The Audio/Video World of

Thomas Dolby

A Guide to MIDI
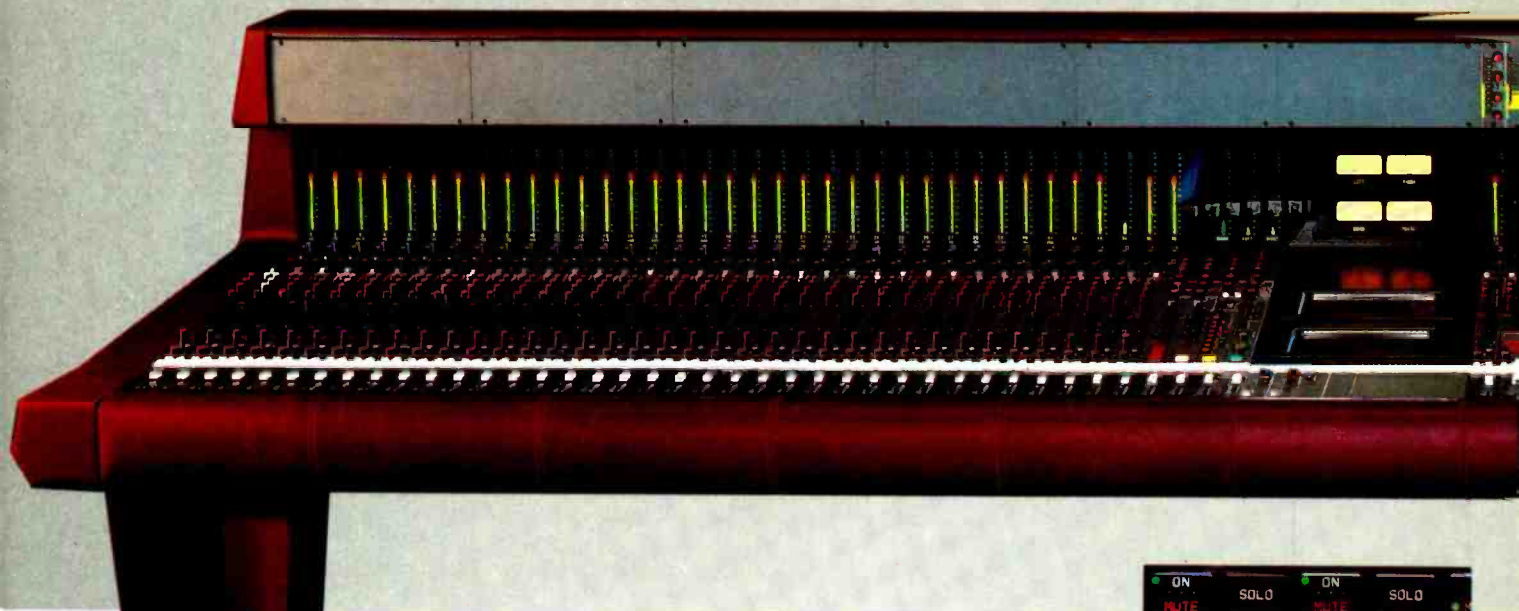
SUPERSTAR

by quad eight

Advanced technology and unparalleled flexibility come together in the SUPERSTAR music recording console. Development of this console centered around the dual requirement of truly high definition sound and low noise, so critical for digital recording.

No other single console offers the combination of superior sound and flexibility in size and layout at such an affordable price. Field expandable, the SUPERSTAR provides ergonomical positioning of the console modules, allowing you to satisfy your own configuration needs. High resolution meters, central bus assignment, Intelligent Digital Faders, and the most comprehensive automation system all add up to SUPERSTAR–your next console.

MODULAR CONSOLE

The SUPERSTAR is a totally modular console using air frame design concepts for strength and rigidity. Individual frame sections are in groups of 8 modules, with plug in wiring for true field expandability. The modular overbridge accepts the new limiter/compressor/gate for use either in-line with the input module or as a peripheral.

60-segment LED bargraph meters use advanced circuitry for precise and stable indication, offering VU, Peak, VCA level, and Spectrum Analyzer displays switch selectable.

Plug-in interchangeable equalizers and preamplifiers in each I/O module give instant user selectability and allow the addition of new technology at any time. Each module is of dual-purpose in-line design with line trim, equalizer, filter, 8 echo/cue sends, and fader switchable into the monitor/mixdown or main channel. Monitor/mixdown can be assigned to two independent stereo output busses for added versatility.

CENTRAL ASSIGNMENT

This electronic output assignment cross-point switching system assures fast and reliable connections from the console to your tape machines with full routing

or mixing capability. 64 output busses are assigned from each input module by a central touch control plasma display panel controlling up to a 96 by 64 electronic switching matrix. Completely software driven, the panel allows instant selection and display of the bus assignment with 10 presets in local memory. Optional unlimited storage to disk is provided. Easy to use, the system prompts for bus assignments and provides help through informative menu displays.

The building block matrix system consists of 16 by 16 switching cards bussed to 16-output summing cards. Logic controlled monolithic switching elements use zero volt current switching for extremely low distortion and feed through.

THE NEXT GENERATION

The introduction of the SUPERSTAR signals a new era in professional sound control. With more and more studio facilities acquiring digital multitrack recording capabilities up to 64-track, larger sophisticated console systems with transparent sound performance are necessary. Digital signal processing (DSP) is neither economically feasible nor technologically advantageous today. A new generation analog console with advanced digital control is required to bridge the gap between the DSP consoles of the 1990s and the currently marketed analog consoles of the 1970s. The SUPERSTAR is such a console system. See it before you decide.

Contents

Recording
ENGINEER/PRODUCER

June 1986   Volume 17   Number 3

Page 26   Page 58   Page 82

Coming in August

The primary spotlight for the August RE/P will be recent advances in digital recording technology, including:
• an analysis of DASH and PD encoding formats;
• details of a live, direct-to-digital session with Joe Jackson;
• digital systems available for mastering stereo audio for VCR duplication;
• The potential of digital audio post-production; and
• A hands-on assessment of a digital recording and random-access editing system.

Editorial

Welcome to the new-look **Recording Engineer/Producer!**

As will be readily apparent, this issue sees the unveiling of a redesigned **RE/P**, with new typestyles and graphics throughout the magazine. It also represents the first issue to be produced from our Kansas City offices.

Although **Rob Tuffly** and I still operate out of Intertec's West Coast editorial offices in Hollywood, all magazine layout, design and production is now being coordinated by art director Kevin Callahan and managing editor Dan Torchia. I am sure that you will join with me in congratulating them on a fine piece of work with this new issue.

For several months we have been planning the various improvements in visuals that you now see before you. I hope that you find the new-look **RE/P** to be easier to read, and that the graphics enhance the high standard of editorial objectivity that we aim to provide in each issue.

The major theme of this issue is electronic music production. We have several articles supporting this increasingly important sector of the recording market. First is an interview with Thomas Dolby, who recently completed the underscore and soundtrack for the new Lucasfilm movie, *Howard the Duck.* We also have a facility spotlight of Image Recording and Composer's Services, a cooperative venture between a studio owner and musician/composer, which is intended to provide complementary preproduction and recording facilities for electronic music clients.

And, to bring us up to speed on the intricacies of MIDI, we have a basic overview of the ways in which the musical instrument digital interface can be used in the studio. Our review this month is of the new Fairlight CMI Series III digital synthesizer.

This issue also sees the start of a new series of regular columns that are intended to cover important topics in the recording, live sound and production industries. Musician and producer Paul Lehrman will be covering current developments in the expanding world of MIDI in "Managing MIDI," while session musician, producer and "futurist" Stephen St. Croix will be keeping us abreast of changing developments throughout the pro audio industry in his column, "Living with Technology." Larry Blake, RE/P's film sound consultant and a regular contributor, will be covering the changing developments in audio for the film industry via his aptly titled column, "Film Sound Today."

I think you will agree that our unique combination of authoritative editorial and enhanced graphics will maintain the status **Recording Engineer/Producer** has earned as the number one operational magazine for the pro audio industry.

Mel Lambert
Editor

Letters

Tissue Paper Phenomenon

From: Ed Evans, New York City

The following is a reply to the recent exchange of letters regarding the use of a tissue-paper layer over the tweeters of Yamaha NS-10M close-field monitors, published in the April issue of **RE/P.**

As a response to the original article by Bob Hodas, and also to his response to Bob Clearmountain's letter, I also had met the mentioned engineer with the "string of hits," and would like to introduce my own thoughts.

As with most anything in the realm of audio reproduction, monitoring is subjective. It matters very little whether the *absolute* electro-acoustic reaction of a monitor is one way or another, only whether its reaction is predictable with relation to the finished product. There is no right or wrong way to use a tool if the end product is predictable.

The argument for total accuracy is moot. There are so many variables in the chain—not the least of which is the engineer's interpretation of the sound emanating from the speaker—that the total effect can only be examined in the context of operation and result. The original concept of the tweeter tissue was to correct for an exaggerated high-end in a consumer speaker, *not* a lab-standard studio monitor. This speaker was close to, but not exactly, the response necessary for *one* engineer to achieve a closer representation of what his record would sound like on home speakers, with their typically exaggerated top- and bottom-end.

The speaker's top-end hype was too exaggerated for the long-haul (listening to the same track for many hours), and a reduction of high-end was in order. If there was a high-end control, it probably would have been used. But even use of that control could come under scrutiny.

What happens to the phase angle when a resistive device is placed in the circuit with the reactive speaker? Non-linearities would occur. What would happen if the speakers were moved six inches, and local-area reflections changed the comb characteristics?

Well, this is the real world; there wasn't a control, and the engineer found this speaker closer to his liking than any other. And, I emphasize, *his* liking. All this talk of absolutes and measurements has to be reviewed with all other factors combined. Many of us have the necessary equipment to make laboratory measurements, but I still believe that the ears are more accurate, and to every conclusion there is a hypothesis—meaning that anything can be read into data to prove a point. Speakers, microphones, digital recorders . . . they all can have identical specs, yet the ears tell us differently.

And that's the bottom line: What the ears tell us, and the brain decodes, is our reality. If it translates to someone else's reality—i.e. they decode it the same way—that engineer will have a string of hits, because he produces an end product that's pleasing to the general public.

But the article also addresses not only the technical ramifications, but also the problem of its widespread use throughout the industry. The original tissue-paper concept was to adjust for one engineer, in one studio, on one set of speakers. If that concept was removed from the environment that spawned it, then let the borrower beware.

It is a tool and a technique. Once one understands the technique's concept, and it is taken outside of that context, I cannot agree with the chastisement of the method because of its adaptation by those with a copy-cat mentality, who may or may not understand what it is or what its ramifications could be. If you get a very definite and singular answer without qualification, that engineer has probably locked himself into a product and not a technique.

Technique is everything in this industry. A drum will sound different on different days, with different drummers, on different songs—there is no mechanical answer. Once again, whatever works, works. It may not work for the other guy but if it works for you. . . .

I appreciate the reams of data that it took to come to the article's conclusion but, in the real world, recording/mixing engineers will use any available or unavailable technique that is necessary—whether it be tissue paper, distorted tube amplifiers, overdriving tape, or whatever—to achieve a desired end product. Their ears tell them when it's right, and the gold record on the wall tells them their opinions were right.

PS: I also happen to know that when the "sophisticated" film and video clients want sound they know will sell, they rely on the likes of "Mr. Tissue Paper" to achieve it for them. Sound had been taken less seriously due to the fact that, up until recently, the end product was strictly a visual one. The public now regards sound as equally important, and that will dictate to the industries where the time and money is spent.

Bob Hodas replies:

I will agree with Ed Evans that technique certainly is very important in this business, and hope that recordings never lose the effects of personality. If you can get a certain type of distortion by using salt in the faders, and the musicians and listeners agree this is a "cool sound," then that is your creative influence and perogative as an engineer.

What's important here is the fact that when one person's technique is taken as gospel, without understanding, we can create more problems than solutions. As Ed himself stated in his letter: "Let the borrower beware."

My article was written as an educational piece to show the users what they are dealing with. The use of a tissue-paper layer to reduce HF output produces an unstable position- and material-dependent frequency response—a result of comb filtering. The odds of **RE/P** readers ever duplicating "Mr. Tissue "Paper's" listening situation are low.

If users want to continue with this method once they have read my article, that's their free choice. At least they now have the information to evaluate how they are affecting their monitors. **R·E/P**

Studer and Philips form joint venture

Willi Studer AG and N.V. Philips Gloeilampenfabrieken have announced their intention to form a 50/50 venture for the research and the development of Compact Disc-related professional studio systems.

According to a joint statement, the venture is intended to "exploit the synergism attained by cooperation in research and development resources and know-how in product engineering."

The venture will not affect ongoing independent developments by either company in magnetic tape recording and optical disc mastering systems.

According to Tom Minter, vice president and general manager of Studer Revox America, the agreement will address two separate but complementary activities: the development of CD-related products for the pro audio industry by the new company and the U.S. marketing of Philips Compact Disc products (including the LHH-2000 professional player and LH-0425 PQ Subcode editor) by Studer Revox America.

AKG establishes digital products division

As of April 1986, AKG acquired all assets and trademarks of Ursa Major, formally establishing a digital products division within the company. In addition to the extensive R&D activities undertaken by the parent company, the new Boston-based facility will become AKG's second R&D center for digital product development.

AKG's Stamford, CT, facility will handle all U.S. sales and the new division's marketing, export and administration.

Christopher Moore, president of Ursa Major and newly appointed executive vice president of the digital products division, will head all future projects.

Although the division will continue to sell off a small residual Ursa Major inventory, manufacturer of the Space Station, 8X32, StarGate 323 and 626 has been discontinued. The MSP-126 will re-enter the market as an AKG product, while the new ADR-68K Aurora will be released now as the AKG Acoustics ADR-68K.

Increase of digital audio applications in video post-production predicted

The increasing emphasis on audio in broadcast television will result in a 400% increase in the use of digital audio in video post-production over the next three years, according to Bradley J. Naples, president of New England Digital.

In explaining that rise, Naples cited recent developments, such as stereo television and the expected arrival of TV receivers with digital audio channels.

"We are at the beginning of a sound revolution in audio-for-video post-production that will be as dramatic as the music revolution of the Sixties," he said. "Sales of consumer Compact Disc players rose more than 300% last year. Consumers want better sound—and they want better sound with their video. That has been demonstrated with the popularity of MTV, stereo TV and such shows as *Miami Vice.*"

NED manufactures the Synclavier, which functions as a "tapeless recording studio" for the composition, synthesis, recording and editing of music, sound effects, ADR and Foley effects.

Klark-Teknik purchases console manufacturer

Klark-Teknik has entered the mixing console business with its recent purchase of Dearden Davies Associates Ltd., an audio console manufacturer based in Ilseworth, England. DDA will become a wholly owned subsidiary of Klark-Teknik.

According to Philip Clarke, K-T chairman, the acquisition "increases Klark-Teknik's product range, and brings us into the mixing console industry, where we can use our technical expertise and distribution network to very good effect." Apart from minor cosmetic touches, there are no changes planned for DDA console products.

3M honors academy nominees

The five audio teams nominated for Academy Awards in the achievement in sound category were recently presented with Lyra Awards from 3M. The fifth annual sound awards were made for work on *Out of Africa*—the Academy Award-winner in the category—*Back to the Future, A Chorus Line, Silverado* and *Ladyhawke.*

For the first time, 3M also recognized the original music-scoring mixers for each of the films. According to the company, this action was taken to further emphasize the important contributions of these less visible members of the motion-picture production crew.

The winners were: for *Out of Africa*, Chris Jenkins (supervisor, re-recording mixers), Larry Stensvold (SFX re-recording mixer), Gary Alexander (music re-recording mixer) and Peter Handford (production sound mixer).

For *Back to the Future*, Bill Varney (supervisor, re-recording mixers), Bob Thirlwell (SFX re-recording mixer), B. Tennyson Sebastian II (music re-recording mixer) and William B. Kaplan (production mixer).

For *A Chorus Line*, Donald O. Mitchell (supervisor, sound mixer), Michael Minkler (music re-recording mixer), Gerry Humphreys (re-recording mixer) and Chris Newman (production sound mixer).

For *Silverado*, Donald O. Mitchell (supervisor, re-recording mixers), Kevin O'Connell (SFX re-recording mixer), Rick Kline (music re-recording mixer) and David Ronne (production sound mixer).

For *Ladyhawke*, Les Fresholtz (supervisor, re-recording mixers), Dick Alexander (SFX re-recording mixer), Vern Poore (music re-recording mixer) and Bud Alper (production sound mixer).

The original music scoring honorees were Danny Wallin, Record Plant Scoring, for *Out of Africa*; Dennis Sands, Group IV, for *Back to the Future*; Mike Farrow, Clinton Recording, for *A Chorus Line*; Armin Steiner, Warner Studios, for *Silverado*; and Bobby Fernandez, Warner, for *Ladyhawke*.

In addition, Stefan Kudelski, developer of the portable Nagra recorder, was presented with a Lyra Technical Award for his work in the motion picture audio field.

Database of synth voices and program patches available

For potential users that do not currently subscribe to an electronic mail and information service, Synth-Bank now offers a substantial discount on Performing Arts Network membership ($100 off the normal $150 fee). PAN is the E-Mail service that will offer access to Synth-Bank initially.

The on-line service of synthesizer sound files will include patches made available by a variety of artists and performers, including Herbie Hancock, Frank Serafine, Tony Williams, Bobby Nathan, Paul Lehrman, Jeff Bova, Tom Metcalf, Sterling Crew, Cory Lerios, Northstar Productions, John Senior, Paul de Benedictis, Doug McKechnie, Henry Kaiser, Howard Leese, Goran Anderson and Bill McCutcheon.

Also available on Synth-Bank will be a public domain library for the following

units: NED Synclavier, Fairlight CMI, Ensoniq Mirage, E-mu Emulator II, Casio, Rhodes, Chroma, Oberheim and Yamaha.

By using a PC, modem and MIDI interface, users can download sounds and sequences over conventional electronic mail networks and then load them into the appropriate instruments.

For more information, contact Bryan Bell via IMC (BELL-US), CompuServe (76327, 3041), WELL (BBELL) or PAN (SYNTH-BANK), or in the Los Angeles area, Bill Hartman at 213-876-8609.

JBL scholarships awarded

In an effort to support education in recording technology, JBL Professional recently awarded four scholarships to students attending the 36th Annual Aspen Music Festival and School.

The recipients were: Augustino Gagliardi, a trumpet student at the Hartt School of Music in Hartford, CT, studying with with Raymond Mase; Maria Vom Lehm, a percussion student who is attending the Peabody Institute in Baltimore, MD, where she studies with Jonathan Haas; Cheeyun Kim, a violinist studying with Dorothy DeLay at the Juilliard School, New York; and Marie Hopper, the principal clarinetist with the Greensboro Symphony Orchestra, Greensboro, NC.

Digital Workshop on the road

Nashville's Music Resources, an electronic music production service, recently went on the road to present its Digital Workshops of electronic music classes to students and faculty at Millikin University, Decatur, IL.

Originally designed by Steve Schaffer, Music Resources' owner, to inform and educate potential clients, the Digital Workshops program is now offered to schools of music throughout the country. Each 18-hour workshop covers four basic areas: sampling, synchronization, synthesizers and microprocessor-controlled devices.

Primary focus is on the New England Digital Synclavier Digital Music System and includes discussion of various synthesizers, samplers, drum machines and similar equipment.

Each workshop offers hands-on programming instruction and focuses special attention on the various forms of time-code and synchronization.

More details are available from Steve Schaffer at 615-794-3700.

Pro audio distribution company formed

Gerald Abeles, formerly vice president of marketing for Ursa Major, has formed A/V Technology International, a new company devoted to representing and distributing professional audio and video products worldwide.

The company will represent products from European, Asian and Pacific manufacturers in the North American market, as well as representing American manufacturers overseas. Lines handled by the company will include products for recording, production and

post-production facilities involved with music, audio, video, film, broadcast and multimedia.

For more information, contact Abeles at P.O. Box 275, Newton Centre, MA 02159; 617-965-5656.

People
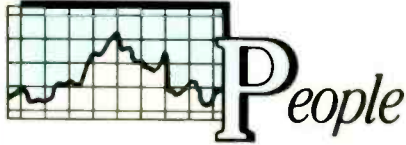
Steven J. Hebrock has been promoted to the newly created position of engineering manager at Audio-Technica. He previously was a design engineer for the company. Philip J. Lantry has been named regional sales manager, professional products, and will work out of the company's Stow, OH, headquarters.

Bill Mead has been promoted to director of special projects, motion picture division, for Dolby Laboratories, where he will be involved with the company's film program.

Abe Hoch has been appointed vice president of marketing at Audio Analysts USA.

Michael Wuellner has been named as product specialist for the professional audio division of Nakamichi USA.

Gerry Eschweiler has been named to the newly created position of vice president/general manager for Digital Entertainment Canada.

James M. Ruse has been named product specialist at DeltaLab, the professional audio division of Analog & Digital Systems.

Ken Meyer has been named Western regional manager of Sony's professional audio division, where he will be responsible for sales and marketing of pro audio equipment. Gary Rosen has been appointed Eastern regional manager/digital audio of the same division.

David "Doc" Goldstein has been promoted to sound department chief engineer at Universal Studio's sound department.

Jim Williamson has joined Studer Revox America as field service engineer, based out of the company's lower Manhattan field office.

Pamela Lyons has been appointed vice president of administration at Turbosound, where she will be responsible for the financial and administrative aspects of the company's American distribution.

R·E/P

fairlight

Managing MIDI

By Paul D. Lehrman

I'm pleased that **RE/P** has asked me to write a regular column on MIDI applications in professional recording. I come to the field with experience both as a recording engineer and producer, and as a developer of MIDI software. With the advanced MIDI tools now available, I have been able to set up a personal use studio in my living room—a computer, a reverb unit, a half-dozen synths and a couple of two-tracks (one analog, one PCM)—which is capable of putting out masters every bit as good as those I've ever been able to achieve in a state-of-the-art 24-track room (provided that you don't mind the absence of vocals or "live" instruments).

And this is where MIDI ultimately could be the death or the salvation of the recording industry. Why should I spend $150 an hour on studio time, and work around the studio's schedule, when I can do anything I want at home, for no more than the price of electricity? And I can work whenever I feel like it, knowing that no one has fiddled with the console or realigned the tape deck since the last time I was in. If a studio wants me to come in, it will have to offer me more—a lot more—than I can accomplish at home. Which is not impossible but it does mean that facility owners will have to look very hard at MIDI, and what it can offer to their clients, if they want their business.

Some of the technology that a studio can install takes the form of hardware that would be too expensive or impractical for a home or personal use studio. A concert grand piano, modified with the Forte MIDI-Mod to turn it into a MIDI controller, might be one such possibility. Some pianists simply cannot play on any other kind of keyboard, no matter how responsive, or how well-weighted. And if the attachments we keep hearing about that allow a piano to receive MIDI information ever become reality, it would be a terrific investment for a studio.

Imagine being able to correct mistakes on a piano track without re-recording it; to edit the volume and length of each note; and then to hear it back on the instrument it was played on originally!

Another wise investment for a studio is MIDI-compatible processing gear. The Lexicon PCM-70, complete with

A MIDI sequencer can convert a 24-track studio into a virtual 40-track facility

Dynamic MIDI, is a harbinger of things to come. A reverb unit or DDL that can change its identity in real-time without audible glitches (and have those changes recorded as part of a musical sequence, so they can be repeated flawlessly) is pretty impressive. Such devices are not cheap, however, and it would be a rare home studio that would have more than one. A good high-end commercial studio, however, might be able to afford, say, six of the devices—think of the damage that could do.

Of course, a studio *must* have synthesizers. There's no reason to have one of everything—if clients are married to a particular, unusual instrument, they can certainly bring it in themselves—but having some of the common ones, like one or more Yamaha DX-7s or a Roland Super-Jupiter, can't hurt. If a studio wants to invest in some of the more saliva-inducing, high-end machines like a Kurzweil K250, or even a Fairlight CMI Series III or New England Digital Synclavier II, these can help bring in those clients who could never dream of owning one themselves. And, if a sampling synthesizer is part of the arsenal, then a large library of instruments and sound effects can and should be on tap.

Interfacing

The other hardware that a studio must have to survive in the age of MIDI is interfacing equipment. Boxes that can convert a client's pre-MIDI synthesizer, such as a Sequential Prophet-5 or a LinnDrum, so that it can follow MIDI signals, are a must. Even more important, however, are the types of devices that can make a MIDI sequencer an integral part of a larger system, involving both audio and video. We'll talk more about MIDI pointers in a subsequent column, but suffice to say there is a way to make some MIDI sequencers "auto-locate" just like a tape machine, and even to chase-lock to a device that reads or generates

timecode. With the right interfacing, a MIDI sequencer can convert a 24-track studio into a virtual 40-track facility.

But more important than the hardware collection is an investment in people, and good training for those people. Clients who come to a conventional studio expect that the staff will be experts in using its tape machines, consoles, microphones and outboard gear. With MIDI equipment, that expertise has to go a lot further.

There have to be synth programmers: Clients may have their own sounds, and discover that they don't work in the context of the final mix, so someone should know how to "tweak" the sounds in the right direction. There have to be sequencer programmers: If a client wants a line transposed, or a different type of feel put on a track, somebody has to explain how to do it without spending half an hour studying the manual. There have to be sampling experts: A client wants to use a particular sound, which is on tape or disc, and someone has to be able to load it, for example, into an Ensoniq Mirage or E-mu Emulator II without fuss, bother, noise, clipped envelopes or munchkinization. And there have to be interfacing experts: Clients may be unfamiliar with SMPTE timecode, but know that they want to be able to slave their sequencer to a timecode track, and someone has to be able to do it quickly and accurately.

In future columns, I'll be considering all of these subjects in detail, and dealing with specific problems that studios can expect to face when they start using MIDI. Your input is welcome: questions, suggestions, tales of horror or of triumph—send them to me at the **RE/P** editorial offices.

