July 1988

Recording ENGINEER/PRODUCER

The Applications Magazine for Audio Professionals

AN INTERTEC PUBLICATION

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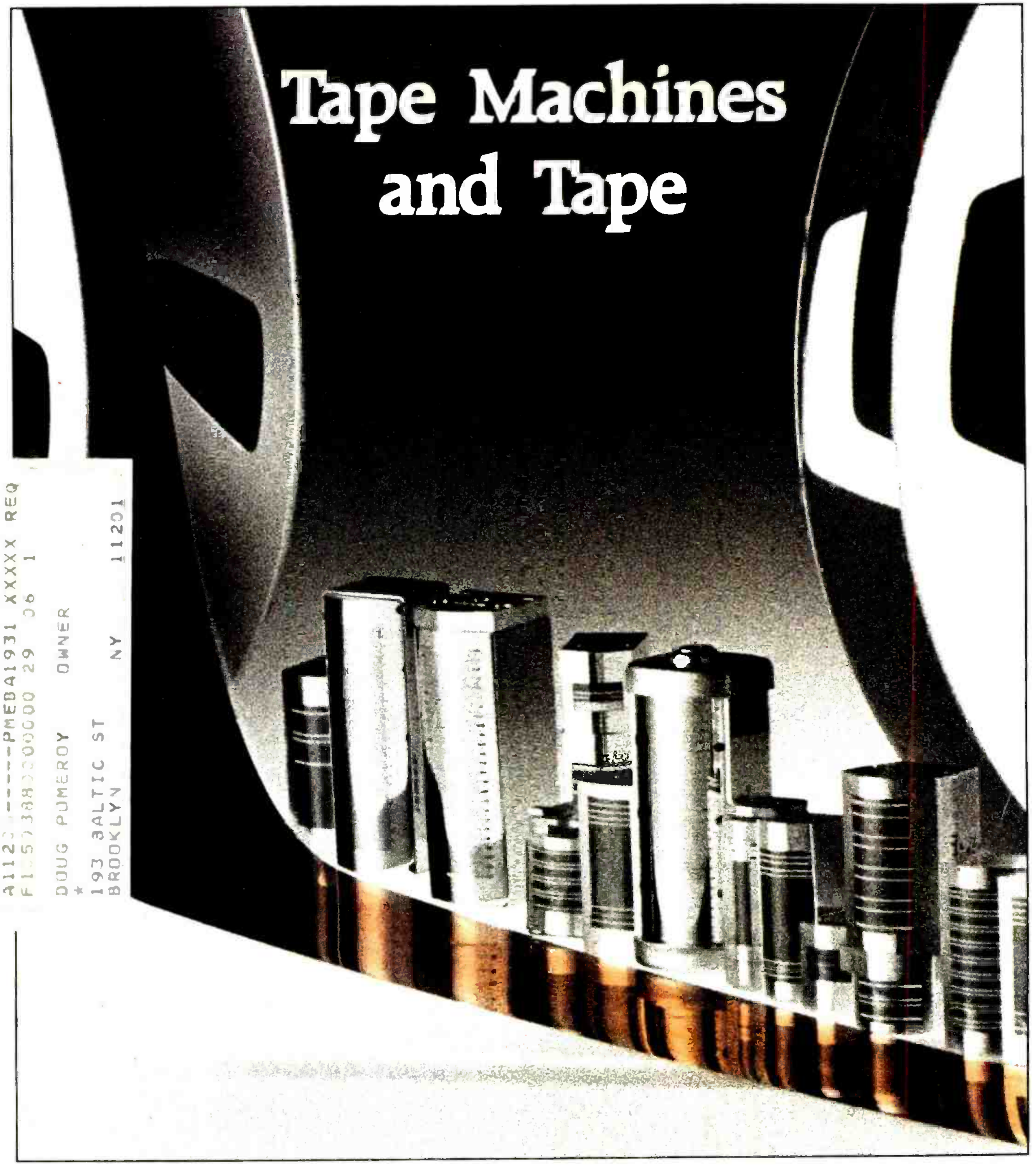
RE/P EXCLUSIVE
Interview with Grammy-Winning Engineer Bruce Swedien
(Page 36)



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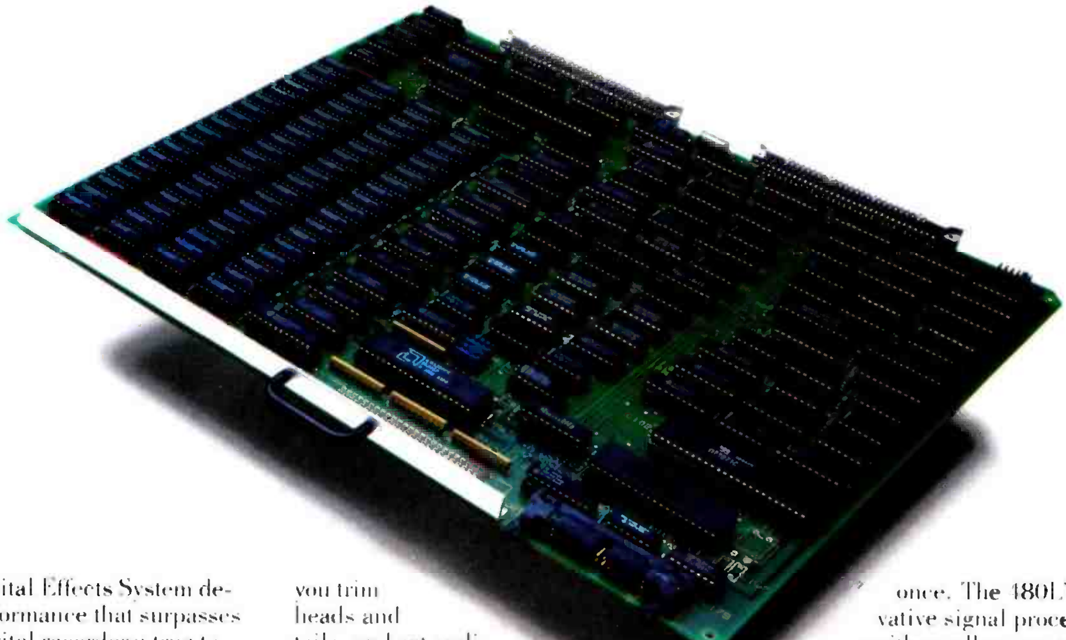
EXPANDING THE SYSTEM

The 480L Sampling Memory Expander. Accurate sampling in phase-locked stereo: a Lexicon applications brief.



The 480L Digital Effects System delivers audio performance that surpasses conventional digital recorders: true to life sampling is a prime example of its advanced engineering. With the optional Sampling Memory Expander, the 480L becomes an astonishingly practical way to copy or move several seconds of audio from point A to points B and C.

Until now, "flying in" or "slipping" vocal and instrumental overdubs meant tradeoffs. Two-track tape is clumsy and degrades the signal: DDLs with long memories are better, but not by much. Now, with the 480L's optional Sampling Memory Expander, you can digitally record 10 seconds of true phase-locked stereo or 20 seconds of mono at a 48 kHz sampling rate. 18 bit equivalent A/D conversion produces a typical dynamic range of 98 dB, with a bandwidth of 20 — 20,000 Hz: output doesn't get any closer to input.

If you're using a digital recorder that's Sony 1610 compatible, the Sampling Memory Expander and the 480L's digital I/O interface can reproduce the original performance at any location on tape without leaving the digital domain. Slip syncing, copying and time shifting of audio segments is faster and cleaner than ever before.

You control sample recording, editing, processing and triggering from your work position using the 480L LARC. Varying up to six parameters at once, you trim heads and tails, and set audio trigger levels. Audio trigger response time, incidentally, is under 300 microseconds — virtually instantaneous. Sampling Memory Expander enhancements include Record Trigger to trigger sampling automatically from audio input, Time Variant Record, and sound on sound digital recording in sync with the original sample. Lexicon Dynamic MIDI® lets you assign MIDI controllers to sampling parameters, for new dimensions of real-time or sequenced control.

Sometimes even the most accurate replica isn't exactly what you're looking for: if so, the 480L will take you beyond imitation into creative sampling. Play samples faster or slower (without changing the sampling rate), backwards or forwards, even both at once. The 480L's innovative signal processing algorithms allow you to enhance sounds with advanced reverbs and effects without leaving the digital domain. Add signal processing as you sample, or process a "dry" sample on playback with digital wet-dry mixing.

The 480L digitally integrates sampling and signal processing because it's designed as a comprehensive audio production tool: a system. And sampling is only the first of many directions in which this Digital Effects System can grow. We're continually examining the possibilities, and as your needs develop we'll implement new tools to meet them. Now is the time to call your Lexicon Advanced Products dealer for a demo of the new production techniques the Sampling Memory Expander makes possible.

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With 16 bit linear PCM, 15K bandwidth, and tons of musical character, MIDIVERB II is the #1 selling signal processor in the business.* That'll only surprise you if you've never used it. Those who have used it love the sound so much they can't resist buying several more. With 99 programs — 50 reverbs, plus choruses, flanges, delays, and innovative special effects — Midiverb II redefines the meaning of cost-effectiveness.

So after today's mix, you deserve something refreshing.

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*Based on *Music and Sound Retailer's* monthly survey of 1,200 audio dealers

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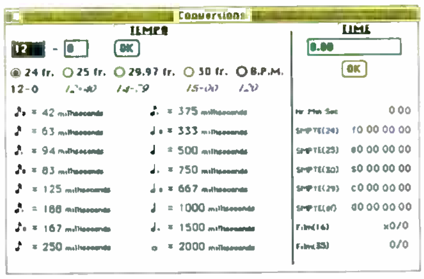

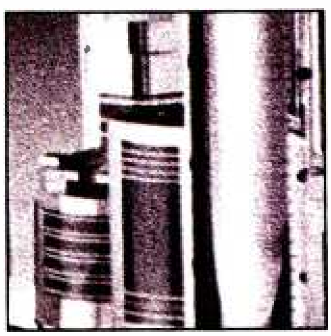

▼ **On the Cover:**
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Marvin Caesar, President

At Aphex we have a problem with the President. Marvin Caesar wants everything the company makes to be the "best." Marvin is *not* an engineer, he is an audio zealot who doesn't understand the word "impossible."

When engineering approached him with a design for a gate, he wasn't satisfied. He wanted the ultimate gate, an *expander/gate* that was extremely fast, perfectly click-free, loaded with features, with traditional Aphex audio quality.

Impossible with available technology they said. Marvin wasn't satisfied.

So, the engineers developed a new VCA, the Aphex VCA 1001. Then they created the Aphex 612 Expander/Gate. Marvin was finally satisfied.

In fact, he is so confident that the Aphex 612 is the world's best expander/gate he is offering $2500 to the person who finds a better one. If you think you know of a commercially available expander/gate that can begin to compete with the Aphex 612, write for full details of this offer.

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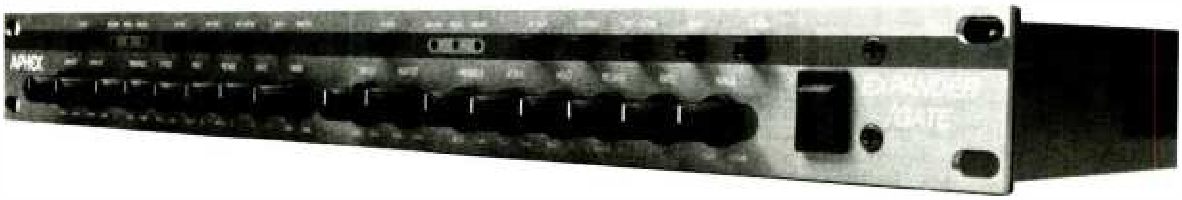

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Circle (5) on Rapid Facts Card

GUEST EDITORIAL

By Wilber W. Caldwell

Being All Things to All People

I used to have this recurring nightmare. In my dream, an important new client booked a big video post session in our studio. He had picture completed on 1-inch type C and wanted to "add the sound."

The morning of the session the guy shows up with the 1-inch video under his arm and a big suit case. He seems pretty knowledgeable. As we thread up the tape, we notice it has two tracks of location dialogue, recorded back and forth between tracks 1 and 2. One track is dbx-encoded; one is not. We sync up to a 24-track and transfer both channels across, decoding only one. No problem. So what's next? The new client opens up the suitcase and begins a monologue that goes something like this:

"I have some location sound here, and I want to go back to the original components." He produces two ¼-inch tapes from the bag; one is stereo FM Nagra format and one is the regular neopilot tone format.

He continues. "We had to replace some of the dialogue. Most of the ADR was done in New York and some in L.A." He pulls out a big 35mm mag roll and a ¼-inch with center-channel time code. "I also have some sound effects," he says, rummaging around in the suitcase and coming up with two 16mm mag loops, three CDs, and a phonograph record. "In addition, in the middle section of the piece I'd like to insert the audio from some documentary material." With this, he presents us with a ¾-inch video cassette and a ½-inch type M video cassette. How big is this suitcase?

"Here's the music." It's a split-track mix with vocals on 1, band on 2, brass on 3 and sync on 4. "I can't remember if the sync is SMPTE or just pilot tone, but it shouldn't matter. Now, we'll need to do some Foley work and add a live announcer, too, and, oh yeah, here is another 1-inch with an alternate dialogue sequence." There is a long silence. "Well, let's mix," he says cheerfully.

In some dreams the suitcase is even bigger and some nights—well, you get the picture.

"It's only a dream," you might say. Yes, but as most psychiatrists will tell you, dreams represent reality, and we can learn from them. They are metaphors.

What can we learn from my nightmares? First, let's discard any notion that the client in this dream is screwed up. He is not. The components are all of good quality, labeled in detail, and they all contain some standard form of synchronous reference. Second, we can learn the importance of preproduction and information exchange, especially at the time of the booking. Third, and most important, we can learn that technologies in the audio for video field are extremely diverse, and that it is incumbent upon the independent studio to be able to deal adroitly with a wide spectrum of formats and configurations. In short, it may be necessary to be all things to all people.

Only in the independent recording studio is sound a powerfully focused specialty.

A common reaction to all of this confusion is to say that our industry is in flux now, and that we are temporarily the victims of "format wars." To an extent this is so, but I submit that this is an industry that will always be in flux. This industry is expanding, diversifying and requiring more and more specialization. The result of expansion, diversification and specialization is not an occasional "hot war" in which a single format victor emerges. It is an ongoing "cold war" characterized by an ever-growing number of format "camps" that don't kill each other off, but only entrench themselves more deeply in their own well-defended, specialized corners of the market.

Another inference can be drawn from the dream. The client's bottomless suitcase not only represents a diverse technology, it is also a metaphor suggesting that the independent recording studio is not in control of its own destiny, regarding the selection of A/V post equipment.

The recording studio can neither begin a production nor end one. The audio mix is just one step in a long, often complex process that begins with the video producer or the animation house. The process ends at the videotape house or at the film lab. Recording engineers cannot control what format comes in the studio door, nor can they control what format goes out. They must honor the conventions, as well as the whims of another industry. They must be able to ask questions such as, "Do you want to layback directly to the 1-inch or would you like a ¼-inch with code? How about a couple of 35mm mags? Should this be 24 or 30 frame? What is the speed reference? Do you want a safety? In what format?" If the large video houses in your marketplace go digital, you must at least be able to supply a digital audio product in the format they prefer, if not to play and layback to the digital videotape itself.

This industry will always be in flux.

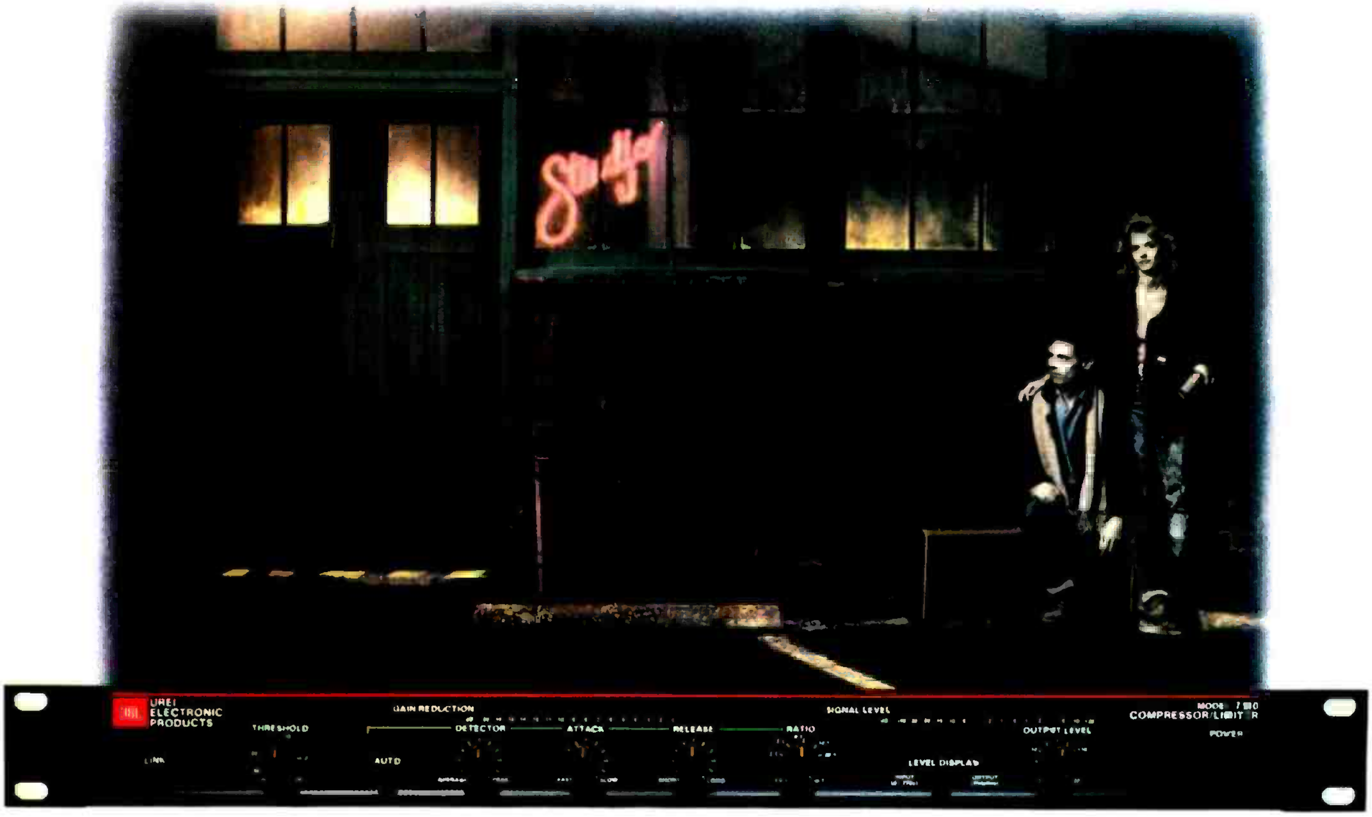
We might infer that the independent recording studio is trapped in a technology that is blindly expanding out of control, a technology that is a black hole for money. This is not true. Remember, the dream is a teacher, not an evil omen. The point is not to insist that a studio must have all formats, but that a studio can select the formats that are most applicable to any potential customer base—the formats that are most prevalent in a particular market area. A studio can target the work it goes after simply by researching the conventions of the producers it wishes to serve. If this proves successful, then another area can be targeted. The more formats added, the greater the potential customer base.

It is all a matter of communication. It is essential that the independent recording studio establishes and maintains active and detailed communication with all audio and video customers. This means that

Continued on page 18.

Wilber W. Caldwell is president of Doppler Studios, Atlanta, and a member of the SPARS Board of Directors.

It's Time To Rack Up Another Hit.



It's hard to follow a great act. Expectations run high. The performance must be flawless. When we decided to carry the legacies of our LA-2A, LA-4 and 1176LN into the next generation, we knew exactly what we were getting into.

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Our new 7110 Limiter/Compressor incorporates the characteristics of its predecessors, is the natural addition to a legendary line and has all the potential to become a major hit in its own right. The 7110 combines both peak and average limiting action, producing smooth, predictable RMS style performance like the LA-2A and LA-4 with the precise automatic peak control of the 1176LN.

The 7110, with our exclusive program dependent Smart-Slope,™ gives you adjustable compression curves from 1.5:1 through infinity:1. You set threshold, attack, release time and output level – the 7110 automatically rides the gain with split second response.

To make set-up as simple as possible, we've included an Automatic Preset function. Punch the button on the front panel – the 7110 automatically defaults to program dependent attack and release times, and presets the peak threshold and ratio to consistently used settings. Perhaps the best news of all, the 7110 produces crystal clean sound and is virtually transparent.

Just another limiter/compressor? We don't believe so. After you've heard it for yourself, we think you'll agree. Stop by your local JBL/UREI dealer and give it a listen. And, get ready to rack up another hit.

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Circle (6) on Rapid Facts Card

LETTERS

Engineers: the first one called

From: Bil VornDick, independent engineer, Nashville.

I enjoyed your editorial in the February issue of *RE/P*. Right on! Except you forgot the first trait an engineer has got to live with. The first one called and the last one paid, usually. He is also the one that is asked to take a pay decrease if the budget is going over, or the studio rate is too high for the budget.

I have been recording live music for over twenty years and in Nashville (Music City USA) for close to ten years. I work mainly on acoustic projects now as I find them more enjoyable and more challenging. A couple that you might be familiar with are two MCA Records Masterseries Artists, Jerry Douglas and Edgar Meyer. I have done just about the full gamut of recording and found your comments to be the way it is.

I am an independent engineer and won't go back to a staff position. I enjoy having the position of being able to say *no*. My first couple of years in Nashville were as chief engineer for Marty Robbins at his studio. Marty was, and still is, the best vocal stylist I have worked with. Being his staff and personal engineer put me in a position that *I had to* record everything that came through the door. The studio manager was a slave driver. You talked about five-day work weeks; those were seven.

Being independent averages me about 5.6 days a week on a 50-week year, which is fine. I enjoy what I do and in most cases do it 14 hours/day. I have enjoyed the gold and Grammy nominations, but the real enjoyment for me is the creative environment of the studio and the music performed there. I love it.

Digital filtering

From: William Sommerwerck, Bellevue, WA.

The piece on digital filtering in February ["Trends in Equalization: Digital Filtering," page 24] has several serious errors.

Analog filtering is considered to be frequency-domain filtering, whereas digital filtering works in the time domain. The author gets this backward. He also fails to point out that digital filtering is but one form of sampled-data filtering. The filter in Figure 3 could be implemented with analog arithmetic and delay just as well.

Figure 3 has two errors. There should be two multiplier boxes after the second delay, not just the one shown. The arrow leading from the input summer to that multiplier is incorrect, because the signal travels only one way.

Finally, the author says that once the impulse response for an FIR filter is determined, its sample values must be subjected to a Fourier transform. This statement is not true. The "waveform" of the impulse response *is* the FIR. Its sample values *are* the FIR weighting factors and need no further processing.

Think audio

From: Thomas W. Earl, chief engineer, Random Bullet Records, Greene, NY.

I want to thank Michael Fay for his editorial "Think Audio" in the April issue. It's nice to hear someone ask questions other than, "What's 3dB overbiased at 10k mean?" There is more to audio than just equipment, there's music. His questions hit right to the heart of the engineer: Do you know music? Too many people are interested in learning how to play with the toys without learning the art of recording and mixing.

However, one comment was made with which I didn't totally agree. Michael's reference to "quickie" audio schools and his doubt of their "usefulness" and their "credibility" sort of wipes out anyone who has neither the time nor the money to go to a four-year college.

That was my case. However, the pendulum swings both ways. Recently, I had a conversation with someone studying music and audio in his third year of college, and he knew nothing of the basics of audio, such as electromagnetics. He never even sat down behind a board in a control room.

In the six months I went to school, I was taking on assignments such as live remote recordings in clubs like the China Club in New York without any supervision, and I was expected to produce a quality recording.

So, let's be careful how we lump everyone into one group and cut down the whole group. Among that group may be an excellent engineer waiting to grow. Thanks again for your advice, Michael.

Michael Fay replies:

I agree that it is not good to "lump everyone into one group and cut down the whole group," but I think I adequately qualified my comments by saying "...Before going further, I admit there are a few good schools out there—mostly they are four-year universities with degree programs."

Doesn't the true measure of any educational system lie in the acceptance of its graduates into the next level up, which, in this case, is the professional audio market? It has been my experience, and the experience of many of this industry's top studio owners, that a six- to nine-month course (often costing thousands of dollars) seldom qualifies a person to enter the job market on any level. Perhaps an important point to bring out is that an "exceptionally talented" person will do well regardless of the quality of education received.

What seems to be missing from both our remarks are the tried-and-true intern/apprenticeship programs that got many of today's top professionals into the business. Has this system become extinct in pro audio? I certainly hope not. If it has, then maybe the facility owners themselves need to take some blame.

I can't help but think that the short-term "trade school" approach is a result of society's need for the "quick fix." Trade schools definitely have their place for those who want to become welders or truck drivers, but in pro audio you just can't buy experience.

Digital limitations

From: Bik Toor, Digital Transformations, Los Angeles.

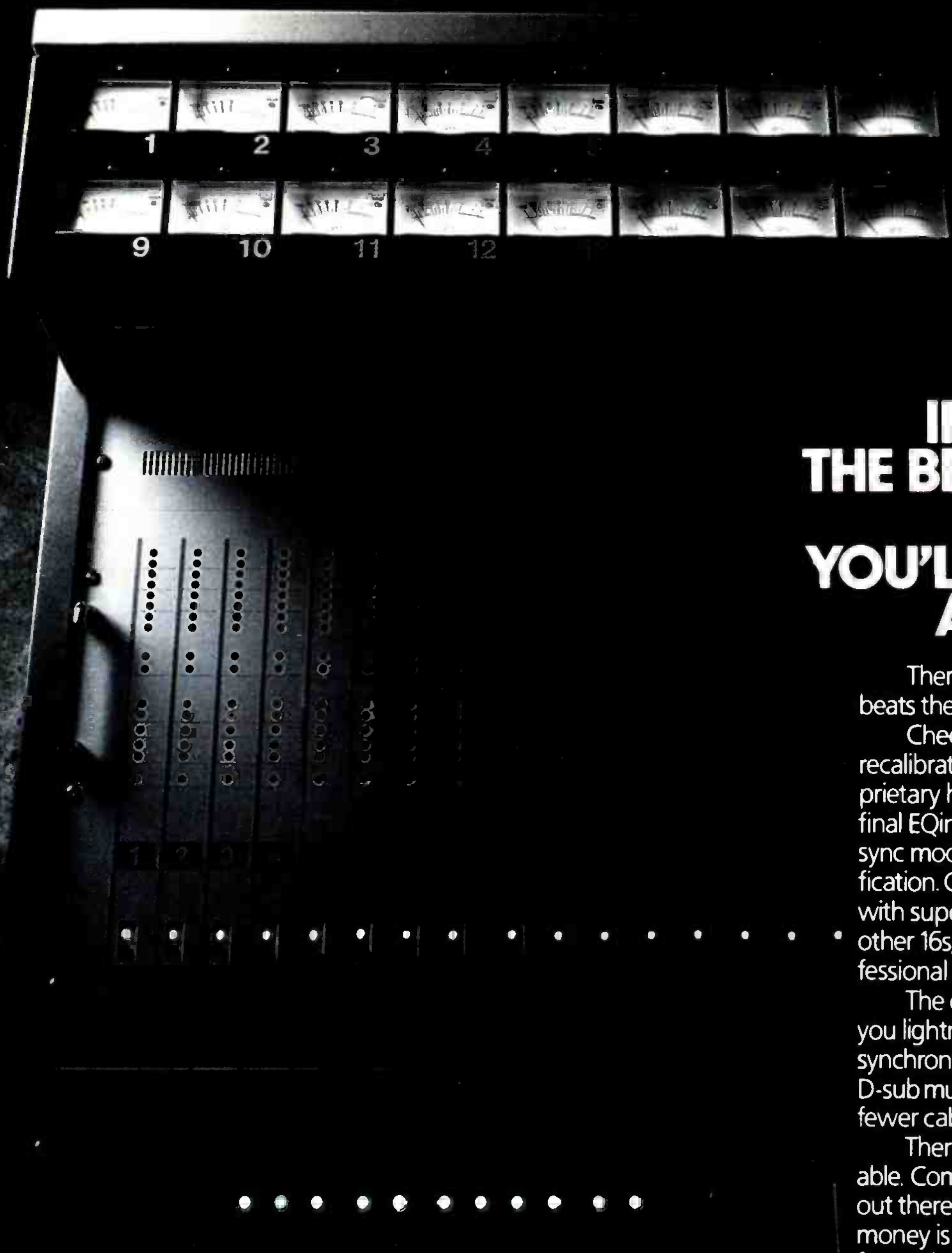
I don't know if I should laugh or thank John Lord ("Letters," December 1987 issue) or rather yourselves for publishing his letter. Such absolutes, as expounded in the letter, are highly irrelevant and I wonder what purpose they serve. Perhaps to assure mankind that digital is not magic?

It would be similar to me informing everyone that they will eventually die and not live forever, that the universe will collapse and not expand forever (although "they" are not yet certain on that).

I'm sure you get the point, as I'm equally sure your readers will.

RE/P

Write to us at Letters, *RE/P*, 8885 Rio San Diego Drive, #107, San Diego, CA 92108. Letters may be edited for length and clarity.



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Circle (7) on Fast Fact Card

NEWS

IRS relaxes capitalization rules

By Kathy Mickelson, editorial assistant

The Internal Revenue Service has relaxed capitalization restrictions on tax deductions for creative projects, by ruling that half of incurred expenses may be deducted in the year they are incurred rather than the period in which they generate royalty income.

Under the 1986 Tax Reform Act, freelance writers and other creative people were required to spread deductions over the entire period of time the project generated royalty income. Before tax reform, 100% of deductions could be taken the year the expenses were incurred, and various creative groups complained that the 1986 rule would impose undue creative hardship.

Although the change was cited as a step toward improvement, various creative groups said they would try to get Congress to restore the pre-1986 rules.

Artists, songwriters and other creative professionals recently have protested the capitalization issue. Under the tax reform act, which requires uniform capitalization, artists are required to capitalize tools of their trades, and most creative people view this requirement as unfair.

Shelly Loomis, staff intern for Sen. Bill Bradlee (D-NJ, one of the architects of the tax reform bill), said creative people have complained that capitalizing their tools is often impossible. They find it difficult to match every loss with every gain.

"This act has created quite a furor," Loomis said. Bradlee and Gina DeSpres, his legal counsel, have been trying to define what artists do and who should be classified in this category to find a solution to the problem.

The House Ways and Means Committee has introduced a technical corrections bill to the Tax Reform Act that would exempt artists; however, according to Loomis, Bradlee doesn't expect it to pass. Technical corrections bills are designed to make minor changes in previously passed acts, but corrections involving potential losses in revenue are usually squelched. A similar bill introduced in 1987 failed to pass for that reason.

Imports and exports up for keyboards, synths

Exports and imports of keyboards and synthesizers increased in 1987, according to the American Music Conference.

Interpreting statistics from the U.S. Department of Commerce, the AMC said that U.S. imports of electronic instruments with one keyboard reached $242 million in 1987, totaling 4 million units. This was a 52% increase compared to 1986 figures. Most of these imports, 94%, were from Japan.

Synthesizer imports to the United States posted larger gains, up 96% to 511,500 units in 1987. Exports from Japan rose 28% to 300,177. Korea was the second biggest exporter.

For U.S. exports to other countries, synthesizers posted the second largest increase in dollar value among all categories of musical instruments—up 106% to more than 13,000 units. Only electric guitars posted a larger increase, 107%. The Netherlands and Germany were the largest importing countries.

Overall, according to AMC, musical instrument exports in 1987 posted the second-largest year-to-year gain in 20 years.

MIDI convention set for West Coast

Following the success of the first MIDI Expo, a second conference and exposition, MIDI Expo West, has been scheduled for Sept. 10-11 at the Anaheim, CA, Convention Center.

The convention will include a seminar program featuring MIDI basics, computer music, song construction, sampling techniques, the MIDI studio and an artists' panel.

For more information, contact Tony Scalisi, show manager, at Expocon Management Associates, 3695 Post Road, Southport, CT 06490; 203-259-5734.

B&K workshops set

Bruel & Kjaer is sponsoring a series of seminars and workshops for various cities throughout the rest of the year. The seminars are designed to help personnel keep current with new developments and to equip them to meet future technical challenges.

Topics being offered include "Acoustical Noise Control," Oct. 18-20, Foster City, CA; "Digital Signal Analysis for Applications in Sound and Vibration," Oct. 4-6, Minneapolis; "Fundamental Measurements in Electroacoustics," Nov. 10, Anaheim, CA, and Dec. 6, Orlando, FL; "Product Noise: Measurement, Evaluation and Control," Sept. 6 in Marlborough, MA, Oct. 24 in Milwaukee, Oct. 26 in Minneapolis, and Nov. 10 in Orlando, FL; "Sound Intensity Theory and Measurements," Nov. 1-3 in Livonia, MI; and "Special Analysis in Sound and Vibration," Oct. 25 in Milwaukee, and Oct. 27 in St. Louis.

For more information, contact Julie Pelz at Bruel & Kjaer, 185 Forest St., Marlborough, MA 01752-3093; 617-481-7000.

Yamaha opens Communication Center

Yamaha has opened the Yamaha Communication Center Show Room at Metropolitan Tower in New York..The center exhibits Yamaha's musical instrument line and pro-audio equipment. The location also features the company's Research and Development unit, which opened in December. Instruments on display include a selection of Yamaha pianos, DX-series synthesizers, an electronic keyboard, and woodwind and brass instruments.

News notes

Cetec Gauss, Sun Valley, CA, has signed an agreement with RCA/Ariola Music to provide the record company with high-speed cassette duplicating equipment for RCA's North Carolina duplicating facility.

Clear-Com Intercom Systems has moved to 945 Camelia St., Berkeley, CA 94701; 415-527-6666.

Bose Corporation, Framingham, MA, is offering music software to people who buy a pair of Bose Pro RoomMate loudspeakers and an Apple Macintosh or MIDI interface. Buyers will receive their choice of Great Wave's ConcertWare + Version 4 or Terpsichore, Opcode's MusicMouse, Coda's Macdrums or Intelligent Music's software package. The promotion runs through Oct. 8.

Citing the increased cost of doing business, **Maxell Corporation of America,** Fairlawn, NJ, has raised the price of its pro-

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For more information regarding TAD speaker components, please contact Leon Sievers at Pioneer Electronics (USA) Inc.

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Circle (8) on Rapid Facts Card

NEWS

fessional/industrial audio/video pancakes and cassettes by 15%, effective June 1.

Fane Acoustics, Des Plaines, IL, has entered into an agreement with systems design consultant Stephen Court, who will design a new range of professional enclosures and monitors that Fane will manufacture and market. The equipment will be sold under the "Fane by Stephen Court" name.

AMS, Seattle, has named the following U.S. companies as factory-appointed representatives for the AudioFile: Douglas Ordon & Company, Chicago; Harris Audio Systems, Miami; Harris Sound Incorporated, Los Angeles; Studio Consultants Incorporated, New York; and Valley Audio, Nashville.

Shape Incorporated, Biddeford, ME, has increased the price of its compact disc jewel cases by 5%, effective June 1.

Promusic Incorporated, Ft. Lauderdale, FL, has been appointed U.S. agent and distributor for the Parry Music, Boosey & Hawkes, and Dwight Cavendish music production libraries.

Westlake Audio, Los Angeles, has been named the exclusive dealer in the Los Angeles area for the Jensen Twin Servo 990 mic pre-amp.

Passport Designs has announced that its first quarter earnings increased by more than 503%, with revenues increasing by more than 178%.

KR Design has teamed up with EagleVision, and is now providing print, audiovisual and video projects for corporate, agency, marketing and public relations companies. The company has also moved to 3001 Summer St., Stamford, CT 06905; 203-348-0722.

Sellmark Electronics, Durham, England, has signed an agreement with Outboard Electronics for the worldwide marketing of Outboard's MF100-S motorized fader.

People

Michael Wood has joined Fane Acoustics, Des Plaines, IL, and is overseeing the company's sales and marketing.

George Armes, general manager for instrumentation and data tape for the Ampex magnetic tape division, Redwood City, CA, has been appointed to the Electronic Instrumentation Technical Advisory Committee.

Phil Guy has been appointed marketing manager for Soundtracs, Surrey, England.

William Dexter has been named sales manager for Audiotechniques, New York.

Lee H. Gray has been appointed vice president and regional sales manager for Recorded Publications Laboratories, Camden, NJ.

J. Michael Hughes has been named vice president of marketing for HM Electronics, San Diego.

Mike Halleck has been named southwest regional manager for Studer Revox America, Nashville.

Michael Mueller has joined the staff of TekCom Corporation, Philadelphia.

Ralph Goldheim has been named national sales manager for Alesis Studio Electronics, Los Angeles.

Agfa-Gavaert, Ridgefield Park, NJ, recently announced its 1987 sales awards for the magnetic tape division. **Walter Bremer,** Pacific region sales manager, was named regional manager of the year. Winners of the sales contest were **Jim Rouse,** Pacific region, first place; **Mike Caputo,** Atlantic region, second; and **Jeff Hamilton,** Central region, third prize. The outstanding performance award went to **Kathleen Smyth,** marketing controller.

Jeremy Bancroft has joined Digital Audio Research, Surrey, England, as sales manager. RE/P

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MANAGING MIDI

By Paul D. Lehrman

MIDI Is Coming of Age

Several anniversaries are being celebrated around my house this month. Ten years ago, I published my first article on professional audio. Five years ago, MIDI was officially adopted. And two years ago, I started writing this column. A lot has happened in professional audio in the past 10 years, but, for me, none of it has been as exciting as what has happened to MIDI in the past two years.

MIDI has had a lot of successes, perhaps the most visible being the extent to which it has been accepted in the music production studio. Two years ago, only the more daring and computer-comfortable studio owner would invest in a MIDI system. Today, in the Western world, there are very few original music facilities that don't have at least a couple of synths, a drum machine, and a hardware- or computer-based sequencer.

Helping MIDI's acceptance into the professional world have been major developments in the area of synchronization. New forms of sync, such as "smart" FSK and MIDI Time Code, can make linking mechanical devices and sequencers much more economical and easier to use.

Also helping are new types of software specifically geared to the audio/video studio. This software can handle multiple events and tag them to real-world (not MIDI) time, provide printed music or pre-timed sequences to follow visual hits, and deal with numerous disparate events from a central controller: the computer.

