October 1986

Recording

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Record Plant, Mk II
Page 30

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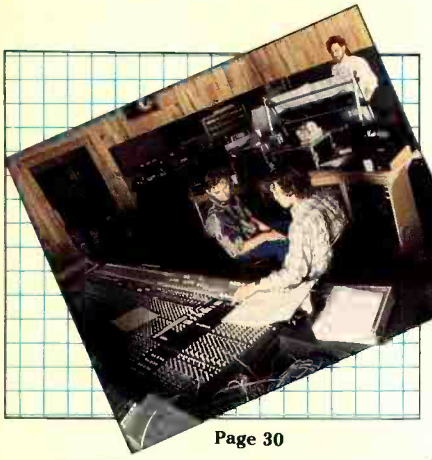
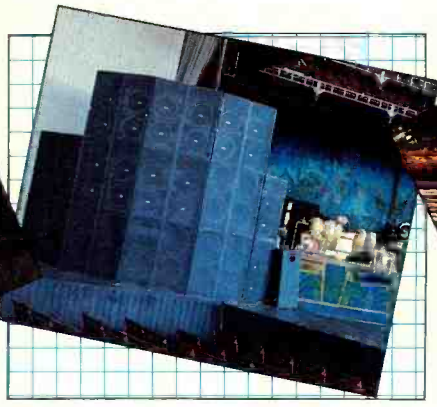
Contents

Recording ENGINEER/PRODUCER

October 1986 Volume 17 Number 5



Page 30



Page 80

Page 96

Audio Engineering Society 81st Convention

This year's theme is "The Digital/Analog Fusion: A Rainbow of Technology." **RE/P**'s pre-show coverage of the L.A. convention can help you plan your visits to the more than 480 booths and demonstration rooms, plus five days of technical sessions and papers.

Other Features

Departments

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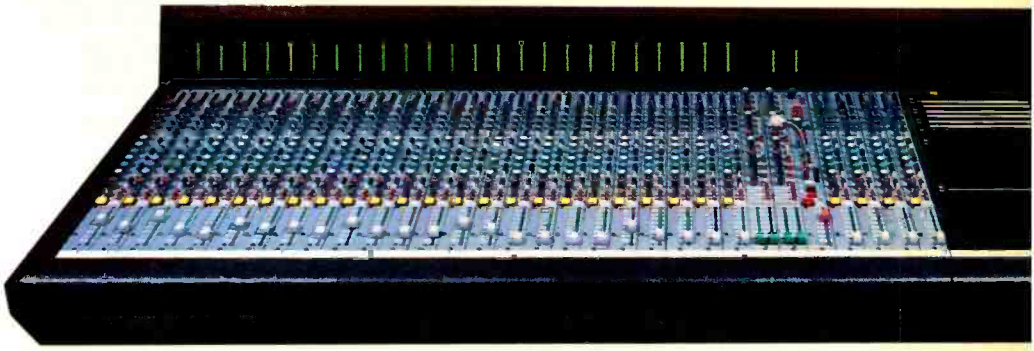
The producer's choice . . .



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Maybe it is AMEK's reputation for reliability. Or maybe it's the ANGELA's wide variety of frame configurations and the ease of access to all of the console's controls. Or it could be the fact that an ANGELA gives you all of the features of a computerized console without the excessive cost. The real reason AMEK has supplied so many ANGELAs to today's leading producers is its outstanding sonic quality.

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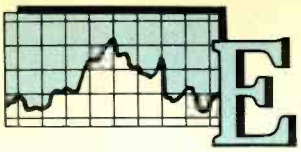

Editorial

Console Topography

It will come as no great surprise to just about anyone working at a recording, audio-for-video or film re-recording facility that console topographies are becoming extremely complex. With the move these days toward more intricate tracking and mixdown sessions—and a growing emphasis on stereo and multichannel production for video and film—the number of control and signal-processing functions that need to be handled by consoles have reached mind-numbing proportions.

Although the advent of in-line designs has helped to reduce their physical dimensions, the centralized role that consoles now play in the control room has resulted in some inefficient compromises. Not only do we expect automation systems to memorize time-dependent level and mute information from upwards of 32 or more input channels, many users now consider automated recall of EQ changes, effect-send levels and group assignments to be absolutely essential. Add a time code synchronization/editing system plus some MIDI-equipped outboard processors, and the control of all this hardware becomes extremely demanding.

And I'm not just referring to the technical considerations of affecting changes in peripheral systems and devices—and heaven knows that a standardized control bus and protocol would go a long way toward simplifying that whole can of worms. Interface architecture aside, I am more concerned with the ergonomics of console layout and how all this myriad information is to be displayed to the engineer.

At the heart of the problem, I would suggest, is that each of these control functions has been added to rather than integrated within the basic design of a console. The end result is a system of discrete microprocessor-based boxes with widely divergent control surface and display designs. And very few, if any, of these subsystems are designed to communicate directly with one another, except through the intervention of the engineer or producer.

A possible way around the problem would be to redefine the console's role as simply a tool for manipulating and modifying audio signals, and concede that it really does serve a very essential centralized control function—if only because that is where the human running the whole operation is located.

Just as form follows function, once we begin to conceive of this centralized role played by the console, and possibly just add the conventional audio manipulation operations to a growing list of duties it should handle, we might have the opportunity to rethink the best way of designing a console to serve these complimentary functions.

While it goes without saying that assignable and virtual consoles will have certain built-in advantages over conventional designs—if only because their functionality can be remapped under software control—there is still no getting away from the fact that just about all mixing boards suffer from fundamental drawbacks in operator convenience.

Instead of simply adding more and more functions to an already overtaxed device, it makes more sense to fabricate a central control system and let the audio functions be handled by a dedicated subsystem. By all means site the controlling computer at the center of the control room—simply because that is where we need to be located to hear the signals we are manipulating and processing—but some of the audio functions could be relegated to a control surface off to one side of the main monitoring axes.

What I envision is a console of roughly the same size as we are using today, but with centralized control and display of relevant subsystems actually built into the topography, rather than tagged on as a remote control, or VDUs mounted in the monitor wall. I would expect to see page-driven menus appear on a master video monitor located at the center of the console, which would allow a master time code, MIDI Sync or MTC track to be used for all timing instructions.

By slaving all control functions to a master time base reference, the sequencing of events—level and mute changes related to audio and/or visual events, as well as EQ, reverb and dynamics changes—could be sequenced and edited more easily.

Of course, there are some extremely elegant systems currently available that use time code as the basis for recording and replaying automation data. But by lightening the control and display load of the console surface, we could effect some major changes in efficiency.

In subsequent issues I'll be looking at specific design approaches. Meanwhile, I encourage **RE/P** readers to call or write the editorial office with ideas and concepts of their own.

Mel Lambert
Editor

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News

Solid State Logic joins UEI Group

The move, announced in late August, brings the SSL Group together with Quantel, the largest member of UEI and a major supplier of digital video post-production graphics, special effects, frame store and animation systems.

According to Peter Michael, UEI chairman, "The combination of technical, marketing and financial resources created by this merger will enable the UEI Group to respond thoroughly to the audio, video, broadcast and film industries' needs for increasingly sophisticated and efficient production equipment." He also said that the merger is intended to advance the development of all-digital audio/video production systems.

Colin Sanders, who will remain chairman and managing director of Solid State Logic, added that the "merger allows detailed cooperation between a number of truly formidable engineering teams. In the process, I believe that it ensures a very bright future, particularly in the digital realm, for professional sound and vision technology.

"In the tradition of UEI, each company in the group will continue to operate in a largely autonomous manner," Sanders explained. "We all recognize, however, that the needs of our clients will best be served by the development of integrated systems running compatible software, all designed to allow efficient coupling between every aspect of production and post-production. This will prove increasingly true as we progress toward high-definition television with multichannel sound capabilities."

In the year ended March 31, 1986, SSL's turnover was £18.6 million (approximately $27.9 million) with a pre-tax profit of £3.3 million ($4.9 million).

For the future, Sanders said that SSL is heavily engaged in digital audio research and committed to the production of a Digital Studio System. The project is based on a proprietary 24-bit audio processor capable of handling more than 1,000 million instructions per second. The proposed Digital Studio System will incorporate disk-based audio storage and editing capabilities, as well as user-definable digital audio processing and switching capabilities. No release date has been announced for the all-digital system.

Industry organization formed to raise money for deaf children

The primary purpose of the Distinguished Engineers Audio Federation is to have fun and be able to laugh at oneself," according to David Hadler, DEAF secretary. "Its goal, though not insignificant, is to raise money for deaf children."

Modeled loosely after the British organization with the same acronym, DEAF meets monthly "to poke fun at the pro audio industry with lighthearted abandon," Hadler says.

Donations will be made to various charities, according to Laurel Cash, DEAF president. "We will choose a charitable organization that doesn't already receive money from any other group affiliated with the music industry."

Other DEAF principals include Bruce Botnick, vice president of Digital Magnetics, and Guy Costa, treasurer of Motown/Hitsville Studio.

Formed last July, DEAF so far has operated generally as an informal affair.

Recording
ENGINEER/PRODUCER

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

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AUDIO LOGIC

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However, early next year it plans to organize and host "The Bogus Awards of the Year" ceremony where DEAF will bestow awards in the following special categories: The Largest Phone Bill for an Album Project; Most Reference Disks Ever Cut; Most Gadgets in a Producer's Car; Most Complaints by a Producer; Longest Down Time for Equipment Maintenance; 3-inch Tape Award for Bad Ideas; Most Studio Time for Non-charting Single; Phone Call Non-return of the Year (2-month minimum); Most Remixes; Most Overdubs; Most Cancellations; Largest Accounts Receivable; and Late-Again-for-a-Session Excuse of the Year.

Frank Zappa is scheduled to appear as keynote speaker at the awards ceremony with all proceeds of the accompanying dinner going to charity.

For further DEAF membership information, contact David Hadler at: 213-459-2743.

APRS guidelines for labeling master tapes

With terms such as Session Master, Production Master, Record Master, Cassette Master and, more recently, CD Master, it is not surprising that record companies sometimes have difficulty tracking down the correct "master" tapes.

To help reduce such problems, the Association of Professional Recording Studios, England, has issued a set of guidelines to clear up some of the anomalies.

Working in conjunction with its Producer's Guild, the APRS has decided that seven colored label designs would cover the most frequently required terms. In addition, a blank label has been added for special requirements, such as Backing Tracks. Examples of the proposed labels are as follows:

- *Session Tape:* For either the original multitrack or 2-channel recording.
- *Original Master* (Edited/Remixed): Mixdown from a session tape, or a first-generation copy when such a copy is modified by editing, so that it becomes a master in its own right. The space after the words "Remixed" can be used to indicate that the tape contains, for example, a 12-inch mix, use for U.S. release, D.J. version, etc.
- *Production Master* (Vinyl/Cassette/CD): Indicates that the tape is to be used for making production parts or export copies. If the production master is equalized for vinyl and cassette manufacture only, then only those boxes would be marked.
- *Not for Production/Listening Copy:* These labels will be bright yellow, in the style of a security barrier or freeway hazard sign, and are intended to show clearly that the recording(s) are *not* suitable for production use.
- *Export Production Copy* (Vinyl/Cassette/CD): Indicates a tape that is a copy for overseas distribution only, and should be marked in a similar way to the production master.
- *1:1 Safety Copy of:* An exact copy of the source tape, which should be detailed on the label.
- *Blank Label:* For all applications where none of the above descriptions are appropriate.

The APRS labels will measure 3"x0.5" and, in addition to the bold wording on each, the text will be duplicated at the left-hand end. By affixing each successive label slightly to the right, rather than directly over its predecessor, the history of the tape can be established. For example, an original master that has been approved for transfer "flat" would therefore become a production master simply be relabeling. However, by leaving the original master legend exposed, a client would know that only *one* tape existed if, for example, he or she wanted to use the original master at a later date.

Because the APRS predicts that there are two different requirements from the studio community—those of recording studios and those of post-production facilities—a set of six labels has been prepared for each sector. The studio package will contain: Session Tape, Original Master, Production Master, 1:1 Copy, Not For Production, and Blank. The second package will omit Session Tape and Original Master, offering in their places Listening Copy and Export Production Master labels.

Once the use of such labels has been fully established in U.K. studios later this year, the APRS intends to approach other associations, including SPARS, to obtain international acceptance for the labeling scheme.

Further details are available from the APRS secretary, 23 Chestnut Avenue, Chorleywood, Hertfordshire, WD3 6LT, England; 01144-923-77907.

Harrison Systems delivers first quartet of Series 10 consoles

Shipments of the automated systems began with delivery of an 80-input system in May to Denny Jaeger Productions, Oakland, CA. Other deliveries were made to Mike Oldfield Music, England; The Marquee Studios, London; and Westlake Audio, Hollywood. All four systems are equipped with the company's 80Mbyte hard-disk automation system, which provides full dynamic automation of all signal-processing functions.

Further deliveries and installations of Series 10 consoles are scheduled for Roger Savage's facility in Soundfirm, Sydney, Australia. Current plans call for the production of two Series 10 systems per month through the balance of 1986, Harrison says.

Anchor Audio purchases ROH

The entire ROH operation has now been moved from Atlanta to Anchor Audio's west-coast facility in Torrance, CA. ROH has manufactured intercom systems, audio-distribution networks and monitors for the last 18 years.

Dan Garrigan, previously western regional sales manager for ROH, has been appointed national sales manager for the ROH division. Jim Van Waay, president of Anchor Audio, says that the purchase of ROH audio products is to "further increase Anchor's involvement in the broadcast and production markets."

Compusonics introduces optical disk recording system

The recently announced optical, disk-based system will record over two hours of stereo audio per double-sided disk. The write once-read many (WORM) recorder, model DSP-1000, is expected to be available in November.

The company has formed agreements with Optotech, N.T.I., and Versatronex to manufacture the recording system and disks, David Schwartz, president, added. Production is scheduled to begin in October with the first completed systems reaching selected dealers by November 10.

The user will be able to determine the amount of audio recorded on each disk, from 37 to 256 minutes per side, depending on the source material and the CSX data compression rate selected.

Via a custom interface, scheduled to be made available next year, the DSP-1000 will randomly access sound files stored on the disk and perform music editing.

flexibility...

Audio consoles were once designed for particular applications. You decided up front what type of clients you were going after, and then picked a console accordingly, keeping your fingers crossed that the clients would approve.

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News

Export Award for Klark Teknik

The Queen's Award for Export Achievement has been awarded to Klark Teknik. The U.K. award was received on behalf of the company by its marketing director, Gaston Goossens.

Founded in 1971 by Philip and Terry Clarke, the company is described as Europe's largest manufacturer of graphic equalizers for the music and broadcast industries, as well as supplying delay lines, digital reverberation and special effects processors.

Discovery Systems begins CD manufacture

In late-July, Discovery Systems officially initiated its optical and Compact Disc manufacturing operations in Dublin, OH.

"The phenomenal demand for CDs worldwide has caused a crucial need for skilled manufacturing operations," says Jeffrey M. Wilkins, company president. "With the rapid advances in Compact Disc technology offering video and text capabilities, as well as audio, the manufacture of optical disks is clearly an exciting opportunity.

"Compact Disc manufacturing is a highly complex process with little similarity to record pressing or tape recording. Our new Facilities Management plan, a service that provides planning design, construction and operation of an independent, modular unit within the main manufacturing facility, allows record companies to move easily into the technology by relying on experienced personnel.

"In addition to ensuring production control, the plan will allow clients to adjust production based on current demand and to prioritize for timely distribution of a new artist or seasonal release."

A complete CD manufacturing "module" will contain molding bays, metallizing, coating and printing. Packaging can be supplied as a separate service. The plant's initial mastering capacity has been set at 36 million CDs per year.

UK work-permit situation for American producers remains unclear

Following a recent meeting of the Association of Professional Recording Studios' executive committee in London, it was reported that producers and musicians who want to record in Britain may still have to secure work permits.

As reported in the August issue, the British Home Office originally stated that a work permit was *not* needed for visiting producers/musicians if they were employed by an American record company and were visiting Britain only for a short period to record at a U.K. facility.

Subsequent to this ruling, however, the Immigration Division of the Department of Employment pointed out that, although it received guidelines from the Home Office, it is up to the Department of Employment to decide how these guidelines are to be interpreted. The APRS was informed that perhaps, it had been misled and work permits were needed in almost all cases.

In the case of producers and engineers, the APRS points out, the crux of the problem is that most of them will indicate that they are visiting Britain to *work* in a studio whereas, in fact, they are *hiring* a U.K. studio. (After all, the association reasons, you don't need a work permit to "hire" a hotel room or a car.)

This point applies *only* to producers and engineers who are being paid in the United States by a U.S. company and not if a U.K. record company has invited them to produce a session; in the latter case, a work permit is certainly needed. Although the position of American musicians working on recording sessions in Britain has still not been clarified, the association says that it is currently working on a solution.

SPARS expands engineer test program

The Society of Professional Audio Recording Studios has expanded its National Studio Exam for recording engineers. The SPARS exam was launched last year with a grant from Sony Corporation and created by the Educational Testing Service (SAT and PSAT).

"The SPARS exam is now a recognized educational gauge for the entire audio industry," Gary Helmers, executive director, said. "For personal evaluation, it's a great tool for the academic student, the seasoned professional engineer, or the beginner. You can't fail the exam; you only learn where your strengths and weaknesses are.

"We are finding that studio owners like the idea of a test profile from prospective employees. An exam score cannot replace a personal interview, but it has become part of the new professional approach."

Beginning in December, educational institutions and recording studios will be permitted to administer the SPARS exam under the society's supervision. SPARS will assist the school or studio with publicity and registration, provide test materials and administration instructions, and score the exams. Additionally, centers have been established in New York and Los Angeles for regular monthly testing.

The multiple-choice exam covers session planning, equipment maintenance and operations. SPARS also sponsors a nationwide internship program which, it is claimed, has resulted in 100% job placement.

People

Correction

In the August letter to the editor, "MIDI in the studio," page 12, we inadvertently misspelled the name *Rod Funston,* general manager of 39th Street Studios, New York.

Jeffrey N. White has been promoted to assistant national sales manager, professional products, at **Audio-Technica U.S.**

Jacquelynn Hebrock has been named product manager for **Audio-Technica U.S.**

Steve Landin recently joined **Amek Systems and Controls** as head of test at the company's Salford, England, headquarters.

W. Kent McGuire has been appointed VP of sales and marketing at **Sound Technology**.

Phil Wagner has joined **Rupert Neve** as eastern regional sales manager.

Clark D. Nail has been named manager, production design engineering, at **Altec Lansing**.

K-Muse has promoted **G. Bob Connelly**, currently VP sales and marketing, to oversee all sales and to direct advertising and promotions. Also, **Don Bird** has moved to national sales manager; **Rick Carlson** is now in charge of direct inquiries as artist relations manager; **Arne Schulze** will head up the Sound Composer's Series as director of the SCS division of sampled software; **Brad Cox** has been hired as national service manager; and **Peter Arkin** will oversee purchasing and production as manufacturing manager. R-E/P

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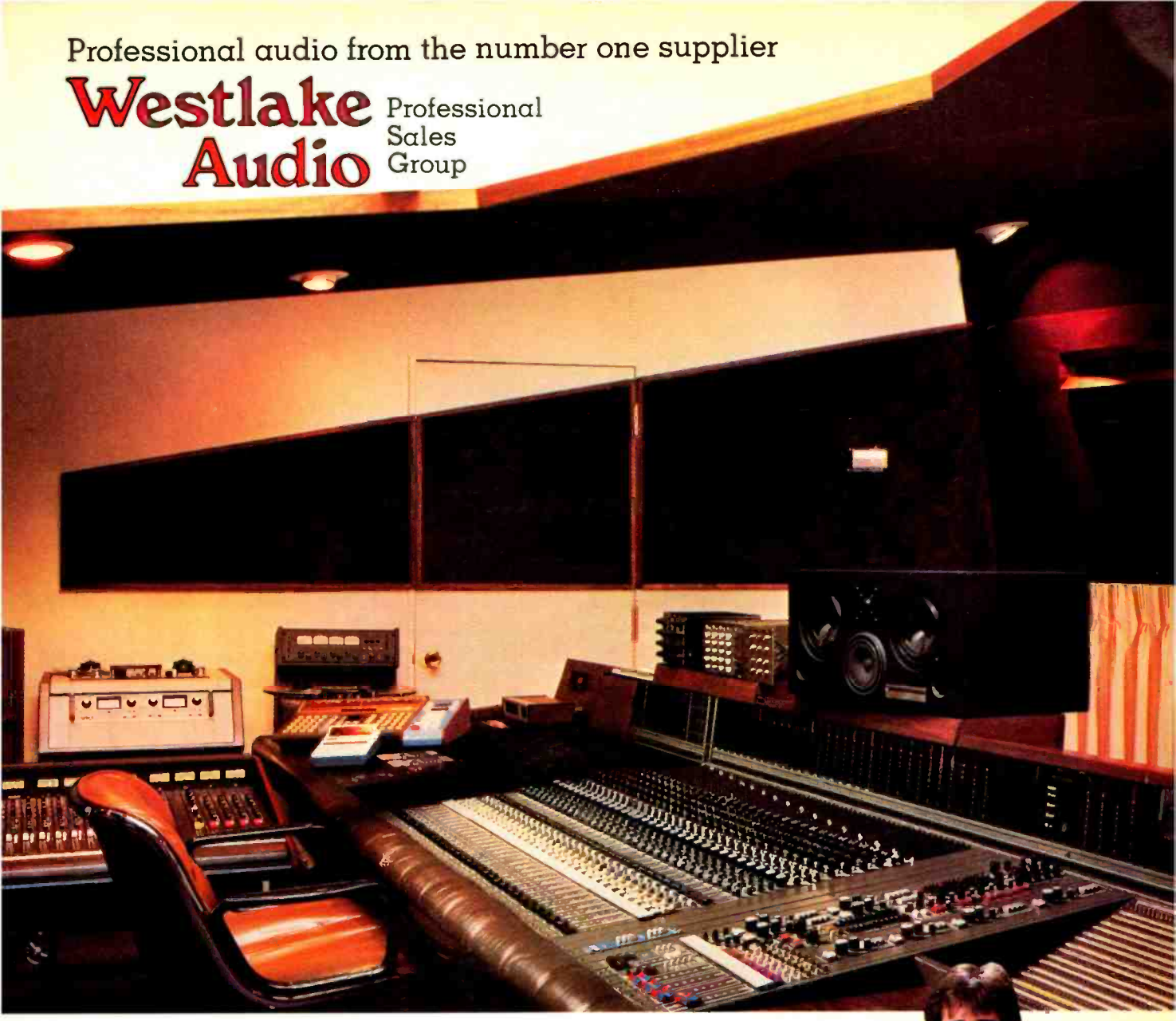

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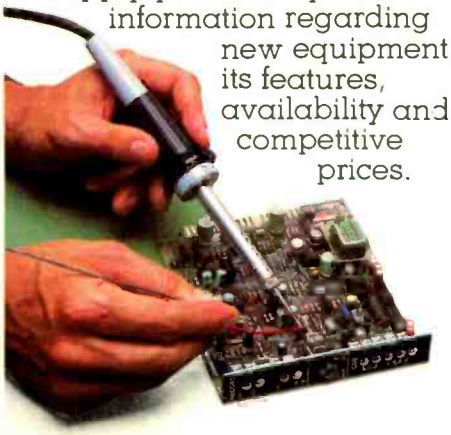

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Letters

Bipolar power supply

From: Harvey A. Rubens, VCA Associates, Canoga Park, CA

The construction article by John Gaines in the August issue of **RE/P**, describing a bipolar power supply, should prove useful to countless studio technicians, experimenters, etc.

Because, by the very nature of a free-standing supply such as this, one cannot predict the load, cable length or the operating circumstances that may be encountered, I would offer the following comments to enhance the stability, safety and performance of the project.

For further understanding, readers are referred to the various manufacturers' manuals for detailed discussions on these and other techniques.

1. The "adjust" feature is not recommended, because it degrades regulation. Instead, use a regulator specified at the desired voltage and the appropriate transformer. Shorting (jumpering) R2 and R4 sets the circuit this way. Delete R1 and R3, because they merely draw excess current.

The experienced technician who needs voltages other than those available with standard parts can rework the printed circuit art for LM317 and LM337 *adjustable* 3-terminal regulators. Consult National or Motorola manuals (among others) for details and, for best results, use *all* "optional" decoupling and protection parts shown in the literature.

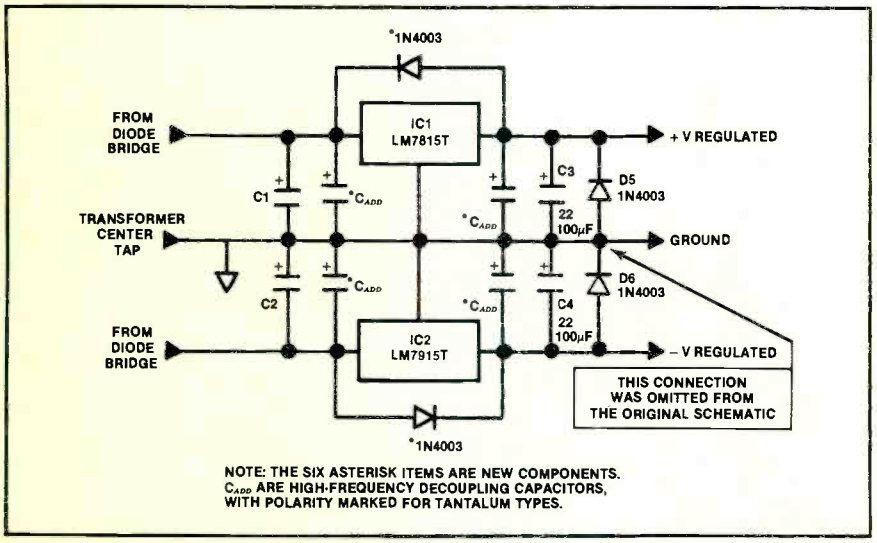
2. The use of high-frequency decoupling capacitors placed near each regulator input and output will prevent oscillation in the regulator. Ceramic caps rated at 0.1μF should suffice in this layout; 1μF/35v tantalum caps will work for sure.

3. C3 and C4 should probably be larger. Values from 22μF to 100μF are available to fit in the space provided.

4. According to the data book, diodes D5 and D6 should be power types. The same 1N4003 diodes specified for D1 thru D4 will be fine. (See Figure 1 on page 70 of the August issue, and the accompanying figure shown here, for correction: diodes D5 and D6 should be connected to ground at their cross point, like C3 and C4.)

5. Adding reversal polarity diodes (see new figure) will protect the power supply from fault conditions where the unregulated source voltage drops below the stored charge in the output capacitors. The 1N4003 diodes provide a path around the regulator under this fault condition. (Since I've suggested deleting R1 and R3 and shorting R2 and R4, it should be easy to use these holes to neatly insert the new diodes and hardwire the leads to perform the suggested surgery; again, type 1N4003 is a good choice.)

PS: Be careful if you use XLR-type connectors for a power supply. Someone may accidentally plug your favorite mic into the power supply with predictably disastrous results!



Suggested additions for the bipolar power supply construction project, which appeared in the August issue. The section shown here refers only to the comments in the above letter from Harvey Rubens; consult Figure 1 on page 70 of the August issue for the full schematic.

Theatrical film sound

From: Daniel Seguin, St-Therese, Quebec, Canada

I am writing in response to Larry Blake's "Film Sound Today" column in the June 1986 issue, which was, by the way, interesting.

I would like to ask a simple question: Why do cinema sound systems differ so much? Isn't there some kind of standard in relation to technical specifications? I wonder if film sound designers realize how badly most of the soundtrack is presented to an audience.

I served as a projectionist for about six years during my college and university career at the University of Waterloo, where I studied electrical engineering. Later, in Montreal, I took a 2-year training course to become an assistant recording engineer. As a free-lance recording engineer, maintenance technician and acoustic consultant, I can very well evaluate cinema sound systems.

A film soundtrack will always sound different from one cinema to another, and 90% of the time it will sound bad. Filmmakers don't seem to realize that most cinemas are 10 to 20 years behind in sound systems technologies—at least in the Montreal and Toronto areas.

Recently, I saw *Top Gun* at the Imperial, Montreal, in 70mm 6-track. At the Dorval cinema I saw the same movie in 70mm 6-track, but not with the same sound as that at the Imperial.

Cinemas differ widely in sound system equipment, frequency response, room acoustics and many other factors. Cinema systems today should be standardized; capable of reproducing high dynamic range music and effects; and should be designed with today's technology.

*Reply from Larry Blake, **RE/P**'s film sound consulting editor:*

The answer to your question—why the same film sounds different in different theaters—is, of course, both very simple and very complex.

All modern theaters equipped to play 35mm and 70mm Dolby Stereo prints have to meet certain standards defined in ISO bulletin 2969. The frequency response is codified (something that the record world has never done), as is the playback level, although I believe the latter is really because of the efforts of Dolby Laboratories.

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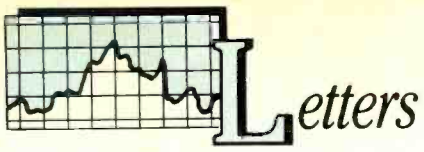

So, regardless of the different speaker systems and acoustics, the mix should sound substantially the same, right?

But you and I both know that a few other factors come into play here. One of the most important is the upkeep of a sound system: a theater that sounds great today might sound just OK a month into a run, after a speaker or two is blown. Also, the print and mag heads are worn down at this point. The latter problem concerns the "A" chain—everything leading up to the main fader—and cannot be compensated for in the "B" chain (the amps, equalizers and speakers), regardless of the quality of the theater's sound system.

As to your comments about filmmakers' knowledge of this problem: *they do know!* But what can sound editors or re-recording mixers do other than to do their best work? I believe the answer to this problem lies in a two-pronged attack: first, paying customers like you and me asking for our money back; and second, directors and distributors playing hardball.

Regarding the former approach, over the years I have written my share of complaint letters, and will continue to do so. But in the future I will try to get them where they breathe and, if the presentation is not up to these phantom standards, I will get my money back. I have found it a useless pursuit to talk to theater managers; they only say that "they came in and checked the system." They must be deaf. I could reel off a litany of instances where on opening weekend, a top film received a bad presentation in a top theater in Los Angeles.

Unless we complain, things will not get any better.

Fostex Hands On

From: Marc Cohen, vp marketing and sales, Fostex Corporation of America, Norwalk, CA.

Obviously we appreciate the efforts of Bod Hodas and Denis Hannigan in their thorough review of the B-16, 20, 4050 and 4030/4035 [August issue of **RE/P**, page 50]. As mentioned in the article, the software in the 4030 and 4050 is being improved constantly; I thought it would be appropriate to bring **RE/P** readers up to date on our latest developments (the software tested is 2 generations old).

First, the 4030 synchronizer has been substantially improved by the latest software, Version 2.0. Here are some of the features:

1) Even faster lock-up (we'll challenge anyone to a race).

2) Wider range of damping control for improved performance with other manufacturer's recorders.

3) Automatic learning of slave's servo characteristics.

4) Automatic learning of slave's tach rate.

5) Master waiting in chase mode. If the master is in auto play, it will wait until the slaves get to the cue point before going into play.

6) Speed readout in the 4035 display for easier and more accurate set-up.

7) VITC compatibility.

8) Code-only master status set by software command instead of jumper plug.

9) Adjustment of slave park point.

10) Play-to-park function.

Older 4030s can be easily upgraded by contacting the Fostex service department. The review mentioned that the locked light comes on at frame accuracy; Version 2.0 indicates locked in subframe mode. Also, feature number 6 (above) eliminates the servo adjustment LED problem. The B-16 should not have taken so long to search for cues (with either the 4030 or the 4050). This problem could have been caused by either an incorrect damping setting (usually the case) or incorrect tension-arm adjustment on the B-16D.

The problems associated with the interface for the JVC BR-8600 have been cleared up with a later-model interface (8756). By the way, the significantly less expensive JVC BR-6400 and 5300 VHS video recorders can be used with this interface, thus making the entire video/audio system very affordable. Also, any video recorder (i.e., home-style Beta and VHS machines) can be used as a code-only master with our 4030 synchronizer; no modifications of the video recorder are necessary.

With regards to the 4050's MIDI Song Pointer disrupting the Midimac software. we are in discussion with Opcode Systems about the problem and expect a solution real soon now. There are no problems when using the 4050 with the latest versions of Roger Powell's excellent Texture program or Performer. The 4050 has had some software updates, the latest version being 1.6 or 2.6 (depends on which ROMs are being used). The latest software supports all modes of punch-in and offers the most consistent MIDI timing we've seen.

We now have an improved manual for the 4030/4035 and are continuing to offer the videotape instruction manuals. Fostex has synchronizer interfaces available for virtually all current models of sophisticated video and audio recorders.

The only error we noticed is that the Model 20 recorder ($1,200) with center-track SMPTE was referred to as the E-2 (also with SMPTE, but uses larger reels and costs $3,600).

Once again, thanks for your efforts.

The Tact Factor

From: Gary Woods, Woods Music, Tujunga, CA.

I am writing concerning an article published in the April issue, titled "The Tact Factor," and specifically the section titled Orchestral Maneuvers. The opinions expressed in this section by the article's author, David Brody, do more to foster ill will between musician and engineer than they do to contribute to a smoothly flowing recording session.

Such comments as "legit musicians are also notorious for not being able to grasp the concept of multitrack recording," is absolutely astounding to me. You're speaking about a group of players, some of whom have played sessions for the past 20 years.

But my favorite comment is: "The trick is to get them up to concert pitch." I have been in situations where the engineer is feeding the orchestra nothing but drum kit, and wondering why they can't find a pitch. Or, where the orchestra is baffled off from the other players in the room so completely that they can't hear the person next to them, let alone the section 10 feet away.

And the height of arrogance is expressed in the statement: "They don't necessarily hear any better than you do." If the engineer would get himself out of that glass box and occasionally come out to the studio floor, he would hear what is going on out there.

I have seen engineers place a microphone less than 18 inches from a bass clarinet and wonder why he was getting "all that key noise." I also have seen engineers bring the microphones down so close to violins that all you hear on the track is a metallic raspiness. And then they wonder, "Why can't those guys play any mellower?"

I'm sure David Brody realizes that

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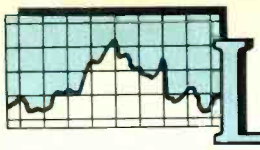

Letters

some brass and reed players carry their own microphones around with them. And he alluded to the fact that some synthesizer players send already processed stereo signals to the mixing console. One of the big reasons this occurs is because musicians are tired of having the sound they have worked on for so long messed up by some guy who has no concept of how a violin or a trumpet, let alone a synthesizer, should really sound.

The most offensive section of all is, "You may even have to erect a few unused microphones so nobody complains of being too far away to be heard." That kind of deception is repugnant. If an engineer doesn't understand the orchestra and how it sounds, then he should not be on a date where there are live musicians; he should be kept working on projects where sequencers and drum machines are the only things being fed into his console.

If I ever have the misfortune to run into this kind of engineer on one of my dates I will tell him to keep his opinions to himself. If it recurred, I would cancel the date and sue the studio and the engineer for the cost.

Reply from David Brody:

It is gratifying to know that there are well-schooled musicians who take the time to read publications like **RE/P,** which approaches the studio from what is essentially the technical side. A careful re-reading of the article will show it to be in substantial agreement with many of the points raised by Mr. Woods. As for where we differ, the article was intended to expose some of the problems to the cold light of rational analysis and to show that *nobody* is immune from critique. The sheer fact that we are debating these matters, in preference to playing the same old cover-up game, indicates that the process is working; a far cry from doing "more to foster ill will."

I wholeheartedly agree with Mr. Woods when he says, "If an engineer doesn't understand the orchestra and how it sounds then he shouldn't be on a date where there are live [orchestral] musicians." But the next statement, "He should be kept working on projects where sequencers and drum machines are the only thing being fed into his console," belies a misconception to which many conservatory-oriented musicians fall victim.

An engineer who cannot hear ensemble in a choir of acoustic instruments and discern how to extract the intended sonority therefrom will fare no better with synthesizers. . .probably worse. With electronics, the instrumental tone color and how it varies with time is, to great extent, controlled by the recording process, not the player. There is no finer example of real-time user-mastery over a great number of technical parameters than a virtuoso string, horn or reed player.

A few years ago, binaural recording—a technique which would have enabled us to record, less obtrusively and more accurately, the sound of an orchestra in a hall—made one of its periodic re-emergences; it was, as through its history, substantially rejected by the marketplace. Although it is being refined all the time (for example: Holophonics from Zuccarelli Communications) and its ideal play hardware may finally have arrived (the CD Walkman), the bottom line is still that orchestral textures are largely regarded by producers as "sweetening," and are so recorded.

Retrofitting "string and horn parts" to a pre-existing track hardly resembles masterworks that are through-composed. And getting them recorded in no way resembles the manner in which *all* of us would like to see an orchestra treated.

That may mean that I have to stick the microphone much closer than I'd like and that you have to put up with more isolation than you care for. *But I certainly concur that you shouldn't have to cope with a bad cue mix,* and I would never defend *any* engineer for giving you one. We can debate the emotional details, but there is *no* excuse for technical incompetence. The sad fact is that you may succeed in hiding a mediocre musician deep within a section, but a poor engineer can do a great deal of audible damage.

In my article, the term concert pitch is used, not only in reference to intonation but, moreover, to define that magic when an orchestra is performing before an audience in a concert hall. Although some of the newer studios (designed using real-time analysis of acoustic phenomena) are good, they are not optimized for any particular size ensemble or type of music. They can't be; they are designed to help expose problems, not mask them. That fact presents the musician trained for the concert hall with some difficult challenges, most of which seem to be psychological, not technical.

At the root of much of this trouble may be the fact that musicians who have spent countless hours in hard work and diligent practice may feel that their talent (and their chops) are being given a back seat to the hitmaker machinery that allows the magic of take 87 to eclipse the ability to get it right the first time the baton comes down. Or that true performance musicianship has been replaced by technology—and it's always easiest to blame the messenger: the engineer and the studio. And, of course, there *are* specialist recordists whose only bag is not legit.

To expect a concert hall and a bravo/encore from the recording studio is as absurd as a producer insisting that tuxedos be worn by the instrumentalists to help capture the esprit-de-corps of a live orchestra.

When something has gone wrong and we all stand to catch the blame for it, we reserve the right to take ameliorative action, even though we may not wish to waste everyone's time with on-the-spot negotiations.

In fact, the article makes a case for supplying the conductor with a track blend and allowing the instrumentalists to function more as an orchestra under their conductor's direction. This simplifies things for all of us but isn't always possible given the producer's need for isolation in the mix and absolute tight adherence to the rhythm track in fast tempi or critical picture hits in video or film.

The Tact Factor articles are a first attempt to stimulate discussion in these entrenched, protected, overlooked areas, and to do so with humor and candor, often the best way to defuse emotional time bombs. It is, by far, not a perfect attempt.

In this less than perfect world, none of us can play in the best tessitura all the time. As one who has spent many hours avoiding a suntan in practice rooms as well as in control rooms, I understand full well the constraints you, the musician, may be placed under. Can you begin to see mine?

Be assured that no competent, tactful engineer would raise these issues on the session. *That is the point of this series of articles.* Here in print where nobody is under pressure of the clock, or putting his next paycheck on the line is the proper forum to address such concerns.

R-E/P

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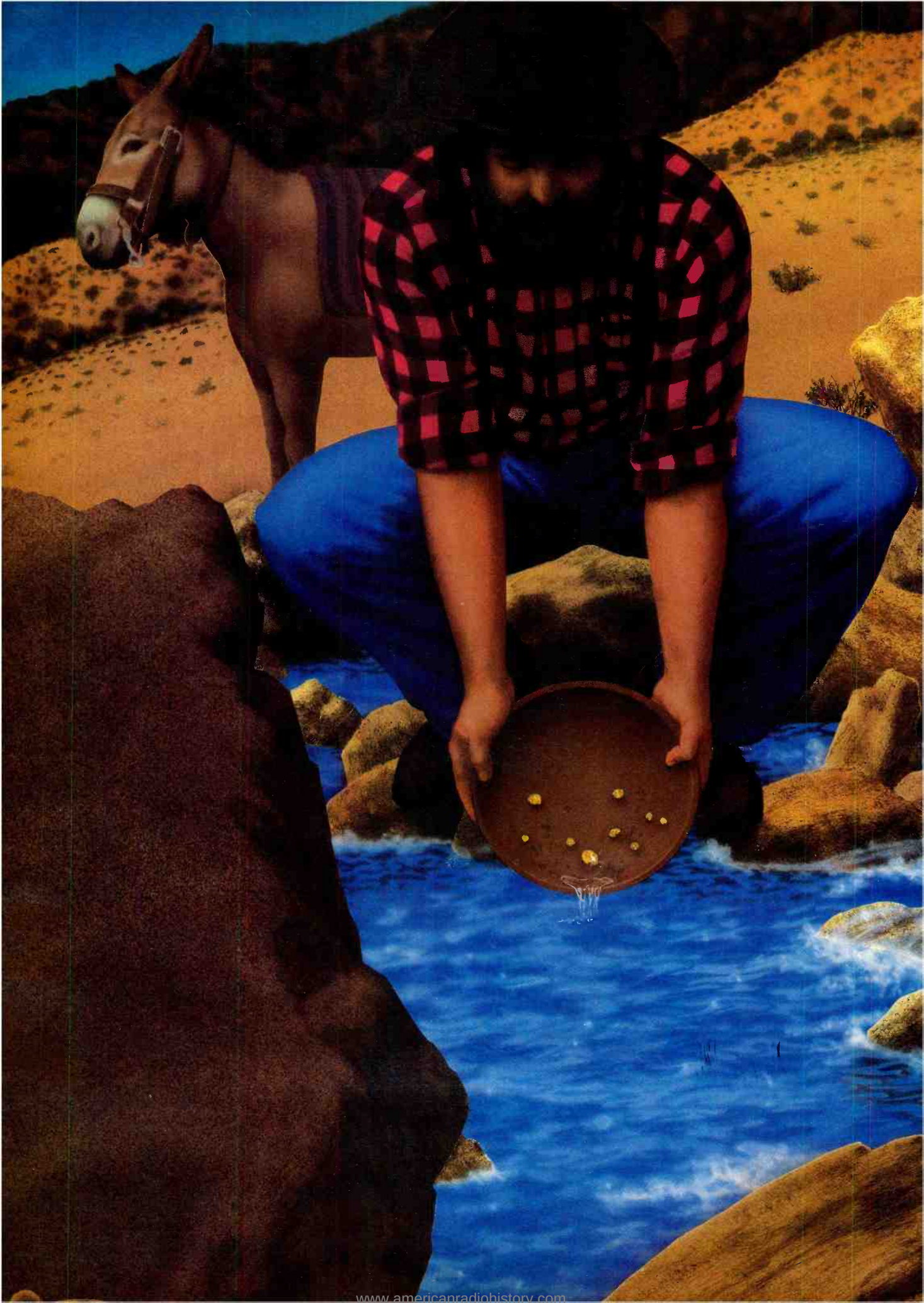

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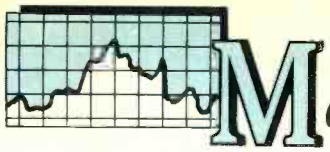

Managing MIDI

By Paul D. Lehrman

Two decades ago, when electronic music sequencers were first introduced, life was simple. Every sequencer ran on its own internal clock. If you wanted to use it for a composition, you adjusted its speed with a knob, maybe checked it against a metronome and rolled tape. If you wanted to use it in conjunction with "live" tracks, you recorded the sequencer first and then used the synth track as a tempo reference for the rest of the piece.

Recently, however, things have gotten more complicated Sequencers, whether they are internal to a synthesizer, stand-alone units or part of a computer system, are being called upon to perform more sophisticated tasks, and to do so at various stages of the production process. Therefore, some way of synchronizing sequencers with other studio gear had to be developed.

Drum sync was the first type of sequencer sync that most studios encountered when they started working with drum machines. It and FSK (for frequency-shift keying, also known as tape sync) remain the most common technique of locking up sequencers and tape machines. With drum and FSK sync-generating and -reading devices, engineers no longer had to record a drum machine's audio output on several tape tracks.

Instead, they could simply lay down a sync track and let it drive the drum machine "live" as other tracks were recorded. This freed up tracks and also improved the quality of the drum sound, because one generation of tape was avoided.

Both drum and FSK sync are tied to the musical tempo of the device creating them—a certain number of pulses or shifts is generated for each quarter-note beat. Both methods share a set of problems: they can't be read off-tape in fast wind; recorded levels have to be adjusted very carefully, usually by trial-and-error; guard tracks are essential; noise reduction will render the signal unreadable; and dropouts can be catastrophic.

They also have a fixed starting point, so if you decide you want a sequencer to start after the sync pulse does, you have to program in the correct number of blank beats. And if you want it to start *before* the sync, you're out of luck. In addition, drum and FSK sync are "dumb": They contain no location information, so every time a sequencer is initiated from tape, it it always has to be started from the beginning.

To make things worse, there are no standard formats. Each manufacturer has its own ideas about the optimum speed of the pulse or the best frequencies for the carrier and modulator signals.

Some way of synchronizing sequencers with other studio gear had to be developed.

This can create a lot of confusion if equipment with different sync rates are going to be used together; for example, if a 48-ppq sync track is being used to drive a sequencer that reads 96 ppq, then the notes recorded in the sequencer will come out at half speed. With FSK, one company's code may look like so much gibberish to another company's reader, and it won't respond at all.

A third popular method of synchronization is audible click tracks. Click tracks work fine for humans (and are less finicky about record levels), but not as well for machines, especially those that use them as a reference for generating another type of sync.

MIDI Sync is something else entirely. It is not a stand-alone signal; instead it is a digital "word" sent out 24 times each quarter-note as part of the MIDI data stream. Besides being an immutable standard, MIDI Sync has a distinct advantage over the other forms: Often accompanying it in the MIDI stream is location information, known as song position pointer. These don't appear continuously in the data stream, but instead are sent only on occasion, usually along with a command that tells a sequencer to start. MIDI pointers allow a sequencer under remote control to be told to start anywhere within a song, and the sequencer will then follow the 24-ppq sync words to stay locked up correctly.

However, because it is part of the MIDI signal, which has a bandwidth of 31.25kHz, MIDI Sync cannot be recorded on tape. (Nor can it drive any device that does not read MIDI, despite the fact that it runs at 24 ppq and appears at a DIN connector, like some forms of drum sync.) But converting MIDI Sync to other forms of sync signals that can be recorded on tape (and vice versa) is a fairly simple matter. The advantages of such conversion to the small studio are obvious: With a recorded sync track, a converter and a multitrack sequencer, one synthesizer can do the work of several, each track recorded in a separate pass with perfect lock-up.

Taking full advantage of MIDI Sync and pointers, however, requires a recordable sync track that contains location information, like SMPTE time code. The combination of the two is one of the most exciting aspects of the development of MIDI. Sequencers can function as virtual tape recorders: Synthesizer tracks can be left in the computer until the final mix, thereby improving sound quality; tracks can be edited on the computer in ways completely impossible with tape; and a virtually unlimited number of tracks is available.

Unlike other forms of recordable sync, SMPTE time code contains no tempo information. As a result, tempos can be determined, and changed, long after the sync track is recorded (provided, they don't conflict with other "live" tracks already on tape). Likewise, the starting point for a sequence with reference to the tape can be easily altered by reprogramming offsets in the device reading the time code.

Several SMPTE-to-MIDI converters are now available, but so far they all suffer from one drawback: Because they generate MIDI Sync internally they bear the responsibility for determining tempo. Because these converters are hardware devices, they cannot possibly be as flexible as a well-designed software-based sequencer; consequently, programming complex meter and tempo changes is more difficult than it should be.

This situation may soon change. Several companies are working on sequencers that can read time code directly, and will use it as a straight timing reference (with location information), while tempo changes—as well as offsets and triggers—will be programmable within the sequencers themselves.

Another development that will help is MIDI Time Code. This addition to the MIDI spec goes several steps further than pointers: It provides *continuous* location information, as well as "setups" that can be used to pre-program any number of MIDI devices. It was formalized at the meeting of the MIDI Manufacturers' Association at last June's NAMM Expo, but at this writing it is still awaiting the approval of the Japanese MIDI Standards Committee. I'll talk more about it in a future column. R-E/P

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

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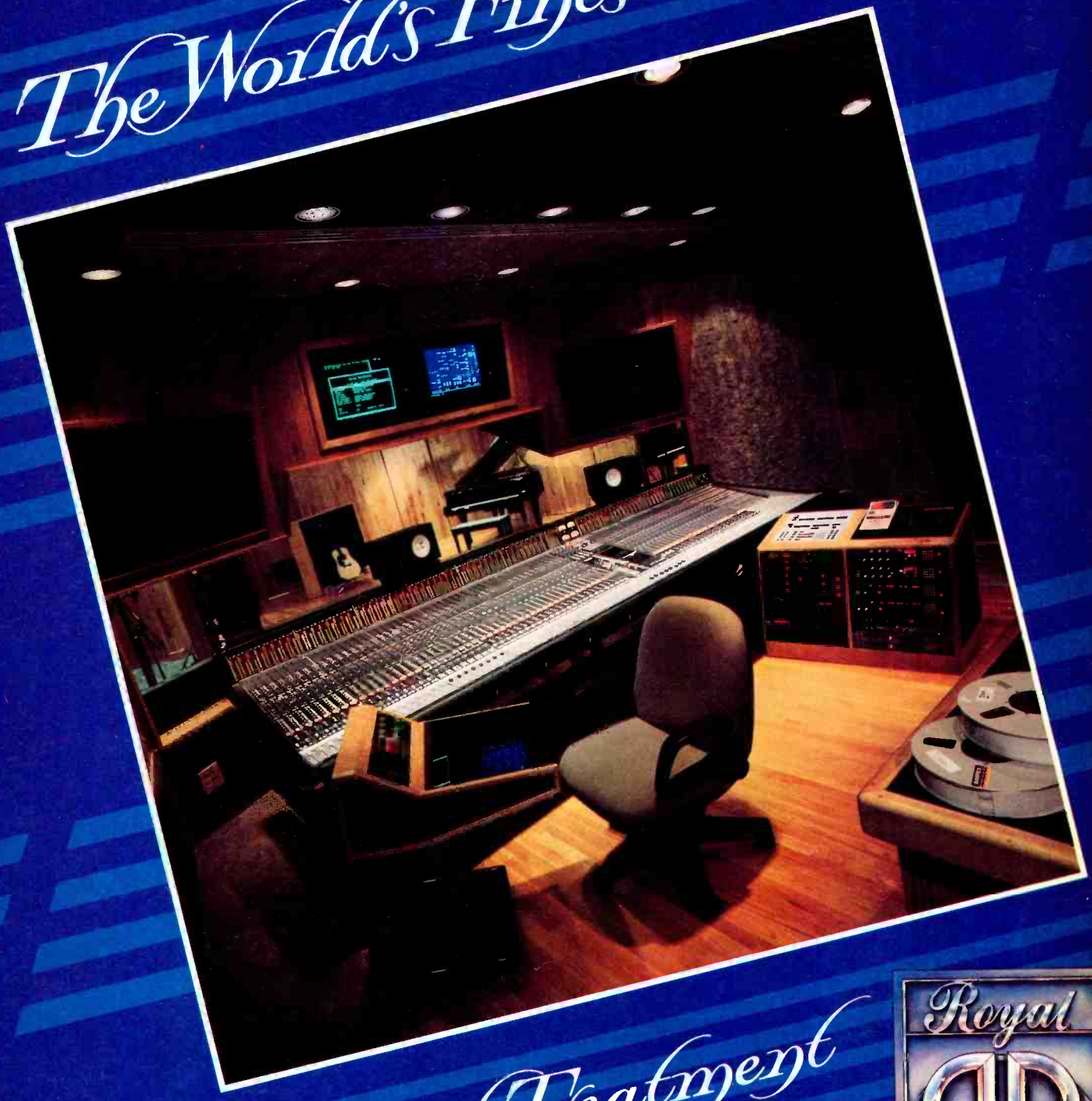

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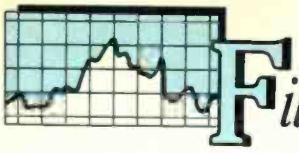

Film sound today

By Larry Blake

Nagra portable recorders have served the film industry for more than 30 years; during more than half that time the company has had the production recording market practically to itself. This situation will almost certainly change, however, when a 1-piece, portable, professional digital recorder comes into the market.

The R-DAT format (rotary-head, digital audio tape recorder) appears to have an early lead. A large amount of momentum has preceded R-DAT's unveiling, which current rumors set for some time next year. In 25 words or less, R-DAT uses a 2.8"x2.1" cassette housing, and records up to two hours at the 48kHz sampling frequency using 16-bit quantization.

This column presents a wish-list of features for the phantom "digital Nagra;" I'll continue in the December issue with a look at the features of R-DAT that make its use in post-production not only possible but inevitable. R-DATs support by more than 81 companies leads one to believe it will be around for a long time.

In rough order of importance, here are some features that I would want to see in a portable digital recorder for film sound production.

• Off-tape monitoring. Call it "confidence playback," or whatever you want, but most film recordists would not use a machine that didn't offer the ability to listen to not only what is going on tape, but also to monitor playback during a take.

• Meters should be legible in bright sunlight. If manufacturers can't make a fancy LED, LCD or plasma meter that meets this important spec, no problem. Give me a lighted, old-fashioned moving-coil meter with peak-reading needles (on the same meter movement).

The metering should also be able to indicate battery voltage and the presence and status of sync, in much the same manner as the Nagra modulometer. (There's one item, however, on the Nagra IV-S metering that we won't really need: LP groove depth according to NAB weighting. I guess there's no analogy to pit depth on a CD.)

While I'm on the subject of monitoring record status, there should be a bright run light that perhaps starts flashing progressively faster when there are under two minutes of record time left, and stays lit when the end of the tape is reached.

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

This light and the time code reader (see below) should be mounted on the front panel near the meters.

• The unit should offer long battery life, either with an easily recharged and quickly replaced pack, or with easily purchased and long-lasting C or D cells. The tradeoff between light weight and long battery life should be made at about 15 hours—long enough to get you through a weekend shoot. (As much as I like the 30-hour life of a set of 12 D cells in a Nagra, your shoulders pay a price for the convenience.)

R-DAT uses a 2.8"x2.1" cassette housing, and records up to two hours of 16-bit/48kHz audio.

• An internal time code generator and reader should be provided. Kudelski, with its Nagra IV-S TC, has set another benchmark for R-DAT to aim for and improve upon. One such improvement would be the ability to accept an external sync source, such as composite video sync via a BNC connector.

It would also be helpful to have a multipin jack that would provide the 60/59.94Hz sinewave stripped from the time code, in addition to a time code output and inputs for external time code and external sync from a 60/59.94Hz crystal. Since the R-DAT format offers two "optional" tracks, a second multipin jack could be used to access their inputs and outputs. This jack could also provide a clapper input for an external "bloop" light used to mark sync in documentary filmmaking.

• The ability to record and play back at the 44.1kHz, 44.056kHz and 48kHz sampling frequencies. As presented in preliminary specifications of the consumer R-DAT format, 44.1kHz would be a playback-only sampling frequency, with recording available at either 48kHz or 32kHz.

The purpose of this design, both to boost prerecorded R-DAT cassette sales and to prevent D-to-D copying of Compact Discs, makes no sense at all. (Wake up guys: If I want to copy CDs to play them on my R-DAT car player, I'll do it through the converters at 48kHz.)

Film and video mixers who pay a few thousand dollars for a *professional* R-DAT machine will need to record at 44.1kHz/60Hz to allow convenient pulldown to 44.056kHz/59.94Hz in conjunction with 24fps film during telecine. Similarly, there are times when the tape must be played back at 44.1kHz during filming to ensure correct sync, after telecine and video editing, with a track prerecorded at 44.056kHz. Such cross-resolving will be necessary in a professional machine, and it will be a sad day indeed if the consumer tail wags the professional dog.

• A multipin accessory connector would conveniently provide production mixers with a slate/talkback mic (with 40Hz subtone), in addition to a feed to the boom operator. The slate mic would also be useful when the mics are planted far away, or when recording from the line inputs.

A single multipin for stereo line inputs and outputs would be useful, although there is something to be said for standard, 3-pin XLR jacks for all audio connections. (I had hoped to avoid mentioning their obvious necessity on the microphone inputs.)

• To control levels, gangable stereo pots should be featured. Slide faders would simply not be as practical in over-the-shoulder field recording, where a stationary hand and twisted thumb and forefinger do the work best. The knobs should be separate, and not dual-concentric.

• Flexible monitoring is important. The Nagra IV-S has a switch that provides stereo or mono monitoring, or assigns either left or right tracks to both ears. In addition, there should be a screwdriver-adjustable headphone level control, and I'd like everything (along with the jack) on the left front panel.

• A rugged, one-hand function control knob is essential, because solenoid switches are just too easy to hit in the field. Admittedly, such a manual knob would preclude remote operation. I'm assuming, however, that such requirements would be handled by a rack-mounted R-DAT deck in the studio.

If I've gone overboard with some of these features, I perhaps should explain that a friend of mine saw the transport for an R-DAT machine pulled out of a *No. 10 envelope;* there *will* be room to spare.

I hope that this column will serve as the catalyst for a dialogue to be established between the professionals who will use the recorders, and the companies that will design and manufacture them. I would really appreciate feedback on this issue from both sides. R-E/P

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Technology

By Stephen St. Croix

Not too long ago a friend and I were engaged in a particularly macho, non-computer-nerd type of activity: cruising the East Coast on big motorcycles.

We had left our terminals and keyboards as far behind us as possible, and were totally involved in the art and science of aiming 700 pounds of metal in the general direction suggested by the dotted white line. Then, out there in the wilderness, his computer crashed. So did he.

After 24 years of walking up to the parts counter of the local bike shop with the same line, "It broke...got one in stock?" It felt a bit strange to walk up to the same guy and say, "My friend's bike fell off the road...you got newer software?" He didn't, by the way; we had to buy a new computer. Good or bad, that was one of those events that stands out as a marker of the further advancement of the byte in my life.

A couple of months ago, I received a UPS package whose contents merely represented a natural evolution of technology, but to me this was still another one of those benchmarks of the "Advancement of the Byte."

Actually, it was just another synthesizer—an additive synthesis machine. With more than 20 synthesizers already in my studio, I am not always thrilled when a new one appears for my evaluation: But this one was a little different. I immediately unpacked it and carried it in my shirt pocket to the synth room.

I viewed this product as the next step, while a friend saw it as "another one of those blue squares." (For the non-Mac literate reader read: 3.5-inch floppy disk.) Sometimes technology is really fun: This blue square is truly the most comprehensive additive synthesis machine that I have ever owned, and it takes up exactly the right amount of space in the synthesizer room: Zero.

I like it when a machine that I already own—in this case a Mac Plus—is suddenly transformed into something new, in this case a 16-bit additive synthesizer. This is my first software-only synthesizer. While I just happen to have the proper hardware on hand to use this program, it's all hardware that I got to use independently; the fact that it can now be used to translate the voices of this program to audio waveforms is pure gravy.

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

If you use a Mac in your studio, and already own a sampler, you owe it to yourself to take a close look at one of these new software synths. The Macintosh is the actual synthesizer, and almost any sampler becomes the DAC and player. Nice. The mouse that roars can now be yours.

With the pricing of technology agreement America and Japan reached months ago (and which will make memory costs higher to you), the concept of redundancy in hardware is even less attractive.

A powerful new toy like this with one moving part and a total hardware cost of less than $2 is a welcome example of an alternate approach to synthesis, and of using computers in the studio. It's an example I hope will soon be explored further soon.

With the general acceptance of MIDI, I feel a bit manipulated when I am forced to get another new synthesizer with a keyboard. What am I going to do with *another* keyboard? All I really wanted was the synthesizer itself. I play one keyboard, and 20 more collect dust as they sing along.

The newer rack synths are a big improvement. But it might be interesting if it were taken a little further. How about rack-mount samplers that can talk to each other? Lack of data-format standardization has been a problem for end users since the beginning of this technology. If our industry would grow up and standardize MIDI Song File formats, and if each manufacturer would make available information for translating their sampler's sound files, everybody's hardware would be more powerful. Properly implemented RS-422 or SCSI (small computer system interface) ports would be another major step in the direction of an integrated studio.

If communication protocol were stabilized and published, various companies could write front ends (software synthesizers) for these different units. Each would have its own graphic screens and appropriate controls: knobs for the emulation of analog synthesizers, Fourier maps for additive units and maybe even an intuitive display for FM. Well, maybe not. Virtual synthesizers; yeah, that's the ticket!

Think of it: the Synth-Of-The-Month club. Every month a new synthesizer comes in the mail...or another blue square, depending on your point of view.