We'll also be discussing new applications for MIDI, particularly in the areas of MIDI-controlled signal processing and mixing. If you stay in the dark about MIDI, you may find that your clients are disappearing into their own living rooms—but if you welcome it into your studio, and treat it right, MIDI may be the best thing since 2-inch tape. **RE/P**

Lehrman is a free lance writer, electronic musician, synthesist, producer and a regular RE/P contributor.
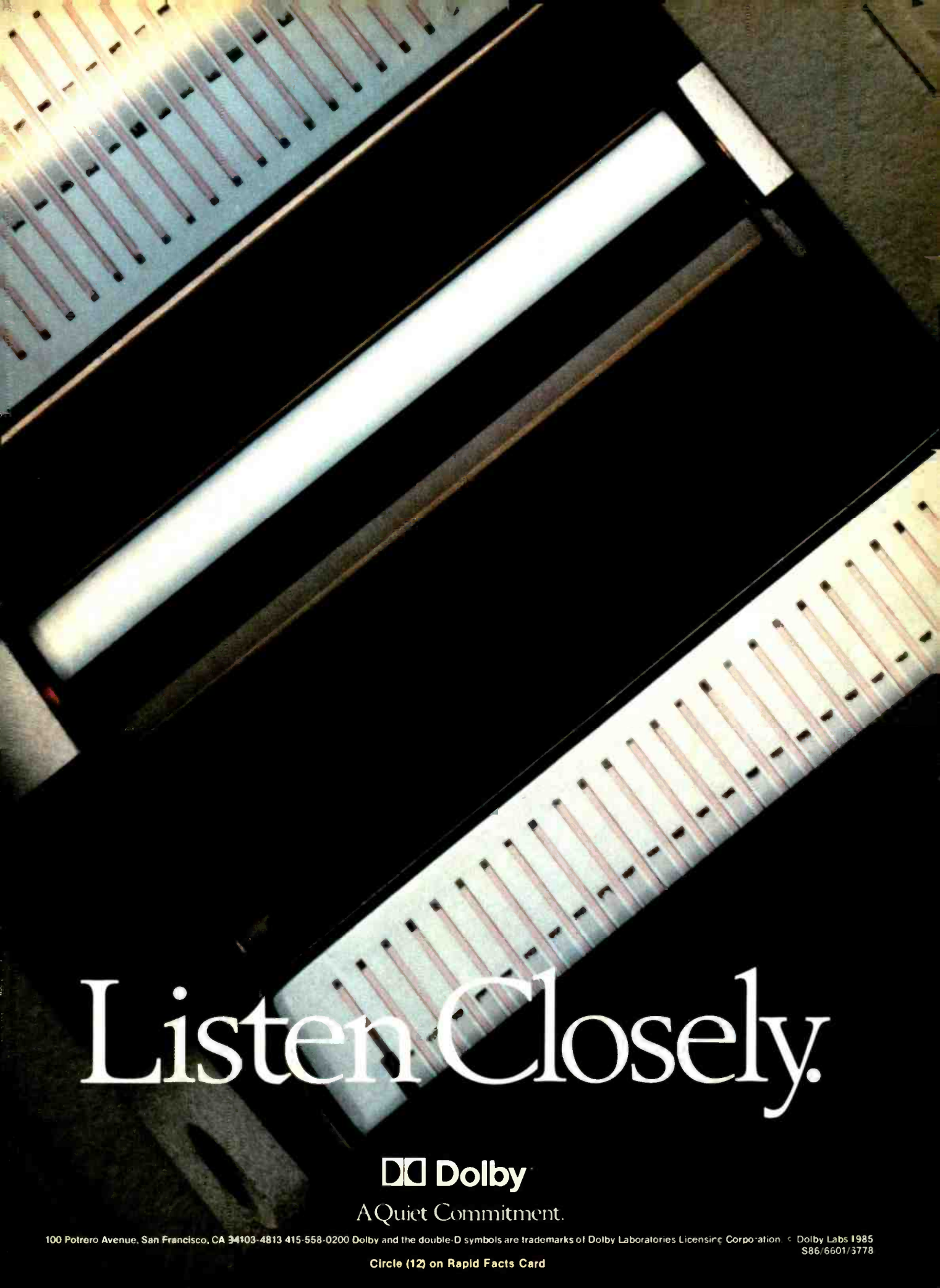
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Film sound today

By Larry Blake

This new column will focus on sound for theatrical motion pictures and will clarify how this discipline differs from its first cousins, video sweetening and album mixing.

Although the recording techniques involved in all three disciplines are merging, feature films still pose unique problems. It is my intention for this column to shed light on some of them. Reader feedback—either criticism of past columns or suggestions for future ones—will be greatly appreciated.

Credit where it is due

Of all the credits that roll up the screen before and after a film, probably the most confusing is that of the production crew. In addition to the "Produced by" card, one sees credits for executive producers, associate producers, unit production managers, assistant directors, executives in charge of production, line producers, and something called co-producers. All of these individuals participated, presumably, in the decision-making of a film, although the credits shed no light on the situation. On some films, the executive producer is really calling the shots (as with George Lucas), and in others that person may just be the person who owns the film rights to the book. The associate producer *may* be the director's girlfriend. Or she may also be the first assistant director, and had a great deal to do with realizing the director's vision.

Similarly, the craft of film sound is divided among many different talents, beginning with the boom operator and production mixer and ending with the re-recording mixers. Although there is no problem regarding the titles—no one confuses a music editor with a Foley walker, for example—there is tremendous difference from film to film concerning the degree of credit/responsibility that the specific job entails.

I'm not talking about misunderstanding on the part of the credit reading public. The industry itself cannot seem to figure out a way to properly attribute responsibility for soundtracks, either in reviews or when handing out Academy Awards.

The two major film trade papers, *Variety* and *The Hollywood Reporter*, simply name the production mixer as the person responsible for the *sound*. Thus the credit, "Sound (Dolby): Joe Smith." I hate to let these papers in on a secret, but something called re-recording was "invented" in 1926; not everything we hear in the final mix was recorded during shooting.

Also, production mixers have nothing to do with the Dolby Stereo aspects of a soundtrack; their goal is to capture a clear dialogue recording with a minimum of background noise, whether or not a film is ultimately released in Academy mono or six-track Dolby. Because this first column will attempt to clarify the "who does what" question, a little history and background may be helpful.

Film sound Oscars

For the first 40 years of sound films, the Academy of Motion Picture Arts and Sciences awarded the Oscar for best sound to the head of the sound department of the studio that produced the film. In almost all instances, this man was not an active mixer and did not record, edit or mix the film's soundtrack. In addition, at many studios the department head received credit for sound on all movies. (To this day many film buffs believe that Douglas Shearer actually recorded the sound for every MGM film for more than 30 years.)

It was not until 1971 that awards were first given to the individual mixers, and in 1976, the Academy began its current policy of presenting Oscar statuettes for best sound to the production mixer and (usually) the three members of the re-recording team. Although this was a major improvement over previous years, an obvious question arose: What about the sound editors? The answer is that they receive a separate Oscar for best achievement in sound effects editing.

Which brings up an important point: Segregation between mixers—production, music and re-recording—and sound editors is consistent at all layers of Hollywood, in spite of the fact that they both work together in sound. They belong to different unions (IATSE Local 695 and Local 776, respectively) and different honorary organizations (editors to Motion Picture Sound Editors, and mixers to the Cinema Audio Society), in addition to receiving different Oscars for creating the *same* soundtrack.

Politics aside, there is no reason why the two groups cannot, and many reasons why they should, merge at all above-mentioned levels. In spite of the fact that the two crafts are synergistic, editors and mixers seem content to play Hatfield and McCoy games instead of acknowledging each other's importance by joining together.

But this column is not about union politics. I want to provide some perspective as to who does what in the creation of soundtracks—no matter how the credits or Academy Award citations read. Let's start at the beginning, with production recording.

Production mixers worth their Nagras give equal credit to the boom operators: "If the boom isn't in the right place, there's nothing that I can do about it." Correct.

Regardless of what is said in conversation, how do the credits read when they crawl up the screen? The production mixer's name will be alone, early on, and the boom operator's will be much later, somewhere near the caterer's. Very few production mixers have the guts to share the credit for "production sound" with the boom operator.

Even if boom operators are not given their proper credit, at least the production sound team can be assured of receiving credit both in the Academy citations and in trade reviews. Such credit is best earned on films such as *Annie Hall* or *Kramer vs. Kramer*, where clear production tracks help carry the drama of the film.

Although those in the industry might notice and appreciate the talent that goes into producing such a soundtrack, with tasteful but subtle background and Foley, and crisp sonorous production dialogue,

Segregation between mixers and sound editors is consistent at all layers of Hollywood.

let's face it: When the press and the public thinks of good film sound, they usually think of spaceships and concerts.

The misconception that everything we hear was recorded during production becomes absurd when one considers that on many special effects spectacles more than 90% of the dialogue heard by the public is recorded during ADR sessions. In almost all of these instances, the high percentage of "looping" is not the fault of the production sound team, but simply a fact of life on films that employ visual and mechanical special effects. (This is not to mention that the consideration given on the set to production mixers is often low even when there is no ready-made excuse.)

Nevertheless, it simply isn't fair for the production mixer to pick up an Oscar for best sound on a film where the overall impact is because of the work of the sound effects editors and re-recording mixers. Also left out are the sound effects recordists, whose often-uncredited skill can make sound editors and effects dubbing mixers look—and sound—very good.

Similarly, music scoring mixers and music editors usually will not receive Academy nominations for their work on musicals, unless they also happen to be part of the re-recording team. It's a sad fact that a music editor's best work, like that of assistant cameraperson who pulls focus, goes unnoticed. These are quintessential examples of work that goes unrewarded if done well, and is noticed only if done badly. As tough as it is for music re-recording mixers to blend the music against the picture in competition with dialogue and sound effects, where would they be without a well recorded and edited track?

Role of the sound designer

In the past six years another title—*sound designer*—has found its way into film credits, further confusing the "who does what" issue. Although some people, such as Alan Splet and Ben Burtt, are truly responsible for the overall sound concept of their films—often including re-recording duties—it pains this writer to see the same credit given to people who might only create a few synthesized laser blasts.

Another point has to be mentioned about these "heir-apparent" sound designers: Although they might talk about their use of multitrack tape to create layers of a sound effects, remember that sound editors who cut on mag film have been doing this for years. The only difference is that the sound designer is able to audition all components in real time, and can mix down a composite effect that goes to the dubbing stage "finished."

What can be done to improve the situation, especially in regard to the Academy Awards? (Regardless of how seriously one takes the Oscars, winning one has an undeniably positive impact on one's livelihood; seen in this light, proper credit attribution is a serious matter.)

My suggestion would be to have only one award for sound personnel, eliminating the Oscar for best sound effects editing. In its place the producer of each film would have *six* positions to fill according to the contributions made to the final film. Whereas today the four statuettes are perfunctorily given to the production mixer and three re-recording mixers, in the proposed scheme the boom operator and supervising sound-effects editor would also be nominated.

However, on a large special effects show using almost all looped dialogue, the sound effects recordist and the sound designer might replace the production mixer and boom operator. The lineup on a musical might include, in addition to one member of the production sound team and the three re-recording mixers, the music scoring mixer and music editor.

These are broad examples used to indicate the possibilities. It should be noted that the total number of statuettes awarded to the soundtrack would not change, because two are frequently given as for best sound effects editing, which would be eliminated under this plan.

It has to be acknowledged that awards are icing on the cake, and the biggest and most important battles fought by sound crews concern getting the track made in the first place. Sooner or later, changes in technology are going to force everyone to rethink how film sound-tracks are put together.

To this end, a mutual understanding by mixers and editors across all union and work boundaries will go a long way to creating not only better soundtracks, but proper acknowledgement by the press and the Academy when they are *really* good. **R·E/P**

Living with Technology

By Stephen St. Croix

Before I got involved with whatever it is I do now, I made a good living playing the ARP 2600 synthesizer. This was by no means because I could play it, but rather that I understood the art. To be a synthesist back then—or electronic musician, as I put on my tax returns—you had to know how to *trick* your equipment.

Anybody who knew someone at a bank could obtain one of these devices, and you could buy patch sheets that taught you how to make the same sounds as everybody else.

Then it blew up. Not the 2600; technology. (That's not strictly accurate: the 2600 blew up every 18 days.) I had been expanding my analog voicing banks as new Oberheim and Moog modules came out, but now there was a new *kind* of device: a digital sequencer from Oberheim that actually remembered what I played.

Then came polyphonic synthesizers, machines that could remember voice patches and so on. I discovered that technology was advancing to such a point that it could make what I had obsolete before it wore out.

That was back then. We all know that today's technology can make what you buy obsolete before UPS gets it to your door. As cynical as that sounds, it *is* true. In fact, it is this precise state of events that has spawned the next level of technology. This new level holds the most promise for the immediate future and, at least temporarily, holds an answer to the question of what you should do when Federal Express brings your hot new Evecon model DPS-999X, and the magazine that comes in the mail announces the newer Yamtec AI-1.

Technical momentum

Finally, technology is beginning to move in our industry. When it first started, the acceleration was uncomfortable. Many became wary of actually purchasing the newest toys because they felt that a *newer* newest was just around the corner. They were right.

As our industry matures, it is natural that computers step in, and it is also natural that they step into our audio gear. Today's new improved engines, support and dedicated chips, coupled with the incredible software coming from some of today's 16-year-old screen jockeys, gives us a new concept in equipment: software-based devices.

Now when your engineer comes in and tells you about the newest DSP power available as you are unpacking your new (and now hopelessly obsolete) monster processor, he might be talking about the latest ROM set for that same machine. It is always nice to save face and money at the same time.

Buying a new device today can be like adopting a child: You expect it to grow and adapt. It is completely possible to buy a digital reverb today and have it transformed into a parametric equalizer next month.

For a significant percentage of functions, it seems to me that power and speed are interchangeable commodities in a computer. I'll use a CAD system as an example.

Let's say there's a schematic for a new circuit that makes anyone who sings into it sound exactly like Pia Zadora. You have just performed a serious edit, and the amended circuit now has to be redrawn and displayed in 16 on-screen colors. A given computer may take 100 seconds to do this. The same basic system made much more powerful with the addition of a math co-processor may perform the same task in 40 seconds. But, the same old not-so-powerful computer simply thinking faster at a higher clock rate, yet without the math co-processor, can also do it in 40 seconds.

Tighter, faster software can also produce similar results. All three of these factors taken together can produce roughly the same feeling you got the first time you put your foot heavily into your best friend's new large-block Chevy.

This is what today's DSP technology is all about: The pursuit of power and speed and the American way.

Just a few examples of this approach include Digidesign's Sound Designer software package for the Emulator II, Mirage and Sequential 2000; the Yamaha SPX-90 and ART DR-1; and, in other areas, the TimeLine Lynx and Fostex 4030/4035 timecode synchronizers.

All these devices can change as a function of customer demand and evolutionary advances in the internal software, but within the limits of how far the hardware can be pushed. The changes that I have already seen in these particular devices include higher accuracy, much faster execution of the same functions, new features and better display communication.

Technical dangers

I have said for a couple of years that technology moves ever sideways. There is so much potential power available now that a great deal of what was doubtless intended as advancement in fact seems to be merely different. We sometimes find ourselves with five different ways of doing the same thing, or a new improved, more powerful way of doing something that is now so complicated to use that we simply give up.

Do we really need a 68000-based bedside clock radio with a three-dimensional holographic plasma display, phone, automatic antenna rotor and a shower massage? Or a Walkman that can duplicate cassettes at twice the speed, and plays AM radio while you wait?

When it comes to our industry, this infatuation with technology seems to appear in a couple of ways. First, we are all learning another new phrase: not yet implemented. Just last week I wanted one of my new monster reverb processors to make me coffee, as mentioned in the ad. But, when I hit the proper keys, I got the message: *not yet implemented*.

I can understand the rush to compete in the marketplace, and that equipment gets shipped with some features unborn. However, I do feel that the manufacturer and dealer should provide a list of these inactive features when you shop.

Second, as the power of these devices increases, the problem of efficient, sane human interface becomes a serious consideration. I have equipment that is so well thought out that it can be used without ever referring to the manual. The front panel tells you just what is going on, and how to get to what you want next. I call this gear *mine*.

On the other hand, I have seen equipment with trick secret functions and a lot of hidden function pages that you have to get to *their* way, which never becomes *my* way. I call this gear *returned*.

All of it became clear to me when I put an unlabeled disk into an Ensoniq Mirage and the synthesizer told me, in plain spoken English: "This is a blank formatted diskette."

I'll never have to work alone in the studio again. **R·E/P**

St. Croix, RE/P's technology developments consultant, is president of Lightning Studios, Brooklandville, MD, and Marshall Electronic, Culver City, CA.

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The Audio/Video World of Thomas Dolby

By Alan di Perna

**The synthesist/musician steps firmly into soundtrack production
with his extensive involvement in the new Lucasfilm movie,
*Howard the Duck.***

According to popular wisdom, our data-congested age of specialization has made the Renaissance Man an extinct species. Although not quite a modern Leonardo da Vinci, Thomas Dolby can lay claim to major accomplishments in just about every entertainment format.

Most people think of Dolby as the bespectacled "Upper Class Twit of the Year." He has cultivated this image with singles and videos such as "Hyperactive" and "She Blinded Me With Science." But these are only the most conspicuous aspects of his multimedia activities.

Since beginning his career as keyboardist for Bruce Wolley and the Camera Club in the late Seventies, Dolby has distinguished himself as a session player (Malcolm McLaren, Foreigner and Joan Armatrading), record producer (Joni Mitchell, Prefab Sprout and George Clinton) and film composer (*Quicksilver* and *Fever Pitch*).

Equally adept at working with videotape as well as audiotape, he directs his own clips and recently collaborated on a longer video project with Japanese musician Ryuichi Sakamoto.

Dolby's varied background also makes him an ideal collaborator for George Lucas, that other noted high-tech eccentric and pop culture iconographer. And there's probably no more appropriate project for the two to team up on than *Howard the Duck*, Lucas' new film based on the Seventies comic-book cult character.

Dolby first became involved with the project via the standard route: The direc-

tor asked the pop star to write a song for the soundtrack. But from there, Dolby's participation became more and more extensive. First, Lucas asked him to write and produce all the music for Cherry Bomb, the all-girl rock group that co-stars with LucasFilm's million-dollar, computer-automated duck from outer space. Then, Dolby persuaded the director to let him handle the underscoring for the film.

To lay the groundwork, Dolby traveled to LucasFilm's San Francisco headquarters, where he spent a lot of time on the set and on location, observing and acting as resident rock-and-roll consultant for the movie's music sequences. It was a lot of work, but Dolby wouldn't have it any other way.

"There's so much more music being used in movies these days," he says. "But, very often, it's used in a completely inappropriate way, just because the music is going to help the movie marketingwise. As a result, I've never just contributed a single song to a score: one of those situations where you have one song by Phil Collins, the next song by someone else and the third one by someone else again. I've always been totally immersed in the projects that I have been involved with."

This type of total immersion is completely characteristic of Dolby, who has an uncanny knack for making his music very visual, and his visuals very musical. The key to this ability, of course, is his impressive command of the technology for both mediums. Dolby's aversion to

being labeled an "electronic whiz kid" has been well chronicled; for him, technology is more a means to an end than a suitable basis for a public image. But there's no denying that his past work has employed contemporary synthesizer, recording and video technology to its full potential.

The Think Tank

The command post for many of Dolby's projects is the Think Tank, an audio/visual studio the British artist maintains in London. The facility used to be a photographic studio, and has provided Dolby with a large control room plus recording area. Dolby converted a darkroom into a kitchen; later he discovered that it made an admirable overdub booth.

Most Think Tank sessions, however, tend to be held in the spacious main area, where there's ample room for the artist's newly acquired Fairlight CMI Series III digital synthesizer and other keyboards; a 32-input Soundtracs CM4400 console (automated by the CS52 interface and a software package for the Commodore 64 computer); a Soundcraft SCM-762 MkIII 24-track; Fairlight CVI (Computer Video Instrument); and five Sony 3640 ¾-inch U-Matic videocassette decks (left over from Dolby's last tour).

Dolby uses the VCRs for a variety of purposes, including video editing with the Fairlight CVI, storage of Sony PCM-F-1-encoded audio tracks and playback of workprints for scoring projects.

"I mostly use ¾-inch," Dolby says,

Photography by Phil Bray

"because when I get tapes from the U.S., they're NTSC, which I can play on ¾-inch equipment. Anything else has to be transferred."

The true centerpiece of the Think Tank, however, is the Fairlight Series III CMI, which serves as Dolby's main compositional tool for both film music and songs. For the former, Dolby synchronizes one of the Fairlight's internal software sequencers (either the 16-track Page R sequencer or Music Composition Language) to the timecode track on his ¾-inch workprint by means of a Friend-Chip SMPTE Reading Clock (SRC).

"The only problem in the past (using the Series II CMI)," Dolby says, "was that

the Fairlight couldn't 'chase' to SMPTE in the way that my E-mu SP-12 (drum machine) did. With the SP-12 you can chase to any point in a song program without having to start at the beginning of the sequence each time.

"But the CMI Series II would *trigger* from SMPTE; in other words, you can set a frame number for it to start playing back a sequence. And, with the SRC, you can set up to eight different cue points for the Fairlight. But once it has been triggered this way, the Series II will only stay in sync with its own timing.

"The Series III, on the other hand, *will* chase to SMPTE. That ability is going to make a big difference in my work,

because I do like to run sequenced tracks live when I'm working on a song. I don't like the idea of having to go back to the top of the song for each pass. What I've been doing (when using the Series II) is to make a couple of slave tracks of the sequence (on multitrack tape) and use those while I'm overdubbing other parts. Then, when I was ready to make a final pass, I'd just retrigger the sequencer and there would be no generation loss on the final master."

On previous film projects recorded at the Think Tank, Dolby has been renting an Adams-Smith synchronizer to lock his multitrack to timecoded video workprints. He indicates, however, that he plans to purchase a Fostex timecode synchronizer to handle the job in the future. In the course of scoring the Ryan O'Neil film *Fever Pitch*, Dolby devised a procedure that allowed him to ensure the continuity of his musical cues while simplifying the problem of coordinating the videocassettes and multitrack audio tapes with which he was working.

"The way I approached the project was to have one roll of film on each of the ¾-inch cassettes I was using. And I actually used one roll of 24-track tape for each reel of film, which is about 9 to 9½ minutes. I would sync up those two, and work one reel at a time. I generally used different tracks for different cues, sometimes filling up all the tracks. Then, on one particular set of tracks, I would give myself a slave stereo pair (i.e., a composite stereo mixdown of all the cues on the reel). This way, I could watch a whole reel at a time while listening to all the cues, which I think is very important."

As a compositional device, the CMI's usefulness for Dolby extends far beyond its multitrack sequencing facilities. He often uses the instrument's sampling capabilities to capture sonic textures that have some thematic relationship to the compositional task at hand. *Fever Pitch*, for example, is a film about gambling. Accordingly, Dolby went to Las Vegas and recorded a variety of croupier noises—clattering poker chips, cards being shuffled—and then sampled these sounds into his CMI. With these samples he built up percussive tracks for his score. This is the kind of approach that Dolby often also uses in his pop writing.

"That's the way I've done it since the *Flat Earth* album (Dolby's second record). It just gets me one step closer to the kind of textures I evenutally want to use. Take a song like 'Mulu the Rain

Forest,' for example. It started with a kind of cricket sound. Then I found a great sample of trees falling over in a forest—in stereo! That became a kind of drum beat. Then I blew over the top of a milk bottle, (sampled it) and got a kind of pan-pipe sound. That became my main melody. Working with those sounds, the rest of the song came. In the back of my mind, I already knew the kind of song I wanted to write. But being able to get straight to the right atmosphere got me there a lot quicker."

The Fairlight CMI is not the only sampling device that Dolby favors. "For samples that need to be of *very* high quality," he says, "I prefer to use the AMS DMX 15-80S delay. For the past year or so, I've generally been conceiving songs and getting together structures, arrangements, etc. using the eight tracks of Fairlight's Page R sequencer. Then, I'll use the Fairlight to trigger the AMS with a simple CV/Gate system."

On some occasions, Dolby also uses Page-R sequences to trigger drum sounds on his E-mu Systems SP-12 drum machine and Linn 9000. In other cases, however, he will synchronize the SP-12 and/or Linn 9000 to master timecode, and run them in tandem with the Fairlight, which is triggered from timecode in the manner described above. Dolby's other keyboard/synthesizers, which are also generally driven by the Fairlight sequencer, include a PPG Wave 2.3, Roland Jupiter 8, and Super Jupiter rack-mount module.

"Those are my main workhorses, really," says Dolby. "More recently, I have been fooling around a bit with an Emulator II and (Oberheim) Matrix 12, both of which have great sounds. But I've got better things to do with my time than to learn new systems constantly, so I tend to stick with what I know. It's always nice, on the other hand, to mess around with an unfamiliar machine. You chance upon things that you would never get on an instrument you know well; an instrument where you know what you're doing."

Howard the Duck: Building a musical identity for Cherry Bomb

Dolby's first task for the *Howard the Duck* soundtrack was to create—virtually from nothing—a sound and style for Cherry Bomb, the film's fictitious female rock group led by Lea Thompson, one of the stars in *Back to the Future*. This meant acting as songwriter, arranger and producer for the group; Dolby had, of course, played all of these roles many times in the past. But for this project he had to work within constraints he had never really faced before.

"It was quite challenging," he reflects, "to try and create this band's sound from scratch, and to create something which has nothing to do with me. LucasFilm wanted the actresses who play the band to actually *sing* the songs. And, while they're all singers, they're not very experienced, so I had to try and write the songs within their limitations. Also, the instrumentation was fairly basic—that was another consideration; it couldn't be too 'high-tech'. It had to sound like a *live* band."