One originally perceived drawback of MIDI, its relatively low transmission speed, is being partially overcome by hardware and software advances that address multiple data streams. This allows the capacity of a MIDI system to be doubled or even quadrupled, without fear of rendering any existing equipment obsolete.

Non-keyboard controllers, originally limited to clumsy, slow, guitar-like devices, or the rather inaccurate (but expensive) pitch converters, are now far more usable and affordable. There are wind drivers, percussion sensors and even "gesture controllers" that you don't have to touch to make music.

Also, there are now MIDI-controlled processors for just about every studio processing function. Even mainstream manufacturers are getting on the bandwagon by either designing MIDI functions directly into their equipment, or hiring outside companies to build MIDI retrofits.

This revolution is far from won, however. The more complex devices have a steep learning curve and require a lot of patience and support if they are to be used to their full potential.

Finally in the success column, do-it-yourself MIDI programming has become a reality. Once, the concept of a "MIDI driver" (an assembly-language program that could read and write MIDI data through a computer's serial interface) was the exclusive domain of a kind of high priesthood of programmers. But today, inexpensive or public-domain MIDI drivers are available for languages such as BASIC, Pascal, C, and even Apple's new Hypercard. Now, anyone interested in programming can write custom MIDI applications.

The promise of MIDI remains only partially fulfilled, however. In the world of text and graphics, there are standard formats for exchanging data among computers, even if those computers are completely different. In the MIDI world, that isn't quite true—yet. The proposed "MIDI Files" standard is a great idea, and a couple of manufacturers have adopted it, but it is still not carved in stone, and discrepancies exist in the way various programs are interpreted.

The industry also needs to adopt a protocol for sending a MIDI File over a MIDI line. Some manufacturers even have stated that they have no plans to make their products MIDI File-compatible, under the assumption that anyone who buys their stuff couldn't possibly want anything else. This is not progress.

One casualty, caused by the delay in standardizing formats, is the idea of being able to "phone in your parts or tracks." This idea, which is even *older* than MIDI (and recently popularized in the Sunday funnies) is rarely practiced. Phoning in your track is possible, but it is still too complex. It requires that the sender and receiver have sophisticated telecommunications software and identical music software (not to mention hardware).

The concept of a multiprogram environment for MIDI applications, in which different program modules (e.g., patch editors, processor controllers, sequencers and automation systems) can operate simultaneously, represents a necessary step forward in MIDI control of production. Although a few manufacturers recognize this need and offer various programs that work in conjunction with each other, as yet, there is no standardization for most programs, even those designed for the same computer.

Heading the "Not Yet" list are products that are completely reliable from their first release. Most manufacturers who survived the first few years of MIDI have cleaned up their acts, and today their products are more solid than their initial offerings. But start-up companies are appearing all the time, and the percentage of their products that actually do what they're supposed to do is still depressingly low.

Then we have the elements that fall into the "Real Soon Now" category. That's the phrase currently used by developers to describe when they're finally going to release that product they've been promising for the past five years. A "Real Soon Now" hasn't happened yet, and it looks as if it may never.

Another "Real Soon Now" capability is MIDI control of mechanical studio equipment, such as audio and video tape decks, hard disk storage systems and CD players. MIDI is a terrific protocol for digital control, and with MIDI Time Code now part of the spec, it can easily be used in both off-line and real-time modes. However, it will never reach its full potential until hardware manufacturers allow their machines to be controlled by MIDI. So far, there hasn't been a lot of enthusiasm for the idea.

An additional step that will be needed to make MIDI hardware control a reality is the introduction of MIDI Local Area Networks (LANs), which will be able to overcome any remaining speed limitations inherent in MIDI and will prepare it for the next generation of hardware and software. A standard LAN protocol would be nice, but, more likely, each manufacturer will come up with its own, and those of us on the sidelines can only pray that they work.

It's been an exciting couple of years. Thanks for reading. RE/P

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

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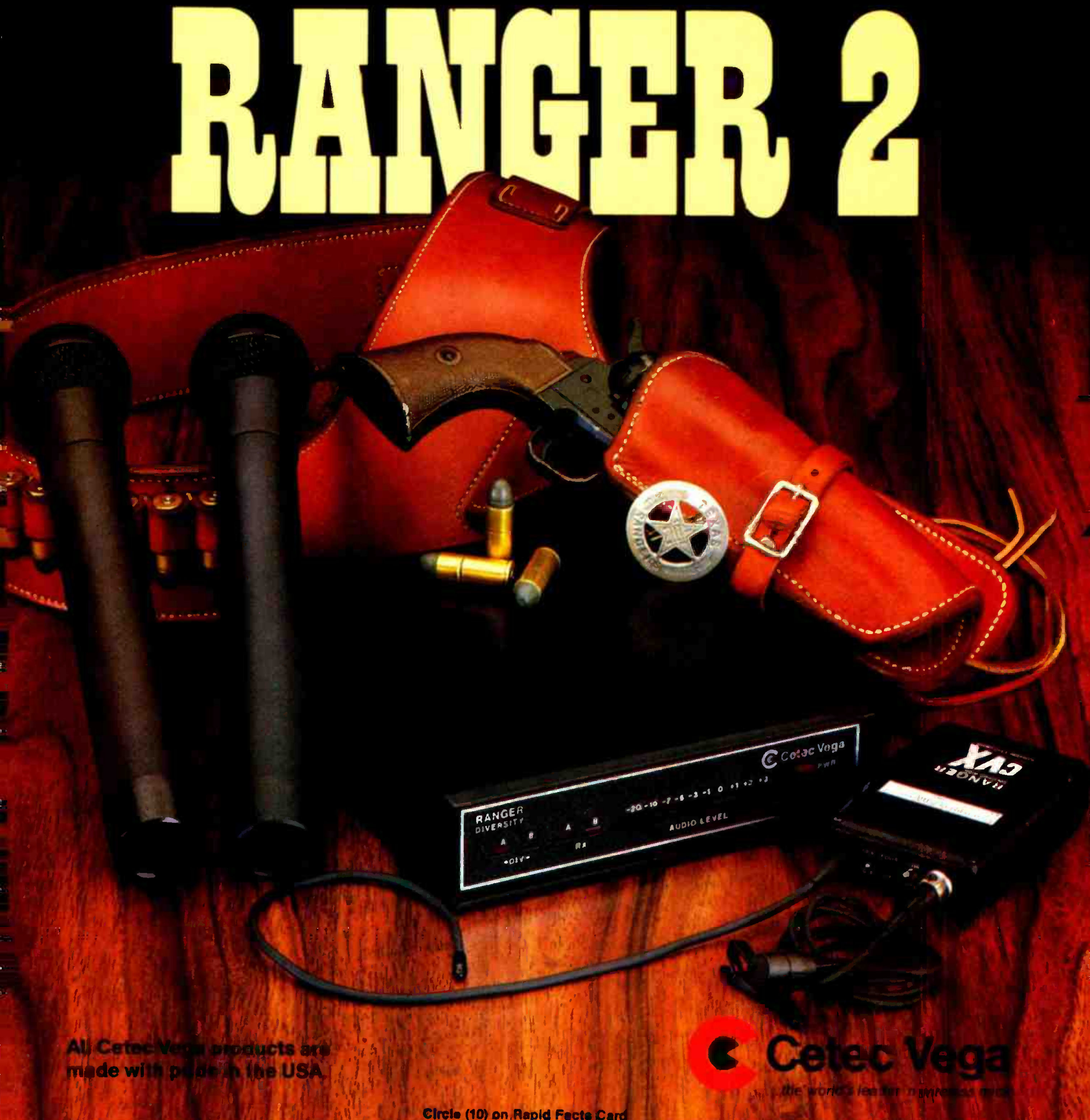


Circle (10) on Rapid Facts Card

SPARS ON-LINE

By Murray R. Allen and Tom Miller

"Multi-Interface Stress Syndrome"

As I regained consciousness in my hospital bed, the doctors were feverishly reading my vital signs. My blood pressure was floating between 40 over 20 to 280 over 120, and my pulse was bouncing between 40 and 120 beats per minute. The young resident commented, "A classic case of recording studio stress."

I was told that I had collapsed in my office and had been rushed to the hospital just five hours ago. The medical team asked me if I remembered what led to this condition. I waited for a few minutes because I was still somewhat confused, but it all started coming back to me.

I remembered that I felt great as I jumped out of the bed that morning. The sun was out, business was good, and I didn't have a hangover. For some reason, it crossed my mind that we still have problems with the standard XLR connector. Even though America joined the rest of the world (at least on paper) in 1972, most American-made equipment still makes pin No. 3 the hot one.

As I started to brush my teeth, my thoughts went to ¼-inch connectors. Most small equipment (de-essers, equalizers, etc.) have ¼-inch connectors. Often, they are stereo connectors for balanced connection. Usually, they are a fraction of an inch too large to work with a standard patch cord (¼-inch TRS telephone type). The mono ¼-inch jacks don't work at all with patch cords. As I rinsed my mouth, that great feeling of a new day was beginning to fade.

The coffee maker beeped that my coffee was ready, but I was preoccupied with the problems of connectors on speakers. I became furious that banana plugs don't fit large speaker cables. The PA industry has been arguing over a standard speaker connector with no success. As I added cream to my coffee, I pondered balanced inputs and outputs. I thought about the fact that floating balanced outputs are compatible with unbalanced inputs, but non-floating balanced outputs are not. All equipment really should be floating balanced—in and out. Lots of equipment isn't even balanced.

As I jumped into the shower, I was really starting to steam. The shower drain was plugged, and as the water level rose, my mind shifted to the way in which audio levels are controlled on professional equipment. I mulled over the specs: In the broadcast industry, "0VU" is +8dBm on most equipment, but on some it's +4dBm; +4dBm is used with most studio equipment; "0dBm" with a lot of PA equipment and industrial video equipment is +8dBm, +4dBm, 0dBm, −10dBm and −20dBm. Music synthesizers and drum machines have a level all their own, and sometimes a mind of their own as well.

I started to shave, and my stomach began to rumble and ache. The ongoing problems with loading made me shiver. Broadcast equipment usually comes with 600Ω input and output impedances, which do not work well over long cables or when using Y-connections. Most studios and hi-fi people use bridging input and low output impedance on their machines.

As I leaned over to tie my shoes, I felt dizzy. I was having trouble aligning my right shoe lace with my left one and getting an even bow. Suddenly, I remembered 1-inch, C-format VTR audio alignment tapes don't match between brands. And ¾-inch videocassette alignment tapes differ between brands and have only two reference tones (1kHz and 10kHz). Tapes from these machines are often used in broadcast and the circulation of music video to television stations.

I really should have stayed home that day, but I forced myself out the door and grabbed a taxi for the studio. As the driver flicked on his meter, I was troubled by the thought that so many meters are marked as VU meters but bounce way too much to match the standard. Identical drum tracks can look very different on different VU meters. Almost no video machine seems to have a true VU meter. And almost no video machines have peak indicators, which are common on even the cheapest consoles.

I tipped the driver and felt as if my body was being invaded by some unknown force. I was possessed with the thought of the slight difference in sound when the polarity of both channels of a stereo pair is flipped. The SMPTE has recommended a practice to standardize polarity on magnetic recordings. No manufacturer or test tape maker seems interested in matching the SMPTE recommended practice.

As I walked into the studio, my heart was palpitating wildly. All those digital formats: PD, DASH, DAT, PCM-1630, D2, CD, AMS, Synclavier, etc. If only all these formats could talk to each other. Professional AES/EBU ports almost work with consumer digital ports. A small change should make conversion between professional and consumer digital links a simple matter of changing voltages, not digital codes. Also, a lot of digital gear doesn't have AES/EBU ports. They are often shown in catalogs as an option not yet available.

As I entered my office, I began to lose my balance. I grabbed for the phone to call for help and then thought about the standard that exists to make all remote control ports work the same. No audiotape machine manufacturers support this standard. Some synchronizer and video editor companies are pursuing it. It is possible, however, to buy consumer remote controls that work with all consumer VTRs and TVs. Why not with professionals?

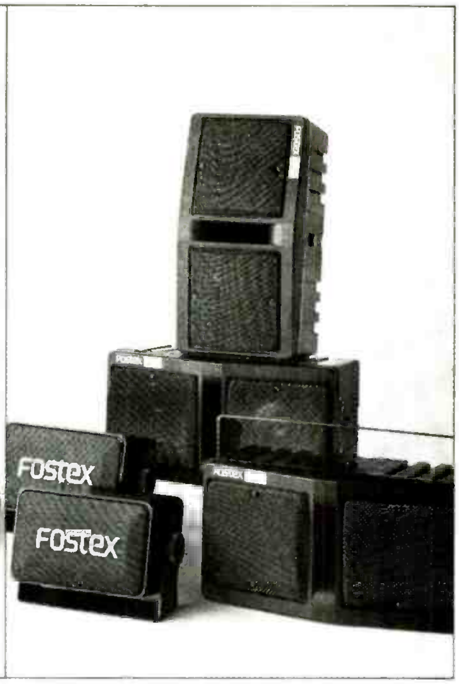
Then I must have blacked out because that's all I remembered when I woke up in the hospital.

My physician told me I was a victim of MISS—Multi-Interface Stress Syndrome. "Those manufacturers probably don't love you as much as they have led you to believe," he remarked. "In fact, they probably think of you only when trying to sell something new or trying to collect on an old invoice. Just remember: It's a real war zone out there, and drink lots of juice."

Feeling somewhat relieved in knowing what was wrong with me, I decided not to think about the lack of interface in our industry. I am quite sure every industry has similar problems. Who ever heard of a Jarvic heart machine being able to talk to a kidney dialysis machine? But studio owners would probably live longer and fuller lives if some of these format, standards and interface problems were solved.

I left the hospital in a wonderful mood. As I meandered up the street, I thought how nice it would be to have a special version of software beyond alpha, beta and final version—something that really had all the bugs worked out.... RE/P

Murray R. Allen is president of Universal Recording Corporation, Chicago, and is a former president and chairman of the board of SPARS. Tom Miller is chief engineer of Universal Recording.



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Circle (11) on Rapid Facts Card

By Jeff Burger

Disk Operating System

If you've been around computers at all, you've probably heard the term DOS, an acronym for Disk Operating System, which is the interface between you and your disk drives for operations like loading, saving, copying and deleting.

A few columns back [May] we likened a formatted disk to a parking lot. Well, DOS is your parking attendant. If we had to deal with tracks and sectors every time we wanted to manipulate disk-based data, the word *computers* would not even be in our vocabulary! You just hand the keys to DOS and let it worry about jockeying your data around.

When you first boot (start) most computers, about the only thing they can do is look for a disk with DOS on it. DOS actually gives computers much of their personality. Besides managing disk operation, most DOS systems actually reside on a disk as well. This provides manufacturers the flexibility of upgrading the operating system in the future by simply supplying a new version on disk instead of dealing with hardware upgrades.

We now demystify Common Computer Question No. 1: Why can't an IBM read an Apple disk and vice versa? Primarily, it's because different operating systems store and keep track of data in different ways. All disks are subdivided into tracks and sectors, but DOS usually labels sectors and files by placing proprietary "header" and "footer" information at the beginning and end to help identify them.

More importantly, DOS maps the location of each file on the disk, just as the parking attendant puts the keys to each car in a cross-referenced place. Complicating the matter, sometimes a disk has enough room to save a file, but it may not be consecutive space because of file deletions and other considerations. This means the file gets broken up and the segments get stuffed into whatever spaces are available. (Let's be glad this doesn't happen with our cars!) A record has to be kept of the location of related segments. As you might have guessed, all this is done differently on each system.

Let's take a look at the major tasks that an operating system performs for you. The command names and implementations vary slightly from system to system, but the concepts are fairly universal. Further, various systems provide different means of specifying which disk drive you wish to address.

The first command that comes into play is formatting. After you buy a blank disk, you must first format it for the computer you're using. Formatting is analogous to painting the lines on the parking lot to define the spaces. Once you have created some type of file (for example, you've used word processing software to write a letter or contract), a save command is necessary to store the file permanently on disk so that you can retrieve it later. Naming your file is part of this process. If you modify your document and save it without changing the file name, it will replace the existing one on disk. Or, you may give the modified version a slightly different name if you want to save both. Files should be named as intuitively and plainly as possible—six months down the road, you'll thank yourself!

DOS actually gives computers much of their personality.

Files should be named as intuitively and plainly as possible.

Once you've saved several files, you'll want a way to verify the disk's contents. This list of files is known as a directory of the disk, and options are usually provided to view the size and date of each file, as well as the amount of free space remaining. Retrieving a file is done by issuing a load, open, or call command and specifying the file name. Rename commands let you change the file name at any time. The delete command removes a file from the directory. Provisions are made for copying a file from one disk to another, as well as replacing the entire contents of a disk with the contents of a different disk.

These basic DOS commands provide the means for just about anything you might need to do with your disk files. Equally important is the arrangement of files on disk. It's not as crucial with floppy disks, but imagine for a moment a 20Mbyte hard disk. If you haphazardly put all the files on the disk, that's like opening the door to your office and throwing your papers on the floor—as a filing system. In real life we use drawers in filing cabinets and files within them to organize our work. The same con-

Continued on page 18.

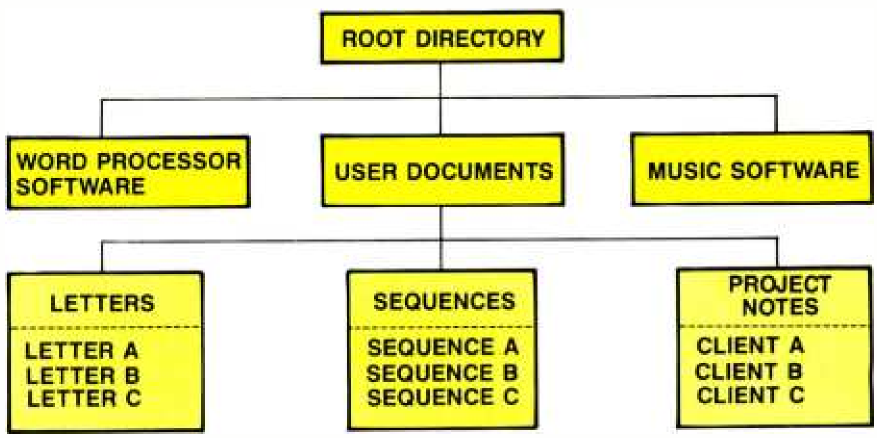

Figure 1. To help organize files, directories can be broken down into subdirectories.

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

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Circle (12) on Fast Fact Card

UNDERSTANDING COMPUTERS

Continued from page 16.
cept applies to today's computers using subdirectories. The main directory (also called the "root directory") should consist of a tree structure of other lesser directories, which, in turn, can contain files or even more subdirectories. The more graphically oriented systems even depict subdirectories as folders.

As an example, your main directory might consist of subdirectories that contain programs, project notes, personal documents and musical sequences. Each of these items might then contain other levels and, finally, the files themselves. (See Figure 1.)

The main directory should consist of a tree structure of other lesser directories.

This technique lets you find files much faster and makes life in the fast lane a lot easier. Another benefit of organizing files into subdirectories lies in the backup process. If your hard disk has lots of files, yet you know that since the last backup you have modified only the business documents subdirectory, you can get away with backing up just that subdirectory.

Some operating systems, such as the MS-DOS found on PCs, provide for more-sophisticated tasks, such as batch files. Batch files allow you to define a series of instructions that are executed automatically with one command or when you boot up your system. For example, you might have your PC automatically load some pop-up resources like your calendar and phone book, then run your word processor program and change subdirectories to access your business correspondence. This process definitely beats typing all those commands each time your start a session!

Next month we'll review the brains behind every computer—the microprocessor. Until then...Happy Computing!

RE/P

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GUEST EDITORIAL

Continued from page 4.
each customer or potential customer must be made aware of exactly what the studio *can* and *cannot* do, and the studio must be aware of the needs and preferences of each customer. Failure in this area is the beginning of the nightmare.

In short, the independent studio must get its head inside the production process from beginning to end. And for video, it can be a very complex process indeed. Every project is different; each is assembled in a different order, by different people, at different facilities, using different equipment. Communication is the name of the game. My nightmare would have been just a pleasant reverie if only I had known what was in the suitcase the day before.

This brings us neatly to a point passed over in my dream. In the beginning, the client was blameless and the components supplied were in every way correct. With most real-life projects, this is not the case.

Recording engineers cannot control what format comes in the studio door, nor can they control what format goes out.

When a director is losing the light and has two more setups to go, he probably will not reshoot a take because of a flaw in the audio. He may have a crew and a cast of 30 or more. At this point, $175 per hour for dialogue replacement and a sound fix is a bargain. Here again, the independent studio is in a position to soar.

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It is easy to forget to listen when you are looking at the picture, but listening sets the audio industry apart from the video industry. Listening is what gives the recording studio a place in the complex, flashy world of video. Remember, half of the art of communication is listening, and listening is what recording studios do *best*.

RE/P

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Noise Modulation in Digital Audio Devices

or Who Wrapped the Mics in Sandpaper?

By Richard C. Cabot, P.E., Ph.D.

Proper machine alignment can make a significant difference in the sonic quality of digital tape machines.

Many studios today are going digital. Some of the presumed benefits are improved sound quality and a reduced need for maintenance. Most types of gear are available in digital configurations, including tape machines, delays, reverbs, equalizers and limiters. Digital delays and reverbs have been around for several years, but they are normally used for effects and are added to the mix at a level significantly below the main signal. Their audible oddities were usually masked by the substantially louder main signal. Besides, other than springs (a little outer-space sound), plates (large and expensive) and special purpose rooms (very large and very expensive), what were the alternatives?

Tape machines are another story. The top-line analog machines with noise reduction really are quite good. Digital machines are expensive and, according to some people, don't sound natural. Others swear by digital machines and find no audible faults at all.

This article discusses one possible reason for the differences and tells you how to measure a digital machine for this effect. What's more, because most professional machines have adjustments that will affect their performance on this test, you can fine-tune a machine once you know what you are looking for.

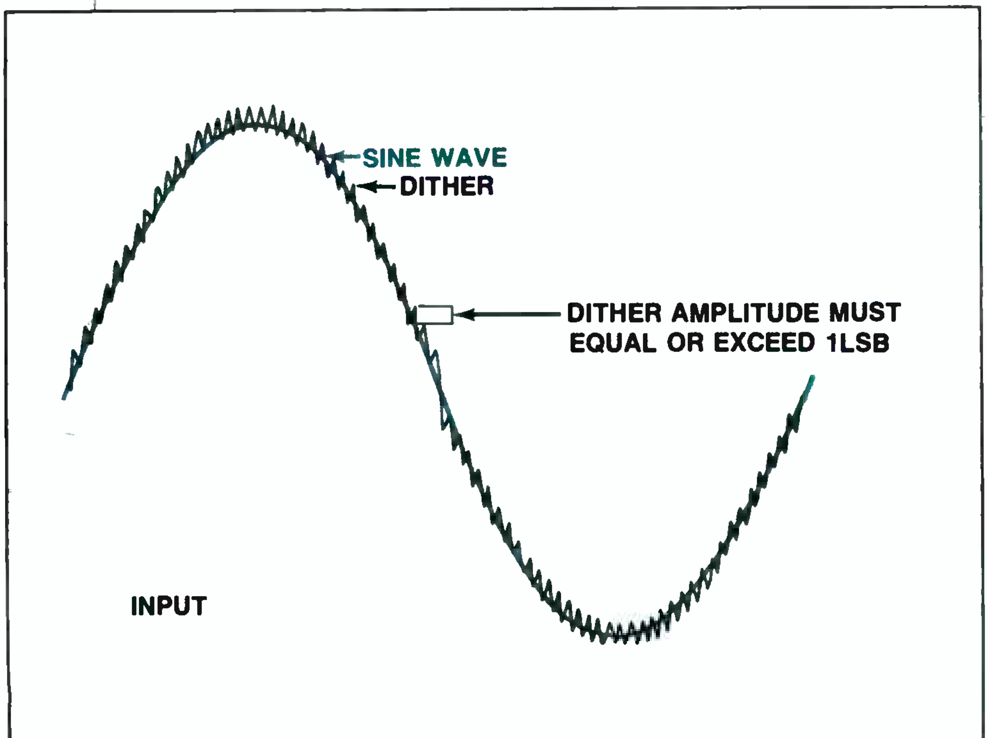

Figure 1A. Dither causes average value out of A/D to be correct. Errors in each sample become random and sound like noise.

Turning audio into bits and bytes

The critical parts of any digital audio device are the A/D and D/A converters. They convert the analog signal to binary numbers the digital circuits can process. The original analog signal can take on an infinite number of voltages between its positive and negative peak values. This signal must be converted into a finite number of different digital words by the A/D. In a 16-bit system, 65,536 different values can represent the signal. If the input to the A/D corresponds to one of the available output values (or codes), the converter uses that code. What happens if the signal doesn't match one of the codes? The converter selects the one closest to the input signal voltage. This process of converting the continuous signal into a finite number of output values is called *quantization.*

The quantization of analog signals in A/D converters introduces errors because of the finite number of levels available in

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the digital code. The errors are commonly referred to as *quantization noise.* The quantized signal may be viewed as the original signal plus an error signal. The error equals the difference between the actual signal voltage and the digital representation. This difference consists of two parts. The first is caused by the quantization process itself. The reduction in resolution of the signal at each sample results in an error signal. The second part is caused by the errors in the converter steps relative to an ideal quantizer.

The signal is converted to binary numbers many times each second, a rate known as the *sampling rate.* In professional audio systems, this sampling generally occurs 48,000 times per second. If the sampling rate is a fixed ratio to the signal frequency, the staircase wave shape results in harmonics of the signal. Aliasing causes all input signals above one-half the sampling rate to fold over into the audio band. For example, a 1kHz tone recorded on a 48kHz sampling rate professional machine will result in 24 harmonics within the passband of the machine. If the signal frequency is not locked to the sampling rate, the harmonic spectrum will shift with time as the number of samples per cycle and their location on the waveform changes. Music and speech consist of signals whose frequencies and amplitudes change constantly. Even a single note from a musical instrument does not keep a constant frequency from beginning to end. (But that's another subject entirely.) The changing parameters of the signal result in a changing spectrum at the output of the converter.

In other words, the error signal is dependent on the original signal. For individual sine waves it consists of discrete frequency components. The sharp discontinuities introduced by the quantizing result in extremely high-order harmonics. For sine waves that are not submultiples of the sampling rate, these components alias down to non-harmonic frequencies. This result of the "folding down" effect occurs when the input frequency is greater than one-half the sampling rate. The fold-over equals the distance from the one-half point of the sampling rate to the actual input frequency. For example, if the input is 25kHz, and the sampling rate is 48kHz, the one-half point is 24kHz, and the resultant fold-down frequency is 23kHz. A 28kHz input would result in a 20kHz sample and so on.

At 20kHz, the fold-over may not seem to be cause for concern, but consider that the 3rd harmonic of a 10kHz signal (30kHz) will result in a fold over sample of 18kHz, which is a non-harmonic component of 10kHz.

When multiple sine waves are present, these quantization products intermodulate, causing many components all across the audio band. Thus, the error signal tends to look like noise, in both the time and frequency domains. However, this noise is dependent on the signal voltage at each sampling instant.

Very small signals result in substantially more distortion than large ones. If the signal is a sine wave smaller than one quantizing step midway between two quantizing levels, the converter doesn't put out any signal at all. If the same signal is applied at one of the quantizing levels, the output from the converter is a square wave.

Correctly dithering an ideal converter will make the noise voltage at any instant statistically independent of the signal voltage. This concept is illustrated in Figure 1A. The error signal will no longer depend on the input and will truly be a noise. However, if the converter is not ideal, or if the converter is not correctly dithered, a dependence on signal level remains. A non-ideal converter has different size steps along its input vs. output characteristic. The dither appropriate for one size step is too large or two small for the other size steps. (In practice, too large a dither waveform is much less of a problem than too small a dither.) Similar problems occur from unequal quantization steps in D/As when the digital signal is reconstructed into analog.

Measuring the effect

One method of measuring the quantization problems in converters is to measure their amplitude linearity. If a signal 20dB below full scale is input to an audio device, the output should be 20dB below full scale. If the input is 40dB down the output should be 40dB down, and so on. In other words the gain should be constant with signal level. To measure the gain, apply a sine wave, usually 997Hz, to the input and measure the amplitude of the output with a meter. (997 is becoming a standard because it has no related integers to any even-numbered digital sampling rates such as 44.1kHz, 48kHz, 50kHz, 96kHz or 100kHz). Change the input by known amounts and note how much the output changes. To enable measurements below the other interfering noise in the system, add a bandpass filter between the output and the meter input. This test is often used on CD players and appears in most home audio magazines' CD player test reviews.

A plot of the gain vs. output level on a graph can help you understand the data. (A picture is worth a thousand words.) The resulting plot is similar to the one in Figure 2. Ideally, this plot should be a straight line. The example in Figure 2 was measured on one channel of a popular professional multitrack digital audio recorder.

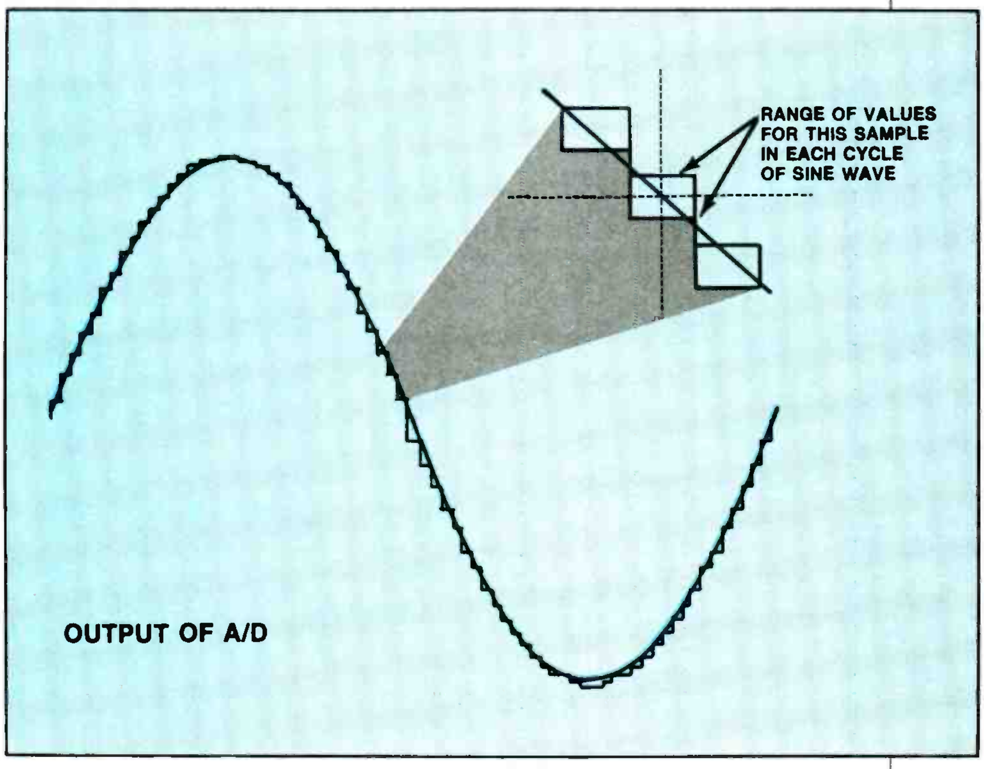

Figure 1B. Expanded view of A/D output.

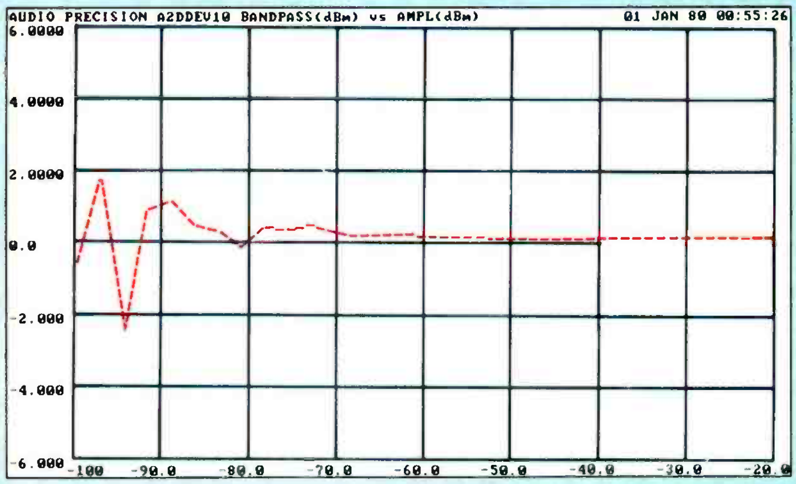

Figure 2. *Deviation from linearity, good channel (channel 10).*

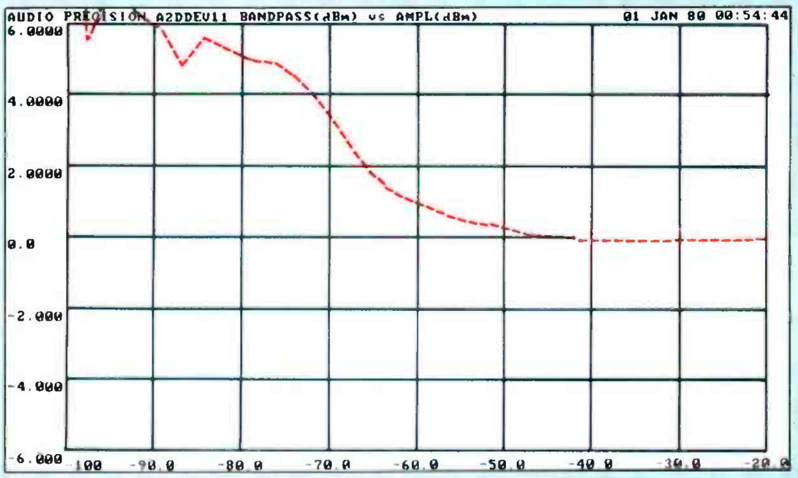

Figure 3. *Deviation from linearity, poor channel (channel 11).*

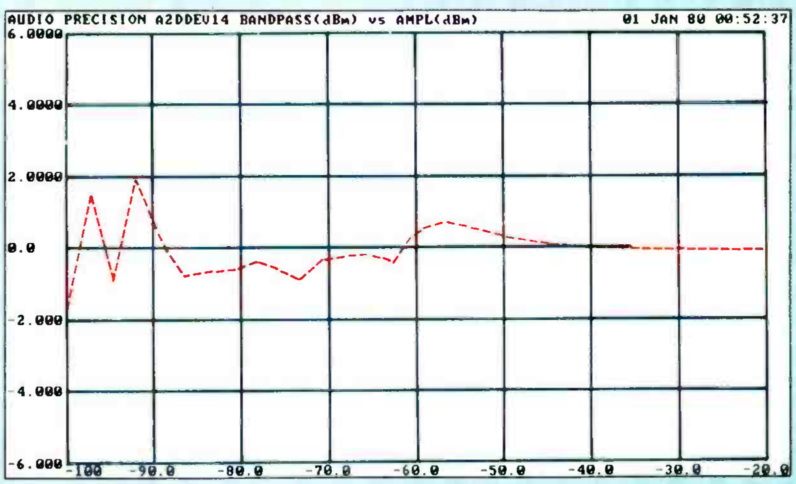

Figure 4. *Deviation from linearity, poor channel (channel 14).*

The measurement shows it to be fairly good, deviating only about 1dB down to the −90 level. However, another channel of the same machine resulted in the graph in Figure 3. This channel looks pretty bad, and it certainly would sound that way, too.

This test usually works well because most converters change all of their bits at 0V, causing the level of small signals to be highly affected by bit errors. However, a poor A/D converter can be improved immensely on this test simply by adding a slight dc offset to the signal. As the signal level is reduced, the 0 point of the converter will be crossed at a higher signal level. At the higher level, the errors are a smaller part of the signal amplitude. When the test sine wave is at very low levels, the offset keeps it from crossing the more significant bit transitions. Although some deviation from linearity exists at a higher level, it is small because it is a lesser percentage of the signal. An example of this relationship is shown in Figure 4. The linearity shows a slight deviation as high as −45dBm input, although the worst case deviation over the range is no more than 1dB, down to −92dBm.

At this point you may be thinking, "I can believe that the 6dB change in the earlier graph is audible, but I can't hear a 1dB change in signal level in music." If it were as simple as that, you might be right. Remember, this article started with a description of how quantization creates noise energy in the signal. What if we instead measure this quantization noise directly?

To measure quantization noise, consider the noise floor as a function of signal level. First, drive the device with a low-frequency sine wave, then remove the sine wave at the output with a distortion analyzer notch filter or a sharp high-pass filter. Next, measure the noise floor of the output with a ⅓-octave analyzer. Change the signal level and measure the noise floor at each level. The largest deviation in any of the ⅓-octave bands is the noise floor modulation.

