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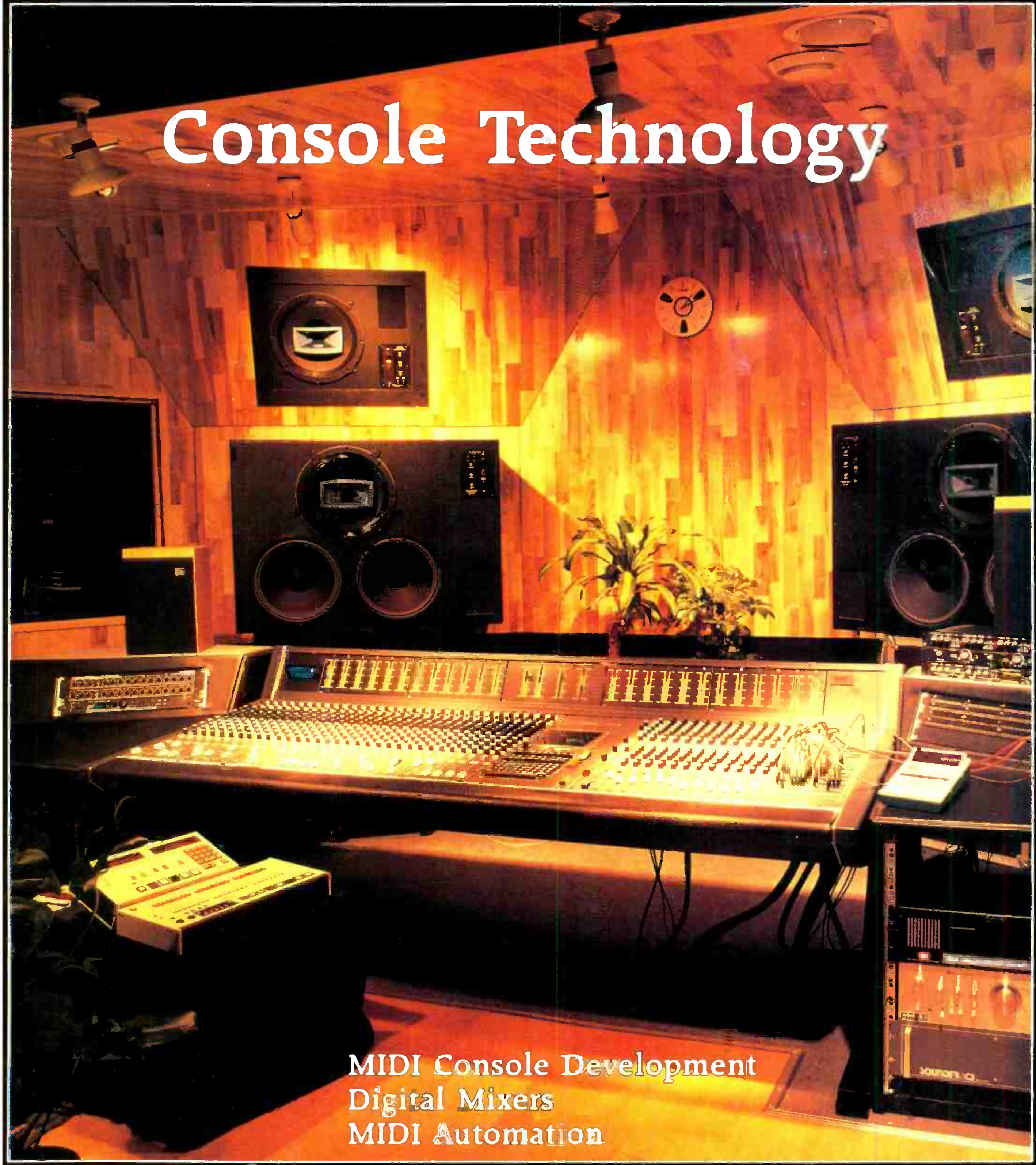

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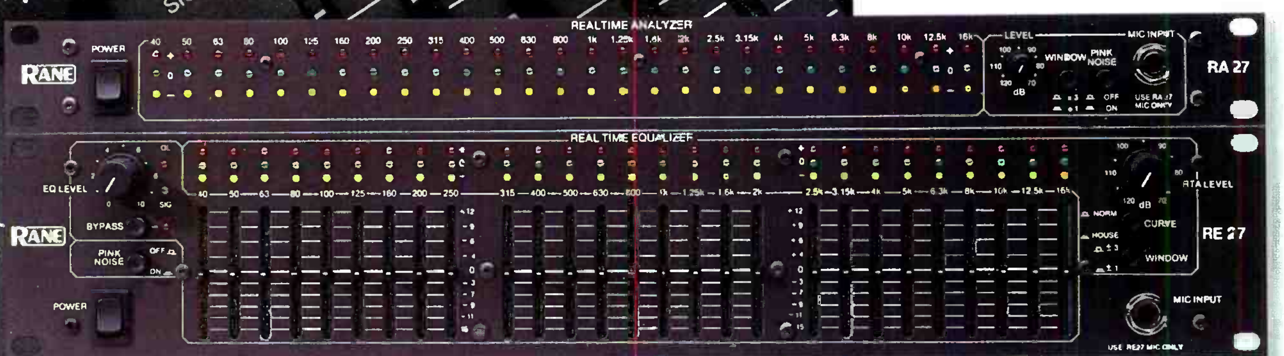

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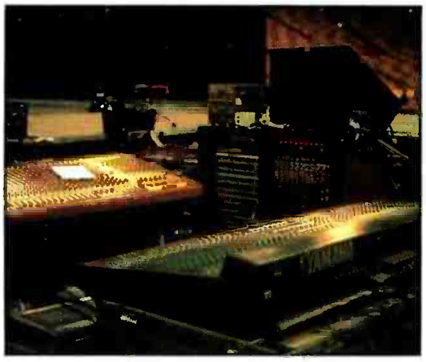

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Bjoern Eie's West Audio, a music and post-production studio in Oslo, Norway, sports a Soundtracs CP6800 console, supplied by Siv Ing Benum, Oslo.

Volume 20, No. 2
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EDITORIAL

Too Many Variables

In professional audio, we exist in an environment where the product of our efforts invites and even craves comparison. Yet listening evaluations are almost never conducted in a single, controlled environment—making accurate or objective comparisons nearly impossible. A recent experience will serve as a good example.

I was invited, along with several other professional and consumer audio journalists, to evaluate the performances of two digital tape machines. The evaluation was conducted using a well-designed concert hall and a separate and equally sophisticated listening room. Our evaluation experience consisted of two classical selections performed in the concert hall, followed by a playback session in the listening room. On the surface, this seems to be quite an acceptable, even ideal way to evaluate the performance results of the two digital tape machines.

Both machines were fed the identical output signal, sourced by the same overall signal path. Again, the *only* difference was the tape (because of different formats), the A/D/A converters and the tape transports themselves. So far, so good.

For reference I'll describe the setup for the second selection. The mixing engineer was in a booth, evaluating the sound over a B&W-801M speaker system. The microphone configuration centered on a Neumann SM-69 stereo mic, positioned roughly 10 feet in front of and 15 feet above the stage. In addition, there were four Schoeps CMT-56Us, one each on two violins, a viola and harpsichord, and two Neumann U-87s, one each on a flute and cello. These mics were placed 2 to 3 feet from their source instruments.

The hall itself had six or seven rows of permanent seats, the first of these beginning about 20 feet from the front of the stage. My seat was in the fourth row, about four chairs right of center (no, this does not allude to my political inclinations), and about 35 feet from the front of the stage. Clearly the sound I would hear would differ from the sound arriving at the various mics. Already there were variables, and a note had yet to be played.

Following the recorded performances, the audience was divided into four groups for playback evaluation. In the listening room, nine chairs were positioned in three rows of three. I sat front row right, to approximate my position in the hall. Only now, I was about eight feet from the source, and because of the position of the monitors in relation to the chairs, I was considerably to the right of center. Furthermore, the room volume was reduced to about one-quarter of the concert hall.

The playback began. What I heard from both machines was tonally acceptable, but I couldn't stop thinking that the instrument blend and the stereo perspective were not at all what I had heard earlier. Soon after the playback started, I moved to the center back of the room, where I thought I'd get a more representative balance. This helped the off-axis problem, but the perspective of the sound still wasn't all that close to what I remembered hearing during the live performance. Also, the engineer, from his booth mixing position, had obviously opted to bring up the cello in the mix and add some artificial reverberation to the program—resulting in a blend that didn't exist where I had sat in the hall.

By now there were five distinct variables: 1. Two storage devices, 2. Mic placement vs. concert hall listening position, 3. Concert hall listening position vs. the engineer's monitoring environment, 4. Instrument position and concert hall volume vs. speaker position and listening room volume, 5. Balance among instruments in the live ensemble vs. balance as captured through the multimicing technique.

This was a reasonably well-planned attempt at a critical listening session. I commend the manufacturer for having such excellent performance and listening facilities, and for risking a head-to-head comparison in front of the audio media. But, in many ways, the evaluation was doomed to failure from the start.

The problem was that the difference between the live performance and the recorded playback was more noticeable than the difference between the two machines. This was a factor I couldn't overlook.

Hindsight suggests that, for test purposes, it might have been better to use a 2-channel micing technique, positioned in the audience, and then to have placed the speakers on the stage for playback. Though not necessarily the best way to record or playback the performance, these steps are exemplary of what might be required to make a more objective comparison.

Rather than the relative merits of the devices I was invited to assess, I left the evaluation reflecting on the constant exchanges among studios, engineers, producers, and manufacturers regarding the quality of their systems or products. Exchanges that are almost always based on non-standardized evaluations and that seemingly overlook the fact that in every listening environment, the sound we hear will be different—not necessarily better or worse.

If your work sounds good to you, that's good. If it sounds good to you and your clients, that's better. If it sounds good to a wide variety of listeners, and translates favorably to an assortment of playback systems, that's great, and you've done your job well.

Michael Fay
Editor

RE/P

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LETTERS

Rosy predictions

From: Kenneth Glaza, K&R's Recording Studios, Southfield, MI.

It is not often I get upset with an article and even less often I will respond. However, I feel that Dr. William Moylan ["Employment Trends," December issue] is painting a much more glamorous picture of the recording industry than it deserves.

His description of great employment opportunities for people entering the audio field paints an encouraging picture "both in numbers and in varieties of positions; the amounts could be moderate to substantial over the next five years." But it is false!

I have a reasonably competent, diverse background in audio. I have been actively searching for employment for 14 months. I have on file 146 companies that I phone-qualified from a list of hundreds. Each received my resume because they showed an interest, and some received Synclavier software I was developing. Each understands that I am not interested in the responsibilities I had when owning a studio, and my objective is to return to school to continue my EE program.

After thousands, yes thousands, of dollars, I have received only one interview with a university, which elected to hire a less-qualified person because it felt I would not be satisfied.

I have found there are too many people looking for too few jobs. Consider the University of Miami School of Audio Engineering, dumping hundreds of "qualified people" per year into this country. How many students does the University of Lowell put into society?

On the other hand, I understand that there are roughly 298 Synclavier systems of substantial size to warrant an operator, and only one new system of that size was sold in the year 1988. And fewer than 60 of these are in establishments large enough to hire operators. How many new major shows were put together that need more than two or three competent audio people? Let's just stack qualified people against job openings.

When I compare the number of students, even the better-educated students, that are entering the audio field with great hopes of employment (especially because they have read an article such as published in the December issue), and I compare the growth of the industry as published in *Pro Sound News*, coupled with my experiences in seeking employment, my having owned a studio in Detroit and my visiting of studio owners in other cities (larger and smaller than Detroit), I feel you are misleading the public.

RE/P

Send letters to RE/P, 8330 Allison Ave., Suite C, La Mesa, CA 92041. Letters may be edited for length and clarity.

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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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NEWS

Electro-Voice restructures management

Electro-Voice has restructured its upper management personnel due to the company's expansion. Robert Pabst, E-V president, said that the addition of several professional sound companies to the Mark IV Audio Group has taken up a majority of his time and attention. Paul McGuire, vice president of marketing, will lead a management team that includes Roger Gaines, vice president of manufacturing, and Alan Watson, director of engineering. The group will focus on present and future operations and will let Pabst concentrate on strategies, planning and investments for the company.

Shape files Chapter 11, sells subsidiaries

Shape Inc. and two subsidiaries, Shape Optimedia and Gage Molding, filed for protection under Chapter 11 of the bankruptcy laws on Nov. 7, 1988. Prior to the filing, Shape also sold its majority ownership interest in Nu-Tec, Apex Engineering and Midwest Engineering. The company said the actions were part of its plans to divest of business entities that are not critical to its long-term success. At Shape Optimedia, production ceased on Dec. 9. Two-thirds of the work force was laid off or transferred in November.

ECM offers MIDI support

East Coast MIDI, a BBS designed for MIDI users, is on-line 24 hours a day, seven days a week. The system incorporates public domain software and supports the Atari ST, Commodore 64/128 and Amiga, IBM and compatibles, and Apple Macintosh computers. ECM membership includes musicians, producers, manufacturers and others from all over the world. The fee is $65 per year. ECM operates at 1,200/2,400/9,600 baud, with 9,600 baud users needing a Courier HST modem. Two data lines are available: 516-928-4986 and 516-474-2450. Voice line information is available at 516-928-4284.

APRS exhibition announced

The Association of Professional Recording Studios (APRS) has set the dates for APRS 89. From June 7-9, more than 5,000 qualified visitors are expected to attend the international exhibition in London at the Olympia exhibition center. Exhibitors should apply early for their first choice locations at the center. For more information, contact APRS at 163A High St., Rickmansworth, Herts WD3 1AY, England; 923 772907; fax 923 773079.

Ampex invests in production facility

Ampex Magnetic Tape Division has invested $17.4 million in a two-year program for a state-of-the-art production facility. The facility will enable Ampex to produce metal particle tapes, thin coatings and thin base films in high volume, as well as increase the production of its existing products. The program began on Jan. 1, and the production center will become operational in January 1991.

Oberheim moves manufacturing

Oberheim has moved all its manufacturing from Japan to Los Angeles, citing an increasing number of problems with its previous manufacturer, Hammond Suzuki.

The company cited a substantial delay in production and delivery of the Matrix-1000. The minimal number of units that were produced had to undergo additional Q.A. and rework before being shipped to dealers. The company said it will now have a greater control of release dates and actual deliveries of products.

KABA Companies restructured

Kenneth A. Bacon Associates has changed its name to KABA Audio Productions, prompted by the company being known through the acronym KABA.

KABA Audio Productions handles all phases of the creation of a marketable audio cassette, including original recording, mastering, duplication, graphic design, assembly and packaging. Toni Lynn is the sales manager.

KABA Research and Development will continue as the developer and distributor the KABA 4-track real-time and double-time duplication systems. Heather Smythe and Greg Glassco are the sales assistants.

The phone number for both companies is 800-231-TAPE or 415-883-5041 in California.

News notes

Deane Jensen of **Jensen Transformers** and Gary Sokolich of **Custom Sound Systems** presented the paper "Spectral Contamination Measurement" at the recent AES Convention. The distortion measurement, with 110dB dynamic range, generates a spectrum graph of cross-modulation products produced by a multi-frequency excitation signal. They have used the measurement to study non-linear distortion products produced by frequency-dependent group delay. Copies of the paper are available from Jensen Transformers, 10735 Burbank Blvd., North Hollywood, CA 91601; 213-876-0059; fax 818-763-4574.

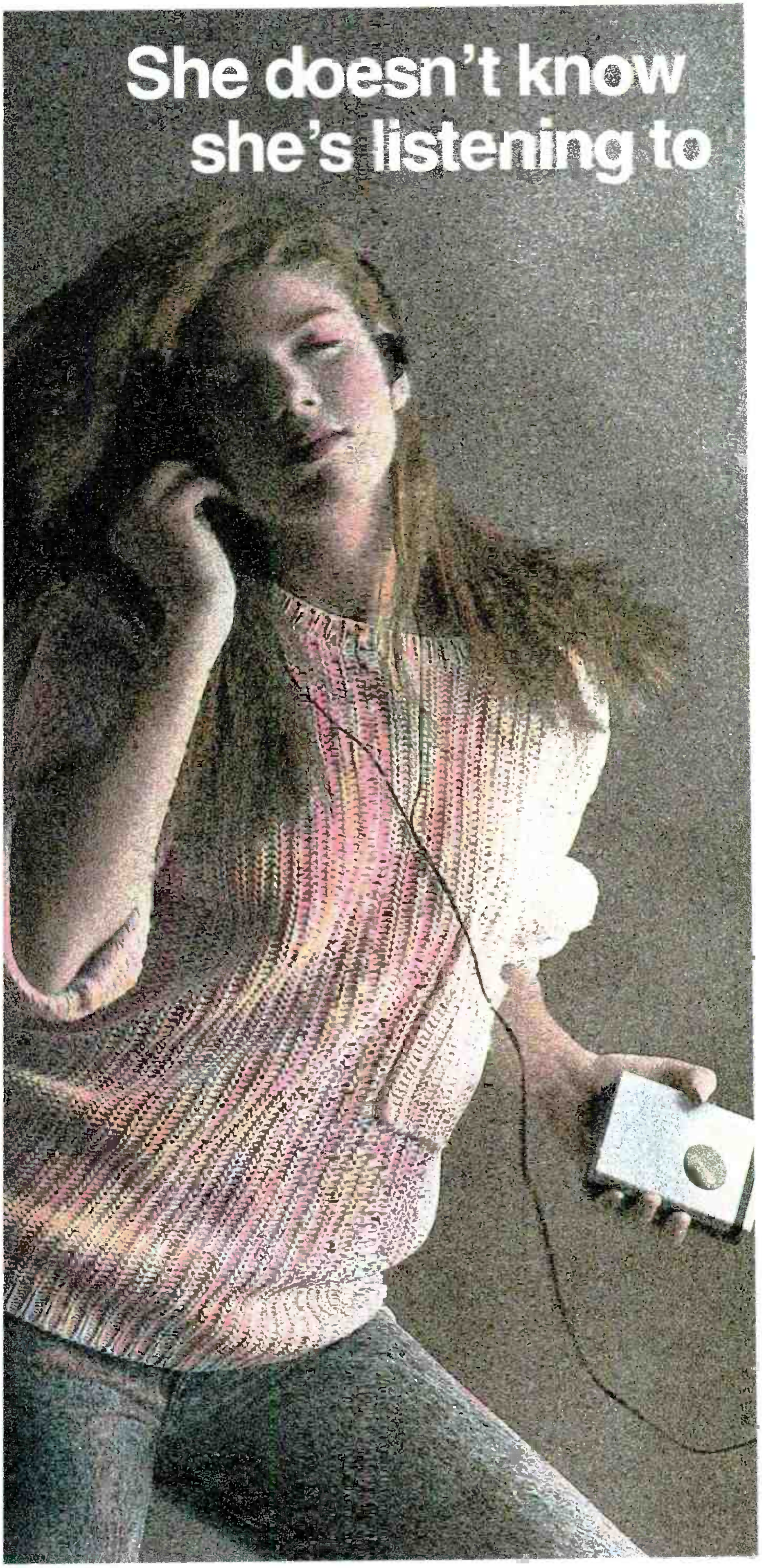
Acoustic Energy is a new London-based company manufacturing close-field monitors. Three models are available, designed by Phil Jones, technical director. For more information in the U.S., contact Acetrain Inc., 8300 Tuckerman Lane, Potomac, MD 20854; 800-527-7161 or 301-983-3966 in Maryland.

Stephen Paul Audio has been named a dealer for Neumann microphones, becoming the first dealer on the West Coast, through an agreement with Gotham Audio, the U.S. distributor. Similarly, SPA has been named an authorized dealer for Sanken microphones through Audio Intervisual Design, the U.S. distributor. Contact the company at 818-508-7720.

Responding to market demand, **Lexicon** has opened up several Opus demonstration rooms. The facilities are located at the company's Waltham, MA, headquarters, Martin Audio in New York and Lexicon's West Coast office in Los Angeles. The additions bring the number of Opus demonstration facilities to 10 worldwide.

Full Sail Center for the Recording Arts has broken ground for a new seven-studio complex in Winter Park, FL, a suburb of Orlando. Design of the 23,000-square-foot, two-story facility is by John Storyk, who designed the existing Full Sail facility. Gary Platt, Full Sail president, will be in charge of installation. Ted Rothstein will be in charge of electronic design. The facility will include a 60-input room with 48 tracks of analog recording; a 1-inch and 3/4-inch video and post suite; a 48-track mobile facility; a MIDI recording studio; and three Tapeless Studios. The complex is scheduled to be completed early this year.

Not content merely to celebrate its 20th anniversary, **QSC Audio Products** celebrated its 21st year in business in January. In keeping with its corporate culture, QSC decided to celebrate its 21st because everyone's 20th birthday was "no big deal. But the 21st! Now that was a birthday," according to Greg McVeigh, QSC's director of marketing. The company started in 1968 when Pat Quilter, vice



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NEWS

president of engineering, built a bass guitar amplifier for a friend. The company also reported that it closed fiscal year 1987-88 with a record-breaking 40% increase in sales compared with the previous year.

New England Digital has entered into a joint development agreement with Lucasfilm Ltd.'s Sprocket Systems to produce film and video sound editing products. The company has also been selected to participate in the research of the Media Lab at the Massachusetts Institute of Technology. The lab is equipping its recording suite with a Synclavier and a Direct-to-Disk system. MIT researchers will use the systems as a platform for developing special applications software.

Trident Audio has entered into an agreement with Digital Creations to use Digital Creations' moving fader automation in all consoles requiring fader automation. All new consoles will feature the automation, and existing consoles may have it retrofitted.

More than 150 editors and assistant editors have completed training on the CMX 6000 editing system at **CMX Laser Systems Inc.** The company, which supplies leasing, sales and rentals of the system, is in the process of completing a certification process based on a test conducted in conjunction with the Editors Guild. For more information on the program, contact CLSI at 6335 Homewood Ave., Hollywood, CA 90028; 213-462-2300.

Apogee Sound has expanded its international distribution network to handle increased export demand. The distributors are Entertainment Services Australia, Australia; Titan Audio, Belgium; Paul Farrah Sound, England; Orthophono Professionale Audio, Italy; Focus Showequipment BV, Netherlands; Sony Espana, Spain; and Audio Rents B-AG, Switzerland.

BASF AG has purchased an **Otari** T-700 MkII TMD laser-based high-speed video duplicator, to be used for the production of BASF chrome tapes for the Thermal Magnetic Duplication process.

Aphex Systems has upped the offer. The company's model 612 expander/gate ad offered $2,500 for people who found a better-performing gate. The offer has been increased to $10,000. The company has also named two sales representatives: Metropolis Audio Marketing, covering metropolitan New York, Westchester County, NY, and northern New Jersey; and Michael Chafee Enterprises, covering Florida and Puerto Rico.

A new manufacturers' rep firm, **Marketec,** has been created by Penny Russell and Gardner Ruffin. The firm's first clients are Morton Hi-Tek Furnishings and Vinten Equipment. Marketec's address is 3330 Purdue Ave., Los Angeles, CA 90066; 213-458-6930; fax 213-458-6936.

LD Systems provided sound and lighting for a Vietnam veterans benefit concert in Corpus Christi, TX, on Nov. 12 and 13. Equipment included 32 stacks of LD 2x4 cabinets, using a total of 64 boxes. Delays contained 16 LD 1x3 boxes. House mix console was a Ramsa 52-input WR-S852; monitor mix console was a 32-input Soundcraft 800B.

Bose has appointed two rep firms: Silver Peak Marketing, for the Rocky Mountain area; and Joseph P. Masseo Associates, for upstate New York.

Intersonics has been granted four basic-design patents by the U.S. Office of Patents and Trademarks. Two of the patents are new types of transducers; the others are related to the latest generation of acoustic levitation systems.

Beyer is offering its new Microphone Clinic to dealers as a marketing tool. The clinic is a computerized, portable anechoic chamber that tests frequency response, polar pattern and output level. A hard copy of the results is available via a support computer and a printer. The clinic can be reserved through Beyer sales representatives.

Apogee Sound supplied six AE-5 loudspeaker systems for the annual Jerry Lewis Telethon.

Martin America has relocated to 21000 Devonshire St., Suite 206, Chatsworth, CA 91311; 818-718-1031; fax 818-718-2886.

David Carroll Electronics has completed the audio wiring of the mix machine rooms at Lucasfilm's Skywalker Ranch. Current projects include audio-for-video facilities at One Pass Video and an audio production laboratory at Zoetrope Studios.

Technical Audio Devices has reduced dealer cost for its entire line of professional loudspeaker components. The reductions, which were made to make the components more affordable for a wider variety of applications, are between 15% and 25%.

Ampex Magnetic Tape Division celebrated its 30th anniversary at the AES convention in November. The division also announced that overall sales increased in 1988 for the fifth straight year.

Leonardo Software provided its Professional Librarian software to Modern Sound, which won an Emmy award for "Outstanding Sound Editing for a Series," for "Star Trek: The Next Generation."

Audio Kinetics has appointed HHB Hire & Sales as its exclusive U.K. distributor for the ESBus audio/video synchronizer for non-broadcast markets. Also, Studio System Technik has been named service spares dealer for West Germany.

Digital Audio Research supplied a Soundstation II for the recording of EMI's "Under Milk Wood," said to be the first fully digitally recorded spoken word recording. Additionally, the company has named three international distributors: THUM + MAHR, Germany; Promovisa, Spain; and 3M, France.

Gentner Electronics' first quarter net revenues for fiscal 1989 totaled more than $1 million, an increase of 50% from last year. Gentner has also named New West Audio as its product rep in Southern California and Hawaii.

Forte Music has relocated to a larger facility at 1951 Colony St., Suite X, Mountain View, CA 94043; 415-965-8880; fax 415-965-0508.

Act III Publishing has acquired **Mix Publications,** which includes *Mix, Electronic Musician* and the Mix Bookshelf.

McGohan Electronics has appointed two rep firms. SRT Marketing will cover Florida, and Admiral Sales will cover Texas, Arkansas, Louisiana and Oklahoma. RE/P

HOW I MIC

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SME7s on
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Shot on location, Stevie Wonder Concert, Valley Forge Music Fair, Devon, Pennsylvania.

ON TOUR WIT

MANAGING MIDI

By Paul D. Lehrman

Sequentially Speaking, Part 1

A sequencer is the heart of any MIDI setup. It is both the conductor and the orchestra. It's the composer's right hand, as well as the arranger and the orchestrator. In many cases, it's also the mixing engineer, and some day, if and when MIDI automation extends to mechanical transports, it may even graduate to tape operator. So having the right features in a sequencer is more important than ever.

Before I talk about those features, I should confess that I'm a software kind of guy. Some of the new hardware sequencers—especially the ones that use disk-based operating systems, so they have a useful life that's longer than the time between NAMM shows—are terrific, but they'll never equal the power and flexibility of a good computer-based system. Besides, can a hardware sequencer do your taxes?

So when I look at a sequencer program for a computer, here's what I like to see:

Graphic editing. When I was a kid studying for my ham radio license, I started learning the Morse code with a book that wrote it out in dots and dashes. An older friend pointed out that I was actually making the job harder for myself—the license test was an aural one, and I would do better to learn what code *sounded* like, not what it looked like.

The visual was adding to the process an extra step that was of no value whatsoever. Sequencers that force me to look at music strictly in terms of numbers I consider to be the equivalent of learning Morse code from a piece of paper. I'd much rather see graphics. Making a connection between lines and graphs perceived by the eye, and notes and other musical events perceived by the ear, is something the mind does readily.

Finding that slip of the fingers in a passage you just played is easy on a screen that shows note durations as line lengths and velocities as colors. Changing a note by picking up an object and moving it is intuitive—far more so than looking at a list of numbers, figuring out which are the offensive ones, and typing in new ones. Showing controller changes as moving lines makes perfect sense, and will do more to further the cause of MIDI-based automation than the most comprehensive and detailed lists of digits.

Paul D. Lehrman is RE/P's electronic music consulting editor and is a Boston-based producer, electronic musician and free-lance writer.

Numerical edit lists. On the other hand, numerical lists are essential in many circumstances. No matter how we dress it up, MIDI is, after all, still numbers. To find that stray controller command, that unpaired note-on, that weirdly arpeggioed piano chord, or that rough-sounding pitchbend, being able to see the numbers is invaluable. And certain kinds of mathematically based editing functions would be impossible if we didn't know what the numbers were to begin with. Graphics makes things easier, but numbers makes them *possible.*

Multiple active windows. A single long note can have all sorts of interesting things going on under it: pitchbend, modulation, breath control, volume. Looking at the note in a lone graphics window tells you almost nothing—you have to see what else is going on at the same time. Sometimes I like to be able to look at several tracks at a time, so I can see how the string and horn parts line up, and whether I should advance the strings a little because their attack is slower.

Therefore, I want to be able to have various windows open simultaneously. And so I don't get bogged down in keeping track of what's where, when I move one, I need the others to go right along with it.

Multiple input and output ports. MIDI's earliest critics were right: 16 MIDI channels and 31.25 kbaud just aren't enough. Of course, it took a while to find that out, but with today's multitimbral expanders, the channels get used up awfully fast, and when you've got flying controllers handling mixing, processing, and Lord knows what else, that bandwidth ain't what it used to be.

Using MIDI Time Code aggravates the problem even more: if MTC isn't on a MIDI line of its own, it can be subject to "jitter," as timing bytes get shoved out of the way to make room for longer commands, and this can wreak havoc with any device depending on them for sync.

The solution is not to rewrite the MIDI spec: it's to add more ports. Even having just one additional output port can relieve the pressure on a MIDI line dramatically, and there's no reason why today's computers can't support four, eight, or even more ports, giving hundreds of channels.

Regarding multiple input ports, remember when recording sessions involved whole groups of musicians playing at one time? Well, your console has more than one mic input, why shouldn't your sequencer have more than one MIDI input? And it's not just for the musicians—engineers can handle a console and several effects devices at the same time, and have all of the movements recorded in real time in the same pass.

MIDI Time Code capability. And speaking of MIDI Time Code, it's getting to the point where I consider any sequencer that doesn't read it to be unacceptable, particularly for film and video work. Tempo editing is the job of the sequencer (especially if it can be done graphically), not some hardware device that only allows a hundred moves and only has room for one song at a time. I recently participated in a record project in which getting a sequencer to sync to tape was a 5-step process: set up the tempo changes in the sequencer, stripe the tape with SMPTE, record a click track onto the same tape with the sequencer (running internal sync), play the tape back into a synchronizer to generate a tempo map, and then dump that map as system-exclusive information *back* into the sequencer for storage. It worked, but can you imagine going through all that every time the film editor comes in with a new cut?