For the most part, George Lucas gave Dolby a free hand in coming up with music for Cherry Bomb. "He did feel quite strongly that the group should be very vocal," Dolby says. "He felt that the vocals were getting a bit lost in the music. And, given that there are four girls in the band, I *have* used a lot of harmonies—but against a very gritty backing. In the film, the girls are all kind of cute; but they're kind of tough and independent as well. So the backing is very

Profile of Live Oak Studios

Floor plan of Live Oak Productions, Berkeley, CA, used during vocal recording, overdub and mixing sessions for the Howard the Duck soundtrack.
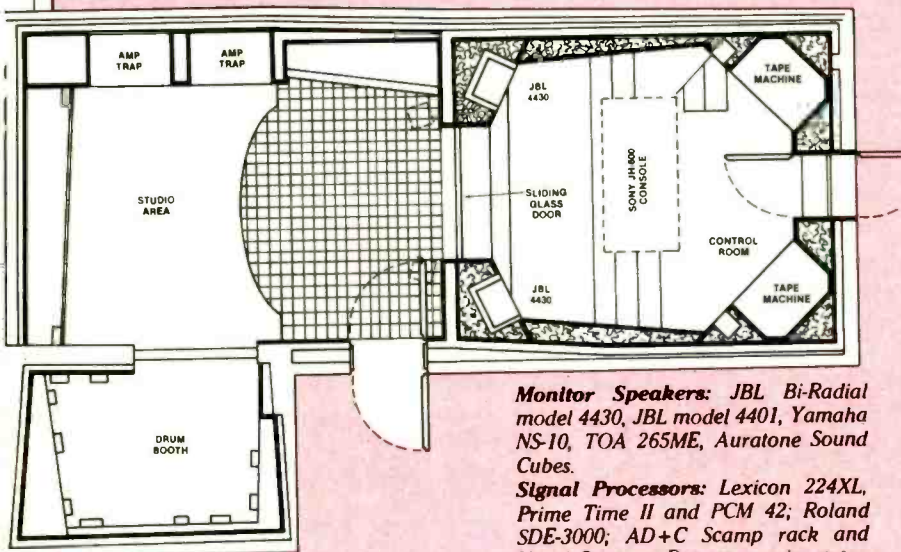
Monitor Speakers: *JBL Bi-Radial model 4430, JBL model 4401, Yamaha NS-10, TOA 265ME, Auratone Sound Cubes.*
Signal Processors: *Lexicon 224XL, Prime Time II and PCM 42; Roland SDE-3000; AD+C Scamp rack and Vocal Stressor; Drawmer noise gates; UREI LA-2, LA-3As and 1176 limiters; Spectra Sonics 610; Yamaha REV-7 reverb.*
Instruments: *Kurzweil 250 with digital sampling; Yamaha DX-7; Oberheim OB-Xa; IBM computer with 32 tracks of MIDI sequencing; Yamaha TX-816; Oberheim Matrix 12.*

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Console: *28-input Sony JH-600 with Sound Workshop Disk Mix II automation.*

Live Oak owner Jim Gardiner.

hard; but the voices provide a contrast."

Another concern of Dolby's was to integrate the music with the film's dramatic action. The songs occur as part of the plot, rather than as discrete set pieces. The film's web-footed protagonist may enter a night club, Dolby explains, and Cherry Bomb will be on stage doing a song. During the song, we see and hear Howard order a drink and ultimately get kicked out of the bar by a bouncer.

"So, as much as possible, I try to interpret that in the music, and try to suggest the gaps where Howard goes to the bar to buy his drink, for example. And, hopefully, the director will go for the 38 seconds I need in order for the band to complete the verse and chorus, and not conflict with the dialogue. But at this stage, a lot of this is just ideas. Basically, I just hope the music will be a kind of light relief for the action."

Dolby wrote all of the Cherry Bomb songs on the Fairlight CMI, using bass, guitar and drum samples from some of his old multitrack masters, a favorite technique. All the drum sounds on his second album, *The Flat Earth,* he reveals, were sampled from the multitrack masters for his first record, *Blinded by Science.* In the case of the Cherry Bomb songs, however, the samples weren't intended for use in the final project; they were only used to help the songwriter visualize the end result.

"It's only a four-piece band," says Dolby, "so it was all very unsophisticated. I'd write the songs on the Fairlight, and then I would get the girls in to check that I was writing in the right registers, and so on."

To record the songs' basic tracks, Dolby brought over from England three of his favorite players: bassist David Leavy, guitarist David Birch (one of Dolby's former Camera Club cronies), and drummer Neil Conti of Prefab Sprout (whose major label debut, *Two Wheels Good,* was produced by Dolby). Dolby handled keyboards. The basic tracks were recorded over a period of two weeks at two noted Bay-area studios:

Russian Hill and The Plant. Using accomplished players to achieve a relatively unpolished sound, Dolby discovered, was not without its own set of difficulties.

"The biggest job with them," he says of his musicians, "was that, because they were in a studio, they tended to behave as though they were in a studio. They were always saying things like, 'Oh can't we just punch-in on that bad note I played?' But I always wanted to leave all of that, because that would keep the tracks authentic."

Vocal recording, overdubs and mixing took place at another Bay-area facility, Live Oak in Berkeley. A synth-oriented 24-track studio tucked into a homey Victorian residence, the studio provided the ideal location, Dolby felt, for wresting the best possible performance from his troupe of actresses turned rock singers. The finished tracks were mixed down to a Sony JH-110 four-track. Along with stereo audio tracks, SMPTE timecode and a 60Hz reference tone were also printed on the master. The 24-track master was striped with identical timecode.

"This gives me the option of remixing the tracks to picture after it has been shot and edited," says Dolby. "Based on the dramatic action, it may turn out that we'll end up with a conversation going on over a vocal after all. Rather than dipping the music down to a point where it's inaudible, I think it would be better to remix and restructure the tracks around the dialogue."

Along with creating the music for the Cherry Bomb segments, Dolby also found himself drawing on his experience as a rock video director and lending a hand with the visual side of Cherry Bomb's presentation. Coincidentally, the band scenes were shot at two local venues: the Mabuhay Gardens, where Dolby played on his first U.S. tour in 1978, and the Warfield Theatre, where he performed during his last U.S. tour in 1984.

"That aspect has all been kind of nostalgic for me. What has happened is that I've become a kind of resident rock and roller on the project. I found that after the first couple of weeks of being here, a wardrobe person or a set designer would come up to me and say: 'Oh, the director would like to talk to you about how this should look.' The director of photography even asked me to draw him up a kind of sample MTV storyboard to use for one segment. You know: 'Guitar solo—shot up the guitarist's trouser leg; drum fill—close up on the

drummer;' that sort of thing. I'm also assuming that some of these songs will be released as singles by MCA, and that I'll probably be directing the videos for the songs, which will be incorporating footage from the movie, but also some outside material as well."

Underscoring

With the completion of filming on *Howard the Duck*, and the recording of the Cherry Bomb tracks, Dolby returned to London to write the film's underscore. (At the time of writing, this phase of the project has not yet begun.) Even before returning to the Think Tank to begin scoring in earnest, Dolby found himself doing preliminary work on the score in an unexpected manner.

"It turns out that they needed a temporary score very quickly," he explains. "So I actually went out and bought a bunch of library music that I can give them to use temporarily—music that I think is more or less in the right direction. That is also useful because it gives

you some ideas, and points out certain things that work timing-wise. If you stop and listen to them a few times while watching the action, you can work out what's good about them, and then try to incorporate those same elements into what you write."

Music sources of this nature can also be helpful in suggesting tempos for specific music cues. Further assistance is offered by the CMI's Music Composition Language software, which Dolby likes to use on scoring projects. MCL allows the composer to type in timecode addresses for all events in a particular scene that have to be "hit" with a musical emphasis of one sort or another. The program then calculates the optimum tempo required for a piece of music to hit all of those events.

Although Dolby has been involved in film projects before, *Howard the Duck* will provide him with his first opportunity to write a score for a large orchestra, an aspect of the project that is especially exciting for the longtime veteran of rock

bands. One feature of his Fairlight CMI that will be particularly useful is this connection is the instrument's music printing capability.

"The thing is, I don't actually write standard music notation," he says, "and I'm going to be using an orchestrator for the orchestral portions of the score. Rather than just giving him a tape, I'll be able to print things fairly accurately."

Although Dolby has never before used it on quite so large a scale, the CMI's music printing feature has proved useful on several of his previous projects. "I was asked to do the opening title sequence for a movie called *Quicksilver*," he recounts. "It featured a saxophone solo played by Tom Scott. I actually played it into the Fairlight, and then printed it out for him. I did the same thing with a four-piece brass section that I had the Brecker Brothers play on a single I did. I got the Fairlight to print out the part, and they were able to play it pretty accurately. So, all in all, the Fairlight music printing facilities are very useful indeed."

Another benefit of Dolby's extended involvement in the early stages of the *Howard the Duck* project at LucasFilm is that it has enabled him to set up a cooperative working relationship with the film's sound designer, Randy Thom.

"I think that's a very luxurious position to be in," he says. "Very often the sound designer and composer don't even meet until the final mix, which can be a problem. Working independently, the composer and sound effects person often hit the same events. You know, a man falls out of a tree and the composer does a huge chord. But the sound effects department has the sound of a tree cracking right in the *same* spot. The two sounds don't work together. So, in the end, you have to choose one or the other. By working with the sound department, I can avoid that and help make things more consistent."

Once Dolby adjourns to the Think Tank to begin intensive work on the score, communication with the LucasFilm sound department will necessarily become more difficult. He has several plans, however, for working with the sound department throughout the scoring process. Not surprisingly, one of these involves the Fairlight CMI.

"If we can get them a Fairlight Series III up there in San Francisco," he speculates, "one way to do it will be simply to send Fairlight floppy disks back and forth. But initially, I think, as they edit scenes, they'll courier them over to me on videotape. I'll score them and

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And we stand behind you—with some of the most advanced research in the industry. All to keep our standing—as number one in the world of the pro.

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send back mixes of the music, actually laid into the audio track of the videocassette. This way, they can see how the mix works with the picture."

The idea of using the CMI as a medium for collaborating over long distances is one that Dolby has used in other areas of his work. While co-creating the "Fieldwork" music video project with Ryuichi Sakamoto, much of the music was arranged and fine-tuned via Fairlight disks, which were sent back and forth between Tokyo and London. This scheme was supplemented by an exchange of data via electronic mail. And even when Dolby co-produced Joni Mitchell's *Dog Eat Dog* album, a great deal of the arranging was accomplished individually by either the producer or the artist, working away from the studio, on a Fairlight.

"For example, the opening song, 'Good Friends,' started with a fairly rough kind of piano part against a drum beat. I picked out a few sounds and textures on the Fairlight, and put them together at home. I came in and played it to Joni; she used various elements and threw others out. The good thing about using the Fairlight on that project was that we had one at the studio, I had one at home, and Joni had one at home, too. So we could work together on the one in the studio, and then each of us could take home disks over the weekend."

Looking toward the future: Feature films helped the Radio Star

Ultimately, our modern-day Renaissance Man would like to add to his list of media accomplishments by directing a full-length feature film of his own. His extensive involvement with the *Howard the Duck* soundtrack, he says, has provided a tremendous education in the mechanics and aesthetics of feature film production.

"Also," he adds, "the whole film industry and the whole music industry will know what my involvement was. And so, if the movie is successful, I think it will be a lot easier for me to approach film studios and try to get some backing for a project of my own."

The idea of learning by observation, after all, was what made Dolby a music video director to begin with. He originally approached rockvid mavens Steve and Siobhan Baron, of Limelight Productions, to direct his clip for "She Blinded Me with Science." When he showed them the scenario he had drafted to the video, however, they sensed he had the potential to direct it himself.

" 'Some of your technical language is incorrect,' they told me, 'but all your ideas are very good. Why don't you just
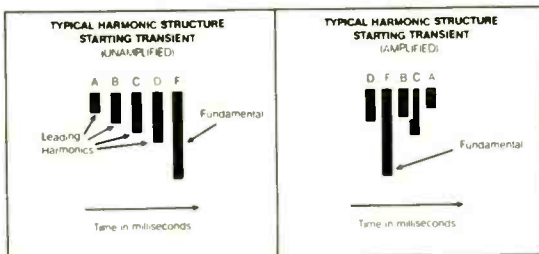
"I like the way BBE™ brings out the 'live' dimension in sound."

Bernie Grundman

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

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

The BBE 202R is a vital link in the chain of precision components used at Bernie Grundman Mastering in Los Angeles. Even the finest components, you see, can be undermined by that all-important interface between amplifier and speaker. That's where phase and "overhang" distortion develop, due to voice coil characteristics, reflected impedance from the environment, crossover impedance anomalies and the mechanical properties of dynamic speakers. The relationships among the fundamental frequencies, their leading harmonics and between the leading harmonics themselves become distorted in both amplitude and time. The result? Muddiness, poor imaging and pinched, colorized sound that lacks the presence and punch of the real thing.

The BBE 202R puts the clarity and sparkle back into amplified and recorded sound. We like to think of it as the "unprocessor." Rather than artificially altering the original source, BBE restores the natural harmonic balances that were present in the live performance. How? First it divides the audio spectrum into three bandwidths. Then it applies phase correction across the full spectrum and dynamic high frequency amplitude compensation as required. BBE's continual sampling of the mid/high frequency relationship allows this correction to take place

TYPICAL HARMONIC STRUCTURE STARTING TRANSIENT (UNAMPLIFIED)

TYPICAL HARMONIC STRUCTURE STARTING TRANSIENT (AMPLIFIED)

SYSTEM PROCESS BLOCK DIAGRAM
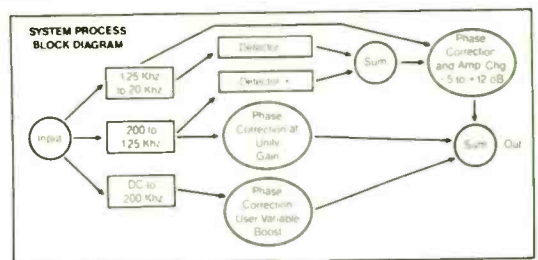
automatically. Convenient front-panel controls let you boost low frequencies and regulate the amount of high-frequency amplitude correction to suit your needs. There's no encoding or decoding involved, so BBE can be used anywhere in the recording chain—from individual tracks on a multitrack tape to a mastering lab monitoring system.

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

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Circle (23) on Rapid Facts Card

come down and hang around one of our shoots? You'll soon pick up the lingo, and be able to do it for yourself.' So it's really because of them and their encouragement that I started directing."

The results of that encouragement are well known. "She Blinded Me with Science" was the video that crystallized Dolby's "loopy" personna, and did so with an abundance of sight gags that accentuated every rhythmic nuance and odd little sound on the record. As Dolby explains, producing records and directing videos for them became part of the same process for him.

"Since I've been making videos, I've always thought of the video when I've recorded a song. I don't go to the extent of saying: 'I want to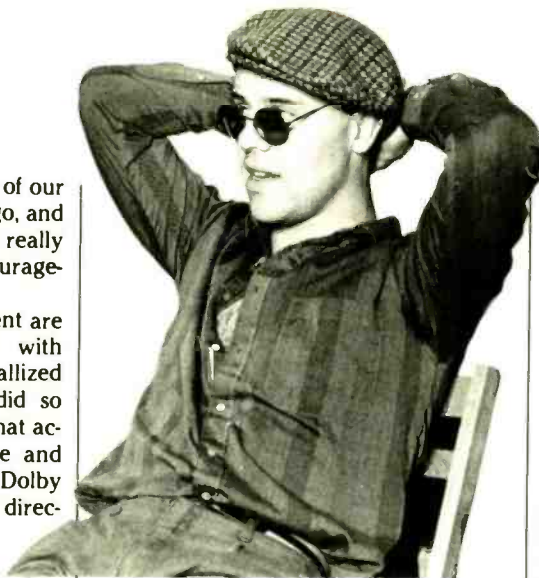 make a video with me on top of a mountain with the wind blowing my hair; so let's write a song to go with that.' That's *not* the way I do it. But on 'She Blinded Me with Science,' for example, a lot of the little sounds that were put in there are very visual. They conjured up pictures for me—even though I didn't know specifically what they were going to be. So I've always left things open in that sense."

Dolby's directorial approach to music also extends to the way in which he handles lyrical themes, a factor that has distinguished his own records and his production work with outside artists. A self-confessed "word man," he says his own lyrics are "self-consciously written so that they could be read off a page and analyzed by literature professors." While producing music for other artists, he places the same kind of emphasis on lyrical content; in a sense, the arrangement becomes a kind of underscore for the song's theme.
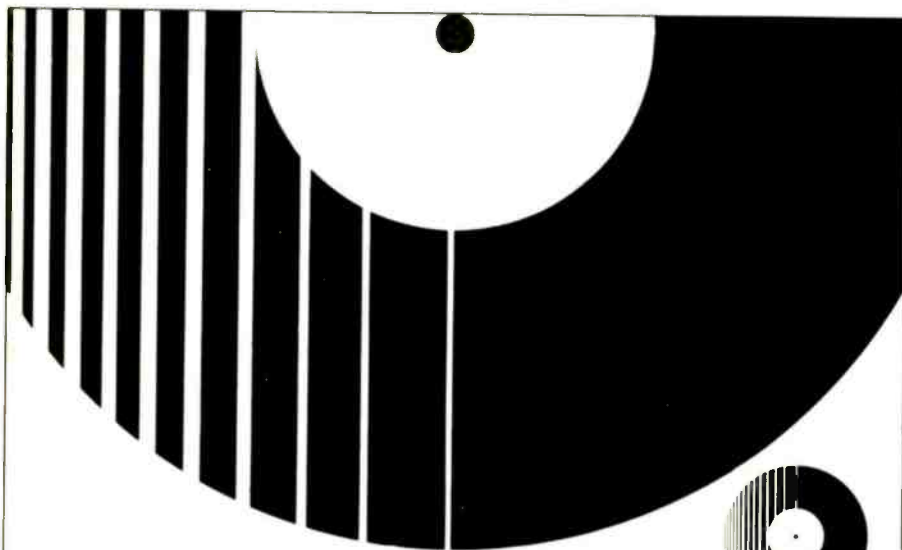
"I try to pick out the key lines," he explains, "and build the arrangement so that everything points to those key lines. Prefab Sprout, for example, are full of lines like that. Lyrically, they're the opposite to me in a sense, because they very often jump tenses, and jump from the first person to the third person. Their lyrics are a lot more of a collage, really. You get these very mysterious, very haunting odd lines that jump out at you. I like to lay a bed against which lines like that can shine."

In more ways than one, then, Thomas Dolby is a "master synthesist." What makes him unique as an artist, director and record producer is his ability to take techniques and attitudes from the various mediums with which he works, and synthesize them into a working procedure for whatever task is at hand. All of which explains the visual elements in his music and the musical elements in his visual work, for instance. Immediately after completing *Howard the Duck*, he plans to turn his "synthetic" creativity loose on the next Thomas Dolby album. In the meanwhile, we can look forward to hearing several new Thomas Dolby tunes on the *Howard the Duck* soundtrack album.

"If this movie is successful," he says, "it will confirm a belief I've had for some time: that I may get to the stage where I don't need radio anymore as a vehicle for my music. Any radio play I get will just be icing on the cake. It has always upset me that I've got, in the first place, record executives and, in the second place, radio programmers between myself and the public. *They* decide whether or not the public will hear my music—it's like a form of censorship, really. You can record whacky, weird stuff, but it will *never* get heard. The public may love it and maybe they would buy it too—if only they could hear it.

"With a movie, on the other hand, if the director put music in a successful film, then you've got a captive audience: they've paid their money, they're sitting there, and they've got no distractions. What's more, they're also sitting between a fairly good speaker system. It's really a great opportunity to have your music heard under nearly ideal conditions."

R·E·P

A User's Guide to MIDI

By Mark Lewer

The studio revolution is here; what remains is learning to use the musical interface to your best advantage.

There are several trends within the pro audio industry that are going to change music and the way it is learned, written, performed and recorded. Because of its formidable powers and inexpensive implementation, the musical instrument digital interface (MIDI) is playing a leading role in revolutionizing the recording and production market.

In a nutshell, MIDI is now widely accepted as the standard interface that defines a format in which musical information can be transferred between synthesizers, sequencers, drum machines, signal processors and other MIDI-capable equipment.

The spec also details other conventions, including electrical levels, transmission speed and connector types. MIDI supports a host of non-trivial uses, most of which are easy-to-use stock features found on the majority of today's performing and processing hardware.

If you are already hip to MIDI, you're probably trying to get a piece of the action. You're also probably finding a lot of overly technical or poorly written documentation. Besides, you may not want to have that engineering perspective; you may only want to know what's possible, what to buy for your studio and how to use it.

Fortunately, MIDI control can be incredibly user-friendly. Most users will never need to know too much about the actual binary *words*, nor be a computer programmer to use the equipment. Your task probably will be to learn the button-pushing sequences for the various units available at the studio or production facility.

However, as with all things, the more you understand the subtleties, the more you can use MIDI to your best advantage in choosing and operating the equipment, without falling victim to misconceptions and ignorance. Certainly, you will need increased sophistication in using this new technology.

This article is intended to provide a useful insight to what MIDI is all about, and what you can do with it, from the perspective of the user. It is *not* intended as a technical review of the spec. (I'll be more technical in the sidebar.)

If you want more technical information, I suggest that you join the International MIDI Association and obtain a copy of the full MIDI specifications. This document should answer any questions you have and provide a solid foundation.

What's so special about MIDI?

The modular approach of MIDI control is software-based. Because it relies on few pieces of hardware, MIDI ensures dependable operations with low costs. As with the computer industry, prices will drop as more musicians and studios purchase user-friendly equipment.

Digital technology offers a lot of other advantages. For most synthesizers controlled by a built-in microprocessor, MIDI is inexpensive to implement. Because it is software-based, future development changes can be made easily with simple ROM exchanges for the instrument's operating system. This will keep such instruments off the obsolete list. Furthermore, ROM and EPROM updates can give MIDI capability to many older models lacking it.

Users can also stop worrying about incompatibility; you needn't be concerned about whether you'll need a 5V or 10V CV output, exponential or linear inputs or even masses of patchcords. MIDI is a system that finally lets synthesizers, drum machines and sequencers communicate with each other easily and quickly.

There are MIDI idiosyncrasies, however, and a lot of potential problems can be avoided by familiarizing yourself with the spec.

MIDI basics

Conceptually speaking, MIDI is a 16-channel party line event code that allows each component in the system to communicate with one another. Actually, MIDI is a music-oriented, microprocessor interface and programming language for systems control, analogous to a vastly powerful player piano.

To accomplish the generation and reception of specific commands, MIDI utilizes a transmitter/receiver setup. A particular unit's transmitter originates the data in MIDI format via a dedicated MIDI Out jack and assumes the role of controller. Transmitters can be either computers, keyboards, sequencers, drum machines or any sort of device with an on-board microprocessor.

Receivers, on the other hand, take the data from the MIDI In jack and either execute or memorize the corresponding instructions. Such units can be tone-generating and sound-processing circuitry, as well as computers, keyboards, sequencers and drum machines. The MIDI Thru jack simply relays a copy of the In data, and is useful for daisy-chaining the instruments together. (Some units, however, have subtle variations on these conventions. The Yamaha KX-88 MIDI keyboard controller, for example, merges the In data with data produced by the keyboard to form the Out data. In this way, instruments downstream get data from *both* sources.

A typical MIDI setup is shown in Figure 1. In this configuration, MIDI information can originate from the master keyboard and then be memorized and played back. The computer and interface could be replaced by a MIDI sequencer.

The difference between a MIDI-equipped computer and a dedicated MIDI sequencer is simple: The sequencer is equipped for MIDI operation, while the computer is capable of other functions. Depending on your software, both types of device can be called a sequencer, and the system can write and perform all sorts of track laying and patchwork.

MIDI operates at a rate of 31.25kbaud or 31,250 bits per second. With 10 bits being used in each byte of command data, we have a period of 320μs per byte. Unlike other serial interfaces, no error correction is used at this relatively slow transfer speed, because it would only tax the host microprocessor. (Historically, the 31.25Kbaud rate was chosen so that it would operate without error correction while still getting the job done.)

With a little math, plus the knowledge

that a complete MIDI message takes up a maximum of 30 bits (most commands are less), you can see that there is time enough for at least 1,040 MIDI messages per second, each requiring $960\mu s$.

Commands use the first (logical-zero) and last bits (logical-one) to indicate the start and stop of each MIDI byte. The inner eight bits comprise the actual MIDI message or instruction, and are of two types: *status* and *data*. Status bytes are the actual commands of the MIDI language and can be identified by the logical one in their most significant bit. (If all of this talk of bits and bytes leaves you a little glassy eyed, refer to the sidebar.) Each status byte is then followed by a certain number of data bytes, depending on the MIDI command being issued.

The data bytes carry the actual content of the message, and can be identified by the logical-zero in their most significant bit. A receiver interprets a status byte, waits for the proper number of data bytes corresponding to that MIDI command, and then either memorizes or executes it.

Status bytes come in two flavors: *channel* and *system*. The latter type of messages are responsible for communication and synchronization of the MIDI system as a whole and are therefore intended for every unit connected to the system. Channel messages are used to communicate with individual or groups of instruments.

System messages, in turn, are of three types: *real-time, common* and the *exclusive* command. Real-time messages are used to synchronize MIDI events to

Figure 1. A typical MIDI hookup, using a personal computer running sequencing software to control synthesizers, drum machines and MIDI-capable outboard effects units.

24 clock pulses per quarter note, as well as *start, stop, continue, reset* and *active sensing* commands. They are 1-byte messages that can be sent at any time, even temporarily interrupting other messages.

Active sensing is an optional feature for both transmitters and receivers. Units that incorporate this feature will operate normally before the command being sent. Once an active sensing command has been sent, however, it must be repeated at least once every $300\mu s$ whenever there is no other MIDI activity.

If a period greater than $300\mu s$ occurs without the active sensing byte being sent, the receiver will turn off all notes and return to normal mode. The active sensing feature, in essence, eliminates the possibility of a note-on lock-up occurring, which can happen if the MIDI bitstream is interrupted at the wrong point.

Common messages are two bytes in length and include *song position pointer, song select, tune request* and *end-of-exclusive* command. The song pointer command is simply a counting register of MIDI beats and facilitates the use of timecode interfaces as well as the continue function. The song select command identifies which song is to be played upon receipt of the start command. The tune request allows analog synthesizers to tune their oscillators.

Exclusive messages use a manufacturer's identification number to transfer data between one brand or type of instruments. After receiving an exclusive command, units without the relevant software identification will temporarily turn off their reception until they receive an end-of-exclusive command. The exclusive message can be used to provide automated control of peripheral gear,

such as digital reverbs and delay units. (Examples include the Roland SRV-2000 digital reverb, which features 24 MIDI-accessed memories, and the SDE-2500 digital delay, which includes 64 MIDI-accessed memories.)