The same tape machine channels (channels 10, 11 and 14) measured in Figures 1, 2 and 3 were measured with the noise floor modulation technique using an Audio Precision System One audio test system. The channels were driven with a 200Hz tone and the spectrum from 500Hz to 20kHz was measured. The level was changed in 10dB steps from −40dB to −100dB, and the spectrum was measured at each point. The resulting graph for the good channel is shown in Figure 5. The noise shifts less than 2dB over most of the frequency range, reaching a worst-case modulation of approximately 3dB. The channel that measured so poorly on the

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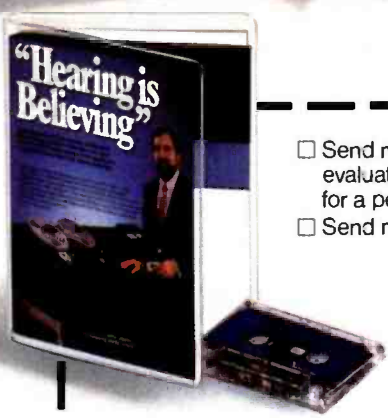

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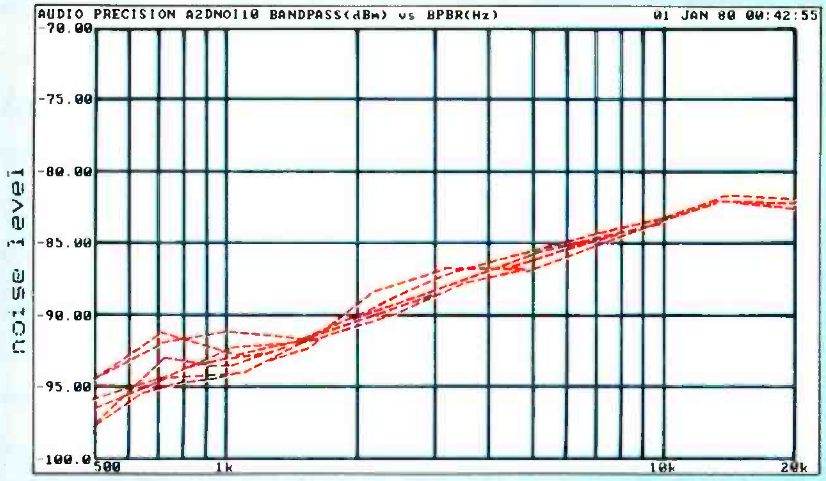

Figure 5. *Noise modulation, good channel (same as Figure 2, channel 10).*

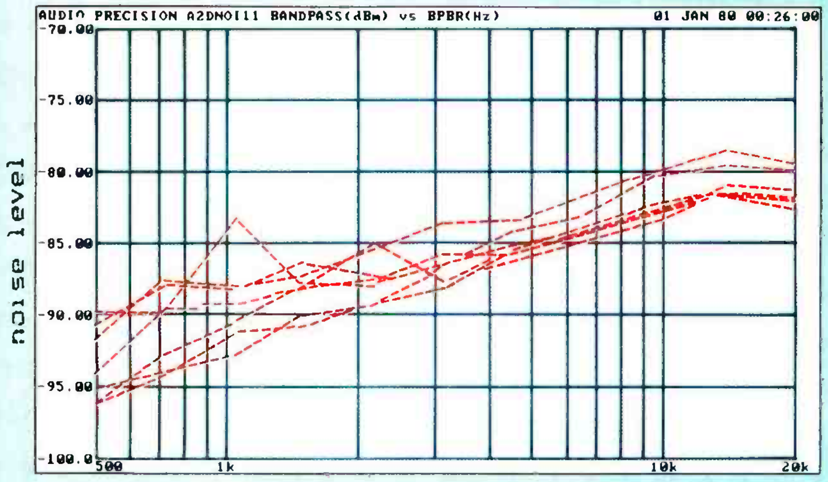

Figure 6. *Noise modulation, poor channel (same as Figure 3 channel 11).*

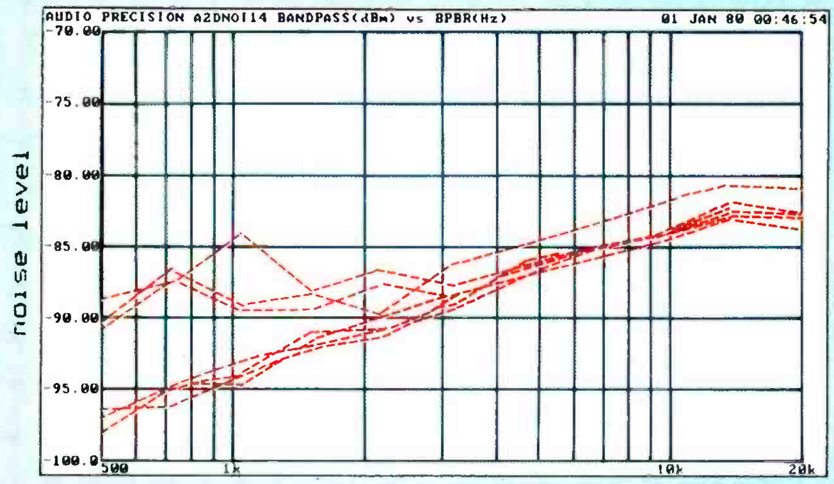

Figure 7. *Noise modulation, poor channel (same as Figure 4 channel 14).*

linearity test in Figure 3 shows an equally poor performance on this test in Figure 6. The noise modulation over most of the range is 5dB with a worst-case deviation of 10dB. More importantly, the channel that showed only small deviation from linearity shows a large noise modulation in Figure 7. The "inaudible" 1dB deviation from linearity now shows a fairly hideous 12dB noise floor modulation. (Time to reach for the tweak tool.)

Addition of an offset voltage to the converter input will not change the results significantly, as it did with the level linearity test. Recall that an offset voltage could cause the converter to shift the most significant byte (MSB) glitch when the signal level was large, making any errors a small percentage of the signal amplitude. (See April's "Understanding Computers" for more information on MSB.) Because the noise modulation test measures the background noise, the signal amplitude does not enter into the measurement and cannot mask poor performance.

How does it sound?

This effect is an audible pumping of the noise with changing signal level, similar to the noise a bad compressor or expander causes. The audibility depends on how far away from the signal frequency the noise modulation occurs and the resulting masking effects in the ear. Pioneering work by Louis Fielder at Dolby Laboratories indicates that a noise floor modulation of less than 2dB is desired for inaudibility of this effect.

If you want to experiment with listening to this effect, you can pull out your digital recorder and apply a low-frequency tone to the input. Run the output through an equalizer with a sharp, low cut and listen to the output. Crank up the gain and listen to the background noise while you change the input level. To make it more musical, you could try using a low piano note or a string bass note and listen to the noise floor as the note decays. Try to get a good S/N ratio on the signal going into the recorder so you don't add dither to the signal. Don't be surprised if some channels of the machine sound fine and some sound poor. RE/P

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Shopping for a Used 2-Inch Analog Tape Machine

By Douglas Beard

A multitrack recorder is a major expense in equipping a studio. A quick check of the new equipment available and current prices might lead you to shop in the used equipment market.

Buying a used multitrack is a little like buying a used car. Both are complex pieces of equipment that can cost a lot. And both may look good on the outside while hiding expensive problems "under the hood." But, if you exercise caution, carefully check the condition of the equipment and do some research before laying down your hard-earned cash, a used multitrack, like a good used car, can offer good value.

Know your needs

When looking for a used recorder, you need to consider your requirements first. How will you use the machine: as a primary multitrack, as a second machine to synchronize for large projects, for an additional small studio for "overflow" projects, or for a MIDI/synth room?

Once you clearly understand how you will be using the machine, you can list essential features that your multitrack must have. Do you need an autolocator and audio channel remote control? Can the machine be synchronized with SMPTE time code? What types of machines do your clients demand?

Where to shop

Possible sources for used machines are classified ads in trade magazines, equipment brokers who specialize in used equipment and equipment manufacturers or dealers who may take "trade-in" machines and offer them for resale. Make your requirements known to as many of these sources as possible. We work in a relatively small industry, and information travels through the grapevine quickly. The fact that you are looking for used equipment will get around soon enough.

Douglas Beard is director of technical and marketing services for Studer Revox America, Nashville.

When you hear about a machine that may fit your needs, it's time to take your shopping seriously by checking out that particular piece of hardware and its condition. If you're lucky, you will find what you're looking for in your city, but chances are your search won't end nearby. In fact, your machine may be thousands of miles away. So, be prepared to put yourself on a plane, or to make arrangements so that you can evaluate the machine by "remote" through someone you know.

What to look for

If the machine still is in use in a studio, and if you can actually get to it and check it out, your task is simplified. In that case, you'll be able to test it in an operating environment. Look for:

1. Capstan motor. Check the shaft for signs of wear. Listen to it in operation. Does it make any "unnatural" noises? Run it through the full range of the vari-speed to be sure it maintains lock at the speed extremes. Is it abnormally slow to lock into speed?

2. Spooling motors. Listen for bearing noise in fast-wind modes. If the bearings are noisy, find out whether they are user-replaceable or have to be rebuilt in a "motor shop."

3. Guides and idlers. Are grooves worn in the fixed guides? Are the rollers worn? Do the bearings sound as if they need replacement?

4. Tape tracking. While the machine is in the play mode, watch the tape as it moves across the heads. Does the tape track straight across them? If not, some components in the tape path (heads, guides, motors) may not be exactly parallel with one another. In most cases, this problem is not major and an experienced technician can correct it. Usually, the technician will adjust the height or angle of the deck components by carefully shimming the problem component.

5. Audio performance. Check frequency response on all channels. If any channels show a drastic roll-off on the high frequency, investigate the heads more closely. Call on a specialist to evaluate the heads and estimate head life. [See "Testing the Heads: A Job for the Specialist" on page 28.] If the heads pass the test, everything else in the audio path is relatively minor. Listen to each channel while recording with no signal input to unveil any noisy channels. (Verify that all channels erase and record.)

Record and play back 1kHz and 10kHz tones while watching the stability of the VU meters. If the meters fluctuate more than ±1dB, look for possible causes such as worn heads, guide rollers damaged or at incorrect height, or a worn pinch roller. Many of these problems are fixed easily. But knowing about them before you buy the machine will help you maximize your performance/dollar ratio.

Check audio switching input/reproduce/sync modes. If the machine has solid-state switching, chances are that it operates well. If relays are used, check carefully and look for intermittent problems.

All of these points to check apply if you can get to the machine and operate it. If the machine you're considering is not "in service," then you'll have to take some test equipment with you to examine it properly.

If you can't get to the machine yourself, call on someone you know to evaluate it in your behalf.

Call the manufacturer

If the machine you're looking at has an elapsed-time counter, it will give you an idea of how much the machine has been

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used. If you know people using a similar machine, ask them about their experiences with it and about the machine's reliability in general.

If you can get the serial number of the machine, you might be able to call the manufacturer or dealer to confirm its age and service history.

Frequently, we get phone calls from people investigating the purchase of a used multitrack machine. Typically, a caller asks about the availability of spare parts (a smart idea) or clarification on model nomenclature (for example, "Is the MK 3 version of the A80 transformerless?"). Callers sometimes request the exact age of a particular machine, which we can determine from the serial number. Generally, we tell callers inquiring about a used machine what updates are possible, desirable or necessary for its optimal performance—and the approximate cost. Most manufacturers will provide this type of information if you ask.

Finally, if callers are contemplating the purchase of used multitracks and haven't had the heads tested, we recommend they do.

Support: is it there?

Machine support varies greatly from brand to brand. Some of the used machines on the market were made by companies that are no longer in business. Finding spare parts for such a machine may be difficult, and sometimes impossible.

Testing the Heads: A Job for the Specialist

Hidden problems in either the heads or capstan motors can require some pretty expensive repairs. Head replacements can run from $4,000 to $8,000 for a 24-track machine. So it is advisable to have the heads tested by a specialist to determine approximately how much life is left in them.

Head life is difficult to determine precisely, but a specialist can give you a good indication if the heads on your machine will be "opening up" within a few months, or if they will serve you for several years.

Several services around the country specialize in testing, repairing and refurbishing magnetic heads for audio equipment. A few of the more widely known services are: JRF Magnetic Sciences, Greendell, NJ; Sprague Magnetics, Van Nuys, CA; AMP Professional Services, Ft. Lauderdale, FL; and Saki Magnetics, Calabasas, CA.

The "head specialist" can determine most accurately if the heads on your machine are OK. Visual inspection of a magnetic head will tell you little or nothing about its condition. To help you accurately estimate the useful life of your machine's heads, a specialist can perform a couple of tests: (See Figure 1.)

1. Inductance. *The inductance of a head decreases as the metal wears away. By measuring the inductance of a worn-out head and a new head from the same manufacturer, and by comparing the results of this test, the specialist can approximate the amount of head life left on the used head.*

2. Tip depth. *Tip depth refers to the amount of core material remaining at the "tip" of the pole pieces (gap area) on the head. The thickness of these pole pieces (tip) at the head gap determines the mechanical life of the magnetic head. The precise spacing between the left and right pole pieces creates a uniform frequency response for each channel. If the tip is worn beyond the depth of the gap, the frequency response of the head deteriorates drastically. At this point, the only thing to do is replace the heads.*

As you can see from the sample test report, the specialist's services are not an absolute measure of head life. But investing a little cash for this type of evaluation could save you a lot of headaches—and a lot of cash—down the road.

Customer ______________________ Job Number __________ Date ______

Equipment ______________________ Assembly No. __________

The following is a summary covering the condition of your magnetic heads:

STACK	S/N	EXPECTED % LIFE					
		New	75%	50%	25%	Marginal	Reject
Erase							
Record							
Repro							

THE EXPECTED LIFE IS BASED ON...
- ☐ Measured tip depth after reconditioning
- ☐ Estimated tip depth after reconditioning as compared to the new depth specification

ADDITIONAL SERVICES PERFORMED
- ☐ Optical Alignment
- ☐ Other: ______

REINSTALLATION: Your head assembly has been set up and optically aligned to exact industry and specific equipment specifications. Variations do exist in tape path, tape handling characteristics and deck plate mounting tolerances in all equipment. As a result, minor azimuth adjustments may be required to achieve phase and optimum performance.

NOTE: Some tape machines are designed without incoming and/or exit tape wrap reference points located on the head assembly. While every effort is made to calculate the exact wrap setting, minor wrap adjustments may be required during initial installation and setup to optimize performance. Symptoms of this condition will show up as insufficient erasure or slight instability on the *Repro* head stack.

To insure maximum head life, be certain to check tape path tensions and adjust (reduce) to the minimum acceptable performance setting. Consult your equipment manual.

***Figure 1**—Audio magnetic head test report.*

Test report courtesy John French of JRF Magnetic Sciences, Greendell, NJ. Copyright 1986 by JRF Magnetic Science. Reprinted with permission.

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Other machines may still be in production, or at least the manufacturer may still support them, so you don't have to worry about the availability of spare parts.

Machines with modular construction throughout might offer an advantage. Modular construction means that a machine will be easy to troubleshoot and repair, if necessary.

Buying a machine from a foreign manufacturer that no longer exports to the United States, has left the audio industry, or is no longer in business may present other problems. If spare parts are available at all, they are usually limited to an independent organization that has taken over the spare parts catalogue from the original manufacturer. They may be in limited supply and, therefore, costly.

Examining all of these "spares" and support considerations will help you determine the value of a particular machine.

Cash and carry

Once you've decided to buy, you still have a couple of hurdles to jump before that bargain multitrack is sitting in your studio.

The first is financial, and in the used equipment business, the name of the game is *cash*—wire, transfer, cashier's check. Once the seller has your cash, the machine is all yours.

Your next concern is transporting the machine to your studio. Because the chances are great (virtually certain, in fact) that the owner didn't keep the original packing for your machine, the most practical shipping option is via "electronics" moving van.

Shipping by an electronics van company is the means of choice because your machine can be shipped intact without fear of damage. All of these companies use "air-ride" vans. They wrap the equipment in blankets and strap it securely in the van. Requesting that the machine be wrapped in plastic (for protection against dust) is worthwhile, although the moving van is probably as clean as most control rooms.

An additional benefit of using an electronics van company is that they have the equipment and personnel to load and unload heavy equipment—something to consider rather than lifting a 300- to 600-pound machine into the back of a truck. Letting the van company worry about this problem eliminates your trying to round up enough muscle from the local gym to get your machine off the truck and up or down the stairs to your studio.

Scan the recording studio directories. You'll notice that there are a lot of 10- to 15-year-old recorders still in use today. This longevity proves to the shopper that the release of a new model does not spell instant death for existing machines.

There are plenty of machines from which to choose. The appropriateness of any one of them depends on your requirements. Beyond that, your preferences take over. The characteristic sound of an older machine, or some quirk of operation particular to a specific make and model, may play a significant part in your decision.

In this age of microprocessor-controlled coffee pots and cars that talk, you might be the type who prefers to get up five minutes earlier to put on the coffee yourself, or take responsibility for shutting off your headlights without being told to do so by your car. For you, the appeal of a particular "previously owned" multitrack may lie in the fact it has nothing more exotic than TTL logic to control the transport, a simple transistor and IC audio amplifier, and trim pots that you adjust with one of those little green screwdrivers. RE/P

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Digital Control of Analog Tape Machines

By Mark Calice

Design engineers have taken an established product and applied a new twist that makes the operator/engineer's job a little easier.

In the garden of digital delights, digitally controlled analog tape machines would seem pretty far down on any list of priorities. After all, this technology does not eliminate tape saturation or tape hiss. It also does nothing to improve generation losses or dynamic range. In fact, it does nothing to improve the analog format itself. So why bother? I'm glad you asked.

Here we address something that all innovative technology in the 20th century has done for us so many times in the past. The design engineers have taken an established product and applied a new twist that makes the operator/engineer's job a little easier.

Analog tape machines still comprise the vast majority of audio storage media in use around the world today and will continue to do so into the foreseeable future, as evidenced by the continuing brisk sales of such devices. In the future, they will undoubtedly share a portion of the market, as well as occupy a fond place in our hearts.

Electronic alignment

Let's define the two basic architectures in use today: The first system introduced might be called, for definition's sake, the "electronic" audio alignment system. Machines in this category include the Scully LJ-12, Sony APR-5002, Soundcraft Saturn, and Studer A810, A812 and A807. Studer pioneered the use of this system when it introduced its A810, 2-track tape machine.

This electronic alignment does away with all the potentiometers for audio alignment parameters (such as equalization, record gain and bias) and substitutes shuttle wheels, or up and down keys. These configurations enable you to step the digitally controlled "trimmer" values up and down manually for the correct results as measured on the VU meters or with outboard test equipment.

Mark Calice is product specialist at Otari Corporation, Foster City, CA.

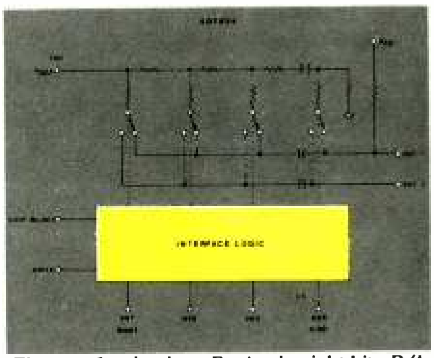

Figure 1. Analog Devices' eight-bit D/A converter.

By eliminating the potentiometers, any possibility of the trimmers going bad or becoming noisy under heavy use is eliminated. No more stripped screwdriver slots or bent trimmers, and you'll never turn the wrong pot or replace one again. It's a little surprising, the first time you encounter one of these machines, to open the panels for adjustment and find no trimmer holes on the front of the audio cards.

In addition, the system lets you store default or user-selected digital trimmer values in non-volatile memories, thus allowing entire setups to be recovered at the next session or to recall different tape formulation settings when switching between different brands.

This ability makes the system user-configurable and contributes to increased efficiency during machine setup. The Sony APR-5000 series took this concept one step further by allowing head stack swapping to retrieve the settings automatically for *that* tape/head stack configuration. This "intelligent" head stack is equipped with an eight-section "identity code" DIP switch, which allows you to program information concerning the number of tracks, speed range, tape width, tape type and three different alignments per speed. The Studer A820 series shares this feature.

This class of machines is a great step forward for the maintenance and session engineer, but, except for the memory recall functions, this system does little more than allow you to put away that tweak tool, which you can never find when you need it. You still must adjust every parameter on every channel separately, and each machine differs as to how many memory locations are available for each speed.

Automatic alignment

We'll call the second system the "automatic" audio alignment system. Machines in this category include the Otari MTR-20, MTR-100A, Studer A820-2 and the A820-24. This system includes all the advantages of the first system and adds the capability for the machine to automate all of the record and repro alignments with minimal human intervention. Of course, the automatic function can be bypassed for manual adjustment, if necessary.

Otari's MTR-20 was the first machine to have this feature. No provision is made to automate the reproduce section on this machine, so all playback alignment needs to be done manually. The MTR-100A and Studer's A820 series have added the capability to automate both the record and reproduce alignments.

These auto alignment tape machines have one or two additional microprocessors dedicated to the control of the audio electronics. Inexpensive eight-bit D/A converters (typically an Analog Devices

AD7524) are used in different circuit configurations to implement different functions. (See Figure 1.) DACs are used instead of voltage-controlled amplifiers (VCAs) to allow more precise adjustments with less noise and distortion.

Digital audio use dictates that the DACs' reference voltage input be tied to a fixed dc voltage. The output conversion will then consist of discrete voltage steps proportional to the digital data input. In some cases, the audio input voltage is applied directly to the chip's reference voltage input, and the ladder network is switched by the CPU to give different overall resistance to the circuit. This configuration changes the circuit's parameters, depending on where the chip resides in the circuit. Configured as digitally controlled amplifiers (DCAs), the DAC network is in the feedback loop of an op-amp, thus affecting the gain. (See Figure 2.) As digitally controlled filters (DCFs), the DAC network is in parallel with the series input resistor. (See Figure 3.)

Because these chips use resistive ladder networks for conversion by switching resistors in and out of circuit, an eight-bit DAC provides 256 possible data steps (0 to 255) programmable by the CPU. Error

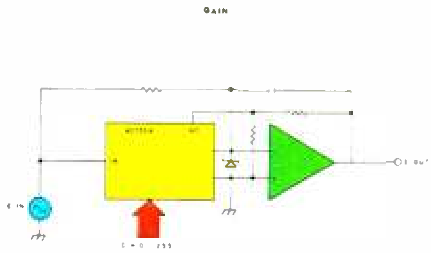

***Figure 2.** Configured as digitally controlled amplifiers (DCAs), the DAC network is in the feedback loop of an op-amp, thus affecting the gain.*

detection also is included in the software so that the machine can indicate incorrectly aligned parameters if a problem arises during auto alignment. This capability is essential because, ideally, the engineer leaves the machine to do other work while the auto alignment is in progress.

Each of these machines provides different features for displaying and retrieving this information. The MTR-100A has a set of "soft" keys located under a large backlit LCD. The keys change function as you step through the menu hierarchy. This topology allows the machine to have many functions controlled by only five buttons. The software also allows the machine to rewind to the beginning of the alignment section and lay down session tones according to a preprogrammed setup. The display can show each trimmer's decimal number currently in use or can show and recall comments the operator entered to identify setups. This capability is helpful in comparing before and after adjustment values. Assuming there are no other mechanical or electrical problems, these trimmer values could be used as a gauge for head wear.

The Studer A820-24 displays its information on the overhead meter panel. The company has elected to display trimmer information in the more compact hex format (0-9, A-F) rather than as decimal numbers. [For more information on hexadecimal notation, see April's "Understanding Computers"—Ed.] So, only two numerics are needed to represent numbers larger than 99. The A820 series also allows you to download the trimmer values through their RS-232C port to a computer or as an FSK audio signal for storage on an open track of the session tape itself.

Alignment sequence

Because adjusting high-frequency equalization affects the overall gain, engineers always follow a certain alignment order

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to minimize the amount of repeat adjustment needed. This order usually consists of first performing the overbias calibration followed by a personally preferred sequence of remaining adjustments. Because the engineer chooses to relinquish the alignment chore to the machine, it would be nice to know that the recorder is attempting to do as good a job as the engineer would be able to do himself. To that end, these systems also follow certain sequences to ensure the most accurate results. As an example, the Otari MTR-100A adjustment sequence for full auto record is as follows:

1. bias
2. gain
3. EQ mid-high
4. EQ high
5. gain
6. EQ mid-high
7. EQ high
8. gain
9. EQ mid-high
10. EQ high
11. phase compensation
12. EQ low
13. gain

These systems require that all mechanical alignments, such as head and tape path alignment, have been done properly. You must also manually set the input level audio alignment and calibrate the internal oscillator and VU meters correctly so that the machine has the correct reference levels. (Yes, you still need to bring your skills with you to work.)

Bias is the first adjustment done to provide the basis for all subsequent adjustments. The MTR-100A does three sweeps for each parameter. The first sweep proceeds full range from 0 through 255 but only outputs every fourth DAC data value so that the CPU can quickly find the peaks and determine a restricted range for the last two sweeps. This peak reading corresponds to a decimal number, and that value is temporarily stored in RAM for comparison. (The value is determined by a level comparator for gain, equalization and bias and by a zero-crossing detector for phase compensation.) At the beginning of each ramp-up, the oscillator outputs a 250Hz marker cue tone for about 50ms. This tells the CPU that valid data follows. The last two sweeps step through every DAC data value, but only in a restricted range determined by the previous sweep. These last two sweeps "fine-tune" the adjustment. This scheme ensures that the final alignment is as accurate as possible.

There are other distinct advantages to automating alignments on an analog tape machine. To achieve the highest-possible recording quality, it is necessary to adjust

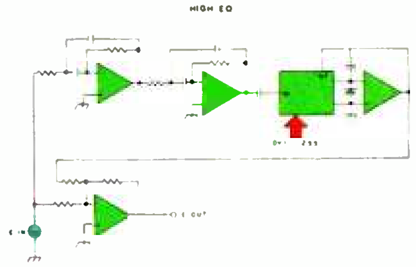

Figure 3. As digitally controlled filters (DCFs), the DAC network is in parallel with the series input resistor.

the record gain, equalization, bias and record phase compensation for each tape formulation. If an automatic alignment is carried out at the beginning of every reel of tape, each reel will be optimized. If all tapes of a particular brand or type were uniform from one batch to the next, such an alignment scheme would not be a benefit and the "electronic" system would suffice. Alas, the real world. This benefit alone will be worth the price of admission to some.

The different requirements for each of the auto adjustments normally would necessitate using various external test equipment. To assist in automating the entire record alignment, a microprocessor-controlled sine/square wave oscillator is provided on-board.

The speed with which these systems operate is awesome. A total 24-track record alignment takes less than five minutes! You don't have to run back and forth to the console to change frequencies and check levels. Even in manual mode, the alignment takes much less time.

If you will be using the optional Dolby or Telcom noise reduction cards in either the MTR-100A or the A820-24 machines, the automatic alignment systems will also calibrate the noise reduction input/output levels automatically, if you wish.

All record adjustments are referenced to the playback head reproduce signal, just as they are in manual record alignment on a conventional machine. The final results of the auto record alignment can only be as accurate as the repro adjustments made during auto reproduce alignment. Completely automated reproduce alignment would require some type of standardized reproduce alignment tape available in all formats, flux levels and equalizations with standardized frequencies, durations and order. Most engineers worldwide would have to agree on all these variables and the chances of this happening are, well, let's say not likely. So, the auto reproduce systems are not totally foolproof. You must tell the machine what parameter you want it to align, (such as repro gain, repro high frequency or repro low frequency) and then play the correct frequency from your test or session tape. You then are free to select the frequencies you would like the machine to align with—a small price to pay for the speed and repeatability gained.

The resolution of the digital gain and equalization trimmers seems to be, on the average, ±0.1dB, depending on where it lies in the range of adjustment. It approximates a bell curve with the extremes at about ±0.15dB. Most engineers seem to agree that this resolution is adequate, considering that a given reel of tape can change much more than that amount from one end to the other. However, the accuracy with which the machine aligns itself is a function of the DAC and associated circuitry. This seems to have settled at an average around ±0.25dB for the moment. Of the engineers polled, 50% say this figure is adequate, and the other 50% say they go back and manually tweak individual parameters to their liking. They also state they still let the machine do most of its auto alignment for very repetitive functions, such as bias, because the machine can do them much more quickly.

All these features—full record and reproduce auto alignment, the ability to compensate for every individual reel of tape, greater speed in setup, consistent setup accuracy, memory backup of audio settings, elimination of mechanical potentiometers, use of DACs instead of VCAs, built-in error detection, automated setup oscillator and the ability to align input/output noise reduction levels automatically—combine to provide the engineer with an impressive arsenal of helpful functions.

The benefits and the consistency with which these machines can align accurately far outpaces any manual method of alignment and allows the technique of digitally controlled audio alignment to really shine. In the future, perhaps any number of cutting-edge computer technologies will be incorporated into our machines. Artificial intelligence and voice recognition may revolutionize our industry. Once you add your variables to the program, you could prepare your session by talking to your machine:

Engineer: "Computer?"

Computer: "Working."

Engineer: "Please align for my tracking session."

Computer: "Align for Argon Advertising, or The Rad Brothers?"

(Most of us already talk to our machines.) RE/P

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Maruyama, K. and Ross, B. "An Implementation of Computerized Record Adjustments for an Analog Tape Recorder."

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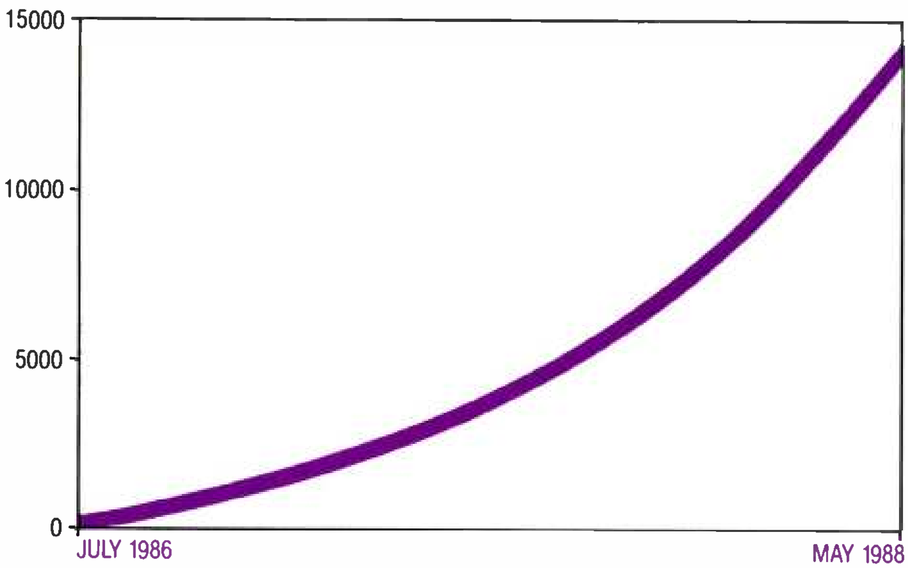

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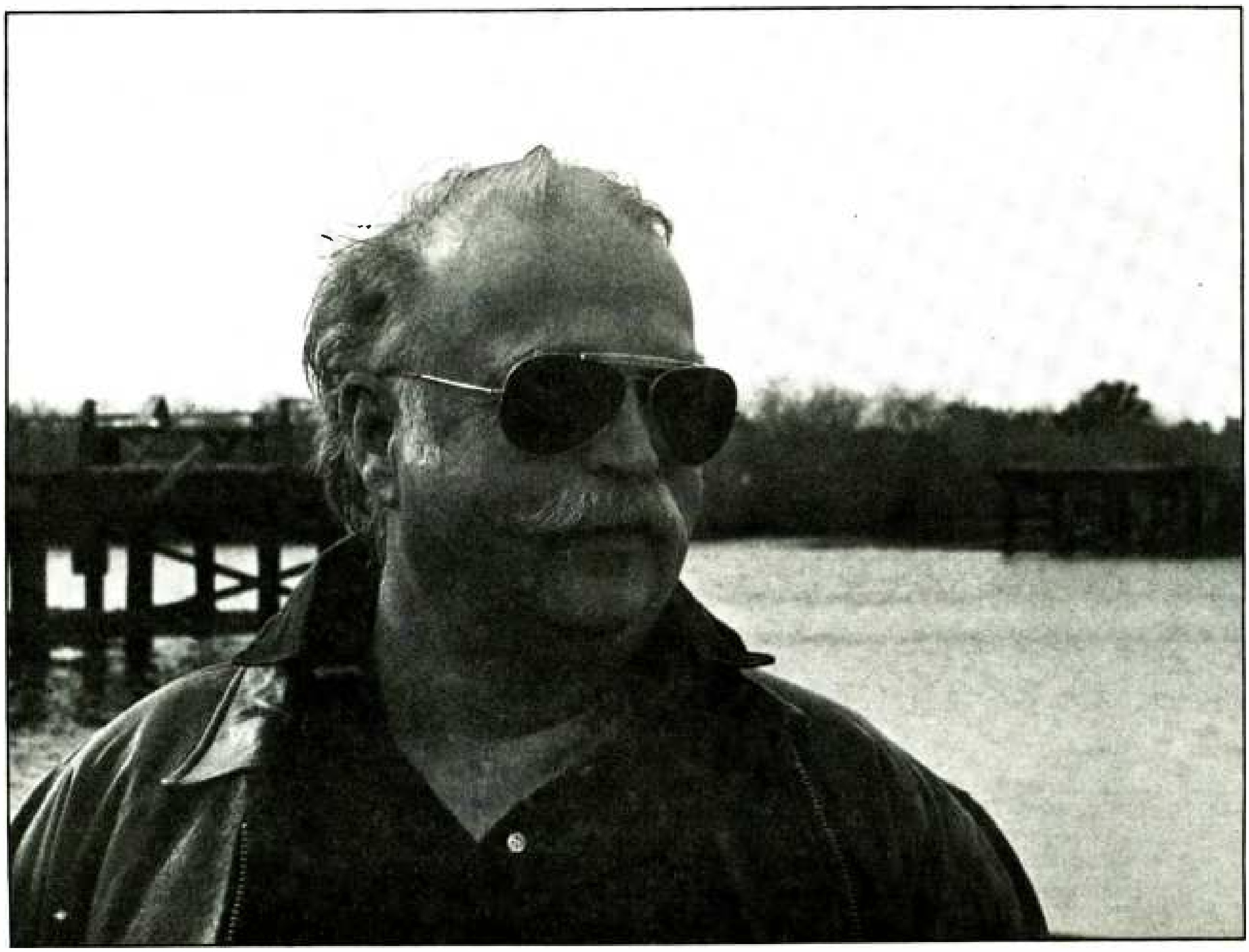
RE/P EXCLUSIVE

Engineer Interview: Bruce Swedien

By Bridget Balthrop Morton

"... The only reason we're in the studio is for the music. Nothing else. Nobody ever left the recording studio humming the speaker or the control console or the tape machine—only the music."



Bruce Swedien's recording career began when he was 10 years old when his father gave him a disc recording machine. By age 14, he was working part-time in a small Minneapolis studio, and he's been in the business ever since. An acknowledged master, he has crossed all barriers and recorded everything from classical music to commercial jingles and film scores. In the process, he's had seven Grammy nominations for his pop music recording. He's won two, for Michael Jackson's "Thriller" and "Bad." At the time *RE/P* talked with Bruce, he was preparing another cut from "Bad" for single release later this year.

***RE/P:** If you were going into the studio tomorrow to record Michael Jackson, how would you approach it?*

BS: If we were starting with the rhythm track, that concept would already be pretty well in hand, but it kind of grows. Every piece of music is different. You want an example?

***RE/P:** Sure.*

BS: The first song that we released as a single from the "Bad" album was Michael's duet with Siedah Garrett, "I Can't Stop Loving You." Michael wrote the song. We decided, probably Quincy more than anybody, that this would be a perfect song to record with the whole rhythm section live. So we had microphones all over the place. There were two keyboards, two guitars, a bass and the drums. (See Figure 1.) Michael and Siedah were on the scratch track together. I think we overdubbed some percussion and strings. That's quite different from another song that might be recorded all with synthesizers.

***RE/P:** You have very particular opinions about the mics you use. Can you tell us about that?*

BS: Microphones to me are like musical instruments. Each microphone, and, in fact, individual microphones within a model or a manufactured type, will have its own color or its own characteristics. I have 14 Anvil cases full of mics that I've been collecting since the beginning of my career. I have two microphones that I bought in 1951. They're Neumann 247 tube types. As a matter of fact, I used one of them on Michael for the "Bad" album. I've been engineering so long that I can listen to people speak and know what mics would work best on their voices. That's a big help. I don't waste a lot of time in the studio trying different microphones.

***RE/P:** Can you tell us why you still frequently use a 16-track analog tape machine?*

BS: The common format for multitrack analog recording is 24-track, but 16-track, 2-inch analog recording, scientifically, is vastly superior. There's a slight gain in track space for each track, so sonically, there's an improvement in the sound. I use my 16-track at the studio almost every day, or whenever we're recording drums and percussion, and I've done that for years as a personal choice. And, I think it affects the sound of my work.

***RE/P:** What precisely is the effect?*

BS: The signal-to-noise ratio is vastly superior on 16-track, so you have the advantage of not having noise reduction between the tape machine and the music. That equipment will color the sound, but that does not happen with 16-track. And because of the greater track size, the transient response is vastly improved. So,

Bridget Balthrop Morton is a free-lance writer and teaches at Florida Institute of Technology in Melbourne, FL.

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A Career Built on Collaboration

The list of artists Bruce Swedien has worked with reads like a "Who's Who" of music in the 20th century: Duke Ellington, Tommy Dorsey, Lena Horne, Sergio Mendes, George Benson—the list goes on and on. Bruce admires artistry and professionalism in the musicians he records. Nowhere is this tact more evident than in his relationship with Quincy Jones.

The two met in 1958 while Bruce was working in Chicago with Universal Studios. Both had classical training and were committed to pop music. According to Bruce, their collaboration has always been founded on tremendous mutual respect. "We have an affinity for each other and the space we occupy together, "he says, adding that Quincy's approach to music is "kaleidoscopic. He's one of the few people I know in the business who always has the big picture in focus."

When Quincy produced "The Wiz" in New York, Michael Jackson played the scarecrow and Bruce Swedien recorded the soundtrack. Another long-term collaboration was born. Bruce says Michael is a "consummate artist. We've been working together more than 10 years now, and Michael is constantly growing. He'll sacrifice anything for the quality of what he does."

Bruce calls himself "a born detail man." He frequently focuses on one kind of detail: He gives credit freely and generously when he believes it is due. During this interview, he praised his wife, Bea, who is his business manager, and his parents for their influence on his career. He volunteered that Bill Putnam, with whom he worked 30 years ago in Chicago and whom he still sees occasionally in Los Angeles, taught him the most as an engineer.

"Bad" received five Grammy nominations, but the only award it won went to Bruce Swedien. Somehow it's no surprise that he says, "I wish I could give mine to Michael and Quincy."

you record the steep wave front in as close to its natural condition as possible. Lately, a lot of people have picked up on the fact that I use a 16-track in all my work. Maybe I'm about the only guy that bothers with it.

I use it in a unique way, though, in that once I have the sound of the percussion recorded on the 16-track, I then transfer it immediately to digital. So what happens is similar to taking a high-quality motion picture camera and right next to it putting a video camera, equally high-quality, and with both these cameras you photograph the same object. You then play back the 35mm of this thing, and it's beautiful; it's soft and it looks wonderful. Then you play back the video, and it's not beautiful; it looks hard. The clarity is there, but it's not the same.

To me, this example is similar to the difference between analog and digital. With analog you get sort of a beautifying effect, a quality to it that is very appealing. One other thing happens that I think is just incredible, that also happens with motion picture film. You can take that 35mm film, and once you've photographed that object and transferred it to videotape, it still looks fantastic. The same thing happens with analog and digital.

So, once I've recorded the drums and percussion, or whatever I use analog recording for, I then transfer it to digital, but it retains that analog quality. Once it's in the digital domain, it's miraculous what you can do with it.

***RE/P:** Can you describe a particular cut where you used that technique?*

BS: "Man in the Mirror" would be a great example, where I recorded all the drums and percussion on the 16-track. There's a snare drum sound that I made up of a real snare drum, and to that we added a handclap sound. That has a tremendous transient noise, and had I recorded that directly to digital, it wouldn't have sounded like it does. It would have been gritty or harsh. But with the 16-track analog, it has the strength and the energy, and it also has a sonic quality that makes it very appealing.