With MIDI Time Code, the process is a little simpler: stripe the tape with SMPTE, and hit "Play."

MIDI File compatibility. A sequencer that doesn't accept MIDI Files is like a page-layout program that won't take input from a word processor. Yeah, you can use it, but who would want to? In this not-exactly-best of all possible worlds, every sequencer excels at some tasks and stinks at others. With MIDI Files, you can use the right program for a particular job, and then move the data into another program to take advantage of its peculiar wonderful features. Now if we can get Clipboard MIDI File compatibility going, we'll really have something!

Next month we'll go into more features I like to see in a sequencer—and some of them are pretty esoteric. RE/P

*Actual retail prices are determined by Individual Fostex Dealers.

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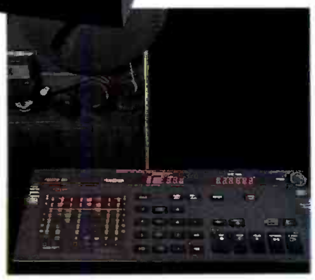

The entire front panel is removable, so you can control all functions — more than ever before — right from your working position. Once you set up your R8 it acts like any other computer. Tell it what to do, and it does it. Faithfully.

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AH

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Among digital equalizers, the Yamaha DEQ7 is unequalled. There's both digital and analog I/O.

SPARS ON-LINE

By Shirley Kaye

The SPARS Exam: Making the Grade

Following is a hypothetical conversation between two engineers during a session break at a prominent recording studio:

Susan Staff: "Hey, Ed, how'd your interview at the new studio in town go?

Ed Freelance: "Pretty good, I guess. I was able to answer most of the questions, but I was surprised when the manager asked me if I had taken the SPARS exam."

Susan: "What did you say?"

Ed: "I said that I didn't need to take any exams. I've been working as a first engineer at World Class Studios for more than a year. Why would I need to take a test?"

Susan: "But it's interesting that the manager asked you about it."

Ed: "Maybe you're right. Do you know anything about the exam?"

Susan: "Sure, I took the exam last year. The SPARS office gave me all the info and notified me when the exam was being administered in our region. I brought my exam results with me when I applied for work here. I can't say that the exam got me the job, but the manager was impressed. Some of my scores were great, and some weren't. He said that he'd help me work on the weak areas. He told me that taking an exam, even though it isn't a requirement, makes him think that I would be a highly motivated employee. I'm still here, so I guess he was right."

Shirley Kaye is executive director of SPARS.

The SPARS National Studio Exam was developed to give individuals a means of evaluating their own production studio knowledge. The test is an inventory tool for affirmation and demonstration of knowledge acquired through education, self-study and work experience.

In July 1984, SPARS signed a contract with ETS (Educational Testing Service) to develop a diagnostic testing program for production studio professionals. During the following year, ETS test development specialists and research scientists conducted on-site visits to studios and conducted interviews with studio personnel. These interviews became the basis for a questionnaire that was reviewed and modified by the SPARS Exam Advisory Committee and then sent out to 600 production studios. The staffs of these studios were asked to rate the tasks listed in the questionnaire in order of importance and relationship to success on the job in a professional work environment.

The results of the survey were then used by a specially appointed examination committee to develop detailed specifications defining the content, areas to be covered, the relative emphasis and the professional knowledge to be sampled by the examination.

The final exam was developed by ETS test specialists in conjunction with a committee composed of studio owners, educators, and production professionals recognized for their expertise in designated subject areas. For obvious reasons, the studio exam must be revised from time to time. The SPARS education committee is currently reviewing the timeliness and effectiveness of the test questions. The results of this process will be an updated test that will be implemented this fall.

The SPARS National Studio Exam gives you a clear picture of your own studio knowledge. If you choose, your scores will be sent only to you, or you can elect to have your exam subsection scores reported to the professional studio community to affirm your mastery of specific knowledge and expertise. This is valuable whether you are being considered for employment, advancement, or just want to share the information with your current employer. If you are applying to schools with an audio engineering program, you can request that your test results be sent to them as an aid to appropriate placement in basic or advanced courses.

Your subsection scores will give you a diagnostic look at just how you compare with your peers in this highly competitive industry. In a market flooded with job applicants, your results in the SPARS exam may give you the edge you need to advance your career.

Should you take the exam, you may decide to use the results in securing an internship position through the SPARS Internship Program. This national program has been developed in cooperation with our studio members nationwide to provide students with exposure to the studio environment over an extended period of time. SPARS interns are not paid, but feedback indicates that students found their experience in a real working environment was invaluable in preparing them for a career in the audio industry. The placement of interns is based on availability, educational background, work experience, and, in most cases, includes the SPARS exam scores. For further information on this program, either as a student or employer, call or write our national office.

General information

The examination consists of 200 multiple-choice questions to be completed in four hours. The three main content areas covered are Equipment Maintenance (77 questions), Session Planning and Setup (50 questions) and Equipment Operation (73 questions).

The diagnostic areas that are tested in the three main content areas are shown in Table 1.

To receive a registration form, contact Shirley Kaye, Executive Director, SPARS, 4300 10th Ave. North, Suite 2, Lake Worth, FL 33461; 407-641-6648. You will be notified when the SPARS National Studio Exam is scheduled to be administered in your area. Your registration form will indicate the location and schedule for the exam. Starting this year, the exam will be given in the spring (April 1 this year) and in the fall. All regional exam locations will give the test on the same dates. The registration form and a check for the fee must be returned by the deadline indicated on the form. RE/P

DIAGNOSTIC AREA	NUMBER OF QUESTIONS
System Interfaces, Synchronizers, Electronic Signal Processing	29
Properties of Sound Environmental Acoustics	18
Electronic Properties of Sound; Electronics Technology	24
Microphones, Loudspeaker Systems, Headphones	18
Analog Recorders/Reproducers	20
Digital Recorders/Reproducers	14
Video and Magnetic Film Recorders	19
Storage Media	15
Consoles	21
Music Theory/Terminology and Musical Instruments	22

Table 1. *The SPARS National Studio Exam consists of 200 questions covering 10 areas.*

UNDERSTANDING COMPUTERS

By Jeff Burger

Telecommunications

This month, let's demystify one of the computer's most powerful capabilities—telecommunications. For our purposes, this refers to the process of sending and receiving digital data via standard telephone lines. While telecommunications encompasses other applications such as fax, credit card verification and electronic banking, we'll take a look at the more tangible application of connecting your computer to the world.

Before we get too far into the mechanics, why would we want to plug our personal computers into the telephone system in the first place? The first application is to exchange files with friends and coworkers. And what better example than this column itself! I write this column on a Macintosh in Los Angeles. *RE/P*'s editorial office in La Mesa uses IBM-compatible PCs to create the final text that goes to the typesetters. Instead of mailing a printout to La Mesa, I can send the text from my computer, across the phone line, directly to the one sitting on Mike Fay's desk, thus eliminating both mailing time and retyping.

When two users are on-line with compatible systems, the potential becomes even more interesting. As discussed earlier, ASCII limits us to the alphanumeric characters that make up a text file. Compatible computers usually provide for the telecommunication transfer of files that are more sophisticated than ASCII text and unique to that system. This more or less entails sending binary files—a stream of bits that the receiving computer doesn't try to interpret as ASCII.

As an example, let's say you and a friend in another state both have a Macintosh, a DX-7 and the same patch librarian software. You can exchange DX-7 sounds by sending the librarian's file as binary information. A songwriting or production team can even send compatible MIDI sequencer files across town or around the world...so don't laugh any more when someone talks about phoning in their parts.

To clear up any misconceptions, however, you might actually be successful at sending an Atari spreadsheet file to an Amiga, but unless a global data file standard (that transcends the individual systems) is employed, the Amiga spreadsheet software won't be able to read the file.

The other enticing aspect of telecommunications is being able to dial up one of the "big computers in the sky." Hundreds of on-line databases are accessible with massive amounts of information, ranging from generic to hobbyist to highly specialized. These services typically require a subscription fee, in addition to per-minute on-line charges. Access is usually gained by dialing a local computer system number, which then ties you into these remote mainframes—without incurring long-distance charges. The on-line cost factor is often based on the time of day, with typical daytime rates being $22/hour and $8/hour at night.

Let's capsulize a hypothetical on-line session. Dial the local access phone number through your computer's modem. Once you've logged on, enter the access number of the on-line service you desire. Once you're greeted at the electronic "door," you must log on by giving your ID number (your electronic "handle" or address) and password. Now we're presented with a menu of choices, such as mail, bulletin board, travel, gateway, news, conference, SIGs, electronic mall or quit.

You can now navigate around the system by going into further submenus or back out to the main menu. Electronic mail essentially means that each subscriber has an electronic mailbox in which other subscribers can leave messages or text files. Since I want this article to be accessible to the editor whenever he wants, rather than being at the whim of our mutual schedules (a scary thought indeed), I can call the database tonight and leave this story in *RE/P*'s mailbox as an ASCII text file. Tomorrow morning Mike can log on while yours truly is still having a meaningful relationship with a pillow. His telecommunications software will let him download the file to his computer, where he can load it into his word processor and have his way with my poor defenseless words. He could extend this process by sending the amended file back to my mailbox for further refinement.

The bulletin board option is really just an electronic version of what the name implies. You can typically specify categories and/or keywords, and ask the computer to search for bulletins that match your criteria. For example, in much the same way you look through the classifieds, you might search for "DX-7" within the music equipment category. Other bulletins might express opinions, offer services or solicit the solution to a problem. Bulletins are, usually, automatically removed after a given interval to avoid a backlog of clutter.

On-line conferencing allows access to different subjects in the form of on-going discussions. You can join a conference in progress, read everyone's contributions and responses to date, and add your own thoughts.

Similarly, SIG (special interest group) files could contain anything from downloadable programs for your computer to banks of patches for you favorite synth. Travel gives you access to the complete OAG (official airline guide) so that you can be you own travel agent. The electronic mall consists of vendors offering a computerized version of what mail-order and the cable-based shopping channels have been doing for years.

News might address a very specific interest or access the complete UPI wire network, depending on the nature of your on-line service. Because of sheer volume, the news sections are usually navigated by specifying keywords. You might direct it to show you any stories that contain the words "copy protection" and "Congress" if you're interested in getting the latest Capitol Hill news on the R-DAT Copycode issue.

Finally, Gateway offers access to additional special services that carry a heavier on-line charge. One that comes to mind is the Songs gateway found on IMC's Esi Street service. Here, producers, publishers and A&R people notify songwriters about material they're looking for. That's worth an extra fee if you're a writer!

Local versions of these remote databases (called BBSs or bulletin board systems) have been launched by hobbyists and computer clubs all across the country. They are often dedicated to one brand of computer and sport a wealth information, downloadable, public domain files and camaraderie. RE/P

Jeff Burger is *RE/P*'s computer consulting editor and is president of Creative Technologies in Los Angeles.

Introducing the first 48-track digital recorder. The Sony PCM-3348.

It's 48 tracks on half-inch tape. On one machine featuring oversampled AD/DA converters and digital filters. Plus 12 newly developed LSIs, increasing reliability while reducing power consumption to 1.2 kW. All with complete half-inch compatibility—in the space of the industry standard DASH 24-channel recorder.


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It works with the tape you've got. Not only can you play any 24-channel DASH tape on the PCM-3348, but you can also add an additional 24 channels to your recording—while maintaining absolute integrity of the original channels.

And the machine is just as compatible as its tape. Because of its built-in synchronizer, connecting the PCM-3348 to Sony's PCM-3324 is simple. So is connecting it to the PCM-3324A—Sony's new 24-channel DASH recorder, which shares much of the remarkable technology we've put into the PCM-3348.

It works like no other machine. Because it's built like no other machine. The PCM-3348 features a newly developed transport that gently shuttles 14"

48 reels of tape with unprecedented speed. Two channels of selectable AES/EBU or SDIF-2 digital inputs/outputs are assignable to any of the 48 tracks. An internal 20 seconds of 16 bit memory can be reinserted back onto tape either manually or by external trigger. Vari-speed control of ± 12.5%. And a revolutionary digital/analog output that can be advanced up to 250 words.

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MIDI and the Mixing Console

By Peter Jostins

With such a variety of MIDI console products on the market, potential users are bound to be confused. This article outlines some of what is currently available and the way in which it operates.

If you think that fitting MIDI sockets to a mixing console is just done to sell a few extra units, think again. At present, more than a dozen consoles on the market are fitted with MIDI, providing extremely useful features such as MIDI muting, MIDI routing and snapshot storage. Add to this the number of MIDI add-ons such as fader automation and you have a vast array of products that provide everything from console automation to total recall, all via the humble MIDI sockets.

With such a variety of MIDI console products on the market, potential users are bound to be confused. Consequently, the following pages outline some currently available products and the way in which they operate, along with a what we can expect to see in the near future.

Peter Jostins is a London-based studio owner, engineer, producer, artist and consulting engineer for Soundtracs PLC.

MIDI muting

One of the simplest and most useful implementations of MIDI in the mixing console is MIDI-controlled muting, as available on the Soundtracs PC, Allen and Heath's Saber and Sigma, the Studiomaster and the Soundcraft 6000 series. With all of these consoles, it is possible to control the console muting via MIDI. So what? Well, if you can control the mute buttons via MIDI and you've got a MIDI sequencer, then you've got yourself a mute automation system at no extra cost.

On all the consoles mentioned, the mute buttons correspond to a note on a keyboard. For example, the mute button on Channel 1 corresponds to C, the mute button on Channel 2 corresponds to C sharp, and so on. Now, when a mute button is pressed, the console *sends* out the note designation along with a Note On message, in exactly the same way as a MIDI keyboard.

Conversely, when a mute button is released the console sends out the Note (C, C sharp, D, etc.) and a Note Off message. Similarly, if the console *receives* a MIDI Note and Note On message, it will mute the relevant channel.

Recording mutes into a sequencer for auto-

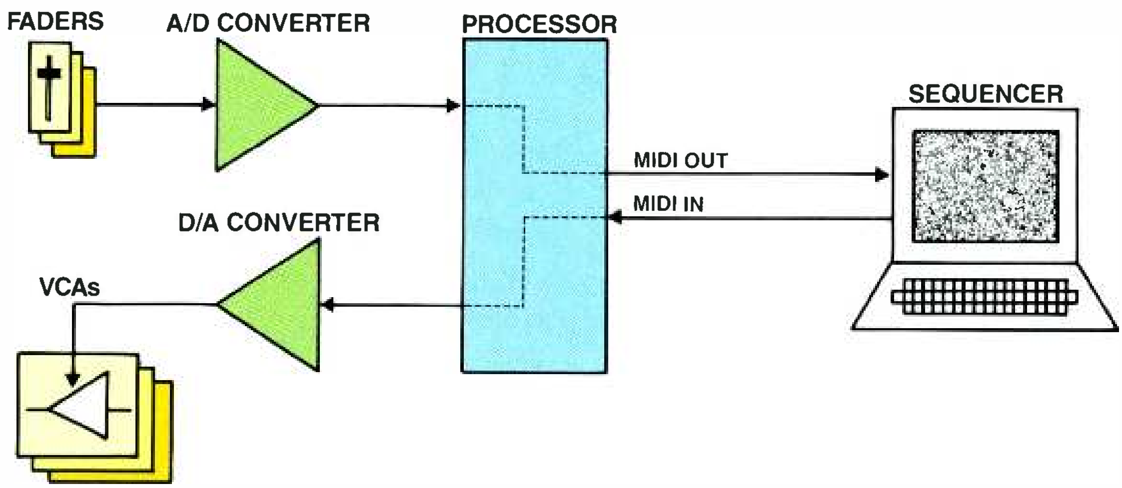


Figure 1. *Fader automation via MIDI.*

matic mixdown is simple. Just connect the console to the sequencer, put the sequencer into record and then record the console mutes exactly as if you were recording a keyboard. If you are using the console with a multitrack machine, then the machine is linked to the sequencer with time code, and the mutes are recorded in exactly the same way.

Of course, the whole point of an automation system is that you can build the mix up gradually, correcting mistakes as you go. With the MIDI system, the mix is built up by overdubbing mutes onto another track and then correcting them using step-time or real-time editing. By overdubbing more tracks, and bouncing them down if necessary, very complicated muting patterns can be achieved.

The beauty of using a sequencer as the mute automation computer is its inherent flexibility. For example, gating percussion with a MIDI mute system is simple because the mutes can be inserted in step time, then copied, repeated, shifted and delayed at the sequencer. With most conventional automation systems, each mute has to be entered manually. (Try muting in between bass drum beats throughout a song.) In addition, it is possible to write mute patterns into a sequencer so that the console produces gated keyboard or bass lines.

MIDI fader automation

Unlike muting, fader automation via MIDI is very difficult to implement. This is not because of any special hardware restrictions, since MIDI automation uses the same fader/VCA (or motorized fader) as a conventional automation system—along with a MIDI sequencer to store the data. (See Figure 1.) The main problem is that the MIDI cable can't easily cope with the massive amounts of data involved.

The reason for this massive amount of data is that the fader is continuously variable. In other words, it can be open or closed or anywhere in between. So, to transmit fader data over the MIDI link, you must use one of the continuous controllers available in the MIDI structure, such as aftertouch, volume, or pitch wheel.

But anyone who has used these with a sequencer knows that you don't have to record much of this continuous data before the music becomes hopelessly out of time. This is because MIDI data is transmitted serially through the cable and has to wait in line for its turn to be transmitted. Even worse, if the "line" becomes too long, then some of the data are simply discarded.

This can be demonstrated with a simple example. Each MIDI command lasts $320\mu s$, and it takes three of these commands to send a continuous controller message. Consequently, each continuous controller message takes $3 \times 320\mu s$, which is about 1ms. So, each time a single fader is moved, a series of continuous controller messages is sent out from the console with each message arriving at the sequencer 1ms later.

If 12 faders are moved simultaneously, then the time taken for the continuous controller messages to reach the sequencer will be 12ms. Of course, this is the delay from console to sequencer, and so you must add another 12ms for the return journey from sequencer to console, which results in a delay of 24ms, or about one frame, for just 12 faders.

Now, 1-frame accuracy is acceptable for fader automation systems, but for music, particularly where percussion is involved, 1-frame accuracy may not be acceptable. Consequently, if a fader automation system is used on the same MIDI network as the keyboards and samplers, the delays caused by the fader data can render the MIDI music data useless. This problem extends to MIDI muting because a delay of one frame on a mute is enough to cut the attack off the kick drum, and a hi-hat might disappear altogether.

To make matters worse, the delay is not constant, but instead, it varies according to how many faders are moving simultaneously. If it were constant, it would be possible to correct the mutes and music at the sequencer by adding predelay to each of them. But, because the delay varies, the predelay must be changed on each mute and musical note every time more fader movements are recorded.

> ***One of the simplest and most useful implementations of MIDI in the mixing console is MIDI-controlled muting.***

There are some ways of relieving this problem, such as spreading the data over a number of MIDI cables or reducing the fader resolution, but these are often either impractical or they degrade the system's performance. Nonetheless, a number of currently available MIDI automation systems work well, such as J.L. Cooper's MIDImation, Soundcraft's Twister, C Mix by Jellinghaus and Yamaha's moving fader automation for the DMP7, provided they are not required to control multiple fader movements simultaneously while the MIDI link is used for muting or music.

No description of fader automation would be complete without a mention of Yamaha's DMP7 and its younger brother, the DMP 11, whose controls (not just faders) can be controlled over MIDI. To overcome the delay problems mentioned earlier, Yamaha has abandoned continuous controllers in favor of MIDI note information along with its velocity data.

In other words, each control on the console is assigned a MIDI note and then the velocity of that note (from 0 to 127) determines the value of that control. Although this system does not have continuous control over the data, it is possible to transmit notes in close succession to approximate dynamic control, without severe delay problems. [See "MIDI-Based Automation" in this issue.]

MIDI program change and console recall

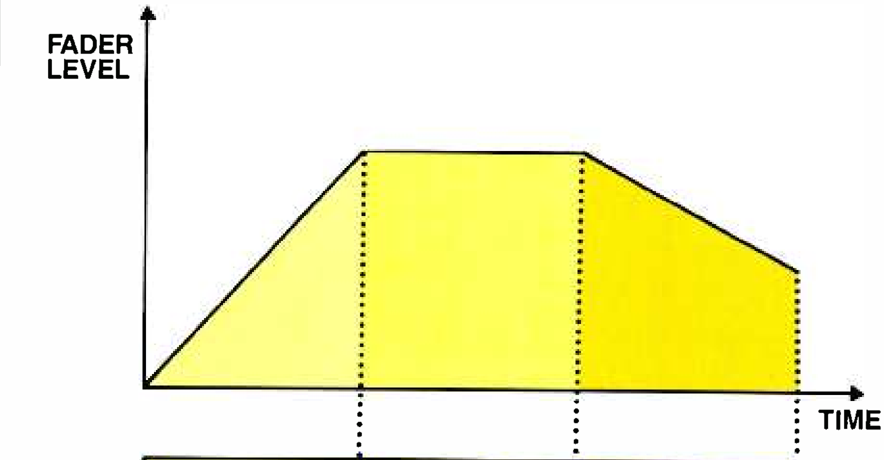
Every time you change the preset on your MIDI keyboard or change the effect preset on your MIDI effects device, it sends out a program change instruction over the MIDI network. If the console can respond to these program change instructions and change its internal settings, then it can be controlled remotely.

For example, if you have a keyboard rig and you want to change the EQ or effects during a performance, you can program the console, during rehearsal, so that when you change the preset on the keyboard during the performance, the console alters the EQ and effects to match.

Consoles that respond to program-change instructions vary considerably. Studio consoles, such as the Soundtracs PC and 6800 and the Soundcraft 6000, store mute and/or routing setups (called patches) in their own on-board computers. These patches can be recalled either from the console or via a MIDI program change instruction sent from an external MIDI device. In addition, they can transmit MIDI patch changes so that effects devices and keyboards can be controlled from the console. In general, this type of console employs electronic switches or FETs (rather than VCAs or DCAs) to mute and route signals in the console, and all the controlling hardware and software is on board.

On the other hand, the Akai MPX 820 and the Simmons SMP 8:2, which have been designed for keyboard and submixing, use VCAs or DCAs for every control on the console. In this way, the on-board computer can store a snapshot of all the console settings. At one time, up to 99 of these snapshots can be stored in memory and then recalled in response to a program change instruction arriving at the console.

Akai has taken this snapshot idea a stage further by allowing crossfades between snapshots. The result is dynamic mixing of all the console settings, in the same way as the Trident Di-An. You may wonder, why aren't there any delays with this system? The answer is simple. All of the snapshot and crossfade data are stored within the console and not the sequencer, so very little data is sent via the MIDI cable. In fact, the MIDI cable just carries the program change instructions that are used to recall the snapshots, so it rarely becomes overloaded.



FADE	UP	NONE	DOWN
CHANNEL	1	1	1
RATE	10dB PER SEC	0dB PER SEC	5dB PER SEC
TIME	00:01:00:00	00:01:05:00	00:01:10:00
	CUE #1	CUE #2	CUE #3

Figure 2. *Fader automation via MIDI Time Code.*

Regrettably, the Akai and Simmons mixers have been designed for keyboard mixing or submixing and, as yet, we still have not seen a resettable MIDI console for recording. In truth, we may never see such a console. The reasons for this have been explained on numerous occasions by SSL when it stopped development of its resettable console: VCAs can be noisy and cause distortion; motorized controls are expensive and sometimes unreliable; each control needs a meter or some display to show what it is doing; and the list goes on. However, although the totally resettable MIDI recording console may never exist, elements of it already do, such as programmable MIDI equalizers (Roland, Yamaha and ADA) and MIDI-controlled patchbays (Akai).

Most people who purchase MIDI consoles and MIDI automation systems do so because they already have a MIDI sequencer and can, therefore, add some form of console automation without breaking the bank. But for these people, a MIDI-automated console is of little use unless it has a large number of inputs to handle all of the individual outputs from their MIDI instruments and the outputs from a multitrack tape machine.

This problem is evident in even the most basic MIDI recording studios. For example, if you have a 16-track machine, a drum machine, a sampler and a couple of keyboards, you would have approximately 36 outputs that need mixing. Add to this a couple of stereo effects and even this fairly basic setup requires a console with 40 inputs!

In addition to this need for more input channels, the console must be reasonably compact to fit into a modest control room. Because of this, many manufacturers are opting to design in-line format consoles, since, for the same number of inputs, they are half the size of a split console.

Problems with MIDI in the console

Noise: Putting MIDI into a console is not as easy as you might think, because computers generate noise—lots of it. This means that the manufacturer has to mount the digital electronics well away from the audio, screen it and, if necessary, feed it with its own independent power supply. Even then, it takes hours of reworking PCB layouts and wiring looms to bring the console up to the professional specifications that are required.

With the amount of work involved, it is inevitable that some consoles don't make the grade. If you intend to buy a MIDI console, make sure that the salesperson "winds up" the volume (without program material going through the board) so that you can check for computer noise or switch clicks.

Software stability: Another problem with MIDI consoles is software stability. Many of you have already experienced bugs and crashes on computers. The same is true of any MIDI consoles. They, too, can crash or behave strangely if the software has not been properly designed.

The severity of the problem will vary from system to system. Some consoles have very complicated on-board computers, along with complex software routines, and are prone to bugs and crashes. On the other hand, some consoles provide little more than a couple of MIDI sockets, so there is less to go wrong. But, because these use third-party hardware and software (such as a sequencer), they are more likely to suffer from interface problems.

Unfortunately, there is no easy way of detecting these faults in the showroom. Before you buy a MIDI console or automation system, seek out people who own one and ask if they've had any problems with bugs, crashes or interfacing.

Cost: When it's possible to buy an 8-bit microprocessor for $2 from the corner shop, you might think that the cost of fitting MIDI facilities into a console would be negligible. It's true that the hardware costs are fairly low, but it's not these costs that significantly affect the price of the console. Instead, the price of the console is dictated by the software development costs.

For example, it might take two people a year to complete the software for a simple MIDI mute system. At a salary of $25,000 each, the price of the software development alone is $50,000. If the manufacturer projects selling 100 of these consoles, then the price increase of each console must be $500 just to cover the development cost. If you take into account licensing, prototyping and hardware fabrication costs, this amount could be double just for mute automation.

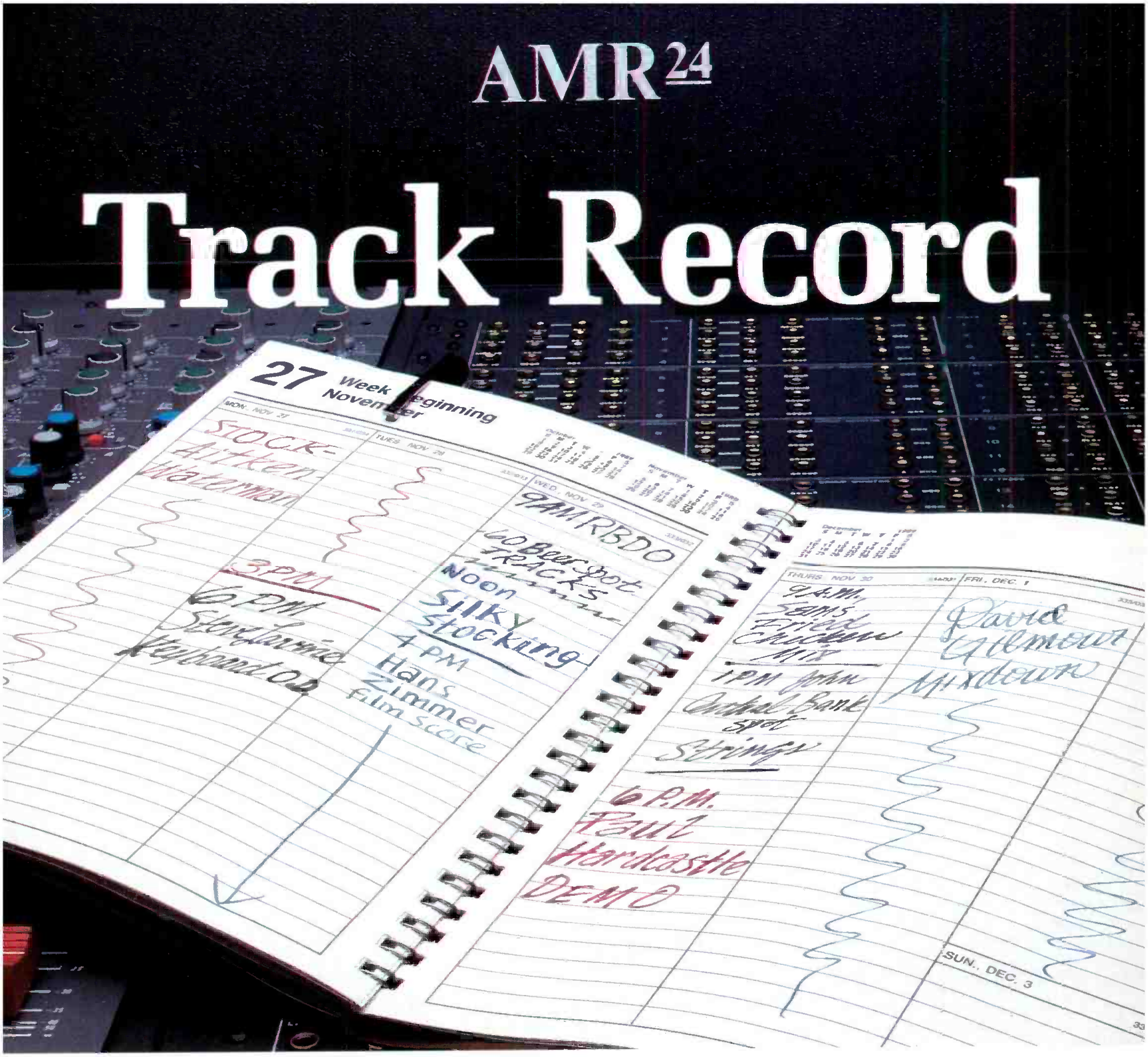
In the case of a more elaborate MIDI-resettable board, the cost of development is much higher and can only be justified if it has a potential for high sales volume.

The future

Increased baud rate: As I have explained, the main problem with MIDI is that it is not capable of transmitting massive amounts of data. However, the current MIDI spec *allows* for a baud rate of 61.5K (instead of 31.25), which doubles the amount of data that can be transmitted and will enable the more complicated devices, such as fader automation systems, to work better. However, this increase will not be enough to eradicate all delay problems within the MIDI network, and there are still many MIDI sequencers on the market that will not accommodate this increased baud rate.