R-E/P

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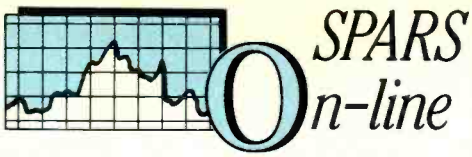


By Gary Helmers

If you have ever been involved with planning or organizing a major industry convention, you know about the tremendous amount of work it entails. You can also appreciate the difficulty in structuring an event that has a clear, unambiguous point of view.

The "big picture" of overall industry trends can easily get lost in the shuffle to serve specific industry needs. Convention attendees are presented with a growing diversity of exhibits, workshops and seminars. As our industry matures, it becomes increasingly difficult to distinguish what's important to see from what's not, or to deduce the direction of the industry, from the information presented at a major convention.

It used to be so easy—just follow the crowds and you'd find yourself at the center of "what's happening now." Lately though, things have become a bit confused.

Just what and where is the *new* emerging technology? There used to be lots of fanfare and a general consensus of opinion at each show regarding the most important announcement or development. Now, who knows? It's likely that you'll wander around the exhibit floor, attend a few papers, go to a workshop or two, and possibly come away feeling bewildered.

What's happening in our industry? What are the dominant technological trends? What will be the structure of the audio production industry in 10 years? In what direction do I develop my business to ensure that it survives?

Borrowing from chemistry (and aesthetic theory), you might say that audio technology is in stasis—a state of dynamic equilibrium. Although things in stasis are constantly changing and mutating, no single change is so dramatic as to attract the attention of a majority of observers.

The last big quake we experienced was the introduction of digital multitrack recording. Since then we have seen many little tremors, several minor revolutions and numerous technological breakthroughs. Yet there has been no dramatic, all-encompassing change.

Moreover, as we look to the future, there seems to be no dramatic revelations on the horizon. The audio industry is by no means stagnant; it is in stasis. It's a useful term, but what are the real-world implications?

We at SPARS believe that one of the most plausible views of the future is the so-called "Mothership Scenario," which recognizes the proliferation of small, specialized facilities that prepare some part of a total production. Let's call the latter "workstations"; they often feed their creations to a larger facility (the mothership or mother assembler) that assembles all the parts. The mothership must be able to speak, interpret and synchronize the language of all the workstation satellites.

You might say that audio technology is in stasis—a state of dynamic equilibrium.

The organizational structure of motherships and workstations will be extremely flexible. Most workstations will have no permanent allegiance to a mothership. Rather, a producer will pull together a team, consisting of a mothership and several workstations, to accomplish a given project. Workstations and motherships may be participating members of several teams at the same time. It won't even be necessary for workstations to be in the same building, or even on the same continent. Today's technology truly allows a workstation to "phone in a part."

There is evidence that this mothership/workstation scenario is the direction of our industry. We are seeing larger and larger control rooms to provide for the simultaneous presence of several workstations. Some of the synthesizer workstations are becoming too cumbersome, or too delicate and time-consuming to move, thus requiring special rooms to house them.

More and more small rooms are being built around a particular synthesizer workstation. Many artists are working at locations quite remote from the mother assembler.

Will all the studios now in existence become motherships? Unlikely. Some studios are making a valiant effort to become a galaxy entirely unto themselves, with a stellar system of workstations interfaced on the premises. Other studios are instead concentrating on a particular workstation technology, with the intention of becoming the best of that type. There also are facilities that will become strictly motherships, emphasizing their assembly and communication skills.

Existing computer networks are already competing to become the most favored form of communication between workstations and motherships. Satellites and telecommunication will become increasingly important as the mothership-workstation teams become more spread out geographically. Of course, there will also be the "Woods Hole School" that gathers a community of workstation artists together at a remote mothership for in-depth artistic communion.

On what terms will motherships and workstations compete? Workstations will differentiate themselves by the uniqueness of their equipment configuration; by their compatibility with other workstations and with a wide variety of motherships; and ultimately by the individual creative talent of the workstation owner/operator.

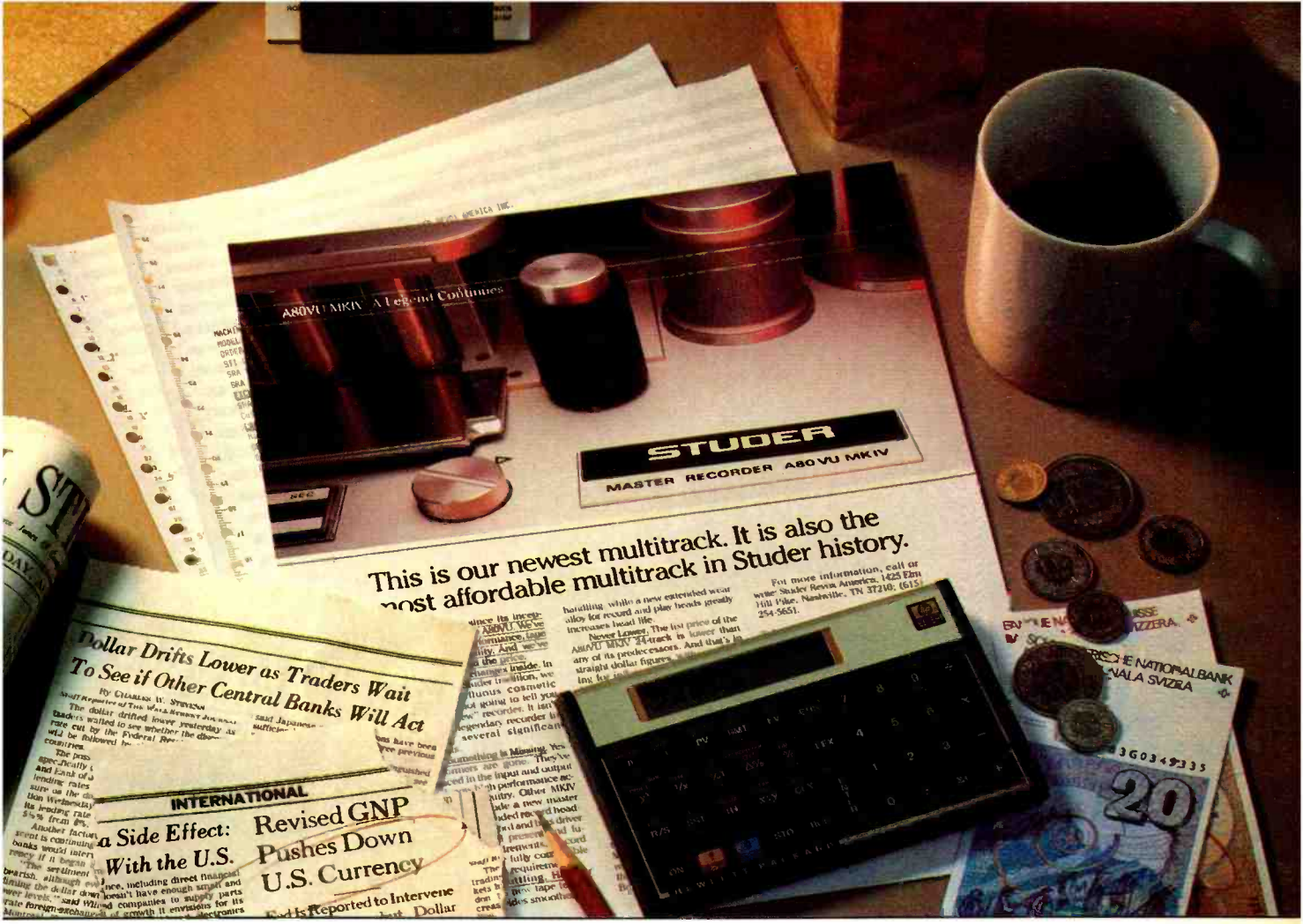
Motherships will be valued for their compatibility with any type of workstation and their ability to translate, coordinate and assemble with efficiency. Motherships will be expected to be able to prepare audio product for eventual release in practically any format, be it film, video, Compact Disc, R-DAT and so on.

We are really forecasting another type of diversification for motherships, and specialization for workstations. Five years ago, the watchword at SPARS was client diversification. Both workstations and motherships must still maintain a diversity of clients. Motherships must diversify the type of interface tools they possess, must specialize in the type of creative tool they utilize.

The mothership scenario provides a possible way to approach the next round of trade shows. Will we be looking at and learning about workstation technology or mothership technology? Will there be seminars and workshops about the interface between workstations and motherships?

SPARS will continue to address the need for a comprehensive view of the overall direction of the industry, and the trends that will shape the studio business in the future. We hope you will join us in our quest to positively affect the course of our industry. R-E/P

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



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Multifunction Studio Design and Construction

The new two-room Record Plant complex has been designed to provide multifunction environments for the changing demands of record and audio-for-video sessions.

By Adrian Zarin

After a year of rumors and speculation surrounding the Record Plant's relocation from its original home on Third Street, the new complex opened its doors to an industry that had become charged with curiosity and expectation.

The first session at the new Sycamore Street, Hollywood, facility was a Judas Priest date in late January, in Studio 2, the smaller of the two rooms. Then, in mid-February, Joe Walsh baptized the larger Studio 1.

These first bookings substantiated the contention of Chris Stone, Record Plant owner, that in shifting its emphasis to audio production for film and television, the facility was by no means turning its back on the rock-and-roll record industry, which had supported the original complex since the day it opened for business. The initial bookings were followed by projects for Paramount Television and the TV series *Fame*, proving that Stone's commitment to the visual medium was no less substantial.

A tour of the new Record Plant facility leads the visitor to one inevitable conclusion: The complex is very much a reflection of Stone's well-publicized conception of the Eighties recording studio as a multipurpose technological facility geared to the audio needs of various electronic mediums.

Facility overview

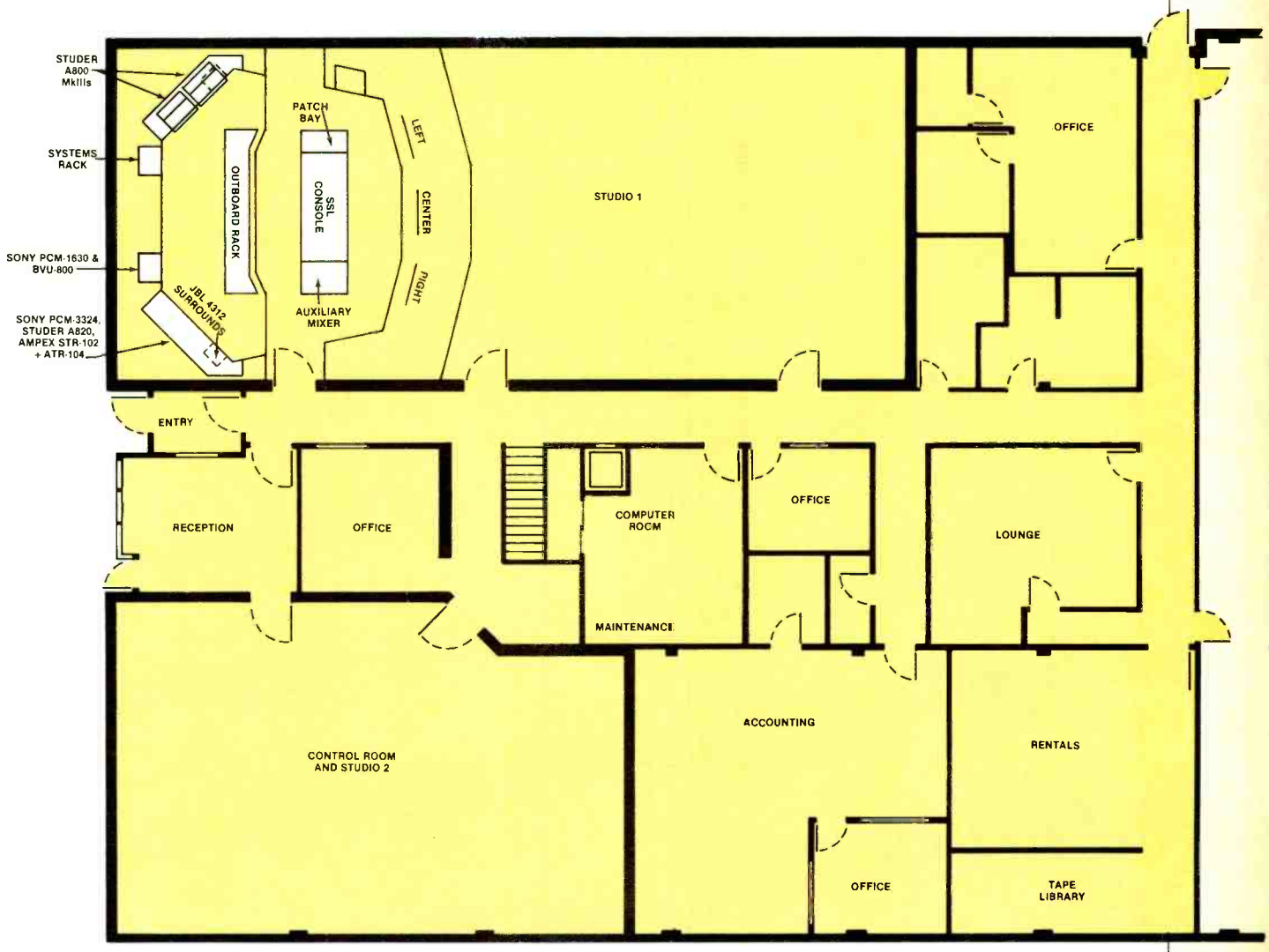
In planning the new Record Plant, a primary objective was to maximize the efficiency of the 2-studio facility. (The Third Street location was laid out as a 4-studio operation.)

Several aspects were involved in achieving this objective. The first was to design the studios so that each could function autonomously, but also allow the two rooms to pool their respective resources and operate synergistically when the occasion demanded. Secondly, the studios were designed not only to provide for the divergent needs of record, film and video production sessions, but also to accommodate the industrywide move toward electronic production using synthesizers and computer-controlled equipment. The third part of the plan was to surround the two studios with the technical and business support facilities necessary to ensure their smooth operation.

"Because of the way the new studios are designed and the way that we are marketing them," Stone says, "we fully expect to get as much volume out of these two rooms as we got out of the four studios we had on Third Street.

"One of the things you have to do in to-

Adrian Zarin is an electronic synthesist, composer and free-lance writer, and a regular contributor to RE/P.



Floor Plan Of New Record Plant Facility.

day's studio business is to take a careful look at your economics and make sure that your utilization factors are as high as possible—that's the only way you can make it pay. If you can make it pay, you can then afford to buy all the equipment you have to provide to clients in order to keep yourself up to date."

In working with acoustician and studio designer Tom Hidley (who in 1969 also designed the original L.A. Record Plant), Stone stressed a need for the new facility to handle everything from a large orchestral scoring date to an all-direct electronic project that would be recorded almost exclusively in the control room. To fulfill the upper end of this requirement, Hidley designed Studio 1, a 40'x35' space with a 25-foot high ceiling, to accommodate up to 40 session players.

"We took into account the fact that we already have a large scoring stage [Stage M] over at Paramount Pictures," Stone explains. "So we knew that we could limit ourselves to 40 pieces at this facility. We also knew that we would have to make Studio 1 an incredible rock room."

The second room, Studio 2, measures 14'x22'x12'. "With the idea, once again, of doing double duty," Stone says. "Studio 2 was designed as a sweetening room and also as an overdub/mix room.

"The Judas Priest project showed us that we could get great rock guitar sounds in there. The first drums recorded in Studio 2 were for an episode of *Fame*—they re-recorded 'Money for Nothing' for one of the shows. The drum sound is just fabulous.

"Our bet is that Studio 2 will bill as much in a year as Studio 1—regardless of its size. Because, basically, the need for a large studio is diminishing. Particularly when you have an overdub room that's big enough to record drums, and will give you an excellent drum sound."

Each of the two studios is also provided with a 6'x8'x10' isolation booth. There are already plans, however, to convert the Studio 1 iso booth into a larger percussion area. This upgrade will be achieved by breaking into an adjoining area behind the iso booth.

Both studios are mated with control rooms that are identical in size, design and equipment complement. By making the control rooms identical, it becomes highly practical to start a project with a large-scale tracking session in Studio 1 and then move over to Studio 2 without sacrificing sound and space consistency.

But, according to Stone, that's only one possible scenario. Both control rooms are large enough—30'x20'—to accommodate a large amount of synthesizer equipment for all-electronic dates. The room design includes a large producer's desk (located directly behind the mixing console) which houses the control room's

selection of outboard gear. Most of the top area was left open, however, to allow convenient placement of synths and related equipment, placing the musician in an ideal central listening spot.

"We looked at things not only in terms of synthesizer equipment," Stone says, "but also in terms of the general industry trends regarding visual music vs. records and 'tapeless' recording. That's why we came up with these very large control rooms; both Hidley and I firmly believe this is the way music is going."

Studios 1 and 2, and their attendant control rooms, are located side by side on the ground floor, separated by a wide, central corridor. By means of numerous tielines, the two studios are linked to various other areas within the facility. One of these latter areas is the Record Plant's maintenance shop, located just behind Studio 2. The shop houses a stockpile of power amps that drive the monitor speakers in all the control rooms and studios. BGW power amps are used for the control room monitors, and Hafler power amps for the studio monitors. The power amp rack also includes a White Instruments room equalizer used for playback in Studio 1.

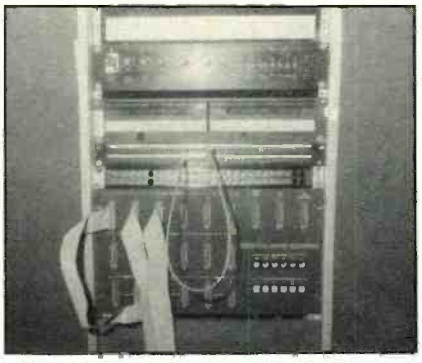
Also located in the maintenance shop is a central patching matrix that routes telecommunication links between studios and ties each studio into the facility's collection of EMT 140 reverb plates housed on the second floor. The shop also provides a central location for the Total Recall automation computers connected to the pair of Solid State Logic SL4000-E consoles.

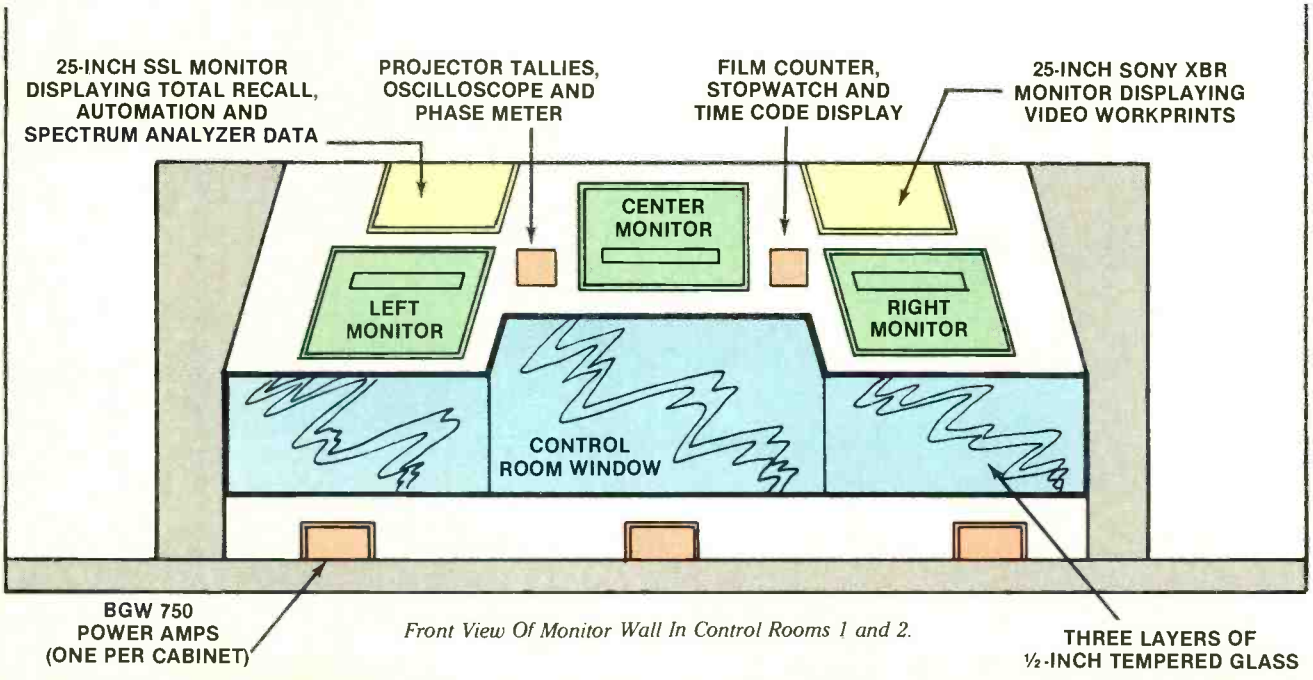
"The two computers are mounted side by side," Stone says. "If the Studio 1 computer goes down in the middle of a 40-piece date, it's just a matter of a simple patch to hook up the other computer—provided Studio 2 isn't using it—and continue the session.

"Also, when we get to 'tapeless' recording [storing digital audio data on hard disk] we know that we're going to be dealing with long cable runs from the control rooms to [a hard disk drive located in] our shop. We know we can

Three custom-designed Hidley-Kinoshita monitors are provided in each control room for 3-channel (left/center/right) monitoring of film and video mixes.

In the rear of Control Room 2 is a custom-built patching system that enables master and slave tape machine reassignments during multiple synchronization setups.





Front View Of Monitor Wall In Control Rooms 1 and 2.

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Pictured in Studio 2 during a music scoring session for the TV series Fame: Michael Stone, session engineer (left) and Phil Jamtaas, assistant engineer with Gary Scott, music producer/composer.

A large selection of outboard signal processors and effects units are within easy reach of the session engineer, assistant and producer.

handle it that way with our SSL computers; data transfer over such distances does work."

Future connections between the shop and the two studios, and between other areas within the complex, will be facilitated by a system of conduits that have been run through the floors and walls. Because the facility essentially was constructed from the ground up (only two outer walls from the original building were left standing), the builders were able to install a generous amount of the conduits, which will be used to route a variety of signals from one location to another.

"When we move toward fiber optics, for example," Stone says, "it will just be a matter of 'pulling on the yellow rope,' as we say—just pulling the fiber-optic lines through the tubes that are already in place. And because we have a number of spare tubes—20 or 25 of them—we can test new technology before committing to it. We can leave a primary system in place while we install a secondary and test it out. Once we make sure the secondary system is really what we want, then we can remove the primary."

Both studios are also linked to a mag/projection room on the second floor above the corridor that runs directly between both studios. Although only Studio 1 has scoring-to-film capabilities (the other room provides scoring-to-video facilities), Studio 2's control room can be used to mix down directly to mag when the application does not require working to picture.

On the ground floor, space has also been allocated for a production/transfer room. When completed, the room will offer transfers "from any format to any format," on a 24-hour basis. It will also serve as a pre-production facility with the capability of handling electronic composition and arrangement, and some video sweetening and layback. Although basic plans for the production/transfer room have been drawn up, the Record Plant design team is still in the process of consulting with Hollywood-based post-production mixers, garnering their suggestions on equipment and services that the room should offer. When completed, the production/transfer room will be fully tied in to both studios.

Control room equipment

The twin control rooms were designed to be equally applicable to both record and film/video work. This means that engineers need to be able to monitor center- and surround-channel audio, as well as the standard left and right channels required for record sessions. Principal monitors in each control room are Hidley/Kinoshita systems, with a quoted low-end capability extending to 30Hz. In addition to the left- and right-channel monitors, an identical center-channel enclosure is centrally positioned above the control room window.

Each of the three enclosures house a pair of TAD TL1601A 15-inch low-frequency drivers, and a TAD TD4001 compression driver. The units are crossed over passively at 630 Hz. Power for each cabinet is derived from a single BGW model 750C amplifier run in bridged mono mode, and rated at 1,000W full load.

"We needed a monitor that would work equally well for film and records," Stone says, "and the new Hidley monitors definitely fill the bill. The 'old' Hidley/Westlake monitors—as we now refer to them—didn't do that, because they were too harsh for most filmmakers. Our hope—and we've had a good response so far—is that these new monitors will have less pressure strain, and be much easier to listen to."

Supplementing the Hidley/Kinoshita systems are JBL 4312 monitors located at the rear of the control room to provide surround-channel monitoring. And as an alternative to the Hidley/Kinoshitas, the studios will offer Meyer Sound model 833 reference monitors. Because the 833s are designed to operate optimally in free-standing applications, Stone explains, they make ideal secondary monitors; they can simply be brought into the control room as needed and placed on stands in front of the console.

To a large degree, equipping the new complex meant selecting the best of the equipment at the old Third Street facility and moving it to Sycamore Street.

"Over the past four or five years that we've been deeply involved in film and video," Stone says, "we've been able to find out by trial and error what equipment works for us, and what doesn't. The things that work were moved over; the others were sold in a giant auction."

Consoles installed in the new control rooms are the pair of SL4000-Es from Studios C and D at Third Street. Before the move, the boards were upgraded from 48 inputs to 56 inputs.

According to Bruce Maddocks, the Record Plant's director of technical engineering, the console's mic pre-amps

Continued on page 38

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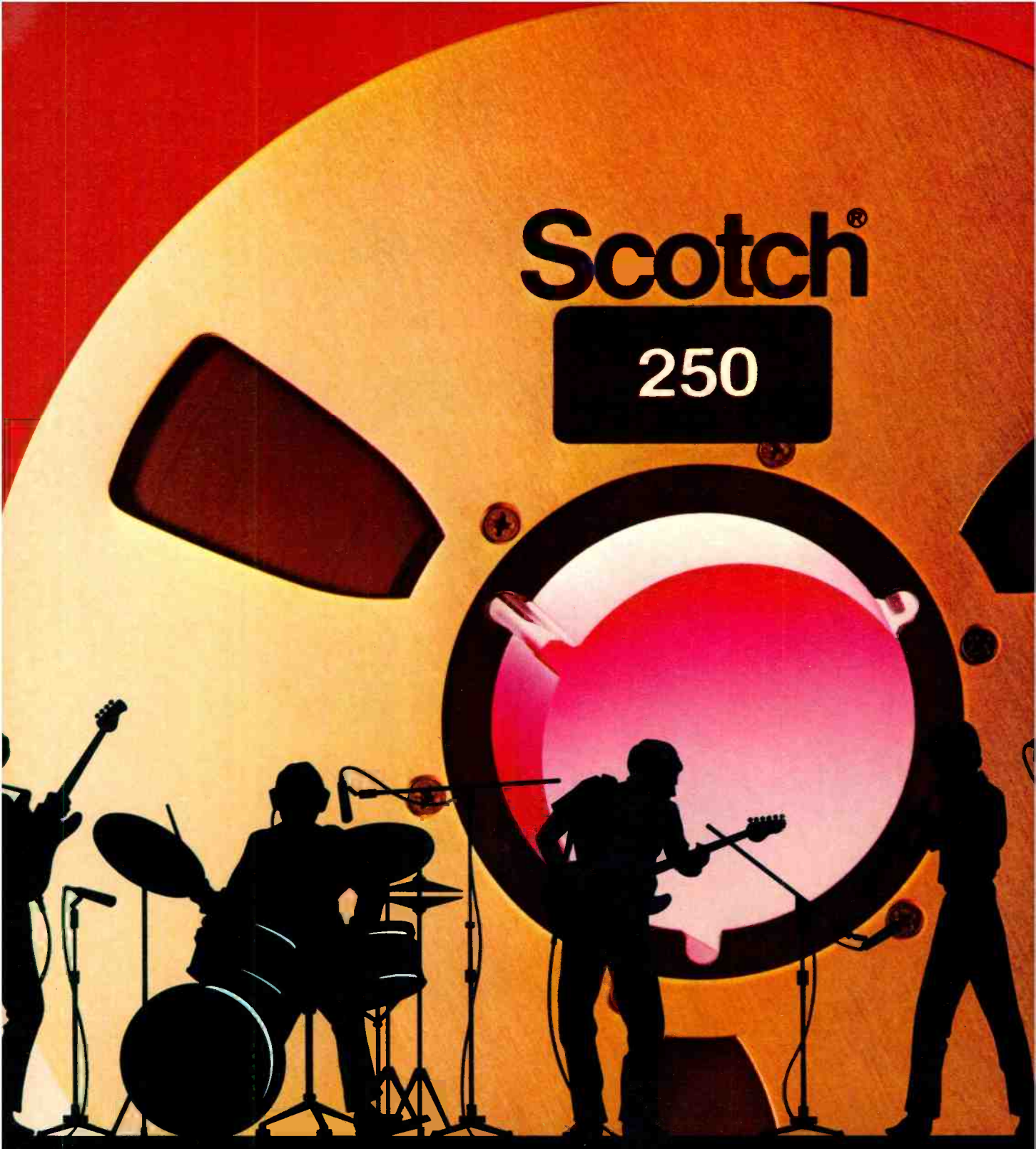

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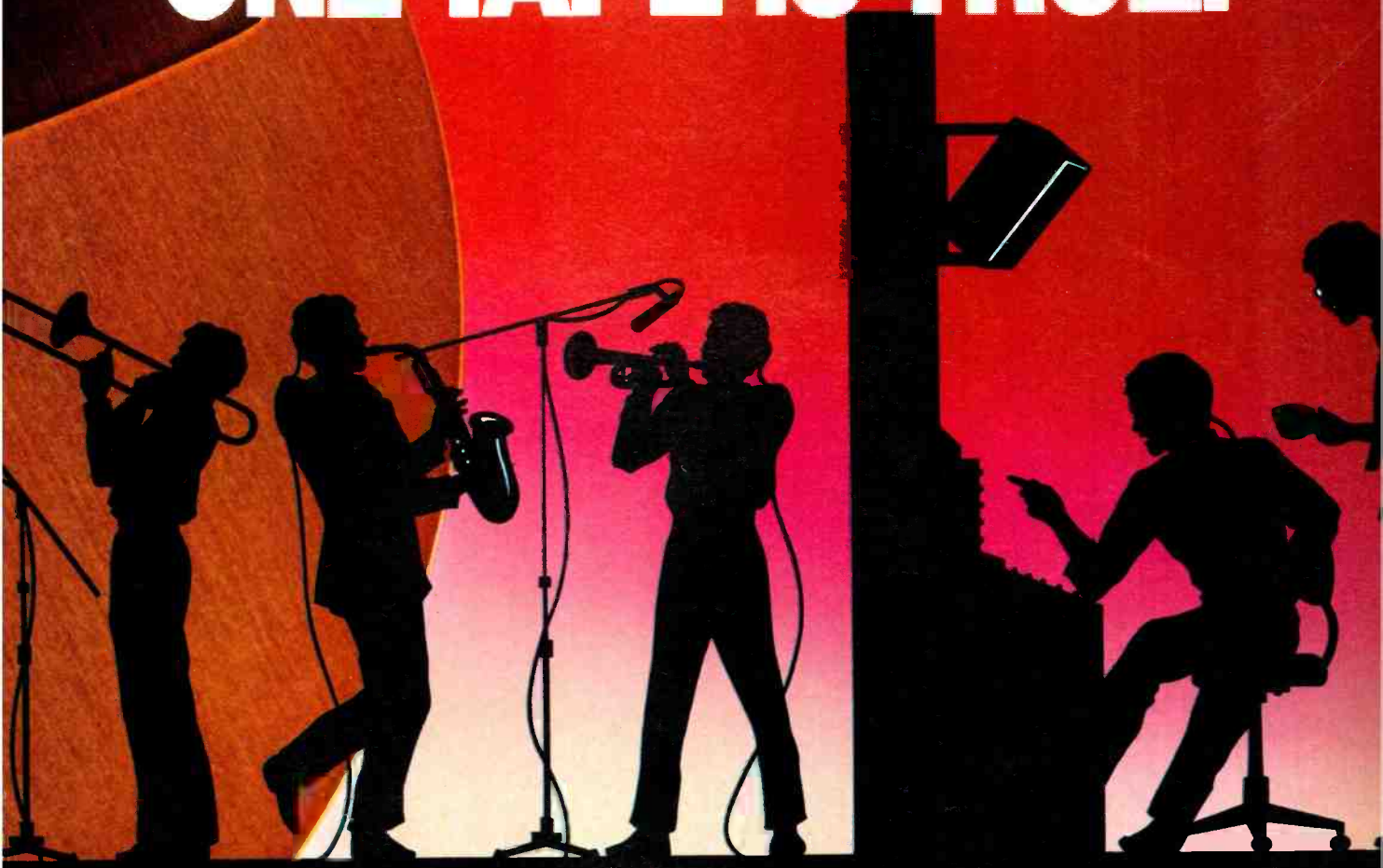

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The left-hand equipment soffit in CR 2 houses a Studer A-820 2-track, an Ampex ATR-102 2-track and a Studer A-800 24-track.

Continued from page 34

were modified "to make them sound smoother, and provide more control over the total amount of gain. Also, the equalization system was converted so that now the sweep frequency and range are closer to that of a Pultec equalizer; it's now a little bit 'warmer' sounding."

To accommodate the left, center, right and surround monitoring systems, the SL651 master facilities module on each console was also modified. As Maddocks explains, this involved reconfiguring the 4-channel monitor source selection.

"In our earlier film-scoring sessions at Third Street, we were working on a 2-monitor system [left and right] and we could produce an electrical phantom center. Now we've gotten away from that, and have changed things to accommodate a discrete center channel.

"We've also worked out 3-channel monitoring for the smaller speakers [JBL 4312s] plus the Dolby Stereo matrix.

Control Room Design Criteria:

A Conversation with Acoustician Tom Hidley

When Chris Stone and Tom Hidley met to discuss the design of the new Record Plant, they discovered that they were both thinking along the same lines in terms of the control room. Impressed with the boom in electronic music production, both Stone and Hidley had become interested in full-sized control rooms that are large enough to handle the demands of synthesizer equipment.

Design and construction of the new facility also occurred at a time when Hidley had just revised some of his thinking on acoustic surface treatment for control rooms. The basic flaw he'd discovered in his older control-room designs was acoustic phase distortion resulting from inconsistent reverberation times among the high-, mid- and low-frequency bands. This acoustic phase distortion, he further concluded, was contributing to listening fatigue in the control room.

Correcting the problem involved achieving better control of first-reflection response around the room. To effect this, Hidley is moving toward the use of a hardwood control room floor and trapping in the rear wall and ceiling. By these techniques, the designer found that he could maintain a consistent, 0.2 second to 0.24 second RT_{60} throughout the mid- and high-frequency ranges.

Both a hardwood floor and trapping are incorporated in the new control rooms. The only modification to Hidley's standard design is the provision of 22-inch equipment soffits on both sides of the rear wall. In other rooms Hidley uses this space, with the rest of the rear wall, for trapping.

Special adaptations on Hidley's basic control room design were also required to accommodate left, center, right and surround monitoring. The extra speakers presented acoustic phase problems not accounted for in Hidley's basic design, geared to left-right monitoring for stereo mixes.

For Hidley, monitor phase coherency and efficiency form a particularly vital area of concern. In co-designing his new control room monitors with Japanese loudspeaker expert Shozo Kinoshita, Hidley was emphatic about using no more than a 2-way system with a crossover network carefully designed to ensure phase coherency between the two speaker components. To eliminate any unwanted resonances, he insists that every Hidley/Kinoshita monitor system be installed to his own specifications. These call for the speaker enclosure to be encased in a 4- to 5-inch thick concrete shell, with an 1/8-inch air gap all around, allowing the monitor cabinet to "breathe."

The additional speakers, particularly the center-channel monitor, presented problems in terms of both mounting position and phase coherency with the other monitors.

"Because of the picture screen, we had to position that center monitor just a little higher than we wanted to," Hidley explains. "And that center monitor is somewhat of a stepchild to an in-line pair of stereo monitors sitting left and right. Fortunately, the center monitor will be sunk back into the stereo imaging picture; it'll enter from a different plane and come down into the room.

"For its application in video mixing, that center monitor does not have to be totally phase corrected to the side monitors that are coming into the room at a different angle. As a result, the monitors will not be working at cross purposes to each other."

In this situation, Hidley's open-trap ceiling design yields an extra advantage: "In the old design, you had a hardwood ceiling. What it amounted to with that center monitor in the old Record Plant Studio C was just unreal—the speaker was blowing directly into the ceiling, with the first reflection coming off the ceiling at whatever angle and then down into the mixing position. The effect was totally different from the left and right monitors, which were a couple of feet away, because the sound had a 4-foot longer path off the ceiling.

"We don't have these problems in the new control rooms, because we're dealing with an open-trap ceiling. The center monitor will drive a lot of ceiling trap; perhaps 40% of its splash efficiency will be in the roof and will never be part of the audio sound field. What comes airborn from the speaker into the listening area will be the bulk of what you're going to hear. In other words, you won't be hearing the first reflection off the ceiling."

Another challenge he faced was the producer's desk located behind the mixing console. The problem, he explains, stems from phase cancellations caused by "unwanted high- and mid-frequency reflections coming back from the producer's desk in an out-of-phase relationship to the direct sound from the front monitors. When that turnaround wave meets the oncoming new signal of the same frequency, and they are not in-phase, what you get is cancellation.

"The problem can be corrected in the mixing position, but if you move backward into the 4-foot area behind the producer's desk, the low end will drop a little bit in power with respect to the rest of the balance. But for a multipurpose room like this, the low-end situation will be acceptable."

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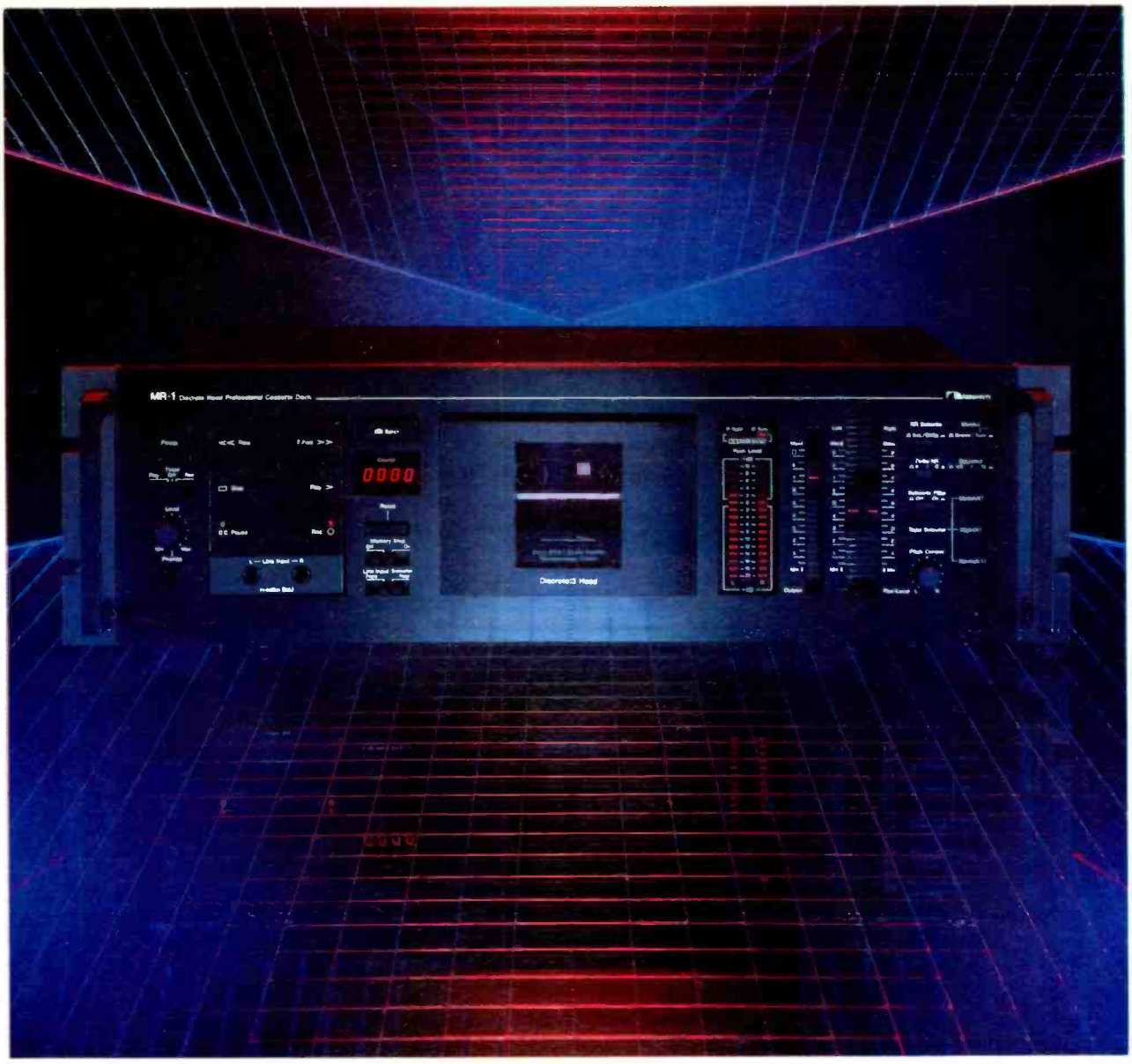

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Studio 2's recording area measures 14'x22'x12' and can accommodate a small rhythm section for sweetening dates or serve as an overdub/mix room.

When the rear speakers are hooked up to the Dolby surround matrix, we can get a full image. Earlier, we just used left, center and right channels; now we can recreate exactly what happens during playback in theaters."

The only operational difference between the control rooms is that Studio 1 is equipped with an 8-bus, 32-channel auxiliary mixing console. An updated version of one that was designed and built by the Record Plant staff for its Studio M scoring facility, the auxiliary console is designed to share important logic functions with the SSL Total Recall automation computer. In this way the auxiliary mixer is said to optimize the music production-oriented SL4000-E for film/video mixing, including dialogue and effects work.

With eight echo sends and eight cue sends, the auxiliary mixer "also comes in handy for building numerous discrete cue mixes for individual players during tracking sessions," Maddocks says.

"We've also incorporated a few new ideas. For example, the design is all modular, which means that there are I/O slices and everything is set up on printed circuit boards. All the summing and distribution amplifiers are mounted on cards in master groups of four. If something breaks, you can just take the card out—it's mounted by means of four Molex connectors—and slide a new card in."

Tape machine control

In each of the rear corners of the control room are 22-inch-wide soffits that contain the tape machines. The left-hand soffit houses the standard analog machines that come with each room: a Studer A-800 MkIII 24-track, a Studer A-820 2-track and an Ampex ATR-104 4-track.

The right-hand soffit contains a Sony PCM-1630 stereo digital processor and BVU-800 ¾-inch VCR. Additional machines, including a Sony PCM-3324 DASH-format digital 24-track, PCM-3302 digital 2-track or a second Studer A-800 MkIII, can be rented.

TimeLine Lynx time code modules are available for synchronizing whatever combination of audio and video machines is required for a given project. Because so many possible combinations of machines need to be accommodated, the studio's staff have designed and built a master system controller. Its function is to simplify interfacing and synchronization among the various tape machines, and to allow control of mag machines located on the second floor.

At the heart of the controller is a system of six machine ports, which allow interface of a master machine and up to three slaves via multiway connectors.

"Through the six ports," Maddocks explains, "you can select any machine as the master and have that machine talk to the Lynx. All the motion controls, tach, time code and direction lines switch over automatically [to control by the master machine] at the push of a button. The other machines will then slave to manual commands from the master machine's function controls, or to commands from the SSL computer, which has motion-control, record, drop-in and rehearse functions."

Another set of master transport controls are located at an "assistant's station" on the producer's desk, which sits behind the mixing console in each control room. In this way an assistant engineer can take care of routine tasks without having to access the master controls located close to either the producer or the first engineer.

"The assistant's station will have tape-machine motion controls, and an auxiliary SSL Total Recall monitor," Maddocks says, "plus a few talkback functions for the 651 [master facilities module]. There will also be a selectable cue system, so that the assistant can track down hums, buzzes and malfunctions. And there's an intercom system that will talk directly to the film room upstairs."

To enable the master systems controller to handle the widely divergent transport logic and ballistics of different machines, the designers have developed a switchable interface card.

"Other people build mystery black boxes," Maddocks notes. "The problem is that you need a *different* black box for every tape machine that rolls in. Instead, we just looked at the cross section of the machines on the market and said, 'OK, everyone's got different tach rates and different direction senses; how can we consolidate all of that?'

"Basically, we came up with one card that contains a series of divide-downs and multiply-ups—depending on what we need. The card also contains tach and direction lines, and a buffer for time code switching.

"Let's say that a Studer machine rolls in. You pull out the card and change the position of a few DIP switches. Slide the card back in and you're ready to go."

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Circle (21) on Rapid Facts Card

The six basic ports of the master controller are supplemented by a system of mults. These serve to make the system expandable, and also enable the Total Recall computer to read time code from any source that is designated as the master.

"As well as direct sends to the machine ports," Maddocks explains, "there are also mults that enable you to add another Lynx module or something like that. In the case of SSL computer mixes, you have to have time code coming into the SSL and going to the Lynx module that has been assigned to the master machine.

"That connection is now made through a distribution system we've devised. Just a push of a button [on the SL651 module] allows you to do one of two things. When the switch is in the up position, you can route time code by means of a patch cord. When the button is depressed, the console automation looks at the system's controller, and whatever machine is selected as master. The latter happens automatically; there are no patch cords, or anything like that involved—it's just a matter of a small relay that makes and breaks connection between the time code read point and the SSL.

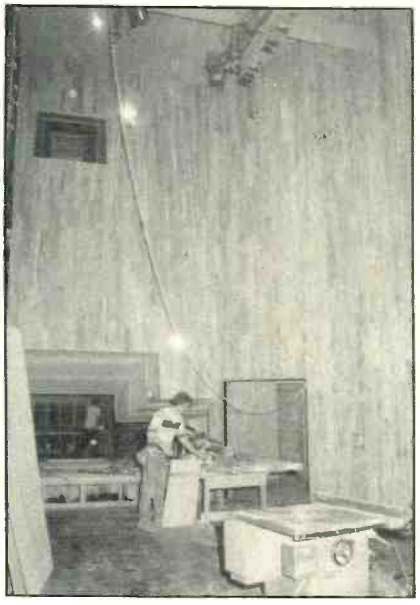
"This system makes it *very* easy to go to a new master machine whenever you need to."

Video facilities

Each control room is equipped with a Sony BVU-800 ¾-inch VCR, which does double duty as a storage medium for PCM-encoded digital audio and as a playback deck for video worktapes. In each control room the VCR works in tandem with a custom-designed video patchbay that provides access to all other video decks in the house. The two studios also provide several options for viewing picture originating from any of these sources. Video playback can be monitored on a Sony XBR 25-inch color monitor, mounted on the right-hand side of the front wall, on a 25-inch SSL multistandard monitor mounted to the left, or on both units.

Studio 2 is also equipped with a Sony model 9000 large-screen video projection system. The screen is positioned so that it can be viewed from inside the control room and from the recording area.

According to Stone, the video projection system provides a very important option for scoring, "because some production companies, such as Lorimar, are now shooting the majority of their footage on videotape, and therefore score the music to videotape. Almost every other television production company does its scoring to 35mm, and so we're prepared for both."



Studio 1's recording area during the construction stages, showing the film projection booth located above the control room.

Mounted beneath one of the control-room video monitors is an LED display that provides a time code readout, along with a readout for time of day and a digital stopwatch. This display furnishes an alternative to viewing time code references that have been burned into a video workprint via a character generator.

Apart from displaying picture and time code, the video monitors in each control room also perform a variety of other functions. The SSL multistandard monitor is used to display the graphics associated with the Total Recall automation computer.

"We've worked out an intelligent video switching system," says Maddocks, "whereby the monitor automatically switches to the appropriate display when it gets a Total Recall command from the SSL computer. You can have, for example, a full RGB display of each individual module's fader position, where the EQ settings were and whether there were any mutes on the bus assigns."

The Sony XBR monitor can also be switched to display the readout from a spectrum analyzer connected to the console's stereo output bus. Further electronic metering is provided in the form of an XY scope mounted beneath the video monitor for checking phase angle of signals on the main stereo bus.

Studio 1 is also serviced by a customized Magna-Tech 35mm projector that has a rewind capability of 8.5X play speed, and can be configured for 16mm as well as 35mm work.

"It's a $50,000 projector that has been hot rodded for scoring," Stone explains. "It was originally designed as an ADR projector, which means it was optimized to do looping. But you only need that kind of loop for ADR and Foley work, and we only do a small amount of that here. So, because of our ability to do our own R&D and manufacturing, we have made it into a scoring projector."

The mag/projector room also houses three MTM 35mm dubbers that currently have a rewind capacity of 6X play speed. Via a custom-designed matrix and a Lynx module, the dubbers and projector have been integrated with the master systems controller in each room, which means they can be operated from the SSL's onboard motion controls. The matrix also allows the output of the control room tape machines to be connected to the dubbers, and provides all the necessary options for monitoring of audio playbacks that originate in the mag/projection room.

Future plans

Apart from the planned production/transfer room and the provisions that have been made for the advent of tapeless recording and fiber optics, Stone has other long-range plans for expanding the studio's facilities. Needless to say, they are characteristically ambitious.

"Our roof is strong enough to install a big satellite dish," he says. "Our feeling is that modems are going to take over, and that more and more people will be calling in their parts from home. Beside that, there's going to be a lot of things happening with uplinks and downlinks. We can have our own downlink here, and we're only three blocks away from Armed Forces Radio and their uplinks.

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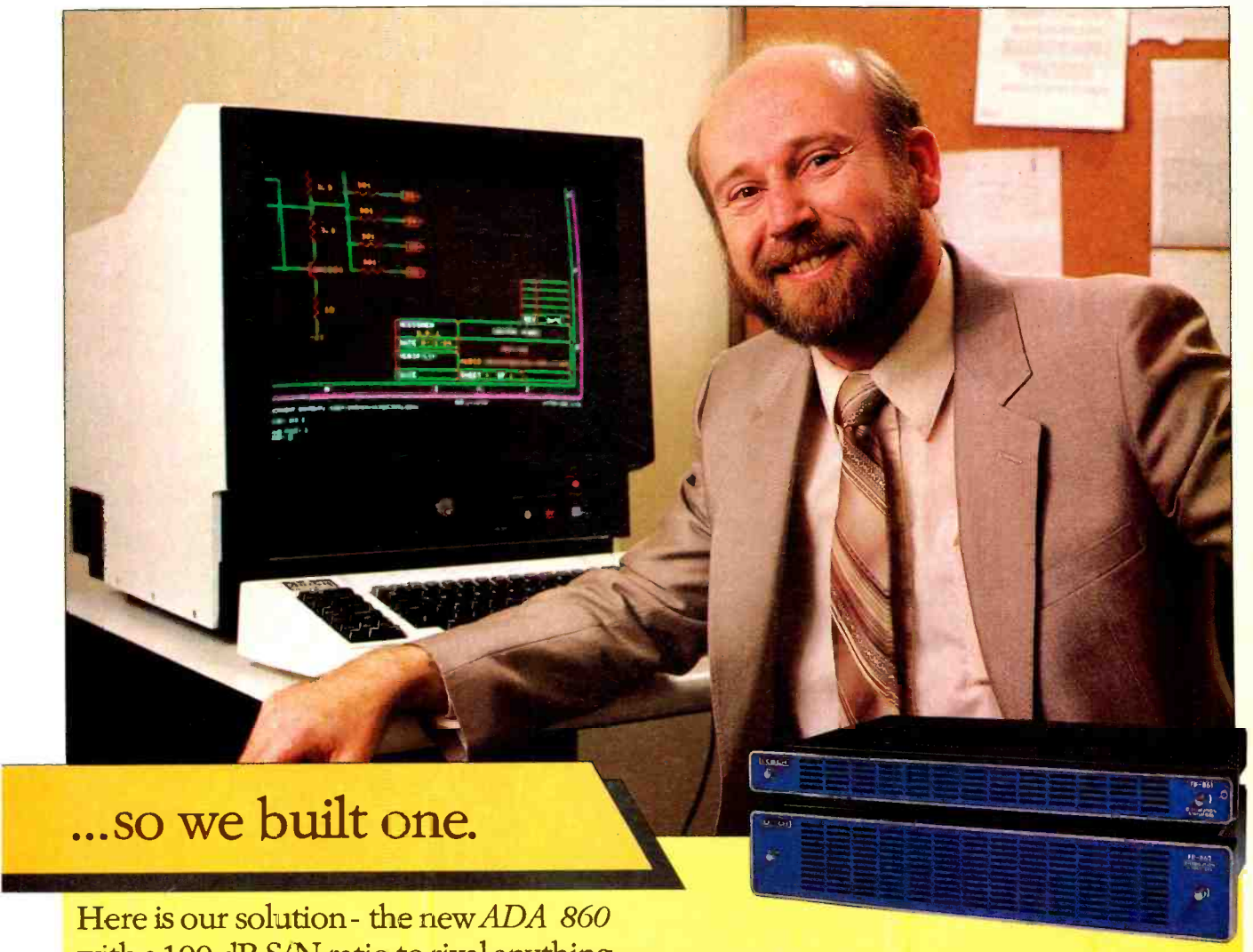

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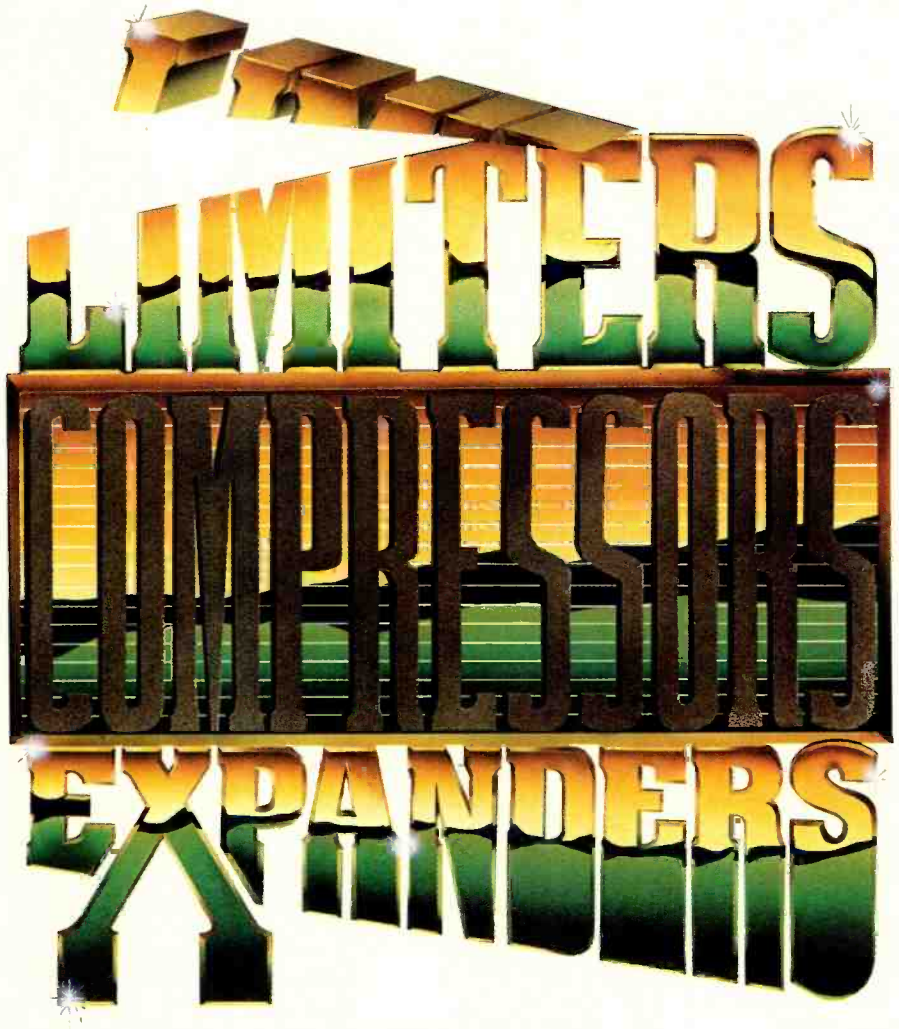

Limiters, compressors and expanders can solve many of today's audio processing problems—just so long as you understand how they operate.

By Richard C. Cabot, P.E.

The limitations of the available media have long been a problem in recording and producing high quality audio. The dynamic range of the human voice or of common musical instruments is much larger than that of conventional discs and tape. Typical vinyl discs and analog tapes provide 60dB to 70dB of dynamic range between the noise floor and the maximum output level. Music can easily exceed a 40dB range between soft and loud passages, yielding only a 20dB S/N ratio on the low-level program material.

Devices were developed to *compress* the dynamic range of signals, making the loud sounds softer and the soft sounds louder. This reduced dynamic range fits comfortably on tape, without the low levels being lost in noise and the high-level sound causing distortion.

Unfortunately, some audio processors actually create problems when dealing with the dynamic range of the human voice and musical instruments. In the case of vocals, these problems include amplifier overload, pickup of room noise and excessive sibilants in speech.

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

Gain changing

Dynamic-range changing devices provide a signal at their output, which *supposedly* differs only in level from the original signal. The device's gain changes with the signal level as needed to alter the dynamic range; the waveform shape theoretically remains the same, but its size (voltage) is made larger or smaller as necessary.

The ideal compressor or expander is like a skilled, and very fast, mixing engineer who rides a fader on the signal, correcting for undesired changes in level. Nothing is changed about the signal but its level.