Another use of the exclusive message is to pass patch parameters between synthesizers. This feature is most useful for instruments of the same brand of synthesizer, because each manufacturer usually has its own method of tone generation and it would be impossible to standardize MIDI patch parameter commands. (In other words, the patch you hear on one synth may not sound the same on another brand after being transmitted via MIDI.)

Another implementation of MIDI simply calls up a patch number from a bank already programmed in the instrument. This feature will be more useful, especially for live patch changing.

Channel messages contain a four-bit number in the status byte that electronically address the messages to one of the 16 possible MIDI *tracks* and are intended for receivers that previously have been assigned to the same channel numbers. System messages do not contain a channel identification number, because they are intended for every unit in the system.
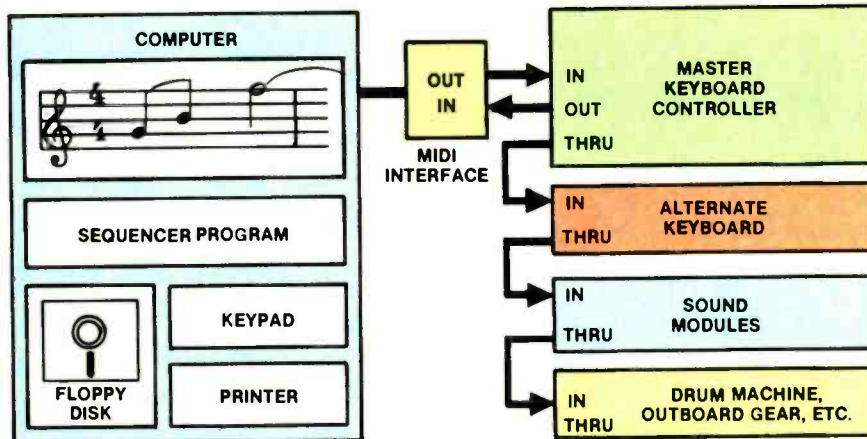
There are, in turn, two types of channel messages: *voice* and *mode*. Voice messages include *note on/off* commands, which carry an integer *note number* and *velocity* information in the data bytes. Voice commands also include *pitch wheel, after touch* and *modulation* commands, as well as those for program (patch) changes, and control changes for controllers (continuous, or on/off) assigned by the manufacturer.

Reassigning functions

Usually, it is possible to assign different controllers to different destinations, which means that you can assign such things as thumb wheels to whatever you like, such as pitch-bend or filter cut-off frequency.

A word of warning concerning the use of controllers: Each unit's individually programmed sensitivity to such controls will determine its reactions to MIDI commands. (If a particular synthesizer is pitch-bending a minor second while another bends a tritone, this is the reason why.) The same applies to such things as modulation sensitivities. These individual sensitivities are selected in the receiver, not via MIDI.

COMPUTER

OUT
IN

MIDI
INTERFACE

SEQUENCER PROGRAM

FLOPPY
DISK

KEYPAD

PRINTER

IN
OUT
THRU
MASTER
KEYBOARD
CONTROLLER

IN
THRU
ALTERNATE
KEYBOARD

IN
THRU
SOUND
MODULES

IN
THRU
DRUM MACHINE,
OUTBOARD GEAR, ETC.

CLEAR REASON

For the music studio owner, no decision is more critical than choosing a console. Both financially and creatively, the success of your operation may well depend on the capabilities and quality of the system you select, and the company that supports it. Clear reason, we suggest, to consider the SL 4000 E Series Master Studio System from Solid State Logic. But certainly not the only reason.
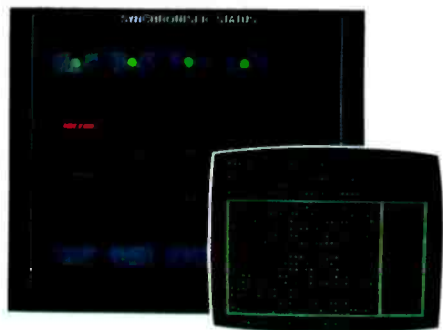
Consider, for instance, that only SSL has built-in track remotes on every channel, integrated with the industry's most versatile monitor fader and foldback facilities. Or that SSL alone provides pushbutton signal processor routing for each channel's noise gate and expander, compressor/limiter, high and low pass filters, and parametric equaliser — plus switchable phantom power, patchfree audio subgrouping, AFL and PFL monitoring, fader start for external devices, and stereo modules with balance and Image Width controls.

Consider that SSL makes the industry's only comprehensive studio control system — with integral synchronisation of up to five audio/video machines, concise English commands, tape location by timecode, foot/frames, cue numbers or key words, and complete session list management. And that SSL alone offers extensive fader, group and mute automation and mix manipulation *plus* optional programmable parametric equalisation and panning, multi-repeatable Events Control, and Automatic Dialogue Replacement.

Mode messages are a subsection of the control change channel messages and are sent and received with select channel numbers. In this way, each MIDI-equipped unit can be used according to its individual design potential.

There are four possible modes of MIDI implementation, each defining the relationship of the 16 MIDI channels to the synthesizer's own voice-assignment algorithm. Put simply, the mode commands determine the way in which the receivers *listen* to the data stream, and is determined by the selection of two toggling functions: *omni on/off* and *poly/mono.*

Omni on enables the receiver to execute the commands on all channels without discrimination. When omni is set to off, however, the receiver will execute commands from its own selected channel.
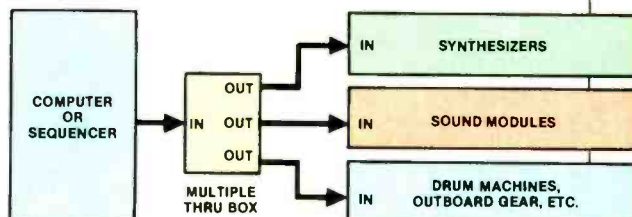
Mono restricts the number of voices per channel to one, and is used with monophonic synthesizers or synthesizers with multiple-voice capabilities, such as the E-mu Systems Emulator II. The second data byte of the command specifies the number of channels over which the monophonic voice messages are to be sent.

Poly mode allows any number of voices to be assigned simultaneously to a single MIDI channel.

Naturally, there are some subtle differences in the modes for the receiver and the transmitter; experience is the best way to really come to grips with their operation. For example, certain transmitter and receiver combinations

Figure 2. Some keyboard instruments can be controlled by either the unit's music keyboard, or from commands received via the MIDI-In port. Local Mode is used to determine which commands—those generated internally or from external speakers and controlling computers—will take command of the unit's sound-processing circuitry.

Figure 3. A Multiple MIDI-Thru box allows MIDI data to be received simultaneously by several devices, and so prevents cumulative timing delays.

send and receive via only one channel.

To avoid problems, it's worth ensuring that either the keyboard controller can select the channel on which it sends, or the sequencer can reassign the incoming channel. Then again, you could just write your own software to do this for you.

Another control change mode message, *local mode*, determines whether the sound-processing modules are controlled by an on-board controller or MIDI In data. As can be seen from Figure 2, a synthesizer's tone-generating circuitry can be controlled by either the keyboard or the In jack, as defined by the local mode. The ability to process the keyboard data, before it comes back to control the tone-generating circuitry, allows software to be developed that can accomplish what the particular unit wasn't made to do.

An example: a transposing program for non-transposing synthesizers, which could be implemented by simply adding or subtracting a constant integer from the note number data byte of the chan-

nel voice message. (A little programming here might also solve the earlier problem with channel-assigning capabilities.)

To ensure that the serial protocol creates no delay problems, the MIDI spec defines a feature called the *running status*, which can be used to speed up transmission of voice and mode messages. When a voice or mode status byte is encountered, the microprocessor controlling the synthesizer remains in that mode and awaits more data of the same type until another status byte (anything with the most-significant bit set to logical-one) comes along. Complete MIDI messages now need only consist of the specified number of data bytes in the specified order, since status bytes do not have to be sent. In this way, long strings of redundant commands can benefit from savings of up to 33% for both the transmission time and memory space normally used to store them.

Lengthy strings of note on/off commands are the most obvious benefactors of the running status. Furthermore, once

in the note-on status, velocity data bytes of zero can be used to *smart program* note-off commands. This kind of efficiency can help reduce the number of bytes present in the bit stream at any one time, effectively speeding up the passage of data.

The MIDI spec also has subtle provisions for continuous controllers, such as pitch wheels, which typically require high degrees of quantification and resolution. Two data bytes are needed to provide 14-bit quantification, while adequate resolution requires frequent sampling of wheel locations. Consequently, a great deal of redundant data is added to the bit stream, which can lead to a problem of the system simply not having enough time to handle all the channel and system messages, without experiencing time expansion and delays. Another potential problem can be the filling up of memory space with data.

To solve such problems, consider the following: When only the least significant byte has changed between subse-

Digital Fundamentals

Byte is the term used in computer terminology to describe a complete data word. In digital technology, this is a string of ones or zeros meaningful to a microprocessor. Bytes commonly are in 8-, 16- and 32-bit formats.

Every numbering system uses symbols to represent different quantities. A numbering system that uses two number symbols, zero and one, is a binary numbering system; this works well for representing on and off states in electronic systems. Unlike our common decimal or base 10 system, which can use 10 digits per column before running out of symbols, the binary system can use only two digits before it has to start a new column, which explains why binary numbers are so long.

As a result, some manufacturer's manuals use hexidecimal numbering to abbreviate them. In this system, 16 symbols are used: binary numbers zero through nine and A(10) through F(15). Regardless of which system is used, counting begins at zero and proceeds up until the possible symbols are exhausted and the next column (left) begins. Here's an example of the same number (15) represented in binary, hexidecimal and decimal notations:

$$1111_2 = F_{16} = 15_{10}$$

Converting from binary to hexidecimal can be done by combining groups of four binary columns, starting with the least significant or right hand group of four bits (a "nibble"), and converting each to its hexidecimal equivalent. Eight-bit binary words can hence be expressed as two hexidecimal numbers, each representing four of the bits. This method can also be used in reverse when going from hexidecimal to binary. Here's an example of this practice:

$$10110111_2 = B7_{16} = 183_{10}$$

If you need to convert either binary or hexademical numbers into their base 10 equivalents, the following formula can be used:

$$X_0 + (X_1 \cdot B^1) + (X_2 \cdot B^2) + \ldots (X_N \cdot B^N) = Answer_{10}$$

In this formula, X is the actual number in the column, N the column number (working from right to left), and B the base number you're converting from. Column numbers count up, starting with 0. The first term, X, represents the number in the first or right-hand column. (Remember, X $\cdot$ $16^0 = X \cdot 1 = X$.) Here's an example:
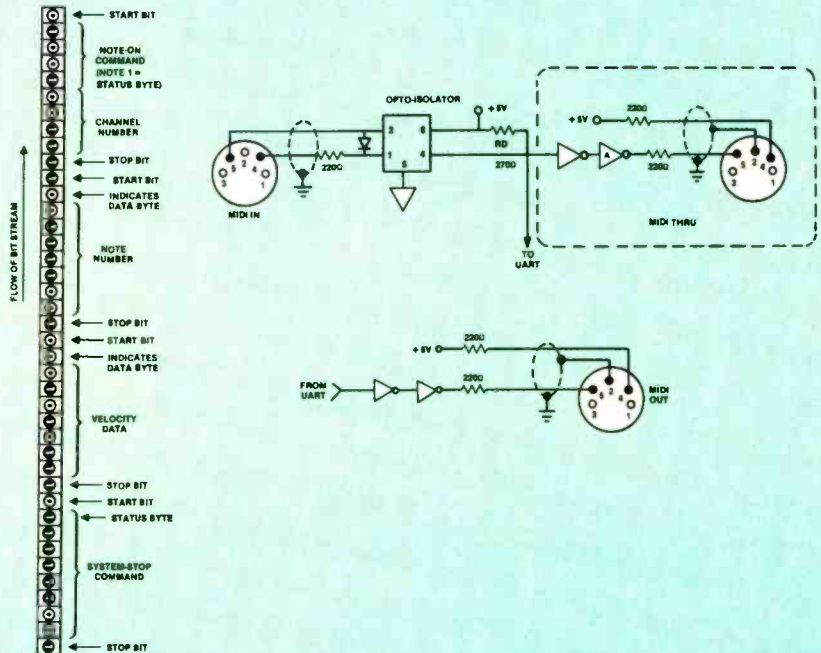
$$23A_{16} = 10 + (3 \cdot 16^1) + (2 \cdot 16^2) = 570_{10}$$

The bitstream in Figure 1 is an example of some imaginary section of a MIDI communication.

In this example, a note on status byte (1001) is sent in channel three (0011). Any unit assigned to this channel or omni mode will respond. The two data bytes that follow indicate that the Note number is 0111100_2 or 60 (middle C), and the velocity 0101011_2 or 43 (soft).

A system real-time command follows, and indicates the system stop message—it consists of a status byte only with no data bytes.

While status bytes have a one in the first, or "most significant bit," data bytes have a zero; this convention makes it easy for the receiver to known which is which.

Observe that different sorts of bytes use the remaining sections of bits differently. The Note On command is a Channel message, and this type of instruction uses the first four bits to carry the commands; the last four bits label the channel to which it belongs. Four bits allows for 16 (zero through 15) possible channels. The system command has no channel information, because it is intended for each unit in the system.

Data bytes can use the last seven bits after the zero. If a status command requires data with more quantization than seven bits can provide, two consecutive data bytes can be combined to provide 14 bits for continuous controllers such as pitch-bend wheels. (Fourteen bits will provide 16,384 discrete levels; 7 bits provide 128.)

Standard MIDI hardware

The second figure details the hardware found inside most units capable of MIDI control.

MIDI data originate in the transmitting microprocessor, and are driven into the MIDI Out port by a UART. The data are then taken by the receiver at the MIDI In, and passed on to another UART via an opto-isolator. This second UART converts the data back into a form that the receiver microprocessor can use. A copy of the In data is presented at the Thru output.

The gates labeled A are actually ICs or transistors. The recommended opto-isolator is a Sharp PC-900; resistor Rd is required to be 270Ω for this particular opto-isolator.

REVO- LUTION
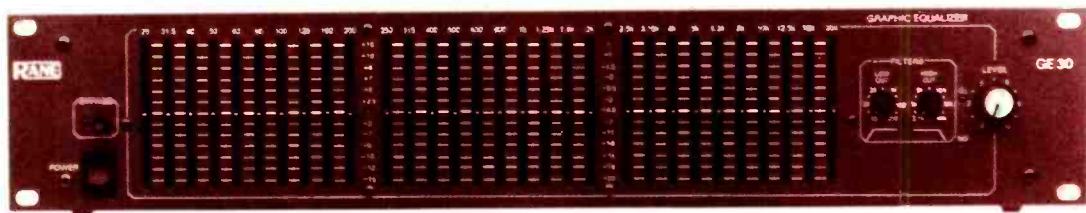
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quent commands (designating a small change), the most significant byte need not be sent. If a particular controller only needs seven bits of resolution, only the most significant byte need be sent. All of these solutions used concurrently can drop the byte count by up to 66%.

Problems can also crop up when long chains of units are tied together, as in Figure 1. The opto-isolator circuitry connected to each MIDI In jack have a limited rise-time characteristic (see sidebar). As a result, passing the data through each unit accumulates the delays added by each.

This problem usually can be solved by means of a multiple Thru box, as shown in Figure 3, which allows the data to arrive at each unit simultaneously. Up-market multiple Thru boxes include a switching matrix that allows selection of the master controller without having to repatch the various MIDI connectors.

MIDI delays can also be caused by the synthesizer itself or can be the result of a slow attack envelope. To solve the prob-lems of timing delays while laying tracks in the studio, a variable MIDI delay box can be used to provide a *relative* sync from the tape. Those units whose audio response is not suffering from a delay should have their respective MIDI information fully delayed, while those sources with audible audio delays do not need to be delayed as much.

Use of a MIDI-to-timecode interface allows a variable offset to be programmed into the timecode synchronizer during track laying. If the tracks are almost finished, and the last synthesizer part is always a smidgen late, a delayed sync tone can be recorded with the tape flipped over. Now when you lay the track the sync is slightly ahead, pulling the synth part back into place.

Hardware interfaces

The MIDI interface consists of a serial protocol, 5mA, current-loop system that is similar to the type used in many printers. Each unit is opto-isolated from the others to avoid ground loops and any type of electrical connection whatsoever. The circuit is simple, as illustrated in the sidebar. MIDI uses a universal asynchronous receiver-transmitter (UART) to drive the circuit.

Because the use of a current-loop format precludes any idea of simple parallel wiring to obtain multiple outputs, a multiple Thru box will be needed to cascade units.

The MIDI connect cable consists of a shielded twisted pair and 5-pin DIN plugs with the shield connected to pin 2 at both ends. Pins 1 and 3 are not used, but are available for future developments. (A word of caution: Many commercially available DIN cables—including some sold by synthesizer manufacturers—tie the shield to the plug's shield, defying the spec and creating the possibilities of ground loops in the system.)

Applications for MIDI

For the synthesist, MIDI provides a system that can serve as both a patch editor and a patch library. MIDI programs for patch editing use the controlling computer's video monitor and keypad to greatly facilitate the creation of custom sounds and storage to floppy disk or cassette tape. With MIDI's ability to control slaved synthesizers from one master keyboard, patches can consist of a number of different sound-generating circuitry (and outboard gear) controlled simultaneously over virtually any length of time.

With some programming ability, it is possible to make a synthesizer do what it was not made to do; the discussion of local mode contained an example of this capability. Don't be misled, however; just because a unit is equipped with MIDI doesn't necessarily mean that it can do things it was not designed to do. Synthesizers without velocity sensing, for example, simply ignore velocity data when presented with it. These kinds of hardware deficiencies may be overcome with a little MIDI creativity.

Keyboard players can really make the most of MIDI. Now all that's needed is one piano-action keyboard and some digital or analog sound modules. (For instance, the Yamaha TX-816 rack system is simply the tone-generating circuitry from eight DX-7s accessed via MIDI.) Finally, keyboard players can leave their equipment in the racks, and require only one keyboard controller on stage or in the control room.

MIDI isn't just for keyboard players,

however. Pitch-to-duration converters, such as the Fairlight Voicetracker or the newer guitar, woodwind and percussion interfaces, allow one instrument to be heard with another instrument's sound. (I like to pad my rock guitar solos with sampled Japanese subway screeches, for example.)

Such devices convert monophonic lines into a digital string of MIDI note on/off commands, with their accompanying note numbers and velocity data. MIDI-equipped drum heads can do the same, enabling drummers to access keyboard voices.

In fact, MIDI is well on its way to becoming *the* standard interface. Zeta System's programmable audio mixer, for instance, has all the features of a normal console—gain control, three-band EQ, fader and effects send/return—plus automatic panning, the ability to fade from one program to another and the ability to memorize 99 different console configurations, all via MIDI.

MIDI can also be a great compositional workhorse. The word processor nature of MIDI enables powerful editing and rehearsal capabilities. Best of all, the programming work a musician/composer does on a demo can be recaptured exactly when and if the project moves to a larger facility. Channels and their patches can be modified right up until mixdown.

If the session is really short of tracks, budget or time, live mixdowns of select live players, prerecorded audio tracks and synchronized synthesizers and drum machines may prove to be the answer.

Studios can benefit from all of this, as well as much more. The budget saved by only having to purchase a single keyboard controller can be spent on sound-generating modules that will take up little space in the rack. Session players might choose to bring their own modules and patch into the studio's MIDI system.

The rest of the money saved can be spent on a controlling computer (if you don't already own one), a computer-to-MIDI interface, MIDI software and MIDI-to-timecode device. Such a combination will require only one tape track—timecode—for total sync.

One such MIDI-to-timecode interface, the Roland SBX-80 Sync Box, can handle tempo variations, move them manually or record the tempo from pre-existing tracks and also includes an offset function to correct any timing problems that might arise. The Friend Chip SMPTE Reading Clock (SRC) features two delay lines for correcting timing problems. Despite the fact that the SRC provides no MIDI outputs as such, a pair of *n-pulses-per-measure* outputs and a pulse width modulation output can be used to drive other units.

Both the SBX-80 and SRC utilize MIDI song pointer, which prevents the controlling sequencers having to be started at the beginning of the song. The song pointer command is simply a count of the number of MIDI beats executed up to that point, and allows each unit to know where it is supposed to be within a particular sequence.

With this feature it is possible to simply roll tape to a point just before the punch-in and then let the sequencer do the work. (Obviously, these and other devices are great tools for video projects, which typically depend on timecode locations for cue spotting.)

MIDI sequencers typically emulate the operations of a 16-track recorder and console. The difference lies in the fact that a tape recorder stores the actual performances while a MIDI sequencer memorizes and stores the commands and data sequentially, to be performed by the connected instruments. Typical sequencers feature the ability to merge and reassign tracks, channel volume control, autolocate functions, and punch in and out.

However, their creative power exceeds that of a tape recorder and console in many ways. Sequencers can transpose and loop tracks, edit and insert or erase notes at any given point and display the whole score. Some are equipped with automatic panning and programmable tempo, as well as the ability to store the entire sequence on floppy disk, instead of expensive 2-inch tape. R·E/P

Lewer is a graduating senior of the music engineering technology program at the University of Miami, Coral Gable, FL, and has been working for Sound International, a Miami-based video and music production facility.

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Synergistic Studio Operations:
Image Recording & Composer's Services

By Adrian Zarin

A complementary production facility demonstrates how studios can follow changing music trends.

The nature of the production and recording industry in the Eighties has been the subject of endless discussion and debate. Here at mid-decade, one thing is becoming abundantly clear: The continual equipment upgrades required by advances in audio technology are only part of the picture for studio owners who wish to stay competitive.

Some of the latest developments in audio technology also require studio professionals to constantly re-evaluate the creative processes by which albums and soundtracks are made.

There's little doubt that the new electronic keyboard/synthesis/sequencing technology has had a profound impact on the way in which audio professionals do their work. Ideas that would have seemed utopian just a few years ago are now commonplace. Without leaving their living rooms, composers, arrangers and orchestrators can often achieve what, in the past, would have required a vast scoring stage and 50 or 60 musicians. Music and other program material can now be recorded at a fraction of what it once would have cost.

To survive these potentially threatening developments, facility owners and managers have been forced to adapt. One such adaptive strategy is illustrated by the recently forged partnership between composer/engineer and studio owner Tom Seufert, and the co-owners of Image Recording, Harry Maslin and John Van Nest.

Pooling their resources and expertise, the trio opened up a new room at Image's Hollywood, CA, location, and named it Studio B. In terms of its physical layout and equipment array, the new facility is designed to provide a variety of audio services—many of them synthesizer-related—at a competitive rate. As an adjunct to the Studio B operation, Seufert also operates a second facility at his home, called Composer's Services.

Referred to as an "audio off-line" studio, CS is in reality a synthesizer pre-production room that principally offers "tapeless recording" via MIDI sequencers. Seufert's intention is to provide an economical environment for composing, arranging and orchestrating just about anything from a pop song to a film score.

Composer's Services and Studio B were designed to work synergistically, the idea being that a project can be brought from concept at Composer's Services to completion at Studio B in one continuous, coordinated effort. At the same time, however, the two facilities were set up to handle any or all stages of virtually any type of recording project.

Facility origins

The building that houses Image Recording is located on Sycamore Street in

Equipment layout at Composer's Services, a pre-production eight-track facility, includes (clockwise from left to right): a Yamaha KX-88, Apple IIe personal computer, Yamaha DX-7 and TX-816 module rack, Linn 9000 MIDI sequencer/drum machine, Roland G-707 guitar synthesizer, Apple Macintosh personal computer and a Tascam 80-8 8-track machine.

one of Hollywood's fastest developing "Studio Rows," and was formerly the home of Allen Zentz' mastering lab. In 1983, John Van Nest and Harry Maslin acquired the facility, converted it into a recording operation and opened Studio A, Image's current automated, 48-track room. Van Nest, a recording engineer with albums by Night Ranger and Giuffria, among others, to his credit, had been working as a studio manager for Allen Zentz. Maslin, who was originally one of Van Nest's clients at Allen Zentz Mastering, is a producer/engineer, formerly with the New York Record Plant.

Soon after Zentz left the facility, Image's Studio B was converted from a mastering room to a synthesizer pre-production room. It became a full-fledged recording studio when Tom Seufert approached Maslin and Van Nest with the idea of relocating his recording studio—a San Fernando Valley facility called Redwing—inside Image Recording's premises. Accordingly, Studio B opened its doors in September 1985.

Conceived as a "MIDI/SMPTE Interlock Room," with a special emphasis on

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production for stereo television and film soundtracks, Studio B has already attracted a diverse clientele, including composer Peter Bernstein *(Battle of the Ewoks)* and record producer Peter Wolf (Jefferson Starship, Heart, El Debarge), Michael Jackson and Maurice White. As an extension of the services offered in Studio B, Seufert opened Composer's Services shortly afterward.

Composer's Services

Projects typically start at Seufert's Composer's Services, a comfortable one-room environment equipped with Altec 604E, Yamaha NS-10 and Walker Sound Cube speakers for monitoring synthesizer playbacks. Although the room features a Tascam 80-8 8-track and a stereo Sony PCM-701 digital processor for composition, arranging and orchestration work, Seufert prefers tapeless recording using his in-house MIDI sequencing units.

In this kind of work, he says, tape-related considerations such as printing signal processing and EQ are not all that essential, whereas the ability to quickly change notes, voices, etc. *is* vital. This, he concludes, is more readily achievable with sequencers.

One of the studio's principal sequencers is a Linn 9000 (equipped with a 3.5-inch floppy-disk storage option), which provides 32 polyphonic tracks,

each of which can be assigned to any one of 16 MIDI channels. The 9000 also serves as Composer's Services' principal drum machine. To enable him to maximize low end on certain voices, and minimize high end "aliasing noise," Seufert has had the drum machine section modified with variable filter circuits on the kick drum, tom and conga voices. And because the Linn 9000 at Composer's Services is equipped with the manufacturer's sampling option, Seufert is able to use his filter modifications to tailor sampled percussion sounds.

