIDI-to-trigger conversion facilities, the PSP includes eight dynamic drum pad inputs and trigger outputs, variable dynamic response, 10-octave MIDI note range, MIDI gate time variable up to 9.9 seconds, dynamic MIDI pitch bend generation, 50-user definable patches and sound processing.

The unit can also play information from up to eight drum pads, and a hi-hat pedal is both digitized and regenerated, enabling the unit to enhance a standard electronic drum kit's playability, the company says, while providing an advanced MIDI interface.

Each pad can be assigned several performance parameters, including MIDI program, note, gate time, pitch bend, feel and channel.

Circle (80) on Rapid Facts Card

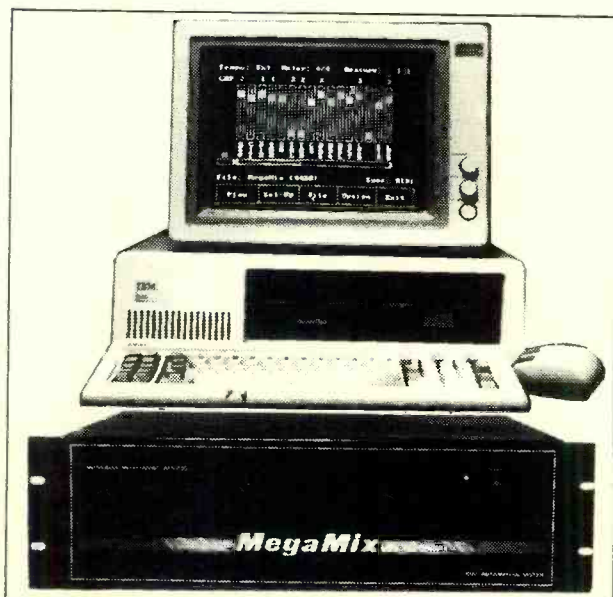
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Circle (37) on Rapid Facts Card

SCS model 2600A amplifier

The MOSFET design features automatic 2-speed fan cooling, coupled with a tunnel-type heat sink to further enhance reliability under abusive conditions, the company claims.

Output power is 350W/8Ω and 600W/4Ω per channel, both channels driven. Slew rate is a quoted 70V/μs. Input level controls, input connectors, 5-way binding post outputs connectors and ac circuit breaker are all rear mounted.

The 3U rack space unit is 12½-inches deep and weighs 49lbs.

Circle (82) on Rapid Facts Card

New Products

Orban 787A

programmable mic processor

The unit is fully programmable with 32 internal memory registers for instant set-up and recall. Features include 3-band parametric equalizer, compressor with adjustable release time, de-esser, noise gate and compressor gate, plus effects send and return.

A security code locks programming controls to prevent tampering once the unit has been set up.

Circle (83) on Rapid Facts Card

Valley International Autogate noise gate/expander

The new 2-channel unit uses auto slope and program variable release shape circuits to remove sound leakage or modify acoustic and electronic drums.

The high- and low-pass filter set can be used in the gate's normal audio chain or switched to the external input. In this way, the unit may attenuate a bandwidth controlled signal, or be used in a frequency-conscious manner to more selectively attenuate signals being processed.

A trigger generator circuitry allows electronic drums or synthesizers to be controlled by acoustic drum kits or other instruments which may be processed through the gate.