Because the characteristic of interest for such units is gain, their steady-state operation may be described with a graph of input level vs. output level. On a log-log scale (dB output vs. dB input) we get a graph similar to Figure 1, which is commonly referred to as a *transfer curve*. For

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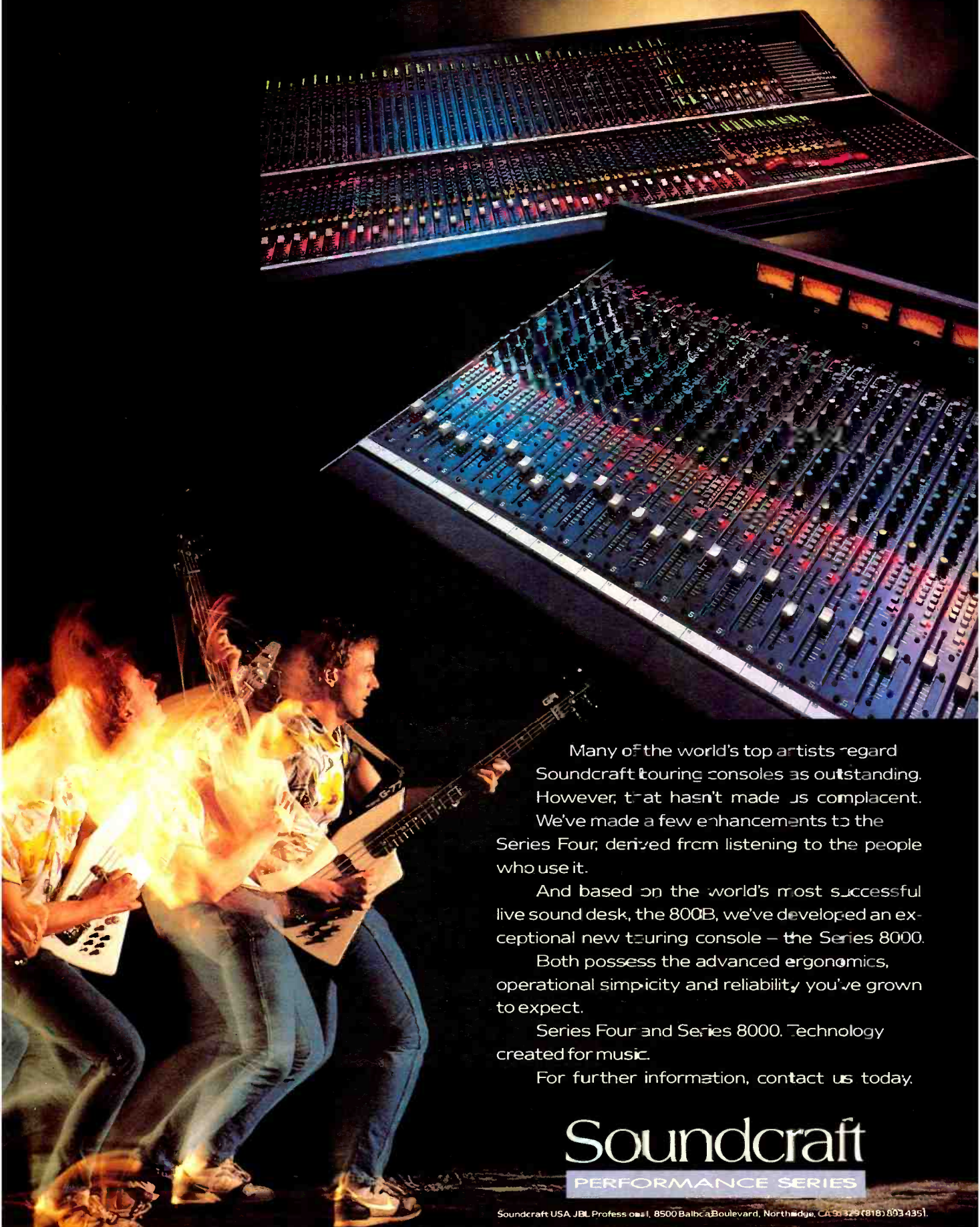

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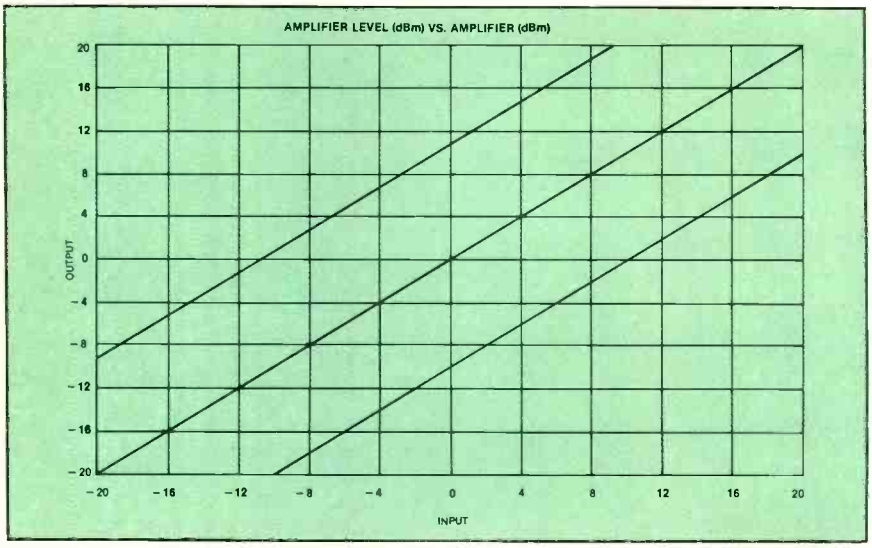

Figure 1. *A basic transfer curve, showing output in dBm vs. input in dBm (log plot). A conventional amplifier produces a line at a 45° angle, with gain determining the actual position on the graph.*

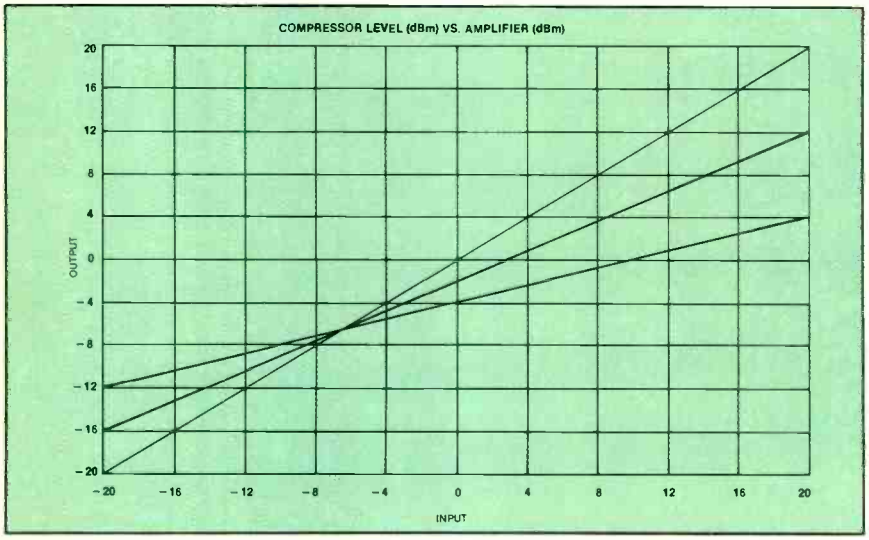

Figure 2. *Typical compression graph showing how the output level will decrease or increase dependent upon the input level.*

a conventional amplifier the graph is a straight line at a 45° angle. The gain of the amplifier determines where the line is positioned on the graph, but the slope is always the same.

Compressors

Musicians and vocalists tend to move around when working into a microphone, causing the signal levels to fluctuate. Sometimes this movement is unavoidable, as with a live performance, and sometimes it is simply due to the vocalist's movement during a take. The solution to such problems usually must be found in the audio processing chain.

When mixing a song for today's competitive airwaves it is often desirable to add more "punch" to the sound in an attempt to capture the listener's attention. The basic philosophy is to make the music fit the listening medium. Often a single's mix, which will get a lot of AM radio play, will be different from the album mix, which will be played on FM or on quality home systems.

This development has led to limiters and compressors that operate in many unusual ways, each optimized for one particular recording application or playback medium.

What we need is a device that will increase the level of the soft sounds and decrease the level of loud sounds, a characteristic that is shown in Figure 2. As the input signal amplitude increases, the output signal amplitude increases by a smaller amount. When the input signal amplitude decreases, the output signal amplitude decreases by a smaller amount. There is always a point where the input level equals the output level, called the unity-gain point.

The slope of the curve is called the *compression ratio*. A compressor whose output level increases by 1dB for every 3dB of input level increase is said to have a 3:1 compression ratio.

The classic block diagrams of a compressor are shown in Figure 3. Compressors come in two flavors: *feedback* and *feedforward*.

The feedback-type compressor is the oldest and most common of the two varieties. In this design, the output signal level is sensed and an appropriate voltage fed back to the gain-control element that precedes it. As the input level is increased, the output level tries to increase. This is sensed by the level-sensor circuit, which in turn drives the gain-control element in an effort to reduce the amplitude. Changing the gain after the level sensor changes the slope of the compression characteristic.

Such circuits are easy to build, and are self-correcting for errors in the gain element or level sensor. However, they guarantee that the output will overshoot its final value when the input level is suddenly increased. This is because the circuits do not see any change in level until the output of the compressor changes, by which time the overshoot has occurred.

Feedforward designs sense the input level and generate the necessary control voltage for the gain element to make the output level change as desired. Because the level sensor sees the signal the instant

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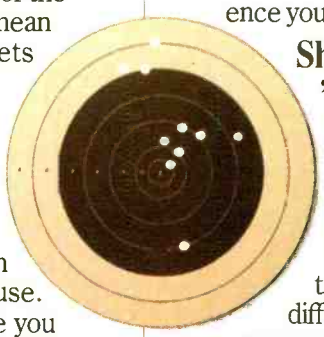

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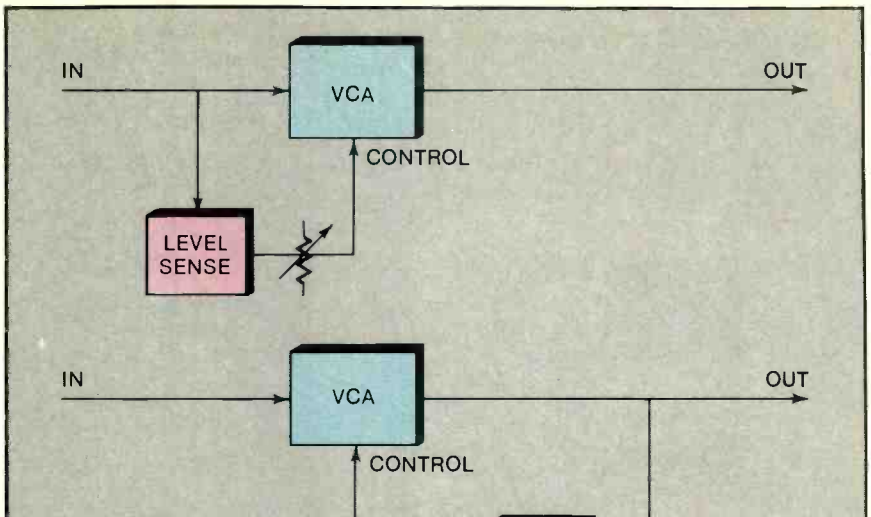

Figure 3. Two basic compressor block diagrams: a typical feedforward compressor circuit (top); and a feedback compressor circuit (bottom).

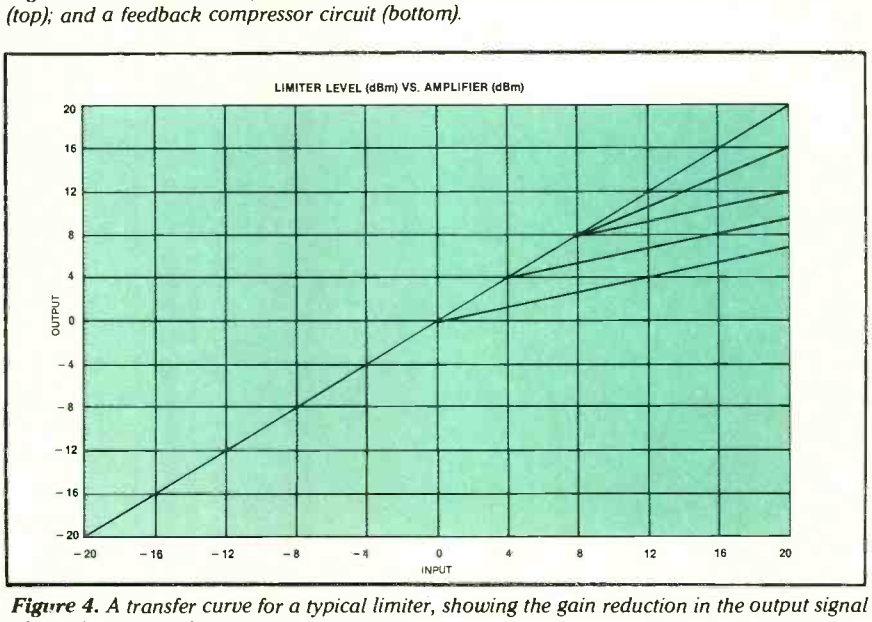

Figure 4. A transfer curve for a typical limiter, showing the gain reduction in the output signal above the threshold level or turnover point.

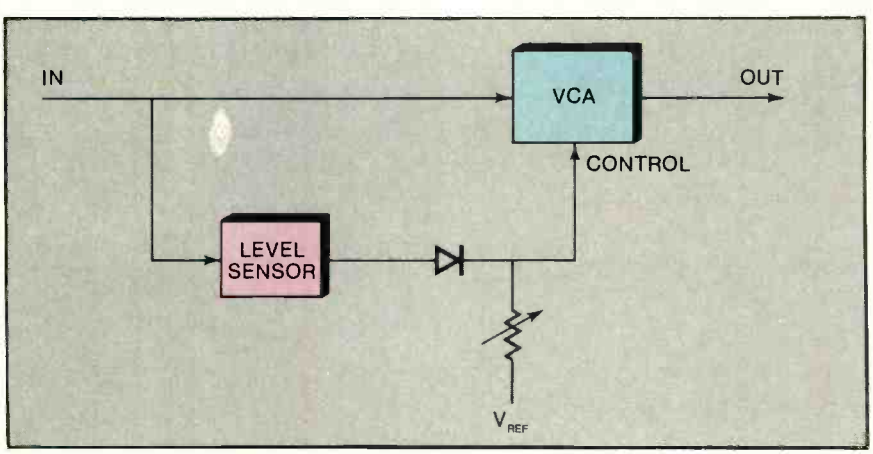

Figure 5. A basic limiter block diagram. A typical compressor circuit may be essentially converted to a limiter by the addition of the diode as shown, for transistor-based circuits.

it is applied; this approach reduces th overshoot problem. However, feedfor ward designs place more stringent re quirements on the accuracy of the level sensing and control circuitry. Any inac curacy in these circuits results in a cor responding error in the gain setting There is no feedback loop to reduce th error.

The graphs shown in Figures 1, 2, 4, and 8 are all straight lines on linear dB scales. This characteristic is obtained with level sensors that output a voltage proportional to the dB signal level Voltage-controlled amplifiers (VCAs) also exhibit similar characteristics, the gain in dB being proportional to the voltage on the control input. It is possible to design a compressor or limiter that has non-linear curves, although they are not very com mon today.

Solid-state circuits lend themselves well to the log and antilog functions re quired for linear dB action. It is possible however, to build a compressor with level sensors and control elements that are not linear dB action. Such circuits are commonly used only in specialized noise-reduction systems, and in simple feedback-type limiters. In any case, the principles are the same.

Compressors can be quite complex devices. Some provide multiband operation in which the compressor divides the frequency spectrum into several bands and then processes each one separately Although this approach produces a subjectively louder sound, the result is a frequency response that varies according to output level.

Limiters

Sometimes a recording or production engineer encounters a signal that is nor mally fairly constant in level, but which occasionally increases suddenly, causing the system to clip or distort. Examples include kick drum, certain vocals, synthesizers and cymbal crashes. To correct such signal fluctuation requires a limiter a device that operates as a normal linear amplifier for signals below some preset input level but becomes a compressor for signals above that level.

Some units found in the studio are composed of both a compressor and limiter. The compressor is used to reduce the signal's dynamic range, while the limiter prevents tape saturation.

A transfer curve for a typical limiter is shown in Figure 4. The level at which the limiter changes from unity-gain operation to compression is referred to as the

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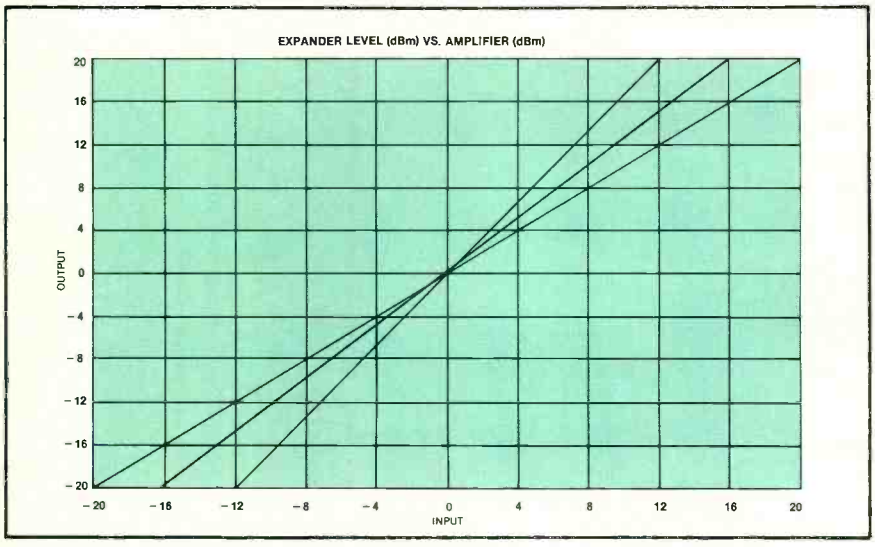


Figure 6. *A typical transfer function graph for an expander circuit, showing the basic function of making loud sounds louder and soft sounds softer.*

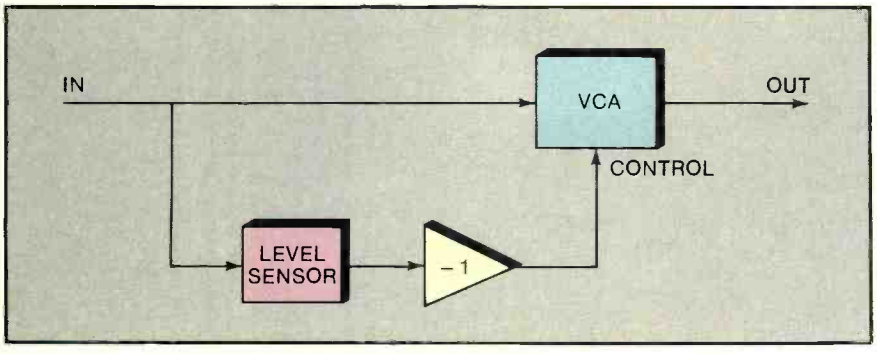


Figure 7. *A basic block diagram of an expander circuit.*

threshold level, or *turnover point*. This point is normally variable, allowing the threshold to be adjusted to match the requirements of the studio operating levels and the program material. As with a conventional compressor, above threshold the compression is characterized by the slope of the transfer curve.

The *knee* in the transfer curve may be sharp, as shown in the graphs, or rounded. Some limiter manufacturers claim that the side effects from a rounded-knee characteristic are less audible; this is a matter for your ears to decide.

A compressor may be converted into a limiter by the addition of a diode before the gain-control element, as shown in Figure 5. The dc voltage from the threshold control is applied to the output side of the diode, forcing the signal level to exceed the threshold before compression can occur.

As with compressors, limiters may be designed in either a feedback or feedforward configuration. A feedforward-type requires predictable characteristics in its level sensor and voltage-controlled element, a straightforward task with transistor-based circuits. If the exact slope of the compression is not of great concern, a feedback-type limiter does not require closely controlled elements.

Limiting thresholds are set by the diode bias voltage, or its equivalent components. The limiting function may be performed with a field-effect transistor (FET), or combination of a light-dependent resistor (LDR) and a light-emitting diode (LED). Such relatively inexpensive designs allow low-cost limiters to be built into power amplifiers or mixing consoles.

Expanders

Expanders are the functional inverse of compressors: They make soft signals softer and loud signals louder, as shown in Figure 6. As can be seen, the slope of the lines is always greater than the 45° slope of a linear amplifier.

If an expander has an increase in output level of 3dB for an increase in input level of 1dB, it is said to have an expansion ratio of 3:1.

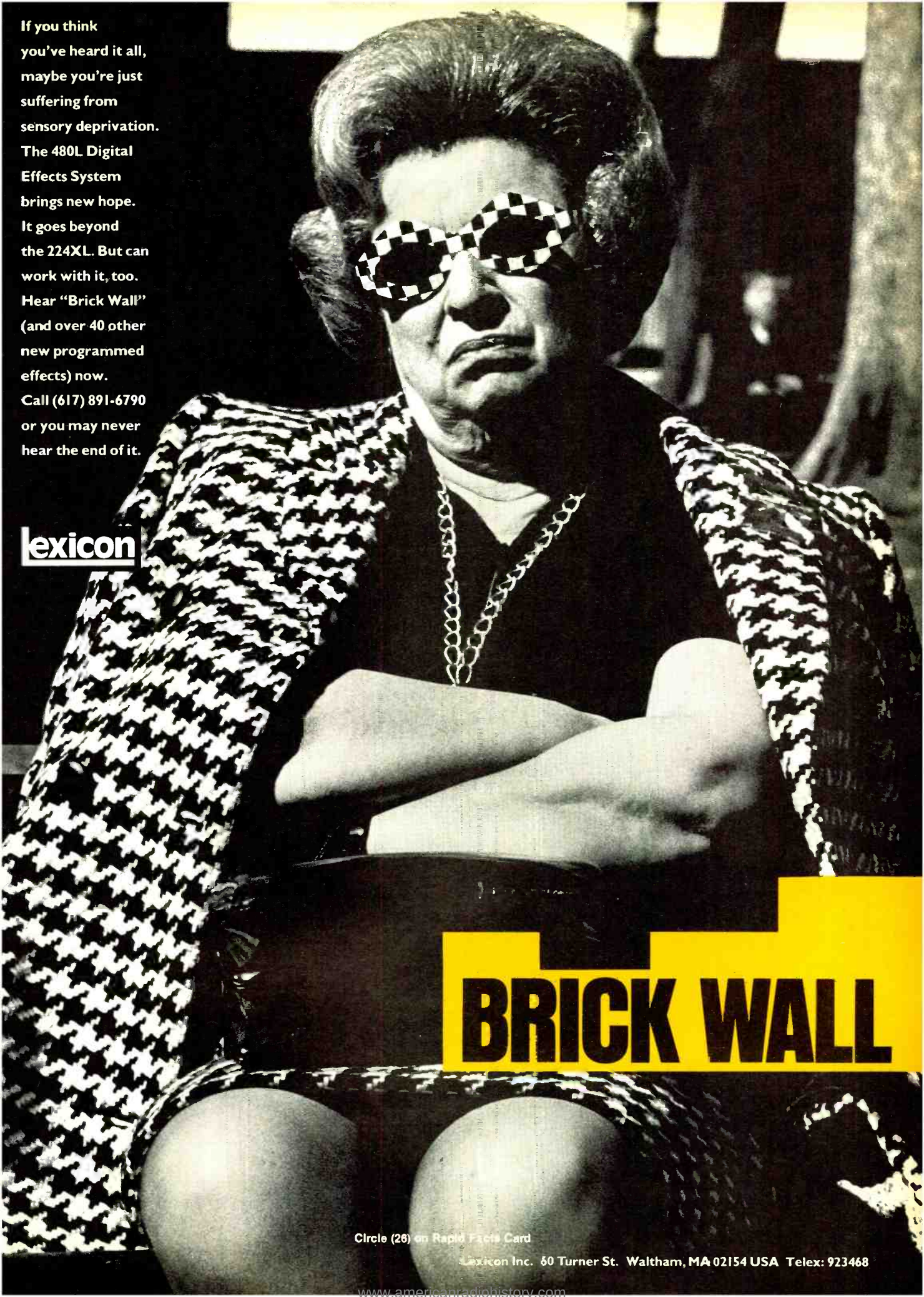
In the block diagram of a typical expander, Figure 7, the only change from a compressor is the addition of an inversion stage to make the gain increase with increasing signal level.

Noise gates

Many recording sessions and live performances use open microphones designed to pick up some desired instrument or vocalist. However, when there is no sound from the desired source the microphone continues to pick up ambient noise. If there are very many of these open mics, the background noise can get out of hand. A similar problem arises when trying to add a track from a noisy tape into an otherwise quiet mix.

What is needed is a way to turn down the gain when the signal level drops below some preset value. A device that does exactly this is the noise gate. Noise gates are to expanders what limiters are to compressors. Above the threshold level a noise gate operates as a normal amplifier; below threshold the gain decreases with decreasing signal level, making soft sounds much softer.

This action effectively gates out, or removes, the background noise, but does not affect the desired signal. The noise-gate characteristic shown in Figure 8 looks much like a limiter's transfer curve



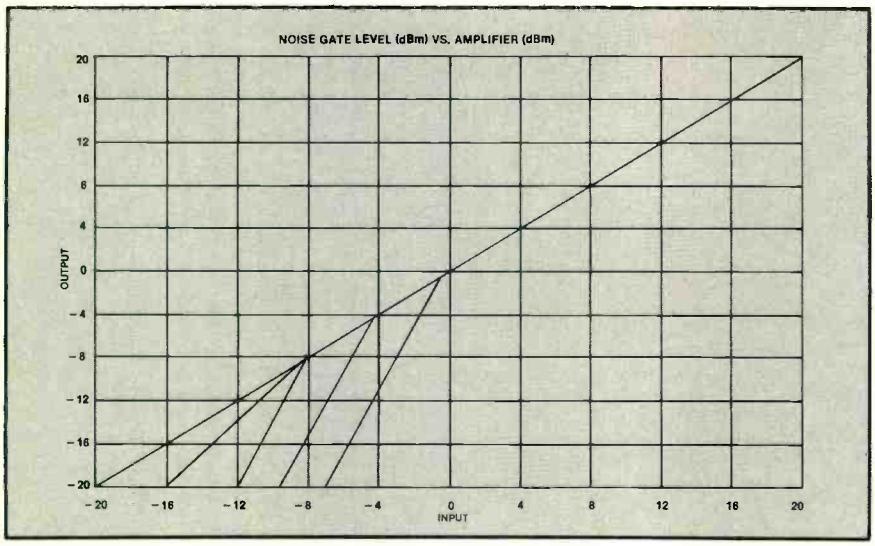


Figure 8. *A transfer function for a typical noise gate circuit. The curve resembles the transfer curve of a typical limiter flipped diagonally from what is shown in Figure 4.*

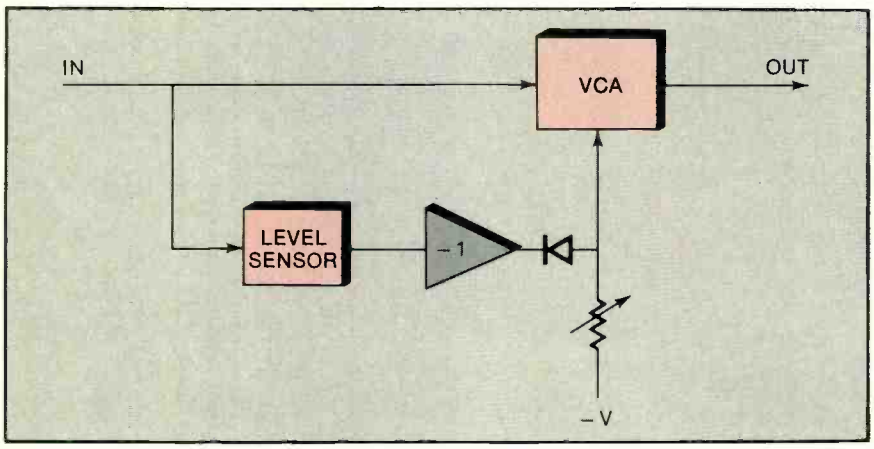


Figure 9. *The basic block diagram of a noise gate.*

that has been flipped over diagonally.

As with a limiter, a noise gate has two important parameters: threshold and expansion ratio. By adjusting the threshold level, the unit can discriminate between the desired signal and unwanted background noise. If there is insufficient level difference between them there will be erratic changes in gain, as the noise gate switches in and out of expansion mode.

The noise gate block diagram shown in Figure 9 looks much like a cross between an expander and a limiter. The inverter is used, as in an expander, to make the gain increase with increasing signal level. However, the diode now prevents the level sensor's output from exceeding the desired threshold—when this occurs the gain is clamped. Below threshold the noise gate functions as a conventional expander.

Noise gates can also be used for special effects. An excellent example of this is the "Phil Collins" gated reverb drum sound. The signal is fed through a reverb unit that has been set to a fairly long RT_{60}. The recirculation signal on the reverb decays to below an appropriate level, which gives a rich sound that ends abruptly when the noise gate cuts in.

Time effects

So far we have considered only the steady-state behavior of these devices. When the signal amplitude changes with time, however, such signal processing is not so easy. Audio signals are, by their nature, ac waveforms that go positive and negative many times per second. It is essential that the signal amplitude be controlled *without* affecting the waveshape of these ac signals.

For example, if the signal amplitude is adjusted too quickly, the waveshape will be changed, causing audible distortion. If it is adjusted too slowly, on the other hand, some signal peaks will pass through unchanged and the compressor has failed to do its job.

Figure 10 illustrates a typical limiter's response to a time-dependent waveform. A tone burst with an instantaneous 20dB change in amplitude is applied to the input. When the signal amplitude increases, the limiter takes a certain amount of time to respond, resulting in overshoot at the output. As the limiter adjusts to the new gain required, the output amplitude decays to the desired value. When the signal amplitude drops, the output level also drops by the same amount. As the limiter readjusts to the new signal level, the output gradually increases.

Many audible problems are related to the time required by the unit to adjust the gain, including the pumping and breathing sounds heard as the medium-level background material is modulated in amplitude by high-level program material.

If, in an attempt to avoid these problems, a limiter is designed to be slow-responding, it will not be able to prevent peak amplitudes from exceeding the desired level. Any design is a tradeoff between audible side effects and incomplete processing.

This situation has led several manufacturers to provide attack- and release-time controls on their units. Such controls

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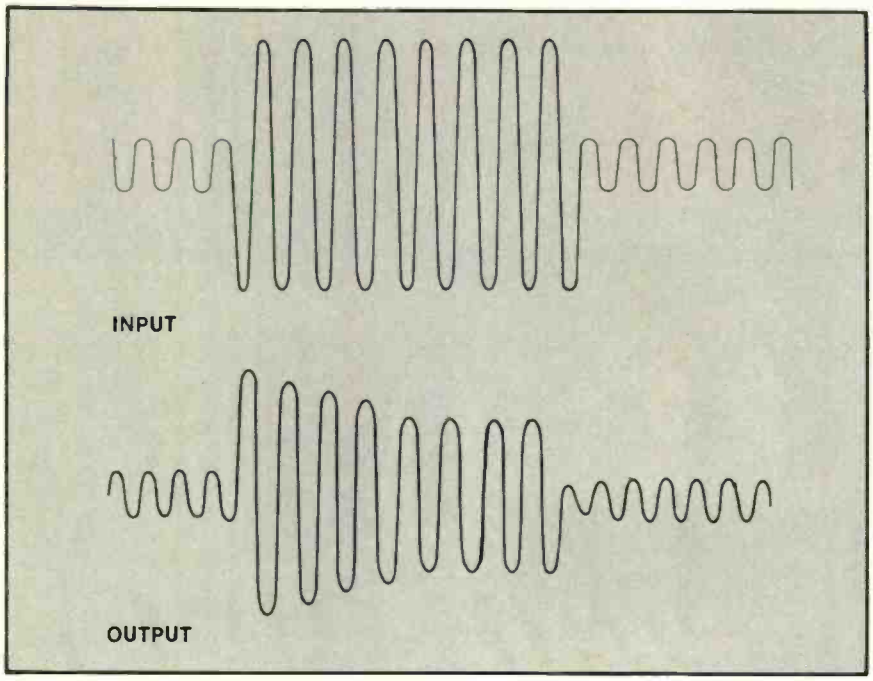

Figure 10. Representative waveforms showing probable tone-burst response for a compression circuit, illustrating the time response of such a unit.

allow an engineer to increase the attack time and reduce the release time until side effects begin to be audible on the particular material being processed.

Adjustable attack and release times also enable the user to operate special effects by adjusting the timing. For example, the attack time may be lengthened to allow the beginning of notes to distort, giving guitars a "gritty" edge to their sound when played hard.

Compressor and limiter features

There are many features available on compressors and limiters that may be important for a particular application. Most professional compressors provide an indication of the gain or gain reduction taking place, and many units also allow monitoring of input or output signal level. This visual aid can be helpful when adjusting drive levels in a system.

Some compressors and limiters allow the control voltages in the level-sense path to be tied to other similar units for use in multichannel systems. If a pair of units is used on the two channels of a stereo mix, tying these points together will prevent the image from shifting because of unequal channel compression.

A few units allow the input to the level-sense circuitry to be patched for special effects. In this way an external equalizer can be inserted into the level-sense path for removing rumble or other low-frequency noise that would otherwise disrupt the level-sensing action. A de-esser is a compressor that has been set to work on high frequencies by the insertion of an HF boost in the level-sense path. Putting a notch filter at the power-line frequency into the level-sense path of an expander or noise gate would allow strong hum to be reduced further than the action provided by a conventional expander. When music is present the filter would mask the hum that would be let through, yet the hum cannot keep the expander gain up.

An unusual, but useful, application for the level-sense input would be to have a microphone picking up background sound to control the level of the main mic channel. With the unit set for a 2:1 expansion, the desired vocal would always be kept a fixed level above the music or noise.

Compressors can sometimes function as remote-controlled attenuators, to provide a remote volume-control function using only dc on the control wires. This might be used, for example, to control the headphone volume in the studio, or could allow a musician to have footpedal control of the instrument level.

Performance specifications

The standard performance specifications of distortion and signal-to-noise ratio are difficult to apply to devices that modify the dynamic range of program material. Because a signal's gain changes with input level, as well as the selected compression or expansion ratio and threshold level, these performance measurements also change.

The noise will generally get worse at high gain values (low signal levels for expanders and high signal levels for compressors). Distortion will sometimes peak at intermediate values or the extremes of gain, depending on the type of gain-control element used. Be wary of units that are specified at only one gain value, or at 1:1 compression ratio. It is best to examine a family of curves of distortion for different gain settings.

Specifications such as frequency response, common mode rejection and maximum input level should be comparable to other types of signal-processing devices. Precautions taken with unbalanced systems apply especially to compressors, which can significantly increase system hum when they have no input signal. Using such a device at different places in the system places differing constraints on residual noise and headroom. Carefully study the system's gain structure before specifying any particular device.

If is difficult to quantify the specifications unique to limiters and compressors in a way that allows meaningful comparison between the audible performance of different units. Attack time and release time are only two aspects of the dynamic behavior. A limiter's distortion performance during the attack portion of its response will significantly alter the perceived distortion for actual program material.

Many compressors and limiters are marginal for headroom, and hard clip on large inputs until the level sensor responds and reduces the gain. Other units are designed with more headroom or an integral softclip circuit, which greatly reduces the level of high-order distortion products during overdrive.

Some compressors and limiters suffer from leakage of the control signal into the output signal path, which results in low-frequency energy at the output during large swings in signal level. This will usually show up in two-tone, difference-frequency intermodulation measurements, if the tones are spaced closely enough in frequency.

Compressors and limiters, more than any other signal-processing device, should pass a listening test before purchase. The necessary performance criteria are not yet well enough understood to simply rely on paper specification. R-E/P

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An Engineer's Guide to Compression and Limiting

By Denis Degher

Specific examples of compressor and limiter settings can provide a valuable starting point when using them to process instrument and vocal sounds.

Compressors and limiters have become to the recording engineer what lens filters are to a camera operator: a way of controlling dissident sound or light elements for enhanced spectrum placement.

You can use compressors and limiters to increase the overall level of program material and hence improve the signal-to-noise ratio, and as a creative device for processing sounds. Compression and limiting ratios, attack and release times, and the degree of compression can help define an engineer's trademark sound, and many of us use specific hardware to record specific instruments and vocals.

The positive implication of this approach is that there is *no* substitute for experience. The negative implication in the artistic world is that mistakes play a major role in the creation of unique techniques and sounds. As with any creative tool, it is the *user* who determines the application and execution of a processing device.

The main control parameters of compression and limiting are peak and averaging (RMS); their response signal levels more or less parallel that of peak and VU meters. An averaging device responds to a signal's average level, like the human ear and a VU meter, and therefore tracks the overall power rather than the transient peaks.

The result is a smoother response characteristic. A peak-responding limiter, on the other hand, is designed to track the audio program peaks like a PPM; it is therefore much quicker responding and can catch instantaneous spikes. Because many of today's most popular compressor/limiters do not offer variable attack and release times, different devices are often used to suit the nature of the sound to be controlled.

Denis Degher is a free-lance engineer and a regular contributor to **RE/P**.

Sound examples

By compressing a **bass drum** by 1dB to 3dB or more during mixdown, using a compression ratio between 2:1 and 4:1, that tight "popping" or "clicking" sound heard on many dance singles can be achieved, without using an inordinate amount of high-end EQ. By using an averaging compressor with a slow attack time, the first transients of the signal envelope will be allowed to punch through before the compressor reduces the sound level above threshold. The result is a sound with a lot of "attack," but without the excessive amount of level that might result from a highly equalized bass drum.

Some form of compression is usually necessary to even out VU levels between notes.

A variation on this theme is to mult the bass drum. By keeping the primary-kick signal natural-sounding for thump, and compressing the paralleled kick signal for click, you can blend the two for power *and* click. By using a very fast attack time the leading edge of the sound envelope can be compressed as well, creating a unique but powerless bass drum effect.

Snare drums can be processed similarly. By heavily compressing the multed snare with a slow release time, a certain amount of sustain can be added by the pumping action of the compressor release. Radical equalization of the compressed channel can be added to create a highly processed sound.

Tambourines and other percussive, metallic sounds that have very high transient peaks can often cause circuit overload and tape saturation. In order to prevent this, peak limiting with a very fast attack time and a compression ratio between 8:1 and 20:1 can be used to help tame those excessive peaks on the leading edge of the sound envelope.

Percussive instruments that are played with sticks or various mallets of wood or metal (including marimba, vibes, steel drum, glockenspiel, tubular bells, triangle, cowbell and timpani) often produce intense overtones and transients. In fact, the overtones of certain notes may almost be as loud as the primary frequency, creating vast swings in level between notes.

Because of the cutting nature of high-frequency instruments, where they will be placed in the mix and the fact that they cause high crosstalk levels between adjacent tracks on analog recorders, correct mixing technique and lower recording levels without limiting or compression is usually more effective than peak limiting or compression.

Midrange and lower frequency instruments are a different proposition. They do not cut through a mix nearly as well as higher frequency instruments, and still may retain heavy harmonic overtones. As a result, some form of compression with relatively fast attack and moderate release times is usually necessary to even out VU levels between notes.

Low-frequency instruments, such as **electric bass,** often need leveling for a variety of reasons, including the way in

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which the pickups are set up, the pickup's frequency response or possibly because of the way the musician is playing. Certain notes tend to jump out audibly and need controlling. Another problem can be caused by level discrepancy between certain notes, which will be obvious when viewing levels on a VU meter.

To ensure proper tape levels, compress the sound by up to 4dB or 5dB, with a 2:1 or 4:1 ratio. If the attack time of the compressor is set too fast, the initial envelope of the note can be affected adversely, causing a punchless, "rubbery" sound. Another problem may occur while using fast release times on low-frequency sounds. Reducing the release time should ensure that the compressor threshold does not pump between the long, low-frequency cycles, or unduly affect the end of the sound envelope.

When recording an **acoustic bass,** many of the above problems are exacerbated by the instrument's acoustical characteristics and the recording environment. **Synthesizer bass** can also create significant level problems, some of which can be reduced by using low compression ratios. For shorter, sequenced **bass synth** sounds, a large amount of compression followed by heavy expansion at a high ratio forces the sound envelope to push its way through the noise gate, thereby creating a tight, expanding sound.

Synthesizers seem to need a certain amount of compression, because no longer are you dealing with a *natural* sound that must be treated carefully to retain certain inherent sonic qualities. In many cases, radical processing may help make the sound more *real.* **Synthesized horns** parts, for example, can often be greatly enhanced by RMS compression at ratios between 1.5:1 and 4:1, and compression from 5dB to 8dB or more, depending on the characteristics of the patch and compressor being used. A thicker, punchier and more dense sound can be achieved as the compressor positively affects the synthesis overtones, by squashing them together.

Synthesizers need a certain amount of compression because you are no longer dealing with a "natural" sound.

Percussive synthesizer sounds often contain jagged and irregular peaks that can be controlled various ways. For level control, use higher limiting ratios (8:1 or greater) on a peak device, or the peak setting on switchable unit, to control excessive level swings without affecting the lower-level program material.

Another approach is to compress the sound at low ratios to reduce both high- and low-level program levels. Again, this technique depends on the sound being created, because you are dealing with synthetic sounds. With **samplers** and **sampling keyboards**, particularly 16-bit devices, the question of what's *real* may become a moot point when sound quality is compared to the analog recording process. The latter's inherent weaknesses and compensations include pre-emphasis, de-emphasis, NAB and AES equalization, record- and playback-head gap loss, and so on.

So, when dealing with high-end sampling technology, some caution must be used to retain the "natural" sound, whatever that may mean today.

The **electric guitar** seems to enjoy compression for many effects. When close-micing a loud rock-and-roll guitar amp, squashing the sound at low compression ratios with a slow release time adds sustain, and definitely makes it easier to place in a mix. Another trick is to heavily compress the ambient mics by between 6dB and 10dB, thereby creating a denser, sustained ambience.

Although heavy compression makes it easier to place a raucus guitar in a com-

What is Gain Reduction?

*There are many misnomers and misunderstandings about how compressors and limiters function. In an attempt to shed some light on their mode of operation and the different types of gain reduction control, **RE/P** offers the following definitions:*

• Leveling is usually taken to mean the long-term control of signal levels. The compression ratio is greater than 10:1, with slow attack and release times. As a result, leveling has negligible or no effect on short-term changes in the average level, or on sudden transients.

• Compression is used to maintain a constant power level; in essence, it prevents levels from falling too low. Benefits include improved signal-to-noise ratio and correction of excessive volume level.

• Limiting prevents a signal from exceeding a preset level and provides protection from overmodulation, overcutting and monitor damage, and elimination of transient peak levels.

There are two basic types: program limiters (referring to the highest VU level occurring within source material); and peak limiters (regarding waveform peak amplitude, measured with a peak program meter or oscilloscope).

When the waveform amplitude cannot be allowed to exceed a predetermined value, peak limiters are used. They are characterized by fast attack and release times, high compression ratios and high threshold levels.

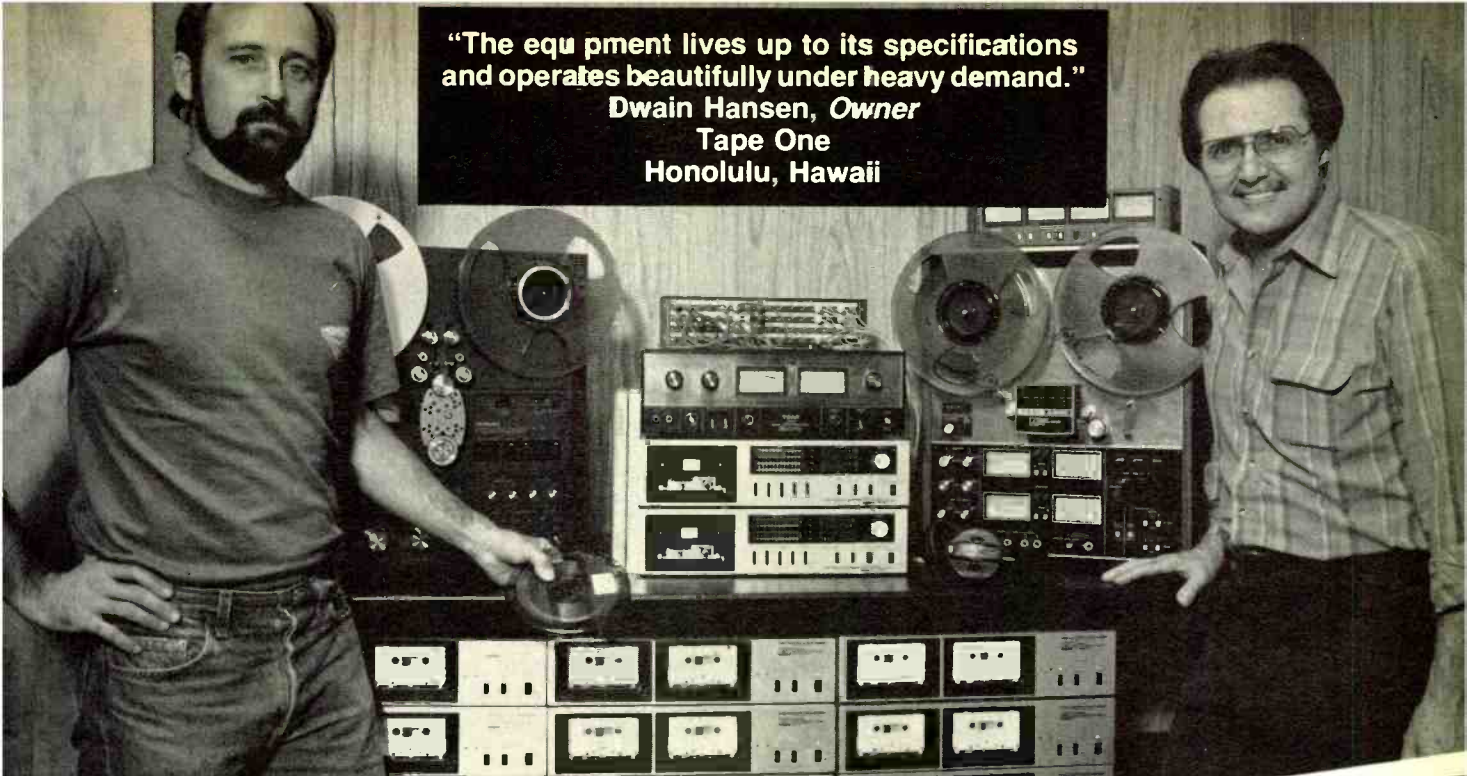
The parameters that characterize limiters and compressors are as follows:

• Compression ratio defines an input-level decibel change corresponding to a user-defined, output-level decibel change. Perfect linear circuits produce a 1:1 compression ratio; compressors and limiters use ratios or slopes that range from 1.1:1 to 30:1. Variable-slope compressors begin with a low compression ratio, and then increase as the level of the input signal rises.

• Threshold refers to user-definable, dynamic range modification. All limiters and most compressors operate below this threshold level as linear amplifiers, with a relative slope of 1:1. In order to be effective, limiters usually operate at a threshold above 0VU.

• Attack time is the time required by a device to bring an input signal under 90% control after it has exceeded a defined threshold. To prevent sudden signal increases from escaping amplitude control, limiters usually incorporate fast attack times. To prevent unwanted effects on transients, compressors use slower attack times.

• Release time is the time required by a device to restore an input signal to 90% of full gain, after it has dropped below a defined threshold. While longer release times result in less apparent volume, faster release times can provide greater volume, but can also introduce more distortion into low-frequency program material.



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plex track, it can have the negative effect of making the overall sound appear smaller. As a result, compression should be approached with care, or avoided, when recording a power-rock trio where the guitar is the only chordal instrument.

When recording direct or close-miced **chordal guitars**, high compression with correct equalization can create a brilliant, shimmering sound with little or no attack. The sound appears to swell after the strum because of the pumping action of the compressor's release.

To help keep **lead guitar solos** highly audible, to add sustain and to avoid level problems during mixdown, compression may be used to compensate for vast level swings between guitar notes. Such level problems may be caused by a variety of factors, including the guitar's fretboard design; the tubes in the guitar amp, which may feature different harmonic frequencies; or the musician's pick attack during the solo, which may send differing amounts of voltage from the pickup.

For **R&B-style, single-note picking**, a slow attack time will allow through the percussive attack before compression begins, and a quick release will pop the level back up between notes, creating the "percolator" effect.

When played as a solo instrument, the **acoustic guitar** usually requires no limiting or compression. Unfortunately, in order to fit a plucked or picked steel string into today's complex mixes, something has to give, and it's usually the guitar's natural fidelity. This is not always bad; many of today's most popular steel string sounds are highly compressed. An example is the popular rock/pop strummed sound that features the string sound over the instrument's resonance.

The **gut string guitar** is another instrument that sounds much better in its natural form than when it's been compressed or limited. For placement in a pop mix, however, it is usually necessary to reduce the dynamic range. Experimentation is the only solution.

Because of an **acoustic piano's** highly transient, percussive nature, level differences between notes and the great dynamic-level possibilities, many engineers favor mild compression with a slow attack time. This lets through the percussive leading edge of the sound envelope before the compressor kicks in.

Personally, I prefer a natural sound with no compression and take my chances with tape compression of the high transients, unless I'm going for that highly compressed, altered sound heard on some records. That particular effect can be achieved with a medium-to-quick attack time at low ratios plus heavy compression, and depends on whether you want compressed sustain (a slower release) or the end of the note to pop back up (a fast release).

Occasionally, real **horn sections** will need a decibel or two of compression at low ratios to blend them onto one track, or in stereo to maintain an even balance between instruments. Overcompression may result in an unnatural sound. **Horn solos** may also need touches of compression to avoid tape saturation on peaks.

Excessive compression can bring up the breath factor too much in comparison to the note.

Flutes of all types can produce large level jumps between notes that wreak havoc on RMS meters, but without being aurally noticeable. Another problem with flute is the breath component of the overall sound—excessive compression for note leveling can have the adverse effect of bringing up the breath factor too much in comparison to the note. The problem should first be tackled with mic placement; finding the correct mic(s) and proper orientation to the flute's mouth piece will be far more successful than using heavy compression or limiting.

Some engineers like to compress the close mics on **violins** and **violas** for blend control. I refrain from compressor limiting of strings, and prefer an ambient, open-micing technique that uses the room acoustics to capture the sound. The only exception to this rule are **cellos**, which usually are more closely miced and sometimes require slight compression to even out different notes caused by terse bowing, or from a pizzicato section.

Theoretically, no compression is the best compression when recording **vocals**. However, this cannot always be accomplished on many of today's dense, rhythm-oriented pop records. Finding the correct vocal perspective within the mix is possibly the most critical aspect of mixing such tracks. The artist must be readily audible without totally dominating the track. Recording vocals requires both sensitivity in dealing with the artist and attention to sonic integrity.

Many considerations enter into the recording of vocals, including the type of microphone being used, the room sound, the use of compression, what device is used and the vocal perspective in the cue mix, which in turn will affect how loud the vocalist sings. Just as certain microphones flatter certain vocalists, certain compressors may also prove to be complimentary to specific voices, depending on the unit's ballistics. What makes certain vocalists a challenge to record are large swings in dynamic range, in some cases 20dB or more between a song's pianissimo and fortissimo sections.

What further complicates the situation is that some voices do not readily accept compression without compromise, resulting in an unnatural sound. For voices that do not like compression, use very low compression ratios, light compression, moderate attack and release times, and gain riding to retain a more natural sound.

By feeding the compressor from the mic pre-amp, very little effective control is provided, and large amounts of level may reach the compressor and cause overcompression. By connecting the compressor across the console's channel bus, the input channel fader can now be used to control the amount of gain that reaches the compressor, thereby avoiding overcompression.

Certain artists with thinner, less intense voices crave compression, and may actually benefit from 5dB to 7dB of compression at ratios from 4:1 to 2:1. The use of compression seems to positively affect the harmonic overtones, creating a richer, thicker and more powerful sound.

Background vocal ensembles may be treated in a variety of ways, depending on the desired sound. If the group cannot blend with itself from a level perspective, compression for level control may be necessary. If you are after punchy, grabbing backing vocals, heavy compression can be used for effect. Another trick that some engineers employ is level *ducking*, where certain instruments are keyed to drop in level when the vocal enters, allowing for placement within the mix, rather than the "over-the-top" perspective.

Heavy compression (8dB to 20dB) of **ambient micing** represents another effective use of compressors that seems to be in vogue for many of today's records. This effect can be accomplished by com-

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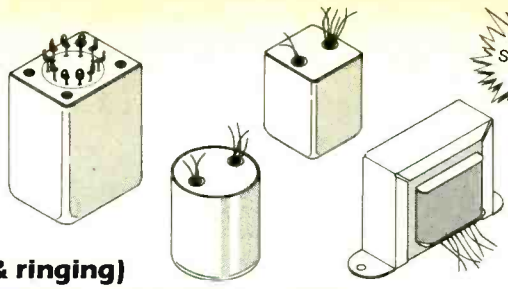

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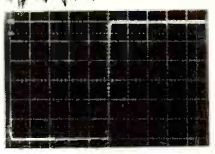

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INPUT TRANSFORMERS AND SPECIAL TYPES

Model	Application	Impedance Ratio Pri-Sec	Turns Ratio Pri:Sec	20Hz Max Input Level[1]	Typical THD Below Saturation (%) 20 Hz / 1 kHz	Frequency Response (dB ref. 1 kHz) 20 Hz / 20 kHz	Band-Width[2] –3 dB @(kHz)	20 kHz Phase Response (degrees)	Over-Shoot (%)	Noise Figure (dB)	Magnetic Shield[4] (dB)	Number of Faraday[4] Shields	Package[5]	PRICES 1-19	100-249	1000
MICROPHONE INPUT																
JE-16-A JE-16-B	Mic in for 990 opamp	150-600	1:2	+8	0.036 / 0.003	–0.08 / –0.05	230	–8	<1	1.7	–30	1	A = 1 B = 2	75.42 82.89	49.87 54.81	34.40 37.81
JE-13K7-A JE-13K7-B	Mic in for 990 or I.C.	150-3750	1:5	+8	0.036 / 0.003	–0.09 / –0.21	85	–19	<2	2.3	–30	1	A = 1 B = 2	75.42 82.89	49.87 54.81	34.40 37.81
* JE-115K-E	Mic in for I.C. opamp	150-15 K	1:10	–6	0.170 / 0.010	–0.50 / +0.10	100	–16	<7	1.5	–30	1	3	54.81	36.24	28.39
LINE INPUT																
JE-11P-9	Line in	15 K-15 K	1:1	+26	0.025 / 0.003	–0.03 / –0.30	52	–28	<3		–30	1	1	122.22	80.82	55.75
JE-11P-1	Line in	15 K-15 K	1:1	+17	0.045 / 0.003	–0.03 / –0.25	85	–23	<1		–30	1	3	52.32	34.59	27.10
JE-6110K-B JE-6110K-BB	Line in bridging	36 K-2200 (10 K-600)	4:1	+24	0.005 / 0.002	–0.02 / –0.09	125	–12	<1		–30	1	B = 1 BB = 2	73.95 85.59	48.90 56.59	35.88 39.04
JE-10KB-C	Line in bridging	30 K-1800 (10 K-600)	4:1	+19	0.033 / 0.003	–0.11 / –0.08	160	–9	<2		–30	1	3	53.17	35.16	24.53
JE-11SSP-8M	Line in / repeat coil	600 / 150-600 / 150	1:1 split	+22	0.035 / 0.003	–0.03 / –0.00	120	–9	<3.5		–30	1	4	194.63	128.69	88.78
JE-11SSP-6M	Line in / repeat coil	600 / 150-600 / 150	1:1 split	+17	0.035 / 0.003	–0.25 / –0.00	160	–5	<3		–30	1	5	98.39	65.06	44.88
SPECIAL TYPES																
JE-MB-C	2-way[3] mic split	150-150	1:1	+1	0.050 / 0.003	–0.16 / –0.13	100	–12	<1		–30	2	3	44.85	29.65	23.24
JE-MB-D	3-way[3] mic split	150-150-150	1:1:1	+2	0.044 / 0.003	–0.14 / –0.16	100	–12	<1		–30	3	3	76.19	50.37	39.42
JE-MB-E	4-way[3] mic split	150-150-150-150	1:1:1:1	+10	0.050 / 0.002	–0.10 / –1.00	40	–18	<1		–30	4	1	114.40	75.64	52.18
JE-DB-E	Direct box for guitar	20 K-150	12:1	+19	0.096 / 0.005	–0.20 / –0.20	80	–18	<1		–30	2	6	54.56	36.07	28.23

1. (dBu) Max input level = 1% THD; dBu = dBv ref. 0.775 V
2. With recommended secondary termination
3. Specifications shown are for max. number of secondaries terminated in 1000 ohm (typical mic preamp)
4. Separate lead supplied for case and for each faraday shield
5. Except as noted, above transformers are cased in 80% nickel mu-metal cans with wire leads.

PACKAGE DIMENSIONS:

	W		L		H
1 =	1 5/16"	Diam.		×	1 9/16"
2 =	1 3/16"	×	1 3/16"	×	1 5/8"
3 =	1 1/8"	Diam.		×	1 1/16"
4 =	1 1/2"	×	1 3/4"		2 1/2" w / solder terminals
5 =	1 5/8"	Diam.		×	1 3/4"
6 =	1 1/8"	Diam.		×	15/16"

NICKEL CORE OUTPUT TRANSFORMERS[6]

Model	Construction	Nominal Impedance Ratio Pri-Sec	Turns Ratio Pri:Sec	20 Hz Max Output Level[7] (dBu)	across (n) windings	600 Ω Load Loss (dB)	DC Resistance per Winding	Typical THD Below Saturation (%) 20 Hz / 1 kHz	Frequency Response (dB ref. 1 kHz) 20 Hz / 20 kHz	Band-Width –3 dB @ (kHz)	20 kHz Phase Response (degrees)	Over-Shoot (%)	Package	PRICES 1-19	100-249	1000
JE-11-BMCF	Bifilar 80% nickel	600-600	1:1	+26	1	–1.1	40 Ω	0.002 / 0.002	–0.02 / –0.00	>10MHz	–0.0	<1[9]	7	81.55	53.92	37.76
JE-11-DMCF	Bifilar 80% nickel	600-600	1:1	+21	1	–1.0	38 Ω	0.004 / 0.002	–0.02 / –0.00	>10MHz	–0.0	<1[9]	8	56.32	37.24	25.69
JE-123-BLCF	Quadfilar	600-600 150-600	1:1 1:2	+32	2	–1.1	20 Ω	0.041 / 0.003	–0.02 / –0.01	>450 170	–1.9 –4.0	<1[8]	7	73.85	43.14	29.76
JE-11SS-DLCF	Bifilar split/split	600-600 150-600	1:1 1:2	+27	2	–1.0	19 Ω	0.065 / 0.003	–0.02 / –0.01	>10MHz 245	–0.0 –2.5	<1[8]	8	53.62	35.45	24.46
JE-11-ELCF	Bifilar	600-600	1:1	+23.5	1	–1.1	40 Ω	0.088 / 0.003	–0.03 / –0.00	>10MHz	–0.0	<1[9]	9	36.36	24.04	16.59
JE-11-FLCF	Bifilar	600-600	1:1	+20.4	1	–1.6	58 Ω	0.114 / 0.003	–0.03 / –0.00	>10MHz	–0.0	<1[9]	10	27.36	18.09	12.48
JE-112-LCF	Quadfilar	600-600 150-600	1:1 1:2	+20.4	2	–1.6	29 Ω	0.114 / 0.003	–0.03 / –0.01	>450 205	–1.2 –3.2	<1[8]	10	32.80	21.69	14.96
JE-123-ALCF	Quadfilar	66.7-600	1:3	+26.5	3	–1.3	8 Ω	0.125 / 0.003	–0.04 / +0.06	190	–4.6	<6[8]	8	50.96	33.69	23.24
JE-11S-LCF	Bifilar w / split pri.	600-600 150-600	1:1 1:2	+30	1 (sec)	–1.7	63 Ω	0.058 / 0.002	–0.02 / +0.01 –0.02 / –0.05	>10MHz 155	+1.1 –4.1	<1[8]	8	50.96	33.69	23.24

6. Multifilar construction has no faraday shield: cannot be used as input transformer. All specifications are for 0 Ω source, 600 Ω load.
7. Max output level = 1% THD; dBu = dBv ref. 0.775 V
8. Source amplifier –3 dB @ 100 kHz
9. Source amplifier –3 dB @ 200 kHz
10. Output transformers are horizontal channel frame type with wire leads, vertical channel frames available. PC types available.

PACKAGE DIMENSIONS:

	W		L		H	Mounting Centers
7 =	1 1/2"	×	2 5/16"	×	1 15/16"	2 13/16"
8 =	1 5/16"	×	1 15/16"	×	1 5/8"	2 3/8"
9 =	1 1/8"	×	1 11/16"	×	1 3/8"	2"
10 =	1 1/16"	×	1 7/16"	×	1 3/16"	1 3/4"

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pressing overall room mics, or keyed ambient mics.

Another area for creative compression is **special effects,** including reverbs, echo and delay lines. Depending on whether you are compressing the sends or returns to such outboard units, certain parts of the sound can be affected, depending on the device and the effect you're after.

Compressing the **stereo bus** during mixdown before the 2-track is a popular effect, particularly on technique consoles that feature an onboard stereo program compressor. This technique can smooth out the entire mix, but may prove unnecessary when various tracks requiring compression within the mix have already been processed. (The net result of this trick is to simulate the heavy compression that occurs prior to the signal reaching an AM or FM transmitter, thereby giving a single or album mix the same sound whether it is played over the air or on a home stereo system. Such processing has serious implications, however, since it can take the life out of the mix by inhibiting the song's dynamics.)

The advent of Compact Disc and digital multitracks in the studio is changing the way we approach recording, and the way in which compressors are being used. With digital's greater dynamic range and absence of tape hiss, packing the tape for maximum level is no longer a major priority. Because the 16-bit binary code does not recognize tape noise like analog tape, lower levels can be recorded without sacrificing signal-to-noise gains.

Care and taste must be exercised when using these compression techniques—they do not apply to all sounds.

In the digital domain, however, a new reason for compression or peak limiting exists, because digital recorders are totally unforgiving when the bit ceiling has been reached—unmistakable distortion and clipping will result. Because the digital domain does offer greater dynamic range, mixdown compression may still be desirable for sound placement within the stereo mix. Using compressors for creative sound, processing will be appropriate for achieving effects that have become popular.

As a recording engineer, I have found that care and taste must be exercised when using these compression techniques, simply because they do not apply to *all* musical styles and sounds. Compressor/limiters can have a great positive value, but may also have a negative impact on audio fidelity, transparency and transient response, making some material sound too clean, controlled and static.

Compression should be used appropriately to achieve specific sound goals, not as matter-of-fact processing.

R-E/P

Acknowledgement: Thanks to Joe Chicarelli, David Sherfey, Peter Chaiken and all the other engineers in the recording industry who may have invented many of the tricks we take for granted.

Part 2

Photo by Les Fincher

Digital Sound for Laurie Anderson's "Home of the Brave"

By Larry Blake

Complex problems of digital post can be minimized by careful planning during preproduction and shooting.

Part 1 of this article, in the August issue, described recording of the live concert audio-to-digital multitrack, various synchronization schemes to ensure audio-video lockup during post-production and the preparations made before the conformation and editing of the digital multitrack tapes following picture editing. In this, the concluding part, we look at the intricate conforming stages, remix-to-picture of the Dolby Stereo 2-track film mix and the soundtrack album mix.

After completing picture editing, the next production step for *Home of the Brave* was to assembly edit the A reel (playback) and B reel (live) PCM-3324 digital multitrack tapes to create an edited, 24-track C reel, which would conform to the edited 10-minute film reels.