***RE/P:** Can you be more specific about the value and performance of digital as opposed to analog equipment?*

BS: I use them both and have favorite applications of each. If you want to address one specific phase of my work directly, generally speaking, I always mix to 2-track stereo digital. My favorite machine system is the Mitsubishi. When I mix, I mix to the Mitsubishi X-86 digital, 2-track master recorder. My multitrack digital machine is a Mitsubishi X-850 32-track recorder. With my 16-track analog, the outputs of that are connected directly to the inputs of the digital. So, the first 15 tracks go to independent channels. There's no mixing involved.

With synthesizers, I record direct to digital. If the sound is originally produced digitally, it seems to record well digitally.

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

BS: I don't like what happens to the sound of vocals digitally recorded. So I record all my vocals on 24-track analog. When I do backgrounds, I will premix in stereo pairs on two or four tracks of the digital tape. I call it premixing because, in pop music, mixing is a lot more that just balancing. And, really, all you do at that early stage is balance harmonies. There are no hard and fast rules.

***RE/P:** What's you opinion on R-DAT copy protection?*

BS: Boy, that's a hard one. Part of me understands the problem of copyright protection, but then there's the technical part that says I don't want anything in the way of the music. I want the music to be reproduced with as much clarity as possible. And all the copy-protection devices offered so far will affect the quality of the music.

***RE/P:** So if you were making the decisions, what would you do?*

BS: I wouldn't have it. But believe me, I do understand the other side of the story.

***RE/P:** Can you tell us about your "acusonic" recording process?*

BS: "Acusonic" recording, or my way of multiplexing multitrack tapes, allows me to record a lot more stereo pairs of tracks, and to keep those tracks as discrete sonic images in the final mix. To give you an example, when 24-track tape came out, I thought, "My God, a 12-track stereo!" But "acusonic" recording is just a term that Quincy and I came up with to describe that organization process. It's not a little black box. It just describes my technique.

***RE/P:** When did you start using this technique?*

BS: About 10 years ago. I'd been dabbling with it long before, but it never really started happening until Quincy and I started doing movies and records where I wanted to create more real stereo in the pop music we were doing.

***RE/P:** What were you aiming for?*

BS: Purely emotional value. It's to present the piano in stereo, and the vocal background in stereo—I'm talking about *true* stereophonic, not somebody's idea of what stereo is. True stereophonic record-

Table 1. *"I Just Can't Stop Loving You" duet, Michael Jackson and Siedah Garrett.*

Microphone used and input designation (live tracking session)

1. lead vocal (Michael)		Neumann U-47 tube	
2. lead vocal (Siedah)		Telefunken 251 tube	
3. acoustic piano (John Barnes)		AKG 414 EB—left	X/Y stereo
4. acoustic piano (John Barnes)		AKG 414 EB—right	X/Y stereo
5. DX-7 (Greg Philenganes)		direct AXE	
6. DX-7 (Greg Philenganes)		direct AXE	
7. synthesizer ("Hawk" Wolinski)		direct AXE	
8. synthesizer ("Hawk" Wolinski)		direct AXE	
9. bass (Nathan East) bass transformer		UTC-LS-10X	
10. guitar (Danny Huff)		direct AXE	
11. guitar (Danny Huff)		direct AXE	
12. guitar (Danny Huff)		AKG C451	
13. guitar (Danny Huff)		AKG C451	
14. drums (overhead L)	N'Dugu (Leon Chancellor)	Neumann U-87	
15. drums (overhead R)	N'Dugu (Leon Chancellor)	Neumann U-87	
16. drums (kick)	N'Dugu (Leon Chancellor)	Sennheiser 42	
17. drums (snare)	N'Dugu (Leon Chancellor)	AKG C451	
18. drums (hat)	N'Dugu (Leon Chancellor)	AKG C451	
19. drums (tom)	N'Dugu (Leon Chancellor)	Neumann U-87	
20. drums (tom)	N'Dugu (Leon Chancellor)	Neumann U-87	
21. drums (tom)	N'Dugu (Leon Chancellor)	Neumann U-87	

ing is usually done with a stereo mic or two single mics in close proximity to each other where the early reflected sounds are heard in their natural order. That's a component that is virtually always overlooked in recording, the early reflections of the sound. There are a lot of them, and they're very, very short in time, but they are a big part of the naturalness of the sound. True stereo micing retains the early reflections in their natural perspective.

RE/P: *Do you believe in enhancing the sound, or do you go after the recording process from a strictly clinical approach?*
BS: I hate the clinical approach. My early years in the business as a recording engineer, and indeed my musical training, were in classical music. To be honest with you—let me say this as nicely as I can—it bored me. I love classical music, and I have my favorites (Vivaldi, and Satie, and Respighi), but recording classical music always seems to me like...taking dictation. The most that we can do is recreate the original sound field. There's no room for the engineer to exercise imagination or creativity in the clinical approach.

RE/P: *So, rather than capturing only the*

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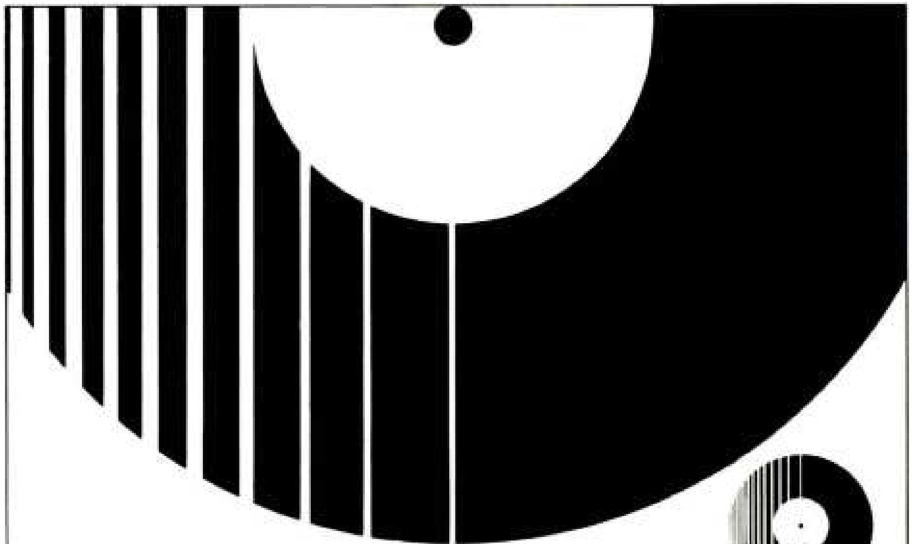

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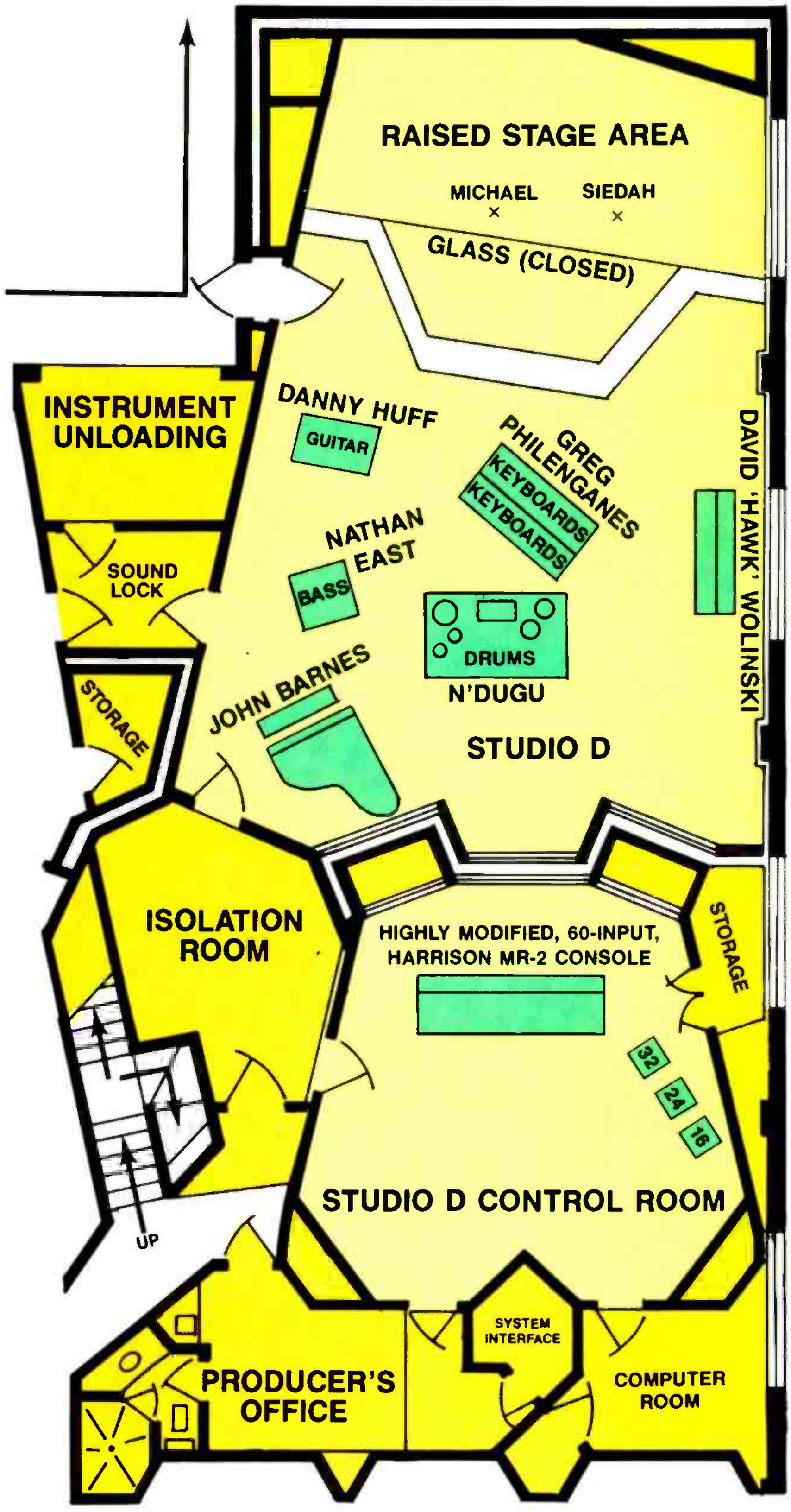

Figure 1. Westlake studio D, "I Just Can't Stop Loving You" setup.

sound the artists are putting out and nothing more, you try to satisfy your own ear?

BS: Absolutely. One of the things I like about popular music is that I can create sound fields that exist originally only in my imagination. They do not have to exist in reality. To me, that is exciting.

I was really fortunate in those years of recording classical music and choirs in the early days of my career, in that, frequently, a very clinical approach, when balanced against the creative approach, makes a wonderful contrast. It gives the music a ground, so to speak.

A perfect example is the choir recording in "Man in the Mirror." That is the most traditional clinical example of recording a choir. I reached back to my old days of recording the St. Olaf's Choir in Chicago and Minneapolis for that one. I used my favorite pair of AKG 414 EBs. Two microphones. Nothing fancy. But it worked so beautifully in that setting. Twenty people in that choir, and the microphones did all the mixing. (See Figure 2.) I just opened up the pots and sat back and listened.

Now that is the clinical approach. But the real trick is knowing when to use it and when not to.

RE/P: *So, you think learning good micing technique is important for younger engineers who have grown up with samplers and synthesizers?*

BS: A lot of people forget that a good sample can originate with a microphone, and many times it does. So, if they don't know good microphone technique, their samples are obviously going to suffer.

RE/P: *How do you go about learning good microphone technique?*

BS: That's a tough one. I think a beginning engineer should go out and listen to good acoustical music in a good acoustical environment. When I give seminars, I stress this. The ear has to have a benchmark to really know what music should sound like. You don't have to be micing it to get the critical ear training. That is the most important and the first step in an engineer's education. Then experience in the studio is the way you correlate that with what goes on the tape. Micing is a way of achieving those emotional values, but it's subsequent to that ear training.

RE/P: *Do you credit your training on early equipment with helping you develop your technique?*

BS: I think so. When I started recording in Minneapolis and Chicago, we didn't have much in the way of equipment. There was no such thing as equalization. So, if you were going to brighten up sound, you had to do it either with a different

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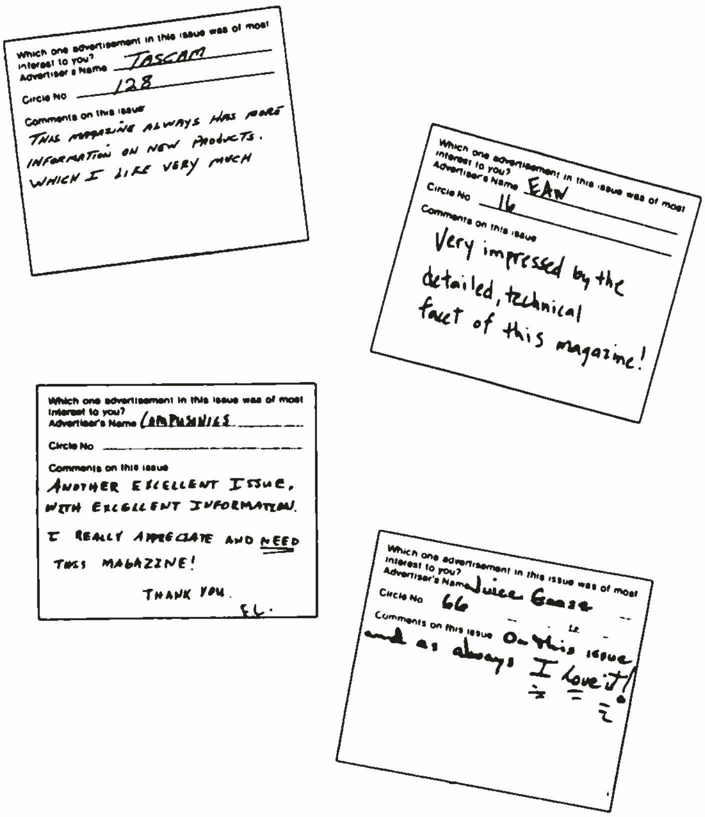

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Discography

The following is a partial list of artists, albums, soundtracks and commercials Bruce Swedien has engineered and/or produced. The complete list is 16 pages!

Artists:
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Dinah Washington
Diana Ross
Nat "King" Cole
Natalie Cole
Roberta Flack
Donny Hathaway
Rufus and Chaka Khan
Patty Austin
The Chicago Symphony
Woody Herman
Duke Ellington

Albums:
Michael Jackson—"Off The Wall," "Thriller," "Bad"
Quincy Jones—"Roots," "The Dude," "I Heard That"
Jennifer Holliday—"Say You Love Me"
Missing Persons—"Rhyme and Reason"
The Jacksons—"Victory"
Sergio Mendes—"Sergio Mendes," "Confetti"
Herb Alpert—"Blow Your Own Horn"
James Ingram—"It's Your Night"
Donna Summer—"Donna Summer"
George Benson—"Give Me The Night"
Patti Austin—"Every Home Should Have One"

Soundtracks:
"The Wiz"
"Nightshift"
"Running Scared"
"The Color Purple"
"Roots"

Commercials:
Budweiser
Miller
Michelob
Wrigley Gum
General Motors
Kellogg's
General Mills
Standard Oil
Braniff
Vick's
7-Up
McDonald's
Procter & Gamble
Pepsi-Cola
Levi Strauss & Company

Two AKG C-414 EB microphones placed one above the other in omnipattern, no EQ, no compression, using Studio Tecnology's Mic Pre-Eminence direct to tape.

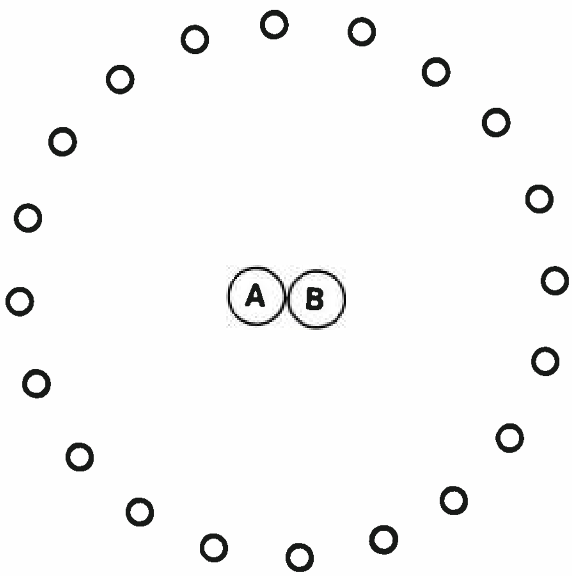

***Figure 2.** Westlake studio D, "Man in the Mirror," Andrae Crouch choir.*

microphone or with some other method. That training kept me from getting really lazy in the studio, and I've never forgotten it.

***RE/P:** What is your opinion of near-field monitoring?*
BS: The real crux of the situation is that the near-field speakers are closer to the ear than to any acoustical border, so there's less reflected sound. In near-field monitoring, I always monitor at a fairly soft level. It's closer to what we use in the home. My feeling with mixing, and, indeed, with all of what I do, is that we make these recordings to be listened to in the home, and not in the studio. The ultimate test is at a home-listening level on close to average speakers.

I always use near-field monitoring in the final mix. I always use both JBLs and Auratones. With my JBL 4310s, I try to keep it at an SPL of no more than 83. The buss peaks are +3. With the Auratones, I adjust speaker levels for an SPL of 80, and buss peaks of +3. I adjust SPL resultant peaks accordingly, if lower buss peaks are to be used. I use a Simpson (Type 2) SPL meter.

***RE/P:** How about main monitoring?*
BS: I use main monitoring for normal recording.

***RE/P:** Do you have a favorite system?*
BS: I prefer the Westlake Audio SM 1 monitoring system, the speaker Westlake Audio has installed in Studio D. It's three main speakers with a five-way system, and the whole system uses 3,000W of amplification. Each speaker consists of two 18-inch woofers, a mid-bass speaker, two compression drivers and a super-tweeter.

The basic system is a 35-cubic-foot enclosure, and it's all built to extremely exact mechanical specifications. To my knowledge, it's one of the widest-bandwidth, non-custom speakers available. The bandwidth is from 18Hz to 20kHz.

But being in the business as long as I have and working in as many studios all over the world as I have, I've tried to learn not to listen to the speakers, but to try to listen through them. That's a real trick. A lot of people in my line of work get very, very hung up on speakers, and will only work with one kind of speaker. I've tried never to be that intimidated by the equipment. My opinion has always been that the only reason we're in the studio is just for the music. Nothing else. Nobody ever left the recording studio humming the speaker or the control console or the tape machine—only the music.

RE/P: *Describe the role of your second engineer.*

BS: I rely on him very heavily, always with guidance in the beginning, but not for any of the creative aspects of what I do. I still place all the microphones myself. With second engineers, I have an incentive program: One mistake and you're through. I'm probably a little hard on them. But all the ones who have lasted more than a year with me have gone on to be very successful engineers on their own.

The second engineer has to be a fanatic for detail. He's responsible for logging tapes, technical details with the equipment, alignment procedures. I give my second engineer a print-out from the Macintosh for his daily responsibilities, setting monitor values, tape speeds and alignments, tone levels and so forth.

My current second engineer is Brad Sundberg. He's 6'3" and very strong. I also look for physical strength, because we work long hours. Staying power is very important to me. I have been known to go through two and three second engineers in one mix day, but I prefer just to have one guy who can hang in there with me.

RE/P: *What technical trends would you like to know more about or get into more?*

BS: I'm a Macintosh fanatic. I really want to get into MIDI archiving. In my work, I get a lot of incredible keyboard players in the studio, anybody from Jimmie Smith to Oscar Peterson, and what I want to be able to do is have them play a MIDI instrument during the session, and record that performance in the Macintosh. Then, later on, I can sit and use that digital information and get the perfect synthesizer color or sound on that performance without having to do that during the performance.

Quincy and I frequently describe synthesizer recording as being like painting a 747 with a toothbrush. It takes a long time when you're as fussy as we are, getting that color, or timbre, right. SMPTE time code lets us take many different formats of recording devices, and even musical instruments, and run them together.

With SMPTE and MIDI instruments, I can recreate a performance on a different instrument. If you sit with a performer, and maybe the color doesn't fit just right, maybe we can't get the ultimate emotional value out of it, by the time we screw around with it, everybody, including the performer, has gone to sleep. The performance won't be a fraction of what it would be if you could get the spontaneity. So, we set up a synthesizer that will send MIDI information and get a color that will work, but it's just not perfect. So, you just have them play it 'til you're pleased with the performance, and then we have that in MIDI archive, and later on I can go back and get exactly the sound I want.

I've always been interested in taking a sound and running it through a device to enhance or change it. But, remember, all music is conceived to be heard with acoustical support. The synthesizer is nothing more that a representation of the orchestra. It doesn't have to sound anything like the orchestra to be that representation. In essence, I guess I'm saying that music is organic. We respond to it emotionally by hearing things balance and sound in a specific way that elicits a different emotional response in the listener. But, even if you want to change that sound, you have to know what the real thing sounds like first. One of the things that bugs me is that the really technical people forget that all you can do with music is listen to it. We don't eat it. You have to keep that in perspective.

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Hands On: Analog Noise Reduction

By Jim Rogers and Jeff Dennerline

Spectrum Studios evaluates Dolby A, dbx, Telcom and Dolby SR.

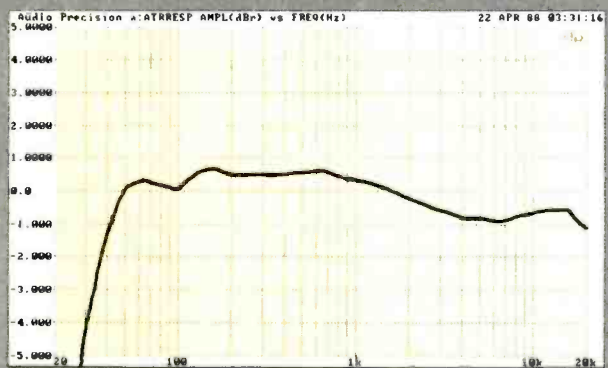


Figure 1A. Frequency response of ATR-102 ½-inch machine at 30ips.

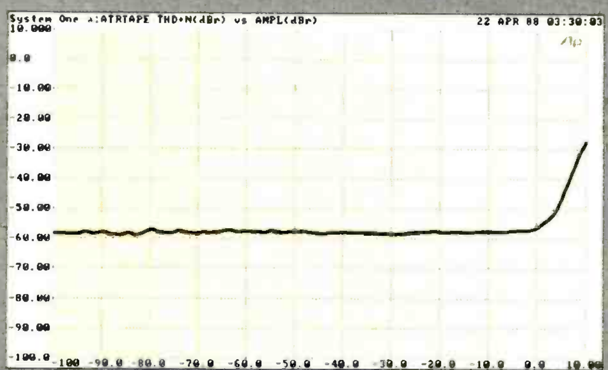
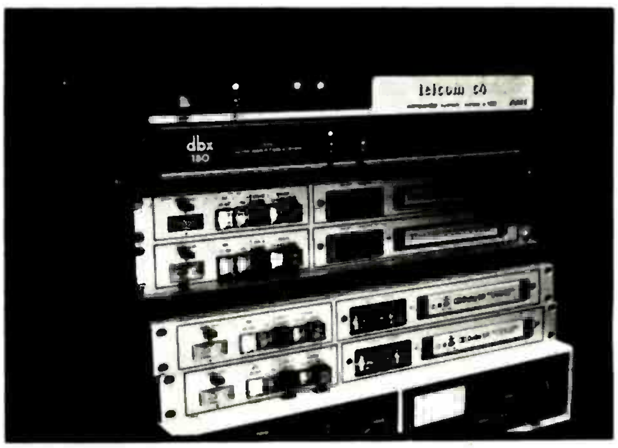


Figure 1B. Noise floor of an ATR-102 ½-inch tape recorder at 30ips (noise minus fundamental vs amplitude).

We are in a wonderful period in the history of recording with technology taking quantum leaps about every six months, it seems. Digital recording was created roughly 10 years ago and now that we can razor-blade-edit digital tape without any trouble, the old analog tape recorders are being used less and less. However, those old analog recorders still have merit. With a good noise reduction system, they can become almost as powerful (and in some ways, more powerful) as the "number-scramblers" out there. Thus, another look at encode/decode noise

Jim Rogers is chief technical engineer, and John Dennerline is a staff engineer at Spectrum Studios, Portland, OR.



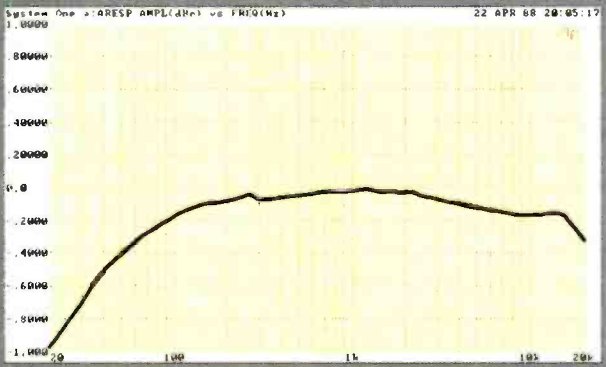


***Figure 2A.** Frequency response of Dolby A-type system, no ATR. (Note expanded scale of ±1dB.)*

***Figure 3A.** Frequency response of dbx system, no ATR. (Note expanded scale of ±1dB.)*

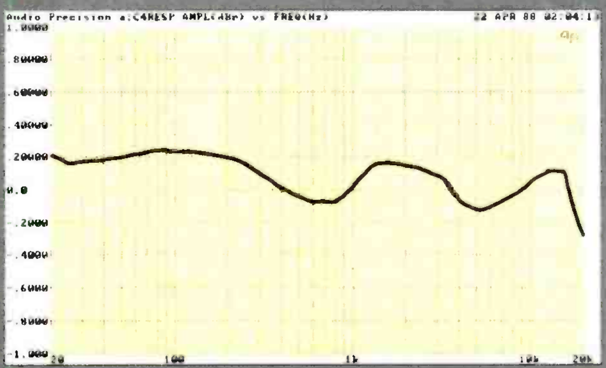


***Figure 4A.** Frequency response characteristic of Telcom c4 system, no ATR. (Note expanded scale of ±1dB.)*

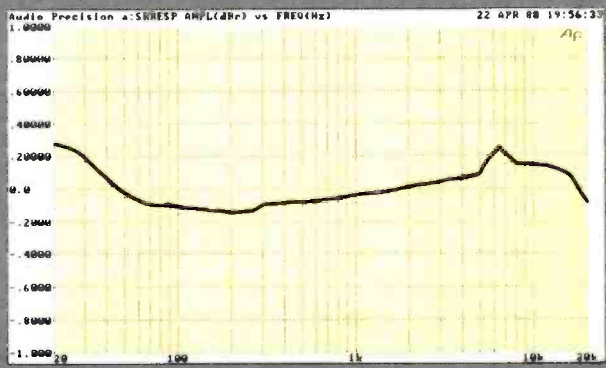


***Figure 5A.** Frequency response of Dolby SR, no ATR. (Note expanded scale of ±1dB.)*

reductions systems is in order.

Dolby A—The first of the modern noise-reduction systems came out in 1966 with the introduction of the Dolby A-type compandor. The first unit was a large, single-channel device, later to be condensed to the now-ubiquitous Cat. 22 card.

The Dolby A-type compandor is a four-band device with a non-linear compression/expansion characteristic. The four bands are 80Hz low-pass, 80Hz to 3kHz bandpass, 3kHz high-pass and 9kHz high-pass. The input signal is conditioned through a 34kHz low-pass filter to prevent high-frequency interference from entering the system and affecting the compandor circuitry. In its encode mode the compressed signal is combined (in a feed-forward manner) with the original signal and then fed to tape. In its decode mode, the expanded signal is subtracted from the full component signal (original plus compressed signals) in a feed-backward manner.

Dolby A-type noise reduction is available in Cat. 22 card form (for 361 single-channel units and M series multichannel main frames) and in the Dolby SP and XP series multichannel mainframes. The Dolby A-type system claims an unweighted noise level 75dB below Dolby Level (reference fluxivity level of the tape recorder) and a frequency response of 30Hz to 20kHz (±1dB).

dbx—To offer an alternative to the Dolby A-type system, dbx decilinear noise reduction (also known as dbx type I) was introduced in about 1972. The dbx noise reduction system features a 2:1 linear compression slope with complimentary 1:2 expansion. In addition to a straight linear compression, the system adds high-frequency pre-emphasis in the encode (compression) mode and high-frequency de-emphasis in the decode (expansion) mode to reduce tape modulation noise. Dbx also uses RMS level detection to drive its compression/expansion circuitry, because it is least affected by any distortion induced by phase-shift.

Dbx has been and is available in many

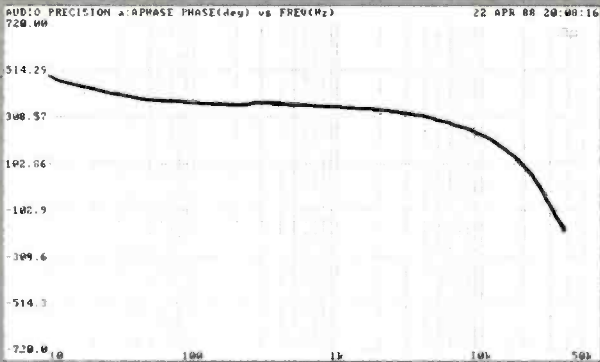


Figure 2B. Phase response of Dolby A-type system, no ATR.

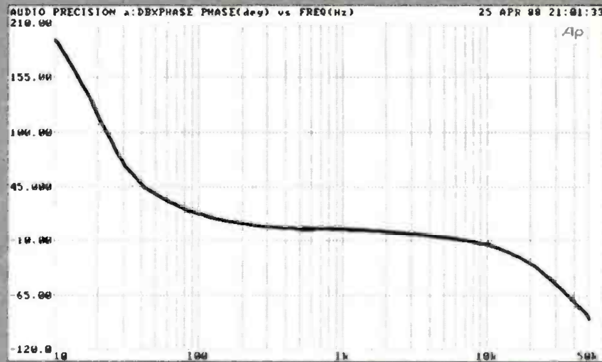


Figure 3B. Phase response of dbx system, no ATR.

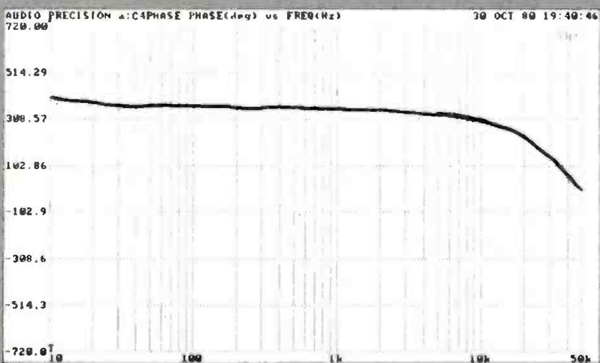


Figure 4B. Phase response of Telcom c4 system, no ATR. (Note similarities to Figure 2B, Dolby A-type system. Both are four-band systems.)

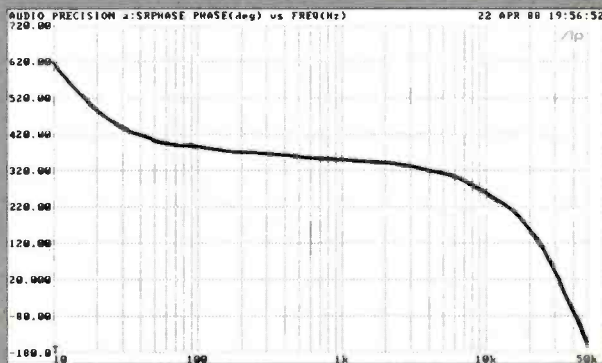


Figure 5B. Phase response of Dolby SR, no ATR.

forms. Among them are the 180A and 150X two-channel units and the 911 single-card unit (which fits dbx's 900 series rack). The dbx type I system claims a frequency response from 30Hz to 20kHz (±0.5dB) and a signal-to-noise ratio of 107dB.

Telcom c4—The Telcom noise-reduction system was developed as a personal project in the early 1970s by an engineer at Telefunken (Germany). The c4 is much like the Dolby A-type system in that it is a four-band companding system. (The bands for c4 are 30Hz to 215Hz bandpass, 215Hz to 1,450Hz bandpass, 1,450Hz to 4,800Hz bandpass and 4.8kHz to 20kHz.) The compression/expansion ratios are also the same as Dolby A-type's ratios (1.5:1 and 1:1.5 compression/expansion, respectively), but the big difference is that the c4 compandor circuit has a linear slope. Telcom c4 noise reduction is available in the 100, 200 and 300 series two-channel units, the ES4 four-channel unit, the 400 series multitrack system and in the c4 DM card, which is Cat. 22-compatible. Telcom c4 claims a frequency response of 20Hz to 25kHz (2dB) and a 115dB dynamic range (A weighted).

Dolby SR—The most recent entry into the realm of noise reduction is Dolby SR. The Spectral Recording process is quite complex and involves situation-dependent, dual-band, multilevel, dynamic signal processing (compression).

The method used to implement band splitting is at the heart of the SR process. Instead of using a conventional fixed-band filter, the SR process uses fixed- and sliding-band filters in each of three level-processing stages (high-, mid- and low-level). The high- and mid-level stages are further split into high- and low-frequency stages. The low-level stage only involves the high-frequency band, where tape hiss is most prevalent. The stages work in such a way that when fixed-band action works best, it is employed, and when sliding-band action works best, it is substituted for the fixed-band filter. Dolby SR is available in card form (Cat. 22 retrofit) in the Cat. 280 card, as well as the Cat. 431, which fits Dolby SP/XP series racks. The SR form of noise reduction claims a usable dynamic range of 90dB to 95dB, with overall frequency response from 20Hz to 20kHz (±1dB).

The objective evaluation

Items involved in this evaluation were unmodified (stock items). Three of the systems were in-house units, and the Telcom system was provided by ANT Telecommunications. Units included: the dbx 180 (serial No. 2006), two Dolby 361 mainframes (serial Nos. 10724 and 10700), two Cat. 22 Dolby A-type cards (serial Nos. 26111 and 31686), two Cat. 280 Dolby SR cards (serial Nos. 13 797 and 17 885) and a Telcom e114 two-channel system (serial No. 01053). The tape recorder was an Ampex ATR-102 ½-inch recorder running at 30ips with Ampex 456 tape. (Graphs of its response and noise characteristics are shown in Figures 1A and 1B.)

The goal of this test was two-fold. First, each unit would be evaluated in and of itself, and, secondly, each unit would be connected to the tape machine and evaluated in a typical-use situation. The test equipment used for all measurements was the Audio Precision System One. In addition to the noise floor measurements, including ATR and tape, four specific test

The c4 is much like the Dolby A-type in that it is a four-band companding system.

parameters have been chosen: each unit's frequency and phase response (tested at 0 reference level [+4dBu] with bandwidth limiting removed), distortion characteristics (third harmonic distortion + noise, also at 0 level) and a sweep in the amplitude domain to show system noise (minus the fundamental frequency vs. amplitude of the noise). The results shown are from a 1kHz downward amplitude sweep. Sweeps at 400Hz and 4kHz yielded similar results, so the 1kHz data is sufficient for demonstration purposes.

Dolby A—Based on our technical analysis, Dolby A-type noise reduction had the flattest frequency response. (See Figure 2A.) The low-end roll-off can be attributed to the bandwidth limiting filters in the Cat. 22 card. (Please note the expanded scale in the figures.) The system phase response is also fairly decent. In spite of the low-end performance, the phase response at low frequencies is quite good, meaning that the filters were either well designed or their action was gentle enough to keep phase distortion to a minimum. (See Figure 2B.)

The system THD+N distortion characteristics (See Figure 2C.) seemed, at first, to be a bit odd with the big bell from 1kHz to 10kHz. But, this characteristic seems normal for four-band companding systems. The rise in distortion at the low end can be attributed to either of two things: the transformer in the 361 mainframe or the low-end roll-off filter in the Cat 22 card. (The rising distortion characteristic in the low end, coupled with the low-end roll-off, is probably why some engineers prefer to bypass their A-type NR when recording drums on multitrack.)

The real test of any noise-reduction system is its ability to reduce tape noise. Figure 2D shows the system noise floor, and Figure 2E shows the system's ability to reduce tape noise. Although this system

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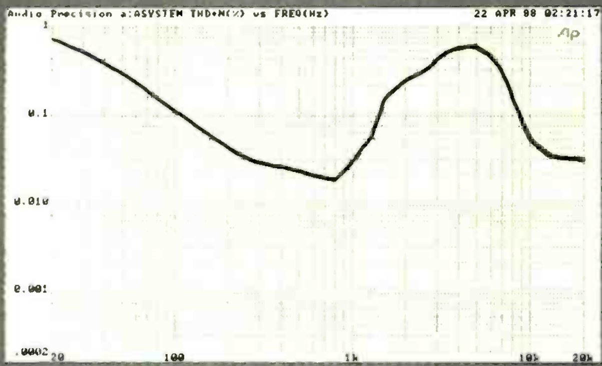

Figure 2C. Distortion characteristic of A-type system, no ATR.

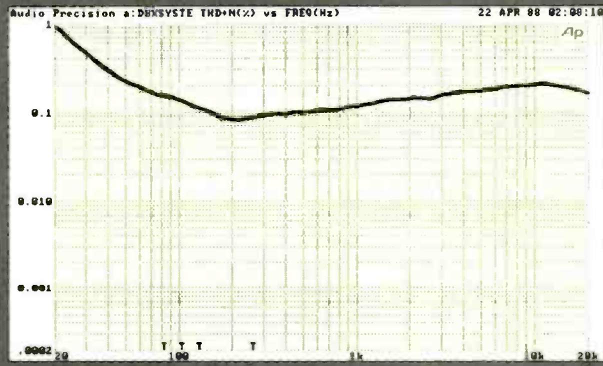

Figure 3C. Distortion characteristic of dbx system, no ATR.

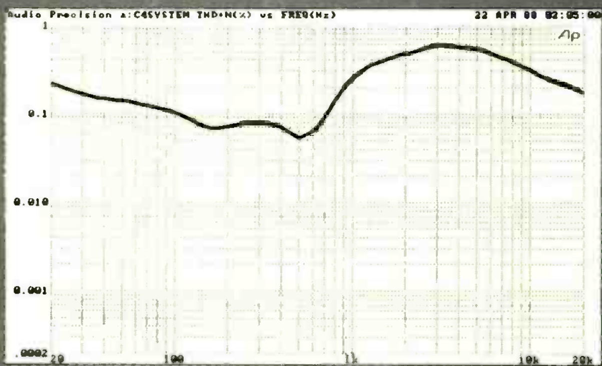

Figure 4C. Distortion characteristic of Telcom c4 system, no ATR.