MIDI Time Code: The next breakthrough comes with the introduction of MIDI Time Code (MTC). MTC can operate over the same network as MIDI, but it is not designed specifically for musical instruments. Instead of transmitting MIDI note messages, it consists of time code and cue lists. The time code portion of MTC is identical to SMPTE, and the cue list is simply a list of commands that tell the studio equipment what to do, and when.

Now, instead of the sequencer constantly sending out the usual-MIDI data (note on, velocity, etc.) it sends out cue lists to each of



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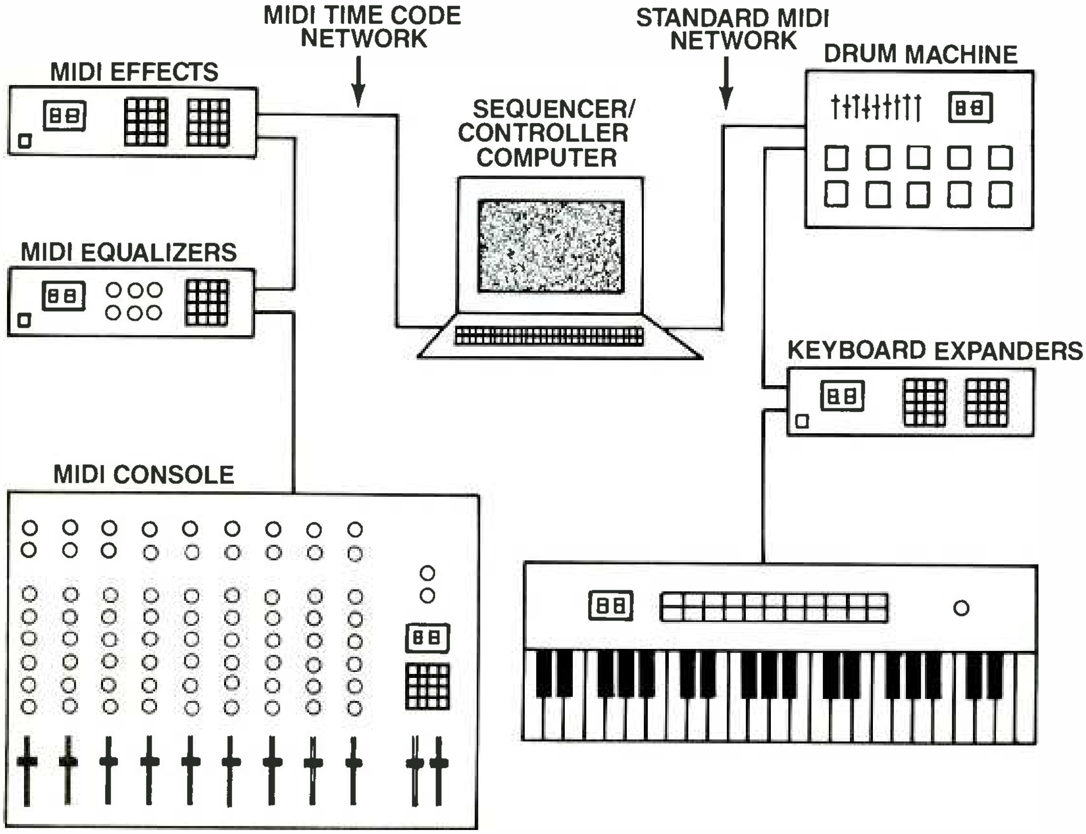

Figure 3. *MIDI Time Code and MIDI networks.*

the MIDI devices. These cue lists are sent prior to the event, or even at the beginning of the song, and then each device waits for the correct time to act upon them. Already there are products on the market that respond to MTC in this way, such as the programmable graphic equalizer by ADA.

Fortunately, this system of sending cues over MIDI can also be applied to mute, route and fader automation. To do this, the automation must consist of console snapshots, which are altered against time code. When the mix is replayed, the console receives its cue list, telling it which snapshots are required and at what time, and then it simply reads the time code and recalls the snapshots accordingly. In the case of fader automation, it will receive not only snapshot data within its cue list, but also fader rates. So a typical cue list for a fader automation system would be:

- Fade up Channel 1 at 10dB per second starting at 00:01:00:00.
- Stop fading Channel 1 at 00:01:05:00.
- Fade down Channel 1 at 5dB per second starting at 00:01:10:00, etc. (See Figure 2.)

Once you get into the realm of fader automation, you end up with a lot of data exchange. Although MTC is extremely resourceful, it is best to keep it away from the standard MIDI network. This has led to the general consensus that we will eventually have two MIDI networks: standard MIDI for keyboards, drum machines, etc., and MTC, for consoles, effects and EQ—controlled from a central computer or sequencer. (See Figure 3.) To this end, both Digidesign's Q sheet and J.L. Cooper's MAGI sequence, generate and read MTC. But to my knowledge, there are as yet no consoles on the market that respond to MIDI TC.

Console remote control: At one time, the mixing console was the focal point of the studio. But today, musicians and engineers are as likely to huddle around a sequencer as they are around a console. Realizing this, some manufacturers are busy developing remote control of tape machines, consoles and effects from the sequencer.

Already, we have seen software developer Steinberg Research develop a remote control facility for its PRO 24 that controls a Fostex E16 from the sequencer. In other words, when you rewind the sequencer, the tape rewinds as well. In addition, the Soundtracs CM and CP consoles have remote control of their routing, so that you can tell the console to record the bass drum onto Track 1 from a remote computer.

This concept of remote control is sure to take off in the next few years, so there will eventually be a dedicated computer (or at least software for your sequencer) that is in charge of machine transport control. Chances are it will link into the MTC network described above.

Conclusions

Now that MIDI has been accepted within the professional audio fraternity, its development and enhancement are sure to flourish. It is my belief that not only will we have professional console automation without the "professional" price tag, but we will have the security of knowing that the equipment we buy today will not be obsolete in six months. A standard, even a somewhat limited one like MIDI, is certain to make manufacturers work together.

RE/P

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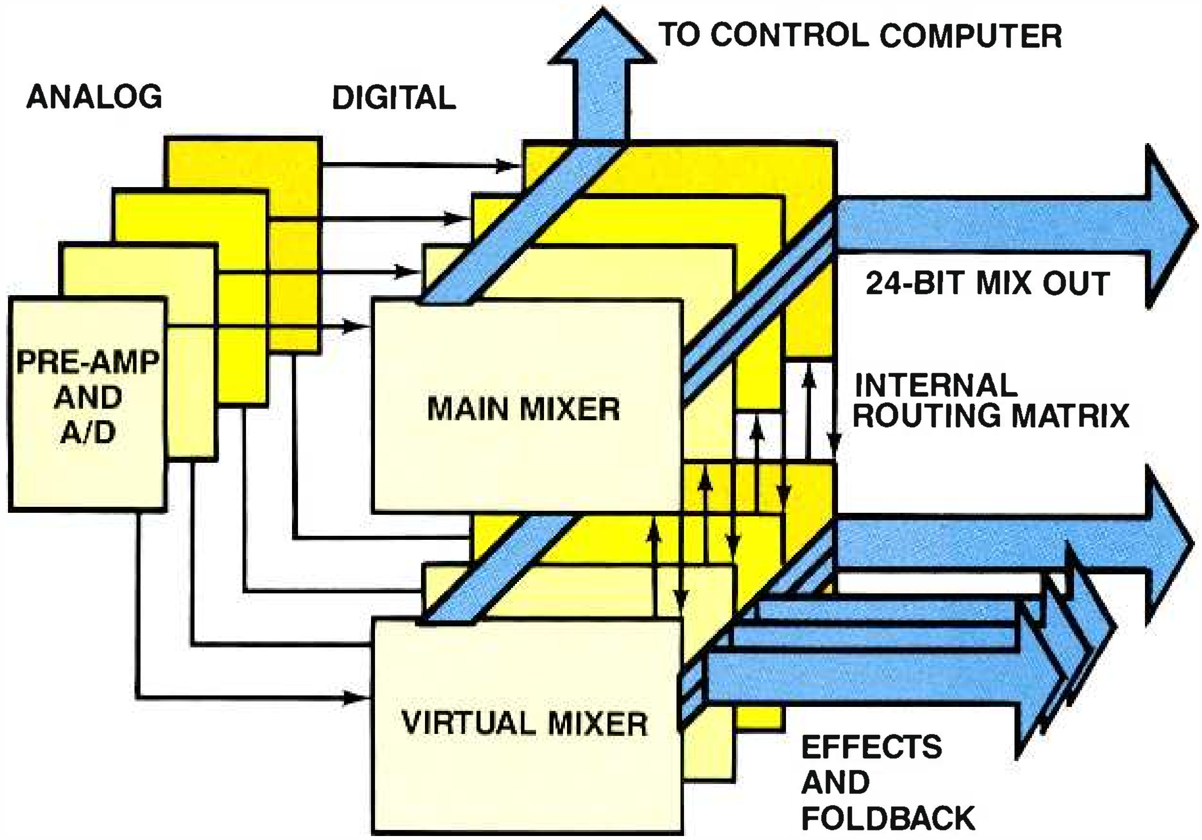
True Digital Audio Mixers

By David Shapton and Mark Mattingley-Scott

The development of all-digital mixing environments presents many new and interesting challenges to both system designers and end-users.



It would be quite reasonable for someone with a basic knowledge of digital audio to ask: Why don't all digital mixers sound the same? After all, 16-bit digital is 16-bit digital, and as long as the arithmetic in the signal processing chain is OK, shouldn't the signal coming out be the same as the signal that went in—plus some volume

David Shapton is head of audio at Digital Automation, Essex, England. He is also a free-lance engineer, producer, musician and composer.

Mark Mattingley-Scott is principal research engineer at Thorn/EMI's central research laboratories. He, too, is a musician.

control, equalization and, of course, mixing?

Despite being rather sweeping, these statements are quite correct. But what they don't tell you is that processes like equalization and volume control are actually rather difficult to get right in the digital domain, and can lead to design compromises and even mistakes.

The perfect digital processor would have an infinite sample rate, use an infinite number of bits (not just 16, 24 or 32 bits) and would be impossible to make. As a result, the digital technology we work with today doesn't provide us with perfect sound. The positive side is that even if digital sound isn't perfect, it can sound nearly perfect to humans. The results *can be* better than analog in just about every respect.

Analog systems suffer from many inherent problems. Conversely, digital processing is not exactly a bed of roses either, there being some areas where even leading-edge technology is struggling to equal the performance of analog designs. But because the problems inherent in the two technologies are so different, there is little point in digital emulating analog technology.

True digital audio mixing

What do we mean by "true digital" as opposed to merely "digital"? True digital means a mixing system in which the audio path is digital from the moment it enters the console to the production of the master tape. A simple and obvious test for this criterion is to count the number of A/D and D/A converters. In the actual signal path (excluding monitoring and foldback), there should be only one A/D and *no* D/A. In fact, the only D/A the signal should encounter is the one in the consumer's CD or DAT player. (Or better still, there would be D/A and a power amp in the loudspeakers!)

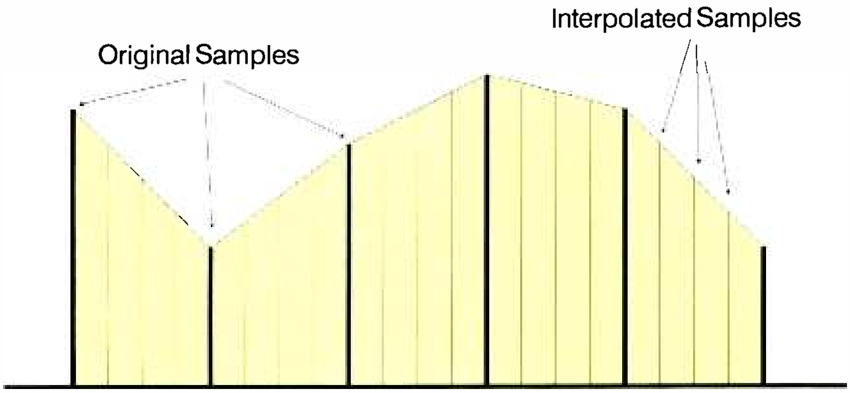
If a digital multitrack and a digital mixer are used, is that true digital? It depends. Even if digital signal processing effects are used, the system still might not be true digital.

To understand why this isn't necessarily the answer, we need to look at how digital effects work. Actually, we don't have to look any further than the input to a digital reverb, where, with one or two exceptions, we will find that the input is analog. So, even in the case of *digital* mixers, the signal has to go through a D/A to reach the analog input of the reverb, then through the reverb's A/D, where it is digitally processed, and then out through the reverb's D/A, and back into the mixer through its A/D.

And it gets worse. "Conventional" digital effects tend to use variable sampling as a means of signal processing. For example, the easiest way to achieve pitch change is to change the sample rate. You only have to look at the history of the digital delay line to realize that digital audio signal processing evolved with the actual sound quality taking a back seat, until high-capacity memory and high-speed microprocessors were cheap enough to incorporate into a system. Most early delay lines simply used DMA (direct memory access) controllers to act in conjunction with memory as simple shift-registers. They were 8-bit and of dubious audio performance, although they did allow phenomena such as infinite repeat for the first time. The bottom line is that "digital" effects do not fit into a digital domain studio unless they have a fixed sample rate.

Why not feed them into a sample rate converter?

Interpolation



Sampling rate conversion is the process of converting between two different sampling rates, for example, between the CD standard of 44.1kHz and DAT's 48kHz. Other applications include pitch change, varispeed, broadcast and NTSC Video Disc Audio Standard.

When you want to convert between two sample rates, it is first necessary to interpolate (multiply). (See Figure 1.) This is required to convert the input sample rate to a higher rate, and then decimate (divide) to convert from this new, higher sampling rate back down to the desired output sample rate. As long as the input and output sample rates can be written as a fraction (like $\frac{97}{116}$) then the conversion process is only extremely difficult. If the two rates do not have a nice relationship, or are constantly changing, such as with a flanging effect, then they are much more difficult to convert and more-sophisticated mathematical techniques have to be used. Here is an example of a relatively simple sample rate conversion problem.

To convert between 44.1kHz and 48kHz, the 44.1kHz must first be interpolated up, then decimated back down. Fortunately, this can be done in stages. The sampling ratio (R) can be written as:

$$R\,\frac{48{,}000}{44{,}100} = \frac{2^5 \bullet 5}{3 \bullet 7^2}$$

This can be factored out to give small individual changes in sampling rate, for example as:

$$R\,\frac{48{,}000}{44{,}100} = \frac{4}{3} \bullet \frac{4}{7} \bullet \frac{10}{7}$$

This can then be implemented as successive interpolations and decimations by factors of 2, 3, 5, and 7. The intermediate frequencies are as shown:

1. Interpolate by 4 from 44,100Hz to 176,000Hz.
2. Decimate by 3 from 176,400Hz to 58,800Hz.
3. Interpolate by 4 from 58,800Hz to 235,200Hz.
4. Decimate by 7 from 235,200Hz to 33,600Hz.
5. Interpolate by 10 from 33,600Hz to 336,000Hz.
6. Decimate by 7 from 336,000Hz to 48,000Hz.

Various other orderings are possible between these two sampling rates, some with lower intermediate rates but more dynamic range, others with higher intermediate rates but lower dynamic range. The particular choice of intermediate sampling rates depends on the application and the digital signal processor being used.

Dynamic range/ signal-to-noise ratio

Dynamic range in the digital domain (not including the effects of A/D conver-

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Derivation of Decibel-to-Bit Ratio

If we consider a number of binary digits, say 24, then the number of different numbers that can be represented by these 24 bits is 2^{24}, which is equal to 16,777,216. A number of bits (h) are used to represent all the numbers from -2^h to $+2^h-1$. To find the equivalent range in decibels of 24 bits, we use the formula:

$$\left(20\log_{10} \frac{\text{biggest number}}{\text{smallest number}} \right)$$

which is the standard equation for finding the dynamic range of a voltage or current. Substituting into this equation for the biggest and smallest number leaves us with the equation,

$$\left(R_{24\text{bits}} = 20\log_{10} \frac{16{,}777{,}216}{1} \right)$$

which anyone with a scientific calculator will tell you is 144.4943979dB (approximately). Because of a fortunate accident of mathematics, another formula allows us to simplify this to $R_{number\ of\ bits} = 20\bullet(\text{number of bits})\log_{10}2$, working out $20\log_{10}2$, gives us 6.020599914, or about 6, if we aren't too fussy. So we now have a relationship between the number of bits and the dynamic range—multiply by 6.

sion) is a function of the digital word size. As a rule of thumb, you get 6dB for every bit—so 8-bit gives you 48dB and 16-bit gives you 96dB. (Putting this in perspective, a vinyl album pressing is capable of about 70dB dynamic range). Even though it might be desirable to incorporate a higher number of bits, the final product will ultimately be quantized to 16-bit resolution if transferred to CD.

In the analog domain, dynamic range is a function of just about everything from the type of integrated circuits used to the wiring layout of the power supply. It is possible, though, to design analog mixing systems with significantly greater dynamic range than 16-bit digital systems can produce, and that is why the critical parts of a digital mixer's internal architecture need more than 16 bits.

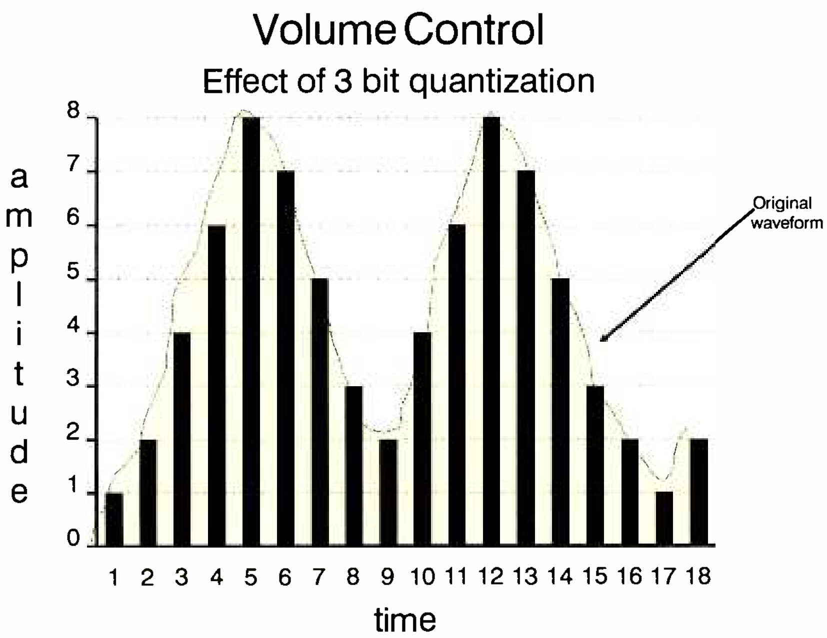
Volume control

Volume control is worth a separate mention because even though it is one of the simplest digital signal processing operations (just multiply the current sample by the current volume coefficient, a number that represents the current position of the fader), it is also one of the most contentious.

The first and easiest problem to dispose of is the resolution of the faders. A convenient number of steps, or possible volume levels, is 256. This is the largest number that can be represented by eight bits. But a fade with only 256 levels, or steps, wouldn't sound very smooth. One solution to this problem is to insert or interpolate a further 256 steps between each of the fader levels, resulting in up to 65,000 steps for a full fade. Some people have suggested a higher fader resolution of 400 levels, but given accurate interpolation, this might not be necessary. And, the next logical level would really be 512 steps, which would equal 9-bit resolution, but 9-bit processors are uncommon and, therefore, impractical.

The second problem is sometimes thought to be of serious significance in digital audio. What happens when the levels are low and quantization is down to only a few bits? Quantization noise is the deviation from the wanted signal that results from having to represent a continuous signal using a finite number of levels or steps. As you can see from Figure 2, it looks severe.

In any digital system, you can hear quantization noise if you have a very low-level signal and you monitor it at very high levels. But to do this is unrealistic. At −48dB (with a 16-bit system), which is actually very quiet, you would still have the benefit of 8-bit resolution. To put it another way, you would not want to watch a television at a distance of 3 inches—if you did, you would complain about seeing all the dots!



Signal processing

The algorithms used in digital audio signal processing are usually quite simple, measured by conventional computer standards, but demand a great deal of processor power because they must be executed in real time. A simple reverberation algorithm could require the execution of 300 lines of program code in one sample period—as little as 1/44,100 second. And digital equalization can be as demanding.

All mixers are complicated devices. They have to route signals to external devices over many channels and bring them back to the right place. These days, the external devices are often digital effects and digital multitrack tape machines that probably use the same sort of processor you would find in a digital mixer. Therefore, it follows that much of this complexity can be avoided if the signal processing is done on board the mixer.

It is now quite common for mixing consoles to offer some signal processing in addition to equalization—dynamics control for example. With digital mixers, this approach can be extended to include all effects, but this is not to say that if a board is digital then it *can* do all effects processing, far from it.

The task of mixing is extremely processor-intensive. (Think of adding 56 channels of digital audio at 44,100 samples per second, then multiplying each sample by a number that represents the position of the fader.) Most digital mixers use emitter-coupled logic (ECL) for their processing, which is fast and powerful, but is essentially a central processing resource. Also, ECL processing consumes a lot of power—kilowatts for a large digital console.

Mixers with this architecture still pass the signal to external effects, which adds a large overhead to the complexity of the machine. As we mentioned earlier, every time the signal path leaves the mixer and returns, it has to undergo several stages of D/A/D conversion and the inevitable loss of quality that this entails.

Matters will improve with the increased availability of digital-domain, fixed-sample-rate effects, but external effect processing is always going to be more complex, less versatile and more expensive than internal processing.

Another approach is to use distributed processing. This is not the same as parallel processing. Perhaps the best-known example of parallel processing is Inmos's Transputer. Digital audio signal processing is essentially a serial phenomenon, unlike video, which occupies a planar domain as well as the time domain.

Distributed processing has the power of ECL technology, but doesn't necessarily use it. The processing power is distributed through the use of many individual DSPs. For example, a 56-input system would probably have at least 56 DSPs. The advantage of distributed processing is that it is more manageable, and the technology is actually simpler because it is modular.

Digital Automation uses a system with at least one and, depending on the task in hand, several DSPs on each channel. (See "The KSd System as an Example of a True Digital System.")

Equalization

This is possibly the most critical and most-criticized aspect of digital mixing. It is normally carried out by digital filters. These operate by adding together scaled copies of the signal. The copies of the signal are kept in a delay line, whose delay is referred to as the filter length. (In very simple terms, *filter length* means the amount of time that the signal copies are held in the delay line. As the duration changes, so does the frequency.) The scale factors applied to each of the copies are referred to as filter coefficients.

Here is an outline of some of the techniques for designing digital filters.

FIR filters: Finite impulse response (FIR) filters implement inherently stable filters. A FIR filter will implement a particular frequency response curve and achieve it by attenuation only. It's analogous in linear circuit design to a passive equalizer. It has one added benefit, though: No matter what frequency is put into it, the phase response of the filter is constant and always equal to exactly half the duration of the filter length. FIR filters are relatively easy to design, and a large number of coefficient design algorithms exist.

IIR filters: Infinite impulse response (IIR) filters are not limited to implementing "passive" filter curves and can therefore achieve a given filter response with a smaller number of coefficients than are required with a FIR filter. IIR filters have a potential disadvantage though, and that is that finite word length (for example, 32 bits) can lead to instability. If a finite word length is used when there is no signal going into the filter, the filter might still produce an output. Another form of instability can occur when a signal is going into the filter and the output oscillates between extreme values, independent of the signal.

Design techniques for IIR filters normally use a common analog filter as a model and, by one or more of a series of mathematical transformations, implement the same or similar filter digitally.

Optimal filters: When you specify a filter (either on a piece of paper, or by setting controls on a console), you are saying to the filter, "I want you to do this to my signal." And, you want the filter to do an accurate job. The problem with most filtering operations comes when trying to define exactly what an "accurate job" is. The accepted way of measuring how accurate a job has been done is with the mean squared error (MSE).

For clarity, we will use frequency response as an example, but what we are about to describe also applies to impulse response, phase response and group delay as well. If you imagine drawing a graph of gain against the frequency of your desired response, and on the same graph drawing a curve that represents the actual response as measured at the output of the mixer, they will not be the same. The difference between the two curves is measurable, and if we draw a third graph that is the difference between them, we should get a rather confused-looking line sitting on the X-axis. This is not really of any use as a measure of the error in the filter because it has an average value of zero (for all filters). But if we take the new curve and square it at every point, this fourth "squared" curve will always be positive, and its average value will be indicative of the error in the filter.

Now, an optimal filter has filter coefficients that make the average value of this error the absolute minimum possible error. Unfortunately, designing these optimal filters isn't easy, but a Russian named Remez invented a method for doing it for FIR filters in the 1940s, which people still use today. The so-called "Remez Multiple Exchange Algorithm" operates by finding the frequencies at which the error curve is a maximum, moving all the frequencies that are less than a precalculated value, and recalculating the error curve. The process of search and recalculation is continued until all the errors are equal to the precalculated value. The resulting frequency response is guaranteed to be the best possible, and is *inverse Fourier transformed* to give the filter coefficients.

Easy? Not really, as the Remez method is concerned with minimizing a large number of small magnitude errors, and problems can occur with convergence. That is, the program can sit there calculating the minimal frequencies, which, because of numerical limitations of the computer, will end up "oscillating" between two close but non-optimal solutions. Other types of optimal filters exist, particularly for applications in which the user doesn't need to change the way the filter works, such as fixed telecommunications systems.

Automation

Digital mixing lends itself to automation. A digital mixer is essentially a computer, and computers don't mind if they are mixing Mahler's Fifth Symphony or working on a model of the economy. We're not suggesting that you should try running Lotus Symphony on a Synclavier, but the point is that the information controlling the signal that is being processed is in a similar form to the signal itself—binary numbers. Every aspect of a digital mix can be stored, given a time code, and recalled in real time. This is dynamic automation. Support functions such as editing (copying, deleting, edit decision lists and mix merging) have to be provided and displayed as well.

The major design decisions are the form in which the mix information is stored and the resolution at which it is stored. If mix data storage and processing power are sufficient, then the optimum method of storage, in terms of accuracy, is to note every point through which a changing parameter (say, volume) passes and the time at which it passes. This is, perhaps, analogous to bit-mapped graphics on a PC or Macintosh.

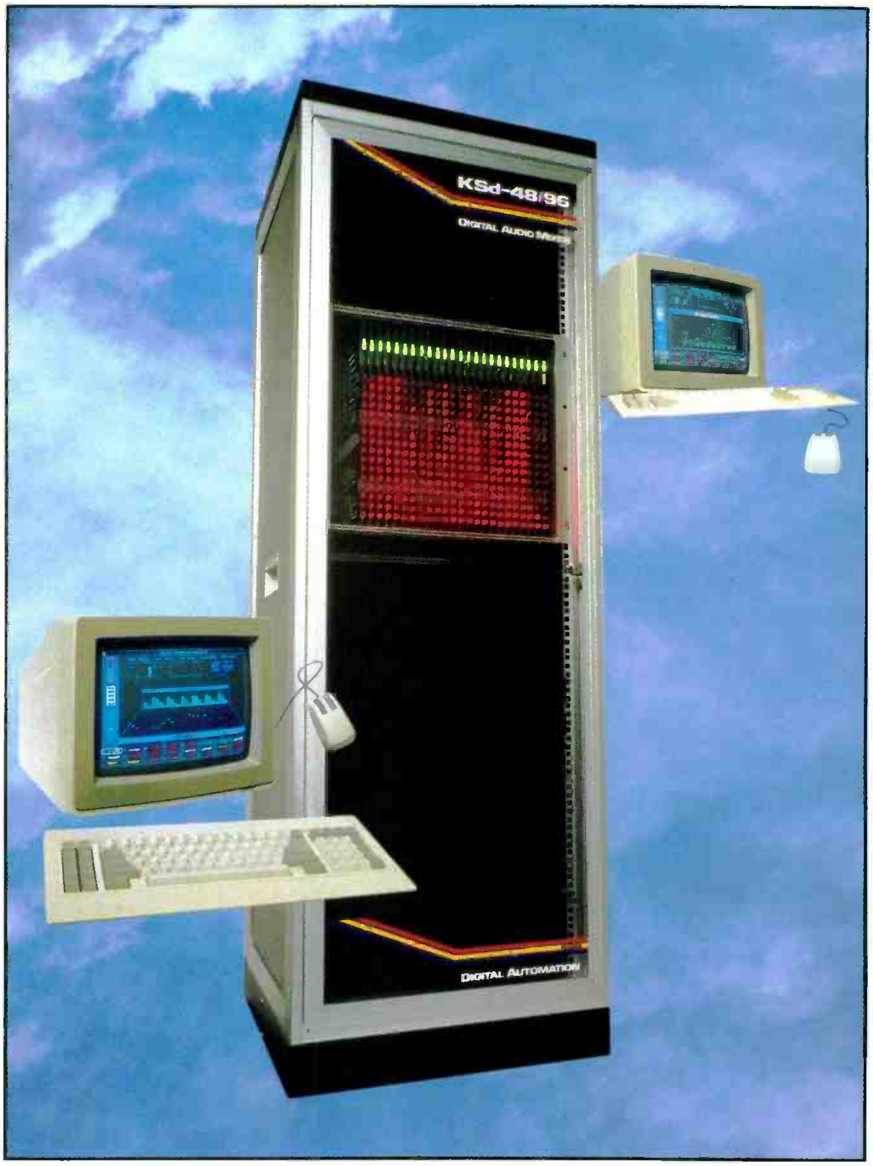
A more economical (and, arguably, equally accurate) method, involves storing vector descriptions of parameter changes (comparable to object-oriented graphics). At its simplest, this means noting the start and end points of a change, and the start and stop times. More-complicated movements can be

The KSd System as an Example of a True Digital Mixer

The KSd system is a digital mixing device based on multiple channel modules, each comprised of a DSP, program and data (sound) memory, and communication ports, for control. Each channel is effectively a digital sound processing computer capable of running its own software and is controlled either directly from a computer, which in addition, provides display, parameter and automation storage facilities.

The system currently under development can be controlled from an assignable physical console. It will feature two sets of channel boards connected back-to-back via a routing system, and will essentially result in a "virtual" mixer. The user will be able to decide, for example, how many effect buses are needed and/or the number of processors allocated for a particular effect. In practice, the user will be "protected" from the architecture by the system software, which will automatically allocate resources after being told what it is expected to achieve.