An audio EDL, made from an edited mag worktrack run through a time code reader, was created for use during the multitrack conforming stage. In preparation for multitrack editing and mixing, the film workprint and audio worktrack were transferred to 1-inch videotape.

A guide track was recorded on track 1 of the videotape, with the new time code for the reel recorded on track 2. This tape was used later to jam sync and prestripe the time code and guide track on the multitrack C reels.

During the conforming stage, Ken Hahn of Sync Sound operated the facility's custom time code editor, while Leanne Ungar kept tabs on channel assignments and, when necessary, combined tracks (Figure 1). Two Sony digital 24-tracks were at Sync Sound for the seven weeks of editing and mixing for *Home of the Brave*, and their first use was in the creation of the C reels. Because they only had access to two machines, Hahn, Ungar and soundtrack producer Roma Baran first transferred the live material from the B reels, and then the A reels were put up and transferred in sync.

This sequence was followed because the cut time code from the balance stripe—which was both burned into the video window and written up in the logs created by the assistant editors—corresponded to locations of these edits (that is, conforming the material on the B reels in sync to the C edit master) they noted the time code of the A reel, which was recorded on track 24 of the B reel.

Ideally, there would be no need to edit the A tracks unless the length of the song was changed, because all takes played back to the same track which, of course, was locked to a stable time base. However, if the rhythm tracks were not prominent in the mono guide track, it was not uncommon to have the A reel location slip a sprocket or two when cutting between different takes.

"The little differences were additive so, even though it might be a quarter-frame, some of the songs had 30, 40, 50 edits,"

Larry Blake is **RE/P**'s film sound consulting editor and is a regular contributor.

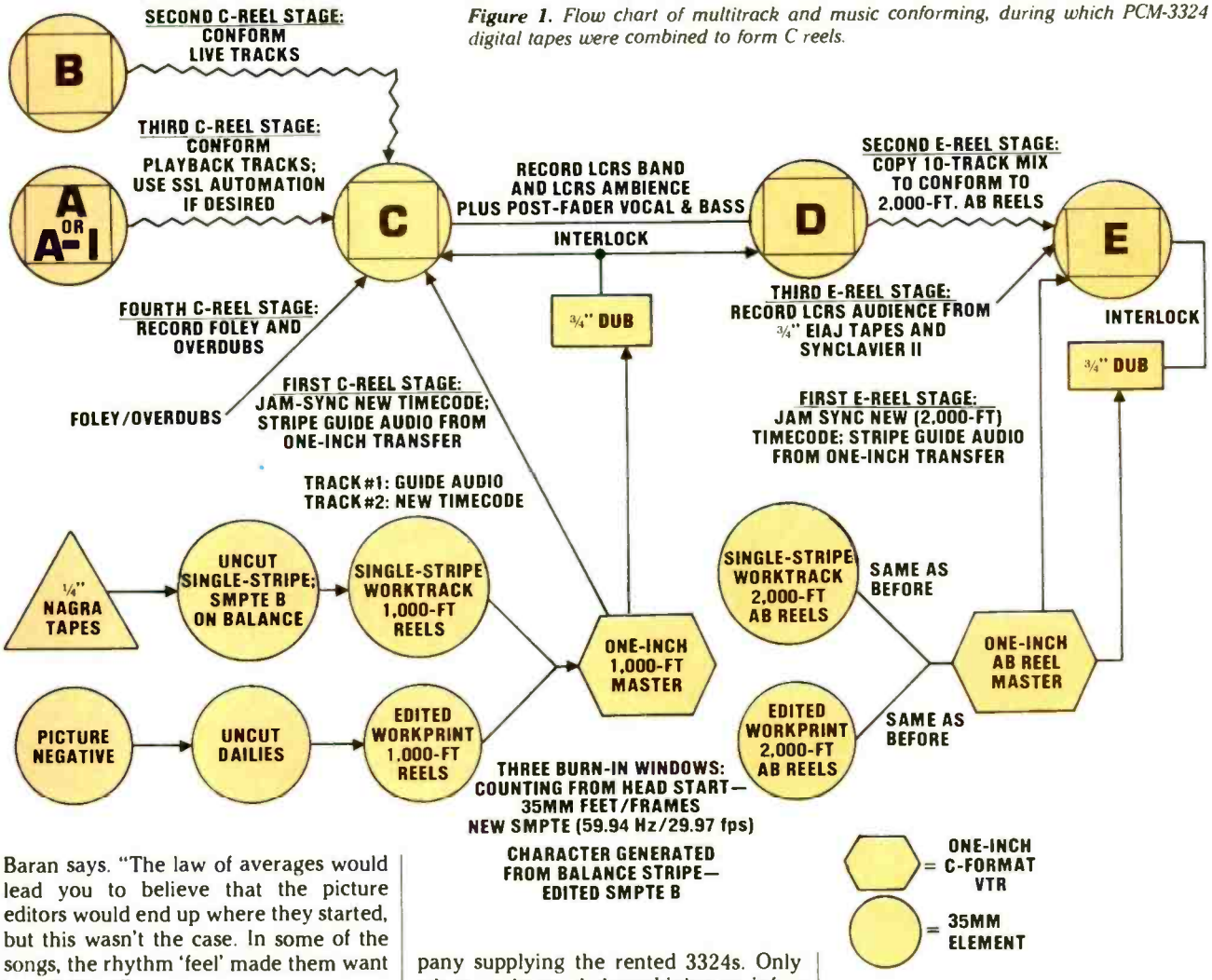

Figure 1. Flow chart of multitrack and music conforming, during which PCM-3324 digital tapes were combined to form C reels.

Baran says. "The law of averages would lead you to believe that the picture editors would end up where they started, but this wasn't the case. In some of the songs, the rhythm 'feel' made them want to pull it up."

One solution to this problem would be for the film editor to work with two sound heads, the second one reading an uncut, playback-only track. By recording on the Nagra production tape both a mono mix of the live track, along with a mono mix of the rhythm track as played back on the set, sync between the cut track and the playback multitrack could be confirmed easily by comparing the rhythm track on the edited production track with the continuous rhythm track replayed on the second sound head.

Baran notes that the problem would be worse, of course, when cutting between different takes of a totally live show (with no common playback), where "every time you try to intercut, it's like a crap shoot . . . at best."

Most of the transfers between the A and B reels to the C reel were done in the digital domain using a digital patchbay built by Digital Services, the company supplying the rented 3324s. Only when tracks needed combining or, infrequently, when a level change was needed to smooth an edit, would the signal pass through the Solid State Logic SL6040 E series console in Sync Sound's Studio B.

The SSL Primary Studio Computer automation system was helpful in recreating the mixes that had been made to the A and A-1 reels while creating the playback rhythm tapes. The digital patchbay not only allowed for easy routing of signals between machines, but also provided the production team with the ability to transfer selected tracks.

"Usually when you go D-to-D, you have to transfer all 24 tracks at the same time, as in making a clone copy," says Hahn. "Also, you must transfer track 1 to track 1, track 2 to track 2, and so on. If you want to transfer track 1 to track 4, you must go D-to-A and A-to-D."

Despite the presence of a frame-accurate EDL, Hahn notes that "not all instruments would cut properly at a chosen edit point. In post, we had to deal with all the problems [the film editors] couldn't hear [on the mono guide track] in the cutting room."

Bill Marino says that "normally you might cut on the downbeat, and use the masking effect of the incoming sound to smooth over any changes in the audio. But, in fact, if you listen to the different instruments, you'll find that the saxophone was in the middle of one line, and [on the cut] he changed what he played. If you had cut the saxophone after he had finished his phrase, it would be much more musical.

"Also, the vocal often goes over the downbeat; in this case we would cut right after the vocal sustain.

"So we end up with a group of tracks staggered around the edit point, producing a seamless and undetectable edit."

In addition to these "split" edits. Ungar,

Hahn and Baran had three other tools at their disposal for fine-tuning sync on the C reel. First, the time code editor at Sync Sound allowed them to easily slip sync in subframes, with 0.01-frame resolution.

"Usually with video or time code editing, you're stuck on the frame edge, and that's it," Hahn notes. "The equivalent of a (video) frame at 15 ips is ½-inch; that's a lot of slop, and most ¼-inch editors can do that. We take that ½-inch and divide it into a hundred parts, which leaves little question as to where you're going to do your edit."

Second, the PCM-3324 allows for variable crossfades during edits, adjustable in length from a 1.45ms to 375ms (at a 44.1kHz sampling frequency). Hahn says that the digital crossfade "is the rough equivalent of bias-ramp time [in analog recording], so you can do anything from almost a hard cut to a slow crossfade."

Third, the team could rehearse and preview the edit, including crossfades. Hahn notes that "rehearse sounded exactly like record, and vice versa; I'm spoiled by it, and I think Roma and Leanne were, too. No longer was it just punching in on the downbeat: We'd say, 'Let's try that a frame earlier, or a frame later. Now let's change the ramp time and make it longer.'

"So we might end up doing a short ramp time at 50 subframes. We would store those parameters and then try it the same way, but a measure earlier. Because we played with it until we got it absolutely right, when Leanne went to mix it, it was more like a [continuous] live performance."

Baran says that Hahn "is a whiz at both the technical operation of [the Sync Sound time code editing] system, and the talent involved at guessing where the edit point is going to be. We weren't going back from 50 [subframes] to 40 then 30, 10 and back the other way.

"Often we didn't have to fix something; it was just a matter of professional pride knowing that we were accurate to within 10 subframes all the way through the picture. If we were 10 subframes off, we'd fix it."

She also had high marks for the speed of their synchronization system: "The lock-up time is quick, about two seconds, and I think people neglect to calculate how high a percentage of your time is wasted if the lock is slow."

Hahn recalls that they "probably spent a week and a half cutting exactly the way the guide track told us to. Then

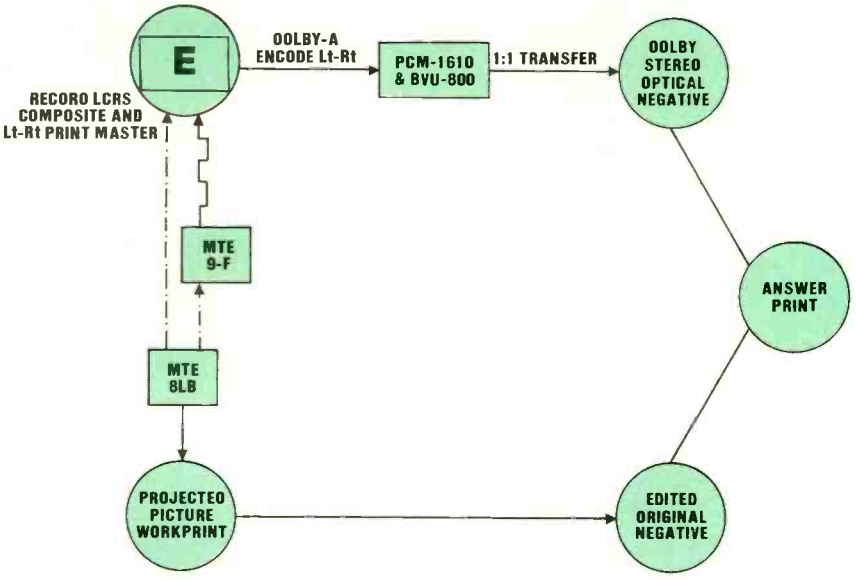

Figure 2. *Dolby Stereo Lt-Rt transfer stage from PCM-3324 tapes to PCM-1610 format, prior to shooting the optical negative.*

Laurie came in and we rebuilt songs for better performances. She had made notes during editing as to what solos she wanted to replace with a better take."

Synchronization between the C reel and the picture was verified in two ways. First, during the conforming, the original mono guide track, which was on track 24 of the C reel, was placed in one speaker; in the other speaker was a mix of the edited tracks recorded on the C reel. If the new and old versions phased against each other, sync was assured—a time-honored procedure used in films to match reprinted dialogue tracks to the editor's worktrack.

Ungar, who came to the project from the record world, recalls that she was "amazed that this was the way it was done. 'You actually sit and listen for it to phase and there's *no* other way to tell?' But that is really the bottom line; with all of the high-tech equipment, we're still listening for it to phase!"

After editing of the A and B reels had been completed, a quick mix of the C reel was transferred to 35mm mag, to allow Lisa Day to confirm sync on her editing bench against the picture workprint.

Most of the 24 tracks on the C reel—and, indeed, what we hear in the final mix—were recorded live at the Park Theater; the playback tracks from the A and A-1 reels were used strictly as rhythm beds.

Some overdubs were also done at Sync Sound. "During the concerts, Laurie would be triggering the Synclavier from different violins and keyboards," Ungar recalls, "which sometimes mistriggered because the interface was a little clumsy. Not wanting to stop during production to figure it out, we just shot with mistriggers knowing we'd fix it later."

To save tracks during the shoot, Ungar occasionally printed a mono LinnDrum track on one of the 3324's analog channels. During overdubbing, the five Linn tracks were retriggered from the Simmons drums recorded on a digital channel.

Foley work is, to put it mildly, rare in concert films, but that didn't stop the crew of *Home of the Brave*. The song "Talk Normal" had so much Foley that it was premixed onto two tracks of a PCM-1610 digital processor, which ran in interlock during the final mix.

Dolby Stereo mix

The chief complaint that re-recording mixers have had in the past with the Dolby Stereo format is that the use of its 4-2-4 encode/decode matrix precludes their placing sounds in the LCRS soundfield with any assurance. Prior to the advent of Dolby Stereo 2-track optical, film mixers had used the simple 4-track mag system: four tracks on the print for left, center, right and surround channels, no matrix.

Rock mixers new to the film re-recording process not only have to con-

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Photo by Sandra Ray

View of the video production screen at Sync Sound's Studio B. Note the dual video monitors displaying time code status information from the facility's custom time code editing system, and the SSL Primary Studio Computer readout.

Photo by Sandra Ray

Sync Sound's Ken Hahn (right) and Leanne Unger operating the facility's Solid State Logic SL6000E console.

tend with the Dolby matrix and its peculiar constraints but also, as Ungar says, to mixing with four channels.

"I'm used to mixing with two monitors, and having a phantom center from left plus right. When you put that center speaker into the room, especially with the surround channel, to hear the space made up of this strange phasing was quite bizarre to me. I kept thinking something was *wrong*.

"Then I got very used to it and had an equally hard time when we went to Soundworks [New York] to do the record mix. Trying to relate to the phantom center was bizarre, now that I was used to listening to all this material on four channels."

During the assembly of the C reels, Ungar, Hahn and Baran primarily monitored in the standard two-speaker mode. Once final mixing began, however, they used 4-track LCRS monitoring exclusively through a Dolby DS-4 unit that simulates the effect of the 4-2-4 matrix used in the stereo optical process.

Home of the Brave was Ungar's first contact with the Dolby Stereo cinema matrix, and she had received warnings from people that "when you turn it on, you'll think it's broken, but it's not," referring to the stereo width restrictions imposed by the matrix process.

"I found that often I would put a lot of high and low end to kind of compensate for the [ISO film monitoring] curve. When I first heard it, I didn't think I was ever going to like what I mixed on it. But that feeling passed."

The film's final music mix entailed Ungar's mixing two 4-track "quartets" containing LCRS mixes of dry music and theater ambience from the edited C reels onto the D reels. Because of the SSL Studio Computer at Sync Sound, most of Ungar's time was spent playing back from the C reel and writing automation data. Only when the mix was finished and ready to be printed did they require the second digital multitrack to print the mix on the D reel.

Although the LCRS music mix contained all instruments and vocals, Anderson's vocal and the bass usually were also recorded (post fader, with all processing) on separate channels, a procedure that allowed levels of these two tracks to be raised in the final print mastering.

On the B live tapes, Ungar had reserved five tracks for the important ambience mics: a Neumann SM-69 stereo tube, two Schoeps CMC-4s, one on each side of the hall, and a Neumann KMR-82 shotgun. The ambience 4-track mix was composed primarily of the Neumann stereo mic and the outputs from assorted digital reverbs used to "roomize" the instruments. Effects like discrete slaps were recorded on the dry band mix.

Ungar "assigned the ambience mic that I recorded in the theater to the ambience LCRS, in addition to the echo sends from the digital reverb devices. We really got lucky with the Park Theater; it just sounded beautiful. I tried to use the Neumann stereo tube mic as much as I could for our ambient sound."

In preparation to adding room tone to the dry instruments on the ambience LCRS, Ungar and Baran tested many available digital reverb devices.

"The one that sounded smoothest and richest to me turned out to be the EMT-250, which really surprised me. I didn't want to use it—partly because it's so silly looking—but it sounds very, very nice and it's also quiet.

"We ended up using the 250 to kind of 'stretch' our Park Theater sound, and make it a little bigger. Also, because they were cutting takes together and occasionally cheating shots, I wasn't always able to use that ambient mic and sometimes had to dump it. When I did dump it, I had to replace it with reverb."

Ungar also found the room programs in the Lexicon 224X digital reverb helpful in such "reverb-replacing" situations. Because the EMT's pre-reverb delay was not long enough, when she wanted "to hear things hit the back wall and come back again," Ungar sometimes used a Lexicon PCM-41 to delay the signal from 120ms to 250ms. She fed this delayed signal to a Lexicon PCM-60 for reverb processing and mixed them back with the EMT-250.

The Dolby Stereo matrix can be very sensitive to the presence of signal processors, and Ungar found that "to make

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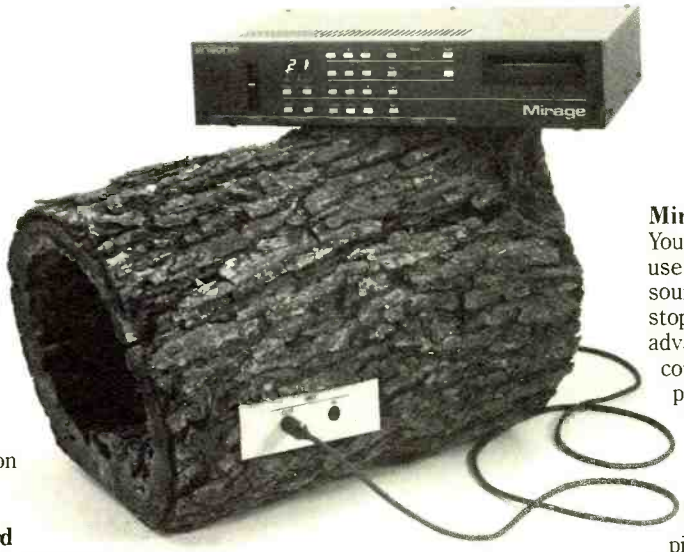

If you're a keyboard player, don't ever let a drummer borrow your Mirage . . . you might never get it back. If you're a drummer, ask a keyboard playing friend to lend you his Mirage . . . "for a while." In either case, if you're into percussion, there's a score of good reasons to get your hands on a Mirage.

The buzz on the buzzword

"Sampled Percussion" is a pretty catchy buzzword. Some high-end electronic percussion systems offer sampling as a creative option. Others offer a selection of sampled sounds on ROM's that plug into the system.

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MIDI makes the magic

Now that we've gotten all these great percussion sounds into a Mirage, how do we get them out? Naturally, all the sounds can be played in real time from the keyboard. Since the Mirage can hold up to 16 samples, you can play a full drum set or complete percussion at any one time.

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Most drummers will argue that playing percussion is no fun unless you get to hit something. We agree. MIDI features that can put you in touch with a Mirage are showing up on electronic drum kits. And our friends at Roland have come up with a MIDI percussion item that's simple and inexpensive — the Octapad*.

As the name implies, the Octapad gives you 8 pads to hit and each pad can be assigned a MIDI channel and MIDI note number. Add a Mirage, a MIDI cable and a pair of drum sticks and you've got a potent percussion instrument.

Let's start by creating an electronic drum kit. Connect the MIDI out of the Octapad to the MIDI in of the Mirage and load Sound 2 (Electronic Drums) from Sound Disk 4. The Mirage now has 12 distinct electronic percussion sounds ready to go, including full octaves of toms, ride cymbals and flanged crash cymbals.

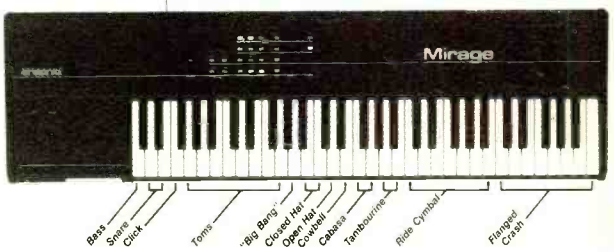
Set the Octapad and Mirage to the same MIDI channel and choose any 8 sounds by entering the MIDI note number into the Octapad for each sound. The keyboard map shown here will give you a guide. You've now got an 8-piece electronic drum kit that's ready to record, sequence or play live.

Mirage means melodic percussion

You can follow the above procedure to use any of the Mirage percussion sounds with the Octapad — but why stop there. One of the strongest advantages of the Mirage/Octapad combo is the ability to play percussion using any sound in the Ensoniq Library.

By selecting MIDI note numbers you can easily program scales into the Octapad to use with Mirage steel drum, marimba, hammered piano, bass, harp, vibes or whatever. Because the Octapads are velocity sensitive, you'll be able to control the dynamics just as you would any acoustic percussion instrument.

If you want to get a bit more outside, try playing brass and orchestra hits, guitar power chords, sound effects and pipe organ through the Octapads — there are over 300 sounds in the Ensoniq Library just waiting to get pounded.



All this and a keyboard, too

As you can see, the Mirage isn't just a great keyboard, but a versatile drum machine, too. With some additional MIDI gear, it can be downright amazing. Visit your authorized Ensoniq dealer for a full demonstration. There's no telling where a Mirage and your imagination can take you.

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Laurie Anderson checking camera angles. Photo by Les Fincher

things read properly in the surrounds [through the 4-2-4 matrix], I couldn't bus very much to the front [LCR channels] at all. It would almost always drift back to the front again from the surrounds.

"Sometimes when I had the original ambience track in the rear, and the original [dry instrument] signal in the front, I would get a pumping noise as the original close miced sound seemed to 'suck' the ambience from the rear to the front speakers. So I had to keep my ambience below a certain threshold in terms of panning it to the rear."

Baran says that "very little discrete information was bused to the surround channel, because we found that it was best used to enhance live ambience."

During this time, an Lt-Rt Dolby Stereo mix was printed on two tracks of the D reel as a reference mix. The Lt-Rt was transferred to mag to allow the crew not only to check the mix in theaters and screening rooms, but also to make a test optical transfer. (The final Lt-Rt negative was transferred digitally.)

Sweetening audience tracks

Once the basic 10-track music mix was locked, the 10-minute D reels were transferred to E reels, joining two editing reels into one projection AB reel. Thus, editing reels 1 and 2 were combined into projection reel 1AB, reels 3 and 4 became reel 2AB, etc. Picture editor Lisa Day combined the workprint reels, and the film was again transferred to 1-inch videotape, this time in 2,000-foot reels, at Tapehouse Editorial, New York. This reel-combining stage usually occurs after all post-production sound has been completed, but before release printing, where films are in 20-minute reels for shipping to theaters. (Few theaters today actually project these reels, however; usually, they are joined together onto a large "platter" for continuous viewing on one projector.)

The main reason that most films stay in their original 10-minute editing reels to the end is that directors and editors, as a matter of course, "lock" the reels (i.e., make no further changes in the picture edit) at the last possible moment before the laboratory's "Give it to us or you won't make your release date" cutoff. Another problem is that 35mm mag machines often resent having to shuttle 2,000 feet of mag back and forth at high speed.

The sound crew of *Home of the Brave* was unanimous in their praise of Laurie Anderson and Lisa Day committing to the film's final length of 90 minutes, 52 seconds and 13 frames before the assembly of the C reels.

"We envisioned having to reconform [the edited 24-track tapes] a couple of times, but ended up not having to do any of that," Ungar says.

Having the film in 2,000-foot reels not only allowed Ungar to adjust the dynamics of the mix in five 20-minute segments (instead of 10 10-minute reels) but, on a more practical level, she only had to worry about matching level and EQ for four reel changeovers instead of nine. To facilitate this process, reels 1AB, 3AB and 5AB were mounted on a single 14-inch 3324 reel, and reels 2AB and 4AB on another.

A third LCRS "quartet," composed of audience reaction and crowd noises, was recorded on the E reel. A primary purpose of this sweetening, as is the case with even the lowliest game show, was to smooth out the difference in audience levels between cuts. (In addition, the audience sounds helped bridge what had previously been reel changes between odd- and even-number editing reels.) The fact that some camera setups were shot without the audience present posed an even greater sound editing problem.

In anticipation of this situation, Ungar ran a "wild" Nakamichi DMP-100 EIAJ-format digital processor to record audience sounds with a split from the Schoeps mics that were also recorded on the multitrack. During the picture editing stage, Baran and Ungar waded through all of these recordings and edited a "greatest hits" collection onto a time-coded ¾-inch digital tape for use in the sweetening session.

Sometimes the take used in the film would be No. 2 or higher, and the audience would no longer be laughing as enthusiastically at a joke as they did the first time. To cover these situations, Ungar and Baran went on a "yuck" field trip to a Broadway comedy with backpacks containing a DMP-100, VCR and mics.

For easy retrieval, some of this material, along with miscellaneous crowd sounds, was later sampled into a New England Digital Synclavier II synthesizer, turning it into a high-tech "laff" machine operated by Ken Hahn. (The real article, which looks more like an ancient Telex terminal than a computer, uses cassette loops controlled by a keyboard and pedals.) In this manner, different keys on the Synclavier could be used to replay room tone, laughs, coughs and such. Applause, whose greater length taxed the Synclavier's memory limitations, was generally rolled in from the EIAJ U-matic digital tapes.

With the final 14-track mix (a trio of four-track LCRS groups, plus bass and lead vocal) locked and agreed upon, the next step was to create the 2-track Lt-Rt Dolby Stereo printing master. It was decided to do this critical fold-down at

Continued on page 74

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Continued from page 70

Sound One Studio C, New York, primarily, according to Ungar, "for a change of perspective. We wanted to make sure that we were in reality; that the bass drifting back from 20 to 30 feet away still sounded tight. We were very close at Sync Sound."

The re-recording stage at Sound One also provided the benefit of standard film monitor loudspeakers (JBL model 4675A), and large, sharp film projection. The room is equipped with a 24-input Neve 5100 series console.

Also, at Sound One the crew could take advantage of re-recording mixer Tom Fleischman's experience with the requirements and headroom restrictions of the Dolby Stereo optical medium. As noted, the sound team had been monitoring through the matrix exclusively dur-

Digital Facility Spotlight: Sync Sound, New York

While building their "dream" post-production facility in August 1984, Ken Hahn and Bill Marino incorporated ideas gleaned from seven years of working together at Regent Sound, New York—Hahn as studio manager and Marino as chief engineer, although both also served as mixers.

One obvious difference between Sync Sound and most other recording/post-production studios is that all audio and videotape machines are located in a central machine room, a layout common in old film studios and new video facilities.

Audio and video transports include Sony BVH-2000 type C 1-inch VTRs; four JVC 8250 ¾-inch VCRs; five Otari MTR-12s; a 4-track Otari MTR-20 and mono (4.2) and stereo (IV-S and IV-S TC) Nagras; and a 4-track Multi-Track Magnetics 35mm recorder. The video decks become digital recorders with the addition of Sony PCM-1610 and 701 digital processors.

The four 24-track machines—three Otari MTR-90 series II analog (with interchangeable 16-track/2-inch and 8-track/1-inch head stacks) and one Sony PCM-3324 digital—connect via 32-channel DL-style connectors to either the Solid State Logic SL 6040 E-series console located in Studio B, or the Soundcraft 2400 in Studio C (there is no Studio A yet).

A welcome side benefit of segregating audio, video and mag machines from people and consoles is the resulting lack of transport noise in the control rooms.

Marino says that "it makes for a much more relaxed environment without all those things chunking away. This is especially true with synchronizers and machines slewing and cuing. The scanners in video decks make a good amount of noise. Digital machines themselves operate very hot, because of the logic chips, and have these big axial blowers that pump a lot of heat into the room. We don't have to make the control rooms 40° to keep the machines happy; it's set at what temperature the people are happy at."

Although Studio B is the size of an average large audio post room (23'x25'x14') the staff at Sync Sound had to modify the room to accommodate Home of the Brave, *its first Dolby Stereo theatrical dub.*

First, Marino and Hahn bought a Panasonic PT101 video projector, "which made us want to make the sound as wide as the picture," according to Marino. "Normally for television shows we monitor on a 25-inch Sony. We were worried that while looking at that small picture we would be more satisfied with a narrower [sound] image. Because the Dolby matrix narrows the image, we wanted to have to fight [the effects of] the matrix."

The speaker system basically remained the same with the addition of another UREI 813B Time Align for the center channel and four JBL model 8325As for the surround channel. White third-octave equalizers were used to match the room acoustics to the ISO X monitoring curve used during Dolby Stereo film dubs. Although the UREI 813 might be a common sight in recording studios, almost all films today are monitored using either a JBL model 4675A or an Altec-Lansing A-4 speaker.

The model 813 "is a speaker that we're all familiar with," says Marino. "Leanne is certainly familiar with them. Our feelings were that if we can come up with a mix that will translate to a 5-inch television speaker—which, in a sense, has some of the same problems of movie theaters in terms of limited dynamic range and high end, and an even more limited low end—then we could certainly make a mix on an 813 that would sound good in a theater."

Marino warns against only monitoring the front portion of the sound field via the three, behind-the-screen speakers.

"People will say, 'What do you care about the surrounds? You don't use that much of it.' Well, if you don't want a lot of surround in your films, and are not so concerned about it, you want to know what the Dolby [matrix] is putting there even if you don't send anything there. People use Dolby [CDS-4 monitoring units] on their dubbing stages, but don't necessarily use it for their music. If music is a big part of your film, the engineer and producer should *mix it on a surround monitoring system with the matrix. Don't leave it up to someone else to balance it for a theater."*

He also emphasizes the process of checking mixes in more than one theater. "Don't trust any one theater or monitoring environment as being gospel. A screening is like mastering: If it is going to affect the way you mix, then you have to make sure that everything is properly set up at that screening. You have to have all the right tones on your playback tape, and make sure that you book enough time to align the playback dubbers."

Machine-to-machine interlock at Sync Sound uses a custom controller that oversees all aspects of the tape machines, and uses a control keyboard as an "intelligent terminal." A video monitor displays all real-time operation, including time code locations and status, while a TeleVideo model 950 ASCII terminal is used for inputting of text and setting up different parameters. Identical units are provided on the SSL console in Studio B and Studio C's Soundcraft board.

Every tape machine in the facility has assigned to it a custom controller interface that incorporates 80K of RAM and two microprocessors along with two serial ports. One of the serial ports is connected to an Adams-Smith series 2600 frame that actually handles synchronization of the tape machines. The system uses a proprietary serial bus that Marino says is faster than the IEC RS-422 standard.

The other serial port on most machines is used only for direct interface with transports capable of serial control or machines with built-in synchronizers. A parallel port goes directly to the tape machines, along with the Adams-Smith synchronizer commands that have been conditioned by the edit system. Wide-band time code readers are provided on every tape machine; in the absence of code, the system uses tach-pulse updating.

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ing the final mix. In the monitor chain, the three quartets were fed into a DS-4 as a single LCRS group, which was folded into two tracks by the encoding matrix. At this point, the 4-channel monitoring output was derived by using a Cat. 150 card found in Dolby CP-55 and CP-200 cinema processors sold to theaters.

Fleischman says that the care taken during mixing was evident: "I really had very little to do in terms of actual mixing," he recalls. "They must have had their monitors set fairly well [at Sync Sound] because, except for a little rebalancing of the surround channel, everything was pretty close."

He also had high regard for the PCM-3324's sound quality, and looks forward to using a digital multitrack to help reduce the generation loss inherent in standard film dubbing. However, the engineer did have one problem with the machine: He couldn't hear the digital tracks while rewinding at sync speed, an old film trick used to save time in adjusting EQ and level settings.

The first step at Sound One was to bounce up a 4-track composite mix from the three quartets, incorporating light peak limiting with four Aphex CX-1 modules. As a result, the Lt-Rt was made almost as a controlled transfer. Regarding the use of limiting, Bill Marino notes that "with digital, the dynamic range is incredible. Even though you don't hear, or necessarily see, some of the peaks go by, they're there. Analog would have squashed some high-frequency peaks that were causing edginess on the high end."

After four days at Sound One, it was back to Sync Sound to transfer the Lt-Rt recorded on the 3324 E reel to a Sony PCM-1610 and U-matic VCR, in preparation for optical negative transfer at PBRS Recording Services, Burbank, CA, as shown in Figure 2 (page 66). Because a 44.1kHz sampling frequency had been used on the Sony multitracks (instead of 48kHz), this transfer could be made without a sampling converter. [The 1610 samples at only 44.1kHz—*Editor.*]

However, the transfer to 1610 was not D-to-D, since it was decided to Dolby-A encode the Lt-Rt mix to permit "shooting" the negative straight across, without having to Dolby-encode the Lt-Rt each time a negative is made. (Dolby-encoding of the optical negative is necessary, of course, because theater cinema processors contain Dolby Cat. 22 cards to noise-reduction decode the two optical tracks on the print prior to matrix decoding into the four LCRS loudspeaker channels.)

At PBRS there were problems in slaving the PCM-1610 to the optical camera's three-phase ac motor, with intermittent sync loss causing the 1610 to mute.

Bill Marino says that "the 1610/1630 doesn't readily accept a variance in the line frequency. We set up a very slow phase-lock loop that integrated the frequency change very easily, and produced the 44.1kHz that is proportional to the line frequency."

At the time of writing, the *Home of the Brave* sound crew and the film's

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distributor, Cinecom, were looking into yet another potentially difficult synchronization problem: interlocking an EIAJ-encoded videocassette to a projector for digital presentation of the film. (This is the same procedure Walt Disney Productions followed in 1985 for digital interlock screenings of *Fantasia,* as described in "Digital Sound for Motion Pictures," in the February 1986 **RE/P.**)

Release for the stereo home video market will use, as is standard policy, the same Dolby Stereo Lt-Rt tracks. At first, consideration was given to a separate mix for home video releases, an idea that was scrapped because, according to Baran, the gain wasn't equal to the extra cost and trouble.

In preparing for the duplication of videocassettes in interlock with a digital audio master, the Lt-Rt from the E reel will be assembled on a continuous piece of 1610-encoded, 1-inch videotape. (By the way, the packaging of the video cassettes and LaserVision discs will warn customers that this is "a film by Laurie Anderson," and not the 1949 Kirk Douglas film of the same name.)

2-track record mix

Two weeks after the film mix was finished, Anderson, Baran and Ungar went to Soundworks Digital Audio/Video Studios, New York, to make the 2-track record mix, using the edited, unmixed C reels as source material. Because they wanted to do minor overdubbing, in addition to bouncing up the final mix, the filled 24-track C reels were transferred (via a digital-to-analog-to-digital transfer) at Soundworks to a 3M DMS 32-track. Allowing two tracks for time code and control, and two for the final mix, four additional tracks were left open on the DMS for overdubs. Baran really liked the ability to bounce up the final mix onto the same multitrack reel should small changes be necessary, since they could easily punch in on the two tracks.

The floppy disks containing Solid State Logic automation data from the mixing of the D reels were brought over from Sync Sound, to be read by the 48-input SL6048 E-series console at Soundworks. Ungar says that they first used the film-mix automation data in the "mutes-only" mode. During the record mix, she "broadened the EQ so it would be more record-like, mainly picking higher frequencies on the high end. Half the record is quite different and, in those cases, I didn't get much help from the old automation data."

In preparation for disc mastering by Bob Ludwig at Masterdisk, New York, the record mix was transferred to the Mitsubishi X-80 format.

The complex editing and synchronization problems solved by the *Home of the Brave* sound crew represents one more step in proving the viability of digital recording in motion picture post-production. Although only a handful of films give digital multitracks such a workout, it is hoped the success of their efforts inspire others to push the technology further. R-E/P

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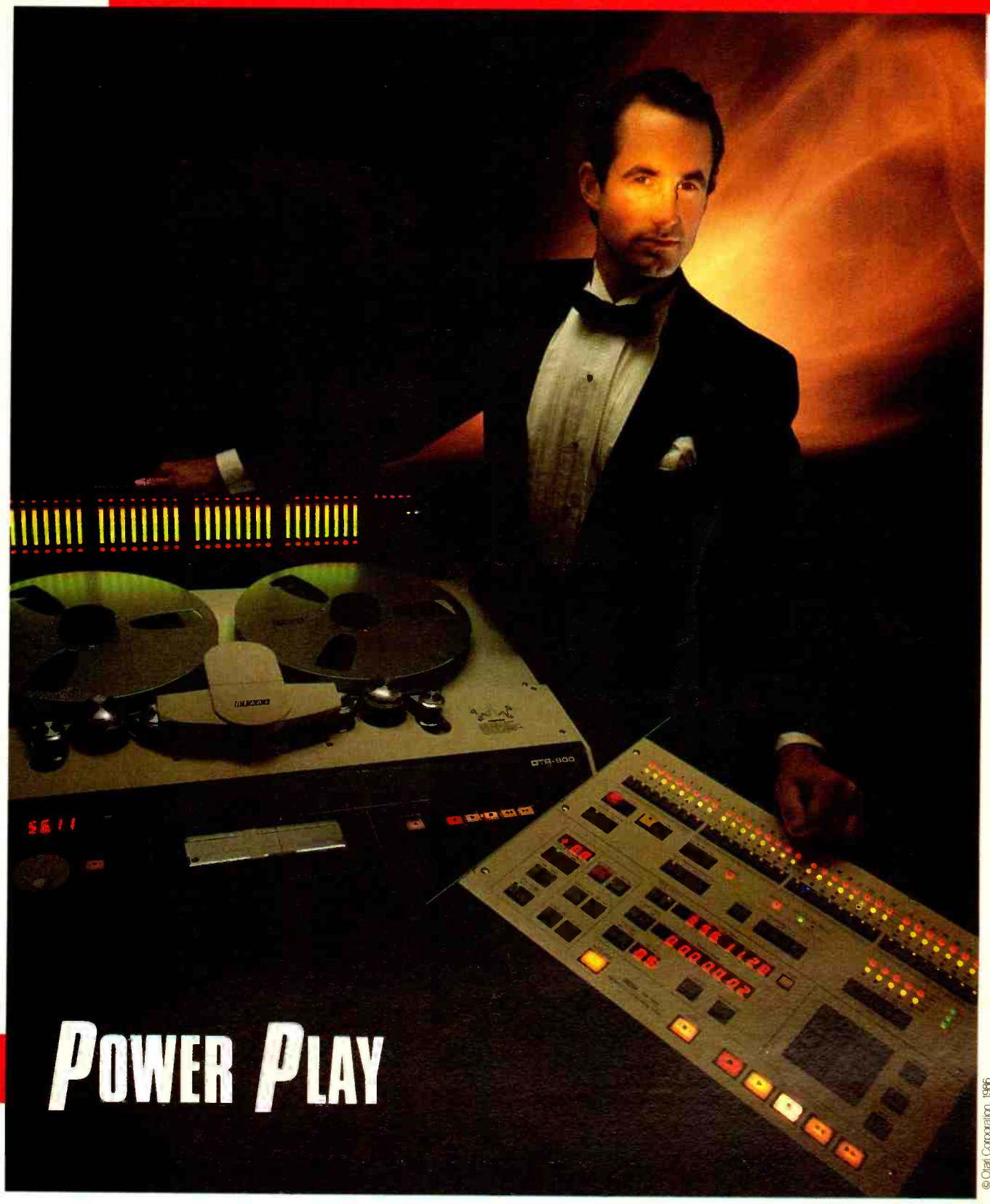

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Jimmy Buffett's "Floridays" Tour

By David Scheirman

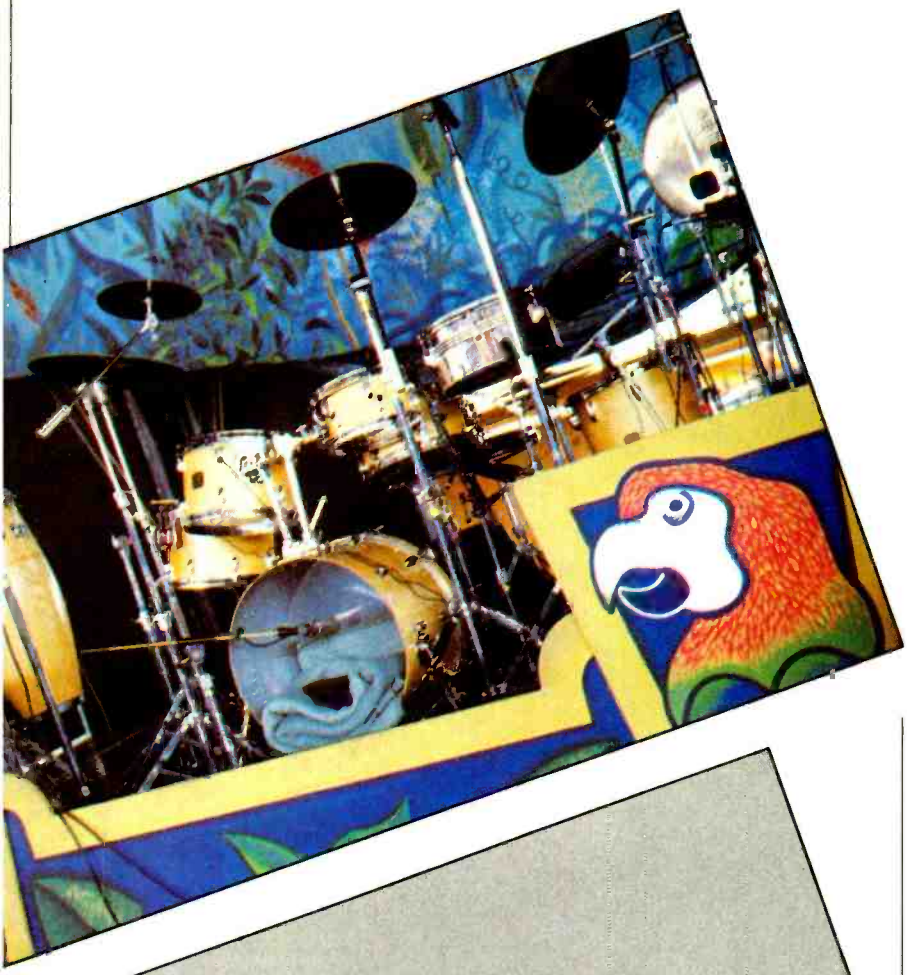
One of this summer's highest grossing tours served as a useful proving ground for new house and monitor consoles, crossovers, amplification, custom-designed loudspeaker cabinets and miniature microphones.

Jimmy Buffett's 1986 *Floridays* summer tour is proof that laid-back, beach-loving singer/songwriters don't necessarily have to receive Top 40 radio airplay to sell out concerts.

Although the artist's recent *Floridays* album has been generating airplay in many markets, the listening public primarily remembers Buffett for "Margaritaville," his party-rock anthem of nearly a decade ago. That tune, along with other Buffett classics, has inspired an upcoming feature film and produced a loyal following of concert fans who call themselves "Parrotheads."

In mid-June 1986, *Performance* magazine ranked Buffett and his Coral Reefer Band as the No. 3 top grossing box office draw in the country. Buffett's summer tour traditionally plays to favorite outdoor venues, such as Michigan's Pine Knob, Ohio's Blossom Music Festival and New Jersey's Garden State Arts Center. Audience capacities for the tour ranged from 8,000 to approximately 20,000.

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



Sound system services for the tour were supplied by Sound Image, San Marcos, CA; company owner Ross Ritto handled the stage monitor mix and Greg McVeigh signed on as an audio technician. This writer joined the crew as house sound mixer and took responsibility for the main sound system, which featured the company's new Phase-Loc loudspeaker enclosures.

Ritto and company have been handling concert sound for Buffett and the Coral Reefer Band for more than a decade.

"Early in his career," recalls Ritto (pictured left), "the owner of a world-famous PA company barged into the dressing room and offered to do sound free for six months if Jimmy would give them the account. Jimmy just smiled and said, 'No thanks.' As you can see, we're still here."

Sound Image's formula for satisfying clients like Buffett, Emmylou Harris, The Untouchables, Pacific Ampitheatre and the San Diego Symphony appears to be a balance between personal service and contemporary hardware.

"You can have the most trendy, expensive gear in the world, but if your people don't relate well to the artists it's just a pile of hardware," Ritto explains. "Tailoring the sound system and the personnel to the act is important."

Sound system design concept

When examining available hardware for use in a large new concert sound system, Sound Image was concerned with the same parameters that any touring sound company must look at: sonic quality (performance); packaging (portability); and availability (cost/delivery and development time).

Before developing the new system in April 1986, the company's rental stock included consoles from Ashly, Pulsar, Soundcraft and Yamaha. Quantities of QSC power amplifiers were available, along with a wide variety of signal processing gear from such manufacturers as Brooke-Siren, Lexicon, dbx, Yamaha and Ashly. Loudspeaker components used were from Emilar, Electro-Voice and JBL. Sound Image chose the new Fostex line of speakers.

Sound Image decided to continue refining the design of its proprietary two-box modular loudspeaker system, which is marketed under the name of Phase-Loc. QSC power amplifiers were chosen, along with new mixing consoles from Total Audio Concepts (TAC). The entire system was specified, assembled and tested in less than 30 days before the tour's production rehearsals in Memphis, TN.

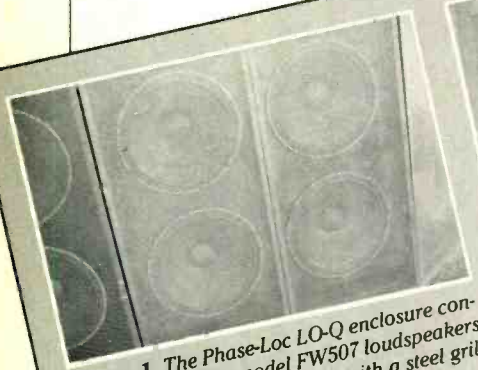


Figure 1. The Phase-Loc LO-Q enclosure contains four 18-inch model FW507 loudspeakers in a vented, trapezoidal box, with a steel grill to protect the components.

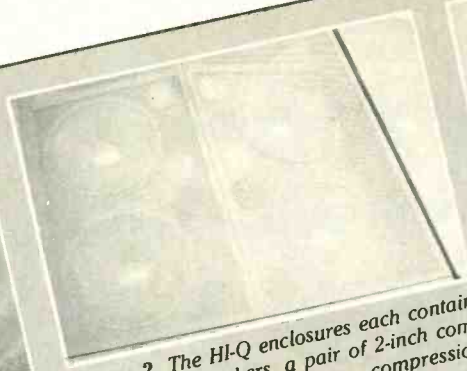


Figure 2. The HI-Q enclosures each contain four 15-inch speakers, a pair of 2-inch compression drivers and four compression tweeters. The front baffleboard is split and angled.

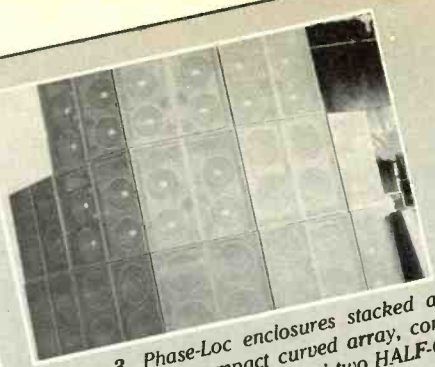


Figure 3. Phase-Loc enclosures stacked at stage level in a compact curved array, containing six LO-Q, five HI-Q and two HALF-Q.

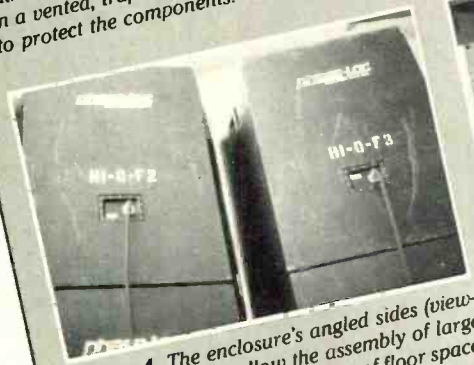


Figure 4. The enclosure's angled sides (viewed from the rear) allow the assembly of large arrays that take up a minimum of floor space.

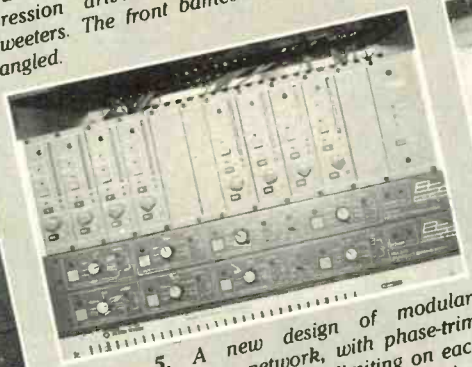


Figure 5. A new design of modular frequency-dividing network, with phase-trim feature and fast-acting FET limiting on each output, was used on the Buffett tour. A pair of standby units are shown below.

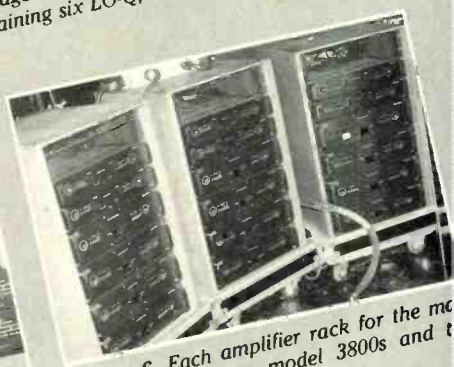


Figure 6. Each amplifier rack for the mo system houses four model 3800s and t model 3350s.

"The concert sound business goes in some pretty strange directions sometimes," Ritto says. "The choice of what hardware components to use can have long-term results for a company's business. In the case of some popular products we've seen lately, marketing seems to have gained the upper hand over engineering and quality control. I don't like to let that happen when we buy gear, because not only do I have to buy it, I have to *use* it."

Loudspeakers and amplification

Low-frequency (LO-Q) and mid/high-frequency (HI-Q) enclosures are used. Each box is constructed of ¾-inch marine plywood with rabbeted joints, and is coated with high-density polyurethane. The boxes are cut into a trapezoidal shape with the sides angled at 20°. The 48"x48"x22.5" cabinets are equipped with 16-gauge expanded steel grills. Recessed handles and castor dollies (attached with Velcro strips) improve portability and ease of handling.

The LO-Q enclosure weighs 227 pounds and contains four Fostex FW507 18-inch transducers in a vented cavity, offering approximately 8.9 cubic feet of internal volume (Figure 1). Fast-hardening foam is sprayed into the interior to dampen cabinet resonances and offer tight seam seals. The exterior of each enclosure is finished with a rugged, black textured finish to stand up to the rugged demands of daily transport.

HI-Q enclosures are separated into two ported chambers, each housing a pair of Fostex L467 15-inch transducers (Figure 2). Designed for high output bass/mid-bass applications, the L467 uses a 4-inch edgewound aluminum voice coil for high power handling. It is capable of response as low as 50Hz in an unassisted, vented enclosure. In the HI-Q box, the speaker operates in the 80Hz to 800Hz bandpass region.

Frequencies from 800Hz to 8kHz are handled in each HI-Q enclosure by a pair of Fostex D502 compression drivers mounted on Northwest 340A fiberglass horns. The combination of a 4-inch diameter edgewound aluminum-ribbon voice coil and Duralumin (eight-element alloy) diaphragm and half-roll surround is said by the manufacturer to produce up to 10dB less harmonic distortion than other 4-inch diaphragm driver designs.

Four Fostex compression tweeters capable of 80W RMS total output (160W program) are grouped on the front baffleboard of each HI-Q enclosure, which is split vertically and the two halves angled out at 10°. These splayed "half-baffles" reduce mid- and high-frequency beaming and allow the assembly of precisely directed speaker arrays with optimum coupling and smooth horizontal frequency response (Figure 3).

The angled sides of the enclosures allow the easy assembly of compact stacked or flown arrays (Figure 4). Cannon EP-8-14 connectors (8-pin, HI-Q) and EP-4-14 (4-pin, LO-Q) are installed on each enclosure to accept program input.

Loaded with Fostex components, the Phase-Loc system was crossed-over at 80Hz, 800Hz and 8kHz. Brooke-Siren FDS-340 frequency-dividing networks were provided for the system, but I elected to use this tour as an initial field-test of TAC's new TX10 modular crossover (Figure 5).

Although it requires an additional rack space compared with the pair of FDS-340s, the TX10 offers an innovative feature that will probably start appearing

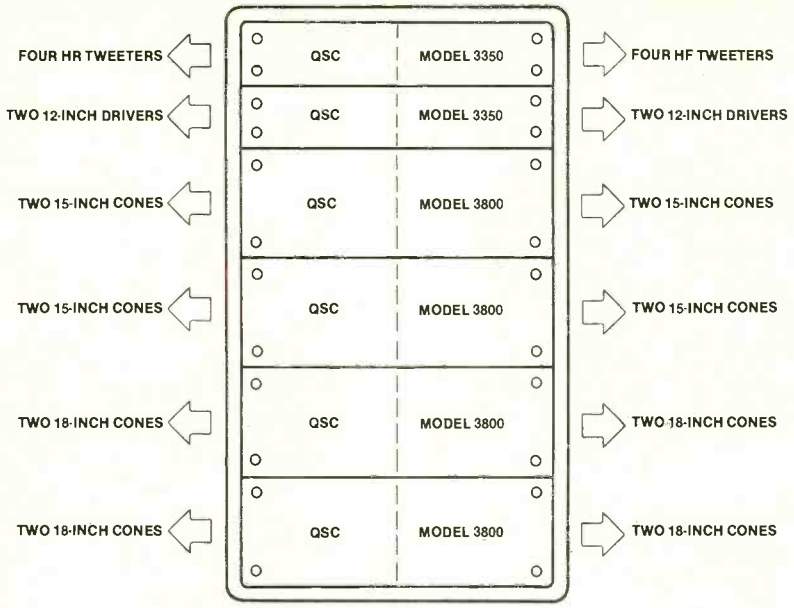


Figure 7. *Amplifier rack loudspeaker loads.*

Figure 8. *The twin-frame Scorpion house console offered 56 inputs, 4 auxiliary returns and 8 auxiliary sends. Pictured here is Don Jarvis, house soundmixer for the tour's second leg.*

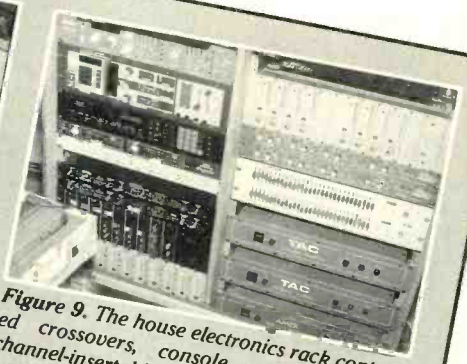


Figure 9. *The house electronics rack contained crossovers, console power supplies, channel-insertable noise gates and compressor/limiters, and effects devices. Note the exceptionally smooth midrange curve on the graphic equalizers.*

on other commercially available crossovers in the near future: phase-trim adjustments for each frequency-band output. Adjustment of the measurable (and *audible*, if your loudspeaker system needs help) phase angle of adjacent bandpass outputs is provided from zero to 180°.

Adjustment of phase angle can be made in the field with a sweep signal generator, frequency counter, some keen ears and a bit of patience. This feature is probably most useful with modular speaker systems, such as the Phase-Loc, which offer consistent arrays due to the identical packaging of various components from box to box.

The system's 24 cabinets were powered by three amplifier racks per side, each housing four QSC model 3800 amplifiers and two model 3350 units. One rack powers two pair of enclosures, for a total of eight 18-inch speakers, eight 15-inch speakers, four 2-inch compression drivers and eight compression tweeters (Figure 6).

Each channel of the model 3800s drives a pair of 18- or 15-inch loudspeakers (8Ω devices), delivering approximately 675W per channel to the 4Ω load (Figure 7).

House system electronics

TAC Scorpion mixing consoles were chosen for the Jimmy Buffett system; the manufacturer custom-modified the units to provide eight discrete auxiliary output sends. A total of 56 microphone input modules and four auxiliary return modules were fitted into a long and short frame combination. The twin-frame setup was then equipped with a multipair cable and Elco connectors (Figure 8).

TAC, a division of Amek Consoles, uses steel chassis construction, which is said to ensure absolute rigidity and minute tolerances. All resistors are of the one metal film/metal oxide type, and all ICs are mounted via sockets to enable fast field replacement.

The Scorpion's EQ section is a 4-band device, with swept-frequency midrange controls. An EQ in/out (bypass) switch is fitted, along with mic/line select, phase reverse, 48V supply and input pad switching. Eight selectable subgroups sum to a stereo main output, with an additional 8-bus output matrix available. Extended-frame, heavy-duty road cases are offered, allowing the pre-patching of console inputs and outputs if multipair disconnects are used.

The house electronics rack used with the twin-frame TAC console was a double-wide package housing crossovers, channel-insertable limiters and noise gates, and effects devices along with the console power supplies (Figure 9).

I brought a Lexicon model 200 with upgrade kit (a set of plug-in chips that dramatically improve the unit's sonic performance and offer more flexible programs) for vocal reverb. The stereo digital reverberator was used typically to add a pleasant ambience to the voices and solo instruments, with 3.5 seconds of midrange reverberation a common setting. A Yamaha REV-7 was used for drum/percussion stereo reverb.

An Orban 245F stereo synthesizer accepted acoustic guitar and steel drum inputs. When the left and right returns of the mono input were hard-panned, I achieved a sense of greater depth and spaciousness throughout the audience listening area for these two relatively challenging instruments.

In reverberant indoor environments (such as Pittsburgh's Civic Arena and some of the wooden-roofed "summer sheds"), I relied on an EXR Projector's psychoacoustic processing to increase the apparent intelligibility of the featured

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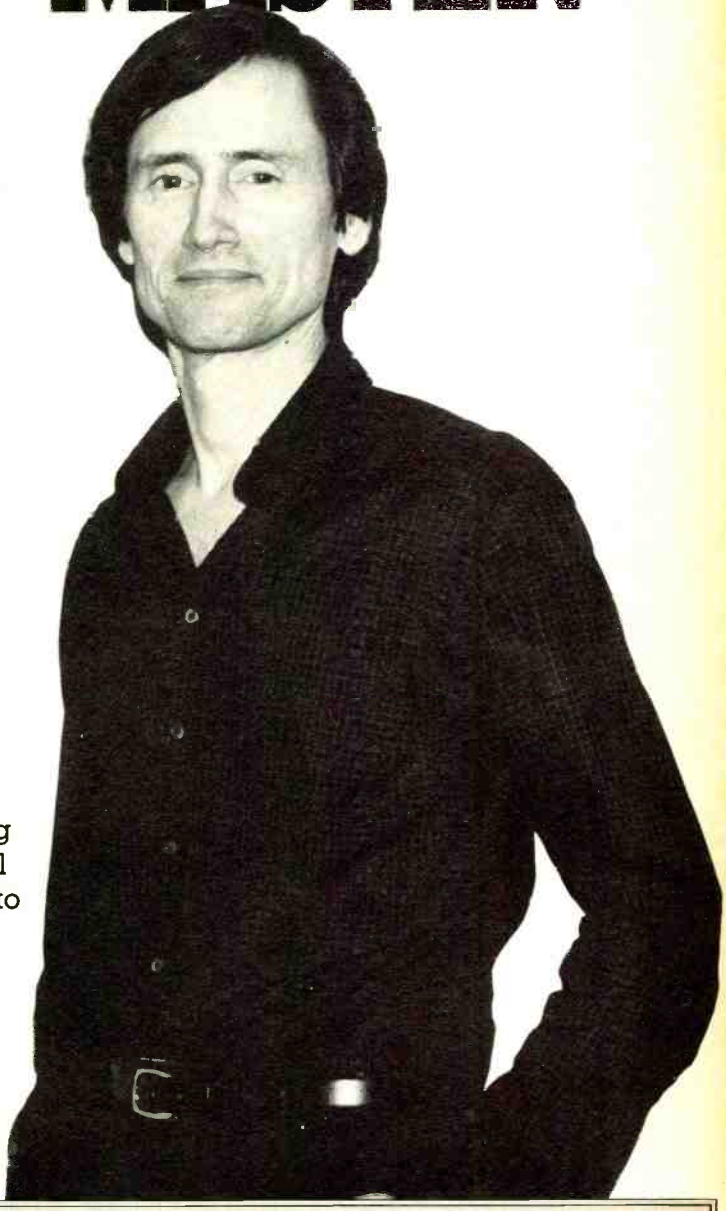

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SEND	DEVICE	FUNCTION
1 2	LEXICON 200 LEFT LEXICON 200 RIGHT	VOCAL AND SOLO INSTRUMENT REVERB
3 4	YAMAHA REV-7 LEFT YAMAHA REV-7 RIGHT	DRUM AND PERCUSSION REVERB
5 6	EXR PROJECTOR LEFT EXR PROJECTOR RIGHT	JIMMY BUFFETT PIANO AND VOCAL
7	ORBAN 2475 STEREO SYNTHESIZER	STEREO RETURN EFFECT FOR ACOUSTIC GUITAR AND STEEL DRUMS
8	LEXICON PRIME TIME DIGITAL DELAY	VOCAL AND LEAD GUITAR EFFECTS

Figure 10. *House console auxiliary send assignments.*

vocal input, without resorting to excessive high-frequency EQ boost. A Lexicon Prime Time II offered special delay effects for lead vocals and lead guitar.