The second sequencing device comprises an Apple Mac running Southworth Total Music MIDI software; Seufert is able to transfer data from the 9000 to the Mac via a single MIDI cable. Total Music is possibly unique, he considers, in allowing simultaneous real-time data entry from two separate MIDI units—an option the composer finds especially useful during ensemble sessions.

"Up until now," Seufert says, "the biggest limitation of using this kind of equipment for composing is that you would have one guy playing a keyboard or programming a drum machine, while two other people sat around doing nothing.

"Now my partner and I can set up a 100-bar rhythmic pattern, and then just start jamming over it. You find that magic things start to happen that can only take place when you have two people

playing together. Bouncing ideas off other people is, after all, an important part of making music."

The next stage will be to have four or five musicians playing simultaneously into a sequencer via a MIDI merging device, such as the MIDIMIX, which allows four separate input devices to be recorded via a single MIDI In connection. During experiments with a MIDIMIX 9 prototype, Seufert has used the device to enter data simultaneously from four different MIDI-equipped instruments, the outputs of which are combined and sent to one of the MIDI inputs provided on the Total Music interface box.

The second input to the interface is used to accept a MIDI clock output from the Linn 9000, which, in turn, can accept input data from yet another MIDI instrument. With MIDI drum pads, guitars and keyboards as input devices, says Seufert, it is now possible to achieve a truly interactive "live-in-the-studio" feel when recording to a sequencer.

"The great thing about the Total Music system is that you can record without any quantization [or timing correction]," he says, "so you can really get a *live* feel. You can then add three different kinds of quantization afterward, to the nearest 16th or 32nd note, for example. Then you can go in and edit the material note by note, or just quantize or correct a certain figure for a musical feel."

Composer's Services offers a variety of instruments for real-time music entry into MIDI sequencers. Originally a guitarist, Seufert's instrument of choice is a Roland G-707 guitar synthesizer that has been modified with MIDI-In capabilities. As well as driving the studio's MIDI sequencers, the G-707 is also used to control a Roland GR-700 analog synthesizer module, which was originally designed by Roland for use with the G-707.

Centrally located in the room is a Yamaha KX-88, which serves as a MIDI keyboard controller. The unit's 88 weighted, wooden keys is said to make the KX-88 a favorite among the pianists and other keyboard musicians that have used Composer's Services. The KX-88 is part of a larger Yamaha system mounted on a Ultimate Support stand to one side of the room. The rack consists of a Yamaha DX-7 synth and a TX-816 rack module, with an Apple IIe computer that is dedicated to patch editing and librarian functions for the RX and TX synthesizers. Seufert principally uses Yamaha DX-Pro software on the Apple to handle such functions, backed up by
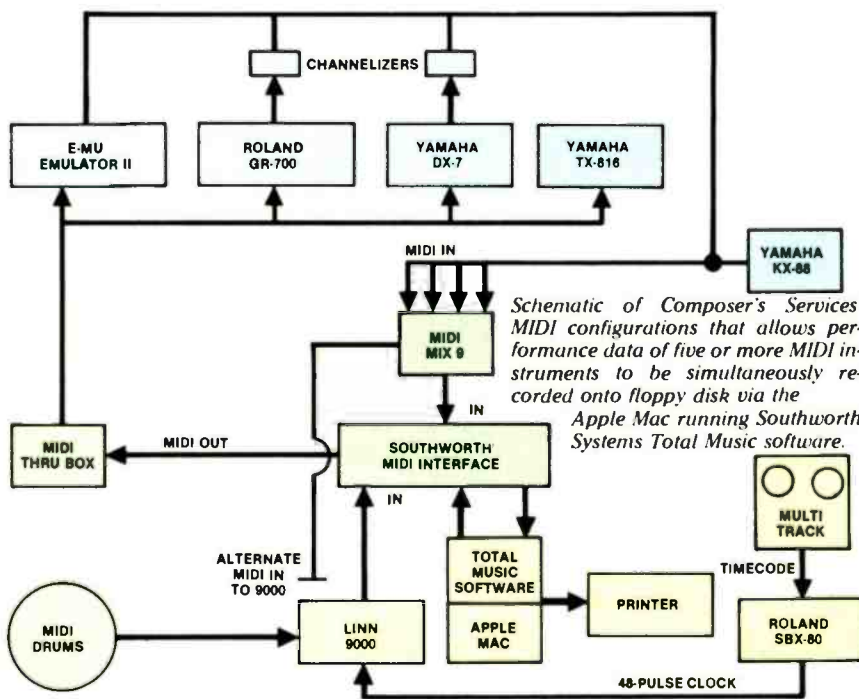
Schematic of Composer's Services' MIDI configurations that allows performance data of five or more MIDI instruments to be simultaneously recorded onto floppy disk via the Apple Mac running Southworth Systems Total Music software.

OpCode editing/librarian software for DX series synthesizers.

Housed on the same stand are two small but extremely useful devices. One is a Casio CZ-101 digital synthesizer, which is mainly used to enhance and fatten the DX-7/TX-816 patches. The second device is a 360 Systems MIDI Bass, for which Seufert has amassed an extensive plug-in library of electric and acoustic bass sounds.

The primary digital sampling instrument available at Composer's Services is an E-mu Systems Emulator II, which has an extensive library of sounds stored on floppy disk. To help streamline the pre-production/orchestration process, Seufert has compiled demo cassettes that catalogue all of the sounds available on his Linn 9000 drum machine, 360 System MIDI Bass and Emulator II.

"I just duplicate the tapes," he explains, "give them to my clients and say, 'We're going to work on bass sounds tomorrow. Listen to this tape, and tell me which sound you like.''

Lurking in one corner of Composer's Services is what Seufert refers to as his "analog museum." It consists of two venerable monophonic synths—a Minimoog and a Sequential Pro-One—plus two analog polyphonics: a Sequential Prophet-Five and T-8. Seufert finds that such units complete the sonic palette offered by his facility's sampling devices and FM digital synthesizers.

Although not a tape-oriented facility, the studio's Tascam 80-8 8-track comes in handy for several applications. "We have done some vocals on it," Seufert says, "along with guitars, scratch tracks, what have you. We use it to see if a song is in the right range for a vocalist, or for other parts that we want to record without sequencing. We can use it to build a rough mix for final approval; and, in some cases, I've even done finished projects on it. Usually I put SMPTE timecode on track 8 of the multitrack, and I can use that to retrigger all the sequencer parts when the project moves to Image Studio B or some other studio.

"In some cases, I've even transferred vocal or guitar tracks from the 80-8 onto 24-track with excellent results. It's surprising how good those [narrow-gauge] machines sound if set up properly."

Not forgetting his interest in film and television work, Seifert has also equipped the facility for scoring to picture. A JVC CR6300-U ¾-inch VCR, which doubles as the studio's storage medium for PCM-701 digital recordings, is used to replay video workprints with timecode. The timecode track on the video workprint is fed into a Roland SBX-80 Sync Box, which, in turn, controls the Linn 9000 and Total Music sequencers via sync and MIDI.

The idea was to create a synthesizer area immediately behind the console.

In this way, composers can enter musical passages in sync with picture. The same timecode used while composing the music can be used later as a master time clock for synchronizing the workprint, multitrack and sequencers during final soundtrack mastering.

Image Recording Studio B

It is at this stage in production—when the basic compositional and arrangement problems have been worked out, and the project is ready for final recording—that Image Recording's Studio B usually enters the picture. Intended to provide the hardware and services necessary during the generation of a finished master recording for album projects or film and TV soundtracks, Studio B is equipped with a transformerless Sony JH-114 24-track, a modified 32-input Trident Series 80B console and a choice of mastering onto half- or quarter-inch analog via Ampex ATR-102 machines. Tracks can also be mixed down to the studio's JVC 6650U ¾-inch, U-Matic, or digitally encoded via the studio's Sony PCM-F1 processor.

In converting Studio B from a mastering room into its present incarnation as a synthesizer-oriented tracking and mixdown facility, Seufert, Maslin and Van Nest decided to capitalize on the room's rectangular 14- by 24-foot shape. The basic idea was to create space for a synthesizer area immediately behind the console, ensuring that musicians working on tracking dates would be hearing the same basic playback as the engineer and producer seated behind the board.

In adapting Zentz' mastering room for recording and mixdown use, the partners enlisted the aid of design engineer Rick Ruggieri and carpenter Johnny Wiggins.

"Since our idea was to locate the synthesist and his equipment behind the console," Van Nest says, "we felt there were many advantages to having a nar- row room with a long throw. Besides hearing the same thing as everyone seated behind the console, the synthesist and his equipment are not blocking any of the waves emanating from the monitors, as they would be if they were located anywhere else in the room."

Together, the team worked out the ideal placement of the studio's UREI 813 Time Align monitors in soffits on one of the shorter walls, and a 25-inch Proton video monitor mounted between the two cabinets. After experimenting with a variety of surfaces, the designers decided to cover the two long walls with an absorptive fabric. The same surfacing is used on the rear wall; but the majority of this wall is taken up by a sliding glass door that gives way on to a 400 square-foot isolation booth.

An acoustically reflective space with brick walls and a hardwood floor, this L-shaped area was designed for music overdubs, voice-overs, dialogue replacement and Foley work. As Van Nest points out, locating the booth immediately behind the control room and providing visual contact via the glass door adds an element of convenience to this type of work. And for music applications, the isolation booth is equipped with a Kawai grand piano, which has been fitted with a Forte Music MIDI modification.

The studio's principal MIDI performance area is located directly behind the console in Studio B's main room. With the multitrack machine on the front wall and the Trident console positioned approximately in the center of the room, roughly half the available space has been left over for synthesizer equipment.

The studio has its own Emulator II, which enables Seufert to take sounds realized at Composer's Services and recreate them in Studio B simply by inserting a floppy disk. Apart from the E2, additional synthesizers are brought into Studio B as needed.

To interface these quickly and neatly, Van Nest also had two separate 16-channel MIDI lines installed in each of the side walls. Each line is connected to five different wall-mounted MIDI In and Out jacks located conveniently along the side walls. With all other studio wiring located in troughs beneath the floor, Studio B provides an obstruction-free setting for the ever-changing array of synthesizer gear that comes and goes on a regular basis.

MIDI/timecode connections
The key to Studio B's SMPTE/MIDI in-

terlock capabilities is its collection of four TimeLine Lynx modules, which allow synchronization of the studio's VCR, 24-track, two-track and any MIDI/SMPTE sequencing equipment that's brought in. Van Nest is particularly enthusiastic about the adaptability afforded by the Lynx modules.

"With most other synchronizers, you must generally use either your video deck or the synchronizer's own master unit as your 'master' shuttle controls. But with the Lynx, you can just cable to the multitrack, cable to the ¾-inch VCR, and simply tell the system what *you* want to use as a master. And that's it. If we decide to use our MCI AutoLocator as our master controller, it will shuttle the video deck while shuttling 24-track."

With four Lynx modules it is possible to synchronize four different tape transports: for example, mixdown with picture to two different mastering machines. Up to 16 Lynx modules can be cascaded, however, enabling synchronization of up to *16* different audio/video transports. Thus, by renting extra modules, Studio B can expand its MIDI/timecode/synchronization facilities fourfold, if a project so requires.

A substantial portion of Studio B's outboard signal processing gear is also tied to the facility's MIDI/SMPTE network. The outboard effects provided in the room include a Yamaha REV-7 digital reverb and Lexicon PCM-70 DDLs, all of which feature MIDI program-shift.

According to Seufert, MIDI control of signal processors is useful: "I find I'm doing things like changing from a non-linear reverb sound for the drums during the verse, to a bigger chamber sound during the chorus—both on the same piece of outboard equipment. This basically saves the necessity of having a second piece of gear set up. You can change reverb or DDL programs automatically through MIDI patch-shift data.

"And, of course, you can program things so that the patch-shifts on your synthesizers correspond with the appropriate program shifts on your effects units."

For additional effects, several tie lines connect Studio B with the effects rack in Image Studio A, which provides access to such additional gear as an AMS DMX 15-80S digital delay, and Dolby noise reduction units. Beyond this, Studio A's large tracking room and a second large, highly ambient warehouse area at Image provide options for recording instruments in a variety of acoustic settings. With all these areas located under the same roof, the studio is able to handle overdub work that might have gone out-of-house in a one-room, "synthesizers-only" facility.

In an era when MIDI synthesizer equipment and timecode-based audio/video work are often twin areas of perplexity for clients, a pair of interrelated facilities such as Composer's Services and Image Studio B can provide a much-needed element of continuity as a film, TV or record project moves through its various stages toward completion. R·E/P

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The Philadelphia Orchestra's use of an on-stage sound reinforcement system probably represents a symphonic first.

Electronics & the Symphony Orchestra

By David Scheirman

Since his appointment in 1980 as music director of the Philadelphia Orchestra, conductor Riccardo Muti has been introducing some new ideas to symphonic orchestral music. Muti conducts the major opera companies of the world, including La Scala, the Vienna State Opera, the Bavarian State Opera and the Royal Opera at Covent Garden, London.

During the 1985 season, the percussion section of the Philadelphia Orchestra began using an on-stage sound reinforcement system to reproduce special effects for that orchestra's own special series of concerts that featured opera themes.

Today, composers are exploring new combinations of woodwind, percussion, stringed and electronic musical instruments. Riccardo Muti, who was reported in a recent interview as saying that ". . . we may be seeing the first light in a dark night—the beginning of the Future of Music," has taken an interesting new step in that direction by allowing a loudspeaker system to actually coexist on stage with the performing musicians.

Rather than forming a house sound-reinforcement system for the entire orchestra, the new development comprises a dedicated system used only by the percussion section. Two mirror-image loudspeaker cabinets, along with mixing and signal processing gear, have become a regular part of the instrumentation used by the orchestra (Figure 1).

Speaker systems were designed and built by Kenton Forsythe, director of engineering for Eastern Acoustic Works. The cabinets, dubbed the FR283AM (the AM standing for Academy of Music, home of the Philadelphia Orchestra), were part of a portable system that was sold and assembled by TekCom, one of Philadelphia's leading pro audio dealers.

Design considerations

Portability and ease of operation by orchestral musicians were the primary considerations. Program input material includes taped sound effects, a digital sampling keyboard and a percussion microphone. The system's end users originally had considered using horn-loaded speaker cabinets.

"When this project first came up, it intrigued me," says EAW's Kenton Forsythe. "In reviewing our preliminary data on the system, I felt it was imperative that the optimum speaker design would be based around a direct-radiating concept. I remember thinking to myself, 'Over my dead body will the Philadelphia Orchestra have horn-loaded speakers on stage,' because I felt that direct-radiating units would be much more appropriate for the intended use."

To simplify setup and operation, power amplifier modules were installed inside the loudspeaker cabinets, yet another reason why direct-radiating enclosures were chosen.

"The internal measurements of bass horns are *very* critical for achieving optimum results," Forsythe explains. "With vented boxes, it is a simple matter of allowing enough internal cubic volume to include the power modules. You don't have to worry about the internal geometry of the horn."

Louis Maresca, a partner in TekCom, recalls that ease of operation was the primary design consideration. "This had to be a very capable sound system that would be set up quickly and used by the orchestra's percussionists," he says. "There was to be no operating technician. The internal power section and crossover made the most sense in terms of packaging."

Maresca found that the musicians were already sold on the idea of using the system; TekCom's major function in the project was to determine the best way to meet the customer's requirements.

Figure 1. A pair of mirror-image, full-bandwidth speaker enclosures with integral power amplifiers from the heart of the percussion section's new sound system.

Figure 2. The compact, portable sound system for the Philadelphia Orchestra includes road cases to protect an Otari tape deck and electronics rack.

Figure 3. Percussionists Alan Abel (left) and Michael Bookspan were instrumental in procuring the sound system for the orchestra. The musicians operate the system during performances without the aid of technicians.

Figure 4. Front and side view of the EAW FR283AM cabinets.

"The percussionists were definitely in favor of using their own sound system to achieve consistent results," he says. "We set up an appointment to demonstrate some signal processing and a speaker system. When Riccardo Muti heard what could be done, he was quick to recognize the creative abilities of a synthesizer and a speaker system. It was then up to us to put together a portable package that was easy to use."

Mixer and signal processing

TekCom chose a Biamp model 683 mixing console, mounted in a custom electronics rack along with an Orban model 622B parametric equalizer for program equalization and a dbx model 160X compressor-limiter. A Yamaha R-1000 digital reverb was also included, along with a Sennheiser MD-421U microphone and an Otari MX-5050-B2 tape deck for program input (Figure 2).

"We had already done two recording systems for the Philadelphia Orchestra and the Academy of Music," says Maresca. "We often provided equipment on a loan basis for various needs, so we were familiar with their situation. The percussion section was often called upon to produce sounds for the opera themes that just could not be obtained from their standard instruments. The idea of the sound system has been to give them a consistent method for recreating effects such as door knocks, thunderclaps and wind noises. They use a variety of methods to produce those effects. The equipment rack is intended to give them a versatile, simple-to-use program mixing tool."

Musicians' involvement

Percussionists Michael Bookspan and Alan Abel operate the new sound system. Bookspán, the Philadelphia Orchestra's principal percussion, is a graduate of the Juilliard School of Music, and teaches at the Philadelphia College of the Performing Arts and the Curtis Institute of Music. Abel brings extensive experience to the orchestra as well, and also currently teaches in the Philadelphia area (Figure 3).

Both men were instrumental in the process that led to incorporating an audio system into the Philadelphia Orchestra's instrumentation. "The Opera Series started it," Abel says. "We had to come up with a method for producing the same sounds every performance, no matter what concert hall we were appearing in. In the past, we once relied on a house sound system to try to get a certain thunder effect. The installed sound system had limited bandwidth capabilities, since it was intended for voice-only use. There was not sufficient low frequencies available, and the location of the speaker cluster made the sound source less than optimum. Muti wanted us to be able to have the sounds issue from our own section. Looking for a sound system, and a means to consistently reproduce our sounds, was the next logical step."

Abel, who teaches at Temple University, was impressed by an Ensoniq Mirage digital sampling keyboard used there by a colleague. "We decided to attempt to use the Ensoniq for creating a specific thunder effect," he explains. "In 'Rigoletto,' Muti wanted to have something that sounded immediate...a brisk thunderclap that was not just a gradual rolling-type sound. We felt that we could get the effect and perform it on the Ensoniq, and that required a sound system."

Abel found examples of thunderclaps on several different sound effects records, and a master tape was made that included about 20 different variations. "We went to some friends in broadcasting and used their Sony digital processor," says Abel. "We put it all on an Otari deck, and then took digital samples. The Ensoniq lab used its Apple computer to 'shape' the thunder sound, and we narrowed it down to nine or 10

different possibilities. We played them for Muti, and eventually we found what we were looking for."

The resulting three- and four-second thunderclaps are stored in the Ensoniq Mirage keyboard, and played through the sound system on cue. "Using the reel-to-reel tape was not as reliable, and we didn't have time to always be cuing it up," Abel says. "The sampling keyboard gives us excellent, *consistent* results."

Loudspeaker system

Measuring 45½ inches high and 29 inches wide, the Eastern Acoustic Works RF283AM units require only program input from the mixer (Figure 4). The portable, full-bandwidth boxes house an AB Systems model 9130 Triamplifier in each. Handles and wheels make the enclosures easily manageable.

Internal contour networks help match the individual drivers to the corresponding amplifier section (Figure 5). Each unit is capable of delivering a maximum sound pressure level of 132dB, measured at one meter.

The enclosure houses two RCF LAB Series L18/651 18-inch speakers; two RCF L10/528 10-inch, polylaminated cone drivers; and a JBL model 2425 driver mounted on an RCF H2009 horn. Five hundred watts is supplied to the pair of 18-inch speakers, 200W for the pair of 10-inch speakers, and 125W for the compression driver.

"The concept of the AB Systems Triamplifier is very interesting," Forsythe says. "It is a great space-saver and helps to make a fairly sophisticated, high-output sound system be easy to use without taking up a lot of space for a crossover and amplifiers."

All Advanced Technology Design amplifiers are designed for mounting in standard 19-inch equipment racks. The basis for the units used in the FR283AM project is the model 7132, which includes a frequency-dividing network, variable equalization filters and three separate power amplifier sections all in a 5¼-inch high package. The power modules were installed in the back of each speaker unit in housings designed by TekCom consultant Jesse Klapholz (Figure 6). The ac disconnect, Cannon connector for program input and two cooling fans are integral parts of the speaker cabinet.

"Design projects like this one are a custom service that we offer to our customers," says Kenneth Berger, president of Eastern Acoustic Works. "On this project, there were a lot of people involved for more reasons than just the sale of some sound equipment; these were totally *custom* devices. We got involved not for profit, but to help advance the state of a particular art, in this case, orchestral music."

The FR283AM units have a quoted frequency response that is virtually flat, with less than 1dB variation from 70Hz to 10kHz, and less than half a decibel variation from 90Hz to 8kHz.

"I don't particularly like working with third-octave measurement devices," Forsythe says. "That only gives you part of the picture. A swept-sinewave test signal enables us to really zero in on specific

THE ADVANTAGES OF A STUDIO CONDENSER WITHOUT A SOUND OF ITS OWN

For all of its virtues, the typical studio condenser imparts a definite character to any recording. These impositions are often considered inevitable technical imperfections: accepted, ignored or tolerated by audio engineers.

Characteristic anomalies of condenser performance such as exaggerated high end response or distortion have even been rationalized as compensation for the high frequency losses inherent in typical analog formats. Nowadays, however, they are increasingly viewed as unnecessary intrusions in critical analog and digital recording situations.

A Condenser For The Digital Era: The Difference is Nothing. The increased dynamic range of digital recording is perfectly complemented by the self-effacing nature of the MC 740. The microphone is virtually inaudible. No coloration, no self-noise — no sonic footprint, not even a fingerprint. All

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The MC 740's freedom from exaggerated sibilance or graininess and its greatly reduced distortion are immediately apparent to critical listeners. European and American engineers have already commented on the startling accuracy of the 740, and the way it reveals the subtle differences between instru-

ments and ambient environments.

Accuracy And Versatility Without Compromise. Uniform (< 2 dB: from actual machine specs, not just published specs) frequency response curves for all five polar patterns may seem a remarkable breakthrough. To Beyer, this is simply a design criterion for the microphone. Similarly, there is no contradiction in the fact that the 740 is exceptionally sensitive, yet also withstands extreme SPLs (up to 144 dB with the 10 dB attenuator in circuit).

Hear What You Could Be Missing. The MC 740's unconventional design offers a clear alternative. The best way to evaluate the difference the MC 740 can make is to work with it in your studio. To arrange a hands-on audition of this remarkable new audio instrument, contact your Beyer dealer or write us: Beyer Dynamic Inc., 5-05 Burns Avenue, Hicksville NY 11801.

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Circle (39) on Rapid Facts Card

Figure 5. *Internal signal flowchart for the FR283AM enclosures.*
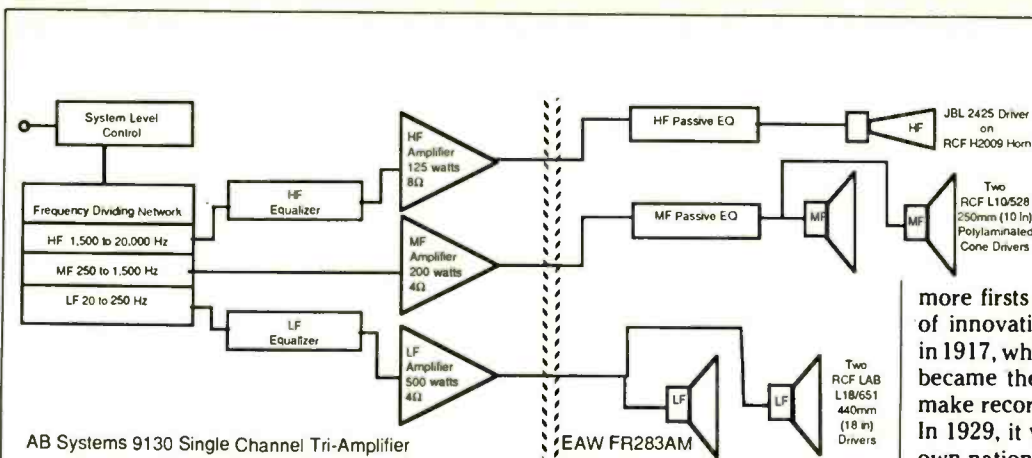
AB Systems 9130 Single Channel Tri-Amplifier

Figure 6. *An integral AB Systems power amplifier section is incorporated into the FM283AM cabinets. An ac disconnect and a pair of cooling fans are also included.*
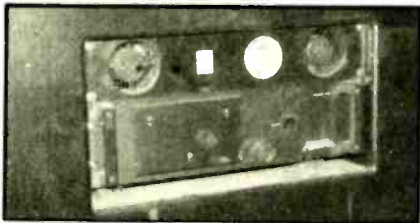
frequencies, but that method takes some working with to get used to. We do look at time relationships within the system— that shows up in the way the response of the whole system shows up on the chart plotter."

Forsythe noted that paying careful attention to the initial engineering and then carefully listening to the resultant designs has given him consistently good results.

Project significance

The Philadelphia Orchestra is arguably one of the finest orchestras in the world. Under the musical direction of Riccardo Muti, the orchestra has begun to rack up more firsts in what has been a long line of innovative developments that began in 1917, when the Philadelphia Orchestra became the first American orchestra to make recordings with its own conductor. In 1929, it was the first to perform on its own national radio broadcasts; in 1948 it was the first to be featured on national television. As if that weren't enough, in 1973 the Philadelphia Orchestra was the first orchestra to go to the People's Republic of China.

This may be the first instance of a symphony orchestra incorporating an audio system into its instrumentation. Although the system is currently being used only for sound effects during the performance of opera themes, the precedent has been set: A sound system is sharing the stage with traditional classical music instruments.

"There have perhaps been others who have used an audio system for special effects at one time or another, such as to simulate the sound of thunder," says percussionist Alan Abel. "As far as I know, however, the Philadelphia Orchestra is the first to purchase and maintain a sound system as a regular part of the stage instrumentation."

Under the direction of Riccardo Muti, the orchestra has used its new sound system for events held away from Philadelphia's Academy of Music, including special shows at Carnegie Hall and the Kennedy Center for the Performing Arts, New York.