Since all effects processing is carried out "on board" (although there is still the option of using external effects), true digital mixing is possible, with no unnecessary excursions from the digital domain.



described in terms of velocity of change, for example. The vector description can be as little as four bytes of information, compared with as much as 65,000 bytes for the "bit-mapped" method.

Ergonomics

What constitutes the optimum layout and functionality of a control surface? There is no simple answer. An arrangement that is ideal for one operator might be unusable for another. This isn't necessarily just a function of the console design, but also, perhaps, of what the operator is used to. You might even have to take into account the reaction of the studio client to the appearance of an all-digital console, which has nothing to do with functionality. Unfortunately, many clients are far more likely to be impressed with a board the size of an aircraft carrier than an anonymous 19-inch rack-mounted module—regardless of the fact that the 19-inch rack might have the power of a supercomputer.

With digital mixers, and this applies to digitally controlled analog desks as well, the temptation is to design the console surface around multifunction controls and controls that can be assigned to any channel. The theory is fine, but the practice is limited by the extent to which the multiple functions of the controls can be understood. Grouping of functions should be logical to the user as well as the designer, with the current status of a control clearly and obviously displayed. Most importantly, the path from one function to another and the options available should be clear and intuitive.

As we approach the 1990s, it's possible that some of you will be considering the purchase of a digital mixing console. By now, understanding the basics of the analog domain is well within the grasp of most audio engineers. This means you have a pretty good idea of what questions to ask and what answers you'll receive when shopping for an analog console. Soon, several systems in development will be vying for attention right along side traditional analog console hardware, and we hope this article has helped define some of the possibilities and limitations of an all-digital mixing system.

RE/P

Theater Sound in England:

A Conversation with Andrew Bruce of Autograph Sound Recording

By David Mellor

"Cats," "Les Miserables" and "Chess": Autograph Sound Recording was the rental company responsible for sound on each of these shows.

Just as The Great American Musical is under threat from the Great European Musical, a trend set by composer Andrew Lloyd-Webber, the British have taken a lesson from American theater sound design. They are not only developing the art on the London stage, they are exporting their ideas to Broadway. "Cats," "Les Miserables" and "Chess" are three recent shows that have crossed the Atlantic from East to West. In each case Autograph Sound Recording has been the rental company responsible for sound in their London productions.

But a rental company just supplies equipment, right? Maybe that was the case a few years ago with "Cats," but today they are often involved in sound design, too. Autograph co-founder Andrew Bruce was the designer for "Chess" in London. Now it is being produced, in a very different version, on Broadway with the same sound designer, and the New York arm of Autograph in charge of the equipment.

Says Bruce: "The sound designer was Abe Jacob, who is the grand old father of Broadway sound design. He taught us a tremendous amount about the way things were currently done on Broadway. He brought in a lot of innovations as far as we were concerned, practical things that they had been doing on Broadway for quite a while that were new to us."

"Chess"

The version of "Chess" seen by New Yorkers is completely different from the one at the Prince Edward Theater in London. Much of the music has been replaced by dialogue to help convey the complicated political theme of the show more clearly—and the ending is completely different.

An Autograph installation at the Shaftesbury Theater in London showing the Cadac console and the outboard signal processing rack.

David Mellor is a free-lance technical writer living in London.

" 'Chess' was written by Benny Andersson and Bjorn Ulvaeus (the writing half of Abba) with lyricist Tim Rice," Bruce says. "It moves between several styles. There are orchestral moments, very reminiscent of Tchaikovsky, and rock and roll that's very definitely Abba, and much in between. We have to have a system that is capable of handling all of these, from orchestral to rock and roll and back again. Some of the changes between one style and another are very swift, shockingly so. It's done for effect. We have a big vocal system that covers the theater pretty well, and a separate orchestral system that uses the same console. They are arranged in two concentric circles, like two systems in one. I like to keep things separate; it seems to work better, and it allows me to use different types of loudspeakers for each job.

"The orchestral system at the Prince Edward Theater consists of four Meyer Sound MSL-3s. They are placed in towers at the sides of the stage and are semiconcealed because that's what the set designer wanted. We started off with four USW subwoofers, but we didn't need all that bottom end so we ended up with two.

"Because of the nature of the MSL-3, it needs to be set back quite a long way from the listener. You need to be a certain distance from it to get the full effect. I originally had some positions blocked into the scenic design, which were perfect. They were sufficiently far away from the audience, so that it didn't hurt even in the front rows. However, part of the design involved three VidiWalls—64 TV monitors in a single eight-by-eight block on stage, and two side walls of eight by four. The walls can be configured to make either one gigantic picture or many individual pictures or anything in between. The side VidiWalls weigh around 3 tons each and need to be suspended from a single point so they can be moved forward for servicing.

"During the survey, the engineers found a place they thought would take the weight of the VidiWalls. When they actually came to do it, those places didn't stand up. They had to find an alternative place from which to suspend these 3-ton monsters. As luck would have it, the place they found meant that the VidiWall hung in exactly the place that I had blocked for my speakers, so I was forced to move them to a position that was much closer to the audience than I had originally wanted. At that stage, it was too late to go back and redesign the system. In the United States, I had to drop the MSL-3s in favor of the smaller UPA-1 clusters. They give me much more flexibility; I can string them out any way I want.

"The vocal system is purposely separate. We are trying to get very high levels out of the vocal system at some points in the show and I favor using the Meyer Ultramonitor (UM-1) for that because it has a very narrow-dispersion high-frequency horn. What I am trying to do with vocals is avoid bouncing mid and high frequencies off the side walls and balcony fronts. Feedback is a huge problem when you are using a head-mounted omni-directional microphone 8 inches from the mouth. I've found that these clusters give me a much higher level before the onset of feedback because there are far fewer reflections than with a wide-dispersion high-frequency horn design. If you use a traditional horn and try and keep it off the side walls then a huge amount leaks back on stage, and if you direct it away from the stage then you can't stop its bouncing off the side walls.

We have a big vocal system that covers the theater pretty well, and a separate orchestral system using the same console.

"I can point the UM-1s in exactly the direction I want so that they miss the balconies and get right underneath. I can throw to the back of the hall with an upper speaker and then, just underneath it, I can have another pair that cover the area directly in front. I can tailor all the levels and patterns so that the coverage is uniform. There is no doubt, however, that you do get a few anomalies on the boundaries where the high-frequency patterns meet. Sometimes I can trace a line straight across the auditorium where they meet, and if you sit in those seats and move your head an inch, you will hear some combing. I am prepared to sacrifice those few seats to cover the majority."

Console

For the New York version of "Chess," a Cadac E-Type console, which is designed specifically for big musicals, is used. It is a 12 sub, 12 group, VCA, computer-controlled console with eight auxiliaries and any number of inputs. Autograph specifies the console with bays of inputs that can be added on to the basic master frame. The master frame has 19 inputs and all the output sections. Bays of 15 or 30 extra inputs (figures constructed around the multicore system) are added as necessary.

"In London I had an earlier A-type version of the console modified so it could be run by a fairly rudimentary computer that just handled muting, and had some relay contact closures that we could program," Bruce says. "It can step sequence effect programs that we have stored in processors like the Klark Teknik DN-780 and Lexicon PCM70. There are 95 cues in the London computer, and it's very useful and works very well. It takes a lot of pressure off the operator. Because of the nature of the writing, you go into a rock and roll section of the piece and there are 20 musicians sitting there playing nothing for five or 10 minutes. You don't have time to kill every fader or mute every channel. The computer mutes them so we lose the muddiness that you associate with open microphones."

Getting clean, high-level vocals in a theater is difficult enough if you're doing rock and roll with close hand-held mics. It's immeasurably more difficult if you have to conceal the mics on the body and attain the same sort of level. Bruce uses the Sennheiser MKE2 on the actors' heads, concealed in the hairline.

"The chest position has been traditional for a long time, but you get the resonance from the chest, the rustle of clothing, people can't embrace and when they move their heads they go off mic," Bruce says. "It's a losing battle; you take up all your console EQ sections just to get rid of the chest resonances and trying to squeeze some kind of presence back in. You immediately have that presence when the mic is placed on the head.

"Almost every show we do has a backstage radio mic person. They are there to take a transmitter off one actor and move it to another because there's a finite limit to the amount of transmitters you can use on a show.

"Because of the finite number of frequencies, we have to make them go 'round, but we may have 40 in the cast. We try to keep principals' radio mics on them at all times—the last thing you want to do is upset a principal when they're trying to concentrate on the part. The remaining transmitters form a block of transmitters that we move around the chorus.

"I've had a fair amount to do with the recent campaign for official recognition and getting a frequency allocation for use by theaters in the UK. No other country has ever officially recognized the need for radio microphones for the use of the theater alone. We now have 18 frequencies, plus four general-user frequencies, making a total of 22 that we can employ. I don't think we could ever really make use of more than that number because the equipment is so expensive that no producer could afford to rent any more.

"We locate the radio receivers on stage so that the backstage person has access

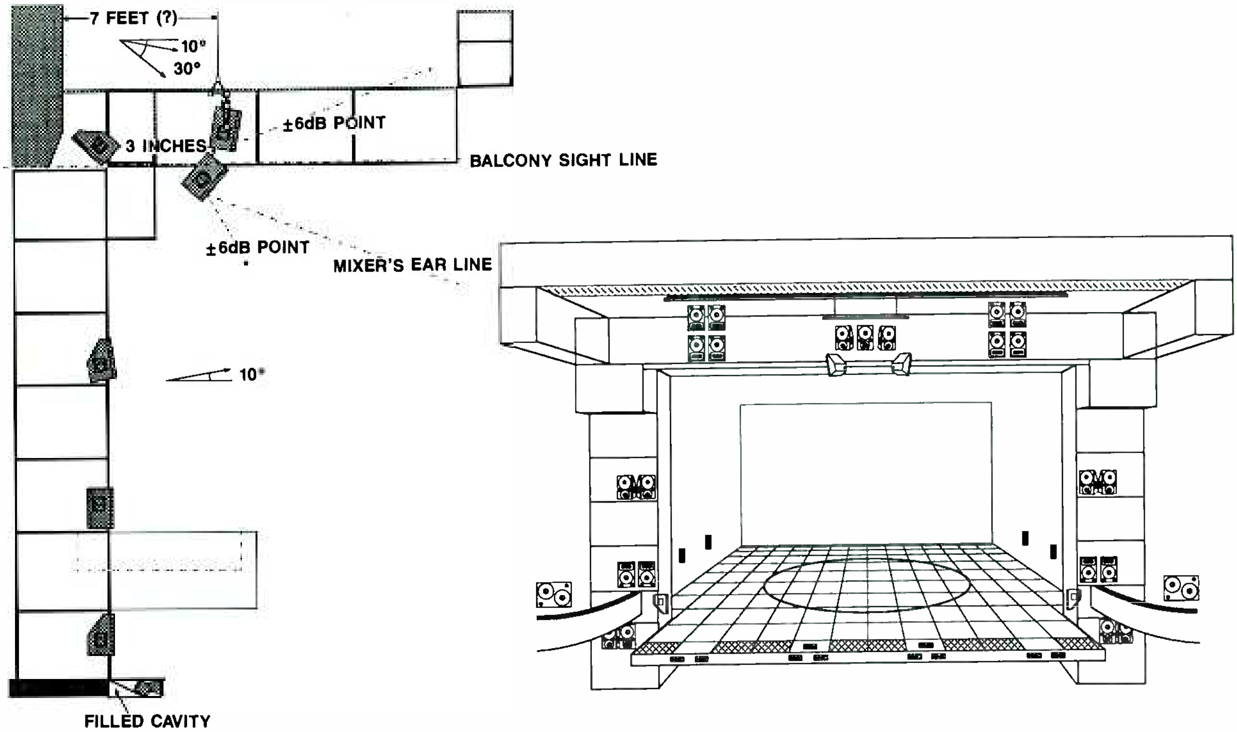


Figure 1. *"Chess" proscenium section and overview.*

to audio and RF monitoring. With transmitters being moved like that, the possibility of damage is quite high. So once our backstage person puts a transmitter on somebody, they go straight to the rack of receivers and check that it's functioning OK."

If the show is to be reinforced, rather than amplified, then a representative number of chorus members will each have a radio mic. If the sound has to be much fuller, then they will all need a mic of their own. The one advantage of using an omnidirectional microphone is that an actor who is not wearing a transmitter can be picked up well on another actor's microphone—as long as he or she is within 4 to 5 feet.

Analysis

An increasingly important part of Bruce's technique is sound systems performance analysis in the auditorium, using the Source Independent Measurement or SIM analysis technique, which uses periodic noise or a music signal as the signal source for measurements. In this way, the system response can be checked and modified during a performance. [For more information, see "Acoustical Measurement Techniques for Sound System Equalization" in June, page 30.—Ed.]

The measurement system consists of a Hewlett-Packard FFT analyzer, a delay unit, Bruel & Kjaer measurement microphones, software-controlled signal switchers, and Meyer CP10 parametrics EQs. Until recently, Autograph used the SIM technique to set up the system, then took away the analyzer, leaving only the equalizers to do their work. Cost plays an important part in this, as it would be impractical to have a SIM operator for every performance during what might be a five- or six-year run.

"We have recently been exploring how to use this system in the most advantageous way for theater," Bruce says. "What we had in mind for the future was to keep the analyzer a bit longer and see how things measure under performance conditions. It's a well-known fact that when you get your first audience, many things change. On your first rainy day, everyone comes into the auditorium wearing damp coats, the humidity in the auditorium rises dramatically and suddenly the sound is completely different—lots of highs and very present. [See "Effects on the Speed of Sound" by Dennis Bohn, April, page 30.—Ed.]

"When I did 'Les Miserables' in Boston, we observed what happened during technicals and moved the analyzer to a room in the basement so that we could sit and observe the system response during a couple of previews. Sure enough, during the first public performance there was an immense change. We had taken a decision to start a very gentle roll off in the vocal system at around 8kHz, and suddenly the response seemed to have become ruler-flat to 12.5kHz. Everything was terribly crisp and bright—distressingly so, and we could confirm this instantly on the analyzer.

"We told the operator that we planned to do something about this, and we brought in the high-cut filters during the intermission. During the second act, it was back to the way we had set it during technicals and the way we wanted it to sound."

Future development

"We are now refining the technique, making the distinction between reinforcement and amplification, but always using amplification with caution," Bruce says. "Reinforcement should draw as little attention to itself as possible, with the focus of attention remaining very firmly on the actor."

Andrew Bruce and Autograph are playing the role that Abe Jacob used to take when American shows came to the West End of London and there was little home-grown product. A British or European show now travels to Broadway with a sound designer in tow. RE/P

Michael Jackson on Tour

By David Scheirman

Sound system design by Clair Bros. Audio features a new subwoofer system from Intersonics.

Any discussion of major concert events in today's entertainment world will bring up the name of Clair Bros. Audio, one of the largest and most active touring sound rental companies in the world. Recent clients have included Madonna, Lionel Richie, Robert Plant, Sting, Bruce Springsteen, and the globe-spanning Amnesty International tour for Human Rights. [For a detailed look at the company's history and hardware, see "The 1983 US Festival", October 1983, page 94—*Ed.*]

In 1988, one of the largest-grossing U.S. tour acts was Michael Jackson, which is also a Clair Bros. account. This artist's show is synonymous with "spectacle" to many concert-goers. Audiences of all ages have come to expect Michael Jackson concerts to treat them to the latest, most outrageous production tricks. From lasers to choreography, staging hardware to live sound, Jackson's shows are some of the most lavishly produced in today's concert market.

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

Sound system design

All Clair Bros. systems rely on the time-tested S-4 modular loudspeaker enclosures. Several design generations have been brought forth since the S-4 was first introduced in the early 1970s. While different cabinet construction styles and internal component layouts have been developed, the box can typically be described as a direct-radiating, 3-way, active, integrated loudspeaker system with passive super high-frequency units.

A pair of 18-inch loudspeakers and an array of 10-inch loudspeakers, along with various combinations of horns and compression drivers, are housed in each enclosure. The box has an approximate weight of 400 pounds (193kg) and measures 43″x45″x22″. (See Figure 1.)

Multiple S-4 enclosures are assembled into large concert arrays. A typical indoor arena system may use from 48 to 64 boxes; larger outdoor systems may require up to 200. For Michael Jackson's indoor arena dates in the United States, 64 boxes were employed. Venues included the 20,000-seat Richfield, OH, Coliseum, the 16,000-seat Los Angeles Sports Arena and the 17,700-seat Bloomington, MN, Met Center. (See Figure 2.)

Half of the available S-4 boxes were arrayed to the audience areas directly in front of the stage. These included some newer enclosures with long-throw characteristics. The rest were distributed to the sides and rear, because all available seats in the arenas were typically sold out.

The 16 enclosures in each of the two main left and right front arrays were suspended in two horizontal rows of six, plus an additional third row of four boxes below. A gentle curvature in the horizontal plane for wider dispersion was accomplished with Clair's unique "bumper bars": custom-built hanging

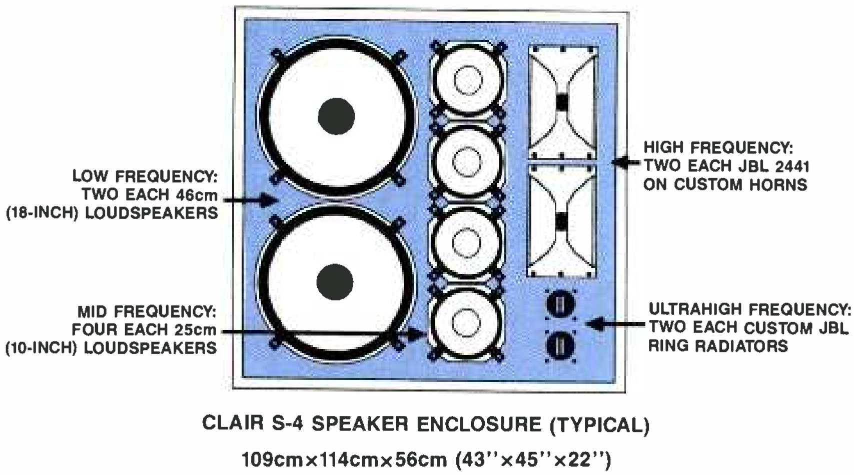


Figure 1. *The Clair S-4 loudspeaker system enclosure.*

beams with curved end-pieces and multiple attachment points.

Each group of 16 S-4s was hoisted into the air with a total of four chain motors. Two 2-ton C/M Lodestar hoists carried the primary weight of the inside bumper bar on each side, and an additional pair of 1-ton motors was used to fly the end of the outer bar. The pair of 1-ton motors was secured to steel bridles set in angled opposition to each other. (See Figure 3.)

"We call this the 'electric bridle' method," says system setup technician Nile Wood. "There are two individual motor controls for the outer bar. That way, we can get the angle we want quickly, without too much fussing around with the hanging points."

Clair's compact, densely packed power amp racks, loaded with Carver amplifiers, were used to drive all S-4s in the system. Each of the racks features a center-hinge point. Each road case opens to reveal a dozen amplifiers with front-panel access for signal cabling and ac power distribution cords.

New subwoofer system

Clair S-4 systems are generally considered to offer a nominal full-range response. The larger number of 18-inch speakers contained in a large system provides a significant quantity of low-frequency energy. For certain shows that require greater bass response, Clair Bros. has typically provided vented subwoofer enclosures, fitted with traditional 18-inch loudspeakers.

A typical indoor arena system may use from 48 to 64 boxes; larger outdoor systems may require up to 200.

"We first started using sub-lows early in the tour when Michael [Jackson] wanted something that would really shake the stage," says Clair systems chief Jimmy James, the person primarily responsible for the sound system while on the road. "We set up a system with the Crown Macro-Tech 10,000 amplifiers—two of those units were used to run all the subs. Originally, when we were running traditional subs, we were losing cones because of overexcursion. When this system came back from Europe, Gene Clair and our engineering staff decided to investigate other options."

The option that solved the problem turned out to be servodrive technology. Pioneered by Intersonics Inc., of Northbrook, IL, servodriven subwoofers employ a high-speed servomotor to convert the amplified audio signal into sound. (For a detailed look at this concept, see "Servodriven Loudspeakers," October 1985, page 85—*Ed.*)

Intersonics' servodrive loudspeakers use a rotary-to-linear converter to push loudspeaker cones back and forth. The piston assembly is capable of large excursions that can exceed 1.25 inches in each cabinet. The pair of loudspeakers are mounted facing each other and are compression-loaded. (See Figure 4.)

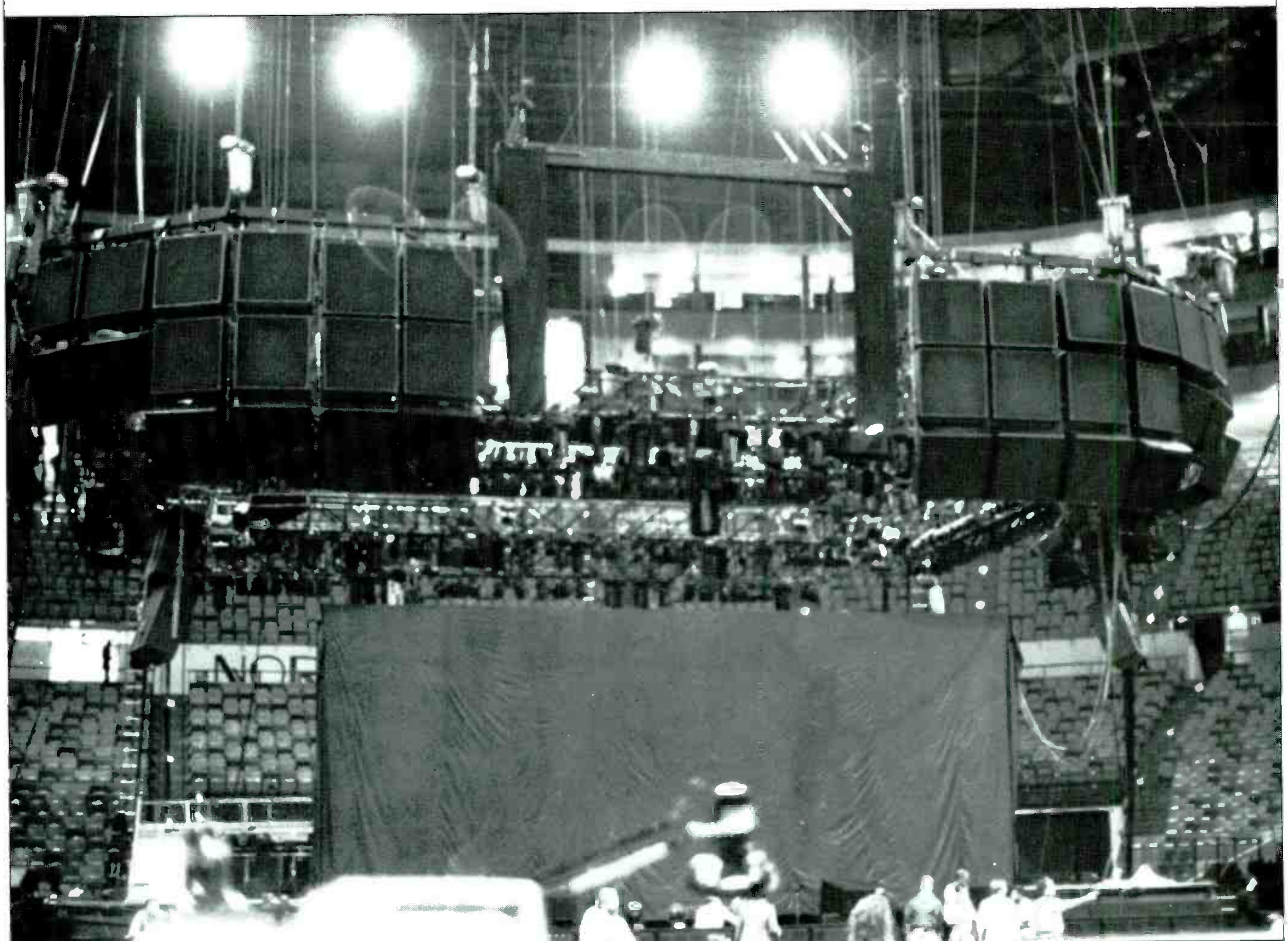


Figure 2. *S-4 Hanging arrays for Michael Jackson's tour, shown here at the Richfield, OH. Coliseum.*

The speed of the servomotor is proportional to input voltage, and the force of the cone movement is proportional to the current applied. The polarity controls the direction of servomotor rotation. The shaft rotations produced by the amplifier signal are converted to linear motion with a push-pull-type belt drive system. (See Figure 5.)

The true limiting factor in any subwoofer system is power compression—that reduction of efficiency that occurs when a speaker's voice coil is heated during high power operation. Subwoofer systems used for arena rock shows are typically run right up to their safe operating limits, while conventional drivers can lose from 2dB to 6dB of sensitivity when they are operated under these conditions. Thus, a point is reached where more input power will not provide higher output; instead, the acoustical output actually begins to drop prior to eventual voice coil failure from thermal overload or overexcursion.

Intersonics' servodriven loudspeakers used a rotary-to-linear converter to push loudspeaker cones back and forth.

The servodriven loudspeaker systems use a patented, integral "power cooling" technique, wherein a blower (which uses only about 1dB of input power) draws cooler air over the motor's coils. This reduces power compression to less than 1dB at rated power and enables the bass system to offer increased acoustical output when operating at extremely high levels. (See Figure 6.)

Clair Bros. chose Intersonics' new high-powered SDL-5 units for the Michael Jackson system. Each unit weighs approximately 270 pounds and measures 45"x 22.5"x 45". The recommended crossover point is 80Hz to 125Hz, and the low-frequency cutoff point for a single unit is 34Hz. With four units coupled together, this moves down to 28Hz. The Michael Jackson system currently on the road employs 16 such devices. (See Figure 7.)

"It's a matter of choosing the right tool for

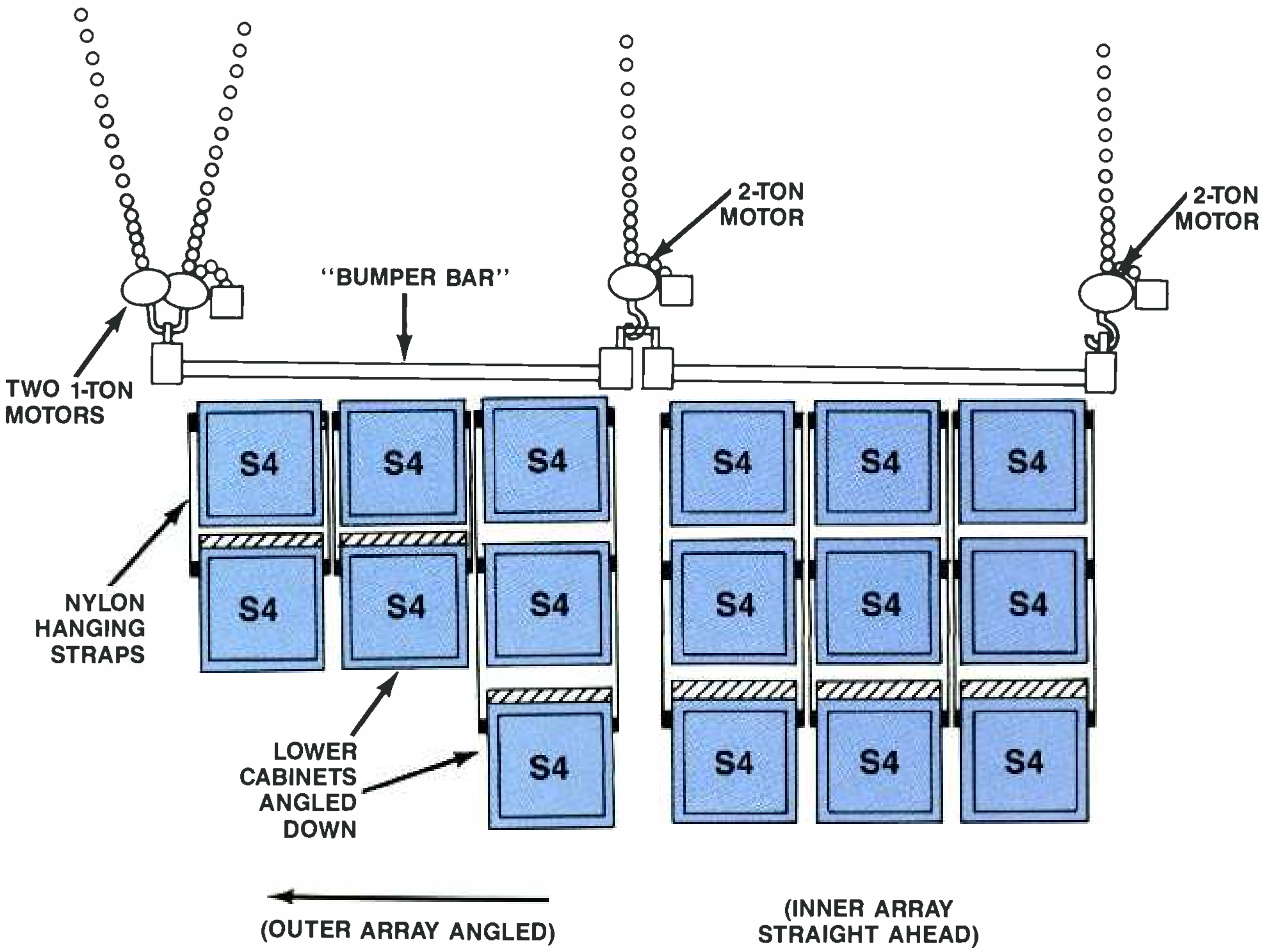


Figure 3. *One of two main front arrays, comprising 16 S-4 enclosures. Four chain motors are used—two 2-ton and two 1-ton.*

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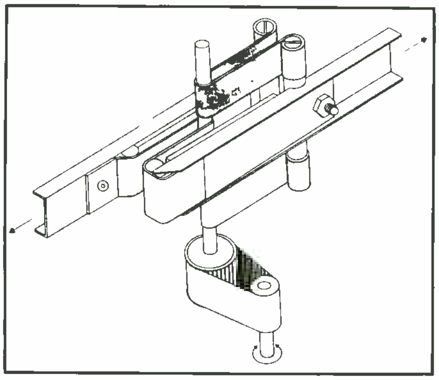

Figure 4. A rotary-to-linear converter inside each Intersonics servodrive enclosure transfers mechanical energy to the loudspeaker cones.