Auxiliary send assignments for outbound effects are shown in Figure 10.

Channel-insertable compressor-limiting on kick drum, snare, bass guitar, lead vocal and the background vocal subgroup was achieved through the use of a dbx series 900 rack. Additional compression and limiting was available for horns, harmonica and keyboards with the Amek RMO1 rack system.

In addition, I used dbx model 166s to provide noise gating and compression for the stereo instrument and rhythm section subgroups. Signal processing interface with the main console was facilitated by the console's liberal supply of rear-panel patchpoints.

Stage monitor system

A pair of Scorpion FB (foldback) 30-by-12 monitor consoles were set up daily on stage left to distribute the 48 stage inputs to the performers' monitor mixes (Figure 11). Yamaha GQ1031 third-octave graphic equalizers were supplied to each of 16 separate stage mixes, and each mix was bi-amplified with TDM Design's new model 24CX-2 crossover, with 24dB per octave slope filters. Two bi-amped mixes required only one 24CX-2 (one unit high), and the units contributed to a clean, quiet signal path housed in a minimum of rack space.

Low-profile floor slant speaker cabinets were loaded with JBL model 2220 15-inch speakers and 2441 compression drivers on flat-front horns. Each monitor cabinet received approximately

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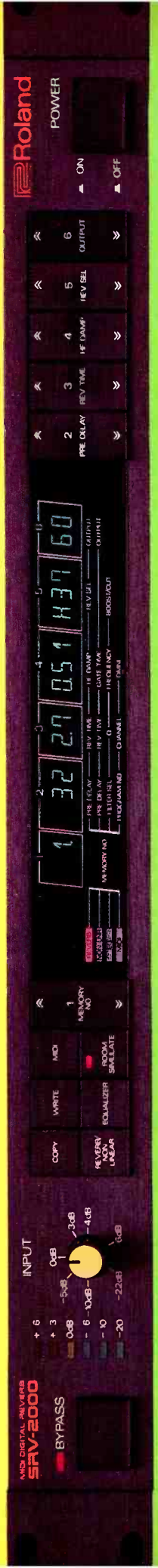

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Figure 11. Two Scorpion FB monitor mixing consoles distributed the 48 stage inputs to the performers.

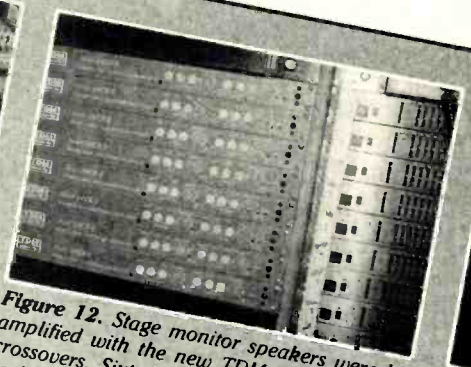

Figure 12. Stage monitor speakers were bi-amplified with the new TDM design 24CX-2 crossovers. Sixteen mixes require only eight rack spaces.

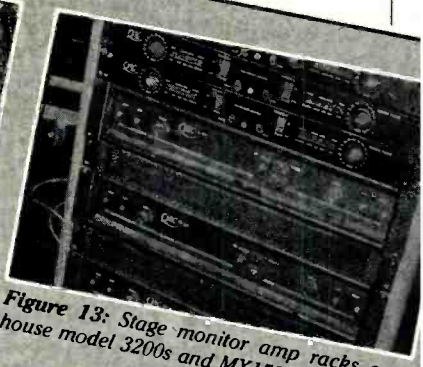

Figure 13: Stage monitor amp racks each house model 3200s and MX1500s.

Figure 14: Miniature condenser mics were used for the tom-toms; note the flexible mounting cup.

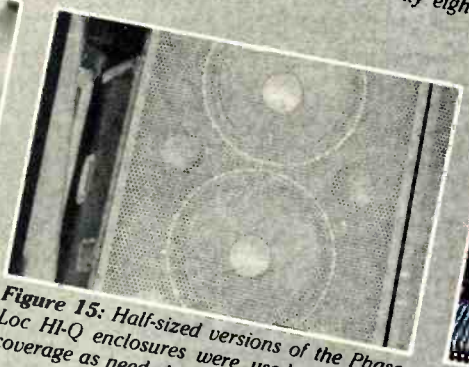

Figure 15: Half-sized versions of the Phase-Loc HI-Q enclosures were used for frontfill coverage as needed.

Figure 16. Author David Scheirman, who mixed house sound for the "Floridays" tour.

675W, with one side of a QSC MX1500 (lows) and model 3200 (highs) dedicated to each box (Figure 13).

Stage inputs

Eighteen stage inputs were dedicated to drums and percussion, while nine vocal channels were required. Congas, timbales and steel drums contributed to a rhythmic calypso feel on many of Buffett's tunes, and Simmons electronic drum pads were added to keep things in tune with 1986.

One of the challenges with large drum sets on today's concert stages is placing microphones, stands and cable. Too much clutter can be unsightly and can hamper the drummer's performance. In an attempt to remedy this and move toward a clean drum riser look, new miniature condenser microphones from Panasonic RAMSA were clipped to the tom-toms (Figure 14). Accessories for the WM-S series include tiny flexible mounting clips that allow placement at practically any angle. The tiny, rugged mics can take abuse, too; the manufacturer claims a maximum input SPL of 158dB (at 1kHz) for the WM-S5E.

Electro-Voice PL95A dynamic cardioid microphones were supplied for all vocal inputs, and a Shure SM53 picked up the harmonica amp. Guitar amplifiers received Shure SM57s, while the bass guitar was taken as a direct input. Shure SM81 condenser mics were placed on hi-hat and overhead cymbals, and a pair of SM57s picked up the snare drum's top head and bottom snares. Reversing the phase of one of the pair reduced cancellation and offered approximately 3dB more gain than when both mics were used in phase.

Five Dean Markley guitar pickups were mounted inside the steel drums, about an inch from the rim of the metal shell on the bottom. These five pickups were summed through a Rane SM26 mixer into two line-level outputs. A 5-element Helpinstill pickup was passed through an active mixer and summed to one line-level output.

System logistics

The majority of this tour's shows took place outdoors during the summer. The Southeast suffered unusual downpours, usually after setup and before showtime. Promoters were instructed to supply clear plastic sheeting and heavy tarps. The combination of heat, humidity, rain and high winds proved to be a rough break-in for the new sound system. Overnight drives of 400 to 500 miles, often four in a row, offered "road adjustment" conditions for the technical crew.

When frontfill coverage was required for tightly packed audience areas, half-sized versions of Sound Image's HI-Q enclosures were available as needed (Figure 15).

Perhaps the most difficult environmental conditions for the brand-new hardware were not created from the weather but from exceptionally lively crowds. It was not uncommon for the sound and lighting consoles to be surrounded by chanting fans forming impromptu conga lines. R-E/P

Editor's note: The mention of specific brand names of audio equipment in this article should not be taken as an endorsement. The information is supplied with reader interest and education in mind.

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Hands-on

Pearl TL-4 Condenser Microphone

By Lowell Cross

Pearl microphones have enjoyed a long and distinguished reputation in Europe, especially in the Scandinavian countries, but have until now remained relatively unknown in the United States.

The guiding force behind the Swedish company is Rune Rosander, who has been an audio designer and engineer since the Thirties. After several years of studying and modifying RCA 44 and 77 ribbon microphones, he began in 1941 to manufacture his own microphones, to fulfill the wartime needs of Swedish broadcasters.

His Pearl microphones were not ribbon designs, but condenser types, and in 1953 culminated in the innovation of the rectangular capsule. The present TL-4 retains this type of design, which is based on a preference for two, minor high-frequency resonances (determined by the respective wavelengths of the capsule's two dimensions) over a single, and possibly more pronounced, resonance at the wavelength of the diameter of a circular capsule.

Other unusual features are incorporated in this design beyond the use of a rectangular capsule. It is a double-capsule microphone, with the separation of the two elements kept to only 5mm (0.2 inches). After experimenting with different materials and metal depositing techniques, Rosander has settled on aluminized membranes.

The outputs of the two capsules, each having a cardioid pattern, are available simultaneously and independently. If both outputs are summed equally in phase to mono, an omnidirectional characteristic is the result; if a similar summation is made, but with the polarity of one capsule inverted, a mono bidirectional signal is obtained. By combining the two capsules at different levels, other polar characteristics are possible: in-phase, wide-angle cardioid; out of phase, super- and hyper-cardioid.

Lowell Cross is professor of music and director of recording studios, University of Iowa, Iowa City, IA, and a regular **RE/P** contributor.

The relative mix and polarity of the capsules, and therefore the resultant mono pattern changes, are thus made at the console. Since the two simultaneous outputs are always present separately, the TL-4 may also be used as a coincident-stereo microphone. In this configuration, the two cardioids are back-to-back, or facing 180° apart. This stereo mode of operation was chosen for our evaluation.

Control unit

The PDV-B accessory module provides switching for the various mono/stereo combinations, if the TL-4 is to be used without a console or mixer. The module offers both balanced XLR and unbalanced RCA/phono outputs for connection to professional, semi-pro or consumer-grade recorders.

The TL-4 has an unusually high output-signal level: 120mV/Pa. Used in a double-cardioid coincident stereo configuration, the Pearl can easily drive the line-level inputs of semi-pro digital audio processors, making it quite adapatable for remote digital recording. The TL-4 operates on either 24V or 48V phantom powering (or any voltage in between), according to DIN 45 596.

The microphone's high output represents one of its potential drawbacks, yet one that can be corrected with attenuator pads. Certain consoles, however, might not offer a sufficiently wide selection of microphone input sensitivities to accommodate the TL-4 when in very loud acoustical environments.

With a console of this type, pre-amp overload is very likely to result, yet the 48V phantom supply voltages are pre-

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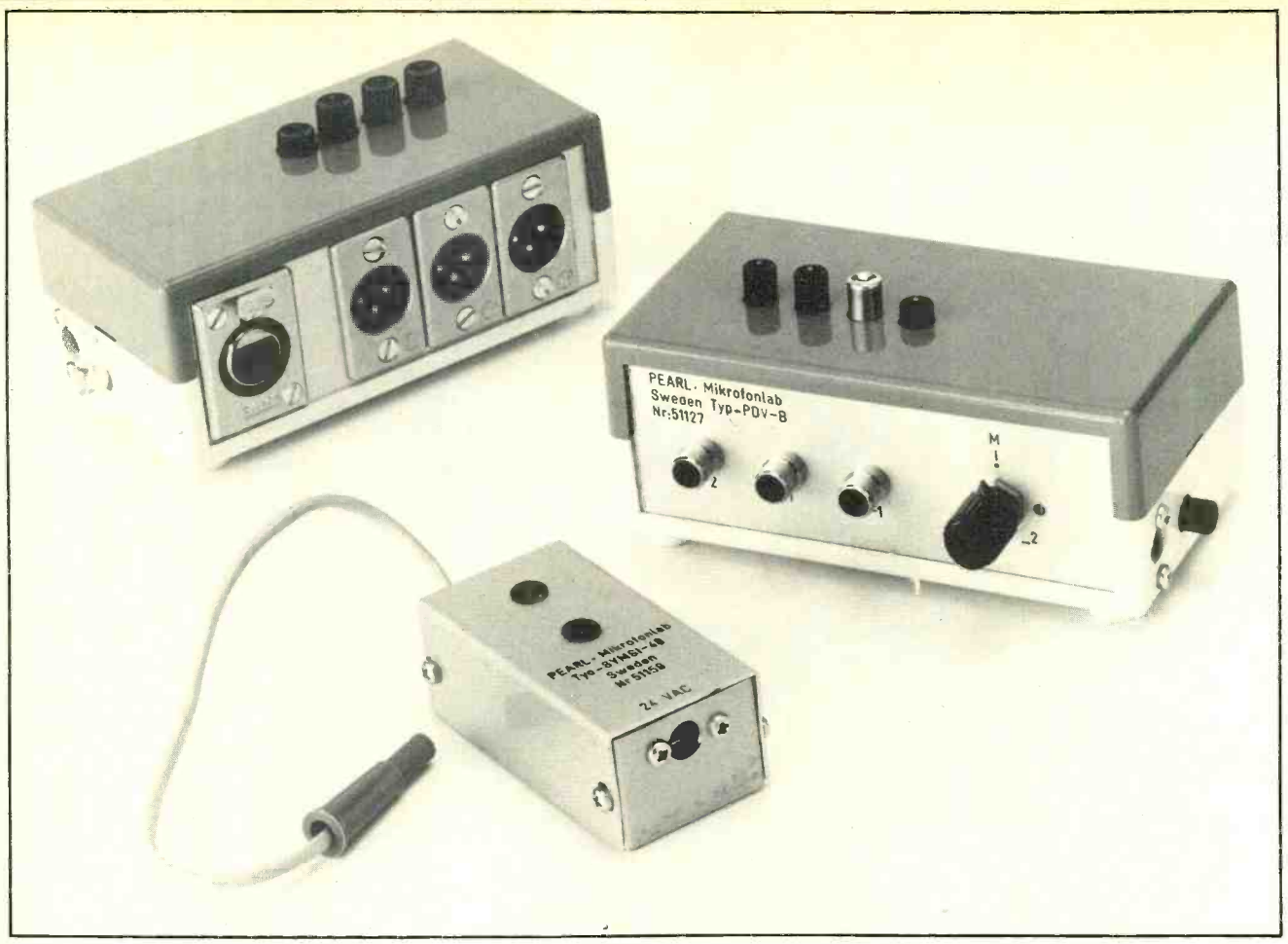

Accessories include the PDV-B switcher unit, which provides various mono/stereo output combinations (shown top left *and* right*), and the SYMSI-48 power supply module.*

sent at the high-gain microphone inputs, *not* the live-level inputs, which may be more appropriate. The manufacturer is prepared to recommend values for attenuators pads if certain consoles are expected to be used with the TL-4.

My other reservations about this microphone are minor, yet worth mentioning. As is sometimes typical of new imported equipment, the unit lacks clear, concise English instructions. The pushbuttons on the PDV-B switching module (not required with professional consoles) are not labeled in any language. Also, no stand adapters or similar accessories are supplied, only a list of recommended items from other manufacturers. I have been assured that the TL-4 will be better documented for English-speaking users once importation begins this fall.

I have no reservations, however, about the TL-4's sound properties. It satisfies all of the criteria of a first-class microphone: low distortion and noise; excellent pattern characteristics; and smooth, wideband frequency response.

Evaluation procedure

Our evaluation took place during a dress rehearsal and concert of the University Symphony Orchestra, including a piano concerto. The performances were monitored live and recorded 2-track for subsequent replay.

James Dixon conducted the orchestra concert in late February 1986, preceded by a dress rehearsal the evening before, performing Schubert's *Overture "in the Italian Style,"* Mendelssohn's *Concerto No. 1* in G minor, opus 25 (Kerry Grippe, pianist) and the Ravel orchestration of Mussorgsky's *Pictures at an Exhibition.*

The TL-4 (coincident stereo) was suspended 12 feet above the stage of the University of Iowa's Hancher Auditorium, 10 feet downstage from the conductor's podium. This configuration placed the microphone just below the line defined by the raised lid of a 9-foot Steinway model D grand piano, which was moved offstage after intermission for the performance of the Mussorgsky-Ravel work. With its back-to-back coincident cardioids, the microphone faced directly left and right.

We attempted to monitor the microphone's self-noise during the quiet period in the hall following our setup. The background ambience of NC 15 to NC 20 completely masked any self-noise from the microphones, as monitored through our Neve 5315/24 console and Klein+Hummel 092 reference monitor loudspeaker systems. As already noted, the TL-4, which was supplied with 48V phantom powering, has a very high output level; console inputs were set for only 25dB gain.

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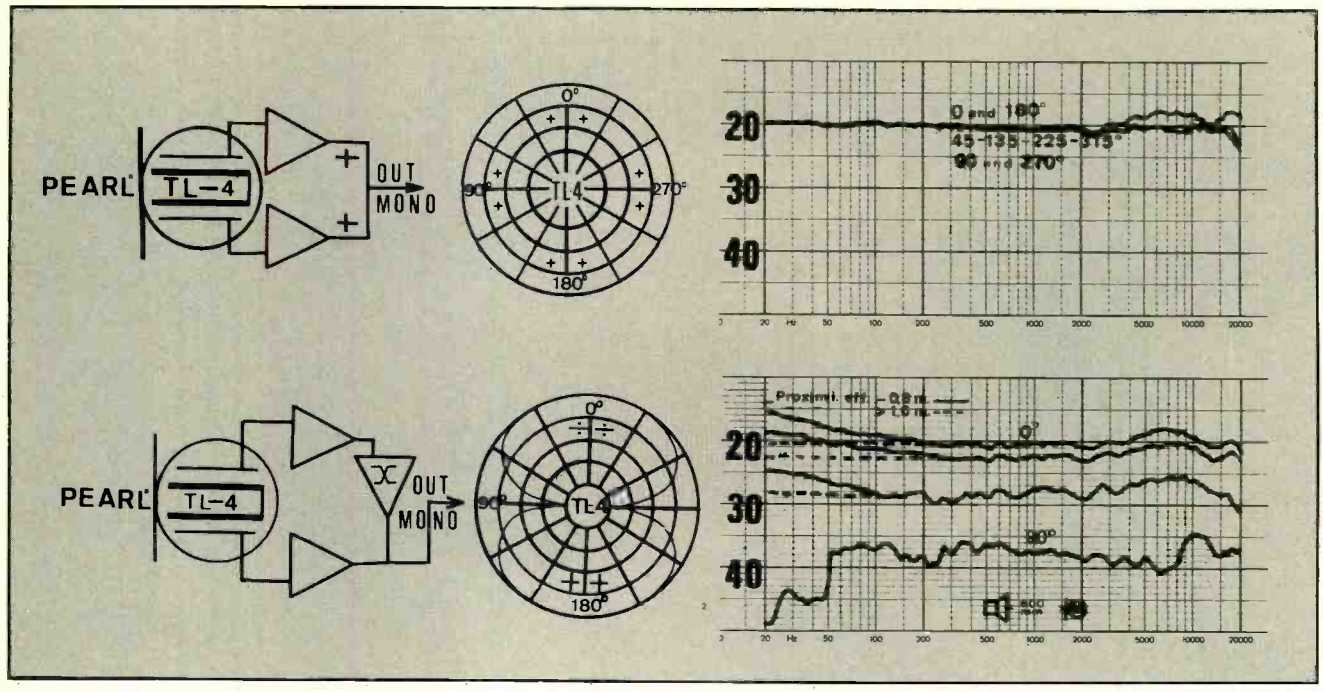

*Polar and frequency response plots of the TL-4 dual-membrane capsule outputs added in-phase (*top*) and inverted phase (*bottom*), showing the differences in directional and biodirectional/figure-of-eight pickup characteristics. Used independently, the two membranes provide a simultaneous pair of cardiod characteristics, with forward and rear-facing orientations.*

Technical Specifications

Patterns: *Back-to-back cardioid (coincident-stereo); omnidirectional through cardioid to bidirectional (totally variable at console; mono applications).*
Frequency range: *16Hz to 20kHz.*
Sensitivity: *120mV/Pa.*
Maximum output: *3.6V, both capsules operating; 5.8V, one capsule operating.*
Self-noise: *less than 20dBa, total system; less than or equal to 0.018mV, A-weighted, amplifier only (noise measurements made according to IEC-179).*
Current drain: *1.3mA, 24V to 48V.*
Maximum SPL *(6 kΩ load, less than 1% THD): 126dB, 48V phantom power, both capsules; 132dB, 24V, both capsules; 132dB, 48V, one capsule; 136dB, 24V, one capsule.*
Source impedance *(balanced, transformerless); 150Ω, 16Hz to 20kHz.*
Load impedance: *may be used with loads of less than 1,000Ω, but with reduced output.*
Connector: *5-pin XLR.*
Dimensions *(length and minimum/maximum diameters): 6.9"x0.9" x1.2".*

Accessories: *505 or 533 cable, model PDV-B pattern controller/recorder interface.*

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Recording was made on an Ampex ATR-104 ½-inch 4-track equipped with Dolby A-Type noise reduction, and operating at 15 ips with Ampex 407-3600 tape.

I was joined in our listening sessions by our two audio engineers, Peter Nothnagle and David Miller. We all agreed that the TL-4 is a truly excellent microphone with regard to subjective listening quality. It picked up the piano, woodwinds and certain other orchestral instruments 90° off-axis from either of its capsules, yet the TL-4 retained the proper frequency balance, clarity and stereo imaging that one expects from high-quality stereo microphones.

Also, because the reverberant characteristics of the hall were quite graciously rendered (again, received for the most part off-axis), one can conclude that the TL-4 maintains very favorable cardioid pattern characteristics across the audible spectrum. No unpleasant colorations were present in any form, either in the sound from the piano and orchestra, or from the hall's reverberation. Good off-axis characteristics are among the hallmarks of high-quality directional microphones.

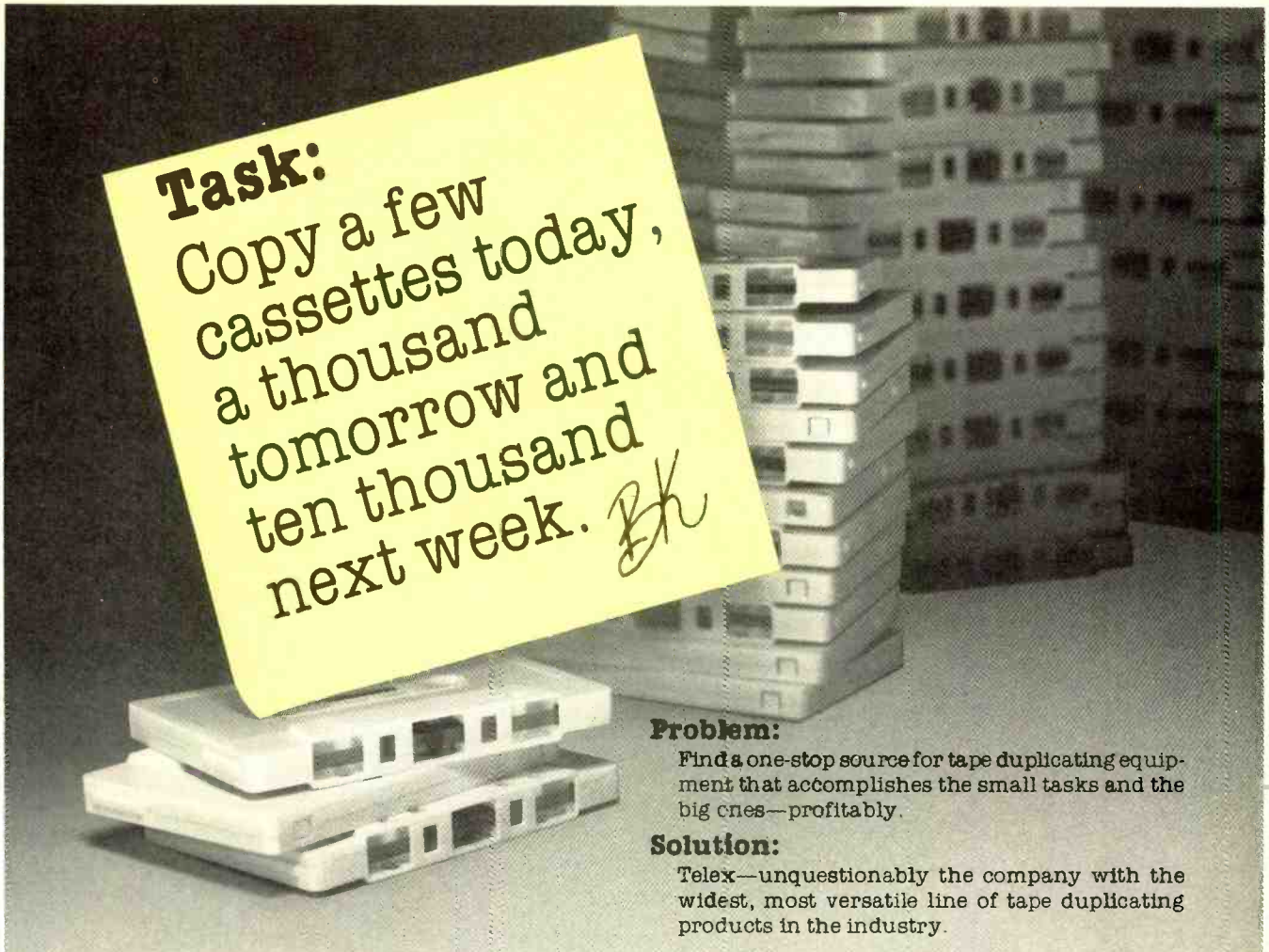
Compared with our normal reference microphones, the TL-4 sounded somewhat softer, with a trace of receding upper-midrange and high-frequency response. As noted, the off-axis pickup of the back-to-back cardioids can partly account for the difference. But this comment is not intended as a criticism; in fact, the soft high-end properties of the TL-4, with its atypical condenser sound, may be used in many recording situations as means for enhancement.

The excellent transient response and wide-band characteristics expected from quality condenser units were certainly present with the TL-4. Its rendition of the piano sound was full and robust. Its orchestral reproduction was very well-rounded, with none of the high-frequency stridency associated with lesser condenser units.

The sounds heard from the lower strings (cellos, basses) were characterized by richness and warmth, prompting a repeat of my admonition to ribbon advocates that they give condenser units a try. Nothing will be lost in smoothness or warmth, and definite gains in pattern integrity and selection, total system signal-to-noise ratio and transient response will become evident over virtually all of the present ribbon designs.

I do not hesitate to recommend the TL-4 as a worthy competitor to the prestigious microphones that have rated high in our previous evaluations. It is an excellent coincident-stereo microphone at a very reasonable price that can also double as a highly flexible variable-pattern studio unit capable of a wide range of solo applications.

R-E/P



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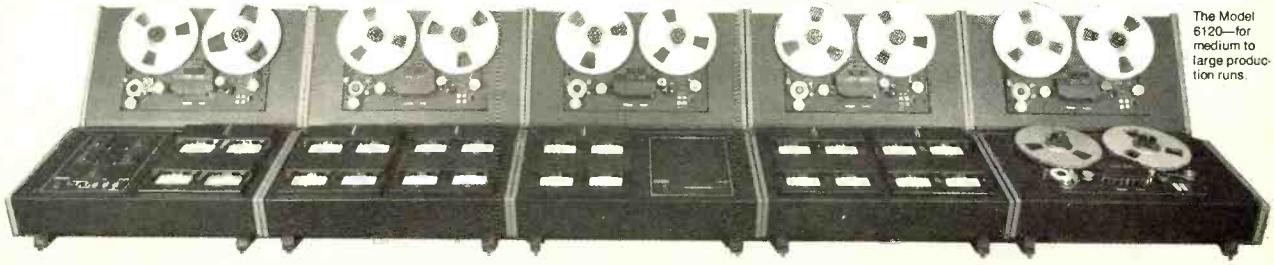

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Preview

Audio Engineering Society 81st Convention Los Angeles, Nov. 12-16

The 81st Audio Engineering Society Convention, whose theme is "The Digital/Analog Fusion: A Rainbow of Technology," will be held at the Los Angeles Hilton Hotel from Wednesday, Nov. 12 through Sunday, Nov. 16. An accompanying technical exhibition, located at the nearby Los Angeles Convention Center, provides more than 150,000 square feet of booth space for almost 200 exhibitors. Additional demonstration rooms will be sited at the Hilton Hotel.

Opening hours for the technical exhibits and demonstration rooms are as follows:

- Thursday, Nov. 13: 12 noon to 8 p.m.
- Friday, Nov. 14: 10 a.m. to 6 p.m.
- Saturday, Nov. 15: 10 a.m. to 5 p.m.
- Sunday, Nov. 16: 10 a.m. to 3 p.m.

Technical papers and workshop sessions will be open from 9 a.m. through 10 p.m.—during the five day convention, except Saturday (the AES Awards Banquet evening results in the afternoon session ending at 5 p.m.) and Sunday, when the convention closes at 12 noon after a morning workshop session.

Marshall Buck, papers chairman, has organized a series of technical sessions that will begin on Wednesday. These sessions will be presented in the Pacific Ballroom at the Los Angeles Hilton.

Wednesday, Nov. 12 (9 a.m.)

A session entitled "Perception" will be chaired by Diana Deutsch, and comprises the following invited papers:

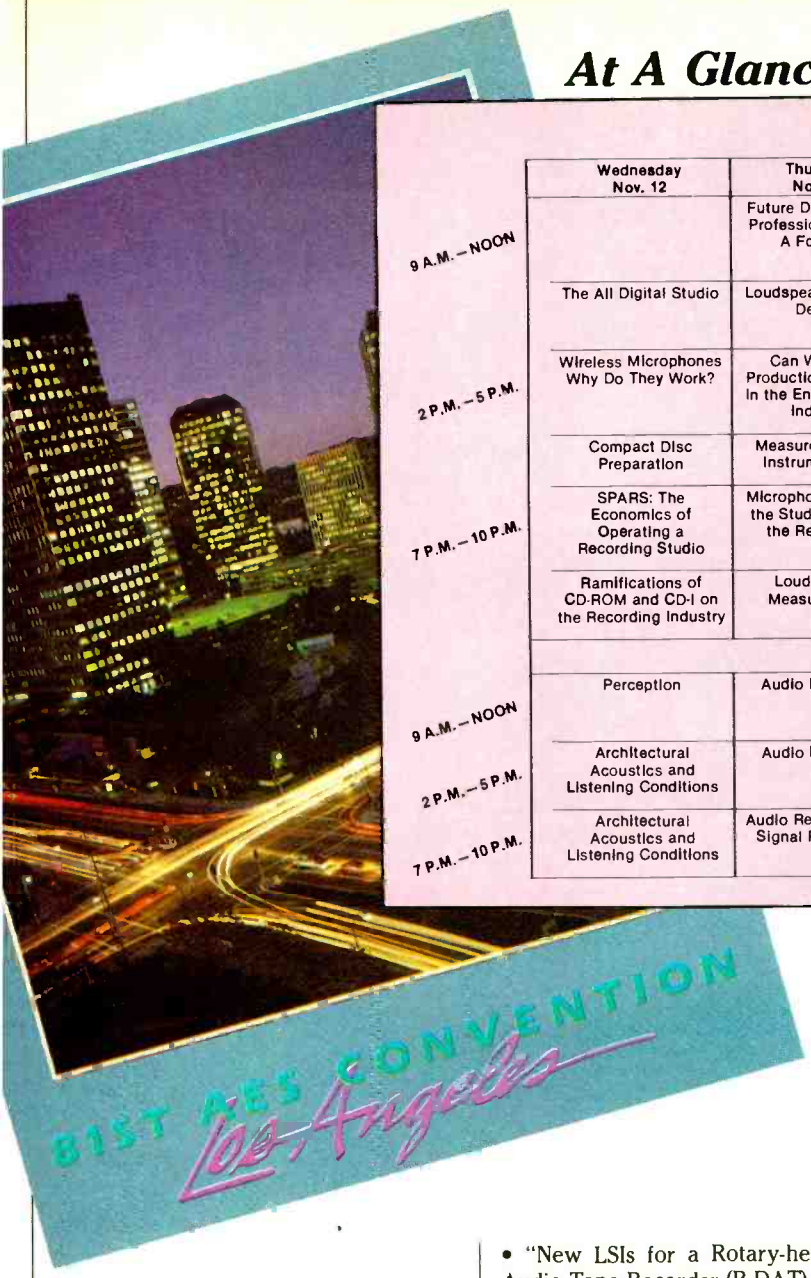
- "Some New Auditory Illusions," by Diana Deutsch, Department of Psychology, University of California at San Diego.
- "What do We Hear?" by John R. Pierce, Center for Computer Research in Music and Acoustics, Stanford University.
- "On the Question of Absolute Pitch," by W. Dixon Ward, Hearing Research Laboratory, University of Minnesota.
- "Hazards in Hearing for the Orchestral Musician," by Edward C. Carterette, Department of Psychology and Brain Research, UCLA, Donald H. Woolford, Australian Broadcasting Corporation, and Donald E. Morgan, Division of Head and Neck Surgery, School of Medicine, UCLA.
- "Acoustic Sequences: Explaining a Perceptual Paradox," by Richard M. Warren, Department of Psychology, University of Wisconsin-Milwaukee.
- "The Perception of Sound Colorations due to Resonances in Loudspeakers and Other Audio Components," by Floyd E. Toole, National Research Council of Canada.

Wednesday, Nov. 12 (2 p.m.)

A 2-part session entitled "Architectural Acoustics and Listening Conditions" will be chaired by Ted Uzzle, and comprises the following technical papers:

- "RT-60: How do I Measure Thee; Let me Count the Ways," by Peter D'Antonio, RPG Diffusor Systems, and Don Eger, Tecron.
- "Decay of Low-Frequency Sound in Rooms," by Arthur Noxon, Acoustic Sciences Corporation.
- "Sound Transmissions Through Partitions—Direct Source Field," by Marshall Long, Marshall Long Acoustics.
- "The Acoustical Design of a 4,000-seat Church Auditorium," by A.H. Marshall, C.W. Day and L.J. Elliott, Marshall Day Associates.
- "Interactive 3-D Graphics Systems for Auditorium/Cluster Design," by David B. Murrell and David G. Meyer, Purdue University.
- "New Types of Acoustic Materials Simplify Room Designs," by Peter D'Antonio, RPG Diffusor Systems.
- "Incorporating Reflection Phase Grating Diffusors in Worship Spaces," by Peter D'Antonio, RPG Diffusor Systems, and W. Peterson, Professional Sound Industries.
- "New Subjective Speaker Test Allows A/B Comparison with Original Sound Equivalent," by Henry O. Wolcott, Wolcott Audio.
- "LEDR: A Subjective Approach to Evaluating and Modifying Control Rooms," by Douglas R. Jones, William L. Martens and Gary S. Kendall, Northwestern University School of Music.
- "The Evaluation of Digital Filters for Controlling Perceived Direction in Stereo Speaker Reproduction," by William L. Martens, Gary Kendall, Richard Karsten and M. Derek Ludwig, Northwestern University School of Music.
- "Aural Analysis of the Spatial Relationships of Sound Sources as Found in 2-Channel Common Practice," by Dr. William Moylan, University of Lowell.
- "The Localization of Phantom Images in an Omni-Directional Stereophonic Speaker System," by Dave Moulton and Michael Ferralli, Phase Coherent Audio, and Steve Hebrock and Mark Reed, Audiotechnica USA.
- "Minimizing Interaural Crosstalk in Near-Field Monitoring by the Use of a Physical Barrier," by Timothy M. Bock, Crown International, and D.B. Keele Jr., Tecron.
- "Subjective Evaluation of Multichannel Stereophony for HDTV," by K. Ohgushi, K. Tsujimoto, S. Komiyama, K. Kurozumi and J. Ujihara, NHK Science and Technical Research Labs.
- "The Influence of Television Images on Auditory Localization," by Bronwell,

At A Glance:



Workshops

	Wednesday Nov. 12	Thursday Nov. 13	Friday Nov. 14	Saturday Nov. 15	Sunday Nov. 16
9 A.M.–NOON		Future Directions in Professional Audio: A Forecast	Stereo TV Mixing: Compromises and Solutions for Mono Compatibility	Preservation and Restoration of Audio	Education in Audio: Does Testing Work?
	The All Digital Studio	Loudspeaker Cluster Design	Tape Machine Maintenance Care and Repair	MIDI and Beyond: Total Stereo Control	The Art and Science of Equalization
2 P.M.–5 P.M.	Wireless Microphones Why Do They Work?	Can We Talk? Production Intercom in the Entertainment Industry	Audio for Video	Film Sound: Dialogue, Music and Effects	Basic System Design of Recording Studios
	Compact Disc Preparation	Measurement and Instrumentation	Computers in Audio	Live Concert Sound	
7 P.M.–10 P.M.	SPARS: The Economics of Operating a Recording Studio	Microphones: Out of the Studio and Into the Real World	Time Code: A Tutorial	AES Awards Banquet featuring The Firesign Theatre and keynote speech by Stan Cornyn Reception–6 p.m.	
	Ramifications of CD-ROM and CD-I on the Recording Industry	Loudspeaker Measurements	Transformers in Audio		

Papers

9 A.M.–NOON	Perception	Audio Recording	Audio Reproduction, Transducers and Sound Reinforcement	Audio Measurements and Instrumentation	
2 P.M.–5 P.M.	Architectural Acoustics and Listening Conditions	Audio Recording	Audio Reproduction, Transducers and Sound Reinforcement	Audio Measurements and Instrumentation	
7 P.M.–10 P.M.	Architectural Acoustics and Listening Conditions	Audio Recording and Signal Processing	Audio Reproduction, Transducers and Sound Reinforcement	AES Awards Banquet Reception–6 p.m.	

L. Jones and Emil L. Torick, CBS Technology Center.

Thursday, Nov. 13 (9 a.m.)

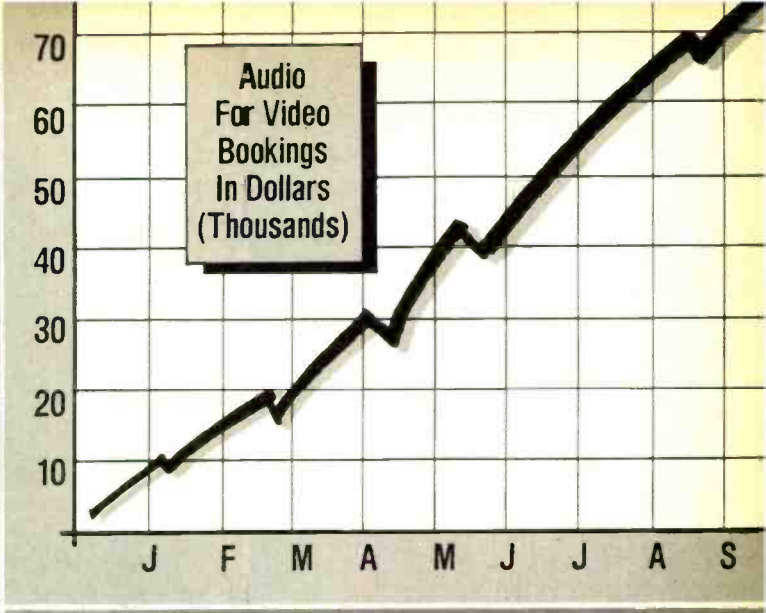
A 3-part session entitled "Audio Recording and Signal Processing" will be chaired by John Eargle, and comprises the following technical papers:

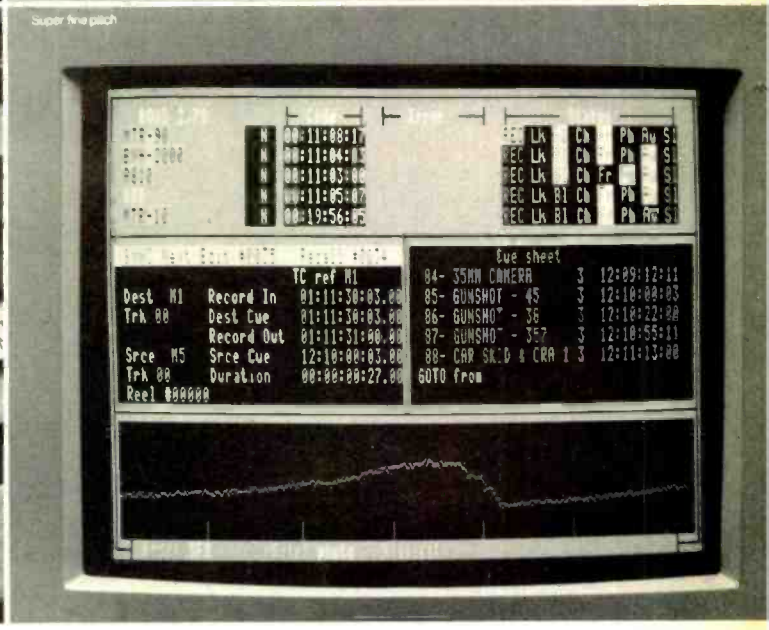
- "Direct Metal Mastering Technology: A Step Toward More Efficient Manufacturing of Compact Discs," by Horst Joschiko and Guenter Joschiko, Teldec GmbH and Gotham Audio.
- "An Automated Approach to Digital Console Software," by William Kentish and David C. Bell, Solid State Logic.
- "PDM for Digital Audio Tape Recorders," by Marcel Schneider, Willi Studer AG.
- "New LSIs for a Rotary-head Digital Audio Tape Recorder (R-DAT) and Their Digital Signal Processing," by Tadashi Fukami, Shinya Ozaki, Kazutoshi Shimizume, Takeshi Uematsu and Kentaro Odaka, Sony Corporation.
- "A New Audio Bit-rate Reduction System for the CD-I Format," by Masayuki Nishiguchi, Kenzo Akagiri and Tadao Suzuki, Information Systems Research Center, Sony Corporation.
- "Music and Sound Design for Motion Pictures and Commercials," by Frank Serafine, Serafine FX.
- "Spectral Recording Process," by Ray Dolby, Dolby Laboratories.
- "The Application of Narrowband Dither Operating Near the Nyquist Frequency in Digital Audio Systems to Provide Improved Signal-to-Noise Ratio over Conventional Dithering," by B. Blesser, Barry Blesser Associates, and B. Locanthi, Pioneer Electronics.
- "Digital Dither," by Stanley P. Lipshitz and John Vanderkooy, Audio Research Group, University of Waterloo.
- "An Optical Disc Recording, Archiving and Editing Device for Digital Audio," by James Moorer and Jeffrey Borish, The Droid Works.
- "A Digital Audio Contact Printing Technique," by Toshikaru Kobayashi, Yoshio Shirai and Shinji Amari, Sony Corporation.
- "On the Design of Digital Oversampling Filters," by Ming-Ting Sun and Lance Wu, Bell Communications Research.
- "A New Compressor Architecture for Digital Audio," by John Strawn, The Droid Works.
- "Software for Controlling Real-Time Digital Audio Processors," by Mike Shantzis and Bernard Mont-Reynaud, The Droid Works.
- "An Integrated Voltage-Controlled Building Block," by Douglas Frey, Computer Science Department, Lehigh University.
- "Music Synthesis Tutor for FM Power Spectra," by Gary L. Gibian, David Bort, Dennis Clements and Eric Harnden, The American University, Physics Department.
- "Digital Audio Processing on a Grand Scale," by Peter Eastty and David C. Bell, Solid State Logic.

• "Automatic Generation of Microcode for a Digital Audio Signal Processor," by C.M. McCulloch, Solid State Logic.
• "Image Model Reverberation for Recirculating Delays," by Gary Kendall, William Martens and Daniel Freed, Northwestern University School of Music.

Friday, Nov. 14 (9 a.m.)

A 3-part session entitled "Audio Perception, Tranducers and Sound Reinforcement" will be chaired by Cal Perkins, and comprises the following technical papers:
• "Complete Analysis of Single and Multiple Loudspeaker Enclosures," by Chris Strahm, Northwest Sound.
• "Automated Measurements of Loudspeaker Small-signal Parameters," by Richard C. Cabot, Audio Precision.
• "Advances in Computer Simulation of Loudspeakers," by Earl Geddes, Ford Motor Company, Electrical Division.
• "The Acoustic Radiation of Line Sources of Finite Length," by Stanley P. Lipshitz and John Vanderkooy, Audio Research Group, University of Waterloo.
• "Are Equalized Closed-boxes Preferable to Vented-boxes?" by Eugene Zuustinsky, Department of Mathematics, State University of New York.
• "A Transmission Line Woofer Model," by Robert M. Bullock III and Peter E. Hillman, Miami University, Mathematics Department.
• "Acoustical Field Studies of Piston-type Vibrating Surfaces Using Boundary Element Method," by J.K. Jiang and M.G. Prasda, Stevens Institute of Technology.
• "Comparison Between 4th-order Linkwitz-Riley and Bessel Active Crossovers," by Dennis A. Bohn, Rane Corporation.
• "Present and Future Trends in High-level Sound Reinforcement Loudspeaker Systems," by Clifford Hendricksen, Electro-Voice.
• "Subtractive Crossover Filters," by Eric Guarin, University of Illinois.
• "Loudspeaker Manifolds for High-level Concert Sound Reinforcement," by David E. Carlson and David W. Gunness, Electro-Voice.
• "Elimination of High-power Compression in Servo-drive Loudspeakers," by Thomas J. Danley, Roy R. Whymark and Charles A. Rey, Intersonics.
• "Cluster Suitability Predictions Simplified," by G.R. Thurmond, G.R. Thurmond & Associates.
• "Electronic Control of Speaker Directivity: A Practical Application," by David Russell, Biola University, Physical Plant.
• "An Analysis of Some Off-axis Stereo Localization Problems," by John M. Eargle and Gregory Timbers, JBL.
• "The Complex Simulation of Acoustical Sound Fields by the Delta Stereophony System DDS," by Dr. Wolfgang Ahnert, Institut fur Kulturbauten.
• "The 'Delta Stereo Compact Processor' (DSCP) to Utilize New Directional Sound Reinforcement System," by Wilheim Nadler, AKG Acoustics.
• "360-degree Dispersion Frequency Invariant Acoustic Transduction System," by Michael Ferralli, Phase Coherent Audio





The Boss™ makes money.

Good business decisions begin with a clear understanding of what an investment will return to you. That's why at Alpha Audio we believe a worthwhile product pays its buyer back: in efficiency, in creativity, and in dollars. And that's why we're so bullish on The BOSS, our premiere audio editing system.

Since its introduction in January 1986, the BOSS has been generating high revenues at studios in Atlanta, Hollywood, Detroit, Houston and Philadelphia. Because our system *works,* from automated track building to looping, from video edit decision list exchange to MIDI control, from sophisticated machine handling to extensive list management.

We feel the measure of a good system is not what it promises, but what it delivers. Isn't that the basis for a good buying decision?

Shake hands with The BOSS.

For further information, contact David Walker, Director of Marketing. (804) 358-3852.

Alpha Audio.
Automation Systems

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Circle (52) on Rapid Facts Card

• "Reducing Off-axis Comb Filter Effects in Highly Direction Microphones," by Yuri Shulman, Shure Brothers.
• "The Impact of High-energy Magnetic Structures on Dynamic Microphones," by D. Ray Kirchoefer and Alan R. Watson, Electro-Voice.
• "Acoustic Load and Transfer Impedance in Listening Rooms," by Tomas Salava, TESLA-VUST.

Saturday, Nov. 15 (9 a.m.)

A 2-part session entitled "Audio Measurements and Instrumentation" will be chaired by Henning Moller, and comprises the following technical papers:
• "High-frequency Phase Response Specifications: Useful or Misleading?" by Deane Jensen, Jensen Transformers.
• "The Analytic Impulse," by Andy Duncan, Cerwin Vega.
• "Measurement of the Dynamic Transfer Characteristics of Multiband Signal Processing Systems by TDS," by Jim Brown, Sound Engineering Associates.
• "Hidden Variables in Electronic/Acoustic Interfacing," by Richard Fay and Jim K. Rhodes, Lenco.
• "Real-time Measurement of CD Player Jitter Distribution," by Dr. Susumu Matsuoka and Masaoki Takai, Kenwood Corporation.
• "On the Use of Computer-generated Dither Test Signals," by Robert A. Finger, CBS Technology Center.
• "STI Measurement Using a Dual-channel Analyzer," by Klaus Hoejbjerg, Bruel & Kjaer.
• "Computing Peak Currents into Loudspeakers," by John Vanderkooy and Stanley P. Lipshitz, Audio Research Group, University of Waterloo.
• "Instantaneous Intensity," by Richard C. Heyser, Jet Propulsion Laboratory.
• "Specific Acoustic Admittance," by Richard C. Heyser, Jet Propulsion Laboratory.
• "Measurement of Transfer-characteristic Discontinuity," by N. H. C. Gilchrist, BBC Research Department.
• "Digital Synthesis of High-quality Test Signals Conforming to the BTSC TV Stereo System," by Jorgen Voldby, RE Instruments A/S.
• "High-performance Dual-channel Signal Analyzer Implementation Teams with Personal Computers for Power and Flexibility," by Jim DuWalt, IQS.

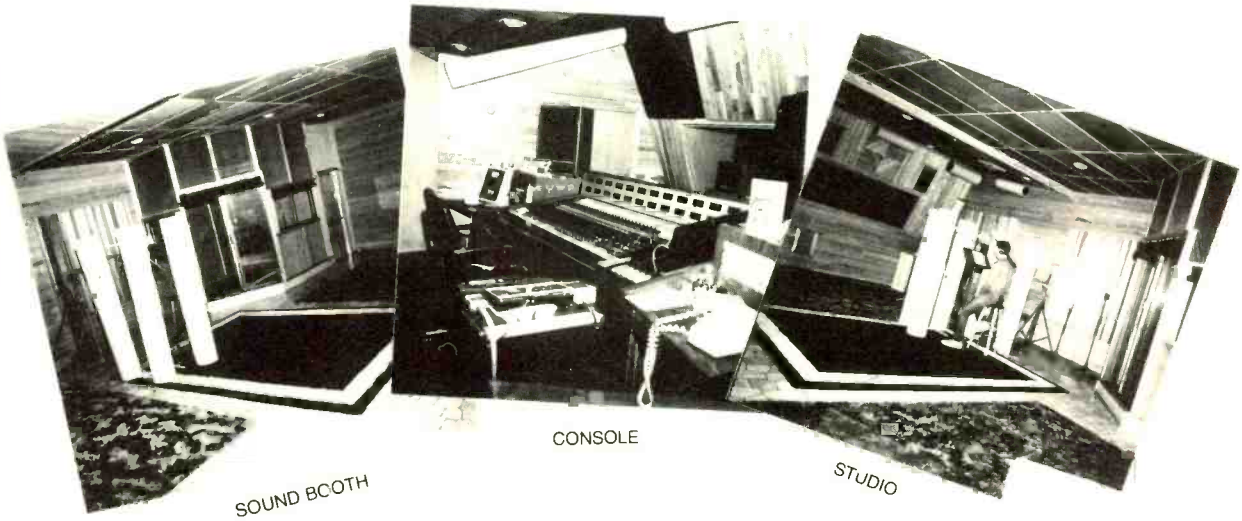
Workshop sessions

In addition to the technical papers sessions, a series of concurrent hands-on workshops has been organized. The dates and subjects are as follows:

Wednesday, Nov. 12

• "The All-digital Studio," with Rhonda Kohler.
• "Wireless Microphones: Why do They Work," with Bill Mayhew.
• "Compact Disc Preparation," with Rhonda Kohler.
• "The Economics of Operating a Recording Studio," with Nick Colleran.
• "Ramifications of CD-ROM and CDI on the Recording Industry," with Rhonda Kohler.

FOR SALE . . .



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CAPABLE OF RECORDING THE PROFESSIONAL QUALITY DEMANDED BY TODAY'S ARTISTS — LOCATED IN THE WILLAMETTE VALLEY BETWEEN THE MAJOR MARKETS OF LOS ANGELES AND SEATTLE — IT HAS RECORDED THE SPECIAL TALENTS OF MANY NATIONALLY KNOWN MUSICIANS.

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- Ampex 1200-24 track recorder, with 20 search to cue and full VSO
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- 2 Lexicon Prime Time digital delays
- Lexicon 224 X digital reverb
- Valley People Kepex II's/Gain Brain II's/Maxi-Q's
- Crown amplifiers
- Orban Parametric EQ
- AKG Spring reverb
- AKG C414 EB's, 452's
- Neumann U-47, U-87's, U-89's
- Neumann KM 84's
- Sennheiser MD 421's
- Shure 57
- Yamaha Grand Piano
- various cables, headphones, etc.

THIS STUDIO AND EQUIPMENT CAN BE SEEN BY APPOINTMENT.

For information or an appointment, call:
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Thursday, Nov. 13

- "The Business of Audio: FX + TX = RX?" with Martin Polon.
- "Loudspeaker Cluster Design," with John Prohs.
- "Can We Talk? Production Intercom in the Entertainment Industry," with Dave Brand.
- "Measurement Instrumentation," with Richard Cabot.
- "Microphones: Out of the Studio and into the Real World," with John Phelan.
- "Loudspeaker Measurements," with Floyd Toole.

Friday, Nov. 14

- "Stereo TV Mixing: Compromises and Solutions for Mono Compatability," with Bill Burnsed.
- "Tape Machine Maintenance: Care and Repair," with Greg Hanks.
- "Audio for Video," with Ed Lever.
- "Computers in Audio," with Russell Berger.
- "Time Code: A Tutorial," with Steve Krampf.
- "Transformers in Audio," with Bill Isenberg.

R-E/P

Welcome to the 81st AES Convention

From Robert B. Schulein,
AES president 1986

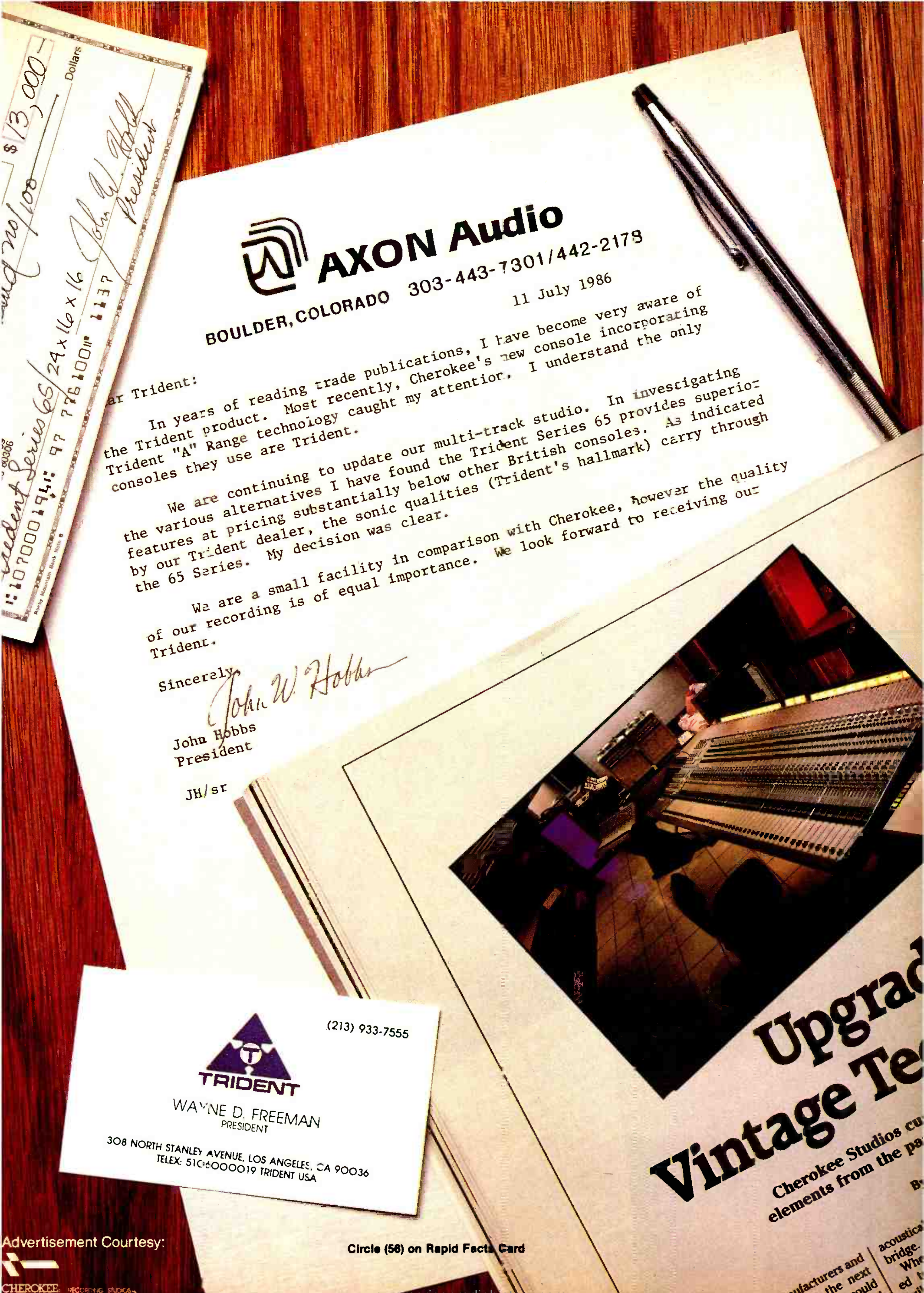
The technical program for the 81st AES Convention provides a clear indication that audio engineering is indeed undergoing significant change. From the vantage point of a discipline serving a broad range of audio communications needs, our industry is driven by both changes in technology and changes in need. It is with a dedication to anticipating the technological needs of the audio professional that the various aspects of this convention have been planned and organized.

There are two major forces significantly influencing our industry today: digital signal-processing technology and the growth of stereo audio-for-video and -television. With an initial emphasis on recording, digital signal-processing techniques have evolved to where they are providing better solutions and expanding the creative aspects of measurement instrumentation and music generation.

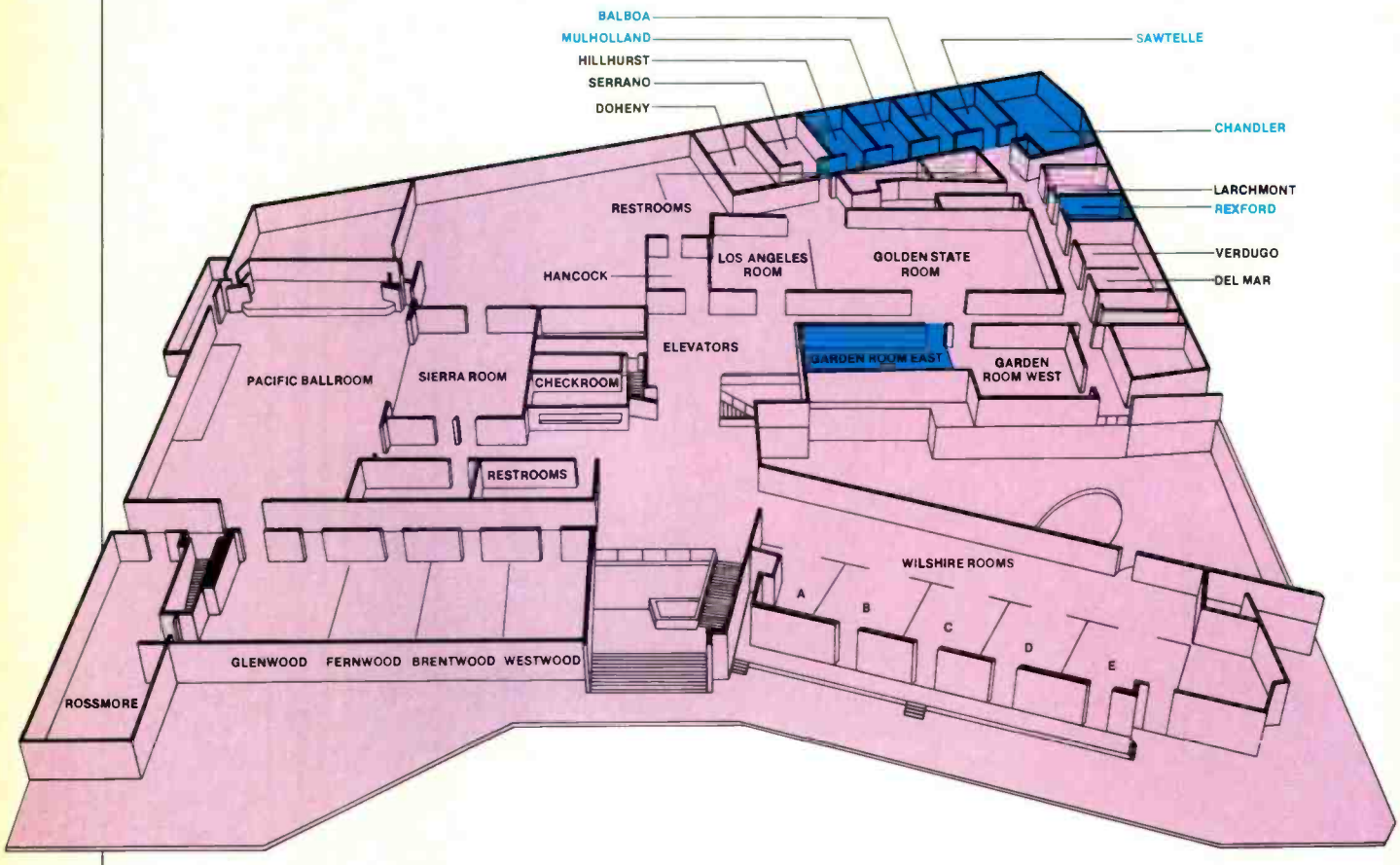
The influence of digital audio is escalating as the cost of circuitry components goes down, and as the technical knowledge base increases.