Circle (79) on Rapid Facts Card

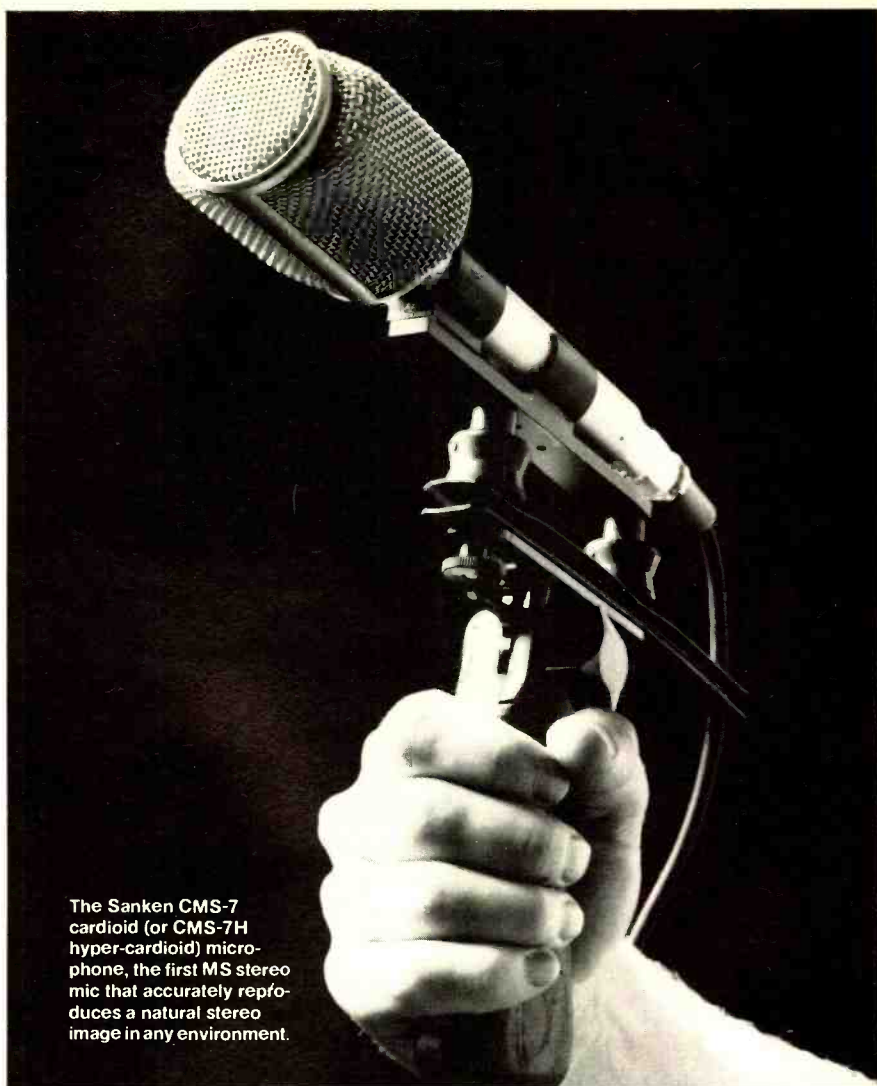
Solid State Logic introduces enhanced G Series Studio Computer

Designed to replace the existing E-Series Studio Computer software and hardware, the enhanced configuration features a removable Bernoulli cartridge drive that holds the equivalent of 80 E-Series floppies. In addition, the provision of 2Mbyte onboard RAM in the new system is said to dramatically reduce disk-access time.

The system also enables two mixes to be simultaneously replayed from memory/disk, and merged into a third. A new integrated processor and memory configuration is said to increase the speed of mix/merge functions by between 20 and 30 times, and also accommodate longer mix sessions.

The new upgrade is compatible with existing E-Series floppy data; disks produced on the G Series, however, are incompatible with existing systems.

Circle (96) on Rapid Facts Card



The Sanken CMS-7 cardioid (or CMS-7H hyper-cardioid) microphone, the first MS stereo mic that accurately reproduces a natural stereo image in any environment.

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Circle (48) on Rapid Facts Card

New Products

Noetek Elan recording console

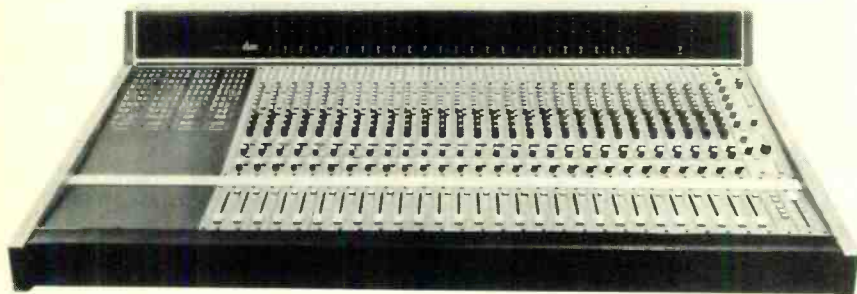
Channel modules feature microphone and line-level inputs, a 4-band sweep EQ section, six auxiliary sends, assignment to 24-multitrack buses and a bargraph meter.

The Elan system also features a second input through each module, doubling the console's capacity to provide 72 inputs

and 30 auxiliary buses in a 6-foot main-frame.

The unit is available in frame formats accommodating 28 or 36 module positions. Standard consoles feature gold ELCO multipin I/O connectors, leg set and patch bay using metal-frame Bantam jacks.

Circle (94) on Rapid Facts Card



Soundforms acoustic control system

The portable, lightweight sound panels can be used to control recording environments on location or in a studio.

The panels can be used independently, hanging on a wall or assembled to create baffle walls of any size and shape, the company says.

The panels weigh 5½ lbs. for a single 22-foot panel, and require no tools for assembly.