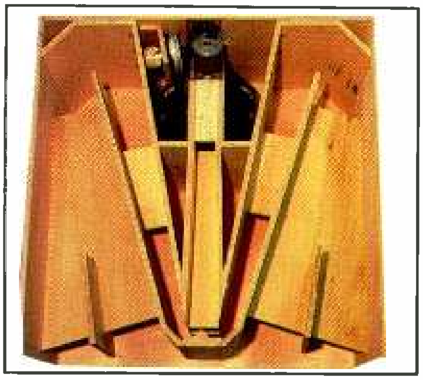

Figure 5. In each SDL-5, a pair of heavy-duty speaker cones are installed facing each other in a compression-loaded chamber.

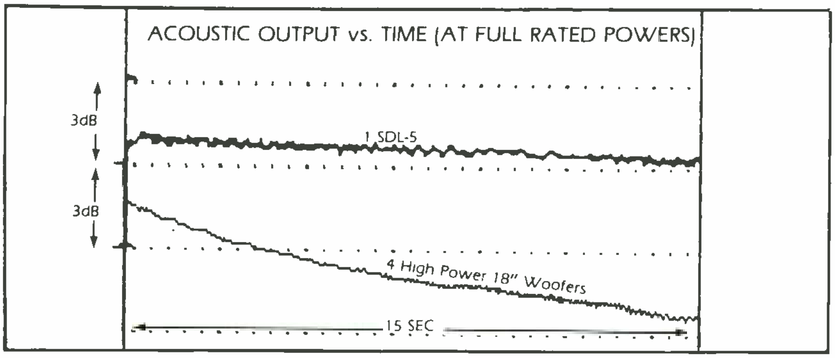

Figure 6. This graph compares the acoustic output of one SDL-5 unit with four 18-inch loudspeakers in vented enclosures. Intersonics maintains that the concept of "power cooling" allows the SDL units not to suffer so much from the effects of power compression as do traditional loudspeaker systems.

Figure 7. Two of the 16 SDL-5 units in use on tour with Michael Jackson.

the job," says Ron Borthwick, senior design engineer for Clair. "We try to interpret what the act wants. For this tour, we were primarily concerned about the lowest audible frequencies, which is where the servodriven loudspeakers work best."

"We line up 14 of the SDLs right across the front of the stage, between the front edge and the crowd security barrier," says Jimmy James. "The other two get pointed toward the stage, just to jazz up the musicians a bit. These things do a good job of keeping the low-frequency energy going where you point it. Low frequencies are supposedly omnidirectional, but these things are horn-loaded and it does make somewhat of a difference."

James is enthusiastic about the performance of the SDL-5s in terms of overall output. "I like this concept for low bass . . . two diaphragms squeezing a column of air between them out into the horn chamber," he says. "They are very efficient, so much so that we run them as an added effect, so they are not on all the time. They appear to be most efficient in the 20Hz to 40Hz region. We like the contrast when they are brought into the sound of the show."

All 16 SDL-5s are powered with a pair of Crown Macro-Tech 10,000s. The manufacturer claims that each amplifier can develop 16,200W of burst power, or the equivalent of 21.7 horsepower (with test measurement parameters of 40ms of 1kHz sine wave at 0.05% THD, bridge-mono). Under actual use conditions with the SDL-5 subwoofers, the amplifiers supplied a minimum of 600W to each of the 16 speaker enclosures, which have a nominal impedance rating of 4.25Ω each. These massive power amplifiers were located beneath the stage in sturdy dollies.

Using the system

Kevin Elson is responsible for Michael Jackson's sound system operation. The veteran live soundmixer chose to feed the Intersonics subwoofers with a signal that could be controlled by a footpedal located on the floor at the house mix position. This method allowed him to bring the subbass effect into the sound of the show as needed, according to existing conditions when the audience was in place and responding to the music.

"It is pretty tough to get an accurate picture of what the low bass frequency activity will be in a particular arena until the crowd is in and the show is on," says Elson. "We make sure that the subsystem is working during sound check, but don't worry about the level settings until we are operating it under real conditions."

To drive the SDL-5 low-bass system, a full-range output from the Clair Bros. custom main house console was fed through the footpedal and into a dbx model 165 compressor-limiter. Up to 10dB of output gain was often provided at this stage. "This unit is

specially calibrated by the Clair shop so that 0dB on the meter is actually −6dBV," says Elson. "That way, we can observe meter deflection in a range that is in alignment with the actual acoustical response of the subbass system during show conditions."

The output of the dbx 165 was connected to an E/V model XEQ-3 electronic crossover, using 12dB per octave filters, including a 16Hz high-pass and 80Hz cutoff. The XEQ-3 is equipped with a phase-reverse switch for each bandpass, and adjustable signal delay. (See Figure 8.)

"We like this crossover for this application," says James. "It has good signal integrity, and we can dial in the optimum sound for the Intersonics sub with the adjustable delay circuit."

When the system was originally set up, a 1/3-octave graphic equalizer was also in-line with the subwoofer feed. "We found that we really didn't need that," Elson recalls. "The subwoofer system has a pretty smooth overall frequency response. We do have graphic equalizers channel-inserted [Audio-Logic SC-31] on both the electric bass direct line input and on the synthesizer bass line-in, so I can fine-tune those inputs as needed."

A Klark-Teknik DN-27A graphic equalizer and dbx 904 noise gate module were channel-inserted on the kick drum input.

Soundmixing facilities

Kevin Elson has a pair of Clair Bros. custom 32-input mixing consoles at his disposal. Although these boards have been in the firm's inventory since the middle of the last decade, they have proved themselves in every conceivable live sound situation. While they only offer four auxiliary output buses, other features were definitely ahead of what was being offered by most commercial console manufacturers at the time.

"We make sure that the subsystem is working during sound check, but don't worry about the level settings until we are operating it under real conditions."

For example, the Clair custom console features a plasma-display metering system, offering both a peak and average reading simultaneously. (For a detailed look at the Clair custom console, see "Clair Bros. on the Road with Bruce Springsteen," February 1981, page 50—*Ed.*)

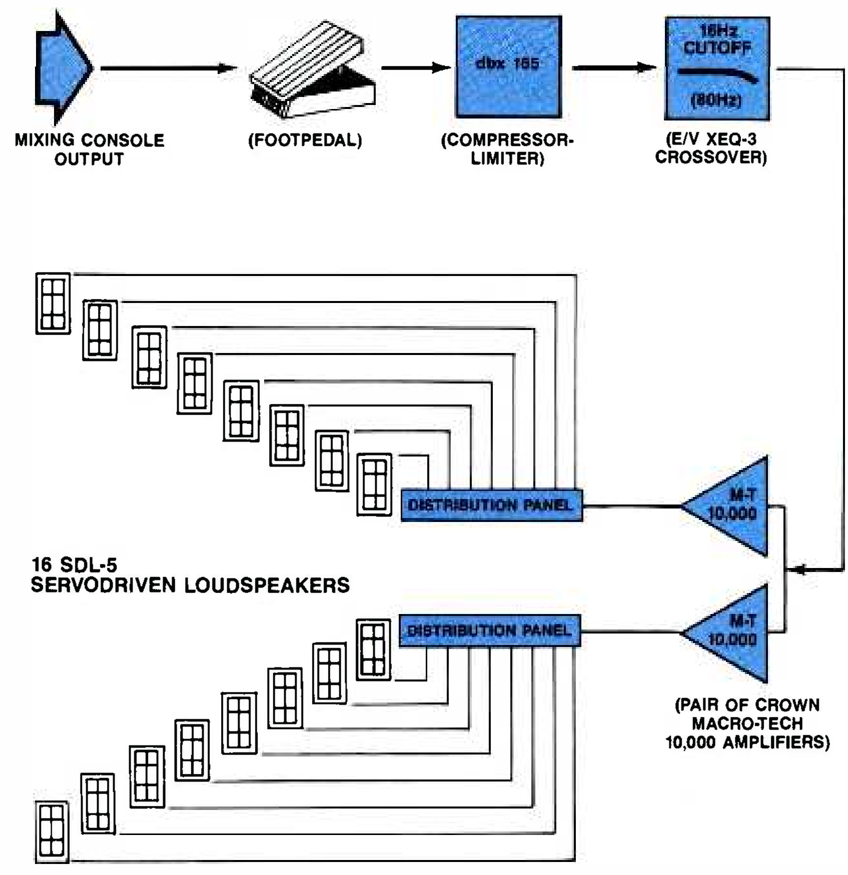

Figure 8. *Signal path of the subwoofer feed.*

Positioned between the pair of mix consoles was a large CRT color video monitor to display the output of a dbx RTA-1 real-time analyzer. (See Figure 9.) The external monitor allowed quick visual access to information regarding the system's frequency response in the room.

Signal processing for the house mix position was housed in Clair's custom road electronics racks, including a new prototype compact double-wide rack with a center hinge. Somewhat smaller and more portable than the company's traditional oak racks with external road cases, the new rack uses the protective case as the frame to hold the electronics, so no case lid needs to be removed for operation. Front latches are undone, and the hinged case opens to reveal the electronics inside. (See Figure 10.)

Special effects devices available included a pair of Yamaha SPX-90s, a Roland SDE-3000 and SDE-2000, two Lexicon PCM-70s, a Lexicon 224X and a Yamaha REV-5.

"Since there are four auxiliary outputs on each console, I keep all of the inputs of a certain type together and then assign them to a particular effects unit," Elson says. "For example, I am using different reverb units for the drum sounds and vocals. I'm able to get around the minimal number of auxiliary buses in that way."

For channel-insertion on various inputs, Drawmer DS-201 noise gates were supplied, along with a dbx 900 rack loaded with both compressor/limiter and noise gate modules. Additional dbx 160 units and graphic equalizers were used for certain inputs.

The stage monitor mixing position was equipped with a Harrison HM-5 and a Yamaha PM-3000k-40. The consoles were set up on stage-right, below the level of the custom-built stage, so that Clair monitor mixer Rick Coberly has a wide-angle view of the entire performance area—from the musicians' ankle height. (See Figure 11.)

More bass with less bulk

In today's touring market, sound systems must be as compact as possible. Yet, the audience's expectations regarding the sound and appearance of live concerts are higher than ever before. Because truck space costs more than ever before, this leads to a serious decision-making process in terms of the overall design of concert sound systems.

Major shows use a fleet of oversized trailers for equipment transportation, yet the list of hardware that needs to be fit into these trucks seems to be constantly growing; custom stages, scenic and drapery packages, exten-

Figure 9. The house mix position features a pair of Clair's custom 32-input mixing consoles. The video monitor is displaying the output of dbx's RTA-1 real-time analyzer.

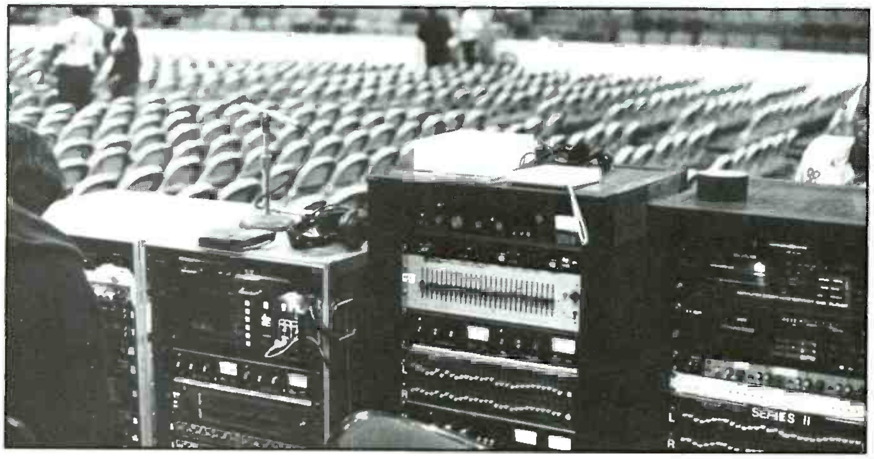

Figure 10. Signal processing in the house electronics racks. Note new, shorter double-wide rack package.

Figure 11. Stage monitor mix position, featuring a Harrison HM-5 and a Yamaha PM-3000.

sive wardrobe cases, neon logo displays, laser systems and video equipment are all part of a major tour today.

The Clair system touring with Michael Jackson represents a package that is more compact than what was available only a few years ago. Downsized amplifier racks, shrinking electronics racks and speaker systems that have increased output for the same amount of mass are all part of this picture.

The move to incorporate the SDL-5 servodriven loudspeakers into the Michael Jackson system has provided the show's soundmixer with greater subbass output in the 20Hz to 30Hz region, without increasing the bulk of the subwoofer system. The challenge of overcoming loudspeaker failure, formerly only solved by either lowering the input level or adding more amps and speakers, appears to have been successful.

The Clair system is designed to be an integrated package that can undergo intercontinental transportation without concern. Following the U.S. leg of the tour that ended in November, Clair shipped the entire sound system to Japan.

For the actual-use situation that I observed at the Richfield, OH, Coliseum, the operation of the new subwoofer system was impressive. Called up by Kevin Elson with the footpedal, the low bass swells demanded the audience's attention as the show started out with an extended-length, sequenced, drum/bass intro.

During the course of the event, I checked the RTA-1's monitor screen from time to time. Bar-graph displays bounced up and down as the average sound pressure level present in each of the 1/3-octave (standard ISO centers) bands was displayed. It was interesting to note that during most of the program material, the 20Hz band floated about 12dB to 16dB below the mid-range bands. Energy in the very lowest octave was usually equivalent to that in the highest (20kHz) band, and the strong audible bass energy was displayed primarily in the 40Hz band. When the SDL subwoofers were brought into action, the 20Hz band display immediately jumped up to nearly equal the 2kHz band.

"It's pretty unusual for us [Clair Bros.] to incorporate speaker systems into our setup that we don't design and build ourselves," system chief James says. "But Gene Clair felt these subs would solve a particular problem on this tour, and they have worked out well. The SDLs have a lot of output and seem to hold up fine."

As touring concert sound systems meet the challenge of offering increased output with less bulk, major rental companies are quick to seek out and field-test new technologies that can make their systems work better. Clair's touring system for Michael Jackson is a good example of this trend. RE/P

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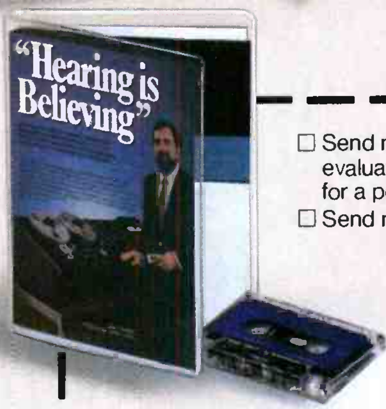

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MIDI-Based Automation

By Jim Cooper

A look at the techniques, terminology, expense, limitations and possible future of MIDI automation.

Just exactly what is MIDI Automation? The term has been bandied about quite a bit in the last couple of years, accompanied by a fair amount of confusion about where MIDI ends and the automation begins.

Jim Cooper is owner of J.L. Cooper Electronics in Los Angeles.

If we add MIDI I/Os to a $100,000 moving fader system, would that make it MIDI automation? Or, would a console that has no built-in "intelligence," and just sends/receives MIDI commands to some external unit (probably a MIDI sequencer) be "MIDI automated"?

Both of these would be, by definition, MIDI automation systems: all that we really need is MIDI connectors (presumably attached to something internally).

In between these two extremes are many other possibilities, each a combination of internal and external capabilities and limitations.

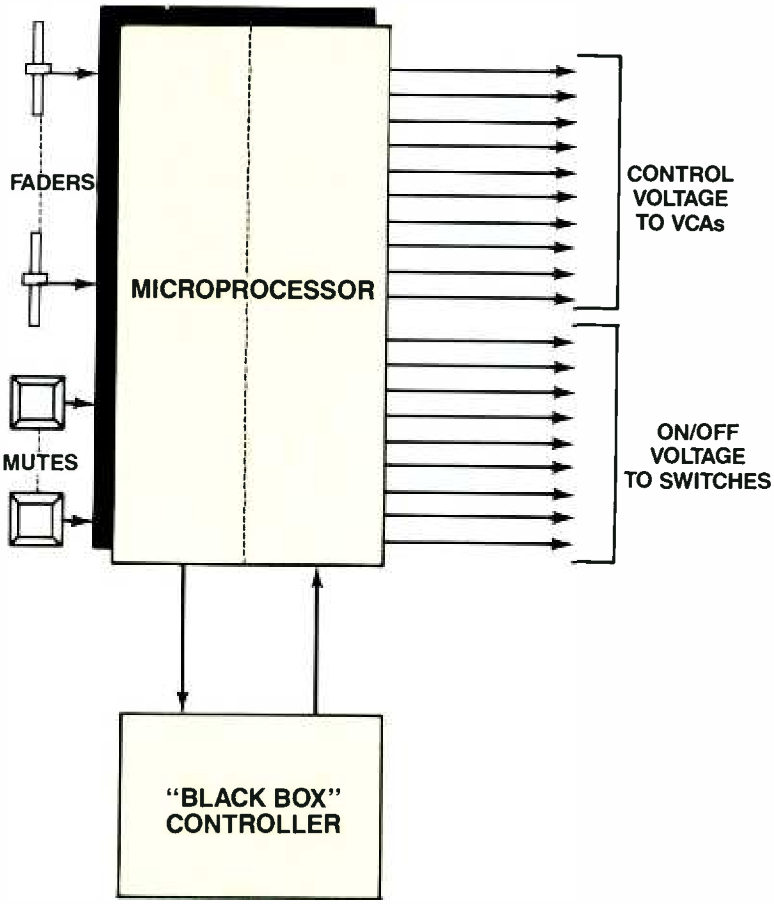


Figure 1. *Separate functions.*

History

As soon as recording and mixdown consoles started increasing in size, engineers found themselves with a dilemma: The consoles could give a great amount of control over the sound, but only if they could control the console.

This need for control begged for computer automation techniques. With automation, the engineer could "teach" the console the fader levels and mutes, perhaps on several successive passes. Ideally, the automation would remember everything done, and would flawlessly recall it all, at will.

Unfortunately, the microprocessor revolution was still a few years away. Instead, clever electronic designers used individual digital "building block" circuits for the logic and used one or two tracks of the tape recorder to store the digital data that represented the fader and mute moves.

Later, lower-cost microprocessors and memory chips made it possible to use them to control the console. Add a floppy disk and you have backup of the automation data. Store the floppy disk away with the master tape, and you can come back in a month and get right back where you left the mix.

Automation has, up until very recently, been the sole providence of the "Big Board." Entry-level automation used to cost at least $30,000. Part of this cost is attributable to the initial development costs and part was due to the "kluge" nature of earlier automation.

Because automation was always sort of an afterthought (usually made by a separate manufacturer), numerous ribbon cables were required to attach the individual fader modules to the automation "black box."

The trend now is for console manufacturers to include at least part of the automation circuitry as part of the basic design of the board. And one magic word has been the driving force behind this trend: MIDI.

The force of MIDI

Since its introduction five years ago, MIDI has changed the soundscape of music. And not long after its introduction, MIDI found its way into accessory equipment, such as reverb and EQ units. Why? MIDI finally represented a "standard" for the transmission of reasonably high-speed digital information.

Up until the development of MIDI, the closest standard was the infamous RS-232 interface. Not only was the exact RS-232 implementation notoriously not standardized, but it commonly operated at a maximum rate of only 19.2Kbits per second (about half that of MIDI.) RS-232 also had no built-in ground loop prevention, which would often lead to lots of digital noise creeping into the audio if the interface was not exactly right.

MIDI has very standard electrical characteristics, operates at 31.25Kbits/second and has opto-isolators as part of the specification. These all make it very attractive as a control medium for applications such as console automation.

The inside story

We can think of console automation as being broken into two halves: the input/output conversion and the actual computer control section. Today, the trend is to have the first half built into the console, consisting of input/output control. The fader and switch positions are continuously scanned by an internal microprocessor, and any change of position or state is sent out as a command of one sort or another. Conversely, commands that are received are translated into control voltages or on/off signals, which in turn control voltage-controlled amplifiers (VCAs), digitally controlled amplifiers (DCAs) or switch functions.

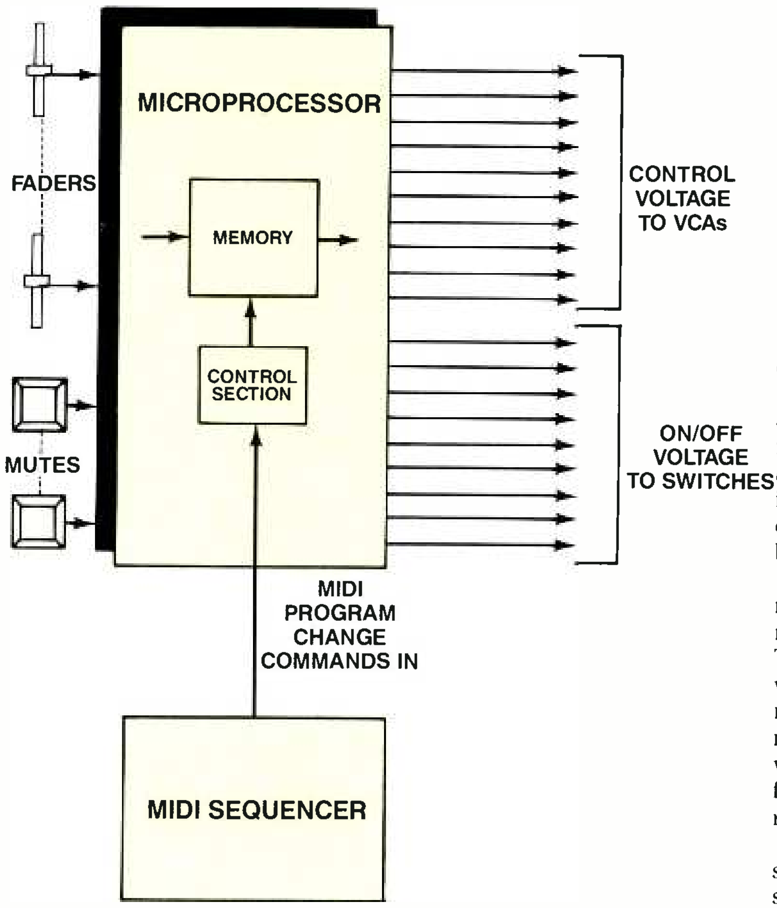


Figure 2. *Snapshot automation.*

VCAs are analog circuits that are amplifiers whose gain characteristics are controlled by a control voltage. In earlier days of automation, VCAs got a bad reputation for altering the sound passing through them. Present VCAs, however, have excellent audio characteristics and should offer little or no change to the sound whatsoever.

DCAs are much like VCAs in function, except that the gain characteristics are controlled by a digital word. This can make interfacing the DCA to the microprocessor a little simpler, but DCAs are plagued with "zipper noise," because the gain changes in little discrete steps rather than smoothly. DCAs may be used for settings you don't expect to change in the middle of a mix, such as EQ circuits.

Switch functions are simple on/off effects, like mute. They can be affected by FET circuits, CMOS logic gates or even small relays. Generally, audio is passed through the circuit, or not, depending on its state.

With these technologies, the console can "know" what the engineer is doing on a real-time basis, and we have means to control as much of the console as we've paid to automate (anything from mutes-only to every switch or knob.) This control information is then sent to (or received from) the *real control function* of the automation. Just how the information is conveyed depends on one of two basically different ways that the control functions work, "snapshot" or "dynamic" automation.

Snapshot automation works on the principle that a representation of the settings (of all attached switches and faders at a given instant) are recorded in memory. This is just like taking a picture of the front of the console. This snapshot would then get a name or number, such as 23. No. 24 would represent another snapshot and would change the faders or switches to a new configuration, and so on.

Later, we could recall No. 23, and the faders and switches would return to their memorized positions. Accessing No. 24 would recall *that* set of positions. This type of automation is particularly suited to stage or theater applications, where the levels and mutes can be broken down into scenes and cues.

It is not particularly suited to normal mixdown automation, since it doesn't accommodate slow, continuous fader movement. True, we could take a hundred "snapshots" while a fader is moving, but that would waste memory, since we would be storing the non-moving faders and switch positions too. And we need to send recall instructions quickly for the hundreds of separate snapshots required during the playback, too.

Mixdown automation really requires some sort of "dynamic" memory. With this type system, all of the faders and switches are "scanned" continuously, approximately every 30ms. During each scan, the fader or switch position is compared to the stored representation of the previous scan. If a position has changed, a burst of data is sent, identifying the control and the new position. In this way, slowly moving and stationary faders can be represented.

Why MIDI?

The way that MIDI gets involved in all of this depends on which type of automation is used. With *snapshot* automation, the actual memory circuitry containing the snapshots is generally housed within the console. Snapshot selection can be made from the front panel of the console or executed by sending a Program Change command to the console from an external MIDI controller. This would probably be some sort of sequencer that is also controlling one or more sound sources and/or effects devices.

Of course, the sequencer would need to be synchronized to the tape. This could be done by using FSK signals or SMPTE time code. Beyond the synchronization requirement, even the simplest sequencers can easily handle the controlling needs of a snapshot automation system. The only requirement is a Program Change command sent at a predetermined instant—just the same as controlling a synthesizer.

With *dynamic* automation, the data representing fader or switch movement is generally sent by either Controller Change or Note On/Off commands. These commands would then be sent to either a regular music sequencer or a special purpose sequencer dedicated to automation needs.

For dynamic automation, very little memory is needed within the console. In fact, the hardware requirement within the console can be provided by a very simple microprocessor system. On the other hand, the requirement of the sequencer section is much more demanding, because of some very basic needs of the dynamic automation.

First, general-purpose MIDI sequencers are all designed with the assumption that a music piece starts from silence; no note is already sounding at the initial downbeat. Since the automation system sends exactly the same type of commands that a synthesizer would, you can see that the sequencer also has no sense of fader position prior to the downbeat. Until the first point in the song, when a fader is actually moved, no data is sent from the sequencer to the console, telling the VCAs where to set their gains—the same is true with switch functions.

One way to get around this problem is to include, within the console, the capability of storing some snapshots that could define the initial settings and have the sequencer send a corresponding Program Change command at the very beginning to recall this snapshot.

Alternately, a button on the console could send a burst of data representing the positions of all faders and switches. Push it at the start of the first sequence record pass. Then, at the beginning, each playback pass would have this burst of data.

Cut to the chase . . .

Starting the tape in the middle of a take creates another problem. In automation terms, this is called "chasing" to a cue, and the ability to chase is essential to modern mixdown efficiency.

If you start a non-dedicated sequencer in the middle of a song, it *may* search through its memory from the beginning to the desired point and calculate the most recent Program Change command and send it. But it almost certainly will not address the status of all Controller commands nor figure which notes might be currently on.

A solution for this limitation might be to have the automation system send a burst of data, representing the position of every fader and switch, every few seconds so that it gets written into the sequence. Then, on playback, you would never be more than a few seconds from one of these bursts.

Some strange problems arise when trying to overdub or update fader or mute data with regular MIDI sequencers. For instance, if you wish to redo one given fader, the sequencer has to be able to erase just one Controller number's data and replace it with the new commands. Only a very sophisticated computer-based sequencer *might* have this capability.

Up to this point, the problems listed above really have nothing to do with MIDI per se. Both hardware units and software packages that address these specific issues, and communicate via MIDI to VCA/fader/mute sections, are currently available today.

The problems arise when, under the umbrella of MIDI, we assume that every combination of MIDI equipment will do what we expect. This is not the case. To avoid confusion and disappointment, the buyer absolutely must become fully acquainted with a systems options *before* committing to purchase.

There is a very real limitation when using MIDI to carry automation data between two units, as from a sequencer to a console. The limitation is MIDI's bandwidth. MIDI has a specification of 31,250 bits per second, which amounts to 3,125 bytes of usable data per second. This is the MIDI bandwidth.

A typical MIDI Controller Change command takes three bytes of data: one byte to define what type of command, one to identify the fader or switch, and one to describe the position. Therefore, a little more than 1,000 commands can be sent each second.

Let's say that we expect our system to be "frame-accurate," which means it must be able to describe any new fader position or mute status within one-thirtieth of a second, or about every 33ms. Dividing 1,000 commands per second by 30 frames per second gives 33 commands per frame.

This means that a maximum of 33 faders and switches may be changed in any frame. If a 34th fader is moved, its position data will slop over into the next frame. If we had 64 faders, all making a gradual fade from full-on to full-off, and had a mute turn on in the middle, its turn-on time might get delayed quite a bit.

This example is pretty extreme. The expectation of more than 33 faders moving at once is unlikely in all but very large systems. And if there are subgrouping capabilities in the system, a single subgroup fader move can affect several channels at once, without cluttering up the MIDI cable with commands.

But it should be clear that there is a real limit to what we can expect from MIDI automation. In a two-part system, we can't expect to have a 32-input board with faders, sends, returns, EQs, pans, mutes and bus assignments all automated and controlled over a MIDI cable.

Future vision

There will be some interesting economic considerations at work in the future. In a small system (say 32 channels of faders and mutes), the cost of the VCA circuits, the microprocessor and its associated circuitry are about the same—if there is not a lot of memory (for dynamic memory storage) or fancy SMPTE decoding circuitry. A very simple microprocessor design can handle the input/output function of the board. At this point, it makes sense to allow the customer to use the sequencer or computer that he already has to provide the "smarts."

But, just about the time that a system runs into the limitations of MIDI's bandwidth, the cost of items such as the VCAs and controllable EQ starts to be high enough to warrant putting more capabilities into the internal microprocessor circuitry. In a large system, why use an external MIDI sequencer, with its problems and limitations? And why rely on MIDI's bandwidth to carry all of the data?

Market pressures are sure to bring down the cost of built-in automation. And I expect that we will see more and more totally self-contained systems, with dynamic storage memory, full functionality and SMPTE synchronization included.

Certainly, MIDI inputs and outputs will be included in many future systems. [See "MIDI and the Mixing Console" in this issue and "MIDI Control of Effects: One Designer's Viewpoint" page 62, November 1988.] This would allow Program Change commands to be sent to peripheral equipment and would allow the transmission of data to an attached (and non-essential) computer that would act as a graphic display only. But the notion that MIDI is an important controlling element within the automation function will be gone. It will just be an appendage.

We started with self-contained automation systems costing tens of thousands of dollars. MIDI opened up the door to economical automation, at least in part, by making us know that it can be economical. But in the long run, MIDI's importance within console automation may well recede.