With the worldwide growth of audio-for-television and -video, many new audio engineering opportunities are opening up, with the potential of providing more realistic and exciting audio experiences for the end-user. Stereo audio with pictures, having been initially limited to the motion-picture industry, is now influencing both the broadcast television, prerecorded video and the recording industries. This influence has not only created a need to solve new problems, but has opened up new employment opportunities as well.

If you are an audio professional or are preparing to be one, the 81st AES Convention continues a tradition of the past 35 years by not only informing you of today's technology but preparing you for the future.



Hilton Hotel Demo Rooms



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Issue advertisers and their rooms are printed in blue

Audio/Digital (Hillhurst Room)
Cerwin-Vega (Wilshire Rooms)
Eastern Acoustic Works (Rexford Room)
E-Mu Systems (Glenwood Room)
JVC Company of America (Serrano Room)
Kurzweil Music Systems (Verdugo and Del Mar Rooms)
Peavey Electronics (Larchmont Room)
Sony Broadcast Products (Garden Room East)
Steinway & Sons (Brentwood Room)
Yamaha (Chandler, Sawtelle, Balboa and Mulholland Rooms)

IN THE PAST WE HAD A BIG ADVANTAGE OVER THE COMPETITION. NOW WE'VE GOT A SMALL ONE.

Until UREI's 813 Time Align® Monitor entered the studio, speaker systems had become a "smear" on the industry. A "time smear," in which high and low frequencies subtly assaulted the ear because they arrived out of sync. The results were general listener fatigue and unrealistic sound, particularly on lead instruments and vocals.

The UREI 813 solved the "time smear" problem with Time Alignment™, unifying sound into a single point source. This dramatic breakthrough, along with other major technical advances, soon established the 813 as the industry standard.

Now UREI introduces less of a good thing: the 809 Time Align® Studio Monitor. The 809 delivers all the engineering depth of its big brother, but at a compact size and price that's ideal for small control rooms and near-field applications.

UREI's 809 features a remarkable, all-new 300mm (12") coaxial driver that achieves a true one-point sound source, superior stereo imaging, and tight bass. It incorporates a unique titanium diaphragm compression driver that unleashes unequalled high frequency response.

The 809 has exceptional power handling capabilities, high sound sensitivity, and low distortion. It accomplishes precise acoustic impedance matching and smooth out-of-band response with UREI's patented high-frequency horn with diffraction buffer. And its ferrite magnet structures assure the system's high sensitivity drivers will not degrade with time and use.

UREI's Model 809 Time Align® Studio Monitor. Smaller package. Smaller price. Same impeccable "813" sound quality. See how it fits into your studio today.

JBL Professional
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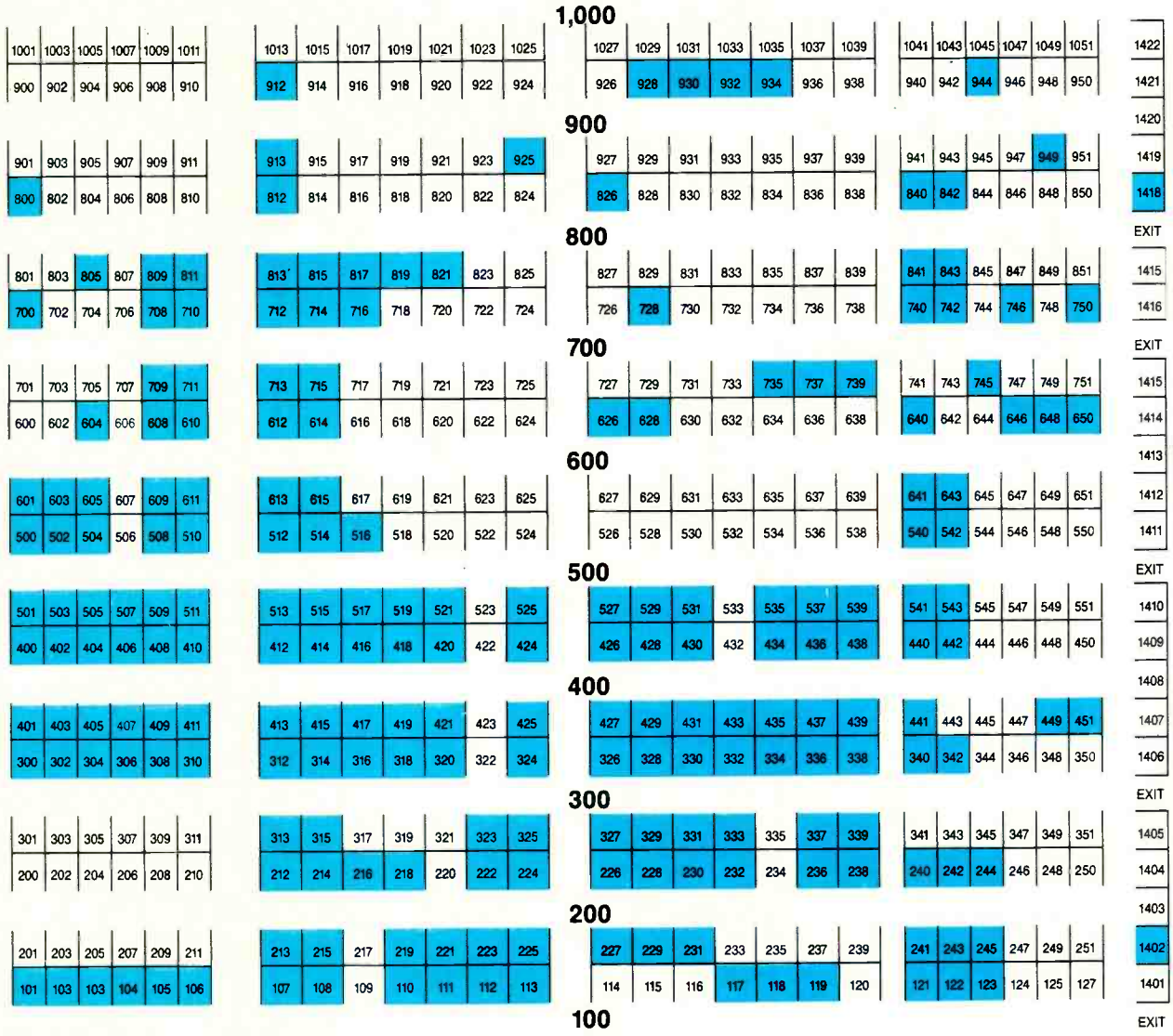
Convention Center Exhibitors



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ACO Pacific (827)
Acoustic Sciences (1418)
Adams-Smith (446, 448, 450, 551)
Advanced Music Systems (718, 720)
AEG (928, 930, 932, 934)
Agfa-Gevaert (208, 210, 309, 311)
AKG Acoustics (236, 238, 337, 339)
ALD Laboratory (915, 917)
Alesis (910)
Allen & Heath Brenell USA (819, 821)
Alpha Audio (508, 510, 609, 611)
Altec Lansing (822, 824)
Amek Consoles (712, 714, 716, 813, 815, 817)
American Multimedia/Concept Design (731, 733)
American Recorder Technologies (906)
Ampex (101-106)
Anchor Audio (604)
ANT Telecommunications (535, 537)
Anvil Cases (911)
Apex Machine Company (1404, 1405)

A.P.G. Ingeniere (921, 923)
Aphex Systems (626, 628)
API Audio Products (728)
Apogee Electronics (547)
Apogee Sound (1402)
Applied Research & Technology (746)
A.T.B. SPA (347)
Audico (211)
Audioanalyse (919)
Audiocast Conseil (927)
Audio Kinetics (117, 118, 119)
Audio Precision (826)
Audio-Technica U.S. (233, 235)
Audio Video Consultants (343, 345)
Kenneth A. Bacon Associates (805)
Barcus-Berry Electronics (912)
BASF (125, 127, 249, 251)
B & B Systems (925)
B.C. Inc. (935)
Beyer Dynamic (516)
BGW Systems (201)
Biamp Systems (1412, 1413)
Boulder Amplifiers/Silver Lake Research (1410)
Bruel & Kjaer Instruments (240, 242, 244)
Bryston Ltd. (800)
CAE Inc. (926)
Cal Switch (205, 207)
Canare Cable (743)
Dwight Cavendish Company (940)
Cetec Gauss (340, 342, 441)
Cetec Ivie (644)
Cetec Vega (350)
Cipher Digital (840, 842)
CMX (828, 830)
Community Light & Sound (506)
Connectronics (730)
J.L. Cooper Electronics (944)
Countryman Associates (704)
Crest Audio (617, 619)
Crown International (227, 229, 231)
CST Sales (1403)
db Magazine-Sagamore Publishing (936)
dbx (121, 122, 123, 241, 243, 245)
DCS Audio Products (602)
Design Direct Sound (630)
Dialight (Kulka Smith) (951)
Digital Creations (445, 447)
Digital Dispatch (801, 803)
DOD Electronics (646, 648, 650)
Dolby Laboratories (222, 224, 323, 325)
Dorrough Electronics (745)
D & R Electronica (747, 749)
Educational Electronics Corporation (904)
Electronic Systems Laboratories (1401)
Electro Sound (724, 825)
ElectroSpace Developments (549)
Electro-Voice (317, 319)
Elison (920)
Eventide (313, 315)
Everything Audio (449, 451)
Fender Musical Instruments (849, 851)
Fife-Pearce Electric Company (950)
FM Acoustics (847)
Focusrite (836)
Fostex Corporation of America (500, 502, 504, 601, 603, 605)
Full Sail-Center For the Recording Arts (829)
Gold Line (443)
Gotham Audio (520, 522, 524, 621, 623, 625)
David Hafler Company (518)
Harrison Systems (618, 620, 622, 624, 719, 721, 723, 725)
HM Electronics (341)
Ibanez (813)
Heino Ilsemann (344)
Industry West Electronics (818)
Innovative Electronic Design (732)
Inovonics (607)
Institute of Audio/Video Engineering (943)
International Music Co./Akai Digital (546, 548, 550, 647, 649, 651)
IPS (848)
IQS Inc. (702)
JBL Professional (326, 328, 330, 332, 334, 336, 338, 427, 429, 431, 433, 435, 437, 439)
Kenwood U.S.A. (933)
King Instrument (346, 348)
Klark-Teknik (608, 610, 709, 711)
Thomas Klotz Musikelektronik (701)
Korg U.S.A. (814, 816)
Kyric (907)
Lexicon (424, 426, 428, 430, 525, 527, 529, 531)
L.T.M. Sound Dept. (834)
Magnefax International (349, 351)
Marshall Electronics (751)
Martin Audio London Ltd./Martin America (913)
Meyer Sound (512, 514, 613, 615)
Milab International (838)
Mitsubishi Pro Audio Group (526, 528, 530, 532, 534, 627, 629, 631, 633, 635)
Mix Magazine (1411)
Monster Cable (833)
Mosses & Mitchell (937)
Nagra Magnetic Recorders (203)
Nakamichi U.S.A. (812)
Neotek (540, 542)
Rupert Neve (200, 202, 204, 301, 303, 305)
Orban Associates (324, 425)
Otari (110, 111, 112, 113, 219, 221, 223, 225)
Oxmoor (942)
Pacific Radio Electronics (929, 931)
Paktec Automation (1406)
Penn Fabrication (939)
Penny + Giles (209)
Pro Co Sound (748)
Professional Audio Services & Supply (703, 705)
Professional Audio Systems (246, 248)
Professional Sound Corporation (1409)
Publison America (634, 636, 638)
Pyral (Rhone-Poulenc Multi-Techniques) (850)
Recording Engineer/Producer, Sound & Video Contractor (918)
RE Instruments (938)
Renkus-Heinz (700)
Rent Music International (914, 916)
Research Technology International (948)
Roland Corp. US (440, 442, 541, 543)
RPG Diffusor Systems (642)
RTS Systems (844, 846)
Saki Magnetics (726)
Samson Products (1407, 1408)
Sanken Microphone/Pan Communications (750)
Schoeps/Posthorn Recordings (640)
S.C.V. Audio (1414, 1415)

Exhibitors continued on page 108

Selco/Sifam (806)
Sennheiser (237, 239)
Shure Brothers (107, 108, 213, 215)
Sierra Audio (1025)
Skotel (947)
Solid State Logic (226, 228, 230, 232, 327, 329, 331, 333)
Sontec Electronics (946)
Sony Broadcast Products (300, 302, 304, 306, 308, 310, 401, 403, 405, 407, 409, 411, 400, 402, 404, 406, 408, 410, 501, 503, 505, 507, 509, 511)
Sound Engineer and Producer (908)
Sound Ideas (632)
Soundmaster International (641, 643)
Sound Technology (808, 810)
Soundtracs (740, 742, 841, 843)
Sound and Vision (949)
Stanton Magnetics (120)
Star Case Manufacturing (820)
Studer Revox America (312, 314, 316, 318, 320, 413, 415, 417, 419, 421)
Studio Master (909)
Studio Sound (727)
Studio Technologies (741)
Sunkyong International (736, 738, 837, 839)
Swintek Enterprises (1416)
Tannoy North America (438, 539)
Tape Automation (802, 804)
TASCAM/TEAC Corporation of America (412, 414, 416, 418, 420, 513, 515, 517, 519, 521)
Technics (901)
Telex Communications (735, 737, 739)
3M/Magnetic Media Division (212, 214, 216, 218)
360 Systems (922, 924)
Timeline (903, 905)
Times One Technology (1001)
Trident USA (612, 614, 713, 715)
TTL USA (729)
Turbosound (708, 710, 809, 811)
27th Dimension (832)
Valley People (250)
V.C.L. Audio SA (900, 902)
Versadyne International (1417)
Whirlwind Music Dist. (434, 436)
Wireworks (538, 639)
Zonal (941)

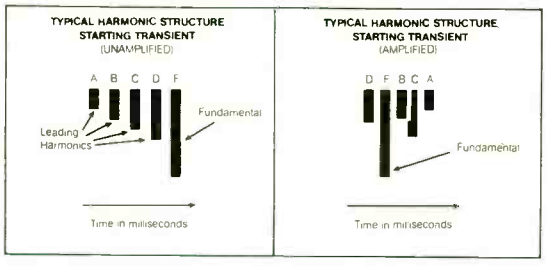
Late Entries

Audiotek (1015)
Carver (1003)
Celestion Industries (921)
Clarity (1013)
El Mar Plastics (1027)
FM Tubecraft Support Systems (1017, 1019)
Midas Audio (1005, 1007)
Optimix International (1009)
PPG America (1021, 1023)
QSC Audio Products (250)
Shape Inc. (1419, 1420)
SPARS (1421)
Stellavox (1011)

"I like the way BBE™ brings out the 'live' dimension in sound."

Bernie Grundman

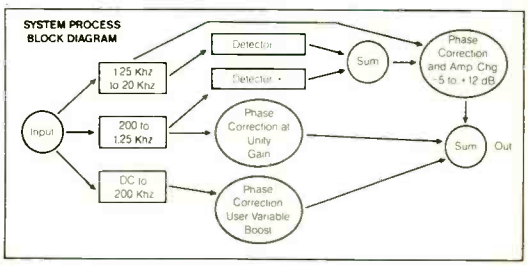
When it comes to cutting discs, Bernie Grundman is the master. Ever since the late 60's, the music industry's top artists have been entrusting Grundman with the delicate task of preparing their record masters. The list of satisfied clients includes Stevie Wonder, Prince, Michael Jackson, Olivia Newton-John and Lionel Ritchie, to name just a few. So when a critical listener like Bernie Grundman lists himself as one of BBE's satisfied customers, it's quite a compliment.



"Unlike an equalizer, BBE is a dynamic process. It gives me the high end I want without the sibilance and without creating energy levels too intense for my cutting lathe to handle. I can foresee BBE becoming a very valuable tool in the mastering process."

BBE is a critical link in the chain of precision components used at Bernie Grundman Mastering in Los Angeles. Even the finest components, you see, can be undermined by that all-important interface between amplifier and speaker. That's where phase and "overhang" distortion develop, due to voice coil characteristics, reflected impedance from the environment, crossover impedance anomalies and the mechanical properties of dynamic speakers. The relationships among the fundamental frequencies, their leading harmonics and between the leading harmonics themselves become distorted in both amplitude and time. The result? Muddiness, poor imaging and pinched, colorized sound that lacks the presence and punch of the real thing.

BBE puts back all the clarity and sparkle into amplified and recorded sound. We like to think of it as the "unprocessor." Rather than artificially altering the original source, BBE restores the natural harmonic balances that were present in the live performance. How? First it divides the audio spectrum into three bandwidths. Then it applies phase correction across the full spectrum and dynamic high frequency amplitude compensation as required. BBE's continual sampling of the mid/high frequency relationship allows this correction to take place automatically. Convenient front-panel controls let you boost low frequencies and regulate the amount of high-frequency amplitude correction to suit your needs. There's no encoding or decoding involved, so BBE can be used anywhere in the recording chain—from individual tracks on a multitrack tape to a mastering lab monitoring system.



For disc mastering perfectionists like Bernie Grundman, BBE makes an audible difference. Shouldn't BBE be a part of your next masterpiece?

To find out what BBE technology can do for your sound, call us toll-free at 1-800-233-8346. (In California, call 1-800-558-3963.) Or write to us at Barcus-Berry Electronics, 5500 Bolsa Avenue, Suite 245, Huntington Beach, CA 92649.

BBE™ All the sound you've never heard.™

Barcus-Berry Electronics, Inc.

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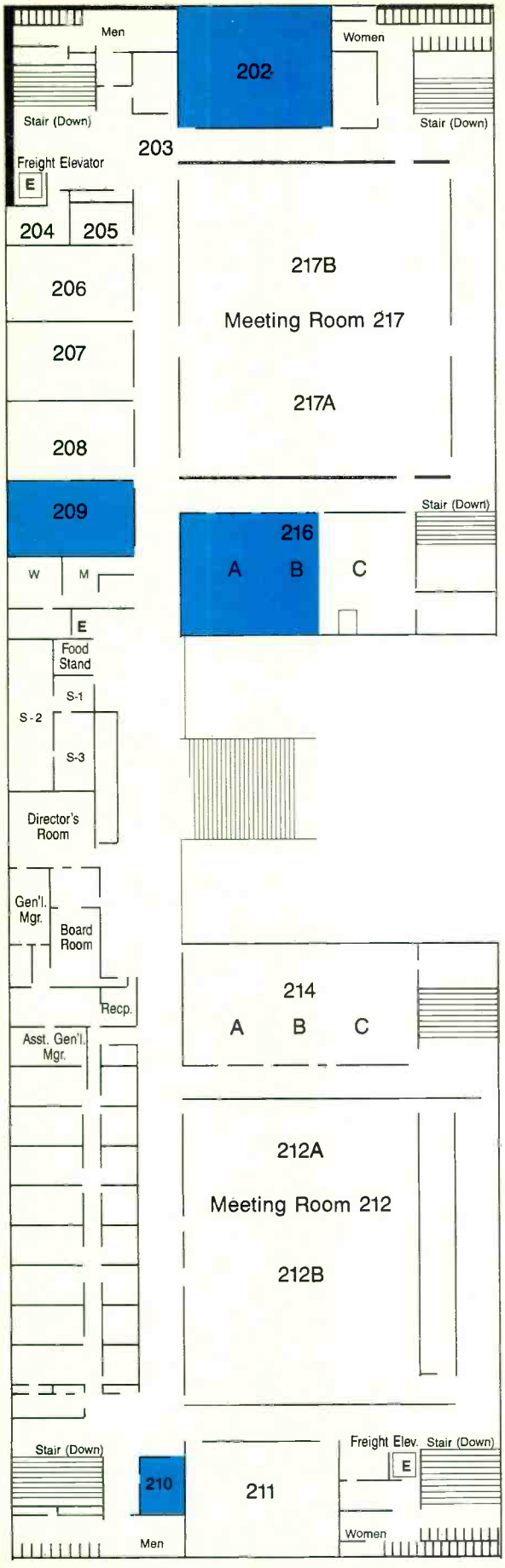

Convention Center Demo Rooms

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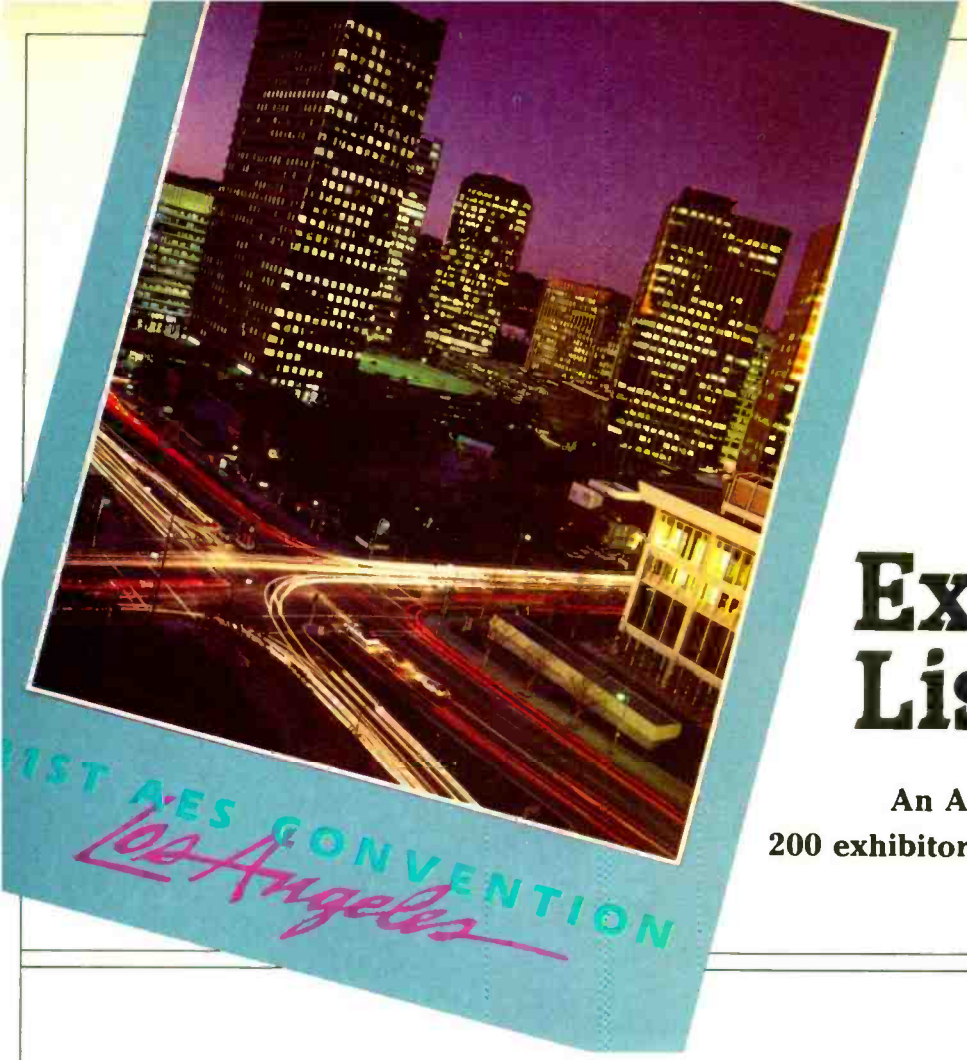
Fairlight Instruments (Room 202)
New England Digital (Rooms 216 A & B)
Panasonic Industrial/RAMSA (Room 209)
Westlake Audio (Room 210)



Exhibitor Listings

An A-Z listing of over 200 exhibitors at the AES Convention.

If you plan to attend the 81st AES Convention and Exhibition in mid-November, **RE/P**'s pre-show coverage can help you plan your visits to the more than 480 booths and demonstration rooms.

On the following pages are listed many of the new products to be introduced at this year's exhibition. During the past two months, we have contacted nearly all of the approximately 200 exhibitors. Although we learned the plans of most of these manufacturers, some were unable to tell all, so expect a few surprise introductions as well.

Below each exhibitor listing are two more numbers. The number on the left that reads "Circle (#)" is the number on the Rapid Facts Card that you may circle to receive additional information from that manufacturer.

If the exhibitor is advertising in this issue, the corresponding page is indicated by the blue callout reading, "See ad page number." You may obtain more information about the company and its products by referring to the ads.

Although this listing is as comprehensive as possible at press time, there are changes and additions occurring every day in exhibitor signups, products to be shown and booth assignments. Check the programs at the AES Convention to make your final plans.

To help you locate the display booths of specific exhibitors, we have provided an exhibitor map of the Los Angeles Convention Center (see page 106). The convention center's lower level includes almost 460 booths. The map of the second floor of the center (see left) includes several demonstration rooms in addition to the main demonstration areas located at the Los Angeles Hilton Hotel (see map on page 104). Manufacturer names printed in blue on the maps indicate **RE/P** advertisers in this issue.

Finally, if you are looking for particular types of pro audio equipment or services, turn to the Product Directory beginning on page 138. There are 51 categories under which companies have been listed for reference. Booth numbers have also been provided.

Acoustic Sciences Corporation
Existing products
Tube Trap acoustic treatment products.
New products
Half-round, wall-mounted Tube Trap.
Booth number: 1418
Circle (381) See ad page 171

Adams-Smith
Existing products
Time code synchronization and audio/video editing systems.
New products
Zeta Three audio, video and MIDI-capable synchronizer.
Booth numbers: 446, 448, 450, 551
Circle (382)

Advanced Music Systems
Existing products
DMX 15-80S stereo digital delay and pitch shifter; RMX-16 digital reverb; DMX 15-80SB broadcast delay; Timeflex digital pitch correction.
New products
AudioFile digital recording and editing system.
Booth numbers: 718, 720
Circle (383)

AEG Corporation
Existing products
Tape-duplicating equipment; reel-to-reel tape machines; cassette winding equipment.
New products
Loop-bin masters and slaves; cassette winders.
Booth numbers: 928, 930, 932, 934
Circle (384) See inside back cover

Agfa-Gevaert
Existing products
Reel-to-reel and cassette audio tape.

Agfa-Gevaert continued

New products
Magnetite 621, PE 627/827 (chromium oxide), PE619/819/1219 and PE 649 (iron oxide) cassette tape; PEM469 ¼-inch mastering tape.
Booth numbers: 208, 210, 309, 311
Circle (385)

Akai Digital
See International Music Company.
Circle (415)

AKG Acoustics
Existing products
Condenser and wireless microphones; headphones; stands; accessories; digital stereo synthesizers, reverbs, signal processors and delay lines.
New products
ADR-68K digital signal processing computer; C-410 condenser microphone headset; D-130NR dynamic microphones; WMS-185 wireless microphone system.
Booth numbers: 236, 238, 337, 339
Circle (386) See ad page 23

ALD Laboratory
No information received.
Booth numbers: 915, 917
Circle (387)

Alesis
Existing products
Midifex stereo digital signal processor; Space and Tonal EPROM enhancement programs; MPX MIDI patch transmitter.
Booth number: 910
Circle (388)

Allen & Heath Brenell
Existing products
CMC MkII multitrack consoles with microprocessor control; SRM stage monitor consoles; Keymix mixers.
New products
System 8 MkIII consoles for recording or sound reinforcement; SR MkII sound reinforcement consoles; Studio 12 production mixer; Sigma modular console series for sound reinforcement.
Booth numbers: 819, 821
Circle (389) See ad page 53

Alpha Audio
Existing products
Boss 8400 automated editor; PCLISTOFF off-line list management software; Sonex anechoic wedge foam; Soundtex acoustical wall fabric; Acosuticlead barrier sheet lead.
New products
Synchronizer control interface; Sonex I anechoic wedge class-A, fire-rated melamine.
Booth numbers: 508, 510, 609, 611
Circle (390) See ads pages 99, 122

Altec Lansing
Existing products
Loudspeakers, horns, drivers, signal processors and amplifiers.
Booth numbers: 822, 824
Circle (391)

Amek Consoles
Existing products
Recording and sound reinforcement consoles.
New products
Model BCII broadcast console; APC1000 assignable production console; TAC SRS9000 sound reinforcement mixer; hardware options for TAC series Scorpion console.
Booth numbers: 712, 714, 716, 813, 815, 817
Circle (392) See ad page 3

American Multimedia/Concept Design
Existing products
Custom tape-duplicating equipment.
Booth numbers: 731, 733
Circle (393)

American Recorder Technology
Existing products
Head cleaning accessories.
New products
Model 711 head cleaning fluid.
Booth number: 906
Circle (394)

Ampex
Existing products
Reel-to-reel and cassette tape, including Grand Master 456; 467 digital U-matic videocassette; 406 studio mastering tape; 600 and 672 cassette duplication tape.
New products
467 digital U-matic videocassette in 30- and 75-minute play lengths.
Booth numbers: 101-106
Circle (395) See ad page 17

Anchor Audio
Existing products
Audio monitors with extended VU or PPM metering; custom intercom systems; AN-1000 powered loudspeaker; AN-1400 powered broadcast monitor; Prota-Com headset intercom system.
New products
Liberty portable sound systems; series 200 audio modules, including DigiMax digital message repeater.
Booth number: 604
Circle (396) See ad page 162

ANT Telecommunications
Existing products
Telcom C4 noise-reduction system.
Booth numbers: 535, 537
Circle (397) See ad pages 18-19

Anvil Cases
Existing products
Transit cases for sound reinforcement consoles, related support equipment and cable reels.
Booth number: 911
Circle (398)

Apex Machine Company
Existing products
On-cassette printing machines.
New products
3-color on-cassette printing press with ultraviolet drier.
Booth numbers: 1404, 1405
Circle (399)

APG Ingenierie
Existing products
Single-, 3- and 4-driver, double-vented monitor loudspeaker systems.
Booth numbers: 921, 923
Circle (400)

Aphex Systems
Existing products
Aural Exciter psychoacoustic enhancer; Compellor compressor, leveler and peak limiter; Dominator peak processor.
New products
Model 303 combined Compellor/Aural Exciter model (mono).
Booth numbers: 626, 628
Circle (401) See ad page 119

API Audio Products
Existing products
Consoles; equalizers; signal processors.
New products
Model 5502 equalizer; model 318A distribu-

API, continued

tion amplifier; pre-amps.
Booth number: 728
Circle (402) See ad page 169

Applied Research and Technology
Existing products
Equalizers; model 190 digital delay; PRO pitch transposer; DR2a digital reverb; DR1 MIDI-capable digital reverb; model PD3 delay system.
Booth number: 746
Circle (403) See ad page 129

Apogee Sound
Existing products
3-way sound reinforcement and studio monitors, with front-end processing.
New products
AE series loudspeakers, including model 1 monitor; model 2 under balcony; model 3 small general-purpose; model 4 passive crossover; model 5 arrayable; model 6 floor monitor; and model 3X-3 3-way loudspeaker.
Booth number: 1402
Circle (404) See ad page 166

ATB SPA
No information received.
Booth number: 347
Circle (405)

Audico
Existing products
Audio and video cassette loaders, rewinders, counters and pressure-sensitive labels.
New products
Videotape cassette counter; 8mm videocassette tape loader; pressure-sensitive label sheets for VHS cassettes.
Booth number: 211
Circle (406)

Audio Analyse Instruments
Existing products
Professional electronic equipment; hi-fi sound equipment; pre-amps; amplifiers; meters.
Booth number: 919
Circle (407)

Audiocast Conseil
Existing products
Software package for scheduling, planning, resource management and budget control, using IBM PC or compatibles.
Booth number: 927
Circle (408)

Audio/Digital
Existing products
TC2, TC3, TC4 and TC5 signal processors for broadcast and effects applications.
New products
ADD-2, ADD-3, ADD-2X and ADD-3X loudspeaker time-alignment and room-synchronization systems; ADX-2000 modular loudspeaker time-alignment system.
Booth number: Hillhurst Room, Hilton Hotel.
Circle (409)

Audio Kinetics
Existing products
Q.Lock 4.10 synchronizer; Eclipse audio editor; Mastermix fader automation system; Timelink "electronic gearbox."
New products
Pacer synchronizer; new software for Eclipse audio editor.
Booth numbers: 117, 118, 119
Circle (410) See ad page 153

Audio Logic
See DOD Electronics.
Circle (416)

Audio Media Research
See Peavey Electronics.
Circle (411) See ad page 71

Audio Precision
Existing products
System One automatic test system.
New products
Wow and flutter option and switcher models for automatic testing of consoles and multitrack transports.
Booth number: 826
Circle (412) See ad page 62

Audio-Technica U.S.
Existing products
Mixing and recording consoles; dynamic and condenser microphones; mic stands, booms and cables; headphones; professional phonograph cartridges.
New products
AT4462 ENG stereo mixer; AT871 UniPlate condenser cardioid mic; AT853W UniPoint unidirectional condenser mic; AT857QMW UniPoint unidirectional condenser podium mic.
Booth numbers: 233, 235
Circle (413) See ad page 67

Audio Video Consultants
Existing products
Audio- and videocassette loaders and winders; audio duplicating, packaging and labeling equipment.
New products
Audio and video loaders.
Booth numbers: 343, 345
Circle (414)

Kenneth A. Bacon Associates
Existing products
Production, cassette duplication and packaging services; supplier of cassette duplicators, loaders and erasers.
New products
Kaba 4-track cassette duplication system; road case version of Kaba system.
Booth number: 805
Circle (421) See ad page 59

Barcus-Berry Electronics
Existing products
BBE 402 stereo signal processor.
New products
BBE 802 stereo signal processor for sound restoration applications.
Booth number: 912
Circle (422) See ad page 109

BASF
Existing products
Audio tape duplicating products; calibration tapes.
New products
Chrome loop bin mastering tape.
Booth numbers: 125, 127, 249, 251.
Circle (423) See ad page 47

B&B Systems
Existing products
AM series stereo phase oscilloscopes and metering systems; Imagescope stereo image display oscilloscope.
New products
AM-1B, AM-2B and AM-3B stereo audio phase oscilloscopes.
Booth number: 925
Circle (424) See ad page 100

BC Inc.
New products
Light-, medium- and heavy-duty shipping cases for audio equipment.
Booth number: 935
Circle (425)

Beyer Dynamic
Existing products
Ribbon, condenser, dynamic microphones in modular, lavalier, shotgun and hand-held versions; headsets and headphones.
New products
MC740 multipattern condenser mic; MC736-PV/737-PV shotgun condensers; MCE6 miniature condenser; MCE10 miniature cardioid condenser.
Booth number: 516
Circle (426) See ad page 157

BGW Systems
Existing products
Amplifiers; model 20 electronic crossover.
New products
Models SPA 1 and SPA 3 signal processing amplifiers; models 750 D&E amplifiers; model 2242 distribution amplifier.
Booth number: 201
Circle (427)

Biamp Systems
Existing products
Recording and mixing consoles; power amps; equalizers, limiters, noise gates, crossovers and reverbs.
New products
Power amplifiers and D.J. mixers.
Booth numbers: 1412, 1413
Circle (428)

Boulder Amplifiers/Silver Lake Research
Existing products
Models 500 and 160 amplifiers based on Jensen 990 op-amps.
New products
Modular pre-amplifier system with dual-servo 990CC phono circuitry; Jensen twin-servo mic pre-amp.
Booth number: 1410
Circle (429)

Brooke Siren Systems
See Klark-Teknik.
Circle (417)

Bruel & Kjaer Instruments
Existing products

They build a

And we've built a great reputation in professional audio. So if you're in the market for a digital multi-track recorder, the only logical choice is Sony.

You can be confident of our experience in professional electronics. And you can be confident of working with an established

great truck.

industry format. With over 200 Sony systems in use, compatibility isn't a problem, it's a plus.

But best of all, Sony equipment is famous for working right out of the box. So the quicker we get the box off the truck, the quicker you can be in business. SONY PRO AUDIO

Circle (64) on Rapid Facts Card

Sony Professional Audio Department, Sony Broadcast Products Company, 1600 Queen Anne Rd., Teaneck NJ 07666.
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Bruel & Kjaer, continued

Studio microphones; loudspeaker test systems; TDS4 and other instrumentation.
New products
Models 1051 and 1049 signal generators, with linear and logarithmic sweep.
Booth numbers: 240, 242, 244
Circle (430) See ad page 143

Bryston
Existing products
Power amplifiers and pre-amplifiers.
New products
DAC digital audio control amplifier.
Booth number: 800
Circle (431) See ad page 130

CAE Inc.
Existing products
Theatrical and stage lighting equipment.
New products
Littlite gooseneck lamps and accessories.
Booth number: 926
Circle (432)

Cal Switch
Existing products
Switches, wire and cable, hardware and test equipment.
New products
Jack panels and connectors.
Booth numbers: 205, 207
Circle (433)

Canare Cable
Existing products
L-4E6S Star Quad mic cable; cable reel systems; junction boxes; snakes; pigtails.
New products
MR200 series multichannel mic cable; L-4E5AT fixed installation mic cable; and L4E5C patch bay cable.
Booth number: 743
Circle (434)

Dwight Cavendish Company
Existing products
Videocassette duplicators in modular formats.
New products
Copymaster 250 videocassette duplicator, expandable up to 250 modules.
Booth number: 940
Circle (435)

Cerwin Vega
Existing products
V-43 3-way portable horn; V31C, V35C and V37C portable loudspeakers; L36PE Junior Earthquake 18-inch driver; PD18B portable playback system.
New products
DSM200 dual driver Spruce Moose horn; SSM200 single midrange driver Spruce Moose horn.
Booth numbers: Wilshire Rooms, Hilton Hotel.
Circle (436)

Cetec Gauss
New products
Loudspeakers; high-speed tape duplicating equipment.
Booth numbers: 340, 342, 441
Circle (437) See ad page 77

Cetec Ivie
Existing products
Modular sound reinforcement mixers, remote controllable consoles, crossovers, equalizers, noise masking generators, compressor/limiters, amplifiers, room combiners and third-octave real-time analyzers.
New products
Automatic mixers in modular and rack-mount versions; portable function generator.
Booth number: 644
Circle (438)

Cetec Vega
Existing products
Dynex II wireless microphone systems.
New products
Model 66B portable wireless microphone receiver.
Booth number: 350
Circle (439)

Cipher Digital
Existing products
Softouch audio editing system; Shadow II time code synchronizer; model 716A time code generator; model 710A time code reader; model 700 quad reader; series 735 time code readers.
New products
Model 750 time code reader, generator and events controller; series 4800 Shadow II synchronizer; Softouch/Softpac audio editing system.
Booth numbers: 840, 842
Circle (440) See ad page 25

CMX
Existing products
CASS I computer-assisted system with synchronization, editing and console automation features.
New products
CASS IE editor-only version providing synchronization and editing features for audio sweetening.
Booth numbers: 828, 830
Circle (441)

Community Light & Sound
Existing products
CS series loudspeaker systems; M4 midrange driver; RS/VB systems; pattern control horns.
New products
CS70 3-way loudspeaker system; SH864 and SH2064M horns for use with midrange.
Booth number: 506
Circle (442)

JL Cooper Electronics
Existing products
MidiMation series of console automation products, including Midi Mute and SAM (SMPTE Automation Manager); MSB 16/20 programmable routing system.
New products
SAM disk storage device for SAM; MAGI (Mixer Automation Gain Interface); ARNI (Automated Routing Network Interface, an automated pageboy); MSB Plus 8-by-8 programmable MIDI switcher.
Booth number: 944
Circle (443) See ad page 163

Crest Audio
Existing products
Power amplifiers.
New products
Power amplifier with built-in limiters on each channel.
Booth numbers: 617, 619
Circle (444)

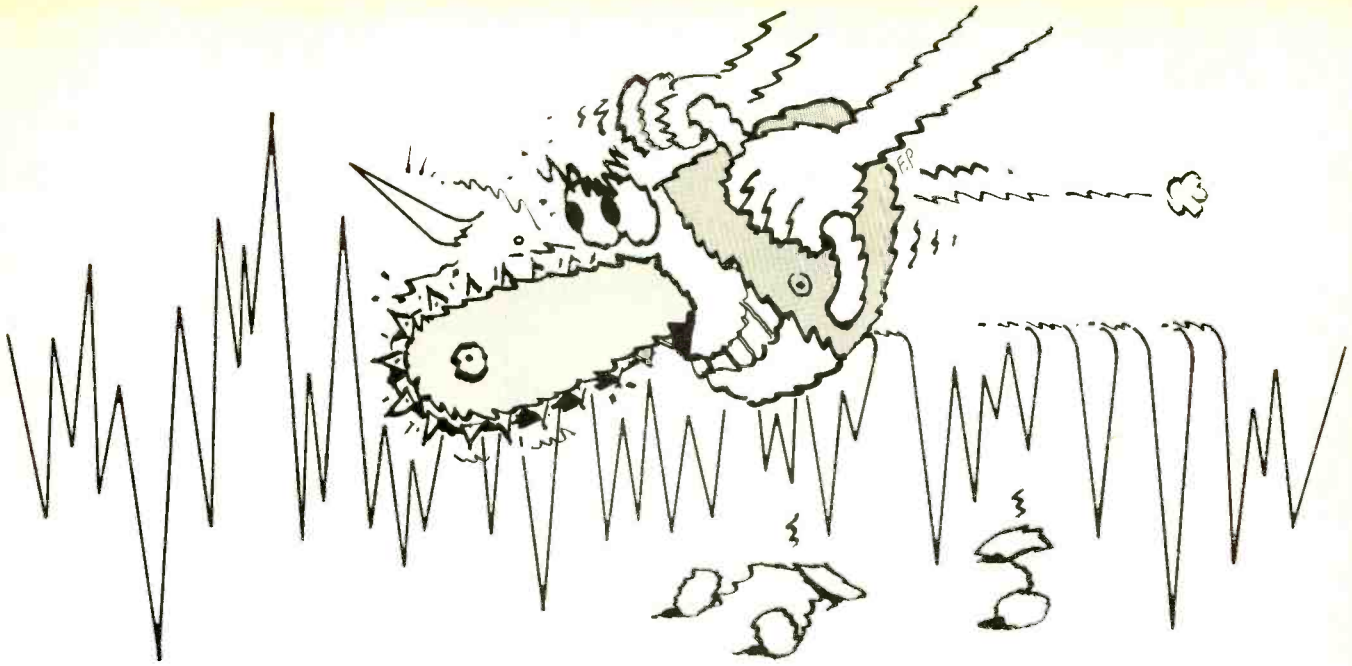
Crown International
Existing products
Industrial and power amplifiers; power supplies; microphones.
New products
Broadcast lavelier and miniature microphones; Power Base 1 amplifier.
Booth numbers: 227, 299, 231
Circle (445) See ad pages 132-133

CST Sales
New products
Audio cassette labeling equipment.
Booth number: 1403
Circle (446)

dbx
Existing products
Compressors; modular series limiters, de-essers, parametric equalizers, noise gates and noise reduction units.
New products
Model 150X 2-channel, Type-1 noise-reduction unit; real-time analyzer.
Booth numbers: 121, 122, 123, 241, 243, 245
Circle (451) See ad page 108

DCS Audio Products
New products
DAP-1 user-programmable digital effects processor with reverb, loop-editing and delay functions; Effects Development System software.
Booth number: 602
Circle (452)

Design Direct Sound
Existing products
October Box, DXB 15H, PM2-65, PM2-110 and P1-100 continuous frequency distribution horns.
New products
CFD1-90, CFS 1-51, series 80, series 40, MB and OM-Tracks stereo mixing boards; 1-inch horns.
Booth number: 630
Circle (453)



DOES YOUR LIMITER MASSACRE YOUR SOUND?

Most limiters are far from perfect...literally chopping up your sound. The new Aphex *Dominator*™ is the perfect solution.

Unlike dumb, over-threshold devices, the *Dominator* is an intelligent 3-band limiter with a proprietary circuit which varies the threshold for limiting. The result is an *absolute* peak ceiling while maintaining a transparent sound. Selectable crossover frequencies, plus high and low frequency drive controls allow creative flexibility.

Aphex Studio *Dominator*

- Provides Absolute Peak Ceiling.
- Total transparency below processing threshold.
- Increased loudness
- Freedom from spectral gain intermodulation
- Minimal loss of transient feel
- High density capability
- Flexible—easy to use
- Multiple applications
- Made in U.S.A.

You can run hotter levels to maximize signal-to-noise levels without fear of overloading your recording or transfer mediums.

The *Dominator* is ideal for *any* situation where clipping is a problem, such as digital audio, disc mastering, video post production and film.

Stop massacring your sound. Ask your audio professional for a free trial of the Aphex *Dominator.* Once you've heard it, you'll never be satisfied with your old limiters.

Aphex Systems Ltd.
13340 Saticoy Street • North Hollywood, Ca 91605 • (818) 765-2212 • TWX: 910-321-5762

Dominator is a trademark of Aphex Systems Ltd.

Circle (65) on Rapid Facts Card

Dialight/Kulka Smith
Existing products
X and D series XLR-type receptacles; phone plugs and jacks; goosenecks; transformers; ac connectors.
New products
Neutricon-A minicircular connector; waterproof XLR-type connector; HH Smth banana plugs; oscilloscope probes.
Booth number: 951
Circle (454)

Digital Creations
Existing products
Diskmix console automation system, compatible with Sound Workshop ARMS II automation system.
New products
Fader retrofit package.
Booth numbers: 445, 447
Circle (455)

Digital Dispatch
Existing products
Rental of equipment for recording and film post-production, including multitracks, synthesizers, microphones and signal processors.
Booth numbers: 801, 803
Circle (456)

DigiTech
See DOD Electronics.
Circle (477)

DOD Electronics
Existing products
Graphic equalizers; rack-mount mixers; digital room delays; compressor/limiters; reverb units, psychoacoustic processors; crossovers; distributor of Audio Logic and DigiTech products.
New products
R-231 dual-channel graphic equalizer; RDS6500 digital reverberation system; Audio Logic SC31 single-channel third-octave graphic equalizer; SC30 2-channel graphic equalizer; and RD3 digital room delay.
Booth numbers: 646, 648, 650
Circle (457) See ad page 7

Dolby Laboratories
Existing products
A-Type, B-Type and C-Type single-through-multichannel noise reduction systems; Soundlink digital audio system.
New products
Spectral Recording noise reduction systems.
Booth numbers: 222, 224, 323, 325
Circle (458) See ad page 28-29

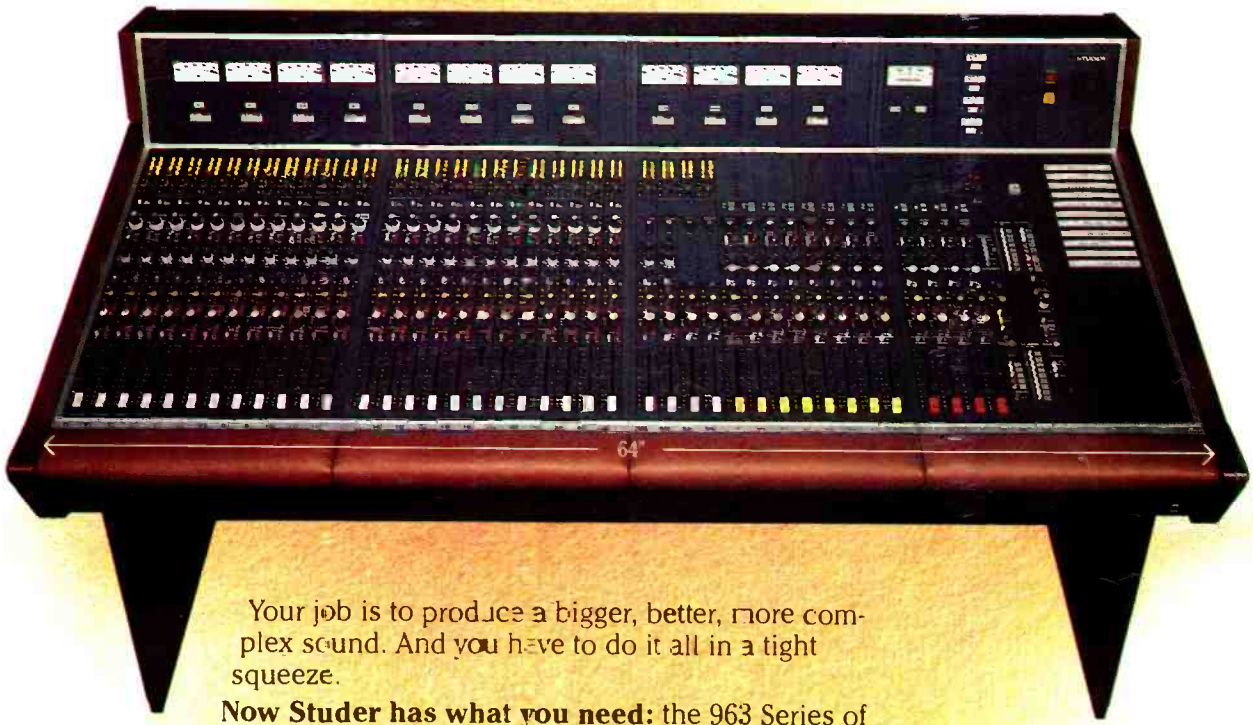
Dorrough Electronics
Existing products
Model 40-A LED loudness meter with simultaneous peak and rms level.
Booth number: 745
Circle (459) See ad page 168

D&R Electronics
Existing products
Mixing consoles; signal processors.
New products
Dayner multitrack console.
Booth numbers: 747, 749
Circle (460)

Educational Electronics Corporation
Existing products
Distributors of Sony high-speed tape cassette duplicators, including CCP-110 mono, CCP-200 4-track stereo and CCP-300 4-track stereo units.
Booth number: 904
Circle (461)

Not-So-Big News

The news is out. Studer's new 963 is big on features, performance and reliability. And not-so-big on size.



Your job is to produce a bigger, better, more complex sound. And you have to do it all in a tight squeeze.

Now Studer has what you need: the 963 Series of compact production consoles. A 963 is ideal for video post-production, video editing, broadcast production, EFP vehicles, smaller recording studios—anyplace where quality and reliability are critical but space is at a premium.

Based on a standard 30 mm module width, the 963 is available in configurations from 16 to 40 inputs. A 28 input console, with 28 direct outputs plus 4 stereo subgroups and 2 stereo masters, is barely more than 5 feet long. A 40 input console, is barely more than 6 feet long.

Standard features on the 963 include balanced insert points, direct outputs, a bantam jack patch bay, and external mute interface for video switchers. A wide variety of module options lets you custom configure your 963 for practically any specialized application.

When it comes to audio performance, the 963 goes head-to-head with the bulkiest of the big-name boards. Noise levels are digital compatible in "real world" conditions with many open faders. Studer engineers gave special attention to mix bus design and reference grounding to assure consistently superior specifications regardless of frame size. For extra reliability, solid state switching is used in all but critical audio paths.

As with all Studer products, the 963 is manufactured and assembled to the highest standards of Swiss craftsmanship.

For more information, call your nearest Studer representative. Find out how the 963 can give you big console capabilities in a not-so-big package.

STUDER REVOX AMERICA, INC.
1425 Elm Hill Pike, Nashville, TN 37210. (615) 254-5651

Offices: Los Angeles (818) 780-4234/New York (212) 255-4462/Chicago (312) 526-1660
Dallas (214) 943-2239/San Francisco (415) 930-9866

STUDER REVOX

Circle (3) on Rapid Facts Card

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

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

The M-21 is microprocessor controlled and user-programmable for any 2 of 4 speeds. It is a totally self-contained package with no external power supplies or cabling, and access to all components for maintenance and alignment is quick and easy. The performance specifications are unexcelled.

It's only natural that the M-21 should be such a fine machine. After all, we invented the modern tape recorder over 50 years ago. To arrange for a free demonstration at your facility, or for information on any of our other high technology products, please give us a call.

In Canada:
AEG BAYLY INC.
167 Hunt Street
Ajax, Ontario L1S 1P6
(416) 683-8200

In U.S.A.:
AEG Corporation
Route 22 — Orr Drive
P.O. Box 3800
Somerville, NJ
08876-1269
(201) 722-9800

AEG

Circle (2) on Rapid Facts Card

Ad index

Advertiser	Page Number	Rapid Facts Number	Hotline
Acoustic Sciences	171	111	800/ASC-TUBE
AEG Corp.	IBC	2	201/722-9800
AKG Acoustics, Inc.	23		203/348-2121
AKIA Electronics	134	77	305/245-2727
Allen & Heath Brenell	53	27	203/795-3594
Alpha Audio	122	67	804/358-3852
Alpha Audio Automation Systems	99	52	804/358-3852
Amek Consoles Inc.	3	4	818/508-9788
Ampex Corp.	17	10	415/367-3809
Anchor Audio, Inc.	162	97	213/533-5984
ANT Telecommunications	18-19	11	301/670-9777
Aphex Systems Ltd.	119	65	818/765-2212
API Audio Products, Inc.	169	108	703/455-8188
Apogee Sound Inc.	166	102	707/778-8887
Applied Research & Technology	129	73	716/436-2720
Atlas Sound	140	22	201/887-7800
Audio Engineering Associates	150	88	213/684-4461
Audio Engineering Society	149		212/661-2355
Audio Kinetics Ltd.	153	91	201/671-8668
Audio Logic	7	6	801/268-8400
Audio Media Research	71	36	601/635-2244
Audio Precision	62	32	800/231-7350
Audio-Technica U.S., Inc.	67	34	216/686-2600
B&B Systems	100	53	800/345-1256
Bacon, Kenneth Assoc.	59	30	800/231-TAPE
Barcus Berry Electronics, Inc.	109	58	800/233-8346
BASF	47	24	800/225-4350
Bertech Organization	170	110	800/992-2272
Beyer Dynamic, Inc.	157	94	516/935-8000
Black Audio Devices	173	116	818/507-8785
Bruel & Kjaer Instruments, Inc.	143	83	617/481-7000
Bryston/Vermont	130	74	802/223-6159
Burlington Audio/Video Tape Inc.	115	63	800/331-3191
Canare Cable Inc.	168	106	
Cetec Gauss	77	40	213/875-1900
Cipher Digital Inc.	25	13	301/695-0200
Circuit Research Labs Inc.	76	39	602/438-0888
Cooper, J.L. Electronics	163	99	213/473-8771
Countryman Associates	170	109	415/364-9988
Crown International	132-133	76	219/294-8000
CSE Audio	148	86	716/436-9830
dbx	108	125	617/964-3210
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Dolby	28-29	15	415/558-0200
Dorrough Electronics, Inc.	168	105	818/999-1132
Eastern Acoustic Works	123	68	617/620-1478
Eastern Acoustic Works	125	69	617/620-1478
Eastern Acoustic Works	127	71	617/620-1478
Editron	160	120	213/275-1790
Electric Valve Communication Corp.	164	100	
Ensoniq Corp.	69	35	
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Everything Audio	55	28	818/995-4175
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Future Disc Systems	171	112	213/876-8733
Hardy Co.	24	90	312/864-8060
HY James	89	47	313/471-0027
Integrated Innovations	130	118	
Intl. Tapetronics Corp/3M	57	29	800/447-0414
JBL Professional	105	57	
Jensen Transformers	61	31	213/876-0059
JRF Magnetic Sciences, Inc.	144	81	201/398-7426
Karlberg Enterprises	102	55	815/968-2022
Klark-Teknik Electronics Inc.	135	78	616/249-3660
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LD Systems, Inc.	165	101	713/695-9400
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LT Sound	167		800/241-3005
3M	36-37	18	
Manny's Music	83	43	212/819-0576
Martin Audio Video Corp.	93	49	212/541-5900
Meyer Sound Labs	75	38	
Micro Audio	142	89	503/292-8896
Musically Intelligent Devices Inc.	136	79	516/864-1683
Nakamichi USA Corp.	39	19	213/538-8150
Neotek Corp.	9	7	312/929-6699
New England Digital	72-73	37	416/868-0528
Ocean Audio Inc.	86	45	213/459-2743
Orban Associates Inc.	126	70	415/957-1067
Orban Associates Inc.	78	41	415/957-1067
Otari Corp.	79	42	415/592-8311
Panasonic (Ramsa Div.)	35	17	
Pierce Arrow	139	80	312/328-8950
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Renkus-Heinz, Inc.	159	95	714/250-0166
Roland Corp. US	87	46	
Royal Recorders	21	12	414/248-9100
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Schoeps, Gmbh	155	92	
Skyelabs, Inc.	112	60	302/697-6226
Solid State Logic	120-121		212/315-1111
Sony Broadcast Products Co.	116-117	64	
Sound & Vision	166	119	805/496-1518
Soundcraft USA	45		818/893-4351
Soundmaster Intl.	147	85	416/741-4034
Soundtracs, Inc.	63	33	
Sprague Magnetics Inc.	172	113	818/994-6602
Standard Tape Laboratory, Inc.	128	72	415/786-3546
Studer Revox/America	BC	3	615/254-5651
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Symetrix	124	20	206/282-2555
Tannoy North America Inc.	85	44	519/745-1158
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Telex Communications, Inc.	95	50	800/328-3771
Trebas Institute of Recording Arts of U.S.A., Inc.	122	66	213/467-6800
Trident USA Inc.	103	56	213/933-7555
Trident USA Inc.	IFC-1	1	213/933-7555
Troisi EDC	169	107	617/692-7768
Turbosound Inc.	91	48	212/460-9940
United Tape Group	174	117	818/980-6700
UREI	105	57	
Vertigo Recording Services	162	98	818/769-5232
Westlake Audio	11		213/851-9800
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New products

23 series consoles from DAX

Available with either 12 or 16 inputs, the consoles offer balanced mic input channels, line-level inputs, input preamp, peak LED indicators, monitor, built-in reverb, effects send and 3-band EQ.

Master control features include switchable metering, panable effects return and reverb, output faders for subgroups 1 and 2, monitor master and main. Rear-panel connections include both balanced XLR-type and unbalanced, 1/4-inch line-level main outputs, plus patch facilities for both subgroup 1 and 2.

Circle (359) on Rapid Facts Card

EAW model SM202P stage monitor system

The units use dual 10-inch drivers for higher output and more gain before feedback, the company says.

An RCF N481 HF compression driver and optimized coverage pattern horn combine with dual low-frequency drivers to offer a quoted frequency response of 150Hz to 120kHz, ±2dB, without external equalization.

The system has a claimed 103dB SPL sensitivity rating at 1W, 1m, and 129dB at 1m maximum output. In addition, the smaller diameter diaphragms are said to provide a response over 800Hz.

Circle (360) on Rapid Facts Card

Apogee model A-2 loudspeaker

The trapezoid-shaped 2-way enclosure is said to be intended for operation where conventional monitors are physically too large. For example, as a balcony fills, the unit can be positioned without affecting sight lines.

Specifications include 65Hz to 19kHz, ±4dB frequency response; 114dB continuous SPL, 120dB peak at 1 meter; and 100W continuous, 300W peak power rating.

The model A-2 is constructed from Finland birch and has an expanded steel grill and protective steel trim.

Circle (361) on Rapid Facts Card

Network CD music library

The library includes contemporary music adaptable to commercials, feature, industrial and educational films, as well as other multimedia productions. It offers various versions of existing music, including 29- and 59-second broadcast-length edits, tags and rhythm tracks.

Circle (362) on Rapid Facts Card

Yamaha MZ series dynamic microphones

The series includes five unidirectional (cardioid) dynamic models. All MZ series mics feature 3-point push-pull suspension to isolate the mic element. Silkscreen photo-engraving technology is said to enable the internal acoustic resistance of each mic to be established for stereo imaging. Photo-engraved acoustic resistors are also said to provide smooth on- and off-axis response, making it easier to control acoustic feedback.

The models 102Be, 103Be and 105Be utilize triple-laminated diaphragms with beryllium domes, offering a quoted HF response to 18kHz. A polyethylene film substrate laminated to the edge of the diaphragm is said to provide controlled damping on the models 101 and 104 mics.

All microphones incorporate protective metal-mesh grilles. Multiple foam layers inside each mic head are said to afford built-in moisture and vocal pop protection; slip-on wind screens are also available.