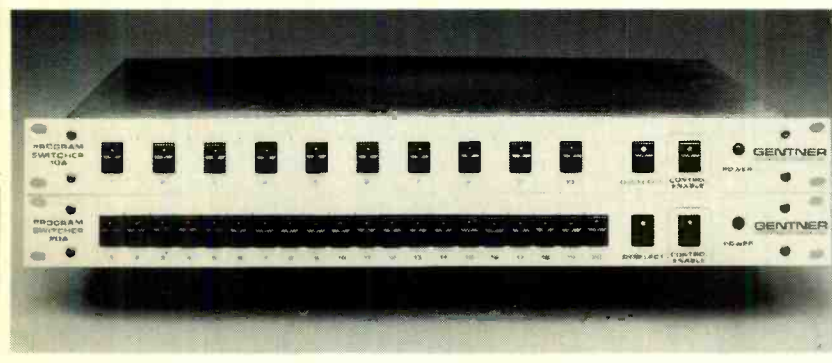
Circle (86) on Rapid Facts Card

Roland MKS-100 digital sampler module

The new unit is an 8-voice instrument featuring 128K of RAM memory, 12-bit linear sampling with 16-bit processing, 30kHz and 15kHz sample rates and total sample times of up to 4.4 seconds at full bandwidth. Quoted audio frequency range is 20Hz to 13kHz.

The MKS-100 can be assigned to dual or split modes with three key-split points, allowing simultaneous access to four samples. Velocity Detune, Velocity

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New Products

Switch and Velocity Mix functions also are featured. An 8-stage envelope generator can be controlled by velocity information.

Samples can be created by setting the loop and end points of a sample using the auto-loop function. Data can be downloaded and stored on a 2.8-inch Quick disk.

Circle (87) on Rapid Facts Card

Altec Lansing model 9812-8A loudspeaker system

The new mid-sized system features grille and tee-nuts for hanging applications.

Output is a quoted 123dB at 1m. One of the unit's three bass reflex ports is located on the back panel, for efficient replacement of the diaphragm assembly, the company says.

Also featured is a Mantaray II constant directivity horn mounted on a large format compression driver.

Circle (89) on Rapid Facts Card

AMR MCR 4/S cassette recorder

The 4-track recorder is said to be synchronization-capable and features a 25-pin synchronization/control port that provides full remote speed and transport control, tachometer and tally outputs.

This port enables the unit to act as a

full-function slave or master when used with the new AMR SyncController.

Other features include all-steel cabinetry, optional rack ears, zero stop, zero play, peak reading LED arrays for each track, variable pitch and LED counter/timers.

Circle (77) on Rapid Facts Card



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Circle (38) on Rapid Facts Card

New Products

Shure SM15 intercom microphone

The head-worn condenser mic is described as the first design to offer hands-free convenience and performance similar to a hand-held mic. Quoted frequency response is 50Hz to 15kHz, and a

SPL handling of 141dB.

Also featured is a newly designed double-braced headband with a durable grip. The unit comes complete with a 4-foot mic and 10-foot amplifier cable, a wind-screen and carrying/storage case.

Circle (91) on Rapid Facts Card

Orban model 464A Co-operator

The new unit comprises a stereo-gated leveler/compressor/HF limiter/peak clipper in one package. It automatically rides gain, controls excessive high-frequency levels and limits peaks, the company says.

The unit can be used with single instrumental tracks, voice or mixed program material and is suited for protecting

recording tape, broadcast cart machines, microwave links, cassette masters and sound systems.

Available as a 1U rackmount unit, the model 464A is switchable for stereo-tracking or independent dual-channel operation. Two LED bargraphs in each channel simultaneously display gain reduction and peak output level.

Circle (84) on Rapid Facts Card



Soundcraft Series 500 live-performance console

The new console features active feedback network input amps, which are said to produce lower distortion figures and improved noise performance.

Each input can be routed individually to 12 sends, switchable pre/post fade, in groups of four. As well as fader, PFL and cut, the monitor input module also features a dim button to help identify feedback problems.

The Series 500 monitor interfaces to the Series 8000 talkback system, allowing total communication between the house engineers and the stage.

Outputs are arranged in pairs on six modules. Each of the 12 outputs has switchable 3-band EQ, the mid-band being parametric/swept with variable bandwidth. The console is available in 16, 24, 32 and 40 channel frame sizes.

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New Products

Quantec QRS/XL programmable digital effects/reverb unit

Designed for stand-alone use, or connected to an IBM PC or compatible for enhanced control capabilities, the new unit features 90 factory-programmed presets, including flanging, chorus, sampling, delay and various digital filters. Connected to the external PC, all algorithm parameters can be adjusted under software control via a standard RS-232 interface. MIDI control also is featured.