A spring tour is underway of 15 cities, including an appearance at the World Exposition in Vancouver, British Columbia, and dates in Chicago, Montreal and Los Angeles.

The percussion section's new sound system may not be in use on the early 1986 tour, but the gate has been opened for the implementation of new music reproduction techniques in orchestral music. "This system enables the percussionists to get their sound effects out over the rest of the music," said EAW's Kenneth Berger. "It is now an accepted part of their instrumentation. The creative possibilities now open to modern composers are very interesting to ponder." **R·E·P**

Mastering Feature Films for Video Release

By Robert Bradford

Consumers' growing demand for high quality sound makes film-to-video soundtrack remastering more critical than ever.

Given the ever-increasing quality of consumer VCRs, the increasing consumer sophistication and the special limitations of presenting entertainment in the home, profound changes are taking place in the new generation of facilities that produce video product for home use. And nowhere are these changes more profound than in the mastering houses that adapt feature films to match the presentation demands of the domestic viewing audience.

The complex process of audio mastering in the film, video and music recording industries can be defined as the specialized technique whereby a master mix is re-recorded for the purpose of generating a new master. This is often for a different medium or for a special-purpose presentation.

In the music industry, for example, mastering traditionally has been the process of re-recording the final release master of an album or single to a lacquer disc. The lacquer master then becomes the first step in a process of plating and pressing that eventually yields the final consumer product, a vinyl record. Recording audio onto a lacquer disk places some stringent demands on the incoming program material, including the control of peak amplitude and phase information. In order to make a product that can be mechanically reproduced as well as aesthetically pleasing, disc-mastering engineers *translate* the program material into the new format—the lacquer disk—by controlling those parameters they know will cause problems during the cutting process. It is this subjective messaging of the program material that creates a requirement for specialized mastering engineers.

As new mediums come into widespread use, new mastering services become necessary. With the introduction of the Compact Disc, new techniques were needed to take advantage of the CD's substantial increase in practical dynamic range and signal-to-noise ratio. As a result, mastering labs, whose primary business had been the cutting of lacquers, expanded their services to meet these new requirements.

Growing demand for high quality sound

Just as a mastering room forms the interface between the recording studio and the record-releasing industry, a film-transfer facility represents the link between the motion picture industry and the home video releasing industry. The film transfer, or telecine, process is the stage during which a film's master picture elements are transferred to videotape to generate a film transfer master. Traditionally, a film's soundtrack has simply been dubbed (transferred) from the soundtrack element to videotape at the same time the picture was being mastered, with the sound given no special care or attention. That situation is changing, however; an increasing number of film transfer facilities are now using their sweetening bays to provide special attention to the soundtrack.

One such company with which this writer is particularly familiar is Crest Video, Hollywood. Crest's audio department handles the mastering of feature films to videotape, primarily for home video, cable and broadcast use. With the ever-increasing demand for video product, the facility's audio department has grown, since its inception five years ago, into a 24-hour-a-day, six- or seven-day-a-week operation.

The reason that cost-conscious clients have begun spending the money for premium audio-mastering services is because of the growing demand for *quality* sound. This change has been catalyzed by the recent introduction of high-fidelity stereo sound on Laserdisc and VHS and Beta Hi-Fi videocassette.

Special problems in film-to-video mastering

Because mastering is the last step before mass release, it brings with it a special responsibility. Although a mandatory function of mastering engineers is to skillfully adapt the program material to conform to the technical specifications of the new format, their real concern lies in accomplishing this task while preserving the many subtle, subjective internal sound balances of a mix. Except in a few special cases, it is not the mastering engineer's job to attempt to superimpose an improved mix onto the one supplied; rather it is to translate, as transparently and elegantly as possible, the artistry of the mixers on the re-recording stage (or studio).

Although all forms of mastering are demanding, the mastering of feature films for video release presents a number of unique challenges, derived mostly from the obvious fact that feature film soundtracks are designed and mixed for presentation in movie theaters under carefully prescribed conditions. Although I think that any mix that is sonically and dynamically well balanced—where the mixer employed com-

Remastering MGM's Classic: Gone With the Wind

It all began in a salt mine somewhere in Kansas, the one that MGM had converted into a maximum security storage facility for archival film elements.

One day a vault operator came across a mysterious box. What was supposed to have been outakes and trailers turned out to be a virgin negative of Gone With The Wind.

The negative was presumed to have been made at the same time as the printing masters for the film's original release in 1939 and had been untouched since that time.

This discovery was a golden find for MGM, which had just negotiated repurchase of the home video rights from CBS, the company to which the rights previously had been sold. The reason MGM was so excited about the discovery of a virgin negative was that all existing film elements have been so damaged by the wear and tear of printing and handling down through the years that, had it been a lesser film, some of them would have been considered unusable. The company would now be able to make a new film transfer that would look better than any ever made.

But what to do with the soundtrack? Any film this old, regardless of its condition, is going to suffer from the limitations of soundtrack recording technology in use at the time. Add the usual pops, ticks, whirrs and crunches endemic to optical soundtracks that have been played even a few times, and it becomes clear that the track would not have done justice to the picture quality.

Peter Anderson, vice president of technical operations for MGM Home Video, New York, was acutely aware of the problem. He knew that because the film was going to be digitally duplicated in both VHS and Beta Hi-Fi formats with MGM's Videophonic process, consumers would be expecting a high quality soundtrack.

He immediately called Jon Truckenmiller at Crest Video to arrange for the picture transfer and to ask about restoration of the soundtrack. Jon asked me if I knew of any new technologies that could be used to improve the audio quality.

I had heard of several people working in the area of experimental computer enhancement techniques, similar to those developed by NASA to clean up video pictures sent back from space. Having tracked down the developers of several different systems, I asked them about the practicality and expense of using such technology for Gone With The Wind.

Although I discovered that it might be possible to use these techniques on the soundtrack, such processes would probably cost several hundred thousand dollars and might take months to complete.

I then remembered a conversation I'd had a couple of years ago with Stephen St. Croix, president of Lightning Studios, a division of Marshall Electronic, during which he discussed the possibility of using logic techniques to discriminate extraneous noise from program material.

When I presented him with the problem, and the time frame involved, he said, "Yeah, probably." It turned out that Steve had just received the first prototype chips for a proprietary audio restoration process he was developing, and which he labeled "Revectorization."

After working around the clock for a week to assemble his prototype Revectorization processor, Steve showed up at Crest with a surprisingly innocuous looking piece of equipment.

He set it up, adjusted a few controls, flipped a switch. It worked.

The noise and distortion disappeared; the processed audio sounded like a new recording made on equipment of that day. The unprocessed audio sounded like that same recording, but with a grit, scratch, pop and noise loop mixed in at a ferocious level. The only other capability I could have hoped for was a way to restore full-frequency bandwidth.

With the noise eliminated—apart from some slight residual grittiness in a few passages that had been virtually pure distortion—it was now possible to use conventional equalizers and enhancers to restore a more natural, overall balance.

So Steve, John Pooley, a fellow audio engineer at Crest, and I strapped ourselves in for a week-long, around-the-clock session to restore the film's soundtrack.

Stereo enhancement techniques

The second part of the assignment was to perform a tasteful stereo enhancement. I am not a believer in conventional fake stereo; for the most part, I would rather hear a properly sweetened mono track than be subjected to the motion sickness and tugging-at-your-ears sensations that can be generated by typical auto-panning, comb-filter circuitry.

In this respect, we were fortunate in having two new pieces of equipment for the session. One was a designer's prototype of the Studio Technologies Stereo Simulator, and the other a Quantec QRS Room Simulator. Our goal in using both was not to produce fake stereo per se, but to emulate the acoustics of real rooms, taking our scene cues from the picture.

The Studio Technologies Stereo Simulator was used to provide varying degrees of stereo spread, and the Quantec QRS to create realistic acoustic spaces.

The prime difference between this type of processing and conventional fake stereo is that what we were adding was essentially real acoustic information. (During subsequent experiments, I found that if I listened in quad using the QRS' four discrete outputs, I could synthesize rooms so realistically that listeners invited into the control room were unaware of the added spatial dimension, except by its absence when the unit was switched out of circuit.)

Because no currently available stereo synthesizer is able to recognize, decode and discretely position actual sound generating sources—such as a person talking, or one kind of musical instrument from an ensemble—the way to make synthesized stereo most effective is to change the simulated acoustic spaces on a scene-to-scene basis, with deference to picture cues and good taste.

Which is precisely what we did for Gone With The Wind. I consider it a compliment when people who have heard the VHS or Beta HiFi home video release say to me, "Sure, it sounds great. But what was I supposed to be listening for?"

I am please to answer: "Just Gone With The Wind."

Photography by Elizabeth Annas

Crest Video's audio-for-video mastering room uses Ampex ATR-102 2-track and -104 4-track, plus an MM 1200 16-track (shown here with 1-inch 8-track head blocks). The room's outboard equipment includes (from top right) an Eventide Harmonizer, Audio+Design Scamp Rack, Dolby 360 units (for use with an outboard CAT. 43 Cinema Processor), Quantec Room Simulator, control panel for an EECO MQS-100 timecode synchronizer, plus custom designed remotes for multitrack and mastering machines.

mon sense and a good working knowledge of acoustics and psychoacoustics—will sound great when reproduced on just about any medium, some effects that sound sensational in theatrical presentation can cause problems in the home.

In a theater, for instance, the dynamic range between stereo crickets and thunderclaps approaches that of real life; the rumble of a train can literally rattle your teeth. But few television receivers include loudspeakers that can rattle more than the neighbor's nerves, and then only if the sound has made it out of the broadcast compression chain—that train rumble may simply disappear as a result of the processing, and take the rest of the sound with it.

The point is that soundtracks prepared for auditioning in the home must successfully accommodate an incredible range, from the worst-case 4-inch TV speaker in a noisy apartment to the relatively sophisticated systems in home screening rooms.

How To Hear Yourself

Getting enough vocals on stage. It's a problem facing every performer. In fact, most smaller bands simply can't hear the vocals clearly when they perform.

If your monitors simply don't cut it, you're probably wandering what to do about it. Well, EAW has a suggestion.

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The SM202P is built around two RCF PRO Series 250mm (10 inch) drivers and a RCF high technology compression driver / horn. And, it includes an advanced Forsythe designed crossover assembly with asymmetrical slopes providing response that is within +- 2 dB over the entire vocal band, and that's without any equalization.

The SM202P's Output Capabilities Swamps The Competition, Too.

The use of two high efficiency 250 mm (10 inch) drivers provides considerably higher efficiency than any design based

on a single 12 or 15 inch driver. The SM202P's 103 dB sensitivity is all usable as there are no peaks to create misleading high specs and lots of feedback as in all other monitors in this price category. And, don't forget the smaller 10-inch drivers provide smoother response above 800 Hz where the main components of vocals are located.

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There's a lot more to the SM202P story than we can talk about in an advertisement. And, it's just one of many EAW solutions to today's sound reproduction problems.

For more information and the name of your local dealer, call EAW right now at 617-620-1478.
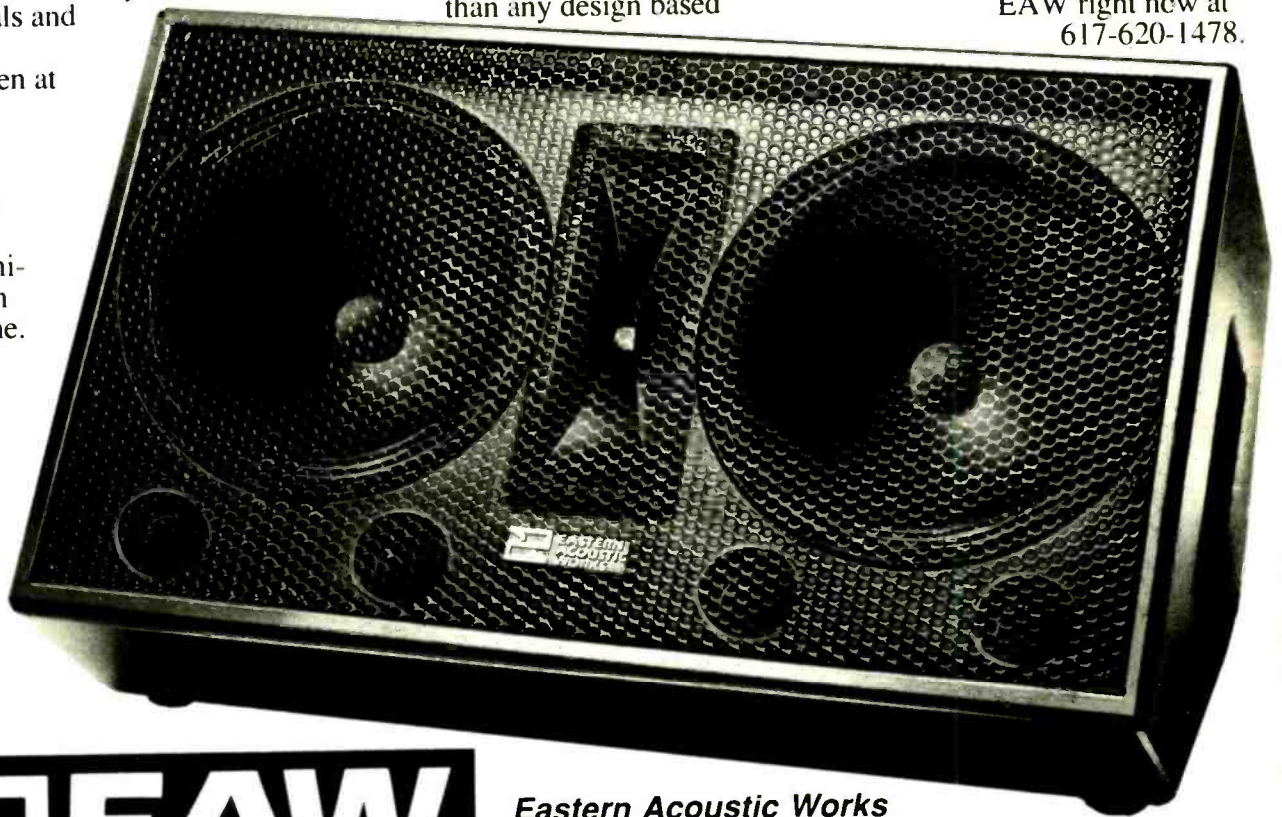
If the typical video post-production house has a vice, it is one of *haste*. Because of the high operating costs involved, everything in video is designed for speed. Nothing is wrong with speed *per se*, but we need to understand that a film soundtrack has taken many people months to prepare. Virtually nothing is present in a soundtrack that was not designed and mixed in during the ADR, Foley, effects, scoring and re-recording processes. Consequently, the challenge becomes one of enabling those sounds that may be barely perceptible in a movie theater to be clearly heard in the home, while keeping them in a believable but controlled relationship to the loudest material.

Understanding source elements

Anyone involved with the mastering of film soundtracks knows that one of the most insidious aspects of the job can be the brute force length of feature films. Unlike most other forms of record-ing—where a mistake may cause several minutes of redo time—having to repeat a feature film mastering session can involve hours or even days. Good mastering engineers rapidly develop a scientific, methodical approach to undertaking a project, much as a pilot uses a preflight checklist.

The first order of business is to calibrate the playback equipment. Even if the mixer is lucky enough to have a competent and friendly tech staff, it is the mixer's responsibility to ensure that the equipment is both working correctly and is properly calibrated. Because nearly all feature film soundtracks arrive at the sweetening bay on 35mm or 16mm magnetic or optical sprocketed film, calibrating the equipment is the first place to start.

Correctly setting the head azimuth and equalization for each reel of film is absolutely critical. Complicating the matter is the fact that, throughout the history of film, there have been many equalization standards and special processes. Often some research is necessary to discover the special circumstances under which a given film may have been made, even for some relatively recent releases.

Setting the correct replay level is not always a simple matter either, particularly when working with film soundtracks that have been noise-reduced. Also, the peak levels often found in film program material may cause clipping of the reproduce amplifiers, noise reduction units and other downstream equipment. *Never assume anything, always check everything and listen, listen, listen.* If, after you are set up, it doesn't sound right, then it probably isn't.

Editing and conforming

Two things must happen simultaneously during the soundtrack mastering process: The audio must be tailored to meet the technical requirements of the home video and broadcast media and the soundtrack must be conformed to the

video, not only to ensure perfect synchronization but also to exactly match the audio edits with the video edits.

Apart from the obvious reason that edits need to be made in order to join film reels together into a continuous video program, other things can happen during the film transfer process that require edits. For example, the film's main and end titles may be supplied on separate reels, particularly for foreign-language dubbed versions; there may also be short film insert *cut-ins* to replace pertinent newspaper headlines and such. And there may be sections in older films where the film element itself is in such bad condition that short portions of it cannot be used. In all of these cases, the telecine operator is forced to make additional edits in the picture.

To ensure perfect audio/video synchronization, precisely the *same* edits must be made in the soundtrack. The plot thickens, as they say, when these forced edits occur in the middle of a music cue, or in the middle of a word. It then becomes the audio mastering engineer's job not only to match the cuts but also, wherever possible, to make restorations. In most cases where there is sufficient music in the clear, it is possible to dub off a portion of the soundtrack to a timecoded four-track tape, and edit a new, small section of music to bridge the hole.

Fortunately, at Crest Video, I had an EECO MQS-100 synchronizer that allowed me to *flywheel* over timecode discontinuities and still maintain phase lock, a capability that enables razor-blade editing of timecode striped tape. Such an option is helpful in cases when there is a need to reconstruct dialogue to replace missing words or parts of words. Using the same technique, clicks, pops and crunches can also be removed virtually without a trace.

The degree to which these techniques are used for any given project depends on the relative severity and extent of the problems. It also depends on how much the client is willing to pay to have this sort of fine *audio detailing* performed. For an old print, where the soundtrack may be severely damaged and several optical generations away from the master, the expense may be prohibitive. In special cases, however, it is indeed worth the effort, as with the extensive processing used for the recent stereo videocassette re-release of MGM's classic movie, *Gone With The Wind*.

Another instance where editing becomes necessary can occur when the end title music does not end, but becomes walkout music: the music level may drop to a background level at the same instant that the picture fades from the screen. It is much better to go to the trouble of properly editing the end title music, rather than replicating some of the terrifying fades I have heard on certain video renditions. The end of the film is like the edge of a building: It is where form meets space. Attention to detail here can reinforce the impact of the entire film.

Synchronization considerations

Before we proceed to the second function of the mastering process, that of signal processing techniques, we need to spend as brief a time as possible with that loathsome gremlin, sync. Let me state the issue as a question and answer:

When is real time not real time?

When relating some video speeds to some film speeds, sometimes!

All of this temporal confusion began when the U.S. television standard, which was designed originally for a monochrome signal, was forced to conjure up a color picture. Until then, the frame rate of a TV picture (30fps) and the film frame rate (24fps) both had a nice even mathematical relationship to one another, and both resolved to 60Hz. When color television arrived, however, because of the mathematical timing changes needed to make color work in a black-and-white system, the color frame rate changed to slightly less than 30fps; hence our standardized color frame rate of 29.97fps resolves to a scanning frequency of 59.94Hz.

What all of this means for the mastering engineer is that, in order for film hardware to run in sync with video, it *must* resolve to video frequencies and not 60Hz. For this reason, in the majority of video houses, all equipment is synchronized to a central house sync generator, which guarantees that *all* hardware runs at precisely the same speed.

If you were to plug a 35mm film dubber into the wall, and let it resolve to the 60Hz line frequency—which would give you very close, if not perfect, resolution to other film hardware running at cine speeds—you would encounter a slow drift compared with an identical machine resolving to video sync.

As a result, the mastering engineer must be *extremely* wary of working with sound and picture elements in video formats that were generated at non-video houses. For instance, a client may do you the favor of having a film soundtrack pre-laid to a timecoded, ½-inch four-track tape. Timecode is an absolute, right? It either syncs or it doesn't, right? Well, yes *and* no.

If the facility that handled the pre-lay was running its film dubber at a 60Hz resolve frequency, but locking its timecode generator to a 59.94Hz video sync reference, the resultant tape is guaranteed *not* to sync with the videotape.

Then, of course, there is also the case of elements that were not resolved to anything, but were simply running wild. (Video machines, particularly the ubiquitous ¾-inch U-Matic deck, can also run off speed.)

Most competent video facilities, particularly those that do a great volume of work for overseas markets, not only have ways of working in both the European PAL and U.S. NTSC video formats and frame rates, but can convert between them and also correct many of the commonly encountered errors.

Signal processing: the key to mastering

To recap: The 35mm mag dubber is calibrated, and the film soundtrack element ready for playback. You are familiar with the format in which the film was presented, and know how to mix it properly, if need be.

You are aware that the film soundtrack reels are going to have to be synchronized with the picture and that all the picture edits made during the film transfer process are going to have to be match-cut. You are prepared to repair holes and remove gross rude noises with either electronic or razor-blade editing.

The key to ensuring that feature films sound effective in the home is to concentrate on the *total* listening circumstance and not so much on the hardware. The typical home video listening room is the virtual opposite of a movie theater equipped, for example, with a LucasFilm THX sound system. Yet the techniques that make for a good-sounding theatrical mix are the same ones that make for a good home video master. The requirement is to establish a consistent, representative listening environment in the mastering room.

As most **RE/P** readers will already be aware, feature film re-recording engineers work on a stage whose monitoring system is sufficiently like that found in an average movie theater to guarantee that the mix will sound good in just about all of them, but accurate enough to allow the mixer to hear the

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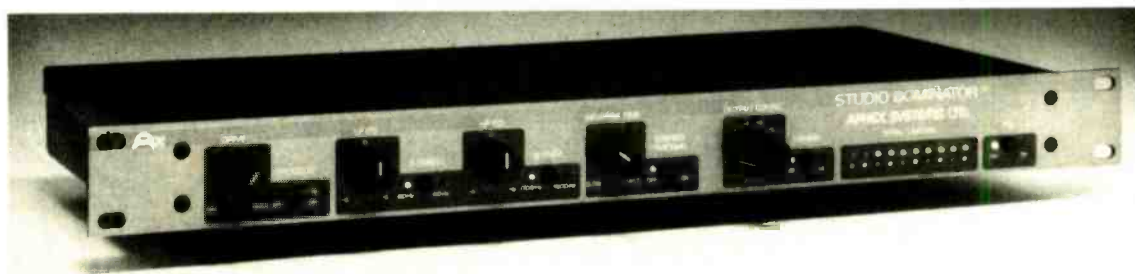
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program nuances without strain or fatigue. In addition, the monitoring level is calibrated to accurately represent the dynamic range available in a typical theater.

Applying these criteria to mixing for the home, I selected a good set of close-field loudspeakers—Calibration Standard Instruments MDM-4 Near Field units—and monitored at a level of 75dB/c, slow per channel. Although this level is substantially lower than the theatrical specification, it is much more representative of a home listening environment. [Dolby Stereo theatrical playback level is specified as 85dB/c, slow, per channel, for 50% modulation—*Editor.*]

A lower monitoring level helps compensate for the fairly high background noise level found in many homes, and guarantees that ambiences will be heard on all but the worst TV sets. The 75dB/c level also seems to be just about right to compensate for the fact that although the frequency response and distribution patterns of my monitors are fairly typical of

midline home speakers, they are much more accurate. And, because extremely phase-accurate speakers enhance the effects of stereo synthesis and room simulation, listening at a slightly lower level helps to make these effects sound more like they would in the home.

Before each session, I use pink noise to calibrate the channel balance and gain of each speaker. I also do this each time I change master recorders. By always monitoring the output of the mastering machine, I can catch any problems in the signal path.

The range of available processing equipment falls into the following basic functional categories:

Dynamic range processors: As mentioned earlier, except in a few special cases the job of the mastering engineer is to adapt, not alter, the mix presented on the film's soundtrack. Ideally, I aim for the film soundtrack to sound to the home video consumer *exactly* as it did in the movie theater; practically speaking, one hopes at least to convey the same overall

impact and impressions.

Dynamic range is usually the single most important parameter to be controlled. With typical home ambient noise levels ranging somewhere between 35dB and 60dB SPL, and a neighbor threshold trigger level of somewhere around 85dB, the usable program dynamic range in a domestic environment is something like 30dB (or less) for broadband sounds that are to be clearly heard.

Yet I have encountered peak program levels on feature film soundtracks that measure as much as 28dB above the calibration tone level. The reason such peak levels are possible is that mag-film oxide is much thicker than that of audio tape. Also, because of the thickness of the film backing material, high levels are much less likely to print through on film than with tape.

As a result, some film mixers make no attempt to control the peak level of sounds such as thunderclaps and crashes. The saturation level of the mag film and the clipping level of the theater

amplifiers are often what determine the final sound pressure level of a screening.

Fortunately, most of the program material that makes up the body of a film soundtrack—normal dialogue, music and effects—falls within the normal listening level range. If you can now contain the really loud sounds to the nominal maximum recording level of the tape, allow the normal DME (dialogue/music/effects) program material to go through relatively unscathed, and increase the level of the very soft ambiences so that they can be heard—and do all of this in such a way that the listener is unaware of any tampering with the sound—you will have achieved your goal in dynamic range processing.

There are definitely ways *not* to handle this task. One is to keep increasing the gain reduction of a limiter until peaks hit zero on tape, and the soft sounds are audible. It has been my experience that such an approach will cause about 8dB of constant compression on the program material, which will be noticeable and

result in an ugly sounding track.

The other incorrect way to control dynamic range is to keep increasing the gain during quiet passages, and then whack the fader down just as you hear a thunderclap. Not only will this sound unspeakably awful, but it will actually *increase* the dynamic range of the film, not control it.

A much better way of controlling dynamic range is by using a good, natural sounding peak limiter to reduce the level of really loud sound and then let the normal DME program remain as it is. If you see a limiter hitting constantly during normal loudness dialogue passages, there is too much compression. On the other hand, if crashes are still pegging the meters after limiting is applied, there is insufficient dynamic range control.