RE/P

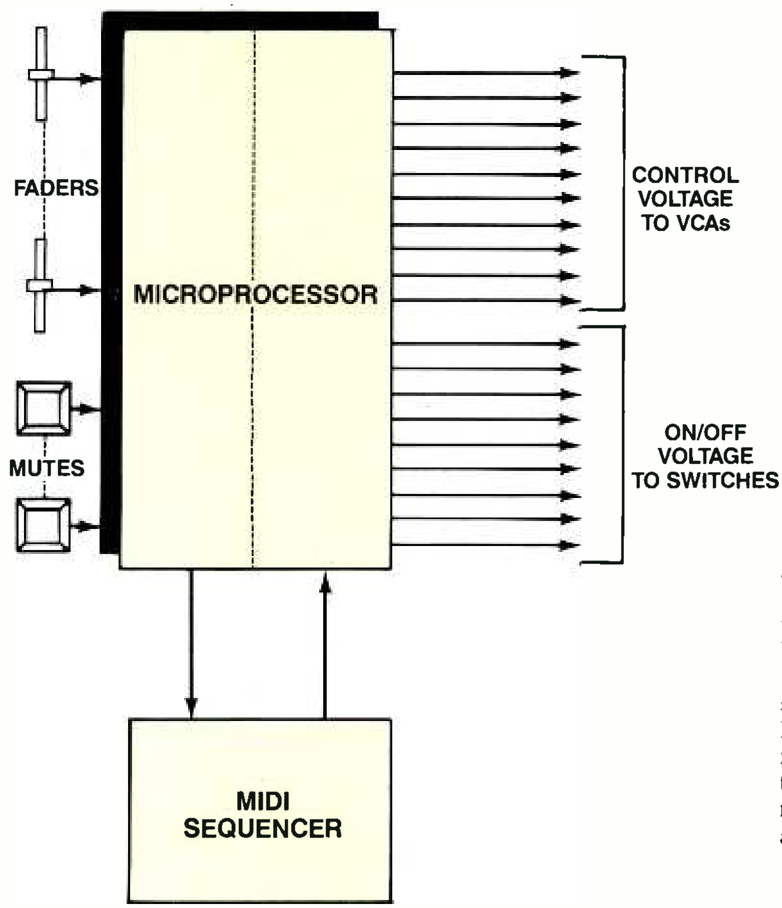


Figure 3. *Dynamic control.*

Post-Production:

The BBC Radiophonic Workshop

By David Mellor

A look at the British Broadcasting Corporation's in-house music production service.

Imagine one organization operating two national TV networks, four national radio networks, local TV and radio services for a country of 60 million people, and an overseas radio service covering most of the world—that is the British Broadcasting Corporation.

I mention this, not to impress by size, but to show how pervasive the BBC is in British society. The number of Britons who cannot receive a full range of BBC services is very small—the odd Scottish islander or Welsh valley-dweller. Doubtless, repeater stations are being built at this moment to serve them.

No BBC broadcasting service carries commercials. Funding is via a license fee—payable by any household that has a television—income from program sales, and a lively merchandising operation. Many people take the view that we have in Britain an ideal system whereby the BBC sets the standard of quality for commercial broadcasting organizations, and the commercial stations make sure the BBC doesn't get too complacent about it.

As with any TV or radio service, the BBC has a large appetite for music. Some of this appetite is satisfied by its in-house music production service, the Radiophonic Workshop. This consists of a team of composers, technicians and administrators, 11 in all, whose job it is to create primarily electronic music for all parts of the BBC operation.

The quaintness of the title "Radiophonic Workshop" can be traced to its beginnings in 1958. Brian Hodgson, formerly a workshop composer, is now in command.

David Mellor is a free-lance technical writer living in London.

Workshop composer Richard Attree.

Workshop composer Roger Limb.

"When the Workshop was first set up it did experimental drama," he says. "Lots of science fiction, using many more effects than we would do nowadays. They would very seldom do things like signature tunes. Music came later on, because the original way of working with laboratory equipment and tape manipulation was not really conducive to getting sounds pitched very accurately. It wasn't really until the introduction of the synthesizer that music became our main function."

Early equipment was primitive, to put it mildly. It consisted of a tape recorder made by a Swiss motorcycle manufacturer, another that needed a 2-hour warm-up before it would run at a steady speed, and a more-sophisticated machine with two speeds!

Sound production relied on lab gear: kit-built sine/square wave oscillators and a frequency-modulated oscillator (known as the Wobbulator). Eventually, an 8-note keyboard was built. It could switch the outputs of the oscillators—each oscillator tuned to a note of the scale.

Current equipment is somewhat more sophisticated. The Radiophonic Workshop now has six composers, each with a studio. All studios, at the BBC premises in the Maida Vale area of London, are available for continuous use.

"It's very important that a composer has access to the place 24 hours a day so that he or she can work when the muse is in, or with TV work, when the film is ready," Hodgson says.

Each studio has a comprehensive collection of synthesizers and recording equipment. Hodgson stresses that the BBC does not endorse any equipment, but buys whatever will be useful.

The tendency now is to work as much as possible with MIDI and reduce the amount of recording necessary. Most of the workshop's mixing consoles are by Soundcraft, tape machines mostly by Studer. But the Studers have been gathering dust recently.

"We are finding that our use of multitrack is diminishing considerably, being able to control and keep so much inside the computer," Hodgson says. "In one program we did last year, multitrack wasn't used at all. It only went to tape so we could take it over to TV center and play it.

"We've actually mastered quite a few programs on R-DAT as an experiment. You can't sync R-DAT to video, but it doesn't really matter whether you can sync it or not. If your speed stability is all right and if you can get it to start on time, you're OK."

Although it used to be the case that some studios were better-equipped than others, the aim has been to bring each studio up to the same standard with Apple Macintosh IIs controlling a selection of synthesizers. This aim has been largely achieved, but there is some traffic of equipment between studios. For example, there are not enough R-DAT machines to go around yet. They are still being evaluated.

Composers

In the early days of the Radiophonic Workshop, the staff members were known as studio managers rather than composers. "Studio manager" is BBC talk for engineer. (A studio control room, for the record, is known as a cubicle). Because the main responsibility was for sound effects, this was, at the time, a sensible title.

"They would all come from the ranks of studio managing in Broadcasting House (the BBC's headquarters), usually from the drama department," Hodgson says. "The music department at that time was suspicious of anyone from the 'funny noise' department.

"The term *studio manager* was changed to *assistant, Radiophonic Workshop*, then to *producer, Radiophonic Music*. When I came as head of department, I said, 'This is ridiculous. They earn their living composing music—we should call them *composers*.' "

Originally, workshop staff were taken from the ranks of studio managers because they could operate the equipment. Eventually it was found that people were leaving music college and entering the BBC as studio managers, purely in the hope of getting transferred to the workshop. It wasn't until 1985 that a composer was directly appointed from outside the BBC. Of course, a composer has to have a solid technical background.

"Composers would have to show that they could make music unassisted in the studio. They would have to send tapes of their work," Hodgson says. "We are not interested in somebody who is using the synthesizer to sound like a bunch of badly played instruments—or even nicely played instruments. We are interested in electronic music, not in the synthetic part of it."

So the composer has to be able to invent music, invent new sounds and engineer, as well. The Workshop *does not* employ recording engineers.

"An engineer will get between the musician and the equipment. I remember all the tensions in the early days of electronic music, when every studio had an engineer who interpreted the composer's wishes," he says. "We always felt that was the wrong way to do it. We were putting a barrier between the composer and the equipment.

"I would expect composers to understand enough about the equipment to make usable sounds, and the sounds *they* wanted rather than the sounds an engineer said were available. There is no substitute for doing it yourself."

Being a Radiophonic Workshop composer may seem a very solitary occupation, working alone in the studio day after day. Fortunately, since there are six composers, and other staff, there is frequent interpersonal contact, which brings several advantages.

"If you have one composer working with an advanced piece of technical equipment, that person is inclined to say that it's his own fault if something goes wrong with it," Hodgson says. "When you've got three composers in the same building working on advanced equipment, the odds are that the one who's having the problem will say to one of the others, 'I'm having some trouble doing that, what am I doing wrong?' and the other guy will probably say 'Yes, I was having the same problem' and the third will say, 'It's not your fault, it's just badly designed.'

"We can go back to the manufacturer and say 'Look, *we've* got a problem. This is going to become a major difficulty for other people using this equipment.' That's why manufacturers like us using their equipment. They get a lot of feedback.

"Also, the composers compete in a friendly way. If they have an artistic problem they will discuss it among themselves. You get that sort of comradeship. The fact that it's a small department (there are only 11 of us) means that we have departmental meetings three times a day—coffee, lunch, and tea in the afternoon."

Although much of the Radiophonic Workshop's output is for major TV productions, even the smallest, most remote BBC local radio station can ask the Workshop for a signature tune, or incidental music.

"In television, because of the size of the budgets, productions are costed individually," he says. "In local radio, where you are talking about a situation where the total budget for a program may be a few hundred dollars, then radio, as a whole, pays its share. It doesn't actually split the costs down for each program.

"It wouldn't be feasible to provide any form of service to local radio if they had to pay for it on a by-the-hour basis. They say, 'we have a job for you,' and if I have somebody free to do it, then we will do it. We don't say you have to mortgage your budget for the next 10 years to pay for the signature tune."

In many cases, the BBC uses more familiar ways of obtaining music—free-lance composers writing for orchestral or other instruments. Producers are given a free hand in sourcing their music. No persuasion is needed to make them use the Radiophonic Workshop. The reason they do use it is, as Hodgson says, the very efficient service provided.

Even though the Workshop is concerned mainly with electronic music, conventional instruments and musicians are often used.

"Strangely enough, the more computers you've got around the place, the more you tend to use live instruments," Hodgson says. "There's a very nice sort of tension you get in music between the strictness of the computer and the freeness of the musicians."

MIDI

The Radiophonic Workshop got into MIDI almost as soon as it happened. The Yamaha DX-7 was bought when it first became available, then the Yamaha QX-1 sequencer was ordered in large quantities. The QX-1, useful though it was at the time, was destined to be replaced by something more versatile—the Apple Macintosh.

"We were interested in the Mac from the moment we first heard about it," he says. "Eventually we managed to track one down and give it to one of our composers to see how he made out with it. It became obvious that it was the way to move.

"We really went for the Mac because the software was there for it. A lot of other computer companies were saying the software is going to be available next year. We were not interested in next year. We bought our first one in October 1986."

Software in current use is Professional Composer and Professional Performer by Mark of the Unicorn. Workshop composers need to get at every aspect of MIDI data. Other software available at the time did not allow as much control. Another advantage is the music printing capability of Professional Composer.

"It's wonderful to be able to print music directly from the computer. When you get down to the session and the musician says you have written it in the wrong key, there is no problem. We can easily replace synth lines with real instruments," Hodgson says. "I think there is a general fashion in the workshop to do so. It's very healthy to get away from the live and electronic being in opposition. We want to enrich both sides."

The change into computers—even a computer so widely regarded as being easy to use—required training for each composer. The first composer to get the Mac was Peter Howell. He then devised a training scheme for all the others. Each composer, in turn, was taken out of front-line service for a month while getting used to the new system, but worked on real projects rather than having more formal "schoolroom" training.

The next change to the workshop's studios will be to bring the mixing console and audio routing under MIDI control. The introduction of small MIDI-controllable consoles like the Yamaha DMP7 makes this possible. In an experimental studio, several DMP7s are used together with a Macintosh-controlled Akai routing matrix, instead of a conventional console. The idea is to make the system more suitable for one-person operation, with all functions within arm's reach, or under the control of the computer. Software for the system (using Hypercard) was designed by the Workshop's development coordinator, Mark Wilson, in partnership with Peter Howell.

The future

The Radiophonic Workshop of 1989 is very different from that of 1958.

"I see a movement back to a lot of sound manipulation," Hodgson says. "The danger at the moment is that there are a lot of synthesizers around that come with 100 or more sounds in them. Everything is going to end up sounding like a glorified Hammond organ if we are not careful.

"The next movement will be back to the creative use of sound. Developments in analysis and resynthesis should open up new areas of sound manipulation and creation. I can see no artistic advantage in sampling a trumpet and using it as a trumpet. I can see more advantages in sampling a trumpet, then taking that sound to pieces, then putting those pieces back together in the way you want.

"We would hire a musician to sample if necessary, but it would be made quite clear why we were hiring him and what we were doing. It wouldn't be to make trumpet notes. It would be to get inside the sound itself. It's a waste of time trying to imitate an instrument. If you have all the expertise of a musician who has learned to play his instrument beautifully, get him to play it beautifully. Very often a producer will say to me, 'I want the sound of an organ chord.' I tell him, 'Go and book an organist; it's cheaper. If I do it, it's going to cost you $375.' They say, 'I never thought of that!' "

By explaining to a producer why he should use conventional musicians rather than an electronically generated imitation, Hodgson may be seen as talking the Radiophonic Workshop out of a job. In fact, he is putting electronic music into its correct perspective—creative rather than imitative—and setting a standard to which many electronic music producers would do well to aspire. RE/P

Engineer/Producer Interview: Ed Seay

By Dan Torchia

With roots in pop/rock and R&B, Ed Seay has recorded and produced some of the most popular country music in recent memory.

Listen to country music lately, you'll hear something different. Sandwiched in between the mainstream artists are new sounds. It's not just in the performances; the changes reach down to the production and engineering values.

Engineer/producer Ed Seay is at the forefront of this revolution. As an engineer/co-producer with Paul Worley, or as an engineer on his own, he has recorded some of the hottest talent in country, such as Highway 101, the Desert Rose Band, Ricky Skaggs, and Foster and Lloyd.

Dan Torchia is staff editor of *RE/P.*

For all his success in country music, Seay is a relative newcomer to the field. He moved to Nashville only four years ago. For 13 years, he engineered and produced in Atlanta, where he worked on pop/rock and R&B, most notably with Paul Davis and Peabo Bryson.

Throughout his career, Seay has shown a willingness to experiment with new technology—such as using drum machines and a Synclavier with Paul Davis when both technologies were in their infancy—while using such time-honored techniques as cutting in a live room with careful mic selection and placement. For Seay, the ultimate judgment is in the performance, not the technology used to get there.

RE/P: Let's start off with your background.

ES: Obviously, I live in Nashville now. Four years ago, I moved from Atlanta, which is really where I started my recording career.

In fifth grade, I started playing trombone and played through high school and college. In high school, I picked up guitar and bass and played in combos and the whole thing. But I didn't want to be a band director because I was looking for something

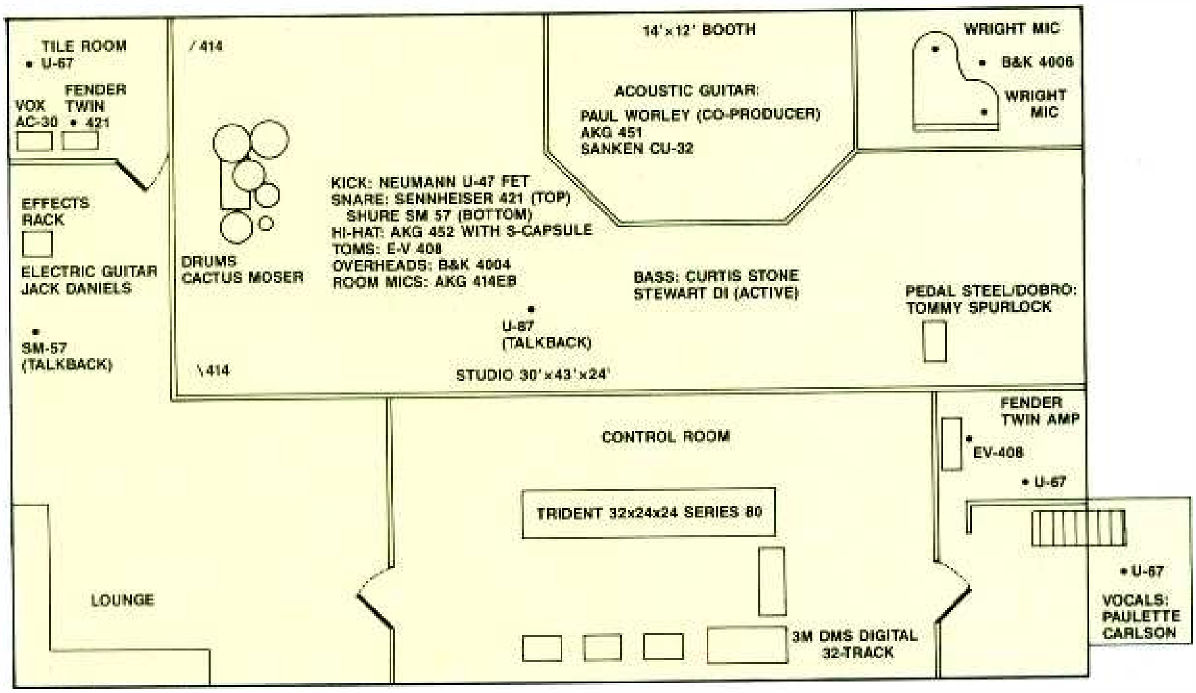


Figure 1. *Room and mic layout for Highway 101 sessions for "Just Say Yes," "Somewhere Tonight" and "All The Reasons Why" at Treasure Isle in Nashville.*

else. So I decided to get into journalism—advertising. I went to the University of Georgia and got a degree in journalism, and moved back to Atlanta trying to find a job in advertising. And there weren't that many jobs in advertising, especially with someone who was half-heartedly qualified. I had very little print material to show, but I would say, "Listen to my tape," which I had done down in the journalism school, bouncing stuff back and fourth.

Finally one man, a creative director, had the vision to say, "You don't need to be here, you need to be in a recording studio." And I said, "You're right." Off came the coat and tie. I went to Master Sound Recording Studios in Atlanta. I popped in and explained, "Hey, I don't have any experience as a recording engineer, but I got a college degree and I'm a musician with good ears, and put me in, coach!" So he did.

I started running high-speed duplicators and then assisted in the voice-over studio. At night, I hung out and learned the studio end of it. That really was my start.

After 2½ years of Master Sound, I moved over to another studio in Atlanta called Web IV. Web IV was a private studio owned by Bang Records. Bang had this studio exclusively for its artists. It wasn't open to the public, and it was the ultimate learning opportunity.

Our duties were to record the writers' demos for the publishing company and the artists who were on Bang Records. We had a lot of mics, and we had a lot of outboard gear, more than anybody in town. To make up for our Electrodyne board—which was a good board, but didn't have the expanded capabilities that some others did—we tried every trick in the book, every mic in every place you could imagine, and we had every compressor that was made at the time.

RE/P: *That sounds like a good environment.*

ES: It was unbelievable, because today when you walk in the door, there is no opportunity to play around with the stuff. And if you don't use it, you can't learn how to do it. It's like reading a book. You can read a book on driving a car, but until you get behind the wheel, run off the curb a little bit and figure it out, you don't know.

RE/P: *While in Atlanta, you had some big records with Paul Davis. How did you get involved with him?*

ES: When I came to Bang Records, Paul was one of the artists who was on the ground floor. He had "Ride 'Em Cowboy," which was a moderately big record, but he was at the point where he was considering being more of a writer than an artist. He played me a piece of a song, "I Go Crazy," just on the Fender Rhodes.

It was really interesting the way it came down. We cut it to a Rhythm Ace—remember there were no drum machines at this point—it was a Korg Rhythm Ace, not really very soulful. Paul played Rhodes, some synthesizer, some fake string pads. I played electric bass. Paul sang lead, and I went out and sang some background. There were still no drums on the thing.

Paul wrote it to be a followup to pitch to Lou Rawls, who was coming off of "You're Gonna Miss My Love." The president of the label, Ilene Berns, said, "There's no way you're going to give this song to anybody, you're going to put it out yourself."

I remember on my birthday I flew to New York to CBS and did a string date—the tracks still had the Rhythm Ace going. A good friend of Paul's, James Stroud, flew in and played drums to what was virtually the finished record and the Rhythm Ace. That became the record, we mixed the thing and put it out.

That was one of the first big records that I had been involved with from the ground up. It was what we used to call a layer cake production. It wasn't a full band or a room full of guys, we layered it up. We did a lot of that in Atlanta.

RE/P: *Why did you decide to move to Nashville?*

ES: I had spent 13 years in Atlanta. Atlanta was one of those cities that was always about to become a great recording mecca, but never quite did. There are three basic markets in the United States—New York, Los Angeles and Nashville. And although other cities now have the same gear that these cities have, they don't have

Selected Discography

Titles with an asterisk (*) indicate co-production.
Brick: "Brick."
Peabo Bryson: "Peabo," "I Am Love."
Paul Davis: "Cool Night,"* "Paul Davis,"* "Singer of Songs-Teller of Tails," "Southern Tracks and Fantasies."
Desert Rose Band: "Desert Rose Band," "Running."*
First Call: "An Evening in December."
Foster and Lloyd: "Foster and Lloyd."
Highway 101: "Highway 101," "Highway 101²."*
Melissa Manchester: "For the Working Girl," "Melissa."
The McCarters: "The Gift."*
Nigel Olsson: "Changing Tides,"* "Nigel Olsson."
Marie Osmond: "I Only Wanted You," "All in Love."*
Sandi Patti: "Morning Like This," "Make His Praise Glorious."
Ricky Skaggs: "Love's Gonna Get Ya," "Coming Home To Stay."

the tech support, they don't have the rental companies, they don't have the manufacturing reps, they don't have such a concentration of the great players. I'd done some work in New York and Los Angeles and liked them both. But coming out of Atlanta, Nashville seemed like the logical place.

RE/P: *Did you have much of an adjustment period?*
ES: Actually, it was a very painless transition. I was obviously worried about this. You always say, "Well, what's going to happen?" But I had a few things working for me, I believe. I did have a track record in the years I'd spent in Atlanta. So when I moved, people paid attention to what I'd done before.

I also was fortunate to have been on the ground floor of some really neat technology before anyone in Nashville, actually. We had the first Linn LM1 drum machine in the Southeast. Paul Davis bought the thing because he'd heard about it. He paid full retail because he needed it and he wanted it. We heard that Stevie Wonder had 12 of them; well, we wanted one, too.

After the synthesizer started catching on, we got wind of the Synclavier. Paul was at a successful stage in his career where he felt he could afford to buy one. To this day, he's still got it, it's still working, and it never became obsolete. So I had worked on the Synclavier.

The third factor that I think helped was that in Atlanta there were bands, lots of rock bands, pop bands, R&B bands. In Nashville, the thrust is more studio players. So it's easier to walk in and get a session started when the five guys on the floor are some of the best players in the world and they're giving you the best sound available. But when you're walking into a band situation, they don't always know how to play in studio, don't understand the studio, don't really know what the phones are about. It's tougher to make that come off.

Actually, it took me two weeks to get work.

RE/P: *Was it a country thing?*
ES: It was a contemporary Christian thing. The producer was Greg Nelson, Sandi Patti's producer, who is very attuned to new sounds. What he was looking for was not a country thing at all. He was looking for more of, for lack of a better term, more of an L.A., more of a pop thing, with verb and explosion and impact. I had come out of that school, so I knew how to deliver it.

RE/P: *How did you get into working with country artists?*
ES: I had worked on a project, a developmental project, with a lady. Her husband was a musician/producer in town, Paul Worley. Paul, through his wife, had heard about me and needed some remixes and a single for Marie Osmond. The song was "There's No Stopping Your Heart." I pumped it up and made it explode a little bit more than perhaps it had before.

Shortly thereafter, Ricky Skaggs, who at the time was the Country Music Association Entertainer of the Year, called and wanted me to do a few overdubs. Then he started saying, "Well, let's book some more time here and there," so that grew.

Prior to moving, I had not really worked on any country music. Because it didn't make sense—like I said, Atlanta was pop, rock, R&B—but if you wanted country, you drove four hours north and worked with the best country musicians in the world.

RE/P: *I think it would be an asset, not having worked on country before, because you have a different perspective.*
ES: I think so. It let me bring a different flavor to it. I wanted to work within the country format, but create something a little more exciting, a little bolder, try to get a little more attention.

RE/P: *Many of the artists you've worked with—Highway 101, Foster and Lloyd, Desert Rose Band and Ricky Skaggs—have distinctive sounds that you don't ordinarily hear.*
ES: I agree with you. I think in those cases, Desert Rose and Highway 101, both are bands. Foster and Lloyd was cut as a band. When you use studio players, there's something gained, the incredible speed of how fast you can cut the stuff. But I think there's also something gained when you cut with a band, because you get each individual member's flair, his own little signature and the way they interact. That and the fact that cutting in a live room makes it sound like it does if you're out in the room or the club, rather than making it sound real "studio."

RE/P: *The opposite of the layer cake.*
ES: Yeah. People, when they turn on the radio, don't say, "Well, that's closed-miced and that's real studio." But they do realize, "That sounds honky-tonk, I can relate to that." So I'm not opposed at all to try and come up with a signature sound.

RE/P: *How do you approach recording the rhythm section, for example with Highway 101?*
ES: The studio that I generally prefer for tracks is Treasure Isle. It's got 30-foot-high ceilings, and there are wood walls, floors and ceiling. What we do is set up the band as live as possible, but with isolation booths, live ones, so that it doesn't sound like you're recording in a closet. We put the drums in the livest part of the room, and close-mic the drums. On snare, I like Sennheiser 421s or Shure SM 57s; on kick, the Neumann U47 FET.

Lately on toms I've been getting into the Electro-Voice ND408s—what I like about them is that they go heavy on the tone; they underplay the attack and overplay the tone. For the same reason, they're not my favorite choice for snare or kick, but for toms, it really fattens them up. On hi-hat, I like an AKG 452 with an S-capsule, overhead—lately I've been getting into B&K 4004 omni mics for the cymbals.

Then I use room mics. I think that when you go to a hall or a club, you don't just hear the dead sound with reverb created from digital box, you hear the room, so we mic the room to get the explosion and make it sound real, to give it some life.

We put the acoustic guitar in the big booth that must be 20'x10' or 18'x10', I guess. The guitarist, or steel man, we put him around the corner in the actual room, but we put the amps in a little booth so that we can get a big sound on the amps, back the mics off if we need to, but still not get it so full of drums that we have no control.

Then there's a piano booth back there that the piano man usually is in. A lot of the time, Paulette Carlson, the singer, will go up to the second floor and we'll have her there. If we had her on the floor, we'd have too much contamination in the vocal, for lack of another booth.

For guitar amps, I like 421s, 57s, the 408s. For steel guitar, 87s. For piano, I've been using three mics, a couple of Wright mics, one on the high end and one on the low end, and a B&K 4006 omni to get the big picture.

***RE/P:** Is that your basic setup?*

ES: Pretty much so. Sometimes, if there are more players, we'll even put people out in the TV lounge, which is not a bad-sounding room.

***RE/P:** How do you record Paulette's vocals?*

ES: The signal path is almost always the same, the only variable has been the mic. From the mic, I use Monster M1000 MkII gray cable run straight to my Massenburg Lab pre-amps, to an LT Sound CLX2, which is a fabulous limiter/compressor/expander. Out of that, I go into the GML parametric equalizer, then straight XLR out the back of that to the tape machine.

That's generally the path, although we're always in a quest for a mic that she wouldn't tear up. Paulette's an incredibly loud singer. She can sing soft, but then she can get very loud, a lot of VU. Not peak as much, just raw VU, power. She's melted down a few U47s before and a few other mics, so what we finally wound up with were 67s, something we could pad. We've got some hot-rodded 87s that we like a lot, too.

We actually did something rather unorthodox for "Cry, Cry, Cry," but we had a No. 1 record, so it must have worked. We stuck a Crown PZM on a board and we mounted the board on a music stand. We set her up and did overdubs at the Money Pit, where we usually overdub. She was singing into this board and she could not break this PZM up. It took a little EQ to bring back the body. Because the mics are hemispherical omnis, it rejects any of the control room glass.

***RE/P:** With Foster and Lloyd, was your approach different because they wrote the material and were the producers?*

ES: In a sense. With Radney and Bill's project, they gave me some demos. They were good, the structure was already there, because they're good songwriters and they write the licks into the song. I heard the demos and said, "Well, yeah, I know what you're looking for, and I know how to get what you're looking for."

They wanted the snare to crack. They wanted it to be big. They didn't want it to be overproduced, they wanted it to be raw. So once again, we miced up the room. We used a direct drum kick so that the room mics would only hear the snares, the toms and the cymbals, and it wouldn't hear the foot. The foot is cool to get into the room mics, but that can sound a touch like Led Zeppelin, and for some people that's the red flag that says, "Hey, wait a minute, this is rock, this is not country."

The main difference with the 101 project and the Desert Rose project, as opposed to Foster and Lloyd, is that I'm co-producing 101 and Desert Rose with Paul. For Foster and Lloyd, I was hired as an engineer, as I am with Skaggs.

***RE/P:** If you're hired just as an engineer, do you approach the project differently than if you were also producing?*

ES: In a way, I do. If I'm hired to engineer vocals on Project X, I'm just hired to engineer vocals. There's very little latitude. We all know what vocals are supposed to sound like. In that capacity, I consider myself to be a photographer using a Hasselblad camera and getting a very accurate representation of the sound source.

When I mix, I can assume the role of an artist. And now, let's paint the guitar, let's paint that tree blue instead of green, or let's tint the face a little redder. And with the projects that I co-produce, obviously, that input is warranted.

***RE/P:** Let's move to the technology, starting off with your use of the 3M DMS.*

ES: When I moved to Nashville, that was the first digital multitrack that I ever worked on, other than 2-track digital systems. So I've been spoiled. I've heard it. It knocks me out. I've never heard a better digital system. The transports can be a little cantankerous, and the error correction is very touchy. They can be a little trouble to maintain, too. However, they sound fabulous. Paul [Worley] and I bought one of these and put it at the Money Pit.

***RE/P:** Have you worked with any of the other digital machines?*

ES: We have used the Mitsubishi. It was pretty heavily modified with Apogee filters, so it sounded OK. The Otari DTR900 with Apogees sounds good. The Sonys sound OK.

I love digital, because it becomes a mirror image of what we're hearing out there. However, I think that good analog is much better than bad digital. But I think that good digital is better than good analog.

***RE/P:** I was interested by the credit given to Monster Cable on the new Highway 101 album. Are you a big fan?*

ES: I'm a big fan of Monster Cable. I'm part of what they call the Famous Monster program. I'd heard of the cable, and we did a shootout one night. I was knocked out. It wasn't just me, there were four other people in here. It was about 10 at night and we heard the difference and it was like "Man! I just can't believe it." So I ran out and bought 80 feet of the stuff, right off of the bat.