Features include oversampling 16-bit A-to-D (2x) and D-to-A (4x) conversion; independent channel processing; linear-phase filtering in processor signal paths; 15kHz bandwidth of the processed signal; 32-bit internal resolution; equal-loudness mixing of direct and processed signals; and fully floating analog I/O section.

Currently available software for desktop or portable PCs and compatibles allows effects programs to be written off-line, compiled and then the controlling parameters downloaded to the QRS/XL via

a high-speed RS-232 serial interface. A total of 40 user-programmable memories are available for external effects generation.

The 1U rackmount unit features front-panel I/O level indicators, backlit LED display of control information and MIDI/RS-232 operation status. In stand-alone mode, effects programs are selected via a single knob, a configuration that is said to simplify everyday use in live performance, broadcast production and theater sound.

Circle (95) on Rapid Facts Card

Kintek KT 904Post mono-stereo converter

Designed for post-production applications, the unit features dynamic width controls for remote mounting on a mixing console, thereby providing full width control of the acoustic image.

Acoustical image width can be adjusted from narrow to a 180° spread.

The unit can be bypassed from the con-

sole, allowing direct feeds to be made without changing the input patch. Remote status indicators show whether the system is in an active or bypass state.

Circle (92) on Rapid Facts Card

Precision Design ROAM-8 mixer

Designed for portable location recording, the unit is housed in a poly case and operates on ac or dc voltages.

Input consists of eight mic or line channels with low, mid and high EQ. Two headphone jacks with individual level controls are provided, along with a limiter, telephone logic and pink-noise generator. Outputs can be interfaced with a line feed or an internal telephone transmission network.

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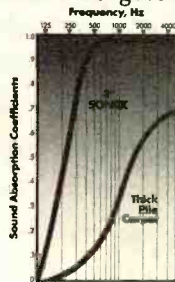
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Circle (44) on Rapid Facts Card

coming in June:

Recording
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First Annual Salary Survey

The primary editorial theme for the June 1987 issue will be "RE/P's FIRST ANNUAL SALARY SURVEY," the results of which will comprise tabular presentations of the following data:

- An analysis of salary trends throughout the audio production industry.
- An analysis by job responsibility and title.
- An analysis by primary type of business and regional location.
- An analysis of fringe benefits and society membership.

Other Features

System Interfacing

A technical look at the type of interconnect schemes necessary to ensure correct interfacing of audio recording and production equipment.

China's Recording and Production Industry

Comprising an analysis of Chinese facilities and techniques.

Electronic Music Production

An overview and future prediction regarding the types of MIDI-based and digital production capabilities that will be required for EMP sessions.

Production Viewpoint: Pat Williams

A leading film and TV music composer, detailing the recording of a new big-band recording produced by Phil Ramone using digital multitracks and mastering.

Plus our regular departments

- Managing MIDI
- Sound on the Road
- Film Sound Today
- Living with Technology
- SPARS On-Line
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- New Products

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AHB SIGMA SERIES

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Top Notch for Your Bottom Line

A top-notch performer, the new Studer A807 has the features and sound quality you need for demanding production applications. And, as the lowest-priced Studer, it also looks good on your bottom line.

The Fruits of "Trickle-Down" Technology. Over the past five years Studer has developed a whole new generation of micro-processor-controlled ATRs. Now, with the A807, budget-minded pros can find this new technology in a compact, ingeniously engineered package.



A Summary of Top-Notch Features:

- Digital setting and storage of audio alignment parameters
- Tape shuttle wheel
- Zero locate, autolocate, start locate, and loop functions
- Multiple timer modes
- Programmable keys for additional locator addresses, fader start ready, or lifter defeat
- Backspace (momentary rewind to play), library wind, varispeed, and reverse play
- Three tape speeds
- Microphone input with phantom powering
- Complete monitoring facilities
- RS 232 port for external computer control
- ...and the list goes on!



Sonically Superior. For superior high frequency dynamics the A807 audio electronics incorporate advanced phase compensation and Dolby HX Pro.™ You won't sacrifice the top end of your sound to enjoy a great bottom-line price.

Tough Stuff. The A807 is 100% Studer, with a die-cast chassis and head-block, rugged AC spooling motors, and a new brushless DC capstan motor. Manufactured with Swiss precision, this ATR is designed to get the job done faithfully—even in adverse outdoor remote assignments.

Suit Yourself. The rack-mountable A807 may be ordered with optional wooden side panels and handles, or in a roll-around console with padded armrest. A wide range of remote controls and options make it suitable for practically any application.

To find out more about this new top-notch performer, call or write for complete information and the name of your nearest Studer Professional Products dealer.



STUDER REVOX

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