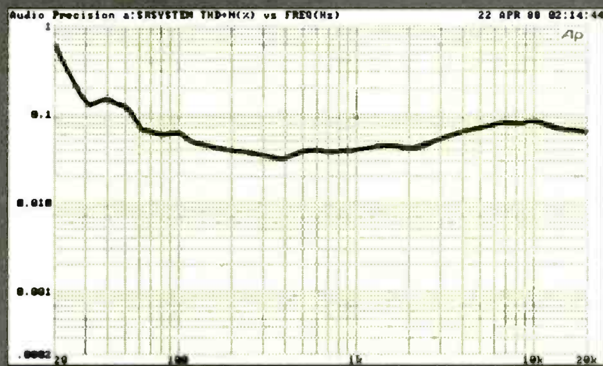

Figure 5C. Distortion characteristic of Dolby SR, no ATR.

has a very admirable noise floor, based on test analysis, its ability to reduce the noise from tape at very low levels is the least effective of all four systems. However, it was the first system available, and it does perform as specified.

dbx—Dbx does well compared to the other systems. The system's frequency response (again, note the expanded scale) is within spec. (See Figure 3A.) The phase response (See Figure 3B.) is excellent down to about 100Hz, and then it swings almost 200° in the bottom three octaves. This response typically indicates either a high-pass filter with a very steep slope or an active balanced input with coupling capacitors that are too small in value. Checking the circuit shows that it is definitely the steeply sloped high-pass filter causing this phenomenon.

Dbx uses the high-pass filter to keep extremely low-frequency information, which causes mistracking, from getting into the system. The distortion characteristics (See Figure 3C.) of the dbx system show that it is a respectable device. The rise in distortion in the low end is most likely caused by the high-pass filter, while the slight peak in the high end could be, in part, from the pre-emphasis/de-emphasis employed to reduce modulation noise.

Compared to the Dolby A-type noise reduction tested, dbx seems to rely more on masking (the phenomenon that occurs when program material is at a much higher level than the tape noise) than its predecessor. The use of this technique is evidenced by the gentle slope in the noise floor from an applied signal amplitude of +10dBV to −50dBV when it settles to its lowest noise floor. (See Figures 3D and 3E.)

Telcom—Telcom c4 is an example of the new generation of noise-reduction systems, even though it's almost 15 years old. As shown in Figure 4A, the unit's frequency response is well within acceptable limits (Again, note the expanded scale.) with a sharp high-end roll-off that occurs above 20kHz. (The unit is 2dB down at 25kHz.) The unit's phase response (See Figure 4B.) measures the best of all units tested. This result can be attributed to the lack of significant signal conditioning in the main signal path. As you can see, Telcom c4 exhibits a bell curve similar in the mid/high end (See Figure 4C.) to the Dolby A-type unit's. This curve is probably normal for four-band-type compandor circuits. However, the low-end distortion is lower because of the lack of conditioning stated previously, as well as the lack of transformers in the circuit. Like dbx, Telcom tends to rely more on masking than Dolby A-type does. Its ultimate noise floor, even with tape, is quite low. (See Figures 4D and 4E.)

Dolby SR—Dolby SR is the most recent analog noise reduction system developed, and it performs well, as shown in Figures 5A and 5C. The only question that might be raised is one of phase response (See Figure 5B.) with a somewhat broad swing, which is evidence of some complex processing going on within. (It was a bit difficult to interpret the data of this test without having schematics to evaluate.) The noise-reduction characteristics of Dolby SR are quite strong. Like the A-type, Dolby SR relies on masking only at amplitudes above the nominal operating level and has a surprisingly low, flat noise floor. (See Figures 5D and 5E.)

The listening evaluation

For the listening portion of this evaluation, four types of program material were recorded: acoustic piano, electric bass, spoken word and a cymbal (crashed and lightly struck at the bell). Technically, all material was recorded through a Solid State Logic 6056E (VCAs bypassed) onto a 24-track Otari MTR 90 MKII using Ampex 456 tape at 30ips. Playback was through three separate monitoring systems: UREI 813Cs, Tannoy NFM 8s and a pair of Stax headphones. (For a further list of equipment, refer to "Facility Profile: Spectrum Studios," in the March issue, page 74.) Once set, a constant level was established by grouping all channels. Individual channels were not listener-adjustable; only overall system level could be adjusted.

One participant claimed anyone who couldn't identify system 1 must be deaf. (He was wrong.)

Seven people participated in the listening evaluation: Six are staff engineers at Spectrum and have widely varied backgrounds, and the seventh participant is a prominent local musician/composer. They did their evaluations alone and noted their answers on a simple questionnaire. (See Table 1.) Most of the questions required simple yes or no answers; comments were optional. In some cases, no response was given when a yes/no reply was indeterminate.

This listening evaluation was not designed to rate or rank the systems in any way; it was developed as a way of perceiving detectable differences between the systems based on variable program sources. This was a blind assessment. A control was not deemed necessary because participants were not told which system they were listening to until all participants had completed the appraisal.

Dolby A-type noise reduction was the oldest design tested and was determined to be the least effective in controlling tape noise. However, it consistently received high marks for transparency. Although some sonic colorations were perceived,

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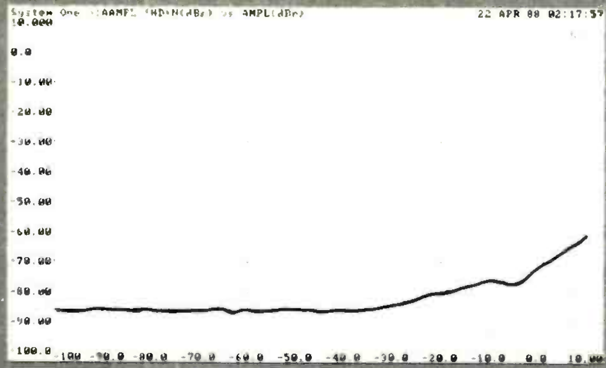

Figure 2D. *Noise chacteristic of A-type system, no ATR. (Noise minus fundamental 1kHz frequency vs. amplitude.) 1kHz swept from +10dBr to 100dBr. ODBR = +4dB.*

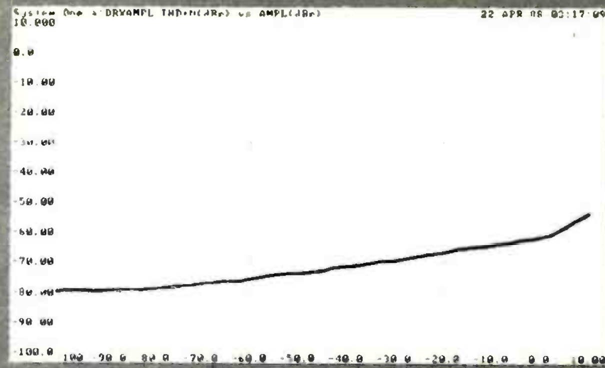

Figure 3D. *Noise characteristic of dbx system, no ATR.*

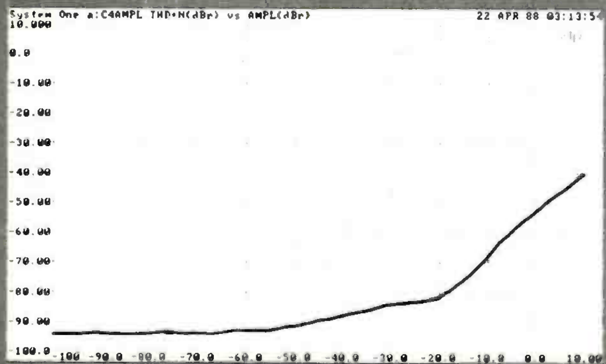

Figure 4D. *Noise characteristic of Telcom c4 system, no ATR.*

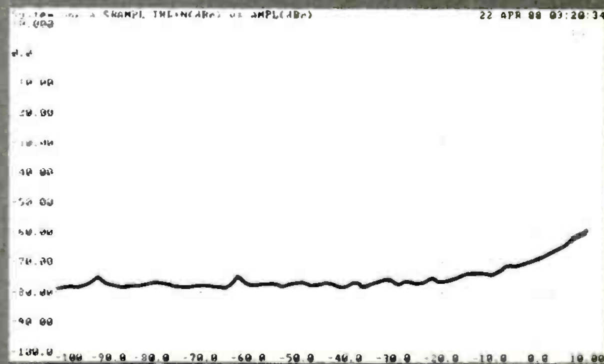

Figure 5D. *Noise characteristic of Dolby SR, no ATR.*

Table 1. *Listening test results.*

	Dolby A		Dolby SR		dbx		Telcom	
Piano	**yes**	**no**	**yes**	**no**	**yes**	**no**	**yes**	**no**
Perceive tape noise?	7	0	1	6	1	6	3	4
Perceive pumping?	2	5	3	4	3	4	1	6
Natural decay?	6	1	6	1	4	2	6	1
Transparent high end?	5	2	6	1	5	1	4	2
Bass	**yes**	**no**	**yes**	**no**	**yes**	**no**	**yes**	**no**
Perceive tape noise?	6	1	0	7	4	3	3	4
Perceive pumping?	2	5	2	5	3	4	1	6
Natural decay?	6	1	7	0	5	2	6	1
Transparent high end?	6	1	2	5	4	3	3	3
Spoken word	**yes**	**no**	**yes**	**no**	**yes**	**no**	**yes**	**no**
Perceive tape noise?	7	0	5	2	5	2	7	0
Perceive pumping?	2	5	2	5	0	7	1	6
Natural decay?	5	1	5	1	6	0	6	0
Transparent high end?	5	1	3	3	4	2	3	3
Cymbal	**yes**	**no**	**yes**	**no**	**yes**	**no**	**yes**	**no**
Perceive tape noise?	6	1	0	7	1	6	2	5
Perceive pumping?	1	6	2	4	3	4	2	5
Natural decay?	6	1	5	2	4	3	7	0
Transparent high end?	5	2	5	2	3	4	5	2

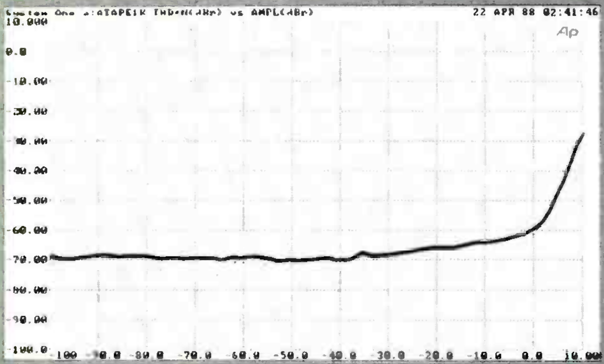

Figure 2E. *Noise floor of Dolby A-type system and ATR-102 ½-inch at 30ips.*

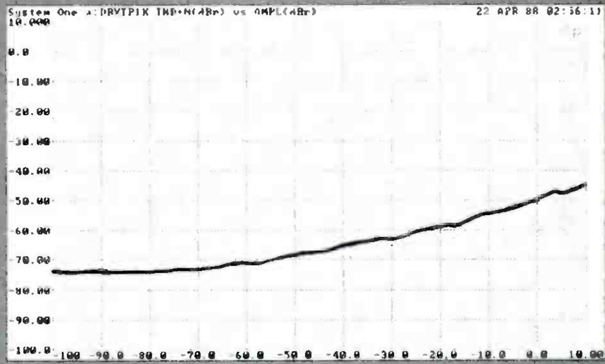

Figure 3E. *Noise characteristic of dbx system with ATR-102 ½-inch at 30ips.*

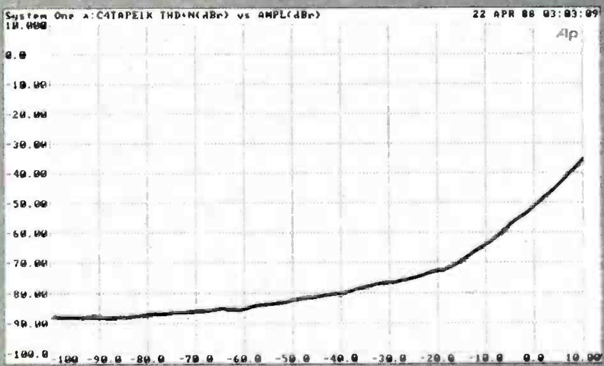

Figure 4E. *Noise characteristic of Telcom c4 system with ATR-102 ½-inch at 30ips.*

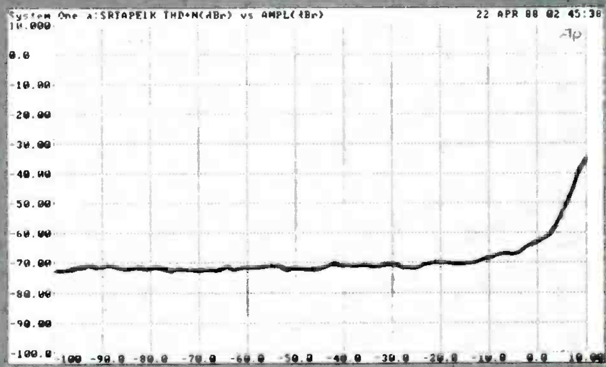

Figure 5E. *Noise characteristic of Dolby SR with ATR-102 ½-inch at 30ips.*

most participants liked the overall sound.

Dbx was most effective with spoken word and piano. It was reasonably good at controlling tape hiss, but most people could hear it "work." It's likely that, because of the pre-emphasis/de-emphasis employed, more people could hear colorations in the high end.

The Telcom c4 system was the only system unfamiliar to the listeners. It exhibited characteristics of all the other systems and often closely paralleled Dolby SR in listener response. The system usually demonstrated the least audible "pumping" effect, regardless of source material used.

Dolby SR produced various responses. It was an obvious favorite with some listeners and a non-favorite with others. It more consistently eliminated tape noise, but participants seemed to notice alterations in the high end when simple signals (bass and voice) were used as a test source.

One interesting note: In only one case, and with only one signal source, could no pumping or "working" be heard. That was the dbx system being used on spoken word. This lack of pumping could be attributed to the fact that dbx is used on all voice/media work at Spectrum, and the staff engineers have become accustomed to hearing it everyday. Interesting!

Print-through was not detectable on any system.

This survey was designed to help engineers better judge the systems available and the viability of noise reduction, in general. It brought a long-awaited death to many of the preconceived notions about noise reduction. Only one of seven persons correctly identified the four noise reduction systems. One participant went so far as to claim that anyone who couldn't identify system No. 1 must be deaf. (Of course, he was totally wrong.) It proved, once again, that a well-aligned analog tape recorder equipped with a professional noise-reduction system is a viable option in today's digital world. RE/P

For more information on the systems please contact the following:

dbx, 71 Chapel St., Newton, MA 02195

Dolby Laboratories, 100 Potrero Ave., San Francisco, CA 94103-4813

Ram Broadcasting Systems, 346 West Colfax St., Palatine, IL 60067 (exclusive U.S. distributors of ANT/Telcom noise reduction)

The results of this evaluation are not to be construed as an endorsement for any of these products, by either the authors or *RE/P.*

Low-Print Mastering Tape

By Helge Kristensen and
Warren K. Simmons

These tapes are designed to have very low print-through while retaining high MOL and low bias noise.

In the late 1960s, mastering tapes were still made from acetate base film and were non-back-coated. In the next generation of mastering tapes, polyester base film with greater tensile strength and less sensitivity to environmental changes was substituted for the acetate base. These tapes were typically in the 290-oersted to 310-oersted coercivity range and required bias levels at the low end of the normal biasing capability of professional analog recorders. In the early 1970s, the first generation of conductive back-coated mastering tapes was introduced. The addition of the conductive back-coating provided improved tape winding performance and cleaner running because of reduced static pickup. In the mid-1970s, higher-performance mastering tapes became available, with coercivity levels in the range of 320 oersteds to 340 oersteds. These improved tapes require slightly more bias current and perform with greater maximum output level and reduced bias noise.

Today, analog mastering tapes can be grouped into three categories, as follows:

High-performance. Tapes characterized by high maximum output level (MOL) and low bias noise.

Standard-performance. Tapes having somewhat lower MOL and slightly higher bias noise.

Low-print mastering tapes. These tapes are designed to have very low print-through while retaining high MOL and low bias noise. Low-print tapes typically use oxide particles requiring 2.0dB to 2.5dB greater bias current and showing print-through levels in the −58dB to −61dB range. (See Figure 1.)

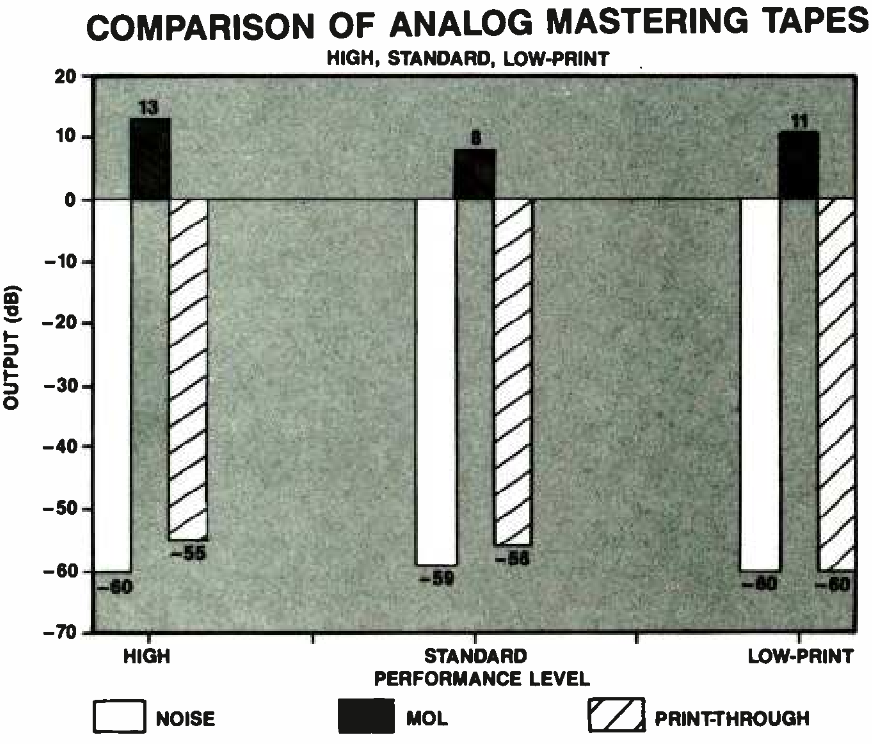

Figure 1. Comparison of analog mastering tapes.

Professional analog recorders generally are designed with tape-biasing capabilities ranging from 290 oersteds at the low end up to 380 to 390 oersteds at the high end.

The latest generation of professional analog recorders have biasing capability for tapes having coercivity levels somewhat higher than 390 oersteds. However, with the large installed base of older recorders, the magnetic tape design engineer today must still limit analog tape design to magnetic particles in the 290 to 390 oersted range.

Print-through phenomenon and measurement methods

A. Print-through is the unintentional magnetic transfer of a recording from one layer of magnetic tape to the adjacent layers when wound and stored on a reel. The print-through is heard at a low level as "previews" (preprint) and "echoes" (post-print) of the recording. The effect is most apparent when a loud section of the re-

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cording is preceded or followed by a quiet section. Time and high temperature aggravate print-through.

B. Various methods for measuring print-through exist. The following method is used for mastering tapes throughout the industry and is intended to simulate actual recording conditions. Wind 10 layers of the tape on a takeup reel at a tape speed of 15ips with operating bias only. Then record one layer with a 1kHz signal at a maximum record level (the level at which the THD reaches 3%). Repeat these two steps at least three times. Store the tape on the takeup reel for 24 hours at 70°F. Without rewinding, play back the tape; i.e., the takeup reel with the test tape becomes the supply reel. Measure the output through a suitable filter (typically 1/10-octave) and display it on a chart recorder for convenience. (See results in Figure 2.) Use the strongest copy signal, normally the preprint, and calculate the average of at least three measurements. Report the print-through level as the difference between the copy signal and the recorded signal level in decibels.

Another method is identical to the first one except for the storage condition, which is four hours at 150°F. This test, which is intended to generate the worst-case condition, typically results in a print-through level 4dB to 6dB higher.

Market segments requiring a low-print tape

A comprehensive look at the marketplace shows four distinct market segments that, either by custom or by actual requirement, have standardized on low-print-through tape performance.

1. In the film sound business, the portable Nagra recorder is commonly used on location to record the original sound as the film shooting takes place. In post-production, dialogue, as well as various sound effects, may be added. Here, low-print tape is necessary to minimize the echoes and previews of door slams, gun shots or other high-level sounds followed by silence. Typically, tapes having a print-through characteristic of −60dB or better are required.

2. Video post-production operates in much the same manner, adding dialogue and sound effects. For the same reasons as film sound, this business requires tape having very low print-through.

3. The international broadcast market puts two demands on magnetic tape. In the European broadcast market, radio broadcasting is often government-controlled. Tape specifications become part of the purchase contract and low-print tapes are specified. European broadcasting operations commonly use the DIN or CCIR hub and fast-forward and rewind speeds as high as 15 meters per second, so the mechanical winding properties of the tape become critical. In this marketplace, not only must the print-through level be −60dB or better, the winding characteristics also must permit smooth pack formation at the higher winding speeds.

The desirable qualities of a high-performance mastering tape need to be maintained while achieving the lowest possible print-through.

4. An important market segment requiring low-print tape is in studio mixdown, in which certain types of music, such as

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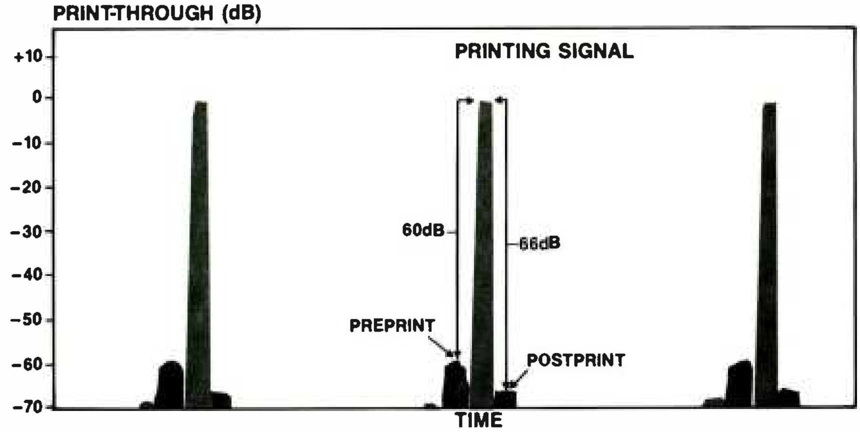

Figure 2. Graphical record of a print-through measurement.

classical, operate with very loud passages followed by very soft passages. Lack of a low-print tape could cause audible previews and echoes of the loud passages.

Design and development of a low-print mastering tape

As in all good product-design practice, a careful, detailed assessment of market needs is an essential first step. The result of market research must establish all critical electrical and mechanical performance criteria.

Electrical design characteristics—In a broad sense, the desirable qualities of a high-performance mastering tape need to be maintained while achieving the lowest possible print-through. A signal-to-noise ratio of 71dB or better, 0.28% THD of a 1.0kHz signal at a record level of 370nWb/m, and modulation noise level of −70dB* should be achieved. An MOL of at least 11.0dB above a record level 370nWb/m is required. Print-through level must be −60dB or better. Sensitivities at all frequencies must be in the same range as in high-performance tapes.

Selection of magnetic material(s)—The print-through level of a magnetic tape is determined mainly by the selection of the oxide particle and its concentration in the formulation. The oxide particles should have relatively high coercivity and maximum temperature stability. Higher-coercivity particles are more difficult to magnetize and, therefore, become more resistant to magnetic transfer. High-coercivity oxides also provide lower bias noise because the particle sizes are generally smaller.

Particle selection also requires consideration of third harmonic distortion, MOL and sensitivities at all audio frequencies.

*Level of 700Hz sideband of 1.0kHz signal.

Tape geometry—Print-through is reduced when the coating thickness is reduced and when the base film thickness is increased. However, a reduced coating thickness also reduces MOL and affects bias setting and frequency response. Therefore, at the same time, thinner coating also affects the recording equalization and S/N ratio. Increased base film thickness, of course, reduces the length of tape that can be wound onto a given-size reel. The design engineer must consider these trade-offs.

Manufacturing considerations—The tape processing method also influences the print-through characteristics. The milling process must be controlled carefully to minimize particle breakage and provide optimum dispersion characteristics. Excessive particle breakage degrades the print-through level.

Modulation noise is highly influenced by manufacturing techniques and, therefore, must also be precisely controlled to achieve the lowest possible modulation noise level.

Recommended studio practices

From a tape/hardware interface point of view, some recorder settings and adjustments are essential in achieving optimum performance. Also, some settings and adjustments are at the discretion of the studio engineer.

Bias current and equalization adjustments are specific and essential.

The proper bias current adjustment is different for each tape and varies depending on the specific tone characteristics. Higher-coercivity tapes require greater bias current than tapes having lower coercivity. From the recorder point of view, bias current varies with tape speed and the length of the record head gap. [Note: "length" of the head gap is the correct description of dimension]. At the proper bias setting, minimum third harmonic distortion and minimum modulation noise are achieved simultaneously.

Equalization adjustments are necessary to achieve flat response within the audio frequency range and vary for each tape. Follow the tape manufacturer's recommendations for proper bias adjustment depending on the tape, recorder speed and record head gap length. Record level is at the discretion of the recording engineer. The setting of record level results in some trade-offs. On the positive side: With high record levels (up to +6dB, 370nWb/m), there is less apparent playback noise. On the negative side: There is a higher apparent print-through level, reduced headroom and higher third harmonic distortion. With lower record levels (0dB or 185nWb/m) the apparent print-through level, distortion and head room all improve at the expense of apparent playback or bias noise.

The use of noise reduction systems at relatively lower record speeds vs. no noise reduction systems at higher record speeds is also at the discretion of the studio engineer.

The care, handling and storage of the master tape is of critical importance.

1. Before and after each recording session, all parts of the recorder in the tape path should be cleaned and inspected for burrs or sharp edges that could scratch the tape or damage the tape edge. In addition, proper, periodic head demagnetization is needed.
2. Tapes should be stored tails-out at the end of each recording session to minimize apparent print-through.
3. Tapes should be stored vertically in the box supplied by the tape producer.
4. To minimize print-through further, stored tapes should be fast-wound and rewound several times before over-dubs or mixing operations are performed.
5. Tapes must be stored under carefully controlled atmospheric conditions. Ideal conditions are 65°F to 75°F and 35% to 45% relative humidity.
6. It is desirable, but not essential, to wind and rewind stored tapes once a year to relieve stresses introduced by temperature and humidity changes during storage.

Low-print tape may be just the type of tape for the work you are doing. However, low-print tape is not necessarily suitable for all recording projects. If you're not sure, call your local tape representative.

RE/P

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Digital Audio Tape: The Key to High-Performance Recording

By Rich Collins and Del Eilers

A comparative look at digital and analog audio tapes as they apply to today's recording technology.

As audio professionals assess the merits of expanding from analog into digital hardware, they face a number of complex issues. Not only must they choose between digital formats, but they must also balance their sizable financial investment against the marketing edge and capabilities that digital recording can bring to their studios.

The digital advantage

Other considerations aside, the technological advantages of digital audio recording are clear. From the beginning, signal-to-noise ratio has topped the list. Digital audio recording handles a wider range of levels, and there is no flutter. As the digital signal feeds into the buffer, the system's crystal clock controls the data rate. Speed variations simply don't exist.

Equally important, digital recording promises successive generations of copies without detectable sound-quality degradation. An analog master, for example, might have a dynamic range of 70dB and lose 3dB in a next-generation copy. By the time a recording reaches the client, its dynamic range could be as low as 60dB, compared to a typical digital recording that holds its 90dB dynamic range from recording through mixdown to mastering to duplication. (Figure 1 compares the sound quality of analog and digital tape after multiple-copy generations.)

Rich Collins is audio products manager, and Del Eilers is senior audio engineer, for the Magnetic Media Division of 3M Company, Minneapolis.

Other audio distortions also increase significantly with each generation in analog recording but not with digital. The deterioration that flutter and modulation noise cause can make an analog multiple-generation recording completely unacceptable, but digital does not deteriorate at all. Digital audio tape also retains its premium sound with time, and the analog phenomenon of print-through is nonexistent because print does not occur at digital recording wavelengths.

Convenience and high-performance design

Those venturing into digital recording reap the benefits of a new technology, but they must also adapt their skills to different equipment nomenclature and parameters. Designers of digital hardware and media are taking pains to make the transition smooth.

Digital audio recording is still evolving, but stationary-head digital recorders are designed to mimic their analog counter-

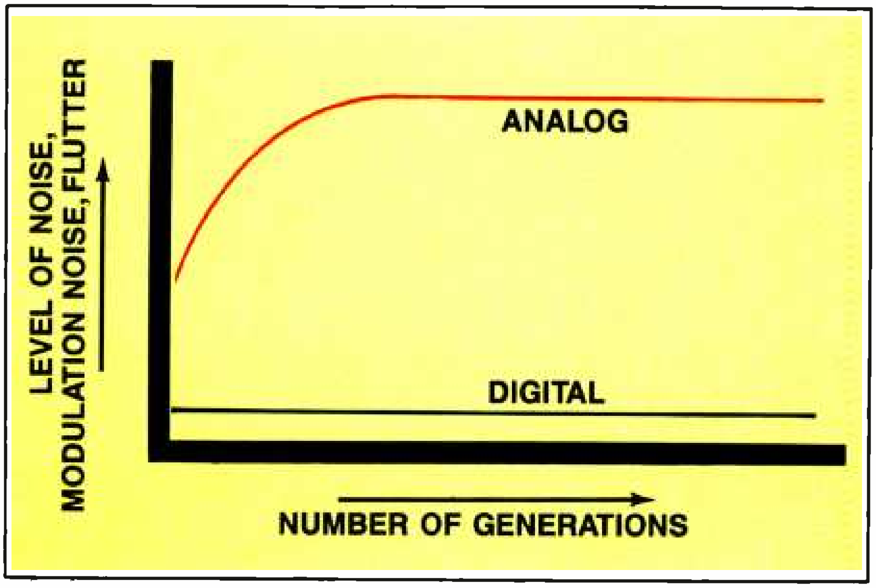

Figure 1. Multigeneration changes, analog vs. digital.

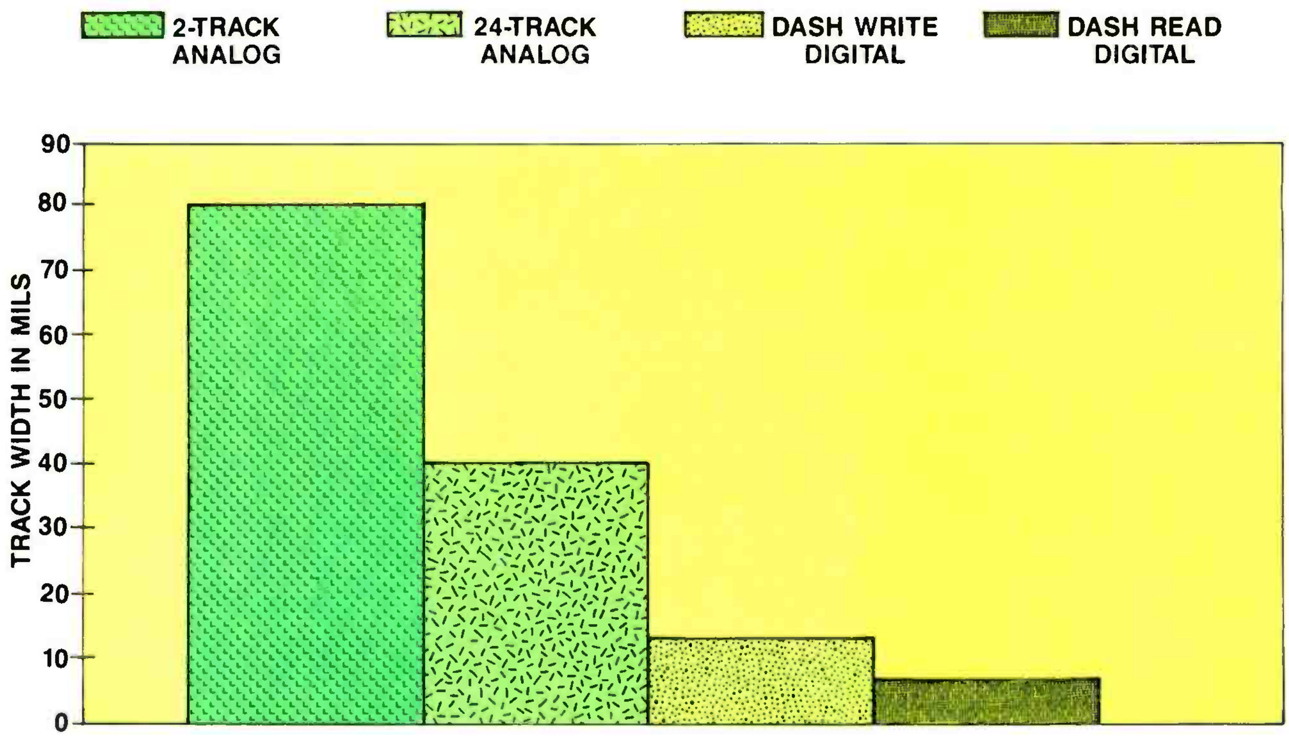

Figure 2. Track width comparison, analog vs. digital.

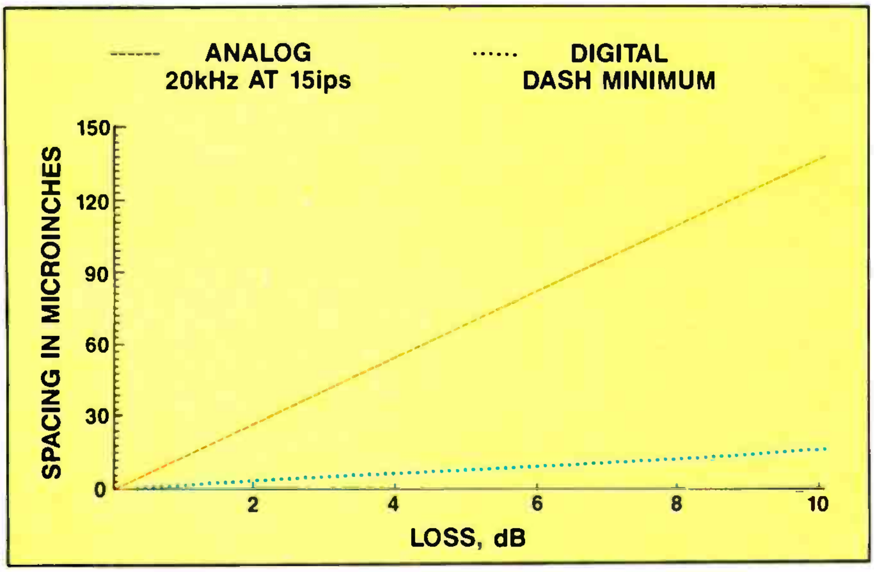

Figure 3. Tape-to-head spacing loss, analog vs. digital.

parts. The familiar reel-to-reel setup remains, along with the typical stop, record, reverse and fast-forward functions.

Although the external trappings cater to traditional habits, the technology itself forces some significant differences. Because of the relatively high tape speed of most digital systems and the importance of providing adequate playing time on a single reel of tape, the base material of digital tape is much thinner than that of a typical analog tape—by about 50%. This thinner base also gives the optimal tape-to-head contact required to record the very short wavelengths of digital signals.

Digital tape requires a much smoother surface than analog tape, and its recording tracks are much narrower. On typical analog recording tape, track widths can range from 40 mils to 80 mils. On today's digital recording tape, however, track widths are much narrower. The Digital Audio Stationary Head (DASH) format, for example, uses tape with track widths on the order of only 13 mils for write operations and 6 mils for playback. The ProDigi (PD) digital recorder format has a track width of 12 mils. (Figure 2 compares typical track widths of analog and digital audio tapes.)

Because digital tape is thin and information is packed densely, cleanliness of the tape and care in handling and editing become critical. In analog recording, for example, debris capable of causing a 3dB loss at the highest frequency (20kHz at 15ips) would be about 41 microinches. In digital recording, much smaller debris—typically as small as 4.3 microinches (1/10 the size)—would cause the same loss. In other words, digital audio tape, which has an information density 10 times greater than analog tape, is about 10 times more sensitive to debris. (Figure 3 illustrates how an increase in tape-to-head spacing caused by debris affects decibel loss in analog and digital tapes.) Although such a comparison is difficult because of the nature of digital recording, it, nevertheless, shows that digital tape requirements easily can have an order of magnitude higher than those of analog recording.

Part of the solution lies in the tape binder, which is much stronger for digital audio tapes than for many of their analog cousins. Because smaller dropouts are more serious in digital recording, a tough binder is crucial for limiting ruboff and generation of debris on the tape; an otherwise insignificant amount of debris can result in disproportionally damaging results.

The magnetic oxide coating of digital tape differs as well, being designed for extremely high packing density and to parameters required to ensure that it maintains its magnetism at very short wavelengths. Typical analog tape has a coercivity in the range of 350 oersteds, but the coercivity of a high-performance digital tape is about 700 oersteds. This means that the amount of magnetic record current required for digital tape is twice that required for typical analog tape. For this reason, digital tape does not function on analog hardware; analog recorders do not produce an adequate bias or erase current to handle the high coercivity of the digital tape. (Figure 4 shows typical coercivity ranges of analog and digital audio mastering tapes.)