Circle (349) on Rapid Facts Card

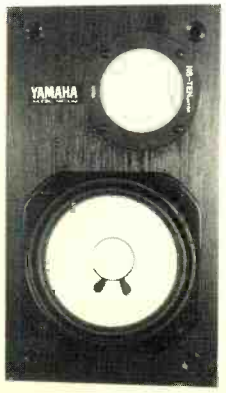
NS-TENuator HF attenuator from Miller Audio

Designed for Yamaha NS-10M close-field monitors, the device holds a layer of tissue paper or other material against the tweeters via a circular steel frame. Using existing mounting holes, the device can be installed or removed with a screwdriver.

[For a detailed explanation of how a tissue paper layer affects the NS-10M's high-end response, see "Examining the Yamaha NS-10M 'Tissue Paper' Phenomenon,' by Bob Modas, published in the February 1986 issue of **RE/P**—*Editor.*]

The NS-TENuator is finished in jet black.

Circle (357) on Rapid Facts Card R-E/P



New products

FilteRing anti-pop filter from Miller Audio

The steel-framed pop filter is designed to replace the "stocking on a coat hanger" microphone screen. It uses a standard mic stand mount and a steel frame holds replaceable nylon filters.

The device also accepts custom filters made by anyone who wishes to use new or time-tested materials. By cutting a simple pattern, many materials can easily adapt to this frame, the company says.

Circle (346) on Rapid Facts Card

Sonex 1 acoustic material from Alpha Audio

Laboratory tests are reported to have shown that uncoated Sonex 1 meets Class 1 requirements for both flame spread and smoke density, while still retaining the anechoic wedge properties for noise reduction.

The acoustic material is available in 2′ x4′ panels, with a depth of 2 inches.

Circle (366) on Rapid Facts Card

Trident series 65 console

The recording console series is available in six frame sizes and features 16 discrete output buses with 16-track monitoring.

Available in 16- through 56-input configurations, the console features separate mic and line inputs with phase reversal, 4-band EQ with variable hi-pass filter, eight auxiliary sends, auto muting, solo-in-place, monitor-fader reverse, direct outputs and four echo returns.

Circle (347) on Rapid Facts Card

E-V SH-1810 loudspeaker system

The 3-way horn-loaded speaker system incorporates a vented-box design and 10-inch cone transducer equipped with proprietary phase plug for a quoted power response of 250Hz to 2.5kHz. A 60°x40° constant-directivity horn and a 1-inch throat, titanium-diaphragm compression driver offers a quoted frequency response to beyond 20kHz.

A solid-state driver protector is said to respond to excessive power levels, attenuating driver input to avoid diaphragm damage, and then automatically resetting when power returns to safe levels.

The system has a quoted overall frequency range of 35Hz to 23kHz, sensitivity rating of 105dB at 1W, 1m and a peak output SPL of 135dB. The system can be operated in 3-way passive or biamp modes, with power handling capacity of 300W and 700W, respectively.

Circle (365) on Rapid Facts Card

New products

Midifex digital signal processor from Alesis

Midifex employs multiple delay lines, multiple high-pass, low-pass and band-pass filters, and stereo digital reverb capabilities. A total of 63 pre-programmed effects are available via front-panel push-buttons or external MIDI control, including stereo generation, multiple delays, echo, ambience generation, multitap panning and reverb.

A separate MPX MIDI patch transmitter also offers remote control of program changes on MIDI-based devices that can receive patch-change data. The unit is said to transmit on all 16 MIDI channels, thus permitting grouping of MIDI devices by individual MIDI channels.

Circle (356) on Rapid Facts Card

Rane GE30 third-octave equalizer

The unit provides switchable +12/−15dB boost/cut mode, or a 20dB cut-only mode via a rear-mounted push-button. It is described as the first equalizer to feature a user-switchable active direct-coupled or transformer-coupled balanced output as standard.

Other features include sweepable ultrasonic and subsonic filters, built-in RF filters, line-surge protectors, fail-safe automatic by-pass, gain control and both 3-pin and barrier-strip input and output terminations.

Circle (364) on Rapid Facts Card

Yamaha model KM802 portable keyboard mixer

Two of eight input channels feature a pre-amp, 5-band EQ section and a single effects-send control and can be used for microphone and keyboard input. The other six channels are optimized for line-level instrument signals and feature three effect sends. All channels include a recessed, straight-line fader and stereo pan control.

To enable simultaneous connection to one or more external processors, and for reassignment without patching, the mixer has three sends and three stereo returns.

Separate left and right master faders and a separate headphone output are featured.

Technical specifications include −122dBm EIN; −90dBu residual output noise; frequency response of 20Hz to 20kHz, +1/−2dB; and a THD of 0.05%.

Circle (352) on Rapid Facts Card

New products

Rebis model RA701
MIDI and noise gate

By using a single MIDI Out socket, the RA701 has two channels to send on/off, key number and velocity data. It can be chained via a merge input, and voices are selected using key number reallocation on a receiving drum machine.

Technical specifications include a 90dB dynamic range, frequency-conscious side chain and 4-stage envelope. Regardless of signal length for fixed envelope shaping, hold and release parameters can be switched to follow attack.

Up to two seconds of envelope delay can be used for slapback effects, particularly on reverb; up to four seconds can be used in trigger-mask mode to remove off-beats or change-accent levels. Also, stereo and duck facilities, external key inputs and 5-volt trigger outputs are provided.

Circle (351) on Rapid Facts Card

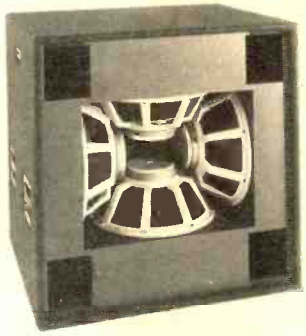
E-V MTL-4
low-frequency monitor system

The enclosure is a vented-box design that contains four 18-inch loudspeakers, each facing into a manifold chamber at the cabinet's center. The design increases acoustic loading, resulting in increased low-frequency efficiency, reduced distortion and 2 to 3dB higher efficiency in the 40Hz to 80Hz region.

In addition, the system is said to enable more than 12% more efficiency over the 40Hz to 225Hz frequency range. Each enclosure has a long-term input power capacity of 1,600W, with an instantaneous peak power rating of 6,400W. These features provide attainable SPLs in excess of 134dB continuously at 1 meter, with 140dB peak capability.

A 30,000W MTL-4 low-frequency system will pack into a 6-foot length of a standard truck floor.

Circle (343) on Rapid Facts Card



New products

Beyer M380 dynamic microphone

Designed for instrument recording and sound reinforcement applications, the M380 features a figure-eight polar pattern, quoted frequency range of 15Hz to 20kHZ, and a claimed SPL measurement of 140dB without overload or distortion.

The unit's moving-coil element is constructed of Mylar. A zinc die-cast casing has a brushed bronze finish and integral shock-mount system to reduce vibration and noise from the stage or mic stand. A built-in wind screen is also provided.

Circle (341) on Rapid Facts Card

Yamaha model S300 loudspeaker system

The 3-way system is designed for small- to medium-size venues, general sound reinforcement rigs and a variety of close-field reinforcement applications. The system has a quoted 90° vertical by 120° horizontal dispersion characteristic throughout its nominal frequency range of 50Hz to 20kHz.

The S300 has a built-in crossover network that allows the system to be connected to a single power amplifier. A 15-inch cone is housed in an optimally tuned bass reflex enclosure. An 8-inch direct radiating, HF compression driver works into a flush-mounted horn. An HF output trim control is provided.

The system is said to exhibit 97dB sensitivity at 1W, 1m. Rated at 200W program power (400W peak), the S300 is claimed to deliver 120dB, SPL continuously.

Circle (350) on Rapid Facts Card

Ross model R12 stereo paragraph EQ

The combined graphic and parametric equalizer features 12 bands per stereo channel, with each band having three selectable 1/4-octave center frequencies.

It has been constructed using low-noise op-amps and a regulated power supply to deliver a quoted frequency range of 10Hz to 65kHz, a THD of 0.02% and a S/N ratio of 100dB.

Circle (353) on Rapid Facts Card

New products

The Hook from Techworks

Constructed from solid steel, the device has an adhesive-backed fabric fastener for attachment to tape machines, effects racks, consoles, desks or other smooth surfaces. It is designed to hold a reel of 1/4-inch leader tape during editing sessions. The Velcro-style fasteners are said to allow quick movement of the device from one location to another.

The unit can be rotated 90° to serve as a spindle for positioning tape reels directly in the tape path, a feature that is said to enable the unreeling of leader tape and also to facilitate handling of machine-to-machine tape loops.

Circle (355) on Rapid Facts Card

Ampex 467 digital U-matic cassettes

The 75-minute play length cassette can be used for Compact Disc mastering, while the 30-minute length can be used for single-song and EP mastering.

The company reports that it uses the correction capability of PCM converters as a final criterion for qualification of the 467 cassettes.

Circle (348) on Rapid Facts Card

Allen & Heath Brenell Sigma consoles

Based on the company's CMC and Syncon series, the new 24-bus modular system is available in two frame sizes and a number of formats.

Input and group/monitor modules feature a 4-band parametric EQ section and six auxiliary sends, enabling group modules with fader reverse to be used as full-function inputs during mixdown. Two independent foldback systems with multisourcing are also provided, while signal monitoring includes stereo solo-in-place, PFL and AFL.

Auto-muting can be achieved via a Sigma Series Remote (SSR) control programmer, which offers MIDI implementation for pre-programmed muting sequences or synchronization using Song Pointers. Up to six different MIDI-controlled effects devices can be changed for each sequence event.

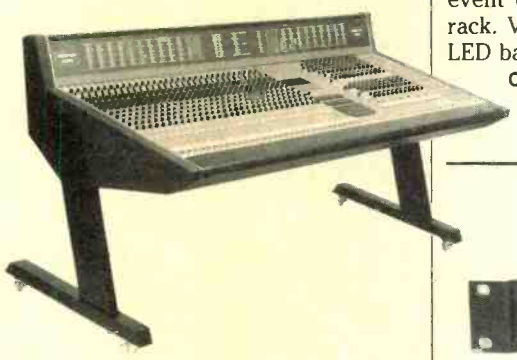
Circle (368) on Rapid Facts Card

New products

input. Simultaneous access to aux master and echo return is provided via a 24-way keypad, which allows level adjustment, EQ selection and pan control.

The first 40-input Di-An console is scheduled for delivery to Keith Olsen's Goodnight LA Studios, Los Angeles, in early 1987.

Circle (370) on Rapid Facts Card



Soundtracs CP6800 console

Designed for recording and post-production, the CP6800 features analog circuitry and programmable digital routing that may be interfaced with existing synchronizer units.

Available in a number of configurations, with or without patchbay, the console is equipped with an internal microprocessor and video monitor, disk storage and time code reader. An 8-way event controller is housed in a remote rack. Versions are available with VU or LED bargraph metering.

Circle (340) on Rapid Facts Card

JBL model 6215 power amplifier

Occupying a single rack space, the unit is designed for powering cue headphones and small speakers.

Output power is a quoted 35W per channel into 8Ω, 45W per channel into 4Ω, and 90W into 8Ω mono-bridge mode.

To reduce audible distortion, the company has used a low transient intermodulation (TIM) design that utilizes predriver-stage local feedback. Minimal overall negative feedback is required to set overall circuit operating points.

Input terminations use a standard 1/4-inch TRS, XL-type connector or barrier strip; output terminations use 5-way binding posts.

Circle (345) on Rapid Facts Card

Circle (104) on Rapid Facts Card

New products

Trident Di-An digitally controlled console

Configured as a digitally controlled console with analog circuitry, the console is said to be the first to store and recall every major console function, from microphone input to group output.

Di-An has the ability to access up to 128 console configurations, which can then be triggered via time code from a multitrack machine. An internal computer can access 1Mbyte of RAM, allowing events to be stored and recalled in real time, or archived to floppy disk. The console will accommodate various fader-automation systems, including VCA-based and moving-fader systems with independent disk-storage options.

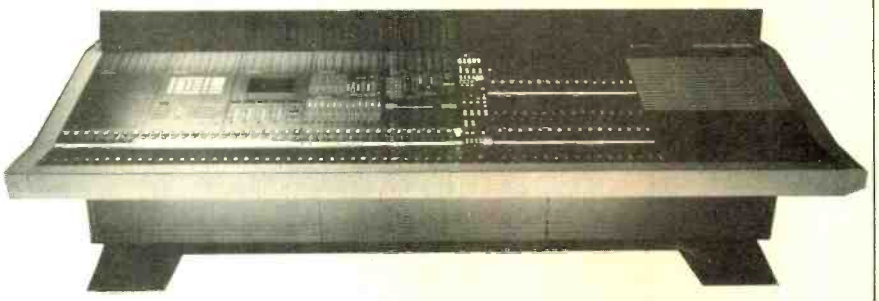
Each input strip has eight inputs, providing channel access, mono after-fade and pre-fade listen, stereo solo, mute and automate. In addition, 24 auxiliary sends can be configured as 12 stereo pairs. A 4-character alphanumeric display, one per input, displays assignments from memory or loaded from disk.

An auto-gain facility on each channel tracks the input signal, and adjusts amplifier gain to within 5dB of peak. Input, group/tape and stereo mixdown buses are monitored by digitally controlled, 100-segment bargraph displays, switchable between VU or PPM characteristics.

Routing functions are configured in eight sections, consisting of LED displays that indicate to which group an input is routed and/or selected, plus phase reverse, EQ and dynamics.

The equalizer has six sections, four of which consist of a 16-frequency sweep, with digital display of center frequency and boost/cut buttons with bargraph display, 3-position bandwidth control, and individual bypass keys. The last two sections are swept high- and low-pass filters, which also have 16 turnover frequencies and a digital display of each range showing a 12dB per octave slope. Memory keys allow each input of the console to store up to four different EQ settings.

The auxiliary-send panel contains 12 sections for level control, pre/post fader selection, mute and re-routing capabilities to enable up to 24 sends from each



Studio update

The Village Recorder (West Los Angeles, CA) has completed a large-scale upgrade of Studio A. Centerpiece of the remodeled studio is a 40-input **Solid State Logic SL4000-E** console. Principle monitors comprise a pair of bi-amped **Vince Van Haaf JBL/TAD** systems, driven by **Crown PSA-2** and **Macintosh model 2105** power amplifiers. Room EQ is provided by a **White model 4400** equalizer.

Studio A's standard multitrack compliment (located in a dedicated machine room adjacent to the control room) consists of a **Studer A-800 Mk II** and an **Otari MTR-90 Mk II,** which can be interlocked via a **Timeline Lynx** synchronizer module. **Ampex ATR-100** 2-and 4-tracks handle mastering duties.

Also: **Nick Smerigan** has been named as the studio's new executive director. He served previously as vice president and general manager of the LA Record Plant, and was formerly studio manager of A&M Recording, Hollywood. *1616 Butler Ave., West Los Angeles, CA 90025; 213-478-8227.*

Northern California

Prairie Sun (Cotati) has installed a **Publison Infernal Machine 90,** equipped with 21-second-per-channel multisampling. *P.O. Box 7084, Cotati, CA 94928; 707-791-7011.*

Live Oak Studio (Berkeley) has expanded its services for the video and film industries with the addition of an **Audio Kinetics Q.Lock 4.10-E Eclipse** editor system. *442A Walnut St., Berkeley, CA 94009; 415-540-0177.*

Northwest

Spectrum Studios (Portland, OR), has added three **Yamaha REV-7** digital reverb units, three **ADR FX-769-R Vocal Stressors,** three **UREI LA-3A** compressors and a **Nagra IV-STC** for on-location sound recording and playback.

Personnel changes include the addition of two new in-house engineers: **Mike Moore,** who joins the studio after five years as an independent engineer in the Portland area, with recent projects including John Fahey and Nu Shooz; and **Jeff Dennerline,** a recent graduate from Mt. Hood Community College's video production program. *905 S.W. Alder, Portland, OR 97250; 503-248-0248.*

Lahania Sound Recording Studio (Maui, HI), owned by **George Benson,** has added a **Sony PCM-3324** DASH-format recorder, linked to the facility's existing **Solid State Logic SL4000-E** console. Outboard gear comprises an **EMT 251** digital reverb, **EMT 140** stereo tube plate, **Lexicon Super Prime Time, Eventide Harmonizer** and **Flanger, UREI 1176LN** and **LA-2A** limiters. New additions include four **dbx model 160X** limiters, a **Lexicon PCM-70** and two **Yamaha SPX-90** effects units, an **Akai S612** sampler and an **AKG Gold Tube** mic. *Lahania Square, H-2, Lahania, Maui, HI 96761; 808-667-2587.*

Studio update

Music Mill (Nashville) has added two new PD-format **Mitsubishi X-850** 32-channel digital recorders, according to **Jim Cotton,** the studio's chief engineer. "We decided on the X-850s after many of our clients began renting the decks on a regular basis from Digital Assoicates and Audioforce Nashville," he said. "But first, we tested the recorder for its sound and for its usability and, in our opinion, the Mitsubishi was a hands-down favorite." *1710 Roy Acuff Place, Nashville, TN 37203; 615-254-5925.*

Dallas Sound Lab (Irving, TX) has added a **Yamaha REV-7** and **Lexicon PCM-70** digital processors to complement its current inventory of **AMS RMX-6, Quantec** and **Lexicon Room Simulator model 224** outboards. "More of our clients are requesting these units because they have been exposed to them in other studios around the world," says **Johnny Marshall,** studio manager. "We would like to be able to provide them with the same amenities they are used to elsewhere."

In addition, the facility's **AMS DMX-1580S** has been upgraded to include stereo sampling with reverse playback. Sample-to-disk with the latest software revisions have also been installed in the studio's **Linn 9000** MIDI-capable drum machine and sequencer. *Three Dallas Communications Complex, 6311 North O'Connor, Suite 246, Irving, TX 75039-3510; 214-869-7657.*

Southern California

Juniper APV (Burbank) has added an **Adams-Smith model 2600** interlock system with three synchronizer modules; a **Sony 5850** U-matic with address track edit modifications; an Otari **MX-5050 MkIII** half-inch 4-track; a **Sony JH-110C** 2-track; and an **Adams-Smith** time code generator with jam sync. The studio has also added a **Kurzweil 250** digital synthesizer to its stock of available rentals. *719 Main St., Burbank, CA 91506; 818-841-1249.*

Fred Jones Recording Services (Hollywood) has added a second audio post for film and a 1-inch video sweetening room, and installed an **Editron** time code synchronization system. In addition to existing **Cipher Digital/BTX** synchronization to film and video, the facility also offers 35mm mag transfers, digital sampling keyboards and a real-time tape copy service. *6565 Sunset Blvd., Suite 211, Hollywood, CA 90028-8521; 213-467-4122.*

Warner/Hollywood Studios (Hollywood) has installed an automated **Harrison PP-1** post-production console in Studio D. The new board replaces a smaller PP-1 that is destined for installation in Studio C. Wired for 72 inputs and 24 submaster modules, with its integral 24-input pre-dub submixer, the console has a total input capacity of 96 dubber channels. *1041 N. Formosa, Los Angeles, CA 90046; 213-850-2948.* R-E/P

To be included in a future issue of RE/P, send your latest news of equipment and facility upgrades and expansions, services and personnel changes to:
Studio Update, **Recording Engineer/Producer,** 1850 Whitley Avenue, Suite 220, Hollywood, CA 90028.
Or via IMC EMail to REP-US, or FAX to 213-856-4985.

Studio update

Teknik DN780, Yamaha REV-7 and **Lexicon PCM-60** digital reverbs. An **EMT 140** stereo plate is also available for either room. *5761 Park Plaza Court, Indianapolis, IN 46220; 317-845-1980.*

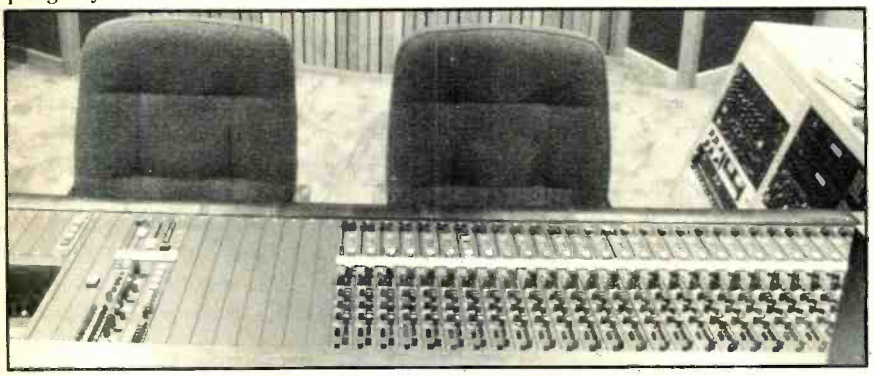
Barn Burn (Annawan, IL) is a new 24-track, MIDI-based studio. Using a Syn-Aud-Con design from **Don Davis** and acoustic consulting by **Jerry Milam,** the studio was constructed in a barn and comprises a 21'x20' control room and 20'x19' studio.

The CR features a 28-input **MCI JH-636** console, with **Sound Workshop ARMS** tape-based automation, linked to an **MCI JH-24** 24-track, a **Sony APR-5002** 2-track, an **Otari MX-5050B** 2-track and **Tascam 22-4** 4-tracks. Outboard gear includes two **Yamaha REV-7** digital reverbs; two **dbx model 905** parametric EQs, **model 902** de-essers, **model 903** and **model 160X** complimiters; a **UREI 1178** stereo limiter; two **ADA** digital delays; two **Omnicraft GT-4** noise gates; and a **Lexicon PCM-60** digital reverb.

Studio manager **Roxanne Heath** says that the new facility is geared toward a wide range of album-oriented productions, in addition to commercial and jingle recordings. *1500 North, Annawan, IL 61234; 309-935-6181.*

South Central

Audio Creations (Paducah, KY) recently added a **Kurzweil 250** digital sampling keyboard with the new **Version 3** software, 50kHz sampling and additional **Soundblock A.** Also added: a new **Neotek series II** 28/24 console; a **Yamaha SPX-90** digital processor: **Lexicon model 200** reverb; **Gold Line GL-30** RTA; **Aphex B; Orban 424; Lexicon PCM-42; Roland SDE-3000** DDL; **Yamaha NS-10s; Fostex RM-780** and **Electro-Voice Sentry 100A** monitors. *4815 Clarks River Road, Paducah, KY 42001; 502-898-5746.*



Barn Burn: new 24-track MIDI studio.

Studio update

been modified to be used as 24 additional inputs for submixes, making available a total of 72 inputs. *1619 Broadway, New York, NY; 212-765-4757.*

Southeast

Sonic Images (Washington, DC) has opened new 24-track audio and video production facilities, featuring two LEDE-type control rooms. New equipment includes an **Audio Kinetics Q.Lock Eclipse** system for audio/video synchronization and audio sweetening. The studio provides narration recording to picture, off-line video editing, and digital music scoring and sound effects composition on a **Kurzweil 250** with **MacAttach** software. *4590 MacArthur Blvd. NW, Washington, DC 20007; 202-333-1063.*

Music Business Institute (Atlanta) has installed two **Harrison MR-4** consoles, each with 28 inputs and 24 output-bus assigns. Input modules on both consoles have the ability to return two inputs in mixdown mode for dual multitrack operation. The consoles will be installed in the institute's newly completed studio complex and will be used during audio recording and production courses.

The studio, designed by **George Augsberger,** of Perception Inc., utilizes one large recording area with variable acoustics. Two identical control rooms connect to the studio, each featuring its own isolation room. Both control rooms are linked for time code interlock, as well as to an adjacent video production facility. *3376 Peachtree Rd. NE, Atlanta, GA 30326; 404-231-3303.*

Sheffield Audio-Video Productions (Phoenix, MD) has purchased a DASH-format **Sony PCM-3202** digital 2-track and a second **Sony PCM-3324** digital multitrack. Outboard additions include four **Yamaha SPX-90** digital processors and an **REV-7** digital reverb. *3816 Sunnybrook Road, Phoenix, MD 21131; 301-628-7260.*

Limelite Video (Miami) has ordered three consoles for its new video production facility: a 36-channel **Neve V series** with **NECAM 96** moving-fader automation for use in its new video sweetening studio; and two **model 5455** video post-production consoles for its new editing suites. *7355 N.W. 41st St., Miami, FL 33166; 305-593-6969.*

Reflection Sound Studios (Charlotte, NC) recently enlarged and rebuilt Control Room A, adding an adjacent isolation booth and installing all-new wiring using **Monster series One** and **Mogami** cable. **Steven Durr** served as acoustic consultant. *1018 Central Ave., Charlotte, NC 28204; 704-377-4596.*

TRC Mid America Recording Center (Indianapolis) has completed a remodeling and equipment upgrade for Studio B. The room now features a 32-input **Sound Workshop series 34B** console with **ARMS II** automation, and a **Sony JH-14** 24-track. Studio A now features a **Sony JH-14** with autolocator and **Klark-**

Studio update

Northeast

Century III (Boston) has added a **CMX CASS-1** audio computer to its mix-to-picture suite. The new system has been adapted for sound editing with a quoted accuracy of 333ms, and includes a hard disk and full console mixing automation.

According to **Jay Rose**, sound designer, "We can drop in the sound effects instantly, program fader moves precisely to the frame and save literally thousands of different mixes or alternative edits for instant recall. It gives producers the option to explore creative alternatives without committing to massive amounts of studio time." *651 Beacon St., Kenmore Square, Boston, MA 02215; 617-267-6400.*

Photo-Magnetic Studios (New York) has ordered two **Neve series V** post-production consoles with **NECAM 96** automation. The consoles, which will be used for video and film post-production audio work, provide 8-track and 4-track monitoring, with stereo music, dialogue and effects plus laugh track or SAP.

A trim facility in the monitor path allows selection to either 2-track or full-coat master recorders with a simultaneous mix, separate from the monitor. Metering is VU/PPM for 8-track/4-track and plasma display for all other 48-track outputs. Independent selections to any or all 4 stereo buses from each input are available, while maintaining regular 2-track output. *222 E. 44th St., New York, NY 10017; 212-687-9030.*

Windsor Total (New York) has installed a **Neve model 5455** 16-output post-production console. *565 Fifth Ave., New York, NY 10017; 212-725-8080.*

Unique Recording (New York) has added four **Studer A-800** Mk III 24-tracks, and three **A-820** ½-inch 2-tracks. Studio B's control room now features **UREI 813** speakers and a complete acoustical redesign of the walls, floors and ceiling. *701 Seventh Ave., New York, NY 10036; 212-921-1711.*

Sound One (New York) has taken delivery of a 48-input **Neve model 8128** console with **NECAM 96** moving-fader automation for its new film mixing studio. The custom console can accommodate eight discrete channels of monitoring for mixing a variety of film formats, including mono, Dolby Stereo and IMAX. The board's small faders have

Circle (120) on Rapid Facts Card

limitation (indicated by rounding of the square wave corners) move the generator past this point in order to get a true indication of what is happening in the next equipment.

Once you are satisfied with the system's frequency response, noise floor and transient response, perform overall distortion measurements. Although low-frequency total harmonic distortion measurements have merit, high-frequency THD measurements are almost meaningless on bandwidth-limited systems, such as low-pass filters found in digital tape machines and audio processors.

If slewing-induced IM (SID) and/or transient intermodulation distortion (TIM) exist within the system, they will only appear at higher frequencies. The best way to detect their presence in a bandwidth-limited system is with the CCIF twin-tone IM distortion measurement. By using 14kHz and 15kHz tones mixed 1:1, a 1kHz IM product is easily detected if SID or TIM exists. It's important that every production facility regularly performs twin-tone IM distortion measurements.

There are a couple of inexpensive tools that may help you identify the sources of problems within a facility. A telephone pickup coil coupled to a mic preamplifier and headphones can track down RF and hum-induced interference. A battery-operated, wideband ac voltmeter is handy for measuring voltage differences between various pieces of equipment.

The bottom line

If all these steps have been performed carefully, you should have an audio system that is capable of outstanding performance, with a residual noise floor close to that of the equipment with the highest output noise. Satisfying these criteria allow your facility to offer a new level of audio performance. R-E/P

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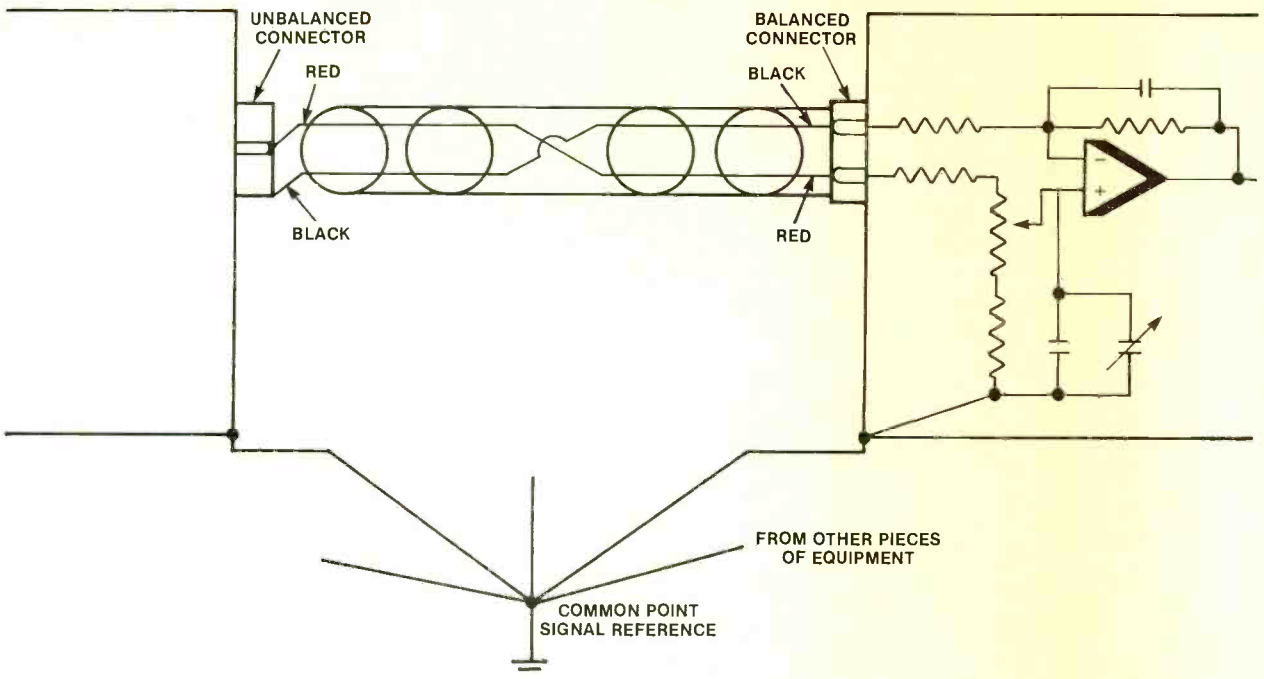


Figure 4. *Interconnection of unbalanced equipment to balanced equipment requires special consideration. Forward referencing will help reduce common mode signals.*

material, which are the source of overload, tape saturation and amplifier clipping. The VU meter, although a universal measuring device in America, Japan, France and Australia, finds little application in most of the rest of the world because of this problem. The problem occurs when adequate headroom has *not* been designed into the system.

Many users have concluded that the peak program meter (PPM) provides a superior indication of program material without lowering average amplitudes, while allowing additional control over excessive program peaks.

Existing VU meters can be converted to operate as PPM by using conversion cards. PPMs can also be installed alongside VU meters. As a bare minimum, add peak overload indicators to any equipment without PPM capability.

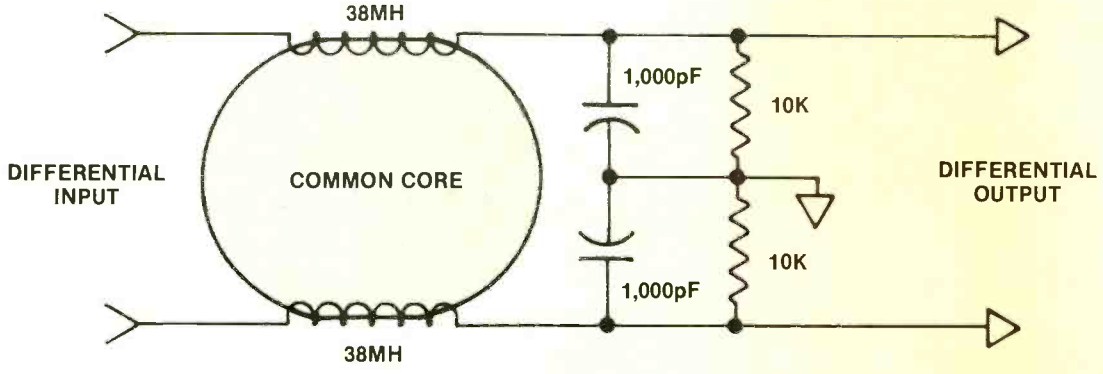
Testing

To check the system, conduct overall noise and frequency response measurements. If the above procedures are followed, wideband (f_3 = 100kHz) audio systems are feasible. It also provides a flatter frequency response and prevents an accumulation of 3dB rolloff points that would upset the system's high-frequency response. Other benefits include low phase shift out to 20kHz and good transient response.

Extensive 10kHz square-wave system tests can be made by placing a function generator at the start of the audio chain, and then looking at every output with an oscilloscope. I would recommend that all response-type measurements be made at −12dBu, which conforms to the new CCIR recommendations and ensures that no slew-rate limitations confuse the picture of system bandwidth.

Check for any overshoot or ringing, which would indicate underdamping of amplifier circuits and thus potential instability and RF susceptibility. It is useful to perform this test with each piece of equipment separately, and then as a system. Make sure that the line and buffer amplifiers under test do not go into saturation during this test; otherwise, the results will be inaccurate. When you find a stage that has significant bandwidth

Figure 5. *If RF interference presents a severe problem, a common mode filter such as the one shown here may be effective in reducing unwanted signals.*



the core cancel and the inductor effectively disappears from the circuit.

However, when the signal is of equal amplitude and the same polarity on both lines (as is the case with the interference we wish to remove), both lines see the L/C low-pass filter. The 2-pole device has a cutoff rate of 12dB per octave, and its response is down 60dB at 1MHz.

Additional shielding may be necessary. A separate conduit for the audio cabling can be very effective in reducing RF as well as with power line-related radiation. Tightening up the shielding of specific audio packages also may be required.

In a few extreme cases, entire studios have had to be constructed inside RF-tight screen rooms. If you need to construct a screen room for your facility, use steel screen for all interfering frequencies up to the FM band. Steel is more lossy and thus more effective than copper (and less expensive). Copper screen is necessary for microwave frequencies.

You will need to solder the seams, and, when your room is finished, be sure to use power-line filters physically located at the point of power entry into the room to prevent RF entering via the ac lines.

I recommend that all audio systems be run at a maximum nominal OVU audio voltage amplitude of +4dBu. As mentioned above, with ±15V power supplies most op-amps will clip around +21dBu or +22dBu. This limitation yields a headroom or overload factor of 17dB, which we saw earlier is a bare minimum for live audio. Peak-to-average ratios of 16dB have been measured on speech, and I have seen at least this high a ratio on percussive music.

My design philosophy is to take a 6dB loss at the input of the equipment, which yields an input clip point of +26dBu to +27dBu (up to +30dBu with 20V supplies). Operating the circuits at a nominal average level of −2dBu yields a headroom factor of 23dB.

The 6dB loss is made up at the inverting follower in the output stage, a configuration that allows both input and output clip points of +26dBu to +27dBu, and provides more headroom factor without sacrificing noise performance. The output noise floor of carefully designed equipment can reach −93dBu, which yields an average S/N ratio of 97dB and peak S/N ratio of 120dB. I recommend, therefore, that all of your equipment be capable of handling this high input and output level.

Setting up your system is also important. Adjust levels so that the gain of each unit allows the various pieces of equipment to reach their clip points at the same time. The majority of the needed gain should come from the amplifier stage that has the lowest noise figure. This is usually the mic pre-amp, the point where the system's dynamic range is established. As simple as it sounds, this adjustment is key to an outstanding system, and will optimize the S/N ratio.

Proper operation

Many engineers and producers find themselves fighting a perplexing and confusing battle with audio levels. The biggest problem stems from the inability of a VU meter to show peaks of program

RF interference

Occasionally, radio frequency-related problems will remain, even after the above procedures have been followed. The presence of RF-related problems may often be heard as an increase in high-frequency noise, a "gurgling" sound or an outright detection of the RF signal by a PN transistor junction in an active device or a poor solder joint. RF field intensities in the studios located close to AM and FM transmitters may easily range from 1V/meter to 100V/meter. As a result, the interconnect scheme described above may need modification.

The open-ended shield recommended above can cause the interconnection cable to act as an antenna. One possible cure is to tie the open end of the shield to its respective chassis ground through a 0.01μF, high-quality disc ceramic capacitor. This capacitor, in essence, provides a ground at RF frequencies, while leaving the shield open at audio and power frequencies. However, all capacitors have their own self-resonant frequency, and you may need to parallel two or three capacitors of different values.

Three basic op-amp gain configurations—non-inverting, inverting and differential—each have different RF sensitivities. The non-inverting, unity-gain buffer amplifier is the most sensitive, and the well-balanced differential input stage the least sensitive. Manufacturers sometimes use an instrumentation amplifier input stage consisting of two unity-gain, non-inverting buffer amplifiers followed by a well-trimmed differential amplifier. In high RF environments, it may be necessary to replace the input buffers with jumpers to lower the input impedance for the sake of RF stability.

At both AM and FM frequencies, series-resonant L/C networks can be placed from each line of a balanced pair to ground to bypass incoming RF. These devices should be placed just inside the chassis and directly at connector terminals. At FM frequencies, self-resonant (parallel) chokes may be placed in series with the incoming lines, which should be shunted with capacitors that yield a maximum of 2Ω reactance to ground at the RF frequency of interest.

A typical self-resonant choke for 100MHz would be a single-layer solenoid approximately 1/8-inch in diameter and 1-inch long. It is tuned by varying the spacing between windings and thus changing the distributed capacitance. You can expect approximately a 45dB reduction of RF energy from such a network.

A helpful but almost unknown device in audio is the common-mode filter (Figure 5). When inserted in a balanced line, these filters are effective in troublesome RF environments. A typical filter will consist of a common-mode choke made up of dual, highly symmetrical windings on a common toroid core, two 1,000pF capacitors, and two 10kΩ termination resistors. A well-designed filter will have a differential bandwidth of greater than 200kHz when driven from a low-impedance source, but a common-mode bandwidth of only 26kHz. When the choke sees a differential signal (equal amplitude and opposite polarity), the magnetic fields created in

The signal references are then tied together via a star-wire system. To allow proper grounding, some equipment manufacturers provide a terminal strip on their equipment, with the signal reference on one terminal and the chassis ground on an adjacent terminal.

With large systems, such as a production center, where many pieces of equipment must be interconnected, a *star of stars* is the correct approach. The hub of the stars is again the central point of the signal distribution, usually the master control or termination area. The time spent in carefully planning a signal reference system will pay large dividends.

- Connect all inputs and outputs together leaving the drain (or shield) floating at one end. Grounding the cable shield at the input end allows a common mode feedback signal to be placed on the shield. Either end may be grounded with the other end left floating, but be consistent with your system. Connecting both ends to the shield can cause power line-related currents to flow between equipment through the shield, and induce a signal into our audio inputs.

Balanced outputs are connected in the traditional fashion. Unbalanced outputs should be connected so that the signal is forward-referenced (Figure 4). This is accomplished by connecting the non-inverting (+) input to the output terminal, and the inverting (−) input to ground (signal-reference) terminal of the unbalanced output. Any voltage difference that still remains between the two pieces of equipment will be ignored by the input as a common mode signal.

- With jackfield and patchbars, use a fully balanced interconnection arrangement, which allows connecting to unbalanced outputs while maintaining forward referencing.

Because the shield lines of the jacks are not usually switched, a jackfield presents an interesting problem in light of the above recommendations. If you wire the jackfield so that the drain wires tie across from output jack to input jack as a part of the normalled wiring, and do not bus together the grounds, everything is fine—until you patch. Then, you have improperly tied the grounds together. The best way to wire a jackfield is to tie all drain wires at the equipment ends, both input and output, but *not* at the jackfield. Then at the jackfield, bus together all grounds, and tie them to the system grounds as though it were a separate piece of equipment, using a separate insulated ground wire.

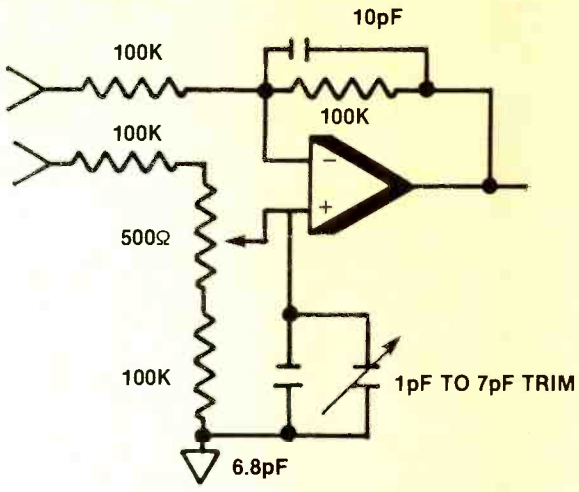


Figure 3. *The best common mode rejection is obtained from circuits that use both precision resistors, a trim pot and a capacitive trim control.*

or differential input to reject a signal that, referenced to ground, has the same amplitude and phase on both inputs. The amount of CMR needed is directly dependent upon the amplitude of the common mode signal.

If, for example, the power line-related voltage difference between two chassis was measured and found to be 200mV (−11.76dBu), and if you wanted a noise floor of −90dBu, the amount of CMR needed is the difference between these two amplitudes, 78.24dB at the frequencies of interest.

Not all differential input stages are capable of the same level of performance. To achieve the degree of common mode rejection to which most op-amps are capable requires some very careful design, PC layout and adjustment.

Low-frequency CMR is achieved by making the resistive portion of the differential-amplifier gain determining network a *balanced* bridge. Some manufacturers use 5% resistors to form the differential input stage, while others use 1% metal-film resistors but do not use trimmers to precisely balance the bridge. The best solution is to use both 1% resistors and a stable trim resistor to achieve high degrees of CMR (Figure 3).

The average CMR that can be expected with 5% resistors is 26dB, whereas 40dB levels are possible with 1% resistors. A carefully trimmed input stage is capable of better than 100dB of CMR at low frequencies. To achieve any kind of high-frequency CMR requires that stray capacitance also be trimmed out.

Pratically speaking, 75dB of CMR is achievable to 20kHz by adding a capacitive trim, and even greater CMR can be achieved using very careful PC layout. The precision balance of the resistor/capacitor bridge around the op-amp makes the common mode signal exactly equal at the device's inputs, and the common mode signal is then rejected by its input stage.

Although it's possible to achieve 90dB of CMR in a transformer, most devices on the market won't begin to come close to that figure.

It is usually preferable to use an active input stage (provided it has been correctly designed), rather than a transformer, because of its freedom from low-frequency distortion, hum susceptibility, and (sometimes) frequency response errors and ringing. If the common mode signal cannot be reduced below 2V or 3V, as is often the case with telco feeds used during broadcast remotes, for example, a transformer input should be used. The transformer has an almost unlimited common mode input voltage range (not rejection) capability.

Interconnect

If you adopt the following rules for the interconnection of your equipment, hum-free audio can almost always be achieved.

- All equipment *must* have well-trimmed, high-impedance balanced inputs. This is absolutely necessary to reject the residual power line-related voltage differences between pieces of equipment. For equipment that does not have balanced inputs, use one of the many interface boxes available. Better yet, install internal balanced inputs, an approach that provides control over the interface grounds.
- All equipment should have balanced, low-impedance outputs. Satisfactory performance from unbalanced outputs can often be obtained by following these same rules, provided that the existing output is low impedance (100Ω or less) and the line is as short as possible and not located in a high RF field.
- Interconnect all pieces of equipment with a separate *insulated* ground wire, size No. 14 or larger. Arrange the ground wires to form a star or single-point ground system. The hub of this star could be a ¼-inch thick copper plate about 4"x6", and located physically at the center of the audio signal hub, such as an audio console or interface rack. The copper plate needs to be insulated from *any* other ground. Add an additional wire to the star to connect the copper plate to the system's house-power ground located within the power box. Better yet, go directly to the power company's local grounding point, which is preferably a water pipe rather than a ground stake, and be sure to use your own clamp.

At times, the star configuration must be created using the third-wire safety ground of the power cord. This method is inferior to using a separate conductor from the power system star ground, because the ground reference third-wire may have a power-related signal induced into it by the parallel runs of power wiring. However, if the equipment does not have a signal ground reference within the chassis that can be floated, then this may be the only recourse.

Do *not* be tempted to eliminate the third-wire safety ground. First of all, it is not wise from the point of operator safety. Secondly, for any equipment not mounted in racks, it can increase the unit's susceptability to RF. Finally, it is *illegal.* In every installation all third-wire safety grounds should have a separate wire that returns to the power box from every isolated-type outlet.

Do *not* use the metal conduit as a substitute for the separate third wire, since its use will disturb a star configuration. Third-wire grounds should tie together in the power box via a copper bus bar that is floating from the box and tied to the power system ground point via a 0000-gauge cable.

In any case, all signal references must be tied together with a separate wire other than the drain of the shielded pair used to interconnect the signals. These references can be tied together via a brute-force system by the equipment racks in which the equipment is installed.

fails to see such equipment as an element of a long audio chain, and potentially the limiting element. Although you will actually never put a signal into that upper portion of the 200kHz bandwidth, it must exist to achieve the necessary 30kHz to 40kHz system bandwidth. (Remember, two pieces of equipment, each with a −3dB point at 20kHz, will have a system response of −6dB at 20kHz.)

Slew-rate limitations, on the other hand, are large voltage-swing limitations; again they have to do with the actual current output required of a stage driving a capacitance. It is important that no amplifier be allowed to slew limit, since high-frequency intermodulation distortion will result. If an amplifier can provide adequate current to a cable to allow full output swing to 30kHz at low THD, the chances are practically zero that it will ever slew rate limit with normal audio.

So, to maximize the interconnect performance in a system, perform the following: move to a low output impedance of 60Ω balanced on all equipment; keep the cable runs as short as possible; where long lines are unavoidable, use low-capacitance cable; and make sure the device driving a long line can supply the necessary current at high frequencies.

Another benefit of a 60Ω output impedance is that there is only a 0.8dB amplitude difference between a bridging input and a 600Ω input. The advantage of this may not be immediately obvious. Consider a typical situation where the nominal system reference (OVU) is +8dBu. Next, you must realize that although the normal peak-to-average ratio (crest factor) of most audio is 8dB to 10dB, it may be as high as 16dB or even higher on very percussive material.

Assume that you are feeding a bridging input from a 600Ω system (the output voltage will be 6dB higher without a 600Ω termination). Now let's add up the amplitudes: +8dBu (system average) +16dB (peak-to-average ratio) + 6dB (for no termination) is equal to +30dBu peak output amplitude. Then, when you understand that a unity gain differential (op-amp) input stage running from ±15V supplies (typical for many pieces of equipment) clips at about +22dBu, you begin to see the need for reducing that last 6dB term. (Even if the nominal reference is +4dBu, the peak output amplitude will be +26dBu).

Common mode rejection

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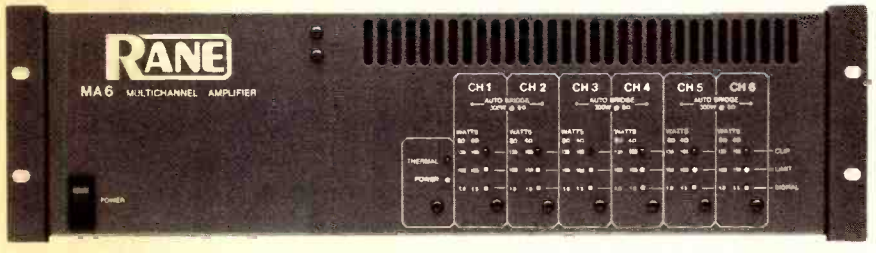

Yes, the MA 6 is a conceptual breakthrough of magnanimous proportions. Yes, the MA 6 is eons ahead of any other amplifier configuration. And yes, when the brains at Rane came up with it they had to take the rest of the afternoon off to recuperate.

The MA 6 is the first and only dedicated 6-channel amplifier. And presents an entirely new way of looking at the system amplifier approach.

Those used to a two channel will find this degree of flexibility amazing. It can act as a monitor amp for up to six separate channels of stage monitor mixing. Or drive a stereo pair of two-way speaker systems bi-amped with two bridged pairs of 300 watts for the low end, and two 100-watt channels for the high end, with limiter drive protection. Or run three bi-amped monitors at 100 watts each section. Or three passive monitors at 300 watts.

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RANE CORPORATION

Circle (82) on Rapid Facts Card

RMS, which yields a peak output voltage of 34.69V. Therefore, SR = 6.529 x 10^6 volts per second.

Now that you have calculated the required slew rate, let's calculate the current necessary to do so. Recall that the 1,000 feet of cable has a capacitance of 32nF (32 pF/ft).

Therefore, using the formula,

$$I = C \frac{dv}{dt}$$

at +26dBu output, I = $32x10^{-9}$x $4.119x10^6$; or 131.8mA. At +30dBu output, I = $32x10^{-9}$ × $6.429x10^6$, or 208.9mA.

Suppose that you wish to drive the same line with time code, and that a 100kHz full-voltage output is required. SR = 2x3.14 x 100,000x21.85, or 13.73 x10^6 volts per second. (At + 30dBu, SR = $21.76x10^6$ volts per second.) In this case, the line length is limited to considerably *less* than 1,000 feet.

If you want a full output to 100kHz, then you really need a small-signal bandwidth of at least 300kHz, which limits the total cable capacitance to 8.84 x 10^{-9} Farads. At 32pF/ft, the maximum cable length we can use is 276 feet. The current required is $8.84x10^{-9}$ x $13.73x10^6$, or 121.4 milliamps. (At +30dBu output, I = 192.4mA.)

If longer cable lengths are required, low capacitance cable—of the order of 6pF/ft—is the only alternative.

All of the above assumes a 30kHz interconnect full-voltage capability, a recommendation for minimum system performance. I would recommend that a 200kHz small-voltage bandwidth capability be the minimum design goal to achieve the flattest response and minimum phase shift at high frequencies for good overall *system* performance.

It is important to recognize the difference between the small-signal bandwidth of an interconnect and its slew-rate limitations. Small-signal bandwidth sets the 3dB cutoff of the interconnect filter, which in turn describes the flatness of the response at 20kHz. Additionally, this 3dB point has phase shift that will also be reflected back to 20kHz.

You must remember that *every* element of an audio system will contribute its 3dB cutoff and associated phase shift to the overall performance of the system. As a result, every element must be viewed as one section of a large, multiple low-pass filter; and although one element may have adequate response to 30kHz or 40kHz, it is the cumulative effect of these filter sections that's of major concern.

At first glance, the proclaimed need for a wide bandwidth of 100kHz to 200kHz in both the equipment *and* the interconnect may seem outlandish. However, when you realize that major recording and production facilities may have from 10 to 20 pieces of equipment in the audio chain, each contributing its own cutoff characteristic, you begin to realize the magnitude of the problem in achieving adequate high-frequency performance through the system.

For years, manufacturers of audio equipment have mistakenly considered an upper bandwidth of 30kHz to 40kHz to be adequate. This narrow viewpoint

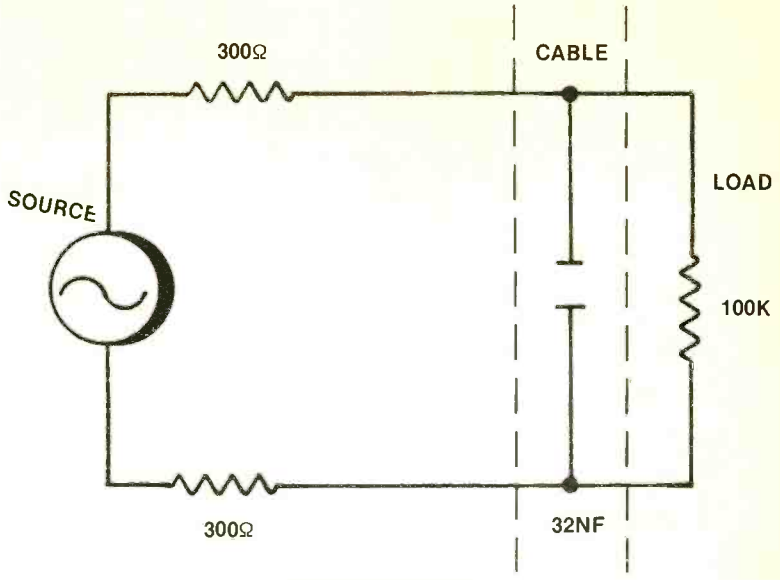

Figure 2. *Using a 600Ω impedance to drive a long line can result in a low cutoff frequency, in this typical case 8.3kHz.*

cutoff you can tolerate. You might think that a lower source impedance would improve things even further. This is not normally the case, however. Because of series inductance in the cable, lower source impedances will result in high-frequency peaking.

In addition to the small-signal bandwidth limitation set by the interconnect low-pass filter, an additional problem is created by the cable's capacitance. If you are to drive the cable to any appreciable voltage swing at high frequencies, you will need a current from the line or buffer amplifier to feed that capacitance. The amount of required current is in direct proportion to the total cable capacitance and output slew rate at the highest frequency to be sent. Slew rate is the rate of voltage change per unit of time. In this usage of the term, slew rate refers to the *linear* slew rate of an amplifier—that is, the amplifier is not allowed to go into saturation.

Linear slew rate is defined as follows:

$$SR = 2\pi f V_p$$

$$\text{Where SR, slew rate} = \frac{dv}{dt}$$

Here, f is the minimum upper frequency for full voltage response, and V_p the peak output voltage from the amplifier.

The amount of current that is required to feed the cable, or in fact any capacitance, is given by:

$$I = C \frac{dv}{dt}$$

Consider the following example: you have 1,000 feet of cable and expect to be able to pass 30kHz at the maximum output of our consoles. (Let's use a maximum output for most equipment of recent design of +26dBu, and another output of +30dBu.) What is the required drive current from your console that meets these criteria? A level of 0dBu is a voltage reference of 0.7746V (the same voltage that 0dBm produces across 600Ω).

$$dBu = 20 \log (V/0.7746)$$

$$V = [10^{[dBu/20]}] 0.7746$$

A level of +26dBu is equivalent to 15.46V RMS output; the peak output voltage is 1.414xRMS or 21.85V. Therefore, SR = 2x3.14x30,000x21.85, or 4.119×10^6 volts per second (or 4.119V per microsecond, if you prefer).

Also, +30dBu out is equal to 24.50V

find yourself in a situation where it is impossible to use conduit, and virtually impossible to have a separate space for audio and power lines, where audio and power lines must cross do so at 90° angles.

• Tie all equipment together in a star configuration—in other words, all power lines should have their grounds referenced to a single point, rather than being daisy-chained—with *insulated* ground wires to minimize any potential differences between pieces of equipment.

Interconnection systems

Although not yet adopted universally, the voltage-sourced-balanced interconnect system is the preferred method. The single-ended, IHF interconnect system found in semi-pro equipment is not usable, because of its inability to reject normal power line-related voltage differences. (Although the latter is now minimized by tying the equipment together with separate ground lines that still exist between various pieces of audio equipment.)

The 600Ω, power-matched system, which was developed in the days of tube equipment to achieve maximum signal-to-noise ratio, is falling from favor. With modern op-amp technology, it is no longer necessary (or even desirable) to terminate audio lines with a matched low impedance. The foil-shielded audio cable used today does not have a characteristic line impedance of 600Ω; it's usually below 100Ω.

The voltage-sourced interconnect system with low (50Ω to 60Ω) source impedance and relatively high (10kΩ and up) input impedances is becoming the accepted practice. An output impedance of 50Ω to 60Ω has been found to be the optimum drive Z for today's foil-shielded audio cables. Advantages include: less power drawn at low frequencies from the source equipment (normally no 600Ω loads) and therefore less heat generated; lower noise pickup by interconnect lines because of the lower equivalent line impedance; and, most importantly, five to 10 times the cable length may be driven for a predetermined high-frequency cutoff point.

To understand this last statement, you need to understand that the equipment's output impedance combines with the capacitance of the interconnecting cable to form an R-C low-pass filter. For example, if you use 1,000 feet of shielded pair (not uncommon in larger, interconnected facilities) whose capacitance is 32-F/ft between conductors, with a drive impedance of 600Ω and a bridging 100kΩ input, then the high-frequency cutoff is only 8.34kHz (Figure 2).

With the line terminated in 600Ω, the situation improves to 16.68kHz. By dropping the source impedance to 60Ω, the bandwidth of the connection is now 83.4kHz.

Total capacitance is determined by the following formula:

$$C = \frac{1}{2\pi f_c R}$$

C is the maximum allowable cable capacitance, R the output impedance (60Ω), and f the lowest small-signal HF

Interconnecting Audio Equipment

By Allen Burdick

Correctly interfacing recording and production equipment to provide clean, noise-free audio presents its own set of unique problems.

Achieving a clean, noise-free audio is a challenge that mixing and production engineers must face daily. Fortunately, the tools are available and the highest price to be paid is usually that of time and care during system design and installation.

There are five major areas, other than choosing quality equipment, that must be properly addressed to achieve dependable, high-quality audio: proper power system installation; interconnection system methods; RF immunity; proper headroom and signal-to-noise ratios; and correct operation within the system's limits.

Power system installation

If all of the lighting and equipment in the world ran on direct current, as Edison had wanted, then the only problems we'd face would be that of RF immunity and the selection of equipment that provided low noise and distortion levels, and wide bandwidth. As everyone knows, however, this is not the case.

Probably the largest single source of audio interference is that of power line-related signals being added to the audio signals. Most of this problem is a result of normal 50Hz or 60Hz voltage differences that exist between equipment chassis (Figure 1). These differences result from the different electromagnetic environments that each piece of equipment experiences. The interference may be caused by nearby ac power lines, magnetic leakage from an internal power transformer or from powering the various pieces of equipment from different legs of a 3-phase power distribution system. In addition, higher frequency audio noise may be *conducted* into the equipment via the power line. Our objective is to reduce or eliminate these potential sources of interference.

Allen Burdick is president of Benchmark Media Systems, Syracuse, NY.

The following generally accepted rules should be observed:

- Use a separate, dedicated feeder for all audio equipment. Install a separate transformer (if required) and breaker panel as close as possible to the audio equipment. This separate feeder should originate as close as possible to the point of power entry into the building, which will provide the lowest possible source impedance. Likewise, the equipment grounding conductor should also originate from the cleanest possible source (see below).
- If the previous arrangement is not possible, try to use only one of the three available phases to power all audio equipment. Perform load balancing on the other two phases with lighting, air conditioning and other equipment.
- Keep all high-power switching equipment—such as SCR lighting controls and motor switching—off the separate feeder (or phase) used for audio equipment.
- To minimize conducted and leakage coupling, use equipment that has input line filters and quality power transformers. Occasionally, a high-isolation transformer with ultra-low coupling capacitance between windings may be necessary to eliminate high-frequency interference conducted from the power line.
- Provide individual (separate) steel rigid conduit runs for power and audio lines, separated physically as much as possible to avoid induced power-related signals. Be sure to use ferrous metals (not aluminum) for magnetic shielding. If you

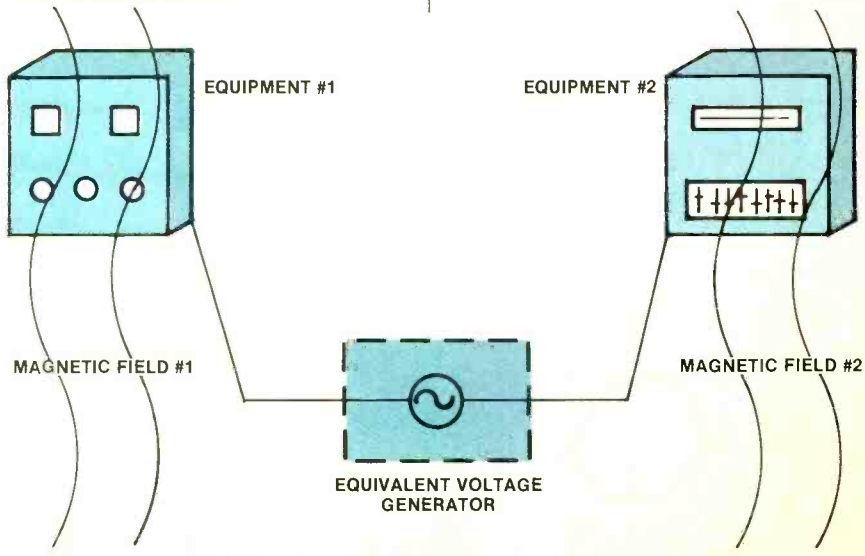

Figure 1. Most interference stems from power-line signals and voltage differences between equipment chassis.