The Dolby Cat. 43 is an extremely useful—albeit little-known—piece of equipment that substantially can solve the problem on the other end of the level spectrum: how to allow soft ambiences to

be heard without obvious gain riding. I suspect the reason the Cat. 43 is so anonymous outside of film re-recording circles is that Dolby Laboratories understands the unit can be an absolutely lethal weapon in the wrong hands. Why? Because, unlike reverberation units, where it is usually obvious if the process is being abused, the effects of a Cat. 43 can be so subtle that using it can be like driving into a blind curve—you don't know you're going too fast until you're through the guard rail.

The device is referred to as a cinema processor on its front panel because Dolby intended it, I think, as a special-purpose device for cleaning up problematic film stems. The company knew it would be in the hands of mixers intimately acquainted with the hazards of processing subtle sounds over long periods of time. I've risked mentioning this device because I believe the Cat. 43 to be an absolutely irreplaceable piece of equipment to anyone working with film sound. (I won't even mention the incredi-

ble things it can do in music recording and mixing.)

The Cat. 43 allows a Cat. 22 card found in a normal Dolby 360 or 361 Type-A NR unit to be used as a single-ended, four-band noise reduction and dynamic range processing device. Further explanation of the system's workings would be beyond the scope of this article, other than to say that its use enables soft sounds in a mix to be brought up by means of frequency-conscious expansion, yet without affecting the dynamic range of the normal program material in any perceptible way. It also allows pinpoint control of many problem sounds in the program, as well as providing an effective, though extremely broadband, way of controlling tape hiss in a single-ended manner.

Psychoacoustic presence enhancers, including the Aphex Aural Exciter and DOD/EXR Exciter, are enhancement systems that allow perceived clarity and brightness to be added to dull, lifeless tracks, and work most efficiently with noise-free, clean program material. Using any device capable of dramatically altering the frequency-spectrum balance of a program is best reserved for the remedial mixing category; in other words, restoring tracks that have obviously been substantially dulled somewhere along the way. And the definite rule of thumb when using such devices is: Less is more. Although new reverberation devices can clearly be used as presence enhancers, see the next category section for more details.

Perceived acoustic environment processors represent a whole new category of equipment that deals in immediate time-domain effects. Most units are capable of creating wonderful, bizarre effects, such as those heard in contemporary music. Their main use for the feature film mastering engineer is to help mask the effects of abused or inappropriate single-ended noise reduction systems at some earlier stage in the film soundtrack's life. (The worst of these single-ended NR systems, by the way, were some early, crude forms of gates, although drawbridges would more accurately describe their modus operandi.)

Many new digital environment simulators, such as the Quantec QRS Room Simulator, and the latest software versions of the Lexicon 224XL, can do breathtaking jobs of duplicating the acoustic environments seen in feature films. The key here is to remember that any use of this type of equipment should

be considered restorative in nature, and should be used only to replace what probably was there originally. And never forget that when using these devices on stereo program material, all new signal information added during the mastering process must be absolutely mono compatible, and free of noticeable artifacts.

The area of noise and distortion removal is where most work is needed.

Graphic and parametric equalizers: Other than the obvious uses for equalizers, and the warning to use them with *utmost* discretion for restorative purposes only, it may be a good idea to mention that in several older film formats, very high-level, high-frequency (usually about 10kHz), or very low-frequency (usually about 20Hz) signals were used to cause surround channels to be turned on. This situation was present in an era when film soundtracks and theatrical playback equipment were so noisy that it would be a distraction for the audience if the surround channels were left on during passages when no information was present in them. Notch filters, often connected several in a series, are necessary to remove such tones. It is much better to notch out the HF tones, than to simply bandstop all of the top-end frequencies. Notch filters are also useful for removing all sorts of other fixed-pitch garbage that manages to get into the signal.

Noise and distortion removing equipment: I have heard few gates, or dynamic noise filter devices, that are successful for sustained use during soundtrack mastering. The primary reason is that the very quiet sounds in a film track are so delicate that almost all of these devices either mask the sound entirely or, if the threshold is able to be lowered enough to allow the sound through, the unit can be heard pumping as it modulates the ambient noise level of the track. The one device I find to be consistently helpful is the Dolby Cat. 43, because the unit's thresholds can be so dramatically altered, or even tricked, and it provides very subtle control.

By all means, do try any device you think might be helpful, but remember to constantly audition the track to make sure that the problem you are treating is not isolated to one scene or sequence. If

that is indeed the case, remember to remove the processing after the problem area has passed.

The area of noise and distortion removal is where the most work remains to be done. The problem faced by such equipment is the ability to distinguish noise from program material, something with which even the ear has occasional trouble.

Pitch shifters find little use in soundtrack mastering, other than for altering the pitch of soundtracks from PAL-format transfers to their normal range. In order to make foreign format versions of domestic, 24fps films, they have to be transferred at 25fps to match the PAL frame rate. Playing back the film at the higher speed causes the audio pitch to be increased; a pitch shifter, such as the Eventide Harmonizer, can be used under computer control to reduce the pitch to its original range. Candidates for pitch shifting are evaluated on a film-by-film basis to determine if the time required to properly handle the pitch-shifting process, which might involve the making of edits to fix glitches, warrants the benefit gained by the correction.

Noise reduction systems are included here as signal processing devices only because that is exactly what they become when misused. Always listen carefully to noise-reduced tracks, particularly if the element supplied to you happens to be a Dolby-encoded videotape. It's not that there is anything wrong with the NR process for videotape, it's just that most often these tapes are made in noisy video *engine rooms* by videotape operators whose main interest is not precision audio. I have actually seen double- and triple-encoded tracks, or tapes where one channel is encoded and the other not.

Mastering many of the MGM and Samuel Goldwyn classic movies has provided me the rare opportunity to audition master mixes of many of the first true stereo recordings; the acoustic beauty of some of the early scoring recordings has been a revelation. By the same token, the sound quality and production values of many recently released films are awesome. I believe that a real contribution of high-fidelity home video is that it gives many people a way to truly appreciate the art of film. R•E/P

Bradford recently joined The Music Design Group, Hollywood, to supervise design and implementation of new studio and digital editing facilities. Previously, he was the senior mixer at Crest National Videotape, Hollywood.

Upgrading Vintage Technology

**Cherokee Studios custom console combines
elements from the past with a futuristic design.**

By Denis Degher

While both console manufacturers and engineers speculate on what the next generation of recording consoles could and should be like in terms of design and operation, Cherokee Studios, Hollywood, has built a new console that marries the best of the past and present, with a futuristic low-profile design.

The concept behind the design of the new Cherokee console evolved from years of experience in updating and refurbishing the studio's beloved Trident A-Range boards.

When Trident Studios in London changed owners a couple of years ago, Cherokee purchased the A-Range console installed in the facility's mixing suite, with the intention of modifying and upgrading it to replace the board currently gracing the control room of its Studio One. What has evolved is a completely redesigned console that retains the A-Range equalizers and mike pre-amps, and incorporates them in a new acoustical design, without a meter bridge.

When queried on why the studio decided to undertake the arduous task of building a custom console around A-Range equalizers and pre-amps, Toby Foster, chief engineer and console designer, outlines five major reasons for

"No. 1, we have had A-Range (boards) for nearly a decade. Client satisfaction, as well as our love for the sound, were both main priorities in retaining the equalizers and mic pre-amp designs; noise and distortion figures have also been improved with component changes.

"Reason No. 2 is the prohibitive cost of today's upscale consoles. With consoles in the $200,000 to $300,000 range, the thought of having to amortize a quarter of million dollars per console, times four control rooms, is staggering.

"No. 3, nothing on the market comes close to what we want sonically, and of-fers the feature that we want on a record-ing console.

"Four, the systems being manufac-tured today are only interim consoles for the next generation of desks, which will either be completely digital, or (based on) an analog signal path with complete-ly automated digital-control circuitry. Our new console will take us through the impending technological watershed, and leave us in a very strong position when the next generation is perfected.

"Reason No. 5 is that the Robbs (Cherokee owners) and myself are all recording engineers; we wanted to build a user-friendly console. We wanted it to be a very musical console, sonically and ergonomically, so that the engineer could be like another musician—freed from technical drudgery of the session, and able to offer creative input."

Console topography
The new 56-input console features 48

input channels and eight dedicated auxiliary returns, and is now housed in Studio One's redesigned control room. A total of 10 auxiliary sends per channel are routed through an aux matrix for easy assignment to outboard effects units without patching. A group muting system without VCAs (voltage-controlled amplifiers) uses transistors configured as electronic switches, the muting action being accomplished by shunting the appropriate signal to ground.

In addition, the console contains no relays, FETs or conventional switches. All routing and switching functions are controlled via transistor switches.

Another unique system design is a mobile monitor section mounted on wheels, which plugs into the console. It is also equipped with controls for level, panning and aux sends.

The console's extremely low physical profile strikes a highly futuristic visual note, as well as a sonic one. With no meter bridge protruding vertically to interrupt the sound path for monitor loudspeakers, sound waves have one less plane to influence their sonic integrity.

Instead, the group-level metering is set into the back of the console, each meter being covered with the same non-reflective Lexan surface that covers the entire console. (The concept was created by Greg Thompson Design, using a computer-assisted drafting system.)

"When we first contemplated the concept of a low-profile console, we had no idea that it would end up looking like this," says Dee Robb, studio owner. "We gave the idea to Greg (Thompson), and were totally amazed when he returned with the CAD-generated design. At first we were concerned with the viewability of the inset meters, but our tests proved that the meters could be seen from a minimum distance of one foot from the console surface."

Undeniably, the visual look of the console is an eye-catcher, but the system's internal electronics may prove to be the ultimate selling point for Cherokee.

"The console was designed around the components, and we have used the highest quality components with sealed pots and switches and all-gold connectors," Robb explains. "By designing around the components, we were able to use parts that would not have physically fit into the corresponding circuit of a standard A-Range."

Other than the modified EQ modules, every other electronic circuit in the desk is new, and was designed by Toby Foster using "990-Type" amplifier modules. Although the EQ modules retain their high- and low-frequency shelving curves, the two midrange controls are now sweepable. The LF and HF filter switches have also been separated from the EQ-insert switch, enabling them to be kept out of the circuit until needed.

One of the weakest design points of the original A-Range, Cherokee owners felt, was its grounding scheme, which was rectified by developing a new grounding system that uses passive summing amplifiers. (The major drawback of such a system, however, is that it must be designed around the precise number of

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INTO THE FUTURE

Overview of 48-input custom Cherokee console. At left is the master control section.

console channels, making its use impractical for mass-production consoles.)

The discrete console also features a highly accessible electronic cardcage layout that simplifies repairs and further updates; individual on/off switches per channel for phantom power; and a host of other bells and whistles not found on the original A-Range line.

"We are not reinventing the wheel," Foster says. "This is an exercise in packaging and application. We have taken all the features that we like on other consoles, and have incorporated them into this console.

"What we have succeeded in doing is building a console where the technical tail is not wagging the musical dog. Today we are receiving so much new technology from the aerospace industry that we have equipment being built by

Close-up of the auxiliary send section of each module (left photo), with two assignable sliders switchable between four auxiliary buses, two dedicated mono sends and a pair of stereo sends.

The master auxiliary sends (right) each have a 2-band switchable equalizer.

Using the Cherokee Console

Having worked on numerous projects in Cherokee's Studio One, I was more than familiar with the control room's acoustic design, and was a great fan of the Trident A-Range board that resided there.

As a result, it was with excitement and trepidation that I entered the refurbished control room—which featured the new custom console as its centerpiece—with members of the band Darius and the Magnets. Visually, the radical look of the ultra low-profile desk created a unique vista into the studio.

For years the masterbridge has served as a useful tool, but it wasn't until I mixed my first session on this new console that I realized what a severe psychological impediment tall meterbridges were, and what a barrier such designs created between the control room and the artist working in the studio.

Gone was the isolation; superceding it was a feeling of openess and air. Adding greatly to this feeling was the direct access to the output of the main control room monitors, which no longer confronted a large vertical plane from the floor through the meterbridge.

With all the analysis that has gone into studio and control-room design, it is amazing that more studies have not been done on the acoustical ramifications of large console designs. Manufacturers seem intent on designing their consoles from a purely electronic viewpoint, with little thought about how their product affects the coherency of the sound field that producers and engineers deal with.

As the session continued, I began to question whether my positive aural impressions were solely acoustical in nature, and considered that the perceived sonic clarity and openness probably had much to do with electronic improvements as well. Although the console retains the popular A-Range front-end and equalizers (with component upgrades) it is, for all intent and purpose, a vastly improved animal, offering many more features than the console it replaces.

Some of the more impressive improvements include 10 auxiliary sends per channel via a switch matrix; group muting without VCAs; a flexible communications module; and an extremely low noise floor as perceived from the console (although no S/N tests were conducted).

The console is straightforward and engineer-friendly with a minimal fatigue factor, which should help the creative flow during marathon sessions. The console seems to benefit immensely from the backgrounds of Dee Robb, studio owner, and Toby Foster, designer, who both have served as recording engineers and musicians.

If creators of the music and the inventors of the technology could work hand-in-hand more in the future, the recording studio would be a more hospitable and conducive environment for the role that such a facility was originally designed to do: make music.
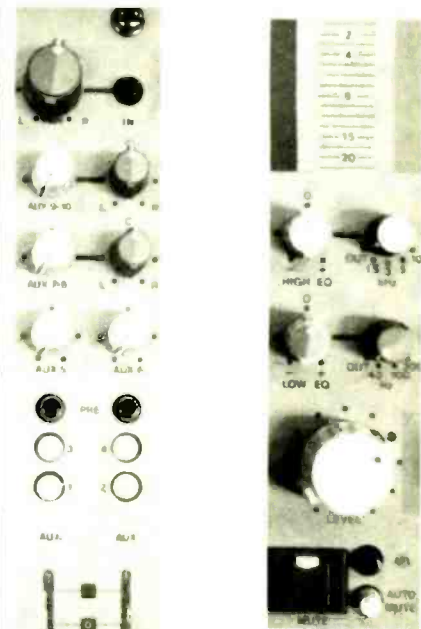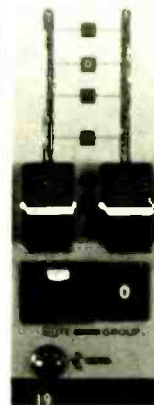
people that have never entered a recording studio. The end result, we feel, is that the music is being controlled by the technology instead of the artist—exactly the opposite of when the industry began."

In the immediate future the console will be outfitted with a servo-controlled, fader-automation system. What else does the future hold for Cherokee Studios?

"We're going to built two or three more (custom consoles) for our other control rooms," says Robb. "Excluding automation, we should be able to build four consoles, including research and development, very cost-effectively."

Cherokee's forward-looking philosophy has always kept them a strong and viable entity. By designing and building its own cost-effective consoles, the studio should be able to place itself in the front row when the next generation of "super-consoles" are perfected. R·E/P

Photos by Greg O'Loughlin and Elizabeth Annas.

Hands-on

Fairlight CMI Series III Digital Synthesizer

By Terry Fryer

What follows represents the first days of a diary written by a new Fairlight CMI Series III owner:

Day 1:

The excitement was high as we tore into the box labeled CMI Series III. Although the crate was the size of a small car, it contained only a video display unit, an alphanumeric keyboard with a graphics pad and the CMI mainframe computer—no music keyboard.

Spread out on the ground amid the rubble of the packing material, the Series III revealed only two things about itself: The cream color of the unit was a better designer color than the old Series II; and the VDU and alphanumeric keyboard look as if they were designed after 1950.

Later, Day 1:

Huffing and puffing as we climbed the stairs, we realized that more technology than meets the eyes is packed into the mainframe. Setting up the instrument was obviously like setting up the Series II. The instruction manual didn't get opened until later.

My Series II is set up for a left-handed humanoid, however, with the disk drives pulled out of the mainframe and the light pen on the left. As a result, the first 20 minutes of operation left me crawling around on the floor trying to boot the disk the wrong way. After that minor loss of brain control, I was ready to make a sound.

One of the joys of being the first to own a new instrument—especially a computer instrument—is that you also get a preliminary manual. In this case, the manual helped turn the Series III into a

The CMI Series III comprises a video display unit (left), music keyboard, alphanumeric keyboard and graphics tablet, plus computer frame housing the central CPU, floppy-disk drive, voice-channel cards and hard-disk unit.

megabuck boat anchor for several hours, while my tech and I pushed, prodded, rebooted and rechecked connections (with time out to answer a flurry of phone calls from well-wishers asking, "Is it great?"). Finally, at 0:dark:thirty Central Standard Time, the Series III produced its first sounds.

All the grief of the prior long hours disappeared as 16 voices of 16-bit samples flooded the room. The neighbors weren't quite as excited; they need a little education to appreciate what they were hearing.

Day 2:

I again loaded and played the demo that came with the Series III, and realized that it really did sound amazingly better than the Series II; 16-bit sampling is more than twice as good as 8-bit. We spent the first part of the morning getting used to the new alphanumeric keyboard. It's smaller than the one fitted to the Series II, and arranged entirely differently.

Because the music keyboard didn't arrive with the rest of the unit, the first serious piece of business was to look for

the instrument's MIDI section. The back of the mainframe bears three MIDI In and four Out ports. Any of the 16 MIDI channels on the three In ports can be routed to any of the MIDI channels of the four Out ports.

All of this routing is extremely easy to do, because of the representational design of the video display. I simply take the graphics pad and click in the correct position to indicate what I want to go where. Presto: instant routing.

The next business was to look at the sounds contained within the library. The Series III organizes pages much as the Series II does, in that separate pages are used to perform different functions on the instrument. Unlike the Series II, however, the Series III provides a directory of files available as a pop-on window from any of the pages. The dual processors are set up in such a way that this directory can be called with the graphics tablet, while the CMI is busy doing another job.

The instrument arrived with a built-in, 143Mbyte hard disk and 8Mbytes of internal waveform memory. Significance: The samples are *truly* impressive in length and take a long time to load—sometimes several minutes. [Editor's note: A fully loaded Series III features 14Mbytes of waveform RAM, providing a total capacity of five minutes of 16-bit, 50kHz mono sampling time.]

flexibility...

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

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BIDIRECTIONAL BUS

MIDI INPUT

MIDI OUTPUT (4 CHAN)

1MB FLOPPY DISC

85/140 MB HARD DISC

85/140 MB HARD DISC

60 MB STREAMING TAPE

VDU

MUSIC KEYBOARD

GRAPHICS TABLET

ALPHA-NUMERIC KEYBOARD

STEREO A.D.C. (16 BIT)

SAMPLE INPUT

AUDIO MODULE (2 CHANNEL) DACS, FILTERS

AUDIO MODULE (2 CHANNEL) DACS, FILTERS

MAX 8 CARDS (16 CHAN)

16 CHANNEL AUDIO OUTPUTS

Block diagram of the CMI Series III Digital Synthesizer.

On the Series II, it was fairly easy to tell when the floppy disk drives were loading a file: They sounded like a pair of coffee grinders. In the name of progress, the Series III has a quiet internal hard disk. If you're not really smart about where you put the mainframe, you will spend lots of time bending over to see if the indicator light for the hard disk is still flickering, because the screen does not indicate when a sound is loaded. Instead, you must look at the hard disk light and wait for it to stop flickering.

The rest of the day was spent listening to files, laughing at how good it sounded (and how long the samples were) and resolving to put the mainframe in a position where the hard disk indicator could be seen.

Day 3:

To make old Series II users feel good, Fairlight has included a bunch of Series II files in the III's library. After several hours of listening to new 16-bit samples, calling up an old 8-bit sample proved to be a painful process; there is *no* way to go back. The manual says you can con-

vert your old Series II disks into Series III files. But, unless you're really masochistic, I don't recommend it.

Now it was time to explore new pages. A breakthrough! Instead of Page 1, Page 2, Page 3 and so on, these pages have some sensible labels. When it's time to sample, you type *SA*; when it's time for waveform editing, *WE*. It's about time somebody thought of it.

The waveform edit page took a little getting used to. Although it is deceptively similar to the Series II, after unlearning a couple of things—like reaching for the non-existent light pen—I found it hard to go back and work on the Series II. Commands such as *zoom in* and *widen window* are available, and allow the user to rapidly zero in and display particular sections of the waveform. It's possible to do pitch analysis and autolooping automatically; no more hours of painstakingly matching the sample rate to the pitch of the sample and looking for the right loop point.

The waveform can be rotated right and left, and specific sections inverted or played backwards. I took an eight-bar

drum break and experimented with reversing the fourth beat of each bar. It sounded great.

The graphics pad is much more controllable and accurate than the light pen. (I just wish that it could be on the left-hand side.) The Series III displays are also easier to read and control.

The rhythm sequencer page is real fun. It is like the old Page R, with a visual display of all the voices in rhythmic notation, but lots better. Because 16 voices are provided on the instrument, there are 16 voices on the sequencer. New notation makes it easier to see the duration of notes, as well as stringing patterns together. Total note and sequence capacity is only limited by the size of the CMI's hard disk.

You can also load new voices while the sequencer is playing. We replaced the drum kit with a motorcycle sample, and played the brass section backward. (It didn't do much for the piece of music or the neighbors, however.)

Looking at the subvoices caused a little head scratching. As I understand it, 16 voices are available on the instrument.

The interesting part is that each of these voices can be divided into 128 subvoices, and assigned to any part of the keyboard. The function became clear when we found a voice labeled *Percuss1* and loaded it. On one voice, mapped across the keyboard was an entire percussion battery: tambourines (lots of them), flexatones, wood blocks, cowbells and on and on. When 16 voices like this are loaded up and played, things can get insane.

The software also allows you to port over Series II sequences from Page R and Page 9. In addition, a command allows two Page R sequences to be merged together into one 16-voice Series III sequence. Although porting over Series II sound files is fairly pointless, merging the sequences could be useful.

These subvoices also have separate filter settings, vibrato and envelopes, so a great deal of control is available over each part of the sound.

Later Day 3:

I ended up with a project to do for Kentucky Fried Chicken, and decided to give the MIDI ports a good workout. Calling up a full complement of voices, including horns, drums, bass and percussion, I assigned each voice of the Series III to a different MIDI channel, and used a Yamaha KX-88 keyboard as a controller and a Linn 9000 as a sequencer. It worked great! Happy client, therefore happy music producer and Series III owner. Again, CD-quality sampling is hard to argue with.

Day 4:

At last, time to play with the sampling function. I almost felt guilty when I realized that I didn't have to be stingy with choosing short samples that loop right away. My first victims were my
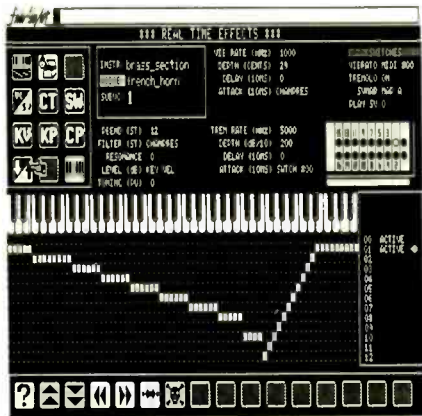
The effects page can be used to show a number of sub-displays to provide control allocations. Here, the subvoices mapped to different areas of the keyboard are displayed graphically.

very young niece and nephew, who wanted to know if I would record them like Stevie Wonder did on *The Cosby Show*. I said, "Sure, step right up to the microphone."

This was no time to be looking at instruction manuals, but the sampling page looked fairly close to the Series II page, so away we went. It gets a little scary setting a sample time of 10 or 20 seconds. There is also a slick little VU meter that displays peak as well as continuous level. When a new voice is created on the Series III, it loads it with sine waves. For us adult types, this is a great way of figuring out if the system is screwing up, or if you just forgot to turn up the proper fader.

But, this feature is hard to explain to young nieces and nephews. The first several samples consisted of 30 seconds of "ummm, uhhh, what should I say, waaaah," and the like. A short rendition of the alphabet song then produced enough suitable material to provide an hour or so of looping, fragmenting, reversing specific letters, transposing and layering. The speed at which the system performed waveform manipulations was incredible. Those tricks and the graphics displays were sufficient to keep the budding sampler's attention for an extraordinary length of time.

Later Day 4:
More Kentucky Fried Chicken. By now my keyboard skills were fast enough that I could take advantage of the keyboard buffer, which stores keystrokes from the alphanumeric keyboard, allowing you to type in future commands while the CMI is processing earlier ones. If you're loading several files that are a couple of megabytes in size, the buffering allows a quick trip to the kitchen for sustenance without losing time on the instrument.

Technical Specifications

Voice system: 90dB dynamic range voice/channel cards, (consisting of dynamic VCF, VCA for each card), with up to 16 voices available in the standard voice system; expandable to 80 voices per system (via external voice racks). Separate 16-bit D/A convertors; 16-bit, 50kHz stereo audio sampling capabilities (100kHz in mono).

Up to 14Mbytes of waveform RAM available per 16 channels, providing more than two minutes of sampling time at 50kHz. Twelve microprocessors operating multitasking functions and programming languages. Waveform Editor software allows editing of waveforms, Fourier analysis, synthesis and resynthesis functions. SMPTE timecode read, write and synchronization capabilities (including chase). Multiple MIDI connectors.

Storage system: 8-inch, double-density, double-sided floppy-disk drive (with controller), capable of storing 1Mbyte of data. Plus 70Mbyte (or optional 100Mbyte) formatted 5.25-inch Winchester hard disk drive, with controller and standard SCSI interface, allowing for connection of additional hard disk drives or other storage units. Optional 60Mbyte streaming tape drive.

Graphics terminal system: 82-key alphanumeric keyboard, including 15 assignable functions keys and high-resolution graphics tablet with stylus. High-resolution, 12-inch video display unit.

Day 5:

Now down to sampling for real. I got out the manual, looked under the section on sampling and found some interesting things to try. On the Series II, it was necessary to adjust the sampling rate to the pitch of the instrument being sampled; the Series III will do it automatically for you.

On to the autolooping function. I have had quite a bit of experience with other digital sampling keyboards, and looping is an area that I would like to eliminate from my life. Well, Series III autolooping is about as close as I'll get during this

Sequencers: Music Compositional Language text-based composer. Rhythm sequencer, 16-track recorder with graphic-note events. Composer arranger performer sequencer (CAPS), with up to 80 polyphonic tracks assignable to internal voices or external devices (via MIDI), programmed in real-time, quantized or in non-real time. Micro- and macrocomputer editing features. Tracks can be viewed as conventional music notation. All sequences synchronize and trigger to timecode.