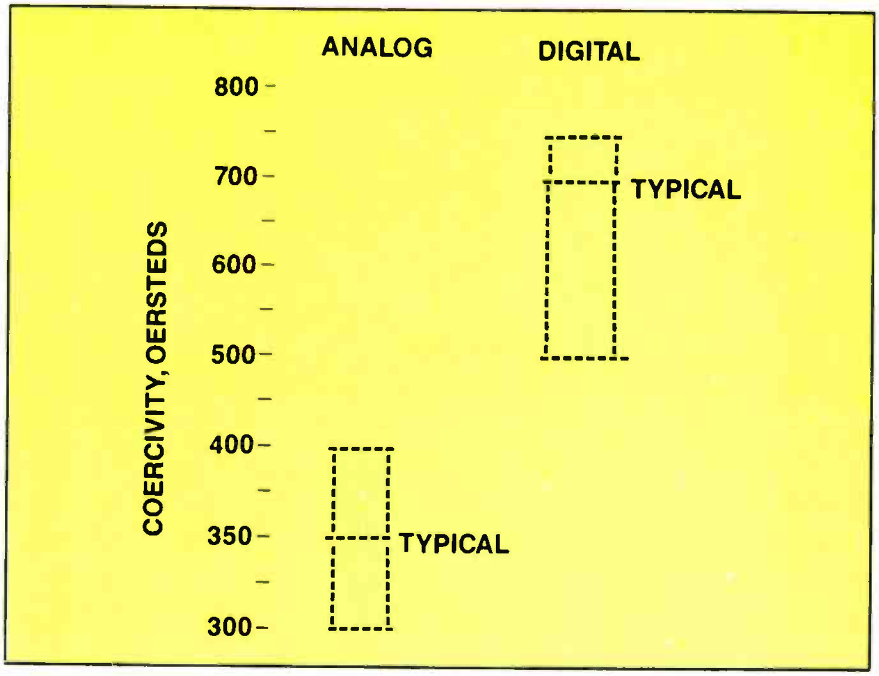

Figure 4. Coercivity ranges in audio mastering tapes, analog vs. digital.

In analog recording, the maximum output level of the tape is a function of the amount of its magnetic material. For digital audio tape, maximum output is not important. The only requirement is that the signal be strong enough to be read clearly above the noise. Hence, analog mastering tape has a thicker magnetic coating and more magnetic material (perhaps 500 or 600 microinches) compared to approximately 200 microinches on a digital audio tape.

Digital audio and videotape

In almost any discussion of the physical characteristics of digital audio tape, questions arise about its similarity to other media. To pinpoint the differences and their relationship to high-performance recording, it's helpful to look back to the beginning of digital audio technology and its roots in the development of videotape.

Digital audio recording and video recording are similar because they both store high information densities. This demand for information density defines the kind of magnetic properties and smoothness required in the tape. Although the two media share similar magnetic characteristics, such as coating weight and surface conditions, their physical properties can differ greatly. Videotape is not designed for use on a digital recorder. The pitfalls to success may not be obvious, however, because one easily can record digital data on videotape. The recording *quality* is the concern.

Video recorders and digital audio hardware have fundamental differences that are likely to become even more pronounced with future development. The trend in digital audio recording is toward stationary head devices, but video recorders use rotating heads. The physical durability and requirements of a tape designed for a rotating-head machine are quite different from those a stationary-head recorder demands. During the stop-motion function, for example, a rotating head can abrade the halted tape, sometimes literally digging away at its surface. To prevent this potential wear, videotape often is made with different binders and lubricants from those used in audio tape.

Like analog tape, videotape has a higher tolerance for flaws. If digital audio hardware can't read data, however, the system either interpolates by inserting data to approximate the lost information or mutes the output entirely (i.e., creates an audio dropout). Because of this low tolerance for flaws, digital audio tape demands near-perfection in the smoothness of its oxide coating, and it must run more cleanly than videotape. A digital audio tape manufacturer takes great pains to minimize the possibility of any flaws by subjecting digital tape to more inspection steps and cleaning operations than either conventional analog tape or videotape.

Three guidelines for selecting digital audio tape

Digital audio tape offers professionals new challenges and opportunities, as well as the greatest potential for optimal sound reproduction. Providing the highest-quality sound means selecting reliable hardware and premium digital audio tape. The guiding principles are simple.

First, for the best performance, select the appropriate tape: one designed and warranted for use with digital audio equipment. While technologies may be similar, tapes made for other systems are not tailored for the rigors of digital audio recording.

Second, for professional applications, reliability is always an issue. Select a tape with a proven record for consistency and reliability. If the tape is a new product, consider the reputation and reliability of the manufacturer.

Finally, choose a tape manufacturer that also provides support and service. Technical support helps studio professionals follow changes, enhance their services and maintain a competitive edge—a must in this high-powered and increasingly competitive industry.

Digital audio technology is constantly evolving as both hardware and recording media become increasingly fine-tuned to the demands of the broadcast, music and motion picture industries. Leading tape manufacturers invest considerable expertise in providing products for specialty applications. As a result of painstaking development, these finely tuned media fill a unique role in their designated niche, and they do so with much better results than video products. Although the building blocks of the design and development process might be similar, the performance of each product is tailored to its own application. RE/P

Digital Mastering

By Gene Shiveley

At the premastering facility, the correct equipment, qualified engineers and informed support personnel help complete your CD project quickly and economically.

Preparing your music for compact disc requires a thorough understanding of CD mastering and premastering processes. Producers and engineers have a responsibility to understand the procedures and production advantages of digital mastering to realize their projects' full sonic potential.

After the creative decisions have been made about the program material, the decision must be made as to what type of recording processes will be used. The Society of Professional Recording Services (SPARS) has established codes that tell the consumer how the CD's music has been recorded. The code is a three-letter format whereby the first letter indicates the original recording technology. The second letter indicates the mastering process, and the third letter indicates the reproduction medium. The letter A indicates analog, and D signifies digital. *DDD* and *ADD* are typical examples of the SPARS code.

Gene Shiveley is vice president of CMS Digital in Pasadena, CA.

The master 2-track

At some point in the recording processes, there is a good chance a digital 2-track will be used if the final product is slated for digital release. Because there are more than a dozen 2-track digital recorders with various formats, choosing a machine that suits your situation is important. Not all formats are accepted at CD mastering houses. At this point, a decision should be made regarding the format to be used, and the trade-offs considered, when selecting a processor/recorder that may not be compatible with established CD premastering systems.

Some of the processor/recorders available are: Sony, JVC, Mitsubishi, 3M, DAT, dbx-700 (which uses Delta Modulation rather than PCM), SoundStream, or mixing to 2-tracks of the digital multitrack. It should be noted that the CD is made from PCM tapes, and that any other format will have to be converted to PCM before the final mastering process.

Editing

These formats require some form of editing capability. If an electronic editor isn't available, DASH (Digital Audio Stationary Head), or PD (ProDigi) machines can be used. These machines are convenient because of their razor-blade-editing capabilities. F-1 and DAT formats also are available, but keep in mind the limitations of such processors. The F-1 and DAT formats are not considered professional digital formats, but they are digital and can be converted to a professional format for CD preparation.

Sonic integrity is probably the most important factor in choosing a digital processor. This is the duty of the A/D (analog-to-digital) converter and the anti-aliasing (low-pass) filter. CDs require processors with a sampling frequency of 44.1kHz. These processors have anti-aliasing filters at one-half this sample frequency, 22.05kHz. This is approaching the audio band and can affect the final sound quality. [See "Noise Modulation in Digital Audio Devices" by Richard Cabot on page 20—Ed.] The characteristics of this filter are important to the recording and reproduction of high frequencies and the stereo image. Custom anti-aliasing filters are available that can improve the A/D stage of many popular processor/recorders.

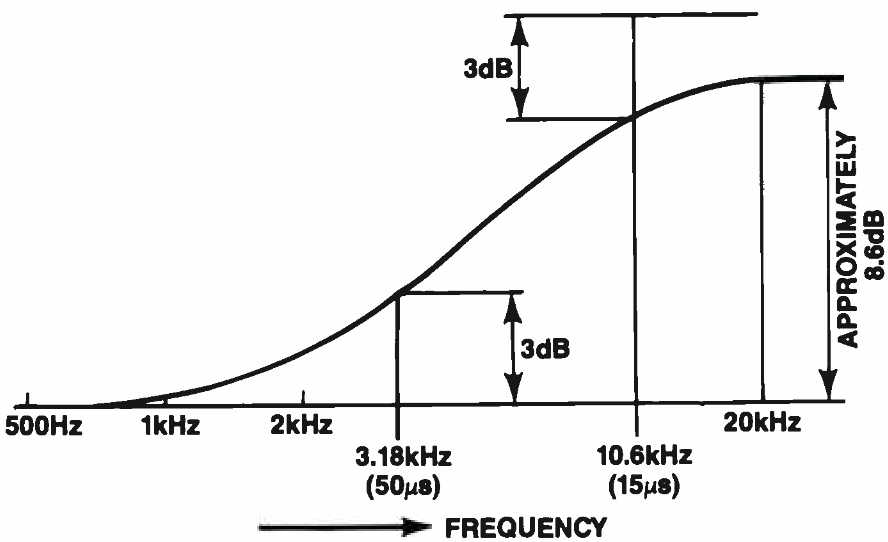

Figure 1. Pre-emphasis frequency response.

Choosing a digital recording processor can be as creative a process as choosing the proper microphone. I know of several producers and record labels that choose one particular format for the initial recording and premastering (because of the "sound" of that particular processor) and then convert to another format for the final CD master tape.

Pre-emphasis

The next creative choice is whether to use pre-emphasis. Pre-emphasis boosts the mid to higher frequencies on a continuously upward slope, with the de-emphasis circuitry mirror imaging with attenuation. (See Figure 1.)

At the point of A/D conversion, this function achieves a better S/N ratio. This process is especially applicable when recording "airy," soft or acoustical pieces. Note that it is not advisable to edit a pre-emphasized piece of music into a non-emphasized piece of music, or vice versa, because it takes the CD player two seconds to detect the pre-emphasis flag and respond. Also, using pre-emphasis and de-emphasis within the same track causes the CD player to decode improperly.

Another easy rule to remember is the greater the level, the less the distortion. However, take care not to go over the processor's maximum peak level, which is established at the A/D conversion point. When a signal exceeds the maximum peak, the processor will try to use more bits than are available. This causes the signal to clip.

During calibration for transfer, levels for non-emphasis should be set leaving approximately 15dB of headroom below 0, and an emphasized signal should have approximately 20dB headroom relative to 0. (Note: 0 level is different on all digital machines and is totally dependent on each processor's A/D/A design. The trade-off when using emphasis is less dynamic range, so examine the music carefully before recording and determine when to use this tool.)

Facilities

A CD prep or premastering facility has the staff and equipment necessary to prepare your master for the actual compact disc glass mastering, which is done at a CD manufacturing plant. CD premastering studios stay in close contact with as many CD plants as possible. With the number of these plants growing every year, this communication is a full-time job. Each plant has specifications that must be followed to make sure your finished product arrives on time and at peak quality.

At the premastering facility, the correct equipment, (such as a digital editor, digital mixer, editing decks and tape checker) qualified engineers and informed support personnel help complete your CD project quickly and economically.

What happens at a CD prep facility, and what is required to do the job correctly?

Hardware

A dedicated digital editor is required for the following reasons:

1. Synchronizing the player and the recorder is difficult. A digital editor automatically controls the synchronization of both player and recorder.

2. Because CDs are mastered to ¾-inch U-matic-type tapes running on video tape machines, the editing accuracy is limited to one video frame (1/30 second). The digital editor stores the digital audio signal around the edit point in RAM. Some of the differences between editors include: how much RAM the editor has, whether the RAM is monitored in mono or stereo and if both sides of the edit point can be

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monitored.

3. With U-matic recorders, it is impossible to search for the edit points by "rocking" the reels by hand, as it was done in the "good old analog days." To overcome this problem, approximately three seconds of data (before and after the edit point) is stored in the memory of an editor. The data can be monitored at the desired speed with the search dial. This procedure gives the "illusion" of moving the reels to select an edit point.

4. On a VTR, it is impossible to do a crossfade edit point by cutting the tape diagonally, as was done with analog tape. Crossfading is done digitally in the memory of an editor. The crossfade times range from 1ms to 99ms. (Note: When using digital editors, the term *crossfade* refers to a 45°-type razor-blade edit of up to 99ms, and not the traditional overlapping crossfade as performed with a fader.) With a digital editor, it is possible to rehearse edits easily and without fear of ruining the source tape, because the source tape is left untouched.

The editor generates a SMPTE time code as a consecutive value and records it on the master tape. When the editing is finished, you will have a perfect ¾-inch master tape onto which audio signals and time codes have been recorded and edited.

Digital EQ

CD prep facilities have specialized digital equalization systems coupled with the "hands-on" experience required for the particular needs of CDs. During the transfer, you can generally EQ the program as much as you like, because the digital medium has a greater capacity to store musical information than analog does. Look for facilities with the latest read-after-write (monitor the playback while recording, like a three-head cassette deck) digital audio editing recorders and dedicated editing recorders for the greatest flexibility, speed and accuracy when preparing the final master.

Digital audio tape

U-matic-type tape designed specifically for digital audio is available through companies such as 3M, Sony and Ampex. This tape is designed for less CRC (cycle redundancy check) or fewer errors than standard U-matic tape designed for video applications. Although a standard U-matic videotape can be used, the results can be severe CRC or digital dropouts in your master. [See "Digital Audio Tape: the Key to High-Performance Recording" on page 55—Ed.]

Dropouts occur when the tape is inferior, damaged, or has dirt or dust particles. The risk of dropouts or other problems is not worth the small difference in cost of tape specifically designed for digital applications.

There are machines that will clean and pre-check your blank tape, such as the RTI (Reference Technologies Incorporated) machine for the U-matic. You can get a reading of the condition of the tape before recording a program. A good CD prep facility will precheck all tapes. These precheck machines should not be confused with the Sony DTA1100 checker, which checks for data errors on the finished master.

Converting analog masters to digital

A CD premastering facility is also required to prepare digital masters from old analog masters. In some cases, the analog masters have lost a good deal of the information that had originally been recorded, because of tape damage or age. Sometimes the only source available is a vinyl record.

In most cases there are techniques available to help clean up old analog masters. With problem masters, it is sometimes impossible to construct a digital master suitable for CD release. When this happens, I advise the client to use an outside source for "that impossible cleanup job." One company is Sonic Solutions, which specializes in master cleanup by transferring the music into a computer with a very large memory. It carefully identifies the wanted program and removes the unwanted program. This process may sound simple, but it is very complex.

Because so many record labels are rereleasing their libraries on CD, many projects come in to the premastering studio in analog format. This material needs to be transferred to a digital format suitable for digital editing.

Time code and logging

All data on CD is formatted by frames. By definition, a frame is the smallest complete section of recognizable data on a disc. Frames consist of sync patterns and parities (These are automatically produced during CD cutting.), digital audio data and the subcode, which is created by the digital subcode editor.

Eight subcode channels are available, designated P, Q, R, S, T, U, V and W. For music, only the P and Q channels are used currently. These subcode channels contain information such as the total number of selections on the Disc, their beginning and ending points, index points within selections, emphasis on/off and end point of the disc. The remaining six channels, (R, S, T, U, V, and W) are available for other applications, such as graphics.

Most of the CD players on the market today are controlled by the Q channel, but the P channel also contains the start-flag bit to conform with older CD players. The P and Q channels contain information such as track numbers, index numbers, elapsed time within a track in minutes, seconds and frames, and elapsed time from the first music track. The data is recorded in each block. The P or Q chan-

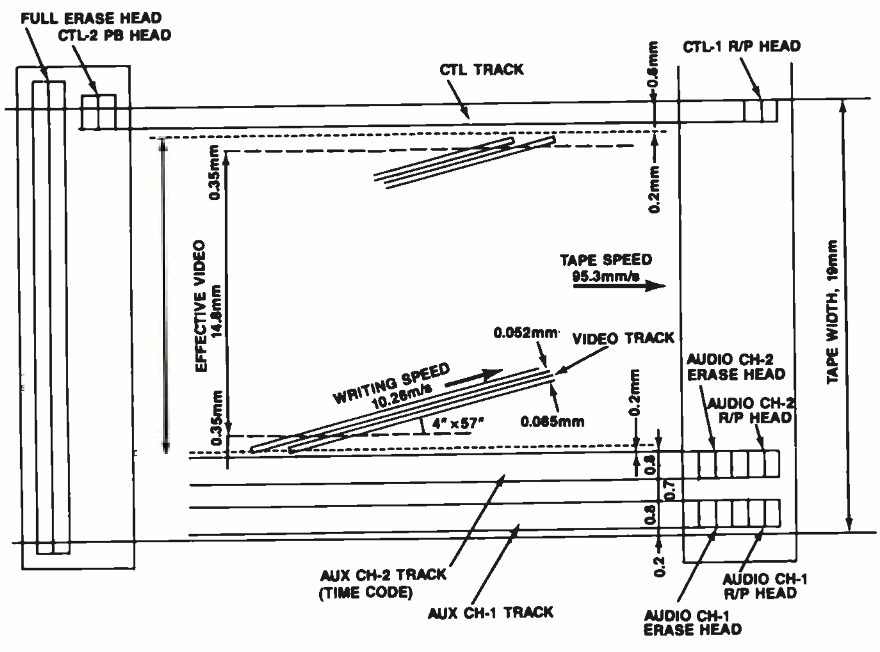

Figure 2. Tape pattern.

nel consists of a block of data made of 98 frames. (See Figure 2.)

Track 1 of the U-matic master is used to store the P and Q codes on the CD master tape. The time code and logging process is achieved using a digital audio editor.

Non-drop frame time code is recorded on analog track 2 of the ¾-inch tape. Time code should start at 0 hours, 0 minutes, 0 seconds, 0 frames (00:00:00:00) at the lead-in to the program. The amount of pre-roll time code needed, along with digital black, mute or 0 data, should be specified by the CD plant that will be cutting the glass master. Usually this time is two minutes minus one frame, meaning the first program (track) starts at 00:02:00:00.

Time code must run continuously through the tape in an ascending manner. At the end of the first piece of program material, an edit is made to record digital black. A log notation is recorded to note the exact end-time of the first track.

The usual amount of time placed between each track is two to three seconds. TM = total time of the previous selection.

BGN	Track 4	15:42:00	
END	Track 4	20:42:10	
	Pau	00:02:00	TM 05:00:10
BGN	Track 5	20:44:10	

On some musical pieces, such as a live recording, no silence exists between tracks. This is listed as a crossfade. Crossfades between tracks are commonly logged as follows:

BEG	Track 1	02:00:00	
END	Track 1	07:20:22	
	Pau X-Fade		TM 05:20:22
BEG	Track 2	07:20:23	

In production libraries or classical music pieces, it is often necessary to access points within the music. A movement within a long concerto may require its own begin index points within a track. Indexes are commonly logged as follows:

Track 2: 1812 Overture
BGN TM: 06:32:19
IND: 09:32:19
END TM: 16:35:24

In this example, the index point may indicate the entrance of the cannon. Information regarding the index point is usually listed on the label. Therefore, the engineer doesn't have to list the program name for the index on the log. According to the N.V. Philips Redbook standard, up to 99 indexes may appear within a track, and 99 tracks on a CD, giving 9,801 possible events.

At the end of all program material, there should be a minimum of one minute lead-out or black video recorded. When timing program material, take into consideration the lead-in at the head and lead-out at the end of a tape. A 60-minute tape holds approximately 57 to 58 minutes of program and still allows room for lead-in, lead-out and pause times. The ideal program length is 58 minutes. The longer tape (75 or 80 minutes) is thinner mill, typically 560 microinches vs. 750 microinches for a 60-minute cassette.

Some CD players have difficulty reading the outer tracks on CD longer than 60 minutes. Cutting the glass master for the longer-playing CDs is often treated as a custom project and is sometimes more expensive than a program that runs less than 60 minutes.

During logging, the mastering engineer

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Sample Log

Client:	GRP Records	Date:	7/28/87
Title:	Portrait	Master No.:	GRCD 1042
Artist:	Lee Ritenour	Project:	CD Master
Producer:	Lee Ritenour	Format:	JVC 900 @#44.1
Engineer:	Fred Mitchell	Time Code:	SMPTE on TRK No. 2
		Emphasis:	No

Memo: All times are absolute. There are no offsets

Lead in>	>	Blank	>>	Lead in	BGN 00:00:00 END 01:59:29		
Title 1>		ASA			BGN 02:00:00 END 07:20:22	PAU X FADE	TM 05:20:22
Title 2>		Turn the heat up			BGN 07:20:23 END 11:21:07	PAU X FADE	TM 04:00:14
Title 3>		Windmill			BGN 11:21:08 END 15:41:29	PAU X FADE	TM 04:20:21
Title 4>		White Water			BGN 15:42:00 END 20:42:10	PAU :02:00	TM 05:00:10
Title 5>		Portrait			BGN 20:44:10 END 25:11:05	PAU :02:00	TM 04:26:25
Title 6>		Grit			BGN 25:13:05 END 28:44:09	PAU X FADE	TM 03:31:04
Title 7>		Shades in the Shades			BGN 28:44:10 END 33:14:19	PAU X FADE	TM 04:30:09
Title 8>		Childrens' Games			BGN 33:14:20 END 37:36:23	PAU X FADE	TM 04:22:03
Title 9>		Runaway			BGN 37:36:24 END 42:20:09	PAU X FADE	TM 04:43:15
Title 10>		Route 17			BGN 42:20:10 END 47:00:00		TM 04:39:20
Lead out>	>	Blank	>>	Lead out	BGN 47:00:01 END 59:59:00		TM 12:58:29

needs to make careful notes regarding any uncorrectable "pops," "clicks," or unwanted noises. It is important that both the CD plant and the client be aware of such noises. Even subtle effects, such as acoustic piano pedal sounds, should be noted. A competent CD plant will call about the slightest defect. Detailed logging avoids the possibility of the CD plant rejecting glass masters and/or production units because of defects in the original master.

In addition to noting defects, the log should include complete information for artist, title, producer, format, time code location, emphasis on/off, engineer and other notes such as if blanks and tones are considered as program material or not. The log should include a contact name for information and clarification.

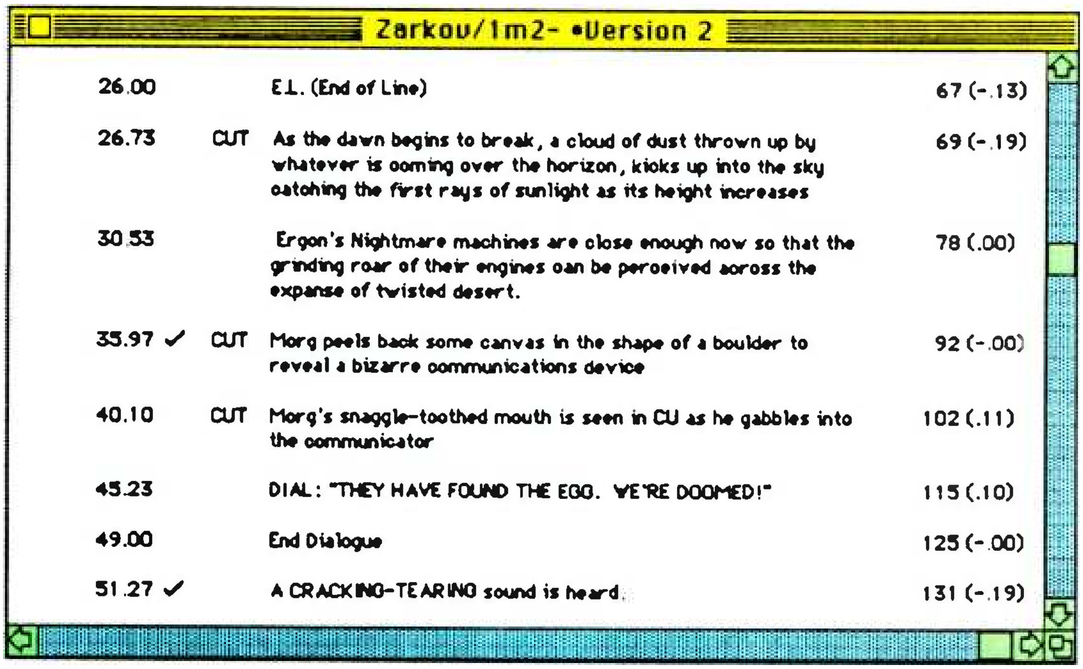
The choice of CD prep facilities affects the final outcome of the project as much as the recording facility does. Choose a facility with a good track record, one that offers the greatest selection of CD premastering systems, capabilities, and format conversions. Keep your options open to use the latest technology available.

RE/P

Software Hands-On: Cue 2.0

By Bill Cavanaugh

They say timing is everything. And to those who make their living by solving the problems of synchronized music/sound to picture, the tools and times have never been better.

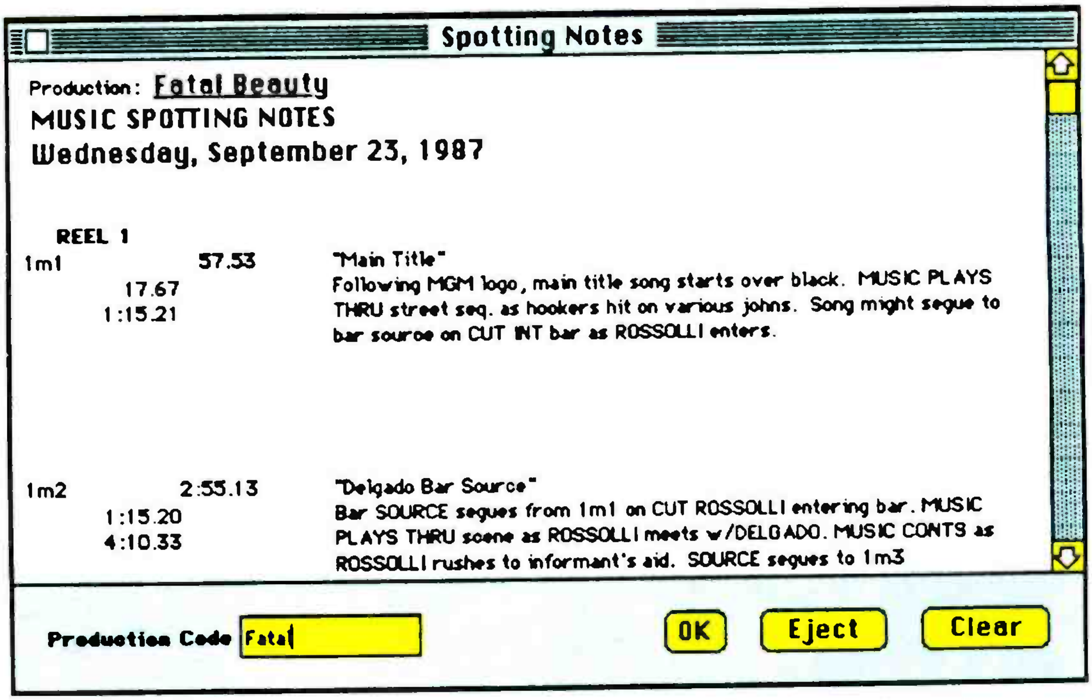


"Accelerando/Ritardando," "Click Tap" display. "Accelerando/Ritardando" makes a number of tempo decisions that speed up or slow down the click track between designated points. "Click Tap" allows the user to tap out the entire click track manually.

Opcode Systems' software program Cue 2.0 ("The Film Music System") will save the user a lot of time, to say the least. It goes a long way toward bringing together the hardware/creative resources for solving the vast number of problems that face anyone who works with sound and picture. Although it doesn't do everything (you still have to write the music and create the sounds), this comprehensive program is a large quantitative leap over the 1.5 version, other software programs or click books. It frees the user from much of the time-consuming tedium, giving more time for creativity. Written by Rick Johnson, Cue 2.0's application begins with spotting notes and continues through the audio recording process.

Bill Cavanaugh is a composer, producer, synthesist and engineer at Ciani Musica in New York.

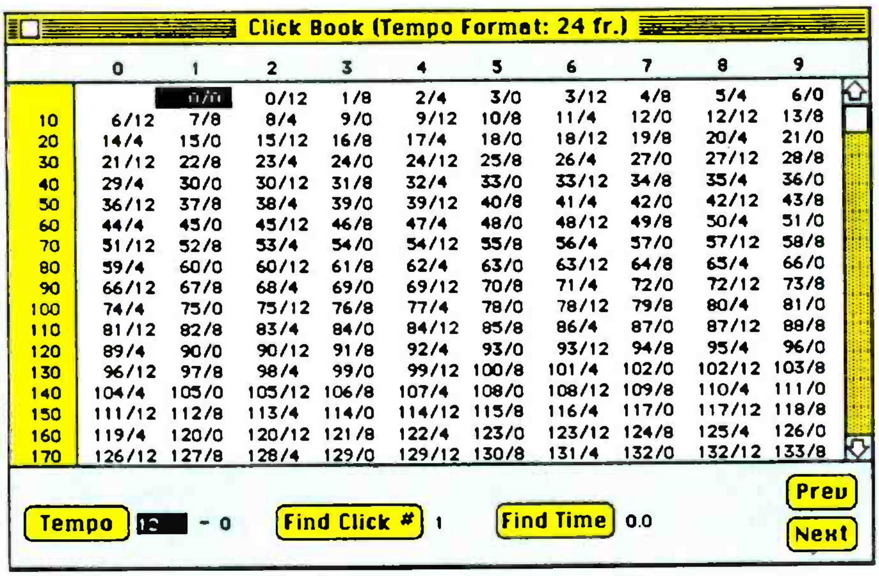
A "hands-off" description of all that this program can do is difficult because of the sheer quantity of options available. This task was put to Bob Walter, who wrote the manual as a "hands-on" tutorial. It works just as well for reference. In a continuing



"Spotting Notes." Each reel number appears in bold. The top line lists the cue number, length and title. Below that are the cue start and stop times in the current "Absolute Format" and the description from the "Spotting Notes" dialogue box. The bottom section lists the cumulative time of all specified music.

age of equipment without manuals or with unintelligible rags that are passed off as manuals, this one is a welcome tool. It's comprehensive with a dash of levity to keep your eyes from drying out, although the "Squeezing the Weasel" section may be the result of too many late nights in front of the word processor.

Getting started with the program requires a few pieces of hardware, not the least of which are a Macintosh 512k Enhanced computer and a graphics quality printer. Although Opcode notes that a Mac 512 is the minimum required, please note up front, however, that this program has problems in pre-Enhanced Mac 512s (the ones with the older 64k ROMs and MFS [Macintosh Filing System]). It does run quite glitch-free on the Enhanced 512s or the Mac Plus (the HFS [Heirarchical Filing System] Macs with 128k ROMs). According to Opcode, the program is compatible for the SE and the Mac II, although I did not test those.


Click Book (Tempo Format: 24 fr.)

	0	1	2	3	4	5	6	7	8	9
		0/0	0/12	1/8	2/4	3/0	3/12	4/8	5/4	6/0
10	6/12	7/8	8/4	9/0	9/12	10/8	11/4	12/0	12/12	13/8
20	14/4	15/0	15/12	16/8	17/4	18/0	18/12	19/8	20/4	21/0
30	21/12	22/8	23/4	24/0	24/12	25/8	26/4	27/0	27/12	28/8
40	29/4	30/0	30/12	31/8	32/4	33/0	33/12	34/8	35/4	36/0
50	36/12	37/8	38/4	39/0	39/12	40/8	41/4	42/0	42/12	43/8
60	44/4	45/0	45/12	46/8	47/4	48/0	48/12	49/8	50/4	51/0
70	51/12	52/8	53/4	54/0	54/12	55/8	56/4	57/0	57/12	58/8
80	59/4	60/0	60/12	61/8	62/4	63/0	63/12	64/8	65/4	66/0
90	66/12	67/8	68/4	69/0	69/12	70/8	71/4	72/0	72/12	73/8
100	74/4	75/0	75/12	76/8	77/4	78/0	78/12	79/8	80/4	81/0
110	81/12	82/8	83/4	84/0	84/12	85/8	86/4	87/0	87/12	88/8
120	89/4	90/0	90/12	91/8	92/4	93/0	93/12	94/8	95/4	96/0
130	96/12	97/8	98/4	99/0	99/12	100/8	101/4	102/0	102/12	103/8
140	104/4	105/0	105/12	106/8	107/4	108/0	108/12	109/8	110/4	111/0
150	111/12	112/8	113/4	114/0	114/12	115/8	116/4	117/0	117/12	118/8
160	119/4	120/0	120/12	121/8	122/4	123/0	123/12	124/8	125/4	126/0
170	126/12	127/8	128/4	129/0	129/12	130/8	131/4	132/0	132/12	133/8

Tempo 12 - 0 Find Click # 1 Find Time 0.0 Prev Next

The "Click Book." Although one of Cue's main purposes is eliminating the need for a traditional click book, sometimes it's useful to consult one.

What it does

Fundamentally, Cue 2.0 inputs timing data, then calculates and manipulates that data and outputs several types of documents that are highly useful in post-production audio/music. The document types are hard, soft and active.

The timing inputs essentially consist of time in hours, minutes, seconds and sub-seconds as expressed in various visual and musical formats. More specifically, any visual format means 35mm or 16mm film footage/frames; basic hours:minutes:seconds; and SMPTE time code in 24-, 25- and 30-frames-per-second (fps) formats. The 30fps SMPTE format is further broken down into B&W non-drop, drop frame (29.97fps) and the obscure "Color" mode which displays in 30fps but actually runs at 29.97fps. The musical format is in measures and/or beats with tempos expressed in beats-per-minute or frames/fractions-per-click. The timing input format is chosen in the "Set Time Formats" dialogue box. (A dialogue box is a temporary window that requests information. As soon as the information is input, the window closes automatically, and the screen reverts to the same place in the file it left.)

As the timing format is chosen, times can be entered in either absolute or relative time. Absolute time is the actual SMPTE time code location on a tape where the cue begins and ends, and relative time is the duration the cue itself.

Although any given project most likely will be in no more than one or two formats (e.g., film that is transferred to videotape and/or back), Cue 2.0 can input data in any of the above formats and translate it into the set format. For example, if Cue 2.0 was formatted for SMPTE 30 non-drop, you could enter information in measures/beats or feet/frames, and Cue 2.0 could translate this information to display it in SMPTE 30 non-drop. If an updated version of your cue appears with changes that range from a few frames added or removed to whole scenes changed, Cue 2.0 can adapt your previous work with very little effort on your part. Or, if the timings are all the same, but the timecode numbers have been changed, the program allows for global adaptation.

SMPTE time code entries, as well as any format entries, are made with the "Input" window. SMPTE numbers can be typed in or grabbed "on the fly" when Cue 2.0 is locked to moving videotape or extracted from a freeze frame using Vertical Interval Time Code (VITC). Using VITC requires encoding of the videotape with VITC and a VITC-to-LTC converter. By offering a SMPTE clock adjust feature that offsets the entry numbers, the program compensates for various VTRs' placements of the noise bar when a frame is frozen on screen.

The timing information is displayed in a window called "Cue Sheet." This primary screen for the program is displayed in a "what-you-see-is-what-you-get" format. The window can be printed as one of several hard documents available. The page, production name, cue title and tempo information are displayed across the top, with the time of the cue start just below. Under that is the optional preface for cue description and the time display for the beginning of any warning clicks that were programmed in. Further down are six vertical columns with the timing specifics for the given cue.

From left to right, the first two are timing information (e.g., your choice of SMPTE numbers, real time or film footage) with either column being in absolute or relative times. The next two columns are displays for "Key Hit" information and "Cut" indication. In the center of the page is Column Five for scene description, dialogue notation, displays of meter and/or tempo changes and the total time of your cue. The sixth column displays click information in either bar/beat or click format. This column shows, at the given tempo, the position of your clicks/beats relative to your first two timing columns and can display the beat divisions of your sequencer, e.g., x/a480 per quarter note for Mark of the Unicorn's Performer.

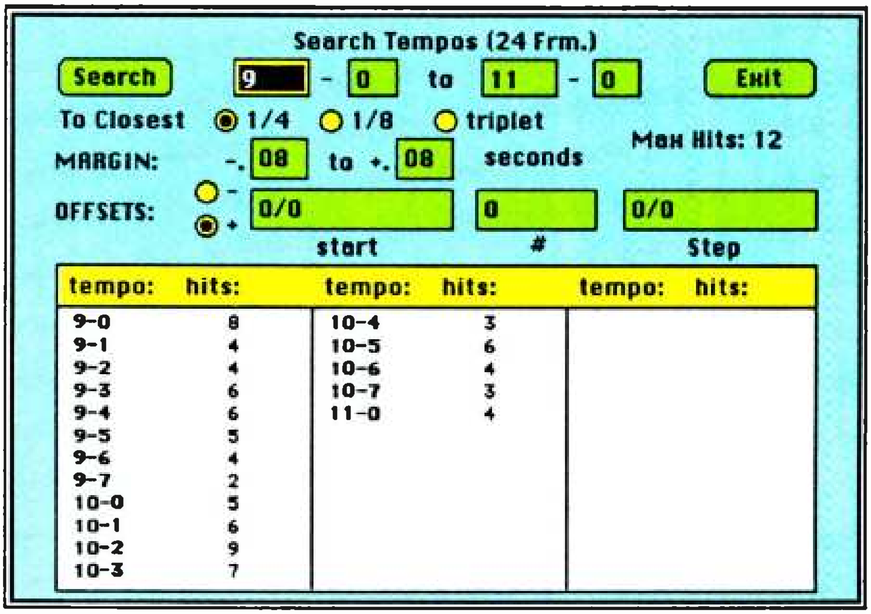
Once all of your timing data has been entered, Cue 2.0 allows you to search for a tempo that will hit all points in the cue that you have designated as "Key Hits." The tempo options, which are set in the "Set Tempo Formats" dialogue box, are clicks in 24, 25, 29.97, 30 frames per second and beats per minute. The standard 24fps click format further subdivides eighth notes into fractional tempos of eightieths. The search can be conducted between a range of tempos to the nearest quarter, eighth or triplet note with a variable margin of frames before and after the hit. The search result yields a display of tempo options and the number of attendant hits. A zoom feature allows the user to zoom in on a selected tempo and see exactly where the key hits are falling in relation to the selected tempo and the exact click. The displayed click is subdivided into hundredths of a click. After a suitable tempo has been found, it is entered in the "Set Tempo" dialog box.

If a simple tempo search doesn't yield enough "key-hitting," then the entire cue can be offset. For example, (if the hit tolerance is two frames, and every hit is hitting within one frame, but one hit is missing by three frames, the entire cue could be offset by a single frame, which would bring all hits within the two-frame tolerance.

If a tempo/offset combination cannot hit all of the hits when using a single tempo, an accelerando or ritardando can be used for "Retiming" between those hits that need to be brought into line. The program allows you to calculate the "Retimings" between a starting click number and an ending time, and takes into consideration the creative options of a desired ending tempo or ending click number. The retiming feature generates a custom click that varies in ways that most "live" musicians can play to easily. Tempo changes are immediately displayed in the "Cue Sheet" window.

Meters are also variable and multiple within a cue. Against a base meter, changes can be inserted between designated points. However, as of this release, the click doesn't automatically follow changes in the denominator. For example, when a meter changes from base of 4/4 to 7/8, the click stays in quarters, giving you seven quarter notes. Solving this problem requires doubling the tempo for the duration of the meter change. With x/16 meter against a x/4 base, you must double the tempo twice for the entire inset. Although not a major or unworkable problem, this anomaly is scheduled to be "corrected" in the next release. Meter changes are immediately displayed in the "Cue Sheet" window.