Tape machines, digital
Advanced Music Systems (718)
Apogee Electronics (1402)
JVC Company of America (Serrano Room, Hilton Hotel)
Lexicon (424)
Mitsubishi Pro Audio Group (526)
New England Digital (Rooms 216 A & B, Convention Center)
Otari (110)
Sony Broadcast Products (300)

Tape machines, mag film
Mitsubishi Pro Audio Group (526)

Tape machines, replacement heads
JRF Magnetic Sciences (726)
Saki Magnetics (726)

Test & measuring equipment
ACO Pacific (827)
Apogee Electronic (1402)
Audio Precision (826)
B&B Systems (925)
Bruel & Kjaer (240)
Cal Switch (205)
Cetec Ivie (644)
dbx (121)
Dorrough Electronics (745)
Gold Line (443)
Gotham Audio (520)
IQS (702)
Kenwood Communications (933)
Klark-Teknik Electronics (608)
Research Technology International (948)
Sound Technology (808)

Time code synchronizers
Adams-Smith (446)
Audio Kinetics (117)
Cipher Digital (840)
Fostex Corporation of America (500)
JVC Company of America (Serrano Room, Hilton Hotel)
Otari (110)
Skotel (947)
Soundmaster International (641)
Studer Revox America (312)
Timeline (903)

VU/PPM meters
Audio Analyse Instruments (919)
Selco/Sifam (806)

R-E/P

Power amplifiers, continued
RTS Systems (844)
Tannoy North American (438)
Technics (901)
Times One (1001)

Racks & stands
Korg USA (814)
Professional Sound Corporation (1409)

Recording tape
Agfa-Gevaert (208)
Ampex (101)
BASF (125)
Magnefax International (349)
Research Technology International (948)
Sukyong International (736)
3M/Magnetic Tape Division (212)
Zonal (941)

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Advanced Music Systems (718)
AKG Acoustics (236)
Alesis (910)
Altec Lansing (822)
Aphex Systems (626)
API Audio Products (728)
Applied Research & Technology (746)
Audio/Digital (Hillhurst Room, Hilton Hotel)
Barcus-Berry Electronics (912)
BGW Systems (201)
Biamp Systems (1412)
Cetec Ivie (644)
Connectronics (730)
DCS Audio Products (602)
DOD Electronics (646)
D&R Electronika (747)
ElectroSpace Developments/ PMI (549)
Eventide (313)
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Ibanez (831)
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JBL Professional (326)
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Korg USA (814)
Lexicon (424)
Orban Associates (324)
Peavey Electronics (Larchmont Room, Hilton Hotel)
Publison America (634)
Renkus-Heinz (700)
Roland Corp. US (440)
Sontec Electronics (946)
Studio Technologies (741)
Valley People (250)
Yamaha (114)

Sound samplers
E-Mu Systems (Glenwood Room, Hilton Hotel)
International Music Company/ Akai Digital (546)
Korg USA (814)
New England Digital (Rooms 216 A & B, Convention Center)
Roland Corp. US (440)
360 Systems (922)

Synthesizers & keyboards
E-Mu Systems (Glenwood Room, Hilton Hotel)
Fairlight Instruments (Room 202, Convention Center)
Korg USA (814)
Kurzweil Music Systems (Verdugo & Del Mar Rooms, Hilton Hotel)
New England Digital (Rooms 216 A & B, Convention Center)
Roland Corp. US (440)
Steinway and Sons (Brentwood Room, Hilton Hotel)
Yamaha (114)

Tape duplication, cassette & reel-to-reel
AEG (928)
American Multimedia/Concept Design (731)
Apex Machine Company (1404)
Audico (211)
Audio Video Consultants (343)
Kenneth A. Bacon (805)
BASF (125)
Dwight Cavendish Company (940)
Cetec Gauss (340)
CST Sales (1403)
Educational Electronics (904)
Electro Sound (724)
Inovonics (607)
IPS (848)
King Instrument (346)
Magnefax (349)
Paktec (1406)
Sukyong International (736)
Telex Communications (735)
TTL USA (729)
Versadyne International (1417)
Zonal (941)

Tape machines, accessories
American Recorder Technology (906)
Fife-Pearce Electronic Company (950)

Tape machines, analog
AEG (928)
Audio Media Research (Larchmont Room, Hilton Hotel)
Fostex Corporation of America (500)
International Music Company/ Akai Digital (546)
JBL Professional (326)
Nagra Magnetic Recorders (203)
Otari (110)
Sony Broadcast Products (300)
Studer Revox America (312)
TASCAM/TEAC Corporation of America (412)
Technics (901)

Tape machines, cartridge
Otari (110)

Tape machines, cassette
Nakamichi (812)
Sony Professional Audio (300)
Studer Revox America (312)
TASCAM/TEAC Corporation of America (412)
Technics (901)
Yamaha (114)

Sony Broadcast Products (300)
Swintek Enterprises (1416)
Telex Communications (735)

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Alesis (910)
JL Cooper Electronics (944)
Fairlight Instruments (Room 202, Convention Center)
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Korg USA (814)
Kurzweil Music Systems (Verdugo & Del Mar Rooms, Hilton Hotel)
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Audio Analyse Instruments (919)
Cal Switch (205)
Canare Cable (743)
Countryman Associates (704)
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Klark-Teknik Electronics (608)
Oxmoor (942)
Pacific Recorders Electronics (929)
Professional Sound Corporation (1409)
SCV Audio (1414)
Selco/Sifam (806)
Technics (901)
Valley People (250)
Whirlwind Music Distributors (434)
Wireworks (538)

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Audio-Technica US (233)
Stanton Magnetics (120)

Power Amplifiers
Altec Lansing (822)
Audio Analyse Instruments (919)
Audio Media Research (Larchmont Room, Hilton Hotel)
BGW Systems (201)
Biamp Systems (1412)
Boulder Amplifiers/Silver Lake Research (1410)
Bryston (800)
Cetec Ivie (644)
Crest Audio (617)
Crown International (227)
Fender Musical Instuments (849)
FM Acoustics (847)
Gotham Audio (520)
David Hafler Company (518)
Innovative Electronic Design (732)
JBL Professional (326)
Meyer Sound (512)
Panasonic Industrial Company/RAMSA (Room 209, Convention Center)
Peavey Electronics (Larchmont Room, Hilton Hotel)
Renkus-Heinz (700)

Loudspeakers, sound reinforcement
Anchor Audio (604)
Apogee Sound (1402)
Audio Media Research (Larchmont Room, Hilton Hotel)
Cerwin Vega (Wilshire Rooms, Hilton Hotel)
Cetec Gauss (340)
Community Light & Sound (506)
Electro-Voice (317)
Fostex Corporation of America (500)
Industry West Electronics (818)
JBL Professional (326)
Martin Audio London Ltd./ Martin America (913)
Meyer Sound (512)
Panasonic Industrial Company/RAMSA (Room 209, Convention Center)
Professional Audio Systems (246)
Renkus-Heinz (700)
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ACO Pacific (827)
AKG Acoustics (236)
Audio-Technica (233)
Crown International (227)
Gold Line (443)
LTM Sound Department (834)
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Professional Sound Corporation (1409)
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International Music Company/Akai Digital (546)
JBL Professional (326)
Klark-Teknik Electronics (608)
Rupert Neve (200)
Panasonic Industrial Company/RAMSA (Room 209, Convention Center)
Shure Brothers (107)
Sony Broadcast Products (300)
Soundtracs (740)
TASCAM/TEAC Corporation of America (412)
Yamaha (114)

Crossovers & frequency dividers
Apogee Sound (547)
BGW Systems (201)
Biamp Systems (1412)
Cetec Ivie (644)
DOD Electronics (646)
FM Acoustics (847)
Gold Line (443)
Klark-Teknik Electronics (608)
Peavy Electronics (Larchmont Room, Hilton Hotel)
SVC Audio (1414)

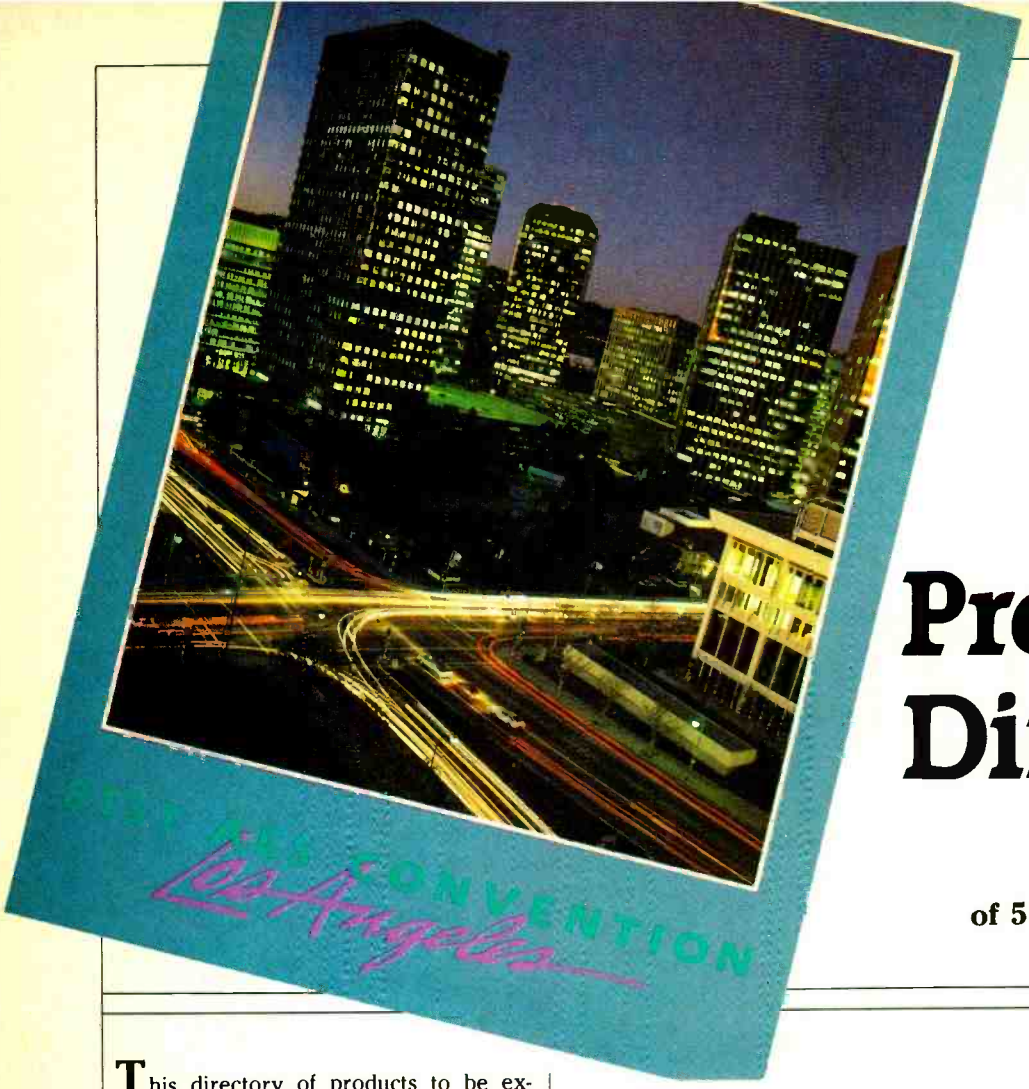
Disc-mastering systems
Gotham Audio (520)
Sontec Electronics (946)

Drum machines
E-mu Systems (Glenwood Room, Hilton Hotel)
Korg USA (814)

Editing systems
Adams-Smith (446)
Advanced Music Systems (718)
Alpha Audio (508)
Audio Kinetics (117)
Cipher Digital (840)
CMX (828)
Fostex Corporation of America (500)
JVC Company of America (Serrano Room, Hilton Hotel)
Lexicon (424)
Soundmaster International (641)

Educational courses & programs
Full Sail Center for the Recording Arts (829)
Institute of Audio-Video Engineering (943)

Equalizers, graphic & parametric
API Audio Products (728)
Applied Research & Technologies (746)
Biamp Systems (1412)



Product Directory

An A-Z listing of 51 product categories.

This directory of products to be exhibited at the AES Convention lists the companies that manufacture equipment or offer services in 51 categories.

To find the companies that make a product in which you are interested, search for the appropriate category printed in bold face. The companies are listed in alphabetical order, along with booth numbers.

To save space, companies that were assigned more than one booth are listed only with the *first* booth number. For companies that have booth numbers *and* a demonstration room, only the first assigned booth number is listed. Companies with only demonstration rooms are identified with the room name or number along with its location (Hilton Hotel or Convention Center).

Complete booth information, as well as information on new and existing products, is available in the exhibitor listings, which begin on page 111. Also, the demo room map of the Hilton (page 104), the Convention Center's main floor map (pages 106-107) and its demo room map (page 110) show the complete assignments for all exhibitors.

Because of last-minute changes, there may be some discrepancies in the booth numbers. Be sure and check your convention program for an update.

Acoustic design & construction
Sierra Audio Acoustics (1025)

Acoustic treatment materials
Acoustic Sciences (1418)
Alpha Audio (508)
Monster Cable (833)
Penn Fabrication USA (939)
RPG Diffusor Systems (642)

Business software
Audiocast Conseil (927)
Studio Master Systems (909)

Cable & connectors
Cal Switch (205)
Canare Cable (743)
Connectronics (730)
Dialight Corporation/ Kulka Smith (951)
FM Acoustics (847)
Marshall Electronics (751)
Monster Cable (833)
Pro Co Sound (748)
Whirlwind Music Distributors (434)
Wireworks (538)

Compressor/limiters/ noise gates
Aphex Systems (625)
Biamp Systems (1412)
Cetec Ivie (644)
dbx (121)
DOD Electronics (646)
ElectroSpace Development/ PMI (549)
Gold Line (443)
Klark-Teknik Electronics (608)
Orban Associates (324)
Publison America (634)
Valley People (250)
Yamaha (114)

Consoles, automation systems
Audio Kinetics (117)
Digital Creations (445)
Harrison Systems (618)
Rupert Neve (200)

Consoles, digital
Rupert Neve (200)
Sony Broadcast Products (300)

Consoles, faders
Penny & Giles (209)

Consoles, recording & production
Allen & Heath Brenell (819)
Amek Consoles (712)
API Audio Products (728)
Audio Media Research (Larchmont Room, Hilton Hotel)
Audio-Technica (233)
Biamp Systems (1412)
Connectronics (730)
Design Direct Sound (630)
DOD Electronics (646)
D&R Electronics (747)
Fender Musical Instruments (849)
Fostex Corporation of America (500)
Gotham Audio (520)
Harrison Systems (618)
Innovative Electronic Design (732)
International Music Company/ Akai Digital (546)
JBL Professional (326)
Klark-Teknik Electronics (608)
Mitsubishi Pro Audio Group (526)
Neotek (540)
Rupert Neve (200)
Panasonic Industrial Company/RAMSA (Room 200, Convention Center)
Professional Sound Corporation (1409)
Shure Brothers (107)
Solid State Logic (226)
Sony Broadcast Products (300)
Soundtracs (740)
Studer Revox America (312)
TASCAM/TEAC Corporation of America (412)
Trident Audio USA (612)

800 series open-reel audio tape; U-matic 3/4-inch PCM videocassette in 60- and 75-minute lengths.
Booth numbers: 212, 214, 216, 218
Circle (580) See ad pages 36-37

360 Systems
Existing products
MIDI Bass sound generator; MIDI Patcher signal router; MIDI Merger data filter and transposer.
New products
Permanent playback audio message, sound effects and multichannel card.
Booth numbers: 922, 924
Circle (581)

Timeline
Existing products
Lynx time code module with reader, generator and chase synchronizer.
Booth numbers: 903, 905
Circle (582)

Times One Technology
Existing products
RF-401 and RF-403 dual mono amplifiers.
New products
RF-700 and RF-200 dual mono amplifiers.
Booth number: 1001
Circle (583)

Trident Audio USA
Existing products
Series 65, 75 and 80B recording and production consoles.
New products
40-input DI-AN digitally controlled analog console.
Booth numbers: 612, 614, 713, 715
Circle (584) See inside front cover-1, 103

27th Dimension
Existing products
Production music library on vinyl and Compact Disc.
Booth number: 832
Circle (585)

TTL USA
Existing products
Model 515-B automated cassette loader.
Booth number: 729
Circle (586)

Turbosound
Existing products
TMS, TSW, TSE, TMW and TPC series loudspeaker enclosures.
New products
TSM series high-frequency manifold device; LS-2403 24-inch loudspeaker with 6-inch voice coil.
Booth numbers: 708, 710, 809, 811
Circle (587) See ad page 91

UREI
See JBL Professional.
Circle (600)

Valley People
Existing products
Noise gates, expanders, compressors, duckers, parametric equalizers, dynamics processors and level-matching interfaces.
New products
Models 415 and 815 dynamic sibilance processor modules; Leveller limiter; model 400 transformerless mic processor.
Booth number: 250
Circle (588)

VCL Audio
Existing products
Modular audio command desk.
New products
Modular video and lighting command desk.
Booth numbers: 900, 902
Circle (589)

Versadyne International
New products
1500 series high-speed cassette duplicator with dual-channel 5MHz bias supply, and capable of handling 12 slave recorders.
Booth number: 1417
Circle (590)

Westlake Audio
Existing products
Equipment sales and service company.
New products
Model BBSM-8 studio monitor loudspeaker.
Booth number: Room 210, Convention Center
Circle (591) See ad page 11

Whirlwind Music Distributors
Existing products
Transformers, interface devices, cable, cable reels and connectors.
Booth numbers: 434, 436
Circle (592) See ad page 174

Wireworks
Existing products
Stage boxes, racks, tails, trunks and transformer isolated splitters; mic and coaxial cables; mic cable tester; audio/video cable assemblies.
New products
Model CR1207 and CR1808 cable reels.
Booth numbers: 538, 639
Circle (593)

Yamaha
Existing products
Recording and live performance mixing consoles; cassette recorders; synthesizers; musical instruments; microphones; monitor loudspeakers and component drivers; signal-processing equipment; equalizers.
New products
KM802 keyboard mixer; S300 2-way loudspeaker system; MZ series dynamic microphones; GC2020B stereo compressor/limiter.
Booth numbers: 114, 115, 116; Chandler, Sawtelle, Balboa and Mulholland rooms, Hilton Hotel
Circle (594) See ad page 13

Zonal
Existing products
Reel-to-reel, cassette and mag-film tape.
New products
1/4- and 2-inch studio tape; chrome, super-chrome and ferric cassette duplicating tape; full-coat mag film.
Booth number: 941
Circle (595)

Late Entries

Audiotek
Booth number: 1015
Circle (601)

Carver
Booth number: 1003
Circle (602)

Celestion Industries
Booth number: 921
Circle (603)

Clarity
Booth number: 1013
Circle (604)

Eastern Acoustic Works
Booth number: Rexford Room, Hilton Hotel
Circle (605) See ads pages 123, 125, 127

El Mar Plastics
Booth number: 1027
Circle (606)

FM Tubecraft Support Systems
Booth numbers: 1017, 1019
Circle (607)

Midas Audio
Booth numbers: 1005, 1007
Circle (608)

Optimix International
Booth number: 1009
Circle (609)

OSC Audio Products
Booth number: 250
Circle (610)

Shape Inc.
Booth numbers: 1419, 1420
Circle (611)

SPARS
Booth number: 1421
Circle (612)

Stellavox
Booth number: 1011
Circle (613)

R-E/P

Steinway, continued

Booth number: Brentwood Room, Hilton Hotel
Circle (564)

Studer Revox America
Existing products
A800 multitracks; A820 and A810 stereo mastering machines; models 900, 961 and 962 mixing consoles; TLS 4000 synchronizers; A710 and B215 cassette decks; model 2706 monitor speakers.
New products
Model A807 tape machine with microprocessor control of alignment and transport; model A12 transport; model 963 production console.
Booth numbers: 312, 314, 316, 318, 320, 413, 415, 417, 419, 421
Circle (565) See ads on back cover, 27

Studiomaster
See International Music Systems.
Circle (597)

Studio Master Systems
Existing products
Studio Master Plus with computer-automation software and hardware package to run on the Apple Macintosh.
New products
Three software packages for the Apple Macintosh, including Studio Master business software for billing and invoicing studio clients; Track Master track sheet and tape label software; and Outboard Master template software for storing outboard equipment environments.
Booth number: 909
Circle (566)

Studio Technologies
Existing products
AN-2 stereo simulator; RCU-1 recognition control unit.
New products
Microphone pre-amplifier.
Booth number: 741
Circle (567)

Sunkyong International
New products
Audiocassette tape; model C-O cassette housings.
Booth numbers: 736, 738, 837, 839
Circle (568)

Sunn
See Fender Musical Instruments.
Circle (598)

Swintek Enterprises
Existing products
Radio headsets and wireless microphones.
New products
Radio interconnect for RTS and Clearcom Series 300 systems.
Booth number: 1416
Circle (569)

Tannoy North America
Existing products
NFM-80 dual concentric desk-top monitor; FSM monitor; SR840 MOSFET amplifier.
New products
Series SGM-10B 10-inch monitor; 12X 12-inch monitor; 1000 15-inch monitor; Little Gold 12-inch, dual-concentric studio monitor.
Booth numbers: 438, 539
Circle (575) See ad page 85

Tape Automation
No information received.
Booth numbers: 802, 804
Circle (576)

TASCAM/TEAC Corporation of America
Existing products
Consoles and tape machines, including ATR-60 series with center-track time code.
New products
ATR-80 24-track machine; M-600 consoles; Porta-Two combination mixer/cassette deck; models 122C and 112 cassette decks.
Booth numbers: 412, 414, 416, 418, 420, 513, 515, 517, 519, 521
Circle (577) See ad page 49

Technics
Existing products
Reel-to-reel and cassette tape machines; preamps; amplifiers; loudspeaker monitors; professional Compact Disc players.
New products
Compact Disc changer unit for multiple CD access.
Booth number: 901
Circle (578)

Teldec
See Gotham Audio.
Circle (599)

Telex Communications
Existing products
Model 6120 cassette duplicator; wired and wireless microphones; modular intercom system.
New products
CD series cassette duplicator; portable cassette duplicator; model HT-400 wireless mic; RMR-4 4-channel wireless receiver.
Booth numbers: 735, 737, 739
Circle (579) See ad page 95

3M/Magnetic Media Division
Existing products
Audio, video and data tape products.
New products

401, 402, 403, 404, 405, 406, 407, 408, 410, 501, 503, 505, 507, 509, 511; Garden Room East, Hilton Hotel
Circle (556) See ad pages 116-117

Sound and Vision
No information received.
Booth number: 949
Circle (557) See ad page 175

Soundcraft
See JBL Professional.
Circle (596) See ad page 45

Sound Ideas
Existing products
Compact Disc sound effects library.
Booth number: 632
Circle (558)

Soundmaster International
Existing products
Electronic audio post-production control software to run on IBM PCs.
New products
Integrated Editing System that incorporates Syncro time code synchronizer module for use with Soundmaster control software.
Booth numbers: 641, 643
Circle (559) See ad page 147

Sound Technology
Existing products
Model 1510A tape machine and audio test system; model 1710A THD analyzer.
New products
Model 3000/3200A programmable analyzer; model 1530A MTS stereo analyzer and monitor; MSAT multichannel switching system; ATS audio test software.
Booth numbers: 808, 810
Circle (560)

Soundtracs
Existing products
T, 8-16, M and MR series of modular consoles.
New products
MC series stage monitor console; FM series modular mixer; MIDI series in-line console; CP6800 console with programmable routing and muting.
Booth numbers: 740, 742, 841, 843
Circle (561) See ad page 63

Stanton Magnetics
Existing products
Phono cartridges, styli, record care products and headphones.
New products
PBR series announcer headphones; model 30M/SR headphones; model 310B stereo phono pre-amplifier; models 981HZ Mk II, 981LZ Mk II, 881S Mk II and 681EEE Mk II cartridges.
Booth number: 120
Circle (562)

Star Case Manufacturing
Existing products
Audio, video and broadcast flight cases.
New products
Enhanced range of colors for existing products.
Booth number: 820
Circle (563)

Steinway & Sons
Existing products
Acoustic pianos.
New products
Model B 7-foot grand piano.

Samson, continued

transmitters.
New products
Stage series wireless system with model SR-1 receiver.
Booth number: 1407, 1408
Circle (545)

Sanken Microphone/Pan Communications

Existing products
CU-41 CD double-capsule condenser mic; CMS-2 MS-stereo condenser; CU-31 and -32 condensers.
New products
CMS-7 and 7H MS portable stereo condenser microphones.
Booth number: 750
Circle (546) See ad page 161

Schoeps/Posthorn Recordings

Existing products
Coincident, ORTF stereo and hand-held vocal condenser microphones.
New products
Tube studio microphone, with adapter ring for using current capsule.
Booth number: 640
Circle (547) See ad page 155

SCV Audio

Existing products
Phase polarity checker; third-octave equalizer; crossover networks; mic splitters.
New products
Model PC80 phase polarity checker; model MD8 rack-mount direct box.
Booth numbers: 1414, 1415
Circle (548)

Selco/Sifam

Existing products
VU and PPM meters; Collet and push-on knobs.
New products
Dual-movement indicator for peak program meters.
Booth number: 806
Circle (549)

Sennheiser

Existing products
Studio and sound reinforcement microphones, including MKH40 P48 condenser and MD409 U3 dynamic drum mic; HD 230 headphones.
New products
MKH20 P48 omnidirectional condenser mic; MKE42 PU cardioid condenser gooseneck mic.
Booth numbers: 237, 239
Circle (550)

Shure Brothers

Existing products
Microphones and mixing consoles, including FP32 portable stereo and FP31 portable mono mixers; SM2 headset.
New products
SM89 shotgun condenser microphone; FP42 portable 4-input stereo mixer.
Booth numbers: 107, 108, 213, 215
Circle (551)

Sierra Audio Acoustics

Existing products
Acoustic design, construction and consulting.
Booth number: 1025
Circle (552)

Skotel

Existing products
TCR-80V/114 time code reader; TCG-80N-005/004 time code generator.
New products
TCR-111 LTC reader, which reads time code at 1/40th to 80X normal play speed; TCR-112 LTC reader.
Booth number: 947
Circle (553)

Solid State Logic

Existing products
SL4000-E series master studio system; SL5000-M series production system; SL6000-E series stereo video system; Studio Computer with Total Recall and AutoScan; stereo submixer; programmable equalizer; events controller; real-time system and integral synchronizer; master transport selector.
New products
Advanced Total Recall with AutoScan; liquid-crystal bargraph metering; additional audio and control cassettes for modular SL-5000M series.
Booth numbers: 226, 228, 230, 232, 327, 329, 331, 333
Circle (554) See ad pages 120-121

Sonosax

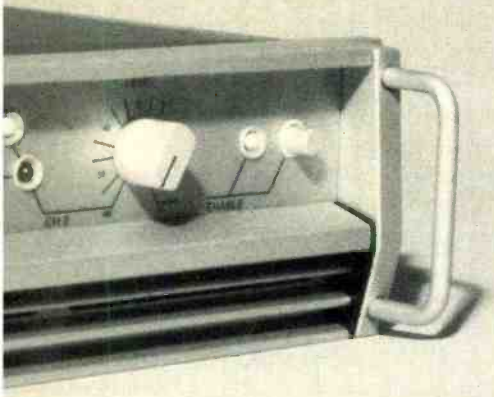
See Professional Sound Corporation.
Circle (574)

Sontec Electronics

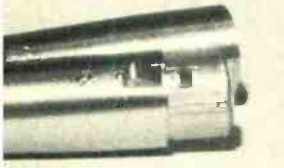
Existing products
Support electronics for disk mastering systems; signal processors.
Booth number: 946
Circle (555)

Sony Broadcast Products

Existing products
Microphones and wireless mics; mixers; mixing consoles; analog recording systems; portable tape recorders; digital audio recording systems; Compact Disc mastering system; digital audio mixing system; compact PA system; conference system.
New products
C-535P/536P condenser microphone; MXP-2000 modular broadcast production console; CDK-006 automatic Compact Disc loader; TC-D5PROII stereo cassette recorder; DFX-2400 sampling rate converter.
Booth numbers: 300, 302, 304, 306, 308, 310,

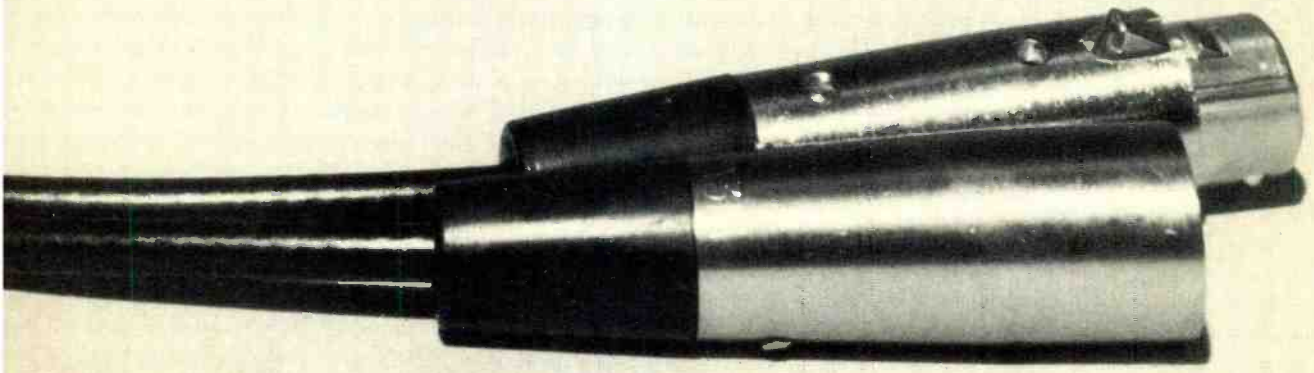


TO GET THE COMPLETE STORY YOU'VE GOT TO GO THROUGH CHANNELS



A superior amplifier has to be experienced firsthand. All the overplayed adjectives used to relate sonic quality of audio equipment in print become moot once the truth leaves the speaker.

Your nearest Crown dealer has the complete story at the flip of a switch.



1718 W. Mishawaka Road, Elkhart, IN 46517 (219) 294-8000

Circle (76) on Rapid Facts Card

crown® International

Automated audio cassette handler and packager.
New products
King winder cassette orientor; cassette box inserting equipment; cassette high-speed rewinder.
Booth number: 1406
Circle (524)

Panasonic Industrial/RAMSA
Existing products
Mixing consoles, loudspeakers, amplifiers and microphones.
New products
Series WR8428 recording and mixing consoles; WS series loudspeakers; recording and sound reinforcement microphones.
Booth number: Room 209, Convention Center
Circle (525) See ad page 35

Peavey Electronics
Existing products
Wireless microphones; IDL-65 digital delay line; PVM-38/PVM-45 dynamic microphones; PCS speaker and processor system; Audio Media Research recording and mixing products.
New products
Model V4X variable 4-way frequency dividing network; PEP 4530 programmable effects processor; DECA/724 power amplifier.
Booth number: Larchmont Room, Hilton Hotel
Circle (526)

Penn Fabrication USA
Existing products
Steel, plastic and aluminum hardware for loudspeaker cabinets and rack enclosures; acoustic foam; microphone stands.
New products
PAR64 and PAR56 lighting cans.
Booth number: 939
Circle (527)

Penny & Giles
Existing products
Slide, rotary and motorized faders; joystick and T-bar controllers.
New products
Motorized and T-bar faders.
Booth number: 209
Circle (528)

Pro Co Sound
Existing products
Models DB-1 and DB-4 single and quad boxes; Helix multipair audio cables; Excelline and Linfeline multitrack and sound reinforcement cables.
New products
Rack-mount Bantam patch bay systems.
Booth number: 748
Circle (529)

Professional Audio Services and Supply
Existing products
Dealer for Amek consoles.
New products
BC-II broadcast console.
Booth numbers: 703, 705
Circle (530)

Professional Audio Systems
Existing products
TOC series loudspeakers with constant coverage horn.
New products
PI series flying sound system with constant coverage horn.
Booth numbers: 246, 248
Circle (531)

Professional Sound Corporation
Existing products
Import and export distributor of Sonosax mixers; manufacturer of communications modules; slating systems; duplex monitoring systems; meter bridges; crystal sync oscillators; portable sound carts; mic power supplies; passive stereo matrix bow.
New products
Sonosax SX-S, SX-T and SX-TR portable stereo mixers.
Booth number: 1409
Circle (532)

Publison America
New products
Infernal Machine 90 (with remote control); Fullmost de-esser; CL-20C compressor/limiter.
Booth numbers: 634, 636, 638
Circle (533)

Pyral (Rhone-Poulenc Multi-Techniques)
No information received.
Booth number: 850
Circle (534)

Quad Eight Westrex
See Mitsubishi Pro Audio Group.
Circle (572)

Recording Engineer/Producer
Come by booth 918 at the L.A. Convention Center to meet **RE/P**'s editorial and sales staff, which will include Cameron Bishop, group vice president and publisher; Mel Lambert, editor; Dan Torchia, managing editor; Sarah Stephenson, associate editor; Stephanie Fagan, promotions manager; Mary Tracy, Midwest sales representative; Herb Schiff, Jason Perlman and Chris Woodbury, West Coast sales reps; and Stan Kashine, East Coast sales rep.

RE Instruments
No information received.
Booth number: 938
Circle (537)

Renkus-Heinz
Existing products
High-frequency compression drivers and horns; signal processing equipment; microprocessor-controlled PA systems.
New products
Smart System amplifiers and power racks with RH-1, RH-2 and RH-3 dynamic processors and loudspeaker systems.
Booth number: 700
Circle (538) See ad page 159

Rent Music International
No information received.
Booth numbers: 914, 916
Circle (539)

Research Technology International
Existing products
Audio and videotape; evaluators; cleaners.
New products
Model DV-5 dropout analyzer and time code generator for Compact Disc premastering.
Booth number: 948
Circle (540)

Roland Corp. US
Existing products
DEP-5 digital effects processor; MC-500 Microcomposer; JX-10 synthesizer; S-50 keyboard sampler.
Booth numbers: 440, 442, 541, 543
Circle (541) See ad page 87

Ross Systems
See International Music Systems.
Circle (573)

RPG Diffusor Systems
Existing products
Computer-designed reflection phase grating sound diffusors.
New products
ABF-FUSOR broad-band panel absorber.
Booth number: 642
Circle (542)

RTS Systems
Existing products
Pre-amplifiers, amplifiers and intercom systems.
New products
Series 2500 amplifiers; series 17 headset and beltpack; series 400 pre-amplifiers; series 800, 810 and 802 intercom systems.
Booth numbers: 844, 846
Circle (543)

Saki Magnetics
Existing products
Ferrite replacement recording heads for Ampex, MCI, 3M, Otari, Scully and Studer tape machines; 24-track metal replacement heads; JRF Magnetic Sciences replacement heads.
New products
2-inch tape heads for Ampex MM1200, Otari MTR90 and Studer A800; 1-inch heads for Teac 8516.
Booth number: 726
Circle (544)

Samson Products
Existing products
Broadcast and concert series wireless systems; hand-held, lavelier and instrument

A. Rupert Neve
See Focusrite
Circle (570)

Rupert Neve
Existing products
Consoles for recording, video post-production, sound reinforcement, broadcast production and film re-recording; NECAM 96 computer-assisted, moving-fader automation system; DSP digital signal processing console.
New products
DTC-1 digital signal processing console; V series recording consoles; model 8232 multitrack mixdown and production console.
Booth numbers: 200, 202, 204, 301, 303, 305
Circle (518)

New England Digital
Existing products
Synclavier II digital synthesizer system, with polyphonic sampling option.
New products
Direct-to-disk multitrack recording option with up to 16 tracks of disk-based recording; sample-to-memory option with stereo sampling; music printing software.
Booth numbers: Rooms 216 A & B, Convention Center
Circle (519) See ad pages 72-73

NTP
See Gotham Audio.
Circle (571)

Orban Associates
Existing products
Signal processing equipment, including parametric equalizers, compressor/limiters, de-essers and stereo synthesizers.
New products
Co Operator level-control unit that combines noise gate, compressor, leveler, clipper and high-frequency filter; programmable mic processor.
Booth numbers: 324, 425
Circle (520) See ads pages 78, 126

Otari
Existing products
Reel-to-reel machines in formats from 1/4-inch 2-track through 2-inch 24-track.
New products
DTR-900 Pro Digital-format 1-inch 32-channel digital tape machine; CTM-10 NAB-type cartridge recorder and reproducer; EC-210 time-code reader; EC-102 chase synchronizer and time-code generator.
Booth numbers: 110, 111, 112, 113, 219, 221, 223, 225
Circle (521) See ad page 79

Oxmoor
Existing products
DCA-2 digital control attenuator and remote control.
New products
DEQ-29 software programmable equalizer.
Booth number: 942
Circle (522)

Pacific Radio Electronics
Existing products
Distributor of broadcast, audio and video parts and accessories.
New products
Plugs, patchbays, tools, cleaners, chemicals and peripherals.
Booth numbers: 929, 931
Circle (523)

Paktec Automation
Existing products

Martin Audio London/Martin Audio America
Existing products
Sound reinforcement systems and accompanying electronics.
New products
Model VRS-800 single-enclosure PA system; model M-10 5-way system controller.
Booth number: 913
Circle (510) See ad page 93

Meyer Sound
Existing products
Sound reinforcement and studio monitor loudspeakers, including MSL-3, UPA-1A Ultramonitor and model 833.
New products
500 series 2-way loudspeaker system with built-in power amplifiers; MS-1000 power amplifier.
Booth numbers: 512, 514, 613, 615
Circle (511) See ad page 75

Mitsubishi Pro Audio Group
Existing products
Models X-850 and X-80 Pro Digital-format digital tape machines; Quad Eight Westar console; Westrex magnetic film recorders and reproducers.
New products
Model X-400 16-channel digital tape machine; model X-86 2-channel tape machine, configured in three formats, including 96kHz sampling rate.
Booth numbers: 526, 528, 530, 532, 534, 627, 629, 631, 633, 635
Circle (512)

Monster Cable
Existing products
Audio cables and connectors; Soundex acoustical treatment system.
New products
Prolink series 3 mic cable; Prolink series 1 tube-mic cable; 8-pair snake.
Booth number: 833
Circle (513)

Mosses & Mitchell
No information received.
Booth number: 937
Circle (514)

Nagra Magnetic Recorders
Existing products
Portable tape machines with center-track time code.
New products
TATC post-production tape machine; model IV STC stereo recorder with time code; and model 4.2 portable tape machine.
Booth number: 105
Circle (515)

Nakamichi USA
Existing products
Professional cassette decks.
New products
MR-1 3-head cassette deck with dual capstan, direct-drive transport.
Booth number: 812
Circle (516) See ad page 39

Neotek Corporation
Existing products
Consoles for recording, video post-production, broadcast production and film re-recording.
New products
24-bus consoles based on the Elite design.
Booth numbers: 540, 542
Circle (517) See ad page 9

Neumann
See Gotham Audio.
Circle (536)

JVC, continued

VP-900 digital audio processor; AE-900 digital audio editor; TC-900 time code unit.
New products
DM-900 digital audio processor; SU-900 synchronizer;
Booth number: Serrano Room, Hilton Hotel.
Circle (495)

Kenwood Communications
Existing products
Test and measuring instruments; Compact Disc encode/decoder jitter analyzer; oscilloscope and power supply.
Booth number: 933
Circle (496)

K&H
See Gotham Audio.
Circle (534)

King Instrument
Existing products
Model 793 automatic cassette loader; model 590 automatic VHS/Beta in-cassette tape loader.
New products
Model 2500 VHS cassette loader.
Booth numbers: 346, 348
Circle (497)

Klark-Teknik Electronics
Existing products
D series 8-input mixing consoles; AMR24 24-input console; S series monitor console; DN780 digital reverb with V2.0 software; DN716 digital delay; DN300 graphic equalizer; DN60 real-time analyzer; DN305B noise masking processor; Brooke Siren System compressor limiters, cable tester and crossovers.
New products
PMC402 4-input stereo mixing console; Jade-1 monitor loudspeakers; Brooke Siren FDS360 crossover.
Booth numbers: 608, 610, 709, 711
Circle (498) See ad page 35

Thomas Klotz Musikelektronik
No information received.
Booth number: 344
Circle (499)

Korg USA
Existing products
DW8000 digital waveform synthesizer; SDD2000 MIDI-capable digital delay and sampler; SDD1200 dual-channel digital delay; SQD1 MIDI sequencer and recorder; DVP1 digital voice processor; Stand Innovations products; Marshall amplification.
New products
SDD3300 programmable digital delay with MIDI capabilities and sampling options; DRV1000 digital reverb; DRV2000 programmable digital reverb; DDD1 dynamic digital drum machine with sampling options; DSS1 digital sampling synthesizer.
Booth numbers: 814, 816
Circle (500)

Kurzweil Music Systems
Existing products
K250 synthesizer; K250 Expander sound module; Midiboard keyboard control instrument.
New products
K150X rack-mount sound module; QLS computer software interface for Apple Macintosh.
Booth numbers: Verdugo and Del Mar rooms, Hilton Hotel.
Circle (501)

Kyric
New products
Audio tape cassettes and accessories.
Booth number: 907
Circle (502)

Lexicon
Existing products
Models 224XL, 200, PCM60 and PCM70 digital reverb and special effects.
New products
Model 480L digital effects system; model 2400 stereo audio time compressor and expander; model RD1 random-access recorder/editor.
Booth numbers: 424, 426, 428, 430, 525, 527, 529, 531
Circle (506) See ad page 51

Loft
See Gold Line.
Circle (535)

LTM Sound Department
Existing products
TV and film lighting products; microphone poles.
Booth number: 834
Circle (507)

Magnefax International
Existing products
Cassette and reel-to-reel duplication systems; model 7574 1/2- and 1/4-inch mastering tape; model LB74 1/4-inch mastering tape; LT72 1/4-inch mastering tape.
New products
As above.
Booth numbers: 349, 351
Circle (508)

Marshall Electronics
Existing products
Mogami and Cable-Worlds multiconductor snake cable in 2- to 48-pair formats; console cable in quad- and 2-conductor types.
New products
Tiny Tel patch cords using Mogami quad wire; quad loudspeaker cable; gold connectors.
Booth number: 751
Circle (509)

IF ANY OTHER "ONE BOX" SYSTEM WAS THIS ADVANCED, AND THIS COST EFFECTIVE, AND THIS WELL BUILT, WE MIGHT HAVE SOME COMPETITION.

HOW ADVANCED?

EAW "One Box" systems utilize advanced horn technology enabling 3 to 8 dB more acoustic output than even the costliest competitive vented systems. And the horns that EAW builds sound like no other horns. They all make use of exceptionally complex throat sections eliminating the beamy hollow sound typical of other horn loaded systems. EAW's horn technology is so advanced that it was selected by the Japanese Audio Consulting Society as the best high output system in the world after they compared it to 14 of the best vented and horn loaded systems from the US, Europe and Japan.

HOW COST EFFECTIVE ?

EAW is the largest manufacturer of "One Box" systems in the United States and our large computer controlled wood working stations enable us economies of scale no other system manufacturer can compete with. As a result EAW s KF Series "One Box" systems offer more performance per dollar than any competitive system. But the real bottom line for rental companies is return on investment, and this is where EAW has no competition. As a result of our reliance on sound engineering, not fad design techniques, our systems remain competitive for a long time. In fact EAW's first "One Box" systems built in 1978 for Carlo Sound are still in demand today, and Carlo Sound is still getting returns on their investment while lesser competitive systems have been long retired.

HOW WELL BUILT?

EAW KF Series are built from the finest materials available, including - exterior surfaces made from 18 ply to the inch cross-grain laminated European birch with the highest sheer strength of any commercially available wood product - horn flares injected with high density polyurethane foam for absolute acoustical integrity - high tech aircraft flying hardware - 100mm voice coil RCF Laboratory Series Cone Drivers featuring exceptionally low distortion and virtually no power compression - heavy duty vinyl dipped perforated steel grill assemblies for indestructible road worthiness. No other competitor has the experience to compete in this category.

WHY EAW KF SERIES?

Because we are the source for professional "One Box" loudspeaker systems. When we invented the "One Box" system we raised the expectations of an entire industry. And now, we are in the process of advancing the technology we pioneered.

The Source For "One Box" Systems.

You can maximize your next quarter revenues by contacting EAW today. Our loudspeaker systems are easy for your crews to use, as well as a positive attribute your customers are looking for. In fact, EAW KF Series have been proven to have a pull through effect on rental of a "B" system. So, if quarterly revenues are important to you, now is the time to contact EAW. To expedite the process, call our president, Frank Loykoat 617-620-1478.

EAW
EASTERN ACOUSTIC WORKS

"One Box" Systems That Defy Competition.

59 Fountain Street • Framingham, MA 01701
Phone: 617-620-1478 • IMC 1651

Circle (71) on Rapid Facts Card

Inovonics, continued

New products
Model 390 insert film record amplifier; high-speed tape duplication electronics, with programmed-control bias system; model 705EM/EMX stereo generator.
Booth number: 607
Circle (489)

Institute of Audio-Video Engineering
Existing products
8-month recording program, offering audio and video engineering and studio maintenance classes.
New products
Various elective classes in mixing, disk mastering and sound reinforcement theory.
Booth number: 943
Circle (490)

International Music Company/Akai Digital
Existing products
Distributor of Akai Professional, Fane, Ross Systems and Studiomaster products.
New products
Akai MG14D/ML14 rack-mount, 14-track tape machine, MPX-820 programmable MIDI-capable mixer and model 5900 multipoint digital sampler; Fane MD2050 1-inch compression driver, CX15 and CX12 coaxial loudspeakers; Ross series 2000 8-track recording console; Studiomaster 12M monitor console and series II programmable MIDI-muting console.
Booth numbers: 546, 548, 550, 647, 649, 651
Circle (491)

IPS
Existing products
Model C0 audio cassette housing and bulk tape; model V0 preloaded videotape.
Booth number: 848
Circle (492)

IQS
Existing products
Single-channel signal analyzers; reference microphone systems; audio analysis software.
New products
Model 416-A dual-channel signal analyzer, with DSP software.
Booth number: 702
Circle (493)

JBL Professional
Existing products
Compression drivers, horns, woofers and monitor loudspeakers, including concert series, and models 4400 and 4311 series; model 6215 power amplifier; Soundcraft recording, production and live performance consoles.
New products
Model 7922 digital audio delay with linear phase filters for loudspeaker alignment; Soundcraft Saturn 24 multitracks, model TS12 and 8000 series consoles and model S500 stage monitor console.
Booth numbers: 326, 328, 330, 332, 334, 336, 338, 427, 429, 431, 433, 435, 437, 439
Circle (494) See ad page 105

Jensen Transformers
See Boulder Amplifiers/Silver Lake Research.
Circle (504) See ad page 61

JRF Magnetic Sciences
See Saki Magnetics.
Circle (505) See ad page 144

JVC Company of America
Existing products

How To Hear Yourself

Getting enough vocals on stage. It's a problem facing every performer. In fact, most smaller bands simply can't hear the vocals clearly when they perform.

If your monitors simply don't cut it, you're probably wandering what to do about it. Well, EAW has a suggestion.

EAW's SM202P High Output Stage Monitor System. It Produces More Vocal Band Gain Before Feedback With Minimum Equalization And Power Than All Other Monitors Any Where Near Its Price.

With EAW's SM202P, performers can hear vocals and instruments accurately even at large concert stage sound levels. And nonengineers can set up effective monitor systems in very little time. Because you don't have to fix the monitors with equalization, the SM202P comes with flat frequency response right out of the box.

The SM202P is built around two RCF PRO Series 250mm (10 inch) drivers and a RCF high technology compression driver / horn. And, it includes an advanced Forsythe designed crossover assembly with asymmetrical slopes providing response that is within +- 2 dB over the entire vocal band, and that's without any equalization.

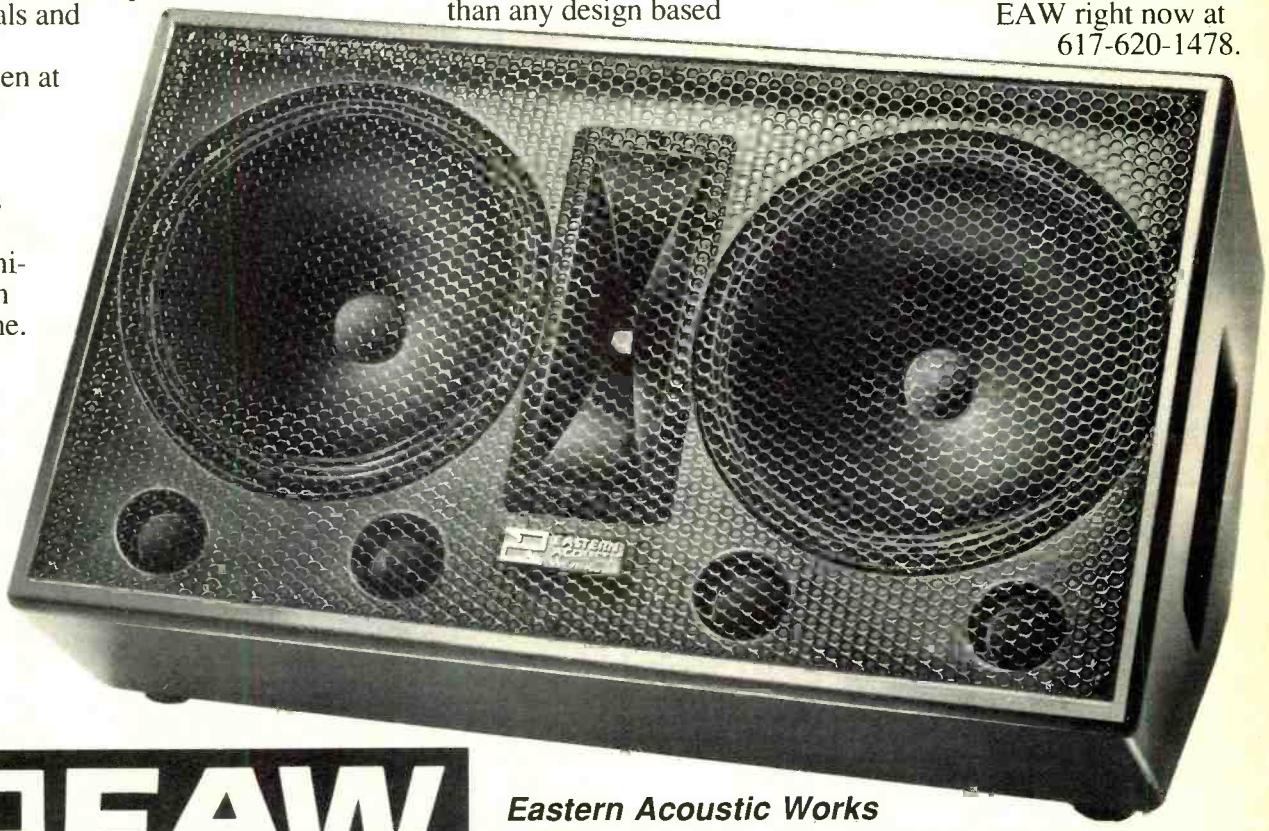
The SM202P's Output Capabilities Swamps The Competition, Too.

The use of two high efficiency 250 mm (10 inch) drivers provides considerably higher efficiency than any design based on a single 12 or 15 inch driver. The SM202P's 103 dB sensitivity is all usable as there are no peaks to create misleading high specs and lots of feedback as in all other monitors in this price category. And, don't forget the smaller 10-inch drivers provide smoother response above 800 Hz where the main components of vocals are located.

Check Out The SM202P At Your Nearest EAW Dealer.

There's a lot more to the SM202P story than we can talk about in an advertisement. And, it's just one of many EAW solutions to today's sound reproduction problems.

For more information and the name of your local dealer, call EAW right now at 617-620-1478.



EAW
EASTERN ACOUSTIC WORKS

Eastern Acoustic Works
59 Fountain St. • Framingham, MA 01701
Phone: 617-620-1478
TWX: 7103807630
IMC: 1653

Circle (69) on Rapid Facts Card

Gold Line, continued

Third- and one-octave real-time analyzers; Loftech TS-1 test set; Loft crossovers and quad noise gate; limiters; and MAD direct boxes.

New products

Loft test set with fine frequency-tune adjustment; electronic crossover with speaker phase adjustment and limiting.

Booth number: 443

Circle (481)

Gotham Audio

Existing products

Distributors of Neumann, EMT, NTP and Harmonia Mundi Acustica products, including microphones, mixing consoles, disk mastering systems, meters, amplifiers, digital reverbs and signal processors.

New products

Teldec/Neumann DMM Compact Disc mastering process; Harmonia Mundi BW102/21 digital equalizer; EMT and NTP analog signal processors; K&H loudspeakers.

Booth numbers: 520, 522, 524, 621, 623, 625

Circle (482)

David Hafler Company

Existing products

Models PM-125, PM-225, P-500 and P-505 power amplifiers.

Booth number: 518

Circle (483)

Harmonia Mundi Acustica

See Gotham Audio.

Circle (503)

Harrison Systems

Existing products

Sound reinforcement, teleproduction and music recording consoles; series 10 automated virtual console system.

New products

HDA-10 80Mbyte hard disk automation system for series 10 console.

Booth numbers: 618, 620, 622, 624, 719, 721, 723, 725

Circle (484)

Heino Ilsemann

No information received.

Booth number: 344

Circle (485)

Ibanez

New products

SDR1000 stereo digital reverb with 16-bit linear parallel digital processing, available with eight modes of operation.

Booth number: 831

Circle (486)

Industry West Electronics

New products

Model RJ-10 multi-input Adapter Box; Hypersound-110 portable sound PA system.

Booth number: 818

Circle (487)

Innovative Electronic Designs

Existing products

4000 series automatic programmable mixer; 5000 series computer-controlled audio system.

New products

6000 series rack-mount power amplifier.

Booth number: 732

Circle (488)

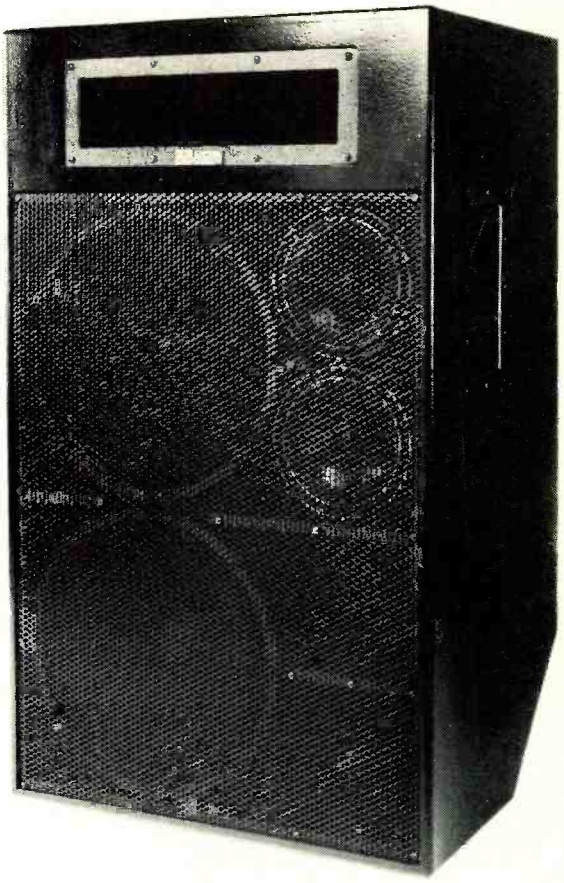
Inovonics

Existing products

Signal processing equipment.

Designing The Future

EAW's New FR253B Is The Future Of High Output Nearfield Loudspeaker Systems



You're looking at what's ahead for high output loudspeaker systems. At EAW, we call it "High Definition Systems".

The FR253B offers features and performance that goes beyond any other brand's "state-of-the art" technology. That's because EAW has led the touring sound industry in system design for years, and now we are bringing our advanced technology to the smaller nearfield market.

What only EAW gives you today others will surely have in the coming years. A demonstration will convince you of the startling difference between EAW and what you're used to. You'll hear definition and depth, not the typical one dimensional sound. All you have to do is listen and the difference is obvious.

Technology:

Poly-Laminated 170mm cone mid range driver operating in the 450 to 3,500 Hz band for seamlessly smooth vocal reproduction, and new standards of distortion-free output.

Advanced third order crossover network employing asymetrical slopes for maximally flat power and phase response.

High technology compression driver utilizes cast reinforcing ridges in the diaphragm for extended high frequency response.

Performance:

Absolute response linearity for faithful tonal balance reproduction
+- 2 dB 55 to 14,000 Hz
+- 5 dB 30 to 20,000 Hz

Very high power handling 625 watts AES standard for unsurpassed reliability

More than 40 acoustic watts maximum output / 131 dB maximum sound pressure, more than enough for even the most demanding nearfield applications

Eastern Acoustic Works

Eastern Acoustic Works, Inc • 59 Fountain Street • Framingham, MA 01701 (617) 620-1478
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Circle (68) on Rapid Facts Card

Fane

See International Music Company.

Circle (480)

Fender Musical Instruments

Existing products

Mixing consoles in 6- through 12-channel formats; Sunn stage lighting products.

New products

Model 2235 and Sunn SPL7000 power amplifiers; RAM mixing consoles in 12- through 24-channel formats; Dynamix D2000, D3000 and D4000 mixing consoles; PMB series headphones.

Booth numbers: 849, 851

Circle (471)

Fife-Pearce Electronic Company

New products

Electro-matic degaussing equipment for reel-to-reel and cassette tape, videotape, computer tape or floppy disks.

Booth number: 950

Circle (472)

FM Acoustics

Existing products

FM300 and FM801 stereo amplifiers; FM1000 mono amplifier; FM236 stereo crossover; Forcelines transfer cable.

New products

FM236/4 linear phase electronic crossover.

Booth number: 847

Circle (473)

Focusrite

Existing products

Distributor of A. Rupert Neve products, including models 85110 and 850109 microphone pre-amplifier/equalizer.

New products

Rack-mount microphone pre-amplifier.

Booth number: 836

Circle (474)

Fostex Corporation of America

Existing products

Consoles; tape machines; time code synchronizers; interfaces; monitor loudspeakers; headphones; microphones; combination mixer/recorders.

New products

E-series tape machines in 2- through 16-track versions, with time code capabilities; FAME assembly editing software for Apple IIc and IIe.

Booth numbers: 500, 502, 504, 601, 603, 605

Circle (475) See ad page 41

Full Sail Center for the Recording Arts

Existing products

Vocational technical school.

New products

Education courses for the professional audio and video fields.

Booth number: 829

Circle (476)

Gold Line

Existing products

Educational Systems Laboratories
No information received.
Booth number: 1401
Circle (462)

Electro-matic
See Fife-Pearce Electronic Company.
Circle (478)

Electro Sound
New products
Model 4800 digitally controlled tape duplicating system, with a 2-speed master and slave with four independent banks for equalization, level, bias, tension and speed.
Booth numbers: 724, 825
Circle (463)

ElectroSpace Developments/PMI
Existing products
Signal processors; noise gates; auto panners; complimiters.
New products
Strate Gate noise gate; The Gate noise gate with internal digital trigger; Pressor comp-limiter; Spanner stereo auto panner.
Booth number: 549
Circle (464)

Electro-Voice
New products
MT-4 4-way sound reinforcement system based on Manifold Technology; N/D series microphones.
Booth numbers: 317, 319
Circle (465)

Elison
Existing products
Noise-reduction systems.
Booth number: 920
Circle (466)

EMT
See Gotham Audio.
Circle (479)

E-Mu Systems
Existing products
Emulator II+ digital sampling keyboard with hard disk storage; SP-12 sampling percussion and drum machine; EMax digital sampling keyboard and rack-mounted digital sampler.
Booth number: Glenwood Room, Hilton Hotel.
Circle (467)

Eventide
Existing products
Models H969 and H949 Harmonizer; SP2016 digital effects and reverb processor.
New products
Model BD980 stereo broadcast delay; MIDI version and retrofit kit for SP2016 processor.
Booth numbers: 313, 315
Circle (468) See ad page 5

Everything Audio
Existing products
Distributor for Sound Workshop, Trident, Harrison and RAMSA consoles; Adams Smith synchronizers; Lexicon, dbx and Valley People signal processing devices.
Booth numbers: 449, 451
Circle (469) See ad page 55

Fairlight Instruments
Existing products
CMI series III sampling, synthesizing and sequencing musical system; Voicetracker programmable pitch-to-MIDI converter; CVI real-time video effects synthesizer.
Booth number: Room 202, Convention Center.
Circle (470) See ad page 15