***RE/P:** What difference do you hear?*

ES: I hear extended top, extended bottom. I hear clarity. I hear less smear in the midrange, the presence. It's most obvious on acoustic instruments: drums, acoustic guitars, things that do have harmonic content. It seems to bring the sound in focus. There are other good wire manufacturers, too, but I've been impressed with the Monster. I hear the difference, and if I didn't like it, I wouldn't use it—even if they gave me miles of it.

***RE/P:** What about consoles?*

ES: My favorite stock production console is the Trident, the 80s in particular and also the Trident 80Bs. The main thing I like about them is that even though they're very simple, they're fabulous for tracking because there's virtually nothing in them. They have incredible punch and clarity, and the transient response is very good. It's almost like the sound's going on out in the studio and it roars through the board and slaps onto the tape. As opposed to some consoles, which shall remain nameless, that I feel like I've got to EQ the sound and pull it through the board. I just like the design philosophy.

I'll tell you another thing that makes a big difference on these boards, and probably all boards, and that's don't stick to the stock power supplies. Use the oversized, or the hot-rodded power supplies, because it really makes a difference in the headroom and the performance. It's like a small Honda Civic vs. a Toyota Supra. You've got the power when you need it. When you're tracking and everybody's rocking, that's when you need the power, not when you're doing one guitar overdub.

***RE/P:** Let's go to your outboard gear.*

ES: I'm a big fan of GML pre-amps. I really like the way they sound. There are other good ones out there that I haven't tried, a few I have tried that I'm not impressed with, but the GML sounds more accurate, more clear, more transparent, more *real.* If there's a problem in the signal path, chances are we can eliminate the pre-amp if it's a GML.

Overall, my favorite compressor is the CLX 2, made by Lacy Thompson, of LT Sound. He's the guy who makes the Vocal Eliminator, with the ads saying "A Singer's Dream" and you think, "What is this?" But the guy is a genius, a musician as well as a tech whiz, and I can do things with that compressor that I can't do with the others.

In addition to that, I have a hybrid EQ box that a tech friend, Les Duncan, who runs a tech support company, built up for me. I said, "I want some EQs from some consoles and some from others and I don't want to buy all these consoles, I just want to buy the EQs." So I've got APIs and Spheres and graphics and three knobs and some outboard pre-amps he built that sound fabulous. There's a power supply that he built up and mounted in this thing.

RE/P: *What about your use of microphones?*

ES: I definitely do not believe that one size fits all. The problem is that some people approach it as, "I bought this mic, I paid $2,500 for it and it *will* sound good."

Case in point: We were doing backgrounds on a project with a female vocalist and a male vocalist. We had a C-12 set up and we also had a U47 set up, a tube U47. The woman was very breathy, not much fundamental and a lot of harmonic in her voice. The guy, John Cowan of the New Grass Revival, has a lot of fundamental and not a lot of harmonic, just a lot of tone. So the suggestion was made, "Well, why don't we put her on the C-12 and John on the U47?" Well initially, I didn't think that was going to be the best plan, but I said, "Let's try it."

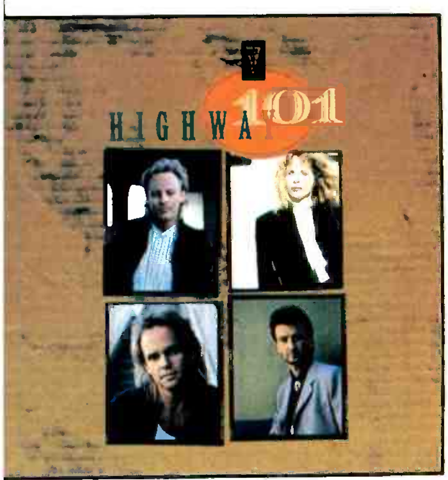
Well, it turned the woman's voice, with a lot of harmonic, to sandpaper. Conversely, it turned Cowan into "tone man." But when we switched, the C-12 added the air to John's voice that he didn't have, and the 47 added the body to the female voice, which she needed. Of course, the EQ can make up for a lot of that, but when the mic is already leaning one way, you might as well use that to your advantage.

RE/P: *How do you work with artists and interpret what they want or what they're trying to tell you?*

ES: Well, if they say, "Hey, I think we need 5dB at 10k," that's pretty clear. I know what they think they want and probably what they do want.

RE/P: *Or something like, "I need it a little warmer."*

ES: Yeah, "I need it a little warmer...it needs more air." Or, "It needs to sound wooden," or "It sounds too wooden," or "It's got a honk to it." Well, generally, the wooden and honk lie in 600 to 800 cycles, maybe even higher. The air is obviously up 10k, 12k, 15k. Generally, being a musician is my biggest advantage because I'm not technical. I'm like a test pilot. I can fly the ship. I can't fix it when it breaks.



But I can tell you when it's wrong. And I can tell you when it's right. And I can get the most performance out of it.

I would much rather have somebody come in and say, "You know, the kick needs to sound less like a basketball." And I'll say, "I know exactly what you mean." It's using your ears. It's training your ears to go out in the room and not be afraid to get out there and listen to what things are sounding like to them.

RE/P: *What advice would you pass on to other engineers?*

ES: You've got to be at the right place at the right time with the goods, or at least with an attitude that implies you're capable of acquiring the goods. When I started, there were no schools. Now they're cranking out all of these kids and there's no place for them to go. You'd better be persistent and try to get in. I actually wish there had been a school, but the school of hard knocks was hard to beat because we got a chance to do so many crazy things.

For example, one of the weirdest examples of creative recording, or painting on an overdub, happened because we were searching for something. We had a very large studio, and for the guitar ride, we hooked up all of these mic booms so that it stretched about 30 feet end-to-end. It drooped, it was so long. We put an AKG C414 on each end, and we had 50 feet of cable, and we wound this configuration up. We had a guy in the middle with blankets over his head and ear muffs, and we had three Marshall cabinets in a triangle and two Marshall heads.

We tapped in on the speaker talkback, "OK, start winding, we're 10 seconds from the ride." We started this thing spinning, the guy played and they were zooming. It was like a Leslie speaker in reverse, but it was moving at the same time, so you got this weird Doppler effect that could *not* be achieved by panning. It was too cool.

Foster and Lloyd: Artists as Producers

It is rare for any new artist to produce his first album, but it is particularly rare in country music, where even established artists usually have a co-producer. In a break from tradition, several new country artists, who grew up in the post-British Invasion era (when self-production first began), have had great success producing themselves. Most notable among these is Foster and Lloyd.

Radney Foster and Bill Lloyd's role as album producers, and the way they interacted with engineer Ed Seay, is instructive for other engineers as they encounter more artists producing themselves. Like most artists, neither has a technical background, and they rely on engineers to achieve their production aims.

Originally, the duo were staff songwriting partners at MTM Music in Nashville, and RCA offered them the chance to become a recording team after other artists were successful recording their songs. After hearing demos recorded by engineer/producer Rick Will at MTM's 8-track studio, the label gave them the go-ahead to record the album—with themselves producing.

"Joe Gallanti [head of RCA in Nashville] said to us, 'If this demo is black and white, give me the same thing in Technicolor,' " says Foster.

Although Foster and Lloyd have a substantial relationship with Will, they didn't feel comfortable enough to go and complete a major record deal, so they called Seay. Aside from his technical expertise, Seay brought good communication skills into the project. Although he could have tried to dictate the final sounds, they say, he always gave them options, with the final decision up to them.

" 'Heroes or goats' was always one of his big phrases," Lloyd says. "We'd go through and make decisions and it was, 'OK, version A, is it a hero or a goat?' He gave us these options, like, 'This is a more traditional route,' or 'This is the route you guys are talking about, and what do you like?' Sometimes you'd blend two ideas together."

Second album

With the success of the first album, Foster and Lloyd resumed their collaboration with Rick Will for their second album, with one change—Will received a co-production credit. Preproduction again occurred at MTM's studio, which, in the interim, was upgraded to 16 tracks. Sixteenth Avenue Sound in Nashville was the site for tracking and mixing.

A graduate of Belmont College's recording program in Nashville, Will has worked with many of the city's up-and-coming rock bands and remixed the four singles from the Foster and Lloyd album. He also engineered several cuts on Giles Reeves' "Nothing Is Lost" album on the MCA Master Series.

"My goal is to give these guys a great platform to stand on," Will says. "We cut as aggressively as possible and then tailored the mixes to country or rock."

"Rick and I are such a team now that I don't have to say anything because he knows what kind of sounds I like," adds Lloyd. "He pretty much knows sonically where to put it all. It really helps to find a great engineer to get your ideas out there."

Working with engineers

Because they do not have technical backgrounds, Foster and Lloyd depend on the engineer to establish a solid technical foundation. But both stress communication is equally important.

"Communication skills are just as important as your technical skills," Foster says. "A good attitude will go a long, long way.

"I suggest that engineers listen to a lot of different kinds of records, too. You can suggest to a guy, 'I want this type of a sound,' and he may understand it technically, but you may be relating to it as a 'rockabilly slap,' like on a Dave Edmunds record, and he has no idea of what you're talking about."

Central to a project's success is the artist/producer being able to rely on the engineer, Lloyd says.

"When we're playing on our own record, there's got to be someone we trust on the board," he says. "They're in there deciding which of your takes are good, and of course, it's up to you to make the final decision, but you have to trust their opinion. That's invaluable."

Bill Lloyd, Rick Will and Radney Foster (L-R) during the mixing of Foster and Lloyd's second album at Sixteenth Avenue Sound in Nashville.

That's something that takes time to do, and if you don't have the time to mess with that, you'll never stumble across something like that. Unfortunately, that record never made the air.

RE/P: What do you have on tap?
ES: A new Dolly Parton album that Ricky Skaggs will produce. That should be fun. I've always admired Dolly's work. Never have worked with her, but we're going to start in a few weeks. There's a Ricky Skaggs album that I've already cut some tracks on. The Desert Rose Band looks like we'll start in February. Highway 101 looks like we'll start in January, and there's already a song search started for that.

We're also looking at a slight diversification here. Paul grew up playing rock, R&B and pop in bands in college, just like I did. Our next iron in the fire is to work on something in a different genre other than straight country, maybe a little more rock-and-roll. So we've got some feelers out there.

It becomes a juggling act. You have to be real careful on scheduling this stuff, because you get too many things going and you do a cruddy job on all of them. I'd rather not do that. I feel like you only get so many times at bat to put out product, to make a great record, before the Great Master Fader in the sky pulls you down.

RE/P: What's the main issue facing engineers today?
ES: Staying on top of technology, not letting it pass you by, but at the same time, not being ruled by technology. I mean, we're still dealing with music. Don't fall into a pattern where there's only one way to do anything, because there's never just one way.

And don't ever fall into a pattern of, "Well, every snare drum has to have +10 at 10k or it's not a snare drum, and when we'll mix it, we'll add +10 again." I don't know snare drums that sound like that. I've never heard any out in the room that sound like that. It's got to be music. It's more than frequencies and numbers and presets and patches.

When you're an engineer, you think in terms of the sonic integrity, but as you start to expand into production values, you start dealing with emotion and trying to get something that communicates. Frequencies don't communicate, but those magic elements of performance and production do. Ultimately, we are in the communication business. RE/P

STUDIO UPDATE

Northeast

Susquehanna Sound (Northumberland, PA) has completed the construction of its 5,000-square-foot video post-production suite and studio. The facility works in conjunction with the adjacent recording studio. *48 A St., Northumberland, PA 17857; 717-473-9733.*

Dreamland Recording (Bearsville, NY) has purchased two UREI LA3As; two Perreaux 6000-Bs for main monitor amps; two API 560-BT graphic EQs; a Drawmer stereo 1960 tube pre-amp limiter; and has upgraded to Monster Cable. *P.O. Box 383, Bearsville, NY 12409; 914-338-7151.*

Round Sound Recording (Cresco, PA) has added an Alesis HR-16 drum computer, an Alesis Midiverb II, a Nakamichi MR-2 cassette deck, an Opcode Timecode Machine and a Roland P-330 digital piano. *R.R. 2, Box 111-C, Cresco, PA 18326; 717-595-3149.*

Sunset Productions (New York) has installed a Studer A820 24-track recorder and a Sony R-DAT machine. *226 E. 54th St., New York, NY 10022; 212-832-8020.*

Ready Or Not Productions (New York) has changed its rate schedule to $100 per song-minute, which allows the studio to put out projects quickly without rushing the artists and producers. A no-time-limit clause also allows clients to experiment with new ideas without being penalized. Equipment includes Fostex recorders, IBM computers, and a variety of synthesizers, drum machines and outboard equipment. *250 W. 57th St., Suite 1527, New York, NY 10019; 212-642-8344.*

SounTec Studios (East Norwalk, CT) has added the FirstCom DigiEffects sound effects library. *25 Van Zant St., East Norwalk, CT 06855; 203-853-3433.*

Southeast

Mangum/Alford Recording Studio (Jacksonville, FL) has purchased a D&R 8000 Series III recording console, with a 32-input mainframe. *3524 Morton St., Jacksonville, FL 32217; 904-737-9242.*

Century III Teleproductions (Orlando, FL) has opened its post-production facility at the Production Service Center at Universal Studios/Florida. *2000 Universal Studios Plaza, Suite 100, Orlando, FL 32819-7606; 407-297-1000.*

Midwest

Audio Recording Unlimited (Chicago) is a new studio serving the advertising community. Located in two floors of Chicago's Wrigley Building, in the center of the advertising and media community, the facility is the first Chicago facility to have the Lexicon Opus digital audio system. Facility principals are Michael King, Don Arbuckle and Betty Rake. *Wrigley Building, 400 N. Michigan Ave., Suite 1900, Chicago, IL 60611; 312-527-7000.*

MIDI Productions (Skokie, IL) has installed two Lexicon LXP-1 effects processors with MRC remote control; a Sony DTC-1000ES R-DAT recorder; a TC Electronic TC 2290; a Lexicon PCM70; and three Yamaha DMP7 automated mixing boards, with Q-sheet and DMP7 Pro software running on an Apple Macintosh. *3418 Main St., Suite 208, Skokie, IL 60076; 312-677-3550.*

Southwest

Arlyn Studios (Austin, TX) has completed a $100,000-plus renovation. A 2,000-square-foot cutting room has been added, and a Studer A820 recorder has been purchased. *5605 Woodrow Ave., No. 7, Austin, TX 78756-1741; 512-467-2247.*

Southern California

Record Plant (Los Angeles) has added full ADR services to its film and TV scoring operation at the Paramount Pictures lot. Linda Corbin has been named to direct the service. *1032 N. Sycamore, Los Angeles, CA 90038; 213-653-0240.*

Track Record (North Hollywood) has expanded its facilities with a second 24-track studio. The main studio is 27'x24' with a 20-foot ceiling; the adjacent iso room is 15'x9'. Equipment includes a Neve 8232 console, Studer tape machines and a variety of outboard gear. *5102 Vineland Ave., North Hollywood, CA 91601; 818-761-0511.*

Interlok Studios (Hollywood) has expanded to include a second room for film and television audio work. Equipment includes a Soundcraft Series 3-B console and an extensive sampled custom effects library. Edie Nelson been named studio manager and marketing representative.

Waves Sound Recorders (Hollywood) has installed the New England Digital Post Pro system. Studio A is being remodeled; a Sony MXP-3000 console with disk-based automation is being installed. *1956 N. Cahuenga Blvd., Hollywood, CA 90068; 213-466-6141; fax 213-466-3751.*

Northern California

Music Annex (San Francisco) has introduced its Tapeless Studio, featuring an NED Post Pro Direct-to-Disk recording system. *69 Green St., Second Floor, San Francisco, CA 94111; 415-421-6622.*

One Pass (San Francisco) has named Jack Schaeffer as president. *One China Basin Building, San Francisco, CA 94107; 415-777-5777.*

Hawaii

Fortunate Sun Recording Studio (Honolulu) has purchased several new pieces of equipment, including an Otari MX-80 24-track recorder, a Studio Technologies mic pre-amp, a Roland D550 module, a UREI 1176N limiter and a Lexicon LXP-1 digital processors with MRC controller. *720 Iwilei Road, Suite 416, Honolulu, HI 96817; 808-531-5744.*

Manufacturer announcements

Studer has delivered a 62-input 900 series console with GML automation to Lighthouse Studio, North Hollywood. This is the first large Studer console to be installed in the United States.

Advanced Music Systems has received an order from Streeterville Studios, Chicago, for nine AudioFiles. This is largest single order that the company has received.

Solid State Logic is installing two SL 4000 G series and two SL 5000 M series consoles at the MOSFILM studios in Moscow.

Alpha Audio has installed the Boss/2 audio editing system at Limelite Studios in Miami.

TimeLine has made a number of recent sales and deliveries: Soundtrack, Keyboard Control Unit (KCU); Studio Marko, three KCUs; Disney, eight Video Systems Interface (VSI) modules; NBC, 16 VSI modules; Pegasus Studios, Time Code Modules (TCMs); Lion Share Recording Studios, KCU; The Church Studios and Hilton Sound, both of London, TCMs; and film modules and TCMs to BBC Pebble Mill Studios, Birmingham, England, and Windmill Lane Studios, Dublin, Ireland.

Digital Audio Research has sold a SoundStation II to TVi, a post-production facility.

Soundtracs has received orders for its PC24 console with MIDI automation from Air Studios, Swanyard Studios, Chrysalis Records, Eden Studios and Westside Studios, all of London; and Jean Michel Jarre Studio, Paris.

Soundcraft has installed consoles at Harriman Communications Center, Washington, DC, 200B; Industrial Audio Film Services, Morton Grove, IL, 200B; Kid Kirk Productions, Agawam, MA, TS-12; and Baldwin Wallace College, Berea, OH, 600.

Lexicon has installed an Opus digital audio production system at Sound Mirror, Boston.

Trident Audio has installed a DI-AN console at Genetics Studio, London.

Mitsubishi has installed consoles at Evolution, Toronto, 32-input Westar; West Oak Recorders, Westlake Village, CA, 52-input SuperStar; Sound Interchange, Toronto, 44-input SuperStar; Triad Studios, Seattle, Moving Fader automation for its 52-input Westar; Traxx Recording Studios, Oak Park, MI, 44-input Westar; SRS Studios, London, Ontario, 28-input Westar; Evolution Studios, Toronto, 32-input Westar; Servisound, New York, 36-input Westar; Sigma Sound Studios, Philadelphia, 60-input SuperStar; and the Bossa Nova Hotel, San Fernando, CA, 52-input SuperStar. On the tape machine front, the company has installed an X-850 and X-86 at Phase One Recording Studios, Toronto; X-86HSs to Conway Recording Studios (two machines), Design FX Audio, Bernie Grundman Mastering and Time Joran Rentals, all of Los Angeles; and an X-86C to Masterfonics, Nashville.

Neve has delivered the first Flying Faders console automation system to Rumbo Recorders, Canoga Park, CA, in a new V60 console. The company has also received console orders from Master Sound Astoria, Astoria, NY, V Series; Streeterville Studios, Chicago, a V60 and a V48, both with Necam 96; Sigma Sound Studios, New York, V Series with Necam 96; Baby'O Recorders, Hollywood, two V60s with Necam 96; AD Productions, Milwaukee, V-48 with Necam 96; and the Burbank Studios, Hollywood, V Series Custom Film Scoring console.

RE/P

THE CUTTING EDGE

By Laurel Cash

Now Available: DAT With Time Code

Fostex D-20 DAT with time code

It's happened. What we've all been wishing for ever since the DAT format was introduced: a DAT machine that will synchronize using SMPTE/EBU time code. It works. I've seen it. And it was a production model, *not* a prototype! It is the Fostex D-20 Digital Master Recorder.

Not only does the D-20 have time code, it has off-the-tape monitoring (also known as "audio confidence" or "read-after-write"), which is accomplished by Fostex's 4-head recording system. The company says you can punch in or out seamlessly, again because of the unique 4-head system. The punches are seamless because the unit employs built-in crossfade timing.

The D-20 works with time code just like any open-reel deck with center track capability. You can take a tape that's recorded on any other DAT machine and add SMPTE time code to it with the D-20, or you can record time code and stereo audio on the D-20 and play the stereo audio back on any other DAT machine. Plus, you don't lose the program/start IDs when time code is added to a previously recorded tape. Fostex says it records the time code every 15ms—in the subcode area.

Another important feature is that the D-20 reads time code at all functions and speeds, even pause (just like a 1-inch type-C video recorder running VITC).

In addition to SMPTE time code, absolute time can be used for locating. There are four memory locations and the relative zero point can be reset as needed.

Fostex has included its standard 20-pin synchronizer port on the back panel, so the D-20 can work with other synchronizers as well as the company's own models 4030 and 4035. The interface cable is the same as the one used on all other Fostex machines. In addition, an RS-422 serial port (DB-9 connector) is available for machine control requiring serial communication. Word sync is also included on both the input and output, as well as an external sync. The external sync feature allows the unit to lock to an outside reference, such as component video or house sync.

Other features include a ±10% pitch control with a complete digital display, so you can precisely reset the pitch change, if necessary. There is also a copy-protection switch on the front panel, which allows you to copy-protect your DAT recordings. (This only works in digital-to-digital copies.)

The Fostex D-20 records and reproduces at selectable sampling rates of 44.1kHz or 48kHz (with or without emphasis), and,

Laurel Cash is *RE/P*'s executive consultant and a free-lance writer based in Los Angeles.

like any other DAT machine, it has a maximum 2-hour record capability.

If this machine lives up to its promises, we may be witnessing a whole new generation in digital post-production. The suggested retail price for the Fostex D-20 is $8,000. The unit is currently available direct from Fostex.

Circle (80) on Rapid Facts Card

Nakamichi's 1000 Digital Audio Recording System

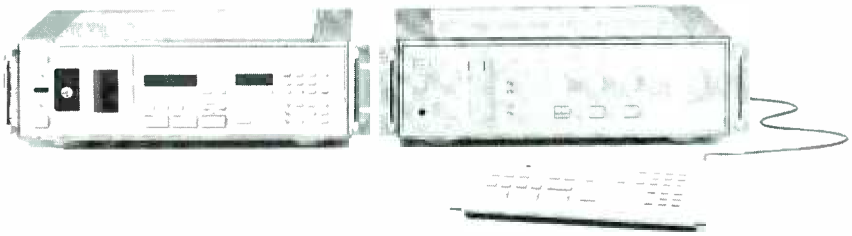
In an unprecedented move, Nakamichi of Tokyo has designated its new digital audio recording system as model 1000. The new DAT product bears the same nomenclature as the world's first 3-head analog cassette deck, which Nakamichi introduced to the studio market 15 years ago.

The system is comprised of two separate units: the Nakamichi 1000 Digital Audio Recorder and the model 1000P Digital Audio Processor. According to the manufacturer, the basic transport/recorder section was separated from the digital signal processing section to make the system easier to upgrade. This separation of transport/recorder and digital processor is certainly not a new technique; it has been used by other digital audio manufacturers in the past.

A notable difference here is what the folks at Nakamichi are calling FAST: *F*ast *A*ccess *S*tationary Tape Guide *T*ransport. It is not derived from existing VCR designs (this in itself is new) and therefore is described as a major departure from conventional DAT mechanisms. The primary advantage of totally stationary tape guides is said to be a much higher degree of tape-path travel precision, which results in lower digital data error rates.

A unique link-arm tape loading system is also credited for the FAST mechanism's smooth tape handling and rapid action. The typical interval between the time a tape is inserted and audio is heard is said to be 1.9 seconds. The 1000 is also said to shuttle tape at 400 times normal play speed.

One welcome feature is the ability to be upgraded and expanded through the use of modular, plug-in circuit boards. The company says, of course, that this product represents the current state-of-the-art in digital audio recording and signal processing technologies. It is interesting to note, however, Nakamichi's awareness that these technologies are advancing and changing rapidly, and that the company is prepared to deal with it in the short term, as well as the long term.



The Nakamichi 1000 Digital Audio Recording System.

As certain technological (and, perhaps, political) obstacles arise, we may even see changes to the current DAT format. Nakamichi claims that the model 1000 will not become obsolete because new plug-in boards will be made available as needed to keep this product at the cutting edge.

Nakamichi says the consumer version will allow direct digital-to-digital recording, but is limited to 48kHz and 32kHz sampling frequencies. The "pro" version is virtually identical to the consumer version except that it will come with 19-inch EIA rack-mount handles and will permit direct digital-to-digital recording from 44.1kHz sampling rate sources.

Although both consumer and professional versions provide balanced XLR-type analog inputs and outputs (in addition to the unbalanced RCA-type connections) the "pro" system will also conform to standard studio line levels, and has both optical and coaxial ins and outs for digital interfacing.

The Nakamichi 1000 is scheduled for delivery in March and has a suggested retail price of about $10,000.

Circle (81) on Rapid Facts Card

Subcode Time Code proposal from NHK

At the recent AES Convention in Los Angeles, NHK proposed a format for STC (Subcode Time Code) recording, which was designed in cooperation with Matsushita Electric Industries and Sony Corporation. The group proposes to standardize the way in which time code is stored, read and displayed on both professional and consumer DAT machines, through a variety of manufacturers worldwide.

When TASCAM, a division of TEAC Corporation of America, was approached, both marketing manager Bill Mohrhoff and product development manager Dave Orin claimed that they would welcome such standardization with open arms.

Obviously, the NHK group proposes that it be possible to record time code information on a DAT tape and play it back from the tape. It also proposes that in the case of asynchronous, non-locked TC frame frequency and DAT frequency, a display and output of the time code are also needed. The proposal suggests that the time code data be recorded in the subcode area (in the same location as the Fostex machine, which, although the Fostex is not part of this proposal, *is* already on the market), which allows for high-speed readout.

The format for recording the time code is what NHK calls the "Pack Format" (time information in Pack form) and can cover many applications, regardless of video-oriented time code standards. The "Pack Format" is the format standard proposed in a recent DAT conference document. It is further suggested that the Pack should be identified by an item number for clarification to the user. Briefly, one pair of subcode data blocks contains up to seven Packs and one DAT track pair. Each track pair carries 16 data blocks and contains up to 112 Packs every 30ms.

RE/P

NEW PRODUCTS

Tracmix automation from Soundtracs

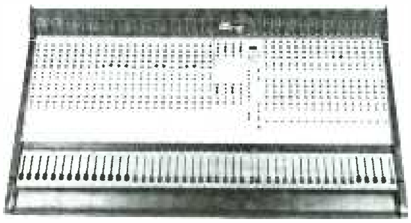
The Tracmix fader and mute automation system is operated via a remote keyboard and color monitor; no additional computer is required for the standalone system. Features include 64 channels of fader and mute automation; mute resolution to half-frame accuracy; fader position to 12-bit resolution; mute attenuation and THD at more than 100dB at 1kHz; and noise level more than −94dB.

Circle (160) on Rapid Facts Card

Peavey Production Series 1600 recording console

The modular console allows channels to be added and removed separately. Features include 60dB input gain; less than 1dB noise figure for 150Ω sources; more than 100dB of common mode rejection; rack-mounted power supply; electronically balanced input circuitry; eight, 16 or 24 submasters; and 4-band sweepable EQ on each input.

Circle (161) on Rapid Facts Card



Kenwood CD mastering and DAT testing systems

The integrated systems are designed to meet the audio industry's utilization of DAT technology in developing new hardware and software. Products for the CD mastering system include the DA-3500D encoder, DR-3552 decoder, DT-3520 digital I/O and DC-3510 analog/digital converter. The DAT testing system includes the DA-5730 encoder, DR-5750A decoder and DB-5740 jitter analyzer.

Circle (162) on Rapid Facts Card

Yamaha SPX1000 processor

The SPX1000 digital multi-effects processor is capable of selecting any two parameters for MIDI control, or control by an external rocker pedal or joystick. Its digital audio input and output system allows direct input and send/return for connection to other external devices. The incorporation of second-generation DSP II chip technology provides the SPX1000 with the capability of simultaneous processing, 40 preset programs with adjustable parameters, and 59 programmable memory locations. The processor offers 16-bit linear quantization and a sampling rate of 44.1kHz. Suggested retail is $1,795.

Circle (163) on Rapid Facts Card

Klark-Teknik signal processors

K-T has introduced three products. The DN500 dual compressor/expander/limiter is a 2-channel dynamic processor that can be used independently or linked for stereo operation. Each channel contains compressor, expander and limiter sections, all with a frequency response of 20Hz to 20kHz, ±0.5dB. The DN510 dual noise gate operates as either a dual mono system or a stereo unit, and features front-panel controls that access four parameters, including gate in/out with a 0dB to −90dB range. Distortion is less than 0.05%, 20Hz to 20kHz. The DN514 quad auto gate features percussive and normal automatic attack settings for fast setup and automatic attack time response. A sync function allows synchronization of parts by interlocking all four gate release times.

Circle (164) on Rapid Facts Card

Turbosound TSE-112 enclosure

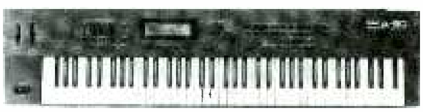
A V-2 high-frequency device provides the difference between this unit and the TSE-111. The unit is designed to be used when a greater degree of high-frequency absorption is detected and would result in an improved spectrum balance. It is switchable bi-amped active/passive, and includes optional flying hardware.

Circle (177) on Rapid Facts Card

Roland A-50 keyboard controller

The mother keyboard controller features a 76-note SK3 keyboard with weighted synthesizer-action, polyphonic aftertouch, four overlapping split zones and 128 memory locations. Four MIDI outputs are provided, as well as the two inputs that merge MIDI information for interfacing with sequencers or synthesizers. A RAM card is optional to supplement the internal RAM.

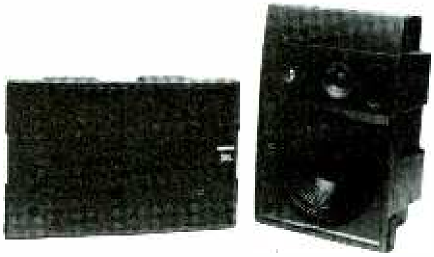
Circle (178) on Rapid Facts Card



JBL Control 10 monitor

The loudspeaker components in this system are magnetically shielded to permit use near television monitors and other electronically sensitive electronics. The Control 10 features a 12-inch low frequency transducer with JBL's SFG magnetic structure for low harmonic distortion; 5-inch cone midrange; and a 1-inch titanium dome tweeter that delivers a smooth performance beyond 20kHz.

Circle (181) on Rapid Facts Card



Studer A827 recorder

Based on the A820 transport, the analog recorder features 14-inch reel capacity; three tape speeds with an integrated varispeed controller; phase compensated MDAC-controlled amplifiers with switchable Dolby HX Pro; and an optional internal synchronizer. Parallel and serial RS-232/-422 control ports make interfacing possible.