The "Cue Sheet" window is only one soft, printable window. Others include: "Custom Score" paper (printed vertically



"Hits and Misses" (the "zoom" feature). 9-0 and 10-2 are promising because they are respectively hitting eight and nine out of 12 "Key Hit" points within two frames of the nearest quarter note.

or horizontally), a "Master Cue List" (the so-called "Bible"), an ASCAP/BMI-type "Performing Rights" document, a "Spotting Notes Sheet" and a "Custom Title Page" that allows for the import of graphics from programs like "MacPaint."

All of these pages share their data, which saves time and reduces the risk of error that can occur in any duplication of effort. More specifically, most information is entered only once, and this data is displayed on every page that uses it, in both soft and hard copy. For example, if a music editor or composer pulls down the "Operations" menu for the "Set Cue Info" dialogue box and enters the production title, cue number, cue title, composer's name, publisher's name and affiliation, orchestrator's name and whether the cue has yet been written, orchestrated, copied or recorded, he does this only once. The timing information entered in the "Cue Sheet" window is automatically displayed in the "Clicks Window," which, in turn, is printed over each music system on the custom score paper. This "Clicks Window" information is the musical display of the "Cue Sheet" window's data. The combination of the "Cue Sheet's" timing data and the "Clicks Window" data is finally activated in the document called "Playback."

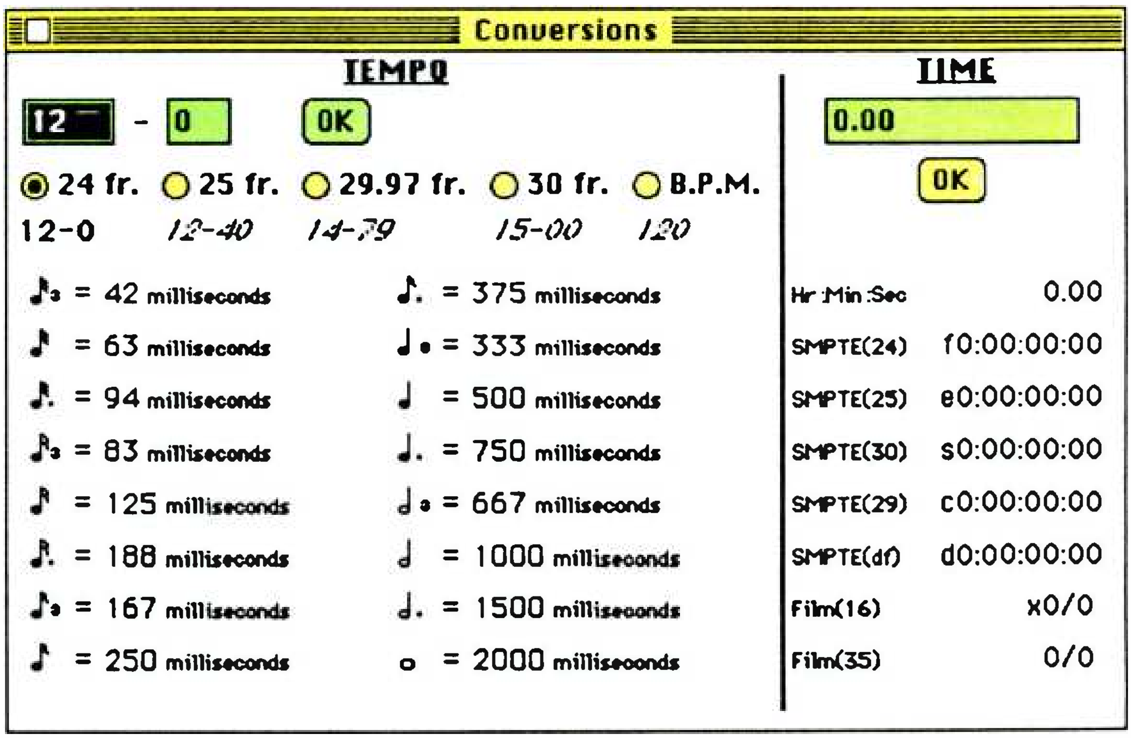
"Playback" is a window from the "Reelworld" pulldown menu. It is the feature of Cue 2.0 that communicates with the outside world. It plays back the custom click track either from the Mac internally or locked to picture via MIDI timecode. The screen is an animated graphic window that displays the current position in SMPTE time, time for the next punch, current bar/beat, frame format, lock indicator, an analog stopwatch with a relative minute sweep, and streamers and punches in sync to picture. These streamers and punches can be superimposed on your video screen using something like the Video Streamer made by CB Electronics of England. Or, because Cue 2.0 sends out MIDI start messages at the beginning of each streamer, a triggered relay closure device can be used with a MIDI-controlled relay box such as J.L. Cooper's MIDI Mute. Streamers can be varied in length between two and four seconds. All of this should help the live conductor to conduct in the "Free-timing" mode. To facilitate the "Free-timing" mode, the click can be programmed to turn on and off at necessary places in the cue sequence. Cue 2.0's click feature generates a MIDI clock for external control of sequencers and MIDI notes for an external custom click sound. Cue 2.0 also receives MIDI start messages.

Although Cue 2.0 is not a sequencer in the common and current understanding of the term, it can sequence, in sync, up to 40 "MIDI Events." These events are entered via the "Input Window" and are listed in the "Cue Sheet" window and the "MIDI Event List" window.

In the "MIDI Event" window, each event is listed with its MIDI note number, description, MIDI channel, up to five separate MIDI notes for each event, velocity level for those notes, and a column for duration. This information is ideal for cuing sound effects at a precise moment via a MIDIed sampler or an event controller that can see MIDI. The MIDI notes can be typed in or entered singly or as chords via the sampler's keyboard to ensure the proper velocities for each of the notes. This helps shape those simultaneous, multiple key-based sounds.

Additional hardware

To use Cue 2.0's MIDI Time Code implementation, a Mac-compatible MIDI interface and a SMPTE-to-MIDI Time Code converter are required. Several such devices are currently available. Relative to Cue 2.0, the SMPTE-to-MIDI Time Code converter that is easiest to use is Opcode's own



"Conversions" window lets the user convert a tempo or time into all possible formats without having to change the "Cue Sheet."

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a software controller for
ime Code Machine, the Gar-
erbeat and Time Commander,
Southworth JamBox/4+ SMPTE
Code converters.

If you are a sequencer-based composer and can't afford two computers, fear not. All timing information and tempo mapping can be transferred to the Southworth JamBox/4+, the Roland SBX-80, or to the Garfield Time Commander. The timing and tempo information can also be saved under a different filename as a version 0.03 MIDI file, which will be transferable as a tempo map to sequencers that can read this version of the MIDI file.

Additional features

Another feature of Cue 2.0 is a custom "tapped" click, which is literally tapped out on the Mac keyboard and either averaged over 17 taps or taken in total from a cue that is locked to picture.

This software supports the use of modems, which speeds up communication between its users. This can be particularly convenient when transmitting data from coast to coast.

Normal Mac file-saving techniques are enhanced here with an optional auto-backup feature and files that save three versions of the same cue, each differing in tempos, meter changes, offsets and warning clicks.

The program includes a "Conversions Window," which displays note durations from a whole note down to a 32nd note triplet, at any given tempo, in any tempo format. The window also displays exact times in all formats for a given time entry.

Conclusion

Although many approaches to post-production timings exist, Cue 2.0 is designed to work in any familiar format; with any working method. Even if your work habits dictate no change from the click book and click paper technique, the program offers the old-timers what they need for their familiar tried and truly tedious method. There is a click book display.

Differing working methods aside, Cue 2.0 should be a welcome weapon in the arsenal of all music editors, sound editors, composers, arrangers and especially those who do all of the above on any given job. It will save the user more than time.

Cue 2.0 is well written; it works; and those who buy it won't feel that they are also beta-testing it for Opcode.

RE/P

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Circle (30) on Rapid Facts Card

Repairing Surface-
PC Boards

By Christopher Fenton

Don't attempt to remove that surface-mount device without having the proper tools and knowing the right procedures.

Today's sophisticated professional audio equipment often relies on surface-mount components. These devices provide extremely compact layout, allowing complex circuits to be entirely self-contained on a single PC board. Although this is a great advantage to the user, repair of surface-mount components is complex and demands special skills and equipment. Attempting to remove these devices with the "ol' soldering iron and needle-nose pliers" can be an expensive proposition.

Unlike leaded components, surface-mount components have short, winged leads for solder attachment to the printed circuit board. So, the components are spaced closer together. Increased circuit density and the smaller size of surface-mount components may reduce the overall size of a PC board by as much as one-third.

Because the devices rely on closely spaced surface-mount multiple leads, rather than through-hole leads, removal can be difficult. Devices with through-hole leads can be removed by desoldering the leads one at a time. However, surface-mount leads cannot be desoldered in the same manner. Surface-mount devices can be removed only when all the solder connections between the component's connections and the printed circuit board have been reflowed—at once.

Conversely, to reattach a new device, all the new solder connections must be made at the same time. With some surface-mount package devices requiring as many as 75 solder connections, the problem becomes how to solder all of them simultaneously. It sounds impossible, but the process can be completed quickly and efficiently.

Currently there are two methods of removing and replacing a failed surface-mount device: conductive heating and convection heating. Conductive heating uses a direct-contact approach to remove and to resolder a surface-mount component. Convection heating uses hot air to remove and resolder a component.

Conductive heating

Conductive heating is identical to using a soldering iron to remove and resolder components, except that all component connections are soldered at t
After a failed device has be
hand-held heated probe is use
and resolder it.

The heated probe has two
nickel tips that are designed to con
reflow the solder joints on all fou
of the component simultaneously.
heating tips are mounted at the end
two short metal tubes containing heat
elements. The tubes are connected to
hinged handle similar to a pair of tongs.
This tong action allows the operator to align the tips of the heated probe to all four sides of the component package.

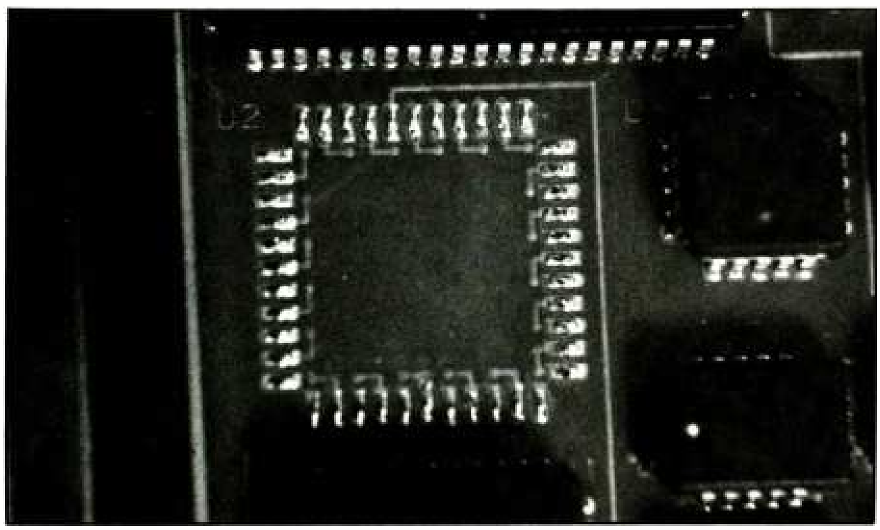
Heated probes are available in sizes ranging from 0.185 inches to 1.5 inches in square or rectangular shapes. Anyone replacing surface-mount components should have several heated probes of various sizes and tip configurations.

Fast, effective component removal and replacement depends on the amount of heat transferred between the component's solder joints, substrate material and the heat source. The key is to transfer the greatest heat from the heat source in the shortest time to reflow solder connections quickly and efficiently.

Inspect the solder fillets at the component's contact tabs and footprint junction on the board. A solder fillet is the concave junction formed by the solder between the footprint pads and the component contact tabs. If the solder fillets are small or are devoid of solder, add solder to the connection. Solder paste used sparingly is usually sufficient. Then add solder to the solder fillet so that when it touches the heated probe, the solder joint will conduct as much heat as possible to the rest of the solder joint. This heat transfer immediately floods the portion of the solder fillet that is bonded to the footprint pad and results in a fast, efficient exchange of heat.

With some components, especially those with lead contacts narrower than the footprint pad, the solder fillet helps spread the heat to the outer edges of the solder joint. Efficient heat transfer is im-

Christopher Fenton is a consultant for the Western Reserve Tool and Machine Co.



The solder pads shown here have been properly prepared for the replacement of the IC. Note the filled solder pads and lack of solder bridges.

perative. Otherwise, you run the risk of ruining an expensive component by lingering too long on the component with the heated probe.

Circuit boards with ceramic substrates and boards with large heat sinks should be preheated before component removal. Preheating reduces the heat-draining effects of heat sinks and also reduces localized thermal expansion that may crack a ceramic substrate.

Removing the component

Before removing a surface-mount component, first secure the board in a grounded holder or place it on a grounded surface. The heated probe is plugged into a controller unit, which enables the technician to vary the probe's temperature for different types of component packages and applications.

Liquid flux should be applied to all solder points and around the component. The flux not only improves the solder fillets' heat transfer characteristics, but also provides a clean surface on the board's footprint pads for soldering a new component.

Position the heated probe tips around the edges of the failed component and parallel with the board's substrate. Firmly grasp the component with the probe tips so that they contact all four sides of the failed device simultaneously. After the solder has reflowed on all joints, raise the heated probe with the failed component held between the probe tips. Because the inside tip dimensions are smaller than the outside component dimensions, the heated probe can be used to pick up the failed component.

To remove or replace a chip capacitor, resistor or other passive component, a different type of heated probe, with parallel tips, is used. This type of heated probe is available with various length tips for removing and resoldering components that have parallel multiple-lead contacts. The basic procedure for removing and resoldering these components is the same as previously described.

Installing the replacement component

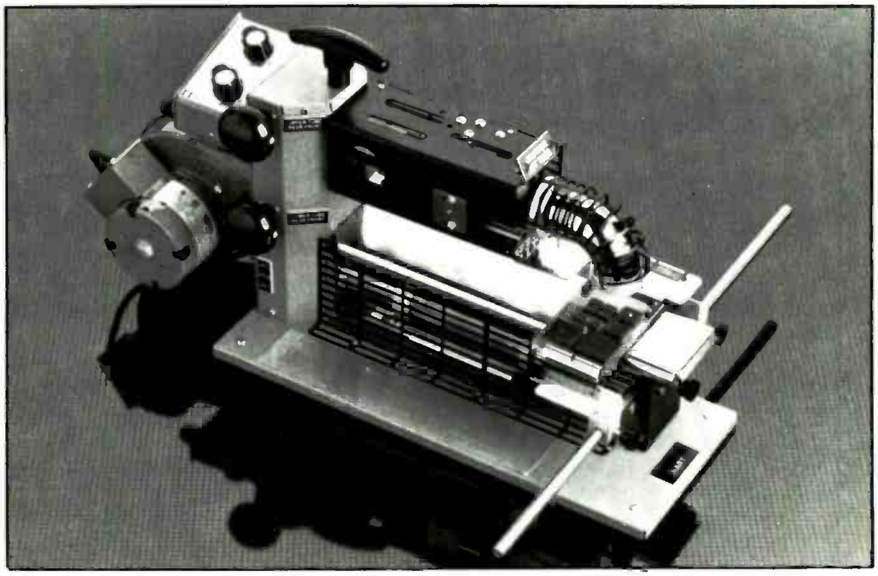
Before resoldering a new component, it's important to pre-tin the component's contacts. This ensures good solder flow between the component and substrate. Pretinning also replaces the solder lost when the defective device is removed from the printed circuit board.

Examine the board footprint pad for a clean solder surface and for consistently sized solder beads. Any solder bridges that may have formed during the removal procedure should be removed before resoldering a new component. If more solder is required on the footprint pads, solder paste should be applied sparingly. Too much solder paste may result in unwanted solder bridges. Too little may result in an open or weak connection.

Place the new component in position and align it by hand with the board's footprint pads. Position the heated probe tips against the component's lead contacts and push lightly against the substrate surface. When the solder has reflowed evenly on all connections, remove the tips of the heated probe and allow the surface tension of the molten solder to pull the component into final alignment with the solder fillet.

The heated probe is fast and effective. It is also the simplest way to remove a surface-mount component. Using the heated probe requires little training and removing a defective component takes an average of four seconds. Resoldering a new component takes about the same amount of time. The process does, however, require a little more dexterity than the removal procedure.

Heated probes have a limitation. Their use is limited to circuit boards with low component densities. It is impossible to operate the heated probe if you don't have room to open and close the probe's tips. For boards with high component densities,



A hot-air repair terminal (HART) simplifies removal of surface-mounted ICs while protecting the board and components from heat-stress fractures.

The control station, shown in the background, allows the probe tip temperature to be carefully controlled to prevent damage to the ICs.

there is another method of removing and replacing surface-mount components: convection heating.

Convection heating

Convection heating uses hot air to remove and replace a failed component. Using a hot-air repair terminal (HART), the convection method preheats the printed circuit board before reflowing the solder connections for component removal or resoldering.

The convection system uses low-pressure hot air directed toward the failed component and the area surrounding the component to reflow all solder connections simultaneously. With this process, there is no direct contact between the heat source, and component and board surface.

The printed circuit board is clamped into a platform between two air tubes that are connected to a blower. The top tube provides hot air to reflow the solder connections and the bottom tube directs a flow of cooler air onto the bottom surface of the printed circuit board. The cooler air prevents the circuit board from overheating, which could ruin components. The desired air temperature is set on the HART, or a temperature-indicating liquid is applied to the top of the failed component.

A temperature-indicating liquid is a milky-colored fluid that becomes clear when it is heated to within 1% of its given temperature. This is the most reliable indicator of solder reflow.

After the circuit board has been secured in the machine's platform, the platform is adjusted so the component is directly in the flow of hot air from the upper tube. Components adjacent to the failed component are not susceptible to degradation or damage because the removal temperature used does not exceed the reflow temperature used during the manufacturing process.

After the machine has warmed up and after the temperature-indicating fluid has become clear, the solder fillets are molten and the failed component can be removed with a pair of tweezers. Circuit boards with large heat sinks may require more time in the airflow to compensate for the thermal-draining effects of heat sinks. The average component removal time is approximately 25 seconds.

Before reattaching a new component, first examine the board's footprint for evenly sized solder beads and for solder bridges. If more solder is required, solder paste can be used as described previously. Solder bridges should be removed at this point.

As with conductive heating, pre-tin the component leads to ensure good solder flow between the leads and the board's footprint pads. Components can be pretinned by applying solder paste to the component's contacts and then placing them upside down on the platform directly in the hot airflow.

Liquid flux is then applied to the board's footprint pads. Next, the component is placed in the liquid flux by hand and generally aligned. When the component is soldered, the surface tension of the solder will float the component package, pulling the component into final alignment with the footprint pad.

When the indicating liquid has become clear, the resoldering process is complete. If the component fails to line up correctly, it may be repositioned with a pair of tweezers while the solder joints are still molten.

With the increasing use of surface-mount components in audio and computer equipment, an engineer must know how to remove and resolder surface-mount components effectively and in the most efficient way possible. The devices and procedures described here may allow you to repair equipment that you would have returned to the factory for service. If so, one or two repairs can more than cover the equipment cost. RE/P

Surface-mount glossary

With the ever-changing state of electronics technology, it helps to know exactly what you're talking about. The following is a basic primer on some of the terms associated with surface-mount technology.

- *Contacts: The wing-shaped leads protruding from the component package body, which are electrically conductive and are used for solder attachment.*
- *DIP (dual in-package): An integrated circuit that has two rows of pins for through-hole mounting along the two longest parallel sides of the component.*
- *Fillet: A junction formed by solder between the board's footprint pads and the component's contacts.*
- *Footprint: The group of board contacts corresponding to the leads of the component package to which it is soldered.*
- *LCCC (leadless ceramic chip carrier): A component package containing an IC, which is mounted to a printed circuit board as a surface-mount component. Made from a ceramic material, it is hermetically sealed and can withstand high temperatures. Instead of wire leads, it has contact tabs around the perimeter of the component package for solder attachment.*
- *Pads (lead contacts): The individual contacts of a printed circuit board's footprint.*
- *Pre-tinning: Applying solder to the component's contacts or tabs and to the board's footprint pads to improve solderability characteristics before soldering a new component in place.*
- *SMC (surface-mounted component): A component that is mounted to the surface of a printed circuit board's substrate, instead of being soldered through plated holes, as in a standard printed circuit board.*
- *SOIC (small outline integrated circuit): A component package that houses an integrated circuit chip for surface-mounting: it is approximately one-third smaller than conventional integrated circuit packages.*
- *Solderability: The capability of solder to reflow and wet the circuit board footprint pads and the component leads during component removal and resoldering.*
- *Solder reflow: The point at which solder paste applied to both the component contacts and the board substrate footprint pads melts or reflows. The solder from both the component and substrate contacts reflows to form or to break the solder fillets depending on the repair operation.*
- *Substrate: The material that forms the base of the printed circuit board, usually made from a ceramic material or a fiberglass-epoxy composite.*
- *TCE (thermal coefficient of expansion): The rate at which a component and substrate expand when exposed to heat, expressed in parts per million per Celsius degree (ppm/°C). The TCE of the component must be matched to the TCE of the substrate to minimize thermal stress from warping or cracking the printed circuit board.*
- *Through-hole board: A printed circuit board that has plated holes through its substrate in which wire leads are inserted and then soldered to the other side. This is currently the industry standard.*

Facility Profile: Pegasus Studios, Ltd.

By Gregory A. DeTogne

The primary concern was to create an environment that would be ideal, regardless of technology.

The southeast side of Pegasus Studios.

Studio A showing the E IV/D monitors and SSL console.

As a concept, Pegasus Studios began on a humble note. The brain child of Butch Trucks, a former drummer with the Allman Brothers Band, it was to be a small, one-room, but well-equipped studio, which would provide him with a chance to settle down. Like most plans, however, that's not how it turned out. In place of the modest room that Trucks envisioned, now stands a state-of-the-art, 11,000-square-foot facility housing everything needed to handle record production, film and video scoring, and even video post-production.

Pegasus is located just outside of Tallahassee, FL. About $3.5 million was spent to complete the project, which included construction costs of $1.3 million for the structure itself. According to Trucks, his initial plans for a small studio changed as part of a marketing strategy because of the increase in film and video projects shot in Florida, now the third-largest center in the nation for such production.

"The more I got into planning the studio, the more I realized that Florida was going to continue playing an important role in film and television production," Trucks recalls. "And, as I looked around the state, I didn't see anyone else with facilities adequate to handle the scoring work. Producers were coming down here to shoot, and then going back to L.A. or New York to take care of post-production. At that point, I decided to build a studio that would meet the needs of the film and video industries, as well as the record business."

The site chosen for the facility was in Gadsden Station, a 550-acre business and entertainment industry park being developed by the Talquin Development Company. Pegasus Studios was Gadsden's first

Gregory DeTogne is a publicist/free-lance technical writer in the Chicago area. He covers recording, video, music, and entertainment industries.

tenant, and the building's construction was completed in early 1988.

Studio design

The studio's design was first put to paper by George Augspurger. Overall, the designers went to extreme lengths to provide the best recording environment possible. "When we showed the plans to some other studio owners in New York, they cussed a bit and told us we were being obscene," Trucks says. "They thought we'd gone too far in obtaining isolation between the rooms, making sure our grounding was correct and in setting up our electrical system, among other things. We did a lot of things most studios would like to do but can't because it's just too expensive. Here in Florida, though, it was within our grasp, so we did it."

Nestled among the rolling hills of Florida's Panhandle, Pegasus Studios was intentionally built in a sparsely populated area conducive to creativity. In all, the area is quiet and free from the usual big-city diversions that have a tendency to distract clients.

A two-studio complex by definition, Pegasus consists of a main studio and a post-production room. Studio A is the larger and features a 2,400-square-foot main floor capable of seating a 115-piece symphony orchestra beneath the studio's 22-foot ceiling. Studio B is primarily a post-production room, outfitted with dialogue replacement, sound effects and video scoring equipment. Although, from the outside, the building appears to be only one story, in reality, it is a two-story, concrete-block, poured-in-place structure built into a steep hill.

Studio B showing the Tannoy FSM monitors and SSL console.

Studio B showing one of two outboard racks.

The original design was developed by Augspurger in 1981, but construction by the development company didn't begin until February 1987. To update the blueprints to current acoustic standards, Trucks hired Dave Engelke to work with architect Rolando Gutierez of Tallahassee, FL-based Clemons, Rutherford & Associates. Engelke was given the role of technical adviser and on-site engineer, while Augspurger supplied additional advice from L.A. during the final construction phases.

Starting from the outside, the first layer of wall material is 12 inches thick and made from steel-reinforced concrete poured in place to form a monolithic slab from the ground up. Talquin poured the walls and footers first, and then, from the same material, poured the bottom slab on which the studio floats. Inside the outer walls, the studio itself begins to form. Once again, to improve isolation, conventional 2″×4″ wall framing was scrapped in favor of a 2″×10″ design. Throughout the structure, the 10-inch gap in the interior walls was filled with fiberglass insulation and given a three-layer-thick coating of sealed gypsum board on both sides.

The interior walls are floated off of the bottom slab with half an inch of neoprene rubber applied to each wall's top, bottom and sides. In the actual interior of the studio, where the different rooms are partitioned, walls of the same design are sandwiched together with a 1-inch, airtight gap between them, creating divisions that are 23 inches thick with 12 layers of wallboard.

Aesthetically, this massive construction was carefully concealed so it would look like a typical wall in a typical home. And, although sensitive audio areas are all located in the portion of the structure that is built into the side of the hill, the feeling of being inside a bunker is completely non-existent.

Architecturally, the building is clean and modern, with many curving lines. The stark gray exterior is a finished concrete surface.

The interior has been tastefully decorated with German hardware; an entrance area is adorned with Greek marble and a huge skylight; and bathrooms feature European shower and sink fixtures.

During construction, consideration was given to the notion that digital recording technology would be the mainstay of the studio's business for years to come. The primary concern was to create an environment that would be ideal, regardless of technology.

Pegasus' technical adviser Dave Engelke says, "From my observations, a basic problem with many older rooms is that they were built to a different set of standards 10 or 20 years ago. As a result, they sometimes have rattling air conditioning systems, creaky floors, buzzes in the lights and water pipes that are too close to the recording environment. In the days of analog, these oversights weren't as noticeable because the tape hiss compensated for a lot of the noise. However, put a digital machine in the same room, and, suddenly, the noise floor drops, you have zero tape hiss, and the dynamic range jumps dramatically. Then, these 'little' things make a big difference. With this scenario in mind, we built this facility to be completely quiet in terms of isolation."

Because of its size, Studio A has a definitive "live" sound. If needed, the room can be deadened by using movable absorption panels, which Augspurger supplied. Engelke explains, "Out intention with the absorptive panels was to be able to change the acoustics of the room according to client needs."

Studio A's floor floats on ⅛-inch closed-cell foam and is made of ¾-inch flatwood glued together into a single-piece unit. A full-size motion-picture screen covers one wall for film scoring, and a stage has been built in for shooting video and film. To complete Studio A's production abilities, theatrical lighting is available with enough dimming capability to deal with 192 circuits. A fully automated Strand light mixing palette can be used to change the room's light gray walls to any color as needed.

Measuring 35 feet from front to back, Studio A's control room was specifically laid out in a geometric shape that ensured proper presentation of the custom E IV/D monitors built by Dave Engelke's company, E-Systems Technologies. (See "Custom Monitors at Pegasus Studios" for a detailed look at these custom, four-way, quad-amped monitors). Special soffits enable the monitors to be hung much lower than is ordinarily possible. This capability facilitates the use of a special panoramic, 2-inch-thick, laminated plate-glass window and brings the monitors into an optimum position. Positioning of these monitors was deemed so critical that Engelke took even greater pains to make sure the walls and ceiling also were constructed in a fashion that would ensure their ideal performance.

At the rear of the room, he placed custom E-Systems Technologies diffusers that are only 8 inches thick. (Standard diffusers are 16 inches thick.) Formed out of high-density Masonite and hardwood, the diffuser panels are finished in polished black lacquer and serve as disguised doors in some cases, providing access to the console's main computer room and other areas.

Purchasing decisions

Purchasing decisions for the equipment in Studio A were ultimately made by Trucks. At the heart of Studio A is an SSL 4056G console equipped with Total Recall. Digital capabilities are provided by a 32-track Mitsubishi X-850, while a Mitsubishi X-86 2-track is used for mastering. Studio A also stocks a large assortment of outboard components. (See "Studio A Equipment List.") One of the most noteworthy components is the new Quantec QRS XL room simulator, which is an open-architecture computer capable of advanced digital processing. Other outboard gear includes an Eventide H-3000 Harmonizer, a Lexicon PCM 70 and 480 L, a Yamaha REV-5 and SPX-90II, and four Kepex IIs from Valley People.

Given the extensive list of gadgetry, Engelke is quick to add that Pegasus isn't completely techno-crazy. "We also place great emphasis on the basics," he says. "As an example, we have a vintage collection of microphones, which includes an AKG C-24 stereo unit and U-47 and U-67 tube mics from Neumann."

Ergonomics

Placement of the outboard equipment rack, as well as the pre-wiring of eight different tape machine positions in the control rooms to suit different engineering styles, are examples of the ergonomic concerns addressed. "I have a degree in cybernetics (the study of information and data flow) music technology, so, in a sense, everything I learned is being put to use here at Pegasus," Engelke observes. "Every aspect of the studio is, in some way, concerned with human interface in its design, and our goal was to make everything as easy to use as possible."

Studio A Equipment List

Console:
Solid State Logic 4056-G with Total Recall

Tape decks:
Mitsubishi X-850 32-track digital
Mitsubishi X-86 2-track mastering deck

Amps: *Power for E-Systems E-IV/D studio monitor system*
(2) Crown Delta Omega
(1) Crown MT 1200 LX
(1) Crown PSA-2
(1) Crown PS-400

Speakers:
(2) E-Systems E-IV/D
(2) Yamaha NS-10M
(2) UREI 813-A
(2) Auratone Cubes
(2) JBL Control One
(2) JBL 4313

Microphones:
(6) Shure SM-57
(4) Shure SM-58
(6) AKG 414-EB
(4) AKG C451
(1) AKG C 24
(2) AKG D-12 E
(6) Neumann U-87
(2) Neumann U-47 FET
(4) Crown PZM
(2) Sennheiser 421
(2) Sennheiser 441
(1) Neumann U-67 tube
(1) Neumann U-47 tube
(4) Beyer 160
(2) E/V RE-20
(4) E/V 757

Headphones:
AKG 240-DF
Sennheiser HD-430
Sennheiser HD-424

Outboard Equipment:
(1) Quantec QRS XL
(1) Lexicon 480 L
(1) Lexicon PCM 70
(2) Yamaha REV-5
(2) Yamaha SPX 90 II
(1) Eventide H-3000
(2) dbx 165
(1) UREI 1178
(1) UREI 546
(2) Orban 622 B
(4) Valley People Kepex II
(1) dbx 900 rack
(1) EMT plate
(2) Audio Digital ADD 3X
(2) JBL/UREI 565 T

Musical Instruments:
(1) Yamaha C7 concert grand piano (with MIDI interface)
(1) Yamaha professional drum kit

Studio B

Directly upstairs from Studio A's control room lies the Studio B control room. Measuring 20 feet wide by 24 feet deep, it is used for projects that require a more intimate atmosphere. Besides post-production, it is well suited for mixing, which works out handily for projects that start in Studio A and move to Studio B.

Studio B is owned by E Systems Technologies. Engelke and staff selected hardware that is centered around an SSL 6056E with a G series computer and Total Recall. The main tape machine is a Studer A820 24-track with Dolby SR noise reduction.

For post work, sound-for-picture and a multitude of other video operations, Studio B houses a Sony BVE editor with A/B roll, variable wipe, a character generator, dual time-base correctors and special effects capabilities. Numerous videotape players and a JVC KM-2000 effects generator are also offered. Studio B also has several synthesizers and samplers. A Synclavier digital music system with 32-voice polyphonic sampling and the latest software upgrades is also available upon request in both studios.

At present, more than 10,000 sound effects are on file in Studio B. The effects are all managed and stored by Leonardo effects software, a database system that allows interactive searches using simple phrases instead of digging through a huge catalog. Also, it lets a client describe to the engineer the type of effect desired; using a sophisticated word search system, the engineer is able to preview, almost instantaneously, many examples that are stored on CD.

Studio B has more than 300 MIDI sequencing tracks available, along with music publishing capabilities, computerized scoring and arranging aids, and full video capabilities centered around the Sony SP format, which is locked into the audio system via Lynx Timeline modules. A wide range of audio monitors is available, and, for video monitoring, everything runs through a 35-inch digital monitor that resides directly in front of the console.

Interface

Two separate mains circuits are used: One handles the infrastructure itself (for example, office light), and one does nothing but run into the receptacles in the control rooms and the main studio. The wiring in the studio areas uses a three-wire configuration at each receptacle. At each point, there's hot and cold, and then a third independent wire that runs all the way back to a grounding stake. The wire measures 4½ inches in diameter and is made from a copper-clad iron pipe buried 87 feet into the ground.

Multipair Canare cable (with its own shielding) was used throughout the facility. For additional protection from possible interference, the multipair was wrapped in aluminum tape before being run through the conduit. The SSL consoles, balanced audio, video, cue and house sync-lines all run through separate conduits. All audio comes in via 4-inch diameter, ½-inch-thick pieces of steel conduit that are also shield-grounded back to the ground stake. For isolation purposes, each conduit was completely surrounded by neoprene and then sealed in concrete. At one point directly beneath the console, all of these conduit feeds were bonded together with strapping wire and then tied to the studio's technical ground. This procedure was done to ensure that the conduit didn't form any ground loops. To conform with international standards, pin 2 is hot in the three-pin XLR connectors.

Another feature of the system interface includes an uninterruptable power supply fed by batteries. It takes up an entire wall in the facility's mechanical room. The batteries provide engineers with a 40-minute time frame in which to turn the systems off without glitching.

A staff of six manages daily activities at Pegasus. Four are technical employees and two are administrative. Engelke's E-Systems Technologies also maintains an office in the building. Studio A's chief engineer is Ralph Moss, who has accumulated 18 years of professional experience at Electric Lady Studios, Regent Sound and many other studios worldwide.

Studio B's chief engineer is Bruce Hensal, whose credits include work for the Eagles, Boston and Dan Fogelberg.

The base rate for Studio A is $200 per hour, while Studio B is $135 per hour for audio only and $200 for audio and video. Both studios are open to the public.

RE/P

Studio B Equipment List

Console:
Solid State Logic 6056 E with G series computer and Total Recall

Audio Tape Decks:
Studer A820 24-track with Dolby SR
Studer A820 2-track CT time code
Studer B215 cassette with remote
Studer CD player

Video:
Sony BVE editor with A/B roll, variable wipe, character generator, dual time base correctors and special effects.
(1) JVC UVCR CR-850U
(2) JVC UVCR CR-600U
(1) JVC TM-2084
(1) JVC TM-13U

Monitor System:
(3) Tannoy FSM U
(2) JBL Control I
(2) Yamaha NS-10 M
(3) Crown Delta/Omega

Synthesizers and Samplers:
Yamaha DX7 II FD
Yamaha TX 802
Yamaha KX 88
Roland JX-10
Roland D-550
Roland S-550 with DT-100 and MESA
E-mu SP-1200 percussion sampler
Akai S 1000 MIDI wind instrument
Akai S-900 sampler
Simmons SDX5 drum kit
Kahler Human Clock
E-mu E-max
Kurzweil 250 RMX

Computers:
IBM AT with 4Mbyte RAM 80Mbyte disk
Atari 1040ST
Apple Macintosh II 4Mbyte RAM 40Mbyte disk
Commodore Amiga
Hewlett-Packard Laserjet II

Interfaces:
Microtech 2400 modem
Adams-Smith 2600
JL Cooper 16/20 MIDI switch box
Akai digital patch bay

Effects Processors:
Yamaha SPX 90 II
Yamaha MEP-4 MIDI events processor
Orban 424A comp/limit de-esser
ART IEQ
Lexicon PCM 70 digital processor
Yamaha REV 5
Lexicon 224 XL with LARC
ART DR1 digital reverb
Korg DVP-1 voice processor
Lexicon 480 L
Valley PR-10
Valley Gain Brain II
Valley DSP
Eventide SP 2016
Quantec QRS XL

Custom Monitors at Pegasus Studios

The E IV/D monitors in Pegasus' Studio A control room (custom-built by E-Systems Technologies, a company owned by Pegasus Studios' technical adviser, Dave Engelke) are based around Community Light & Sound's M4 compression loudspeakers.

"Monitoring was a paramount consideration when we built Studio A's control room, so we literally built the room around the speakers," Engelke reveals. "Our goal was to have a system that was accurate and provided a true representation of the actual sounds in the studio. For that reason, we went to great lengths to create a physical environment that would be conducive to proper acoustical presentation and were highly selective about which components went to the cabinets themselves.

After producing a series of prototype monitors, Engelke settled on a proprietary four-way design he named E IV/D. The cabinets each weigh 600 pounds, stand 45 inches tall and measure 38 inches wide by 24 inches deep. Not your average square box by definition, the cabinets, like the shape of the control room itself, are highly geometric with many angular planes. Power for the cabinets comes from five Crown amplifiers. The crossover network was provided by Creative Electronics of Nashville.

Engelke says he chose the M4s because of their strong mid-range performance, intelligibility, low distortion characteristics and power handling capabilities. "In my estimation, the mid-range section is too often ignored when choosing the proper components for a loudspeaker system," he says.

"I think people tend to ignore the mid-range because they're so preoccupied with high-end response and low-end bass. As a result, when you listen to some monitors, you find that the mid-range band simply isn't accurate in its representation of sound, especially with snare drums or the male vocal range. To illustrate my point, try running a signal off of a snare drum into a woofer.

E-Systems Technologies E IV/D monitor.

Without question, you will not obtain true reproduction of sound. Conversely, the same thing will occur when you try to run sound from a low-frequency instrument into a high-frequency driver. Over the years, it became obvious to me that something had to be done to get accuracy back into the mid-range spectrum, and that's where the M4 came in. When you look at the M4's total response, the mid-range frequencies between 200Hz and 2,000Hz are just about as optimum as you can get for accurate performance."