Music keyboard controller: six octave, F to F; MIDI implementation; pitch and modulation wheels, including programmable switches and controls.

Serial interface: dual printer ports. Telenet telecommunications software.

Configurations:

• Series III 16.8 system, consisting of a 16-bit stereo-sampling central processing unit, 16 voices, 8Mbytes of waveform RAM (expandable to 14Mbytes), MIDI/SMPTE capabilities, 60Mbyte hard disk drive, alphanumeric keyboard and graphics tablet, MIDI-controlled music keyboard, one floppy disk drive, video display unit and all corresponding software.

• Series III 8.4 System, consisting of a 16-bit stereo-sampling central processing unit, eight voices (expandable to 16 voices), 4Mbytes of waveform RAM (expandable to 14Mbytes), MIDI/SMPTE capabilities, 60Mbyte formatted hard disk drive, alphanumeric keyboard, one floppy disk drive, video display unit and corresponding software.

Manufacturer: Fairlight Instruments.
Circle (180) on Rapid Facts Card

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lifetime. The function actually *works.* I tried a choir sample that I had labored fruitlessly to capture into certain other instruments (that shall remain nameless). On the Series III, it took all of two minutes to set the correct level, adjust the sample rate and the sampling into the machine.

Series III—I could get used to this.

R·E/P

Fryer is a co-partner in Colnot-Fryer Music, a Chicago-based jingles, commercials and music production facility.

Northeast:

Production Masters (Pittsburgh), an audio and video production facility that provides creative and technical services to advertising agencies and corporations, has hired **Eric Trow** as account representative and **Chris Anderson** as audio engineer. Trow is a graduate of Calrion University with a bachelor's degree in communications. Anderson has a B.A. from the Berkelee College of Music, and will be responsible for maintaining and installing audio equipment and recording/mixing audio projects. *321 First Ave., Pittsburgh, PA 15222; 412-281-8500.*

Digital Music Products (Stamford, CT) has installed a second **Mitsubishi X-80 digital two-track**. "The machine's portable configuration has been a real plus for us," says label president **Tom Jung**, whose facility is noted for live to two-track digital recordings. "There are really no [tape machine] setups or alignments to worry about." *175 Dolphin Cove Qual, Stamford, CT 06902; 203-327-3800.*

Cove City Sound (Glen Cove, NY) is a new 24-track studio featuring a **Neve 8068 console** linked to a **Studer A-80**
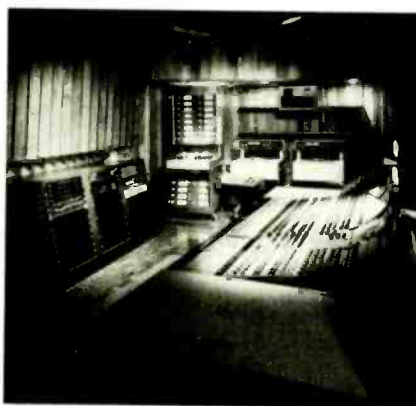
Cove City: new facility featuring Neve 8068 console.

24-track. Outboard gear comprises an **AMS DMS**, **Lexicon PCM-60**, **Yamaha REV-7** digital effects processor, **Valley People Kepex II** noise gates, **dbx model 160** stereo compressor/limiter, **Orban model 662Bs** and **Eventide H969 Harmonizer**. In-house instruments include a **Yamaha DX-7**, **Memorymoog**, and **E-Mu Systems Emulator II**. Studio monitoring is provided by **JBL model 813Bs** driven by two **McIntosh model 2255** amplifiers; a pair of **Yamaha NS10Ms** are powered by a single **McIntosh model 2155** amp. *7 Pratt Blvd., Glen Cove, NY 11542; 516-759-9110.*

Quadrasonic Sound Systems (New York City) has installed a second 48-input **Solid State Logic SSL 4000E console** with **Total Recall** and **Studio Computer** in Studio A. Slaved to the desk are two **Studer A80** 24-track machines, synchronized by a **Cipher Digital BTX Softouch** unit. Owner **Lou Gonzalez** says that Studio B will continue its album- and jingle-production focus. *723 7th Ave., New York, NY; 212-730-1035.*

La Guardia High School for the Performing Arts (New York City) has a new four-track studio for recording student performances. Housed in the facility are a 12-input **Sony JH-618 console**; a **Sony JH-110** ½-inch four-track and two-track machine; **Inovonics model 201** limiters; and **AKG BXE25E** reverb units. Monitoring is provided by **JBL 4411** speakers and microphones are supplied by **Neumann**, **AKG**, **Crown** and **Shure**. **Westec Audio**, New York City, supplied the equipment and training for the new facility. *Amsterdam Ave., New York, NY; 212-877-8908.*

39th Street Music Productions (New York City) has completed renovation of its main control room, following the installation of a **Solid State Logic 4000E console**. Recent outboard acquisitions include a **Publison Infernal Machine 90**, **Drawmer noise gates**, and a **Lexicon PCM-70** Dynamic MIDI effects processor. The facility has also installed a computer-based synthesizer area, which features **Yamaha TX-816** and **DX-7** synthesizers, **Oberheim Matrix-12** and **OB-8**, **E-Mu Systems Emulator II** synth and **SP-12** sampling drum machine, a **Sequential Prophet V**, a **Minimoog**, a **Linndrum** and a **Casio CZ-101** synthesizer. **IBM** and **Macintosh** personal

39th Street: SSL 4000E installation.

Southeast:

Omega Recording (Rockville, MD) has installed a **New England Digital Synclavier II** digital synthesizer system into its three-studio complex. The facility has also added a **Cipher Digital Softouch** synchronizer (with BP.2 software) to lock one of the facility's **Studer MkIV A80** 24-tracks to a **Sony 5850** VCR. In addition, the facility has ordered three sound effects libraries on Compact Disc from **Sound Ideas, Network** and **Dewolfe.** *5609 Fishers Lane, Rockville, MD 20852; 301-946-4686.*

Sheffield Audio-Video Productions (Phoenix, MD) has purchased two **Otari MTR-90s** for its audio remote truck. The facility says that although the truck is normally loaded with a **Sony PCM-3324** digital multitrack, the new analog acquisitions will "allow for maximum recording flexibility." *13816 Sunny Brook Road, Phoenix, MD 21131; 301-628-7260.*

South Central:

Ardent Recordings (Memphis) has remodeled Studio B, the control room of which now measures 25'x15' (studio: 24'x28') and features a **Solid State Logic**

Ardent: renovated CR equipped with SSL 4000E console.

computers are also used for sequencing, patching and sound synthesis applications, with overdub setups manipulated via **Octave Plateau Patch Master** software, and sequencing options controlled via **Sequencer Plus.** *260 W. 39th St., 17th Floor, New York, NY 10018; 212-840-3285.*

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4000E Series console with **Total Recall** and **Studio Computer**. A Mitsubishi **X-850** digital 32-track and two **X-80** two-tracks also have been installed, plus a new **Fairlight CMI Series III 16.8** system was also purchased. Studio B's outboard equipment array is also upgraded to include a **Lexicon 224XL** digital reverb; **Valley People** expander/compressor and **Dynamite model 440s**; **DOD Exciters**; a **Yamaha TX-16** rack; six **Yamaha REV-7** digital processors; and two **Macintosh 512** personal computers with **Opcode** MIDI software. *2000 Madison Ave., Memphis, TN 38104; 901-725-0855.*

AAmerican Professional (Houston) will open a new branch office in Nashville, to function primarily as a music publishing, management and pre-production firm. To be located on Music Row, this new arm of the company will liaise with the main Houston office "to better market Houston music groups," the company says. The current Houston staff includes **Rom Favor**, promotions; **Joe Saenz**, producer and production manager; **Jimmy Duran**, arranger and composer; **Fran Moore**, engineer and composer; **Linda Saenz** and **Roxy Niles**, office management. *838 Courtesy, Suite 2, Houston, TX 77032; 713-987-3272.*

Digital Recorders (Nashville) has purchased a new **Sony PCM-3102** DASH-format digital two-track. The facility says that the acquisition will complement its existing pair of **Sony PCM-3324** digital 24-tracks and **Sony MXP-3000** console. *54 Music Square East, Suite 305, Nashville, TN 37203; 615-244-0100.*

Midwest:

ARS Recording (Alsip, IL) has installed a new keyboard array consisting of **Yamaha DX-7** and **TX-7** synthesizers, **Ensoniq Mirage**, **Casio CZ101**, **Roland Planet-S**, **360 Systems MIDI Bass**, **Linn 9000** MIDI sequencer/drum machine and **LinnDrum** machine. *11628 S. Pulaski, Alsip, IL 60658; 312-371-8424.*

Northern Entertainment (Cleveland) has upgraded Studio B with the installation of a 32-input **Soundcraft model 1600 console** slaved to a **Sony JH-24 multitrack**. New MIDI-capable equipment includes an **E-Mu Systems Emulator II**; **Yamaha DX-7** and **TX-16** synthesizers; **Roland Super Jupiter** and **GR-700** guitar synthesizer; **Linn 9000** MIDI sequencer/drum machine; and **Akai** samplers. Studio design was accomplished by **Tom Irby** of Valley People, Nashville. *500 S. Marginal Road, Cleveland, OH 44103; 213-361-2650.*

Southern California

Sunset Sound (Hollywood) has installed three **Studer A800 MkIII** 24-tracks into Studio 2. The facility has also undergone cosmetic changes, in-

cluding the addition of a 25-inch video monitor; a new annex for tape machines and ancillary equipment; and an improved working area for electronic instruments. An additional isolation booth was also built into the studio area. *6650 Sunset Blvd., Hollywood, CA 90028; 213-469-1186.*

The Burbank Studios (Burbank) has ordered 60 Micron CTR-501 wireless microphone systems, each consisting of a model TX-501 pocket transmitter and model MR-510 mobile receiver with CNS noise suppression. Reportedly, the units offer a 115dB dynamic range, extended operating range, enhanced low-frequency performance and improved multichannel performance. Upon delivery of this latest order, TBS will have more than 150 Micron wireless microphones in operation. *4000 Warner Blvd., Burbank, CA 91510-7803.*

Craig Harris Music (Studio City) has added a MIDI-controlling option to its New England Digital Synclavier synthesizer, to allow eight separate MIDI outputs (with 16 channels each) for control of the following recently acquired MIDI-capable equipment: a Yamaha TX-16 synthesizer rack and REV-7 digital reverb processor; a Roland Super Jupiter; and a Lexicon PCM-70 Dynamic MIDI effects processor. Also added was an AMS DMX 15-80S digital delay and pitch changer. *P.O. Box 36A45, Studio City, CA 90036; 818-508-8000.*

Northern California

City Sound Recording (San Francisco) has added an ADR Pan Scan autopanner, dbx model 165 limiter and a fully updated AMS RMX-16 digital reverb. *245 Hyde St., 2nd Floor, San Francisco, CA 94102; 415-474-0377.*

OTR Studios (Belmont) has acquired an Aphex Compellor, a Macintosh personal computer with Marc of the Unicorn Performance and Composer MIDI software, and a Sequential 2002 sampling unit with Digidesign Sound Designer software. *P.O. Box 874, Belmont, CA 94002; 415-595-8475.*

Northwest

Steve Lawson Productions (Seattle) has unveiled a new electronic tracking MIDI room, which features a Kurzweil

K250 digital sampling keyboard, Oberheim DMX, LinnDrum, Yamaha RX-11 and four DX-7 sound modules, Roland Juno 106 and a Macintosh personal computer with Southworth Total Music software. According to the studio's owners, the new room is completely MIDI capable and can link to Studios A, B or C. Also purchased was the Kaba RTDS tape duplicator, which can duplicate eight cassettes simultaneously. Finally, Carol Howell has joined Lawson's staff as operator of the new MIDI room. *Sixth and Battery Bldg. 2322 6th Ave., Seattle, WA 98121; 206-443-1500.* R-E/P

Adams-Smith System 2600 controller

The hand-held model 2600-CC incorporates video-style editing procedures, as well as standard audio procedures for soundtrack assembly. The remote-control unit is said to implement a new way of editing TV audio, from ideas suggested by System 2600 users as well as company engineers.

The controller can operate five synchronizer modules and a timecode generator to control up to five audio or video transports. Features include: fully automated audio rehearsing (previewing); recording and replaying functions by use of system-in and -out points (with all offsets, pre-rolls, post-rolls and durations calculated automatically); automatic tape cuing, loop, stop and record; eight non-volatile functions keys (for storing pre-programmed routines); 100-position, non-volatile scratch-pad memory; timecode to feet-and-frames conversion; drop/non-drop-frame conversion; varispeed operation; and optional 0.01-frame accurate audio punch-in and -out and/or auxiliary equipment turn-on and -off capability.

The model 2600-CC is connected to the mainframe chassis via a small diameter flexible cable, and offers compatibility with other model 2600 modules.

Circle (153) on Rapid Facts Card

Dolby announces new Spectral Recording process

Under development for six years, the new process is intended for use with professional analog tape machines, and is available in the form of single-channel plug-in modules to fit existing and new mainframes.

The SR process exploits the spectral diversity of audio signals, taking advantage of a previously unused mechanism to improve the information capacity of analog recording. The system employs a "powerful new coding algorithm" sensitive to variations in signal spectrum as well as to level changes. This technique is in contrast to noise reduction systems, which respond primarily to level variations.

Among other advantages, the new technique is said to provide professional analog tape machines with a useful dynamic range equal to or greater than that of 16-bit digital recording systems. At the same time, SR is said to effectively suppress modulation noise and other ef-

fects introduced by conventional analog recording.

According to Dolby, SR has a linear and continuous transfer characteristic and does not add high-order distortion or step-switching artifacts to a recording, even at critical, very low signal levels. At very high recorded levels, SR is described as reducing spurious high-order harmonics and intermodulation components caused by tape saturation, "improving the integrity of signal transients even under near overload conditions."

A new feature, *auto compare*, is said to allow rapid verification of correct recorder alignment and operation. Each SR module contains a built-in generator that applies pink noise to the recorder channel in an easily identified way (Dolby Noise) and at a standard level. When auto compare is switched on during playback, the monitored signal automatically alternates between the Dolby Noise recorded on the tape and the internal reference pink noise generator, with visual indication of which signal is being reproduced. In this way, an engineer can make an instant auditory comparison of the two signals without additional equipment, measurements, or a break in the recording session.

Circle (151) on Rapid Facts Card

Sony DFX-2400 sampling-rate converter

The new two-channel unit permits transfer of digitally sampled signals between different formats totally within the digital domain.

The rack-mountable device is said to provide 96dB conversion accuracy, and handles the four most commonly used sampling frequencies: 32, 44.056, 44.1 and 48kHz. The unit also offers Sony PCM-1610/1630 and AES/EBU-format inputs and outputs, and can transmit auxiliary data in the AES/EBU mode.

Circle (155) on Rapid Facts Card

Cable tester from Whirlwind

Designed as a pocket-size package, the new tester accepts all standard audio connectors. Via LED displays labeled "pin 1 shield," "pin 2 ring," "phase reverse," and "pin 3 tip," the unit tests the condition and phasing of a cable.

The unit does not use switches or buttons, and is housed in a heavy-duty die-cast aluminum box with belt clip.

Circle (163) on Rapid Facts Card

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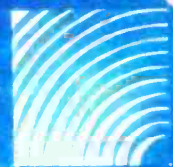
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Circle (74) on Rapid Facts Card
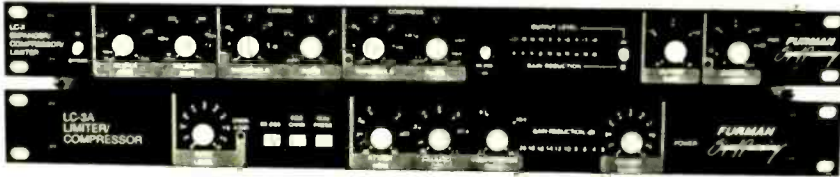
Two compressor/limiters from Furman

The LC-3A complimiter, which replaces the LC-3, features control over input and output levels, attack and release times and compression ratio. Also provided are a detector circuit, which the company says virtually eliminates distortion caused by waveforms following gain changes, and a 10-segment LED meter.

The LC-X is a combination expander, limiter and compressor, and features balanced and unbalanced input and outputs; a quoted +0/−0.5, 20Hz to 20kHz frequency response; 104dB dynamic range; variable attack and release times (30μs to 1s, and 50ms to 5s, respectively); expand and compression ratios (1:1 to 5:1, and 1:1 to infinity:1, respectively); output-limit ratio of 20:1; and output-limit release time of 0.5s.

Both units also feature de-ess and side-chain capabilities, ground-lift switch, low- and high-level inputs, and a stereo-link capability.

Suggested pro user prices are $249 for the LC-3A, and $349 for the LC-X.

Circle (152) on Rapid Facts Card

Alpha introduces PC-LISTOFF for Boss editor

Designed as an off-line companion for the 8400 series on-line automated editing system, the PC-LISTOFF software provides off-line list management for an IBM PC and compatibles.

PC-LISTOFF is said to provide session preparation before a recording date. Because the model 8400 can access files prepared by PC-LISTOFF, an audio decision list or event list can be sketched out in a small off-line room for pre-production work.

The new off-line software is described as forming just one component in a system that streamlines data entry. For example, a CMX disk interface is available for the Boss 8400, to allow a CMX EDL to be loaded automatically to an audio decision list, and manipulated like any other Boss file.

Circle (154) on Rapid Facts Card

New products

Club MIDI unveils Prolib Librarian software

A universal patch librarian for the IBM PC, Prolib stores sound patches from 18 different instruments onto floppy disk and provides data management and editing features. During operation the user selects an instrument with the cursor key to receive a patch data bank or to transmit previously stored patches from the computer.

Patches can be named, copied, moved, updated and deleted via commands from the computer's keyboard. In addition, patch banks can be stored alphabetically, titled and saved to disk. An audition feature allows the replay of selected features.

Optionally, up to one full line of text can be entered to accompany each patch, allowing the user to document non-storable sound attributes, such as function parameters, wheel settings, real-time knob adjustments, signal chains or outboard gear settings. Patches and their accompanying remarks can be printed for later reference.

In addition, users can create and save function environments to automate any number of the program's procedures.

Prolib has a suggested list price of $99.95, and runs on the IBM PC and compatibles equipped with a Roland MPU-401 MIDI interface. The software supports Yamaha DX-7, DX-9, DX-21, TX-7 and TX-16 formats (voice and performance data); Oberheim OB-8, Matrix-6, Xpander and Matrix 12 (single- and multipatches); Sequential Prophet-V, T-8 and 600; Casio CZ-101 and CZ-1000; Korg DW6000 and DW8000; plus Roland MKS-80 (Super Jupiter) and JX8P.

Circle (166) on Rapid Facts Card

Klark-Teknik DN716 digital delay

Intended to replace models DN700 and DN701, the new unit is designed for sound-reinforcement applications, and features one input, three outputs, 16-bit linear converters, 90dB dynamic range, and a quoted frequency response of 20Hz to 20kHz.

Delay time of each of the three outputs is independently adjustable from zero to 1.3s, in $20\mu s$ increments. All memory settings are non-volatile. Input and output level controls are located on the front panel.

Optional features include a fail-safe bypass relay, security cover and input or output transformer balancing.

Circle (156) on Rapid Facts Card

New products

Model A807 broadcast tape machine from Studer

The "low-cost" model A807 is constructed with a die-cast chassis and three servo-controlled motors. Tape deck functions and phase-compensated audio electronics (including audio-alignment) are under full microprocessor control.

The machine can handle 11.1-inch tape reels and runs at three tape speeds. A wide range of versions covers all popular ¼-inch professional formats, including portable units (with mic inputs and phantom powering), rack-mount and console-mounted models.

Circle (160) on Rapid Facts Card

AMEK assignable production console

The new APC-1000 is described as a large architecture console (LAC) or "semi-virtual" console, which allows custom configuration to suit user requirements. Every switch and routing function is controlled by a central microprocessor, including channel-input selection, EQ in/out and auxiliary output bus selection with pre/post and in/out. Up to 48 routing busses and eight stereo busses can be controlled simultaneously, along with an optional dynamics section, with an expander/compressor module per channel.

In contrast to conventional designs, there is no tape monitor on the channel nor a separate monitor channel. The APC-1000 is neither in-line or split-monitor, but consists purely of channels that may be dedicated to whatever function the operator requires.

On-board internal memory allows for storage and recall of up to 99 console setups, dependent on RAM capacity. The console is supplied as standard with VCA faders for interface with the Audio-Kinetics Mastermix computer automation system, enabling digital grouping and fader-level storage. Alternatively, the APC-1000 can be supplied with GML Moving Fader automation, allowing for expanded computer-control options, such as real-time update of switch configuration from timecode locations—which AMEK refers to as *synchronous reset*—and increased RAM page storage from the system's hard disk drive.

An optional memory system allows rotary knob positions to be stored for later manual reset. A band of display LEDs allows current pot positions to be matched with a particular setting stored in on-board memory.

The console can be supplied with up to 128 computer-controlled modules, and AFV (audio-follows-video) ports can be retrofitted as standard equipment.

Pro user price for a 48-channel APC-1000 is $100,000; a 64-channel version with GML Moving Fader automation is priced at $200,000.

Circle (158) on Rapid Facts Card

New products

Otari DTR-900 digital multitracks

The fixed-head, 1-inch, 32-channel transport utilizes the ProDigital (PD) format to record a total of 45 data tracks: 32 for digital audio; eight for parity; two for auxiliary analog information; one for timecode; and two for auxiliary digital data.

Features include cyclic redundancy check and Reed Solomon code error-detection and correction circuits; razor blade and electronic editing; seamless punch-in/-out functions; long-life ferrite heads; and switchable 48 or 44.1kHz sampling rates.

The machine comes standard with peak-reading meters, pre-emphasis control, built-in timecode generator, over-dubbing and ping-pong recording, parallel or serial interface, and active balanced-line inputs and outputs. User-assignable transport switches and 100 cue-memory autolocator are provided.

The DRT-900 is offered in 24/32 (24 expandable to 32) and 32-track configurations. Deliveries are expected to begin this summer. The company plans to introduce a 16-track on ½-inch version by the fall.

Circle (159) on Rapid Facts Card

360 Systems MIDI Patcher

Designed as a 4-input, 8-output routing system, MIDI Patcher allows the programming of up to eight routing configurations, and storage in a battery-backed memory. Configurations can be recalled via front panel buttons or via MIDI program change commands.

Eight groups of four LEDs on the front panel display the current routing configuration. To verify MIDI continuity, a test button sends a short MIDI sequence over a selected channel.

MIDI Patcher is rack-mountable, and carries an SRP of $295.

Circle (167) on Rapid Facts Card

Roland RD-1000 digital piano

Featuring a touch-sensitive, weighted-action wooden keyboard, the RD-1000 is said to deliver dynamic sensitivity to fully recreate timbre variations characteristic of different ranges on acoustic instruments, utilizing Roland's proprietary Structured/Adaptive Synthesis technology. The 16-voice instrument offers eight preset sounds: three acoustic pianos, harpsichord, clavichord, vibraphone and two electric pianos.

Fifty-six programs of voice function data—such as level, equalization setting, chorus effects and tremelo effects—can be entered, and then stored either in the unit's internal memory, or on an optional M-16C memory cartridge.

The instrument also offers MIDI-control options in the form of program change, volume control, key transposition and velocity information. Balanced and unbalanced stereo outputs, a headphone jack, power supply, a damper pedal, soft pedal and external pedal are also provided.

Circle (164) on Rapid Facts Card

Timecode reader from ROS Software

Designed to run on 64Kbyte Apple IIe or II+ personal computers, the new Time Counter software utilizes the Apple's cassette input port to read timecode and act as a 10-point events controller.

An events controller can be set to perform events at specific times, or softkeys pre-programmed for event initiation. Up to 100 files, each containing 10 event codes or event times, can be stored on a single disk.

Time Counter can send a timecode-locked frequency in the form of a click track, or for routing to a frequency-driven transport, or to synthesizers for phase-locking to a timecoded tape. Events can also be caused to switch the position of the II's annunciator outputs for sending TTL levels to an exterior device. Events can be loaded automatically, allowing up to 1,000 separate events to be performed in a 24-hour period. The software will support serial-, parallel- or MIDI-interface cards.

Printing functions of all events and timecode information are available, along with a "help" menu.

The software package with manual and program disk has a suggested list price of $140.

Circle (162) on Rapid Facts Card

R·E/P

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Soundworkshop series 30 32 ch. w/meter bridge and flight case. Asking $12,000.00. Otari 5050 ¼" 4 and 2 tracks, Lexicon Primetime, Eventide 910 Harmonizer, Korg-SDD3000, 48 Point PRS ¼" patch bay (prewired to pigtails), API Console Main Frame (only). To make an offer and/or for more information please contact Steven at P.O. Box 791, Flushing, NY 11352/718-886-6500.

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Even if your music starts as a piece of junk, your sampling mic better not.

The new Shure SM94 Condenser Mic can make a big improvement in your digital sampling—at a surprisingly affordable price.

If you've made a major investment in a sampling keyboard or drum machine, don't overlook the importance of the microphone you're using. A vocal mic, for example, might "color" instruments you are sampling.

To capture your sample as accurately as possible, we suggest the new SM94. Unlike many popular mics, the SM94 has no high-frequency peaks, accentuated presence boost, or excessive low-end rolloff. This prevents overemphasis of high frequencies on instruments like strings and brass, while allowing you to retain the important low-frequency response essential to capturing the fullness and richness of many live sounds.

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In addition to offering a unique combination of features not normally found in condenser mics in its price range, the SM94 is built with Shure's legendary emphasis on ruggedness and reliability. Features like a protective steel case, machined grille and tri-point shock mount make it rugged enough to go wherever your inspiration takes you.

And for voice sampling, we suggest the new SM96 with its vocal contoured response and built-in three-stage pop filter. Both these fine microphones can bring a new dimension of realism to your digital sampling.

For more information, write or call: Shure Brothers Inc., 222 Hartrey Avenue, Evanston, IL 60202-3696, (312) 866-2553.

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Circle (3) on Rapid Facts Card