Circle (182) on Rapid Facts Card

Yamaha FMC1 digital format converter

The FMC1 converts the digital outputs of the DMP7 and DEQ7, allowing the units to be used in existing digital recording and processing systems. The dual-channel stereo digital audio format converter provides an on-board master word clock, switchable to 44.1kHz or 48kHz, and transforms the Yamaha proprietary format to three common digital audio formats: unbalanced SDIF-2, Sony; CD/DAT (S/P); and AES/EBU. Suggested retail price is $595.

Circle (183) on Rapid Facts Card

Cibley music library

The Studio C Music Library from Cibley Music features seven volumes of music in all styles. A volume of musical sound effects is also available, include wa-was, ballpark organs and comedy drum hits. The library is available in a variety of formats.

Circle (110) on Rapid Facts Card

Intelix Graphic-DSP

The graphic digital signal process limits, compresses, expands, gates and peak-limits simultaneously in the digital domain. The unit graphically displays the input/output transfer curve with real-time overlay of input and output levels. Other features include a variable sampling rate, dynamic EQ, digital rate conversion, FIR filtering and spectral power density sensing. The system is software-based, allowing for future updates.

Circle (159) on Rapid Facts Card

Hybrid Arts hard drives

The Hardrive HDX series are 77 to 780Mbyte unformatted hard disk drives for Apple Macintosh and Atari ST computers. Custom-tailored for the company's ADAP I and II, each drive contains Hybrid Arts' proprietary controller and software, double-shock mounting and can be switched between 110V and 220V. Models are available in rack-mount or desktop configurations; formatting and partitioning software is included. Prices range from $1,395 to $6,995.

Circle (107) on Rapid Facts Card

PhantomAcoustics Shadow

The Shadow is an active system for the control of low frequency room resonances. By using active electro/acoustic servo systems to suppress room pressure zones, the unit performs more effectively in less physical space than passive systems. Two active suppression modules contain a mic, servo-amplifier and a transducer, each capable of reducing pressure energy by 90% at 50Hz at the module, for a total of 20dB suppression. Price is $1,790 a pair.

Circle (108) on Rapid Facts Card

HED Productions MIDI utility package

HED's MIDI utility package is a software program that will capture any MIDI datastream. Once captured, the data can be displayed, modified, loaded and saved to disk and retransmitted on the MIDI bus. Multiple internal buffers allow for merge, cut/paste, and insert/delete operations. It can be used for education, initializing/debugging MIDI systems or as a universal librarian for any MIDI-based unit. System requirements are an IBM PC/XT/AT or compatible with a Roland MPU-401 or compatible MIDI interface. Price is $59.95.

Circle (109) on Rapid Facts Card

New England Digital remote controller/editor/locator

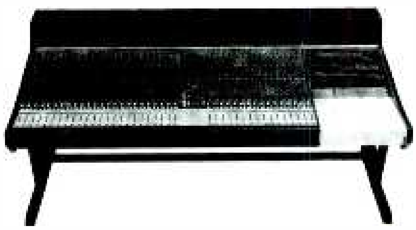
The remote box, when used with the Mac II workstation, provides a complete control center for digital audio production. Basic functions of the company's Direct-to-Disk digital multitrack systems, including motion control, track arming and soloing, auto-locator functions and scrub editing, are possible with the remote unit.

Circle (165) on Rapid Facts Card

Tascam M-700 40-channel mixer

The console offers an in-line configuration that provides two signal paths in each input-output module, and maximum input capacity in relation to console size. Other features include group output, quad mix buses, 4-band equalization assignable to channel or monitor signal paths, I/O module channel and monitor FLIP switches, and a hard-wire patch bay. Retail price is $70,000.

Circle (166) on Rapid Facts Card



UltraAnalog dual 20-bit audio DAC

The DAC D20200 includes two complete 20-bit D/A converters, a stable bipolar reference, a serial CMOS/TTL-compatible digital interface circuit and two distortion suppressing output deglitcher amplifiers. The unit provides a 108dB dynamic range; a THD and noise range from 20Hz to 20kHz; and output signal amplitudes from 0dB to −60dB. Price is $159, in quantities of 1,000.

Circle (167) on Rapid Facts Card

Turbosound TMW-210 floor monitor

The model TMW-210 incorporates two specially developed 10-inch drivers and a high-frequency unit for broadcast and film foldback applications. The model features a frequency response of 150Hz to 18kHz, ±3dB, and 18dB/octave, high-pass at 4kHz crossover.

Circle (168) on Rapid Facts Card

NEW PRODUCTS

SDS crossover design software

Scientific Design Software has released CACD, a computer-aided crossover design package. CACD is a circuit optimization program and circuit analyzer designed to develop passive and active crossover networks for loudspeaker systems. In addition to predicting the response of the network and driver combination, the program also predicts the input impedance for amplifier load verification. The program interfaces to a data library of more than 750 drivers, and a library of standard circuits allows schematics to be made easily. CACD is currently available for the IBM PC. A Macintosh version was scheduled to be available by the end of the year. Suggested retail price for the IBM version is $349.95.

Circle (114) on Rapid Facts Card

NED instrumental library

The Denny Jaeger Master Violin Library for the Synclavier offers more than two gigabytes of digitally recorded section violins. Included are legato sustains, staccato and marcato attacks, tremolos, whole-step trills, half-step trills and pizzicatos. Various tuning levels are supplied, giving users a wide variety of tonal colors. In total, the library contains more than 1,000 different attacks and 600 different sustain files.

Circle (115) on Rapid Facts Card

RTS Systems modular loudspeaker

The MSA325 is a full range loudspeaker that fits into a standard 19-inch rack. The 1U speaker is designed to be used with the RTS 810 master station or 410 monitor amplifier, but can be used in any monitoring application where rack space is limited. A 4"x8" hi-compliance transducer is mounted at an angle within the enclosure; maximum handling capability is 10W. List price is $140.

Circle (116) on Rapid Facts Card

Roland S-MRC software

For use with Roland MC series sequencers, the software greatly expands the system capabilities. The software provides eight tracks for recording MIDI information. Editing capabilities include data timing shift by track, channel or MIDI event. Functions such as erase, extract, transpose and change MIDI channel can be performed by track, MIDI channel, note range and MIDI event. The software sends and receives Song Position Pointer, enabling synchronization to external rhythm machines or SMPTE devices.

Circle (117) on Rapid Facts Card

Alacra Systems Lock-A-Bin

Lock-A-Bin is a closeable, locking plastic bin designed for storing, handling and transporting small parts. It can be stacked or be connected horizontally and vertically. The bin is molded in one piece and is designed to improve pickup time, minimize parts handling and provide maximum parts protection. The bin can also be used as a shipping container.

Circle (104) on Rapid Facts Card

Commodore hard disk controller

For the Amiga 2000, the A2090A hard disk controller enables the Amiga to access a variety of high-speed mass storage devices. It contains both ST-506 and SCSI interfaces and provides a buffered direct memory access with high-speed burst data transfer. Up to two ST-506 devices and up to seven SCSI devices can be connected simultaneously. Suggested retail price is $399.

Circle (119) on Rapid Facts Card

Oberheim Navigator

Part of the PerF/X series, the Navigator is designed to provide flexible mapping of MIDI messages. Three mapping modes are included. Controller Mapping enables a controller value to be modified and reassigned to different channels and controller numbers. Note Mapping enables note-on messages output as a different note. Patch Mapping allows multiple patch changes to be sent on different MIDI channels. Each mapping setup comprises a lower and upper map, with the lower used for controller or note mapping and the upper used for patch mapping. Suggested retail is $249.

Circle (120) on Rapid Facts Card

FSR sequential ac switcher

The SP-3R is a solid-state, sequential ac switcher that contains a fully regulated 24V, 2.5A power supply with switched and unswitched outputs. The unit will turn on the entire rack, and after a pre-set delay, will turn on the amplifiers. At system off, the process is reversed. A remote connector allows a remote control to be used. Using the system elimiates speaker pops and reduces possible damage to the speaker systems and amps from transients.

Circle (121) on Rapid Facts Card

Publications

E-mu training video

E-mu Systems has released a training video for the Emulator III, designed to supplement the reference and tutorial manuals. The video goes through each of the module function menus in detail and offers specific application ideas. A printed outline is included to allow users to locate and focus on areas of interest. The video's running time is 112 minutes, and is available in VHS and Beta. List price is $29.95.

Circle (123) on Rapid Facts Card

HAVE product catalog

Hudson Audio Video Enterprises has released its 1988 Fall/Winter Catalog. Called "Your Necessities Catalog," it contains audio and video products as well as sections devoted to audio and video cable, adapters, connectors, tape, equipment, production necessities and general supplies.

Circle (124) on Rapid Facts Card

E-V mic catalog

Electro-Voice has released a catalog featuring its line of broadcast/production microphones. Included are sections on mic selection and applications, as well as model information referenced by microphone type.

Circle (125) on Rapid Facts Card

Chromium shielding brochure

Chromium has published a brochure that discusses Compushield, a cost-effective electromagnetic shielding system that combines shielding effectiveness with environmental stability. The brochure is free.

Circle (146) on Rapid Facts Card

Rane DC 24 dynamic controller

The DC 24 is a compressor/limiter/expander/noise gate package that the company says represents a new approach to dynamic control. Configured as four separate audio tools in one, the unit features two compressors, two limiters, two expander/noise gates and a built-in 24dB/octave crossover. The crossover feature allows band-split processing of high and low frequencies to eliminate side effects.

Circle (100) on Rapid Facts Card

Acoustic Energy close field monitors

The new company has introduced three loudspeakers, all designed for close field monitoring. The drivers have metal diaphragms and the bass/mid drivers have a patented metal cone technology that enables high power handling with distortion levels of about 0.1%. Models available are the AE1 (two 5-inch LF plus 1-inch HF) at $1,500, the AE2 (two 5-inch LF plus 1-inch HF), at $2,500, and the AE4 (four 5-inch LF plus 1-inch HF), at $3,750.

Circle (103) on Rapid Facts Card

NEW PRODUCTS

Hardware and software updates

Commodore memory expansion for Amiga

The A2058 memory expansion card has been introduced for the Amiga. When fully configured, the 8Mbyte expansion card provides the maximum memory in a single expansion slot. The standard card comes configured with 2Mbytes of RAM, and users can add an additional 6Mbytes using 1megabit DRAM chips. Suggested retail price is $799.

Circle (131) on Rapid Facts Card

Roland MC-500 upgrade

Owners of the Roland MC-500 can have their units upgraded to a Mark II with the OM-500 upgrade kit. The upgrade gives the unit 100,000 notes of internal storeage when used with the S-MRC Mark II software. The S-MRC system also resides in memory at once, so new editing features and functions are instantly accessible. The upgrade requires the installation of a new motherboard, which must be performed by the service department at Roland corporate headquarters or by an authorized Roland service center.

Circle (132) on Rapid Facts Card

Hybrid Arts ADAP I revision

ADAP I V. 1.3 features several new features not available on earlier versions, including merging of cue lists, improved stereo handling, sample loading from CD-ROM, waveform display configurable to display in time units of samples and SMPTE triggering in Edit Page. Automatic zero-finding of SMPTE off-time has also been added.

Circle (133) on Rapid Facts Card

TAC Matchless console update

New features to the 24-bus in-line console include an input reverse switch, which coupled with the provision of both line and tape monitor inputs on the jackfield enhances the console's routing system. Two signal paths on each input module can be accessed from the jackfield and the tape return multiway. The console's electronic design has been refined, improving its performance specifications, and a 32-track version is available with an all-VU meter option.

Circle (126) on Rapid Facts Card

Fader automation for DDA console

The DDA DCM 232 in-line console is now available with a VCA fader automation system, which adds recording and replay of the fader and mute information to the existing offline editing and preparation of channel switch settings. Storing and recalling existing console settings is possible using the new automation or the on-board CAT automation. Three levels of operation are allowed: VCA-only, channel-switch only and a combination of the two, available on a per-channel basis. Price is $368.50 per channel.

Circle (127) on Rapid Facts Card

New modules for DDA D-Series console

DDA has introduced a new module set for its D-Series consoles. The VCA input modules and master modules offer direct application advantages for global audio control with the inclusion of VCA and mute groups. The modules offer separate trim for the mic and line inputs, in addition to offering a four-stage, quasi-parametric EQ. Also included are eight individual aux sends and eight individual group assigns.

Circle (128) on Rapid Facts Card

Sonus SuperScore update

SuperScore 1.3 is a more comprehensive, versatile and interactive scoring and sequencing package. Note entry methods have been expanded to allow music input using the mouse. Quantize features have been enhanced. QMagic is an intelligent quantize feature that allows users to selectively quantize rests. Editing enhancements include set/unset notes to staccato, legato or accented. Two different noteheads are also available: "x" for non-pitched noteheads and diamond for string harmonics. Users who have sent in a warranty card will be sent the update; users who have not sent the card in should contact the company.

Circle (129) on Rapid Facts Card

E-mu EIII software update

Software V.2.0 for the Emulator III includes four new features. SCSI compatibility with the Macintosh allows data to be exchanged from the EIII and the computer using third-party software and peripherals. MIDI Load Bank allows EIII sound banks to be loaded into RAM memory by MIDI program change commands. MIDI Sample Dump allows sample data to be digitally transferred from other MIDI devices. Quick Zone allows efficient manipulation of keyboard parameters and can quickly define alternate tunings. The software update is free to registered owners.

Circle (130) on Rapid Facts Card

Tascam MIDiiZER

The MIDiiZER functions as an auto locator for transport, a MIDI synchronizer that syncs MIDI machines to transports and a transport synchronizer that chases two transports. Capabilities include 1/100 sub-frame offset, memory-by-memory card with MIDI bulk dump information and information input by LCD, 10-key, rotary dial or cursor key. Functions include tempo mapping, SMPTE-based locating, auto punch-in/-out, 20 points cue, pre-post roll, end-limit, locate play, insert, copy, delete and self-learning setup function. Suggested retail is less than $2,000.

Circle (102) on Rapid Facts Card

ART high-definition equalizers

The EQs feature a newly developed high-performance circuitry. The units feature faders with 60mm travel, switchable subsonic and ultrasonic filters, clip/signal metering, fail-safe hardwire bypass in the event of a power loss, and balanced XLR, TRS and terminal block connectors. Two versions are available. A one-third octave version features 31 bands from 20Hz to 20kHz, switchable scales, 7.5dB and 15dB, and a transformer balancing option. The two-thirds octave version features 15 bands per channel with independent level controls.

Circle (101) on Rapid Facts Card

Clear-Com monitor speakers

The 1020 and 1020M amplified monitor speakers are self-contained two-channel audio monitoring systems that occupy a single rack space. Features include 100Hz to 12kHz frequency response, XL-3 type balanced line-level inputs and RCA unbalanced inputs and an optional LED bar-type input level meters. The units are biamplified devices that combine both channels' low frequency information into a signal amplifier and a specially baffled speaker to provide extended bass response. Prices are $525 for the 1020 and $595 for the 1020M.

Circle (118) on Rapid Facts Card

Yamaha EMX mixer series

The three series models of stereo powered mixers feature a built-in digital signal processor, dual graphic equalizers and built-in high-power stereo power amplifiers. Each includes two auxiliary sends per channel, 3-band ±15dB equalization, balanced XLR and unbalanced phone jack inputs, a 20dB pad followed by a variable gain control and an LED peak overload indicator on each input channel. The EMX2150 offers six inputs and stereo 150W at 4Ω power amplifiers. The EMX2200 and EMX2300 include eight and 12 inputs, respectively, and dual 250W at 4Ω. Suggested retail prices are $1,795 for the EMX2150, $1,995 for the EMX2200 and $2,195 for the EMX2300.

Circle (184) on Rapid Facts Card

Roland U-110 RS-PCM sound module

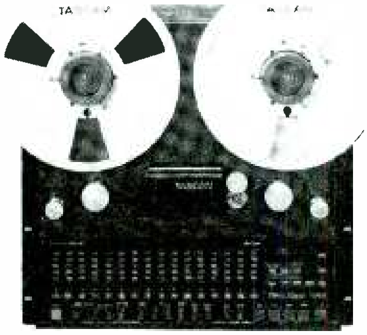
The U-110 provides access to an extensive sampled sound library and features multi-timbral capabilities, 31-voice polyphony, individual and stereo/mono outputs, and built-in digital effects. The 16Mbyte ROM stores 99 sounds in addition to the 36 percussion sounds in a separate memory location. Seven optional ROM cards are available for the four ROM card slots on the module, and more will be available in the future.

Circle (179) on Rapid Facts Card

Apogee 3X3 loudspeaker

The 3X3 tri-amplified concert loudspeaker is comprised of dual 15-inch high-power cone drivers coupled to two separate low-frequency horns combined at the horn mouths; a 2-inch throat titanium diaphragm compression driver, coupled to an advanced, constant directivity 60×40° midrange horn; and an array of four 1-inch throat compression driver/horn assemblies for high-frequency reproduction. The loudspeaker is designed to be used with the A-3X3 dual-channel signal processor.

Circle (192) on Rapid Facts Card



Tascam MSR-16 recorder/reproducer

The ½-inch 16-track tape deck features built-in dbx Type I noise reduction. A serial port for external control via computer or Tascam MIDiiZER and a parallel port for control via a synchronizer are provided. The model offers two speed settings, dump and manual edit, spot erase and a sync lock switch. Suggested retail is $7,500.

Circle (170) on Rapid Facts Card

DDA Q Series Mute modules

The Q Series Mute group includes input and master modules that feature eight select switches with LED indicators. Ten-segment LED level indicators are included. The Master Mute module provides for eight master Mute switches.

Circle (171) on Rapid Facts Card

Soundcraft SAC200 console

Designed for on-air broadcast and audio-visual production applications, the console features mono and stereo inputs that offer control logic for cart start and other remote functions. A Telco module is available, allowing any number of modules to be interfaced with any combination of mono or stereo modules in any position. Other features include main stereo mix output, mono mix output, control room and studio monitoring facilities, and frame sizes for 8-, 16- and 24-input module formats. A remote cue is provided for cuing stereo sources.

Circle (169) on Rapid Facts Card

NEW PRODUCTS

Tascam M-06ST signal mixer

The compact 6-in mixer accommodates microphones and electronic instruments and includes six stereo input channels, two stereo phone and four line-level inputs. Each channel features input selectors and 2-band equalizers (100Hz to 10kHz). Dual RCA-jack stereo outputs and one pair of phone jack stereo outputs are provided. Suggested retail is $500.

Circle (172) on Rapid Facts Card

Martin Audio VRS1000 system

This full-range horn-loaded system incorporates a method of bass loading, or delay-line porting, to provide solid extended pass. The system features an 18-inch Martin bass driver, 2-inch compression drivers and a 1.4-inch exit compression driver and horn. If interfaced to the Martin MX4 system controller and dedicated VRS1000 plug-in card, the system can be configured for 3- or 4-way operation.

Circle (190) on Rapid Facts Card

Denon DN-060 adaptive PCM encoder/decoder

The DN-060 for compact disc interactive (CD-I) mastering and authoring is available in two configurations: Type A for CD-I authoring and Type B for extended play. Type A processes two channels of audio at a time for Level A, B or C audio quality. Four-channel real-time processing for Level A and B is standard, and a plug-in board for Level C is optional.

Circle (191) on Rapid Facts Card

Yamaha MR mixing consoles

The consoles are designed for recording studios, sound productions and small sound reinforcement applications. Each of the MR series' three consoles have four mixing buses, a stereo master bus, and complete monitoring assign and talkback facilities. The MR842 has eight inputs, the MR1242 12 inputs, and the MR1642 16 inputs. Each input features a choice of electronically balanced, low-impedance XLR-type inputs or balanced high-impedance TRS phone jack inputs. Channels offer 3-band ±15dB equalization and sweepable peaking mid-frequency control. Suggested retail prices are $1,295 for the MR842, $1,595 for the MR1242 and $1,895 for the MR1642.

Circle (185) on Rapid Facts Card

Peavey DECA 528 power amp

The 1U amp delivers 210W into 8Ω and 250W into 4Ω, and gives maximum efficiency performance for applications where one single 8Ω or 4Ω speaker enclosure is used. Other features include 90% power transfer efficiency, DDT compression circuitry with LEDs to indicate DDT activity or a line fault condition, compression defeat circuits and fast output protection design. Suggested list price is $749.99.

Circle (112) on Rapid Facts Card

Fluke 9132 memory interface pod

For use with the 9100A digital test system, the 9132 is a software-configurable pod that allows the 9100A to test 80386- and 68020-based boards. The pod uses the HyperTEST RAM test algorithm, allowing it to test 1Mbyte of the unit under test's RAM in as little as one second. Included with the pod is a support floppy, providing advanced diagnostic capability for isolating bus faults, and a sync module adapter to help speed up the troubleshooting process.

Circle (113) on Rapid Facts Card

Opcode software for SPX-90

Opcode's Macintosh software package is an editor/librarian for the Yamaha SPX-90 and -90II. Using the software makes changing a parameter simple. More than one patch can be edited at a time, and multiple parameters can be copied and pasted between different effects and reverb patches. The librarian features the "Bundle," which allows users to combine multiple librarians into a single applications, simplifying complex setups. Retail price is $150.

Circle (111) on Rapid Facts Card

Yamaha PM2800-32/40 console

The console offers extensive signal-routing and control flexibility for on-stage monitoring applications. Available in 32- and 40-channel configurations, the console features eight mix buses and a stereo master bus; eight master mute groups; four matrix mixes with level controls; 4-band sweep equalizers with 15dB of boost/cut; 16 VU meters; and electronically balanced XLR-type connectors.

Circle (186) on Rapid Facts Card

Digidesign Desktop Audio Production System

The three-part system consists of the Sound Accelerator digital signal processing card, the AD IN A/D converter and the Sound Designer II audio editing software. The system runs on Apple Macintosh models II and SE, and offers hard disk recording and playback of 16-bit sounds. The system features two channels of stereo hard disk recording; visually enhanced waveform scrubbing; flexible buffer allocation; 14-band programmable graphic equalization; and variable time compression/expansion. System list price is $3,285.

Circle (187) on Rapid Facts Card

DDA D-Series VCA modules

The VCA input modules and master modules offer direct application advantages for global audio control with the inclusion of VCA and Mute groups. The modules provide greater program audio manipulation and include eight aux sends and eight group assigns. VCA master defeat switches, two Mute group masters per module and an aux return section are included.

Circle (194) on Rapid Facts Card

Denon America anechoic CD

The disc contains performances from every era of European art music, which were recorded in a temporary anechoic box that conformed to ISO 3745 for semi-anechoic rooms. The disc is designed for research, education and for testing audio systems. In addition to musical selections, the disc contains a comparison of different sound-gathering methods and test signals for room measurement, including single-frequency sine wave, pulses for FFT testing, wideband pink noise and white noise, and 1/3-octave band noise. Cost is $50.

Circle (122) on Rapid Facts Card

DDA fader automation system

The VCA fader automation system is now available for DDA's DCM 232 in-line console. The system adds recording and replay of the fader and mute information to the existing settings. The automation package allows for operation in VCA only, channel switch only, or the combination of the two functions. Price is $358.50 per channel.

Circle (176) on Rapid Facts Card

Beyer M 58 microphone

The M 58 omnidirectional dynamic microphone was designed for ENG and EFP applications. The microphone incorporates an internal shock mount system that reduces handling noise. To enhance vocal and speech clarity, the microphone's extended response is contoured with upper frequency rise.

Circle (193) on Rapid Facts Card

Studer PR99 MKIII recorder

The PR99 MKIII features a true autolocator; built-in varispeed and fader start; VU meters with peak LEDs; balanced and floating line-in and line-out; a digital counter and Zero-Locator; an edit switch and a tape dump. The recorder is available in 3.5/7.5ips and 7.5/15ips configurations.

Circle (188) on Rapid Facts Card

Apogee Sound hanging hardware

Apogee's expanding hanging hardware is available for its line of microprocessor-controlled loudspeaker systems. Accessories to make hanging easier and safer are included. All eight Apogee speaker enclosures can use the hardware.

Circle (180) on Rapid Facts Card

Tascam CD-701/RC-701 system

This CD player/control unit system features an autocue function, event play, link play and an optional BU-1 RAM buffer. The CD-701 features a clamping system that uses a rigid-free disc clamping device for precision; oversampling digital filters; 16-bit D/A converters; internally switchable monitor-mode line output; 98dB S/N; and a frequency response of 20Hz to 20kHz, ±0.5dB. The RC-701 control unit is capable of controlling four CD players and features ±6% pitch control with frame-accurate searching. Suggested retail price is $1,249.

Circle (173) on Rapid Facts Card

Roland R-8 drum machine

The R-8 Human Rhythm Composer incorporates artificial intelligence for the imitation of nuances and variations of actual percussive performances. The machine features 16-bit sounds and a 44.1kHz sampling rate; a copy voice function; two ROM/RAM card slots; and memory for 120 sounds and 2,600 notes.

Circle (174) on Rapid Facts Card

Turbosound TXD-580 enclosure

The TXD-580 is a high-power, bi-amped, 3-way wide-dispersion enclosure that will deliver high SPL levels over a wide area with outstanding clarity. A passive crossover on a retrofit backplate is available as an option, as is load-certified flying and mounting hardware.

Circle (175) on Rapid Facts Card

HME wireless sytems

HME has added the System 515 and System 525 to its line of wireless systems. The 515 is a body-pac system, featuring a mic mute switch, mic gain adjust, low battery LED indicator and reversible belt clip. The 525 is a hand-held unit that features locking mic mute and power on switch, an integral antenna, low battery LED indicator and rugged ABS housing. Both feature NRX 11 noise reduction circuitry, ac or dc power and are compatible with other HME companderized systems.

Circle (106) on Rapid Facts Card

Yamaha SPX50D digital multi-effect processor

The SPX50D provides 50 factory preset programs and 50 user-programmable memory locations. The 1U-high digital multi-effect signal processor features 16-bit linear quantization; a sampling frequency of 31.25kHz; a ¼-inch jack for feeding an unbalanced input; footswitch jacks for bypass and memory toggle; two unbalanced-output ¼-inch phone jacks. Suggested retail price is $695. RE/P

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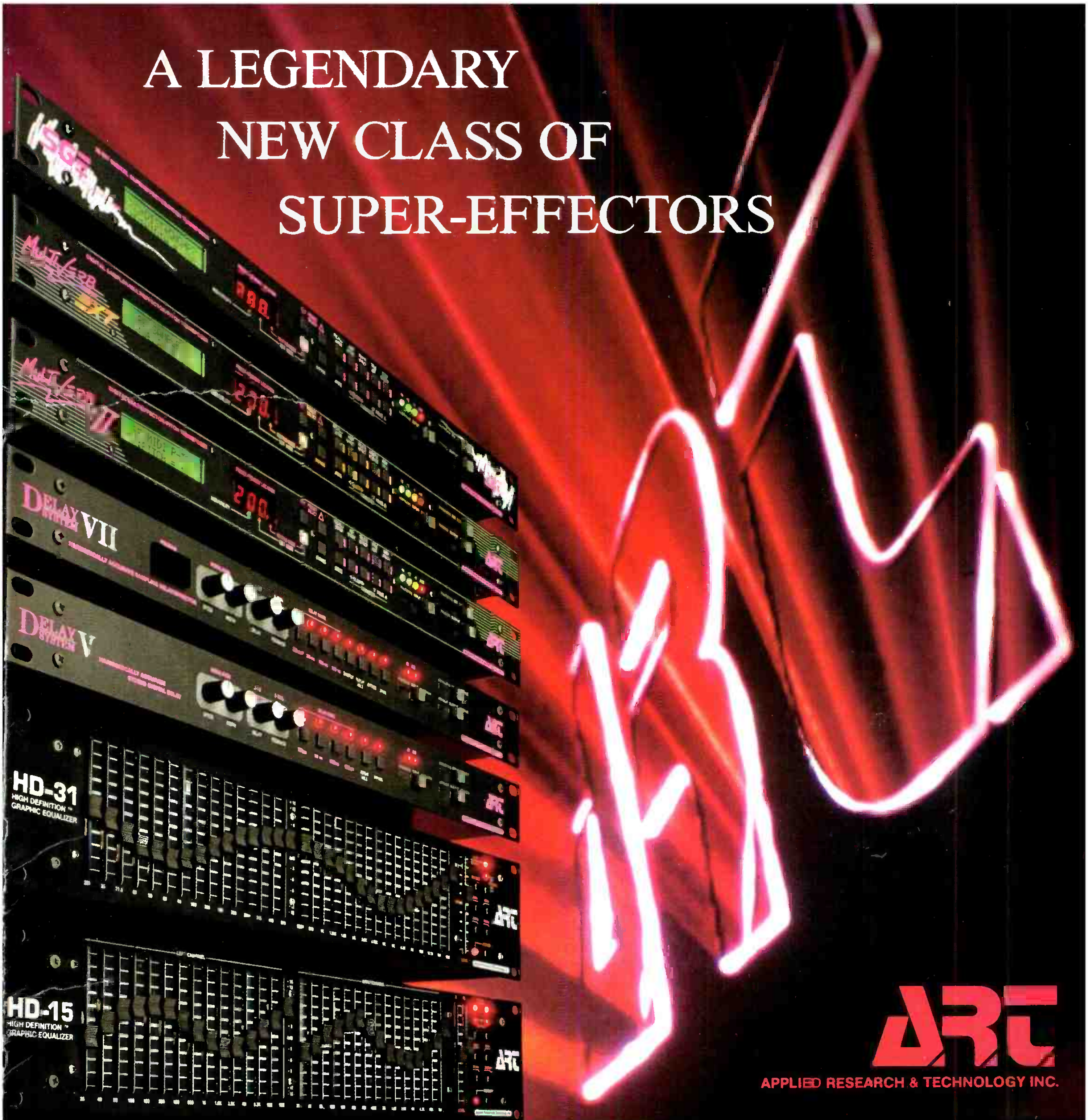

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When you record with the TS12, its in-line monitoring means you create a control room mix on the main faders.

In mixdown this becomes the basis for your master mix. Saving time and helping you keep a train of thought as to how the final mix will sound.

Unlike other in-line desks, though, the TS12 also allows true stereo subgrouping on long-throw group faders.

In this mode the routing matrix offers either six stereo groups or four extra auxiliary sends – totalling ten sends – plus four stereo groups.

No other console in the world provides such versatility.

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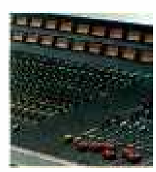

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