To fit Engelke's design parameters for the E IV/D monitors, the M4s were modified slightly to accept a custom-tailored version of Community's SH1894M horn. "In accordance with the plans I drew, we removed the front of the M4 (the portion of the unit containing the phase plug) and exposed the diaphragm. Next, we massaged the Community SH1894M horn into a hypex curve configuration, which extended its entrance enough to adapt directly to what was now a 7-inch exit area on the M4. The result was a 1:1 design, which reduces the distortion to an astonishingly low level."

In each cabinet at the bottom end, signals below 120Hz are directed to an 18-inch JBL 2245 driver, which has had its mass ring removed from the voice coil. By taking this step, Engelke was once again going for greater accuracy. "Accuracy problems result in a lot of low-end devices because of the mass ring on the voice coil, which makes the cone resonate when you punch it and produces a booming effect. Although that's desirable in some systems, we designed our system from the amplifiers up, so for this portion of the low end, we decided to go with a velocity-controlled Crown Delta Omega mono amp for each driver. With the velocity-controlled amps, we have a situation in which the amp is literally capable of sensing the position of the voice coil. By removing the mass ring, you let the amp have as much control over the voice coil as possible and dramatically increase accuracy."

For low-end frequencies of 120Hz to 400Hz, a Crown Micro-Tech 1200LX powers twin TAD 1601-A 15-inch woofers in each cabinet. At 400Hz, signals cross over into the M4s, which are driven by a Crown PSA 2. The M4s carry the signals up to 1,600Hz, where they cross over into a TAD 4001 coupled with a Community SH864 horn, which theoretically can carry them up to 22kHz with the help of a Crown PS 400 amplifier.

Engelke's E IV/D were made phase-coherent by physically adjusting the drivers in each cabinet. This adjustment is part of the reason for their appearance. Each cabinet was built around the tweeter and its SH864 horn, which reside at the center of the enclosure. Directly underneath the tweeter and its horn lies the M4 and its custom horn, which was positioned as close as possible to the tweeter by bonding both devices' horns together with a wood emulsion. The three low-end transducers are situated side by side at the top of the cabinet with the single 18-inch element located between the twin 15-inch woofers.

"We conducted extensive tests with every driver," Engelke points out. "Then we matched the position of each driver in such a way that they all had the same relative acoustic center. The resulting configuration provides a high degree of phase-coherent monitoring."

Marketing: New Business Development

By Sarah Coleman and Michael Fay

A studio's image has many different tiers: business, technical, maintenance, physical plant and the image of the personnel.

You want more clients, more work and more income. But you're not the only studio owner who wants more business. In today's competition for business, to be successful studio owners must constantly look at ways to adapt to the changes in the recording industry.

Most studio owners will agree their primary marketing tools are providing good service, maintaining a good image through positive word-of-mouth promotion and staying current with technological changes. But, these tools also can benefit from some additional planning and well-conceived marketing ideas.

Marketing your studio should involve more than taking your client to lunch. A complete plan involves everyone in your business. Everything your studio does, including billing, office routines and phone calls, should have a marketing slant. Secondly, effective marketing plans focus on client retention as well as acquisition. There is an 80/20 rule in business that says 80% of your income is generated by 20% of your clients.

Your marketing begins with the needs and wants of clients as they perceive their needs, not necessarily what you as a studio owner think.

Marketing research

The first and most important step is research. Market research doesn't have to be complicated or expensive. The following research questions won't cost anything but your time. Ask yourself these questions and list the answers.

1. Who are our existing clients?
2. Who are our potential clients?
3. What are our rates?
4. In what industries are our clients, and are their companies growing, shrinking or stagnant?
5. What niche do we fill? What niche could we fill?
6. Have we lost any clients? Why?
7. What do our clients think of our work, our image and our reputation? (To find this answer, simply ask your clients and listen to their comments. You may be surprised. See "A Client Review Program" for examples.)
8. What are our style, fee structure, strengths and weaknesses, equipment, gross profit, invoicing and collection procedures, billable hours, client base and share of market?
9. Who is the competition?
10. What new business is available—beginning with current clients?
11. What is our current marketing plan and activities? What was effective? What was not effective?
12. Who are our people? What are their strengths and weaknesses?

After answering these questions, identify opportunities, and write a position statement that clearly defines your studio. You can't develop a marketing plan until you know what you are marketing. Therefore, the more complete, original and solidly developed your position statement, the easier your entire marketing plan will fall into place. The statement can include as many as five or six points, or maybe more, depending on the size of your studio and what you find from your research. An example of a position statement follows:

Position Statement for Studio Reel, 1410 Main, Baltimore, MD.

•Studio Reel is the second-largest commercial production facility in Baltimore.

Sarah Coleman is a free-lance writer based in Washington, DC. Michael Fay is editor of *RE/P*.

A Client Review Program

Keep in mind that the sole purpose of meeting with a client under the client review program is to obtain feedback on your studio's services. Do not try to sell additional services to the client. Ask and listen.

If possible, the client review should be conducted by a partner other than the one in direct charge of the client's account. The points to cover are:

1. What is the client's opinion of our facility and services?

2. What is the client's opinion of the engineers and support staff with whom they have had contact?

3. Have we done adequate follow-up upon completion of the projects?

4. Have we helped clients in other areas besides their specific projects?

5. Does the client feel let down? If so, why?

6. Would the client recommend us to other people, if asked?

Write a brief report afterward, with distribution to the appropriate personnel.

Example taken from the "Practical Accountant" magazine.

•The facility has increased billable hours 15% in the last six months.
•Combined, the staff has 30 years experience in advertising and corporate commercial production work.
•Rates are competitive because the studio has low overhead.
•Clients are pleased with the studio's work and 75% have been with the studio since it opened eight years ago.
•Two new competitor studios are also working with some of the facility's clients. Reason: equipment and quick turn-around time.
•Three new clients from New York are coming to the area and working with our studio.
•The No. 3 production facility has recently added a computer/MIDI/sampling package.

From a marketing aspect, look at your position statement and focus on areas of growth and weakness for your studio.

Strengths—Studio Reel is stable and has passed the five-year danger zone of small businesses. In its eighth year, 75% of its client base began with the studio and billable hours increased 15% in the last six months. What you didn't anticipate this past year was that you were serving three new clients from New York.

Weaknesses—Clients have no major complaints; however, two new facilities also are serving two of the studio's major clients, and the No. 3 facility has added the new MIDI package.

After completing the position statement, pick three to five major objectives to address in your marketing plan. At this point, make two separate objective lists, one emphasizing the specific plans you have for the next year and the other, strategic ideas for the next five years.

For example, the main one-year objectives could include:

•Keep the 75% original client base content and coming back.
•Increase the billable hours 10% in the next six months.
•Research what other facilities are providing your clients. Can you also provide the same services?
•Will the addition of a computer/MIDI/sampler system bring in additional business?
•Will the additional business offset the capital expense of the hardware purchase?
•Continue to target more clients from New York.

Its main five-year objectives might be:

•Add duplicating studio/service.
•Start a rental company.
•Upgrade to the next level of technology; for example, move from 16-track to 24-track or from analog to digital tape machines.

The marketing plan

At this point, don't become frustrated with following a certain format. Depending on the size of your studio, your plan could be two typed pages or an entire notebook, bound and copied for your staff. After you have completed the research, position statement and objectives (one- and five-year plans), you are ready to decide on the marketing steps. The three areas to address in your marketing plan are present clients, new business and development of the company's image.

Remember that an effective marketing plan should identify opportunities and turn them into satisfied clients. Most of all, keep your ideas simple and attainable. Your marketing ideas should reflect the same technical accuracy and creative energy you produce in the studio. A simple, small plan that is carried through is better than an elaborate plan that sits on the equipment case.

Present clients

One obvious, but often overlooked, marketing aspect is an organized method of knowing your clients. For example, who are your clients' decision-makers? How is the client doing compared to its industry or field? What are the new developments in that field? Do you really understand what the client does? Who are your client's clients? If you don't know the answers to these questions, find them. It will improve and solidify your current client relations.

Make sure you are meeting all of your clients' needs. Are they going to another studio for work that you could do? If you don't have the equipment or people needed, is it possible to upgrade?

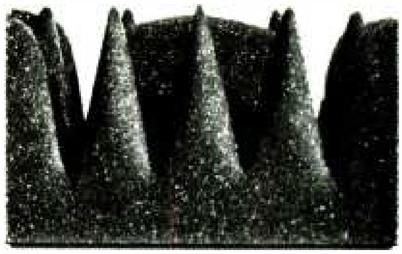
Maintain a complete, up-to-date client database. Allow one line in the entry for a category, such as corporate, advertising, music or radio. Use the database for client information besides just their addresses and phone numbers. For example, use details from previous sessions, special microphone preferences, favorite foods or restaurants, and spouse's name.

Maintain records on client sessions. Try to transcribe your notes for future reference in case a client asks you to "do what you did last time." With a reference point, you can more efficiently give your clients exactly what they're looking for.

The most important aspect of keeping present clients content is to know them and continue to grow with them in their businesses. Go beyond the control room and know what their businesses entail.

Dropping a present client doesn't sound like a growth strategy, but, in certain situations, it is. Some of your clients may be small, have little potential, and require more work than is paying off. Try to work out an objective system to prune out clients. But remember, just because a client is small doesn't mean it is a waste of time. Small clients become big clients, too. Try to justify your billing, time and manpower. This process is tricky and sensitive. So, handle with care and learn how

A pointed attack on noise.

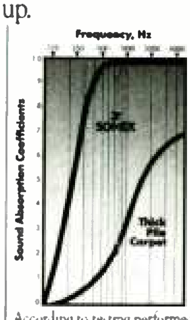


SONEX

SONEX acoustic foam wages a two-front war on noise.

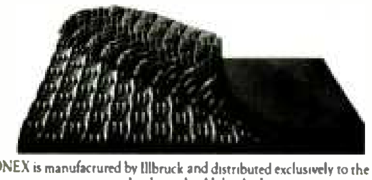
First, the patented SONEX wedge traps, deflects, and scatters noise. The wedge's depth and angle carry noise waves down into the lowest point of each anechoic foam valley. Most of it doesn't have the energy to come back up.

Then the foam itself converts sound energy to silent kinetic energy. Sound literally gets lost within the open cell pores of this special foam. What the wedge doesn't dissipate, the acoustic foam converts to silence. Together this two-pronged attack kills background noise every time. Call or write us for all the facts and prices.



According to testing performed in strict accordance with ASTM C423-77 procedures, at 500 Hz 3" SONEX has a weighted noise reduction coefficient (NRC) of almost 0.98, while thick carpeting is 0.15

AlphaAudio. Acoustics
2049 West Broad Street
Richmond, Virginia 23220 USA (804) 358-3852
Telex: 469037 (ALPHAAUD CI) FAX: (804) 358-9496
Acoustic Products for the Audio Industry



SONEX is manufactured by Illbruck and distributed exclusively to the pro sound industry by Alpha Audio.

Circle (35) on Rapid Facts Card

Table 1. Example of different marketing techniques.

	Marketing Techniques			
	Advertising	Publicity	Promotion	Direct Sales
Message:	Sales	News and education	"Good guy"	Friend
Tools:	Paid ads	Editorial	Event	One-on-one
Example:	TV commercial	TV news story	Charity fund-raiser	Personal meeting
Tone:	Hard-hitting	Subtle	Subtle	Either
Cost:	$$$$$	$	$$	$
Affect:	Guaranteed awareness	High credibility	Goodwill	Human

Taken from Kansas City Press Relations seminar. Prepared by Candace Keohler and Nancy Werner.

to cut off relationships with poor clients and discourage similar potential clients without hurting your studio's reputation.

New business

Acquiring new people, new and different work and more revenue sounds good. But, it is also one of the toughest tasks, requiring the most organization and time. Sometimes clients just walk in the door or call you "out of the blue." But, other times obtaining new business requires a lot of work, especially for a start-up facility.

Begin a database of target clients as you did for present clients. (Keep this as a separate file from your current clients.) Again, list the clients and everything you know about them. What are they doing now, and what studios are they using, if any?

How do you find these targets? First, decide on your target geographic area: local, regional or national. Start with the city telephone directory and yellow pages, state industrial directories, "Who's Who" directories and "Contacts Influential." Define and identify those who are "influentials," people with position, and include them in your database. Keep your eyes open for opportunities to meet them. They may not be prospects for your company, but they may be business references if you get to know them well enough. Don't forget your banker, accountant, insurance agent, lawyer and realtor; they know your business and care about your success. Ask them for leads and ideas.

Develop a plan for each good target. Each client is different. Taking one client out to dinner may work, but another client may rather just hear your tracks.

The initial contact with your lead can be the most crucial. Providing a cost incentive is usually not going to pull an established client away from another facility. The elements that are attractive to such clients are consistency, reliability, efficiency, and quality—usually in that order. Unfortunately, these characteristics are the most difficult for a new or competing studio to demonstrate. As an owner, you must create a situation in which the clients you seek can work in your facility without budgetary, time or completion pressures. In other words, they shouldn't have to count on using the work that's done in your facility—the first time.

Another effective way to demonstrate your facility's capabilities is actually to hire your future clients to work on a project in your studio. This way, they will be able to absorb the working environment at *your* risk, not theirs.

As Bob Yesbek, owner of Omega Studios, Rockville, MD, says, "Anyone can lay a track, but not just anyone can drag someone in to hear it."

Omega, now in its 20th year of operation, is one of the largest and oldest studios in the Washington, DC, area. The 48-track, four-room studio also features a recording school.

Marketing Omega Studios hit a turning point in 1976 when the owners completely redesigned the image. Known as the 1960s rock-and-roll studio, they decided they also wanted the coat-and-tie crowd. With the assistance of a marketing consultant (who is also a client), the studio designed new business cards, stationery and the on-site image.

Now, the studio maintains a database of "thousands" of names, advertises in pro-audio trade journals and yellow pages, and works with public relations companies in Rockville and New York.

"In this industry, you have a narrow market to hit," Yesbek says. "We found that direct mail was a complete failure. People shop around for studios, for instance, advertising in the yellow pages.

"But most important is on-site image. We pride ourselves on being a place where our clients can bring their clients. In the past, we spent more money on active marketing because we needed more people to know about us. Now, we like to put our money in state-of-the-art equipment and staying up-to-date."

After you have made the initial contact with your target clients, you are ready to make a proposal, which is usually a list of goods and services, at specified rates and terms, that you are offering the prospective client. For example, you know through your research that this new client books 20 hours a month on a regular basis. Your proposal may be a guarantee that they can: block out favorite days and time slots, select the engineer and get a 20% discount from book rates, providing they pay net 20 days. Any or all of these terms may be an improvement over the client's current terms. Notice the studio gets something in return for what it gives up in other areas, that is, early terms on payment. Proposals are written forms of negotiation, and both parties must benefit from the transaction.

After you have made the proposal, close the sale. Don't forget to ask for the work. It's the most important part of reaching a new client. Simply say, "I would like to work with you. I can produce exactly what you want."

Finally, follow up with your clients. If you don't hear from them, call back. If they decided to go with someone else, ask why. Even if you didn't get the job, you have gained something if you know why someone didn't choose your studio.

Developing the studio image

First, step outside and look at your building. Do people know you are there? Is your address legible? Your studio's image should remain clear and consistent throughout your identity, including signage, letterhead, business cards and invoices. This consistency breeds quality and precision in the work that comes from the front office, as well as the control room.

As a recording studio owner, you have a speciality. Spread your knowledge and abilities by writing for trade journals, or give speeches at university audio and music classes. Join business and trade associations and charitable, religious, civic and educational groups; these can be avenues to more exposure for yourself and for your studio.

Also important is a studio referral system. Depending on how "friendly" your competition is, you can build a network of other facilities and benefit from referrals.

If you have the capabilities, and time,

consider publishing a local audio newsletter on a desktop publishing system. A collaborative effort by several studios would put you at the forefront of studio activities and news in the area. Send the newsletter to local and regional studios, clients, vendors and manufacturers, as well.

Consider giving parties or holding open houses at the studio to unveil new equipment, introduce a new engineer or celebrate a holiday. Invite clients, vendors, neighbors, community leaders, media and other studio owners. The bigger the mix, the better.

Advertising

If you decide to place a display ad in a local newspaper or trade journal, contact an agency or free-lancer to design the print ad. Unless you are skilled in design, you could do more harm than good with your advertisement. Remember, you are a studio owner, not a graphic designer. As you select an agency, shop around. Ask to meet the person who would manage your account. Ask what budgets their other clients have. If yours is also in that range, this agency is about the right size for you to work with. However, if the other clients appear to have higher budgets, you will be one of the agency's "small" clients, and you many not receive the attention you deserve. Also, make sure you understand the breakdown of fees.

While you are working with an agency, consider designing a complete studio identity, including business cards, Rolodex cards, rate card, reel labels, brochure and letterhead. In the end, you will save money if you have it all designed and printed at the same time. Don't let an over-zealous art director talk you into a slick brochure. Good design is not necessarily splashy. The printed pieces are intended only for communication to clients, and they should do just that: communicate. Keep your designs clean and simple, and state the important information clearly.

To help you communicate with an art director, collect some examples of graphic design you like. This portfolio will give the designer an idea of your preferences.

When selecting the medium in which to place your ad, carefully screen the publication's readers. If you have completely researched your position, you should have a clear idea of who you want to reach, so make sure the newspaper or magazine reaches those clients. Request to see a Business Publications Audit of circulation (BPA) statement, if available, and ask for an explanation of the form. Two points you should understand are: who the readers are and what the renewal rate is for the publication. A print ad is primarily used to build and/or maintain an image, and indicate a presence in the market. It is not an active marketing tool that will bring in a rush of new clients.

Personal marketing

A well-organized marketing plan, a successful party at the studio and an effective brochure and rate cards are all just parts of your business growth. But, you must also remember the personal aspect to marketing. Ultimately, it is you and your staff that will sell your facility.

A marketing plan can't run on its own. Even if it is a good plan, if no one knows how to make it work, it won't serve its purpose. Keep your ideas simple, personal and organized, and allow each person in your studio to play a part.

Studios need people, equipment and the opportunity to provide services. When you do get someone's attention through your marketing plan, remember, now it's time to deliver. RE/P

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Talkback

Binder Breakdown in Back-Coated Tapes

By Scott Kent

The phenomenon of binder breakdown is becoming more noticeable now that record producers are pulling old tapes out of storage for remastering.

In recent years, the number of complaints regarding oxide shredding from back-coated tapes has increased. Often, these complaints come from tape recorder owners who say their machines are making a squealing noise during playback, and they wonder whether they should bring the machines in for service. An examination of the machines shows a much heavier than usual oxide deposit on both heads and fixed guides. The deposit is not dry, as in normal oxide shed, but somewhat sticky and resistant to removal.

The problem appears to be binder breakdown: Solvents in the binder react with constituents outgassing from the Mylar base and make a "goo" out of the oxide, which then builds up on heads and guides. The problem involves the length of long chain molecules in the polymer chains in the urethane binder compound. If the molecule chains are too long, the binder does not properly grip the oxide particles; if too short, the binder does not remain cohesive. The short chain components tend to migrate to the surface in time, causing the tape surface to become sticky. To a lesser degree, the back-coating binder layer also has this problem, causing the back-coating to stick to adjacent layers of oxide.

What tapes are affected?

This phenomenon is becoming more noticeable now that record producers are pulling old tapes out of storage to remaster them to CD. This problem does not appear to involve humidity or storage conditions, as was originally thought. However, the problem seems worse in playback of these tapes under conditions of high humidity, or when the tape recorder heads and guides are colder than 60°F or warmer than 80°F. These problems commonly affect back-coated tapes manufactured from the late 1960s (when they were introduced) to the early 1980s. This is a continuing problem; some tapes manufactured in 1983 already are affected. In addition to Ampex 406, 407 and 456, there are reports of 3M 226, 256 and 250, as well as AGFA 369 and 469, exhibiting binder breakdown. AFGA 468 does not appear affected. Some BASF tape stocks may be affected. There are no reports of Maxell UD, videotape, or digital audiotape stocks being affected.

If you play a portion of a tape on which the oxide is coming off, and if there is any signal on it, a screech or squeak caused by the oxide piling up on either a fixed guide or a head (Often the erase head clogs first.) modulates the magnetized particles and redistributes them mechanically in the pattern of the scrape flutter. If this happens, you may then have a permanent and unrepairable squeak magnetically

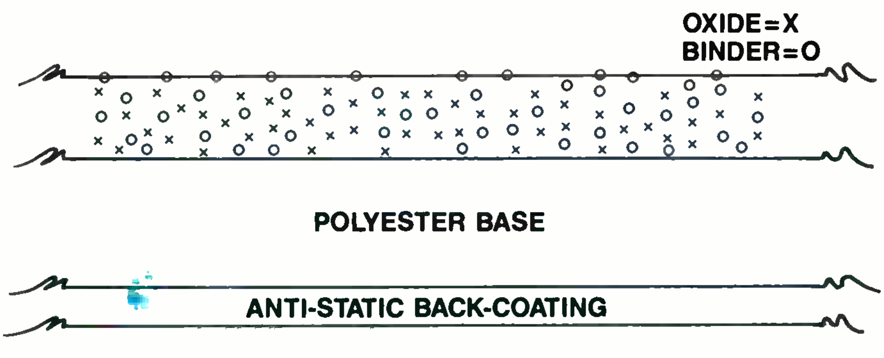

Figure 1. Tape cross section.

During a recent job, have you encountered a problem or unusual request that required a unique solution? We would like to share it with the industry. Send it to "Talkback"; if we use it, we'll pay you $50. "Talkback" is a forum for sharing your solutions to difficult production situations other engineers may encounter. In a continuing effort to educate, we believe that this type of information is helpful and will display your professional abilities. This is not a tech tips column; rather, the focus is on solutions to problems—technical or non-technical.

To submit, in 1-2 pages describe the job, what the problem was and what you did to solve the problem. Include any supporting documentation, such as diagrams or photos, that would help explain the situation. If we publish your entry, you and your company will be fully credited.

Send material or inquiries to Michael Fay, Editor, *RE/P*, 8885 Rio San Diego Drive, #107, San Diego, CA 92108.

Scott Kent is a recording engineer, producer, and president of BKM Associates/AFKA Records, Wilmington, MA. He also is technical director for Northeastern Records and a faculty member of the Sound Recording Technology Program, College of Music, University of Lowell.

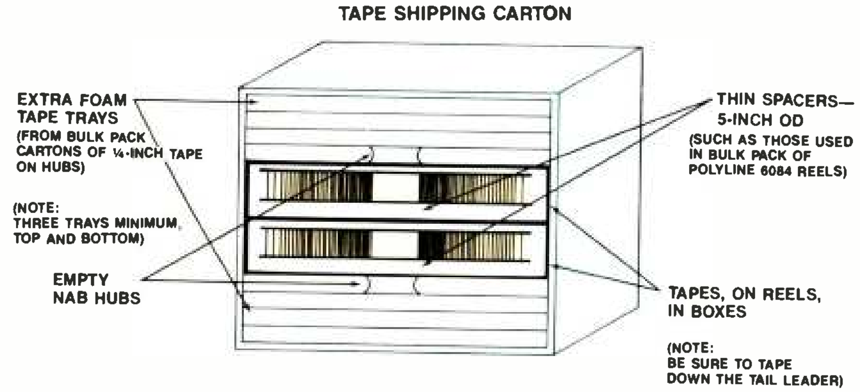

Figure 2. *Careful packaging is essential to proper shipment of scatter-wound tapes.*

recorded on the tape. When there is no signal, it appears that the redistribution may cause only the mechanically audible squeak without applying a squeak to the tape magnetically.

If you have rewound the tape to the head of the reel, for example, you also have the problem of how to get it off the machine without further damage. One way to approach a suspected problem tape is to play an unrecorded portion of the tail. Alternatively, play the test tones. Ideally, these are at the tail. A slight removal of oxide, first noticeable on the erase head, may give warning of a problem before a squeak occurs. In short, if in doubt, it is better to have the tape treated before trying to play it. Even a slight removal of oxide means that the highs are being removed from the tape, because the short wavelengths are stored closer to the oxide surface.

Is a cure available?

A remedy for this condition is available, and we are using it successfully to recondition masters. However, the process is time-consuming and somewhat risky for the inexperienced.

The requirement, basically, is a carefully controlled time-temperature cycling process to reverse the binder breakdown. The heat allows the binder system to rebond temporarily. The treatment is not permanent, but gives at least a 30-day time window in which to work with the tape. Provided the time-temperature cycling is done correctly, there is no measurable high-frequency loss, noise increase or increase in print-through. Treated tapes appear to play with no difficulty and no apparent damage. It also appears that the tape could be re-treated later for another use period.

Because the exposure temperature must be controlled within +3°F, this process requires an industrial oven with circulating air-flow and precise temperature control, not a kitchen oven. The oven used must be free of any magnetic fields (caused by relays, for example) and any radiant heating effects.

Caveats

To do this treatment correctly, the tapes need to be flat-wound onto plastic reels, which may be a problem if you have fast-wound the tape to the head of a reel. To correct the winding requires a specially designed spooler that has no fixed guide surfaces, yet provides a smooth wind. Our experience suggests that shipping a scatter-wound tape, correctly packed for shipping, is far safer than attempting to flat wind it on any conventional equipment. (See Figure 2.)

Certain kinds of splicing tape (Radio Shack's, for example) can bleed, ripping off the oxide when played, but 3M ST-67 and BASF seem to be fine. 3M (Scotch) Type 41 or 621 are not recommended for use on back-coated tapes. Often, splicing tapes need to be replaced even before a binder problem appears, and they bleed with temperature and age. Splices made with 3M Type 620 sometimes come apart after the oven treatment and require replacement. Fortunately, most people only used these latter types on non-back-coated tape. RE/P

For more information on reconditioning problem tapes, contact BKM Associates, Box 22, Wilmington, MA 01887.

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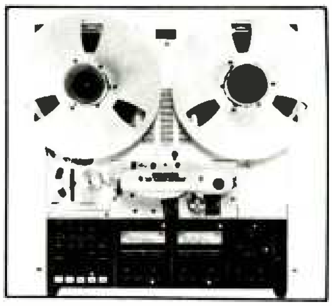

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Spotlight

Between the Ears

By Dan Torchia

Located just outside of Manhattan, BTE divides its time between being a personal-use studio and a workshop for up-and-coming artists.

Between The Ears started out as a personal-use studio and ended up being a full-time music facility.

Owned by Cy Curnin, lead singer of The Fixx, BTE began as a place where he could write when he was not on the road. Located in Douglaston, NY, just outside of Manhattan in Queens, BTE is attracting music clients who need to make high-quality recordings at a moderate rate.

Working at a recent session at Between the Ears are (left to right): Andrew Sedgwick, co-owner, manager and programmer; Cy Curnin, co-owner and lead singer of The Fixx; and Rob Bergston, chief engineer.

"There is a great vibe here for a writer," Curnin says. "I began recording here years ago. In fact, a good deal of the 'Phantoms' LP was written and recorded here."

A two-room facility, BTE caters to several segments of the music recording market. Studio A is BTE's live room and features a Sound Workshop Series 40 console with Otari and MCI tape machines. Studio B is a MIDI preproduction room with a Ramsa WRT-820 console, Tascam tape machines and a variety of synthesizers, drum machines and sequencers.

Although the focus is on the live room, having the MIDI room enables BTE to cater to a larger variety of clients, Curnin says.

"If somebody's in the main room, a writer can come in and work on his own and have a complete song together in a couple of hours," he says.

Background

Curnin's quest for a studio resulted in his wanting to have a place to write and store gear. Early in The Fixx's career, its record company referred the band to Workshoppe Recording studios. He liked the studio and began spending a lot of time there. When it came to finding a studio to buy, BTE seemed the place.

"I initially thought about buying a small

Dan Torchia is staff editor of RE/P.

place," he says, "but I thought, 'Why don't you buy a place that's larger?' I spend a lot of time on the road, and the place can operate on its own while I'm gone."

Andrew Sedgwick, the studio's programmer and manager, runs the studio from day to day; the chief engineer is Bob Bengston. Together, they foster an atmosphere in which bands and songwriters can develop their craft without worrying about money.

This philosophy stems from Curnin's early days with The Fixx, which gained initial interest from publishing companies, not record companies. The group needed high-quality recordings but did not have a lot of money. With that in mind, Curnin says, BTE is meant to have a workshop atmosphere.

"I thought that the industry was running away with itself," he says. "Until record companies feel they spend a lot of money on a record, they don't believe in it. To me that's ridiculous."

Because of the Queens location, Curnin believes that BTE does not directly compete with Manhattan studios. Top-quality equipment is important, he says, but BTE is not out to compete with larger studios that actively acquire the latest technology. BTE's mission is to provide a high-quality environment where people can get a good initial recording, then transfer to a larger studio if there's still work to be done.

For Curnin, the location is ideal. Although he lives in Manhattan, he does not like to work there and calls it "Studio Hell." Douglaston is about a 20-minute drive from Manhattan, is easily accessible by car or train, and was recommended to him by a former member of Billy Idol's band, Steve Stevens, who grew up in the area.

"The atmosphere reminded me of England," Curnin says. "I never specifically thought of locating in Queens, but it just worked out that way."

BTE has been relying on word-of-mouth to attract customers, which seems to be working, Curnin says. He's been trying to get into the studio but has been having trouble because of the number of bookings.

"It's not so much a profit venture, but a place where I can write and a place for Steve to work," he says. "And, we can help bands out and have it pay for itself."

RE/P

At a Glance

Owners: *Cy Curnin (Studio A); Andrew Sedgwick (Studio B).*
Studio manager/programmer: *Andrew Sedgwick.*
Chief engineer: *Bob Bengston.*

Studio A

Console: *Sound Workshop Series 40 console, 36×16×24, with Diskmix automation.*

Tape machines: *Otari MTR-90II 24-track with remote, autolocator and 16-track record/reproduce head assembly; MCI JH-100 ½-inch, four-track recorder with ¼-inch, two-track and ¼-inch, full-track head assemblies; Technics RM-85II cassette decks.*

Video and sync equipment: *JVC 6650 ¾-inch video recorder; BTX Shadow SMPTE-based machine synchronizer; BTX controller; Otari IFC Shadow/Otari MTR-90 interface; JVC IFC Shadow/JVC interface; MCI IFC Shadow/MCI-110 interface; Roland SBX-80 SMPTE-to-MIDI/clock converter; Panasonic 19- and 5-inch color video monitors/receivers.*

Gain reduction signal processors: *UREI 1176LN leveling amplifier/peak limiters; UREI LA-3A leveling amplifier/peak limiter; dbx 160 compressor/limiter; Drawmer 201 stereo gate; Kepex gates.*

Outboard EQ: *Pultec PEQ-1 tube EQs; Pultec PEQ-1A; Audio Arts stereo parametric; MXR-15 band stereo graphic; White ⅓-octave passive filter set.*

Effects signal processors: *Yamaha REV-7 digital reverb; Roland SRV-2000 digital reverb; Lexicon Prime Time digital delay; Delta Lab DDI digital delay; Delta Lab Effectron II digital delays; Korg SDD-2000 sampling digital delay; Eventide 910 Harmonizer; Eventide PS-201 stereo phase shifter; Eventide FL-201 stereo flanger; Roland Vocoder; Sound Technology Echoplate II.*

Speakers: *UREI 813s; Yamaha NS-10Ms; Auratone 5Cs; EQ Boy speaker selector.*

Power amplifiers: *Bryston; Phase Linear; Dynoco; Bogen.*

Musical instruments: *Yamaha C-7 grand piano; Hammond B-3 organ with Leslie speaker; Oberheim OB-8 synthesizer with MIDI retrofit; Oberheim DMX digital drum computer with MIDI retrofit; Oberheim DSX sequencer; Korg DDD-1 digital drum computer; Ludwig drum set; assorted other drums, cymbals and percussion.*

Studio B

Console: *Ramsa WRT-820.*
Tape recorders: *Tascam MS-16; Tascam 32-2; Tascam cassette deck.*

Signal processors: *Roland SRV-2000 digital reverbs; Delta Lab Effectron II digital delay.*

Speakers: *UREI 809s; Tascam CM-10s.*

Power amplifiers: *Yamaha.*

Musical instruments: *PPG 2.3 digital wave synthesizer; Oberheim Xpander; Sequential Studio 440 drum computer/sampler/sequencer; Roland Juno 106; Kramer bass.*

Address: *40-35 235th St., Douglaston, NY 11363; 718-229-5057.*

STUDIO UPDATE

Studio News

Northeast

Iris Sound (Royersford, PA) has purchased a Soundcraft series 760 MkIII 24-track recorder. Its Allen & Heath mixing console has been expanded to 32 channels, and the studio has added outboard effects and EQ. *237 Main St., Royersford, PA 19468; 215-948-3448.*

Flite Three Recordings (Baltimore) has named Kirk Davis to its post-production staff.

Quantum Sound (Jersey City, NJ) has purchased a Studer A820 2-inch recorder for its Synclavier room and a Studer A820 ½-inch machine for its SSL mixing suite. Other recent purchases include three TC Electronic 11-second samplers and two PCM 42s. *512 Paterson Plank Road, Jersey City, NJ 07307; 201-656-7023.*

Shelton Leigh Palmer & Company (New York) has named Syd Weiss as executive producer. *19 W. 36th St., New York, NY 10018; 212-714-1710.*

Midwest

Cedarwood Recording (Ashland, OH) has remodeled and expanded its production capabilities. Equipment purchases include an Ampex MM-1000 16-track recorder. *1628 St. Rt. 511-S, Ashland, OH.*

The Disc Ltd. (East Detroit, MI) has installed a Solid State Logic 4000 G series console with 32 ins/outs on a 40-I/O mainframe. *14611 E. Nine Mile Road, East Detroit, MI 48021.*

Producers Color Service (Southfield, MI) has named Michael Suggs as editor. *24242 Northwestern Highway, Southfield, MI 48075-2583; 313-352-5353.*

Tone Zone Recording (Chicago) has added several pieces of new equipment, including a TC Electronics 2290 digital delay and effects processor, Tube Tech PE-1B program equalizer, Tube Tech CL-1A compressor, Teletronix LA-2 leveling amplifier and a Roland DEP-5 digital effects processor. *1316 N. Clybourn, Chicago, IL 60610; 312-664-5353.*

Southern California

Waves Sound Recorders (Hollywood) has opened Studio E, which runs in conjunction with Bert, Barz & Kirby and features a Soundcraft console with MCI machines. Construction on two new video sweetening rooms was scheduled to begin in June. *1956 N. Cahuenga Blvd., Hollywood, CA 90068; 213-466-6141.*

Summa Music Group (Los Angeles) has opened up a recording studio with design by George Augspurger and Lakeside Associates. Studio A features a 64-input Solid State Logic SL-4000 Master Studio System with Total Recall and G-Series EQ and fader automation. Tape machines include a Studer A-820 and A-800 analog multitracks, a Mitsubishi X-850 digital 32-track and Ampex ATR-102 2-tracks. *8507 Sunset Blvd., Penthouse 1, Los Angeles, CA 90069; 213-854-6300.*

Interlok Studios (Hollywood) has upgraded its operations to include audio post capability. New equipment includes a Sony BVH 3100 1-inch video machine, an Otari MTR 90-2 24-track recorder and a Cipher Softouch synchronizer. *1522 Crossroads of the World, Hollywood, CA 90028; 213-469-3986.*

Cantrax Recorders (Long Beach) has added a Yamaha SPX-9011, Loft TS-1 RMX test set, Rane HC 6 headphone console, Rane real-time analyzer and new software for its Studer A820 2-track. *2119 Fidler Ave., Long Beach, CA 90815; 213-498-6492.*

Northern California

Dave Wellhausen Studios (San Francisco) has upgraded its equipment to include Westlake BBSM8 monitors and a Digital Creations Diskmix automation system for its new Soundworkshop 34C console. *1310 20th Ave., San Francisco, CA 94122; 415-564-4910.*

Northwest

Steve Lawson Productions (Seattle) has named Vince Werner as studio manager and chief producer/engineer. The studio has also created a marketing department, directed by Celia Congdon with the help of Kim MacQueen. Bob Israel has been named sales manager. *Sixth and Battery Building, 2322 Sixth Ave., Seattle, WA 98121; 206-443-1500.*

Canada

Cinar Studio Center (Montreal) opened June 1 with three integrated rooms, all designed by Tom Hidley. Studio One, designed for music recording, features a Solid State Logic 56-input console with Total Recall. Studio Two is designed for film post-synchronization; Studio Three is designed for electronic music, sound effects and synchronized pictures. *1207 rue St-Andre, Montreal, Quebec, Canada H2L 3S8; 514-843-7070.*

Spain

Atanor Studios (Madrid) has installed an AMS AudioFile with two-hour recording capability in its newly built studio. *Agastia 5, Madrid (28027) Spain; 34 1 407-8011.*

Kash Productions (Madrid) has added a two-hour version of the AMS AudioFile to its list of studio equipment rentals. *Agastia 20, Madrid (28027) Spain; 34 1 267-5222.*

Manufacturer announcements

New England Digital has sold a Synclavier with a Direct-to-Disk digital multitrack recorder to Pete Townshend.

Solid State Logic has taken an order from HBO Productions for an SL 6000 E Series console for its audio post room.

Soundtracs has installed PC24 consoles in the home studios of Mark Knopfler and Alan Clark of Dire Straits.

Harrison Systems has installed a PP-1 post-production mixing console at Warner Hollywood's Re-recording Room A.

Mitsubishi has recently sold four consoles in Toronto. McClear Place has installed an X-850. Sounds Interchange has ordered two Westar consoles and one Superstar console.

Neve has installed V series consoles at Conway Recording Studios, Hollywood; Westlake Audio, Los Angeles; Cherokee Recording Studios, Hollywood; Track Records, Seattle; the Record Plant, Los Angeles; the Village Recorder, West Los Angeles; Gravity Recording, Nogales, AZ; Andora/Smoketree; Pacifique, North Hollywood; and Amigo Studios. RE/P

NEW PRODUCTS

Studer A807 recorders

Studer has added two models to the A807 line. The A807 VUK HS two-track recorder features three speeds (7.5ips, 15ips and 30ips) and is designed for console mounting with overbridge metering. The A807 4½-inch, four-track VUK is designed for broadcast applications and available only in the high-speed version with overbridge metering.

Circle (150) on Rapid Facts Card

Siemens Crossmatic D audio routing system

Crossmatic D contains 16/32-bit microprocessors and computer control for single-user systems. Up to six levels, including stereo, SAP, feedback monitor or intercom, are available. It can be controlled by the Centronics interface or data terminal control via an RS-232 interface. External host control is achieved via an RS-422 interface.

Circle (152) on Rapid Facts Card

Soundtracs In Line consoles

The series comes in a 36-input version, the IL3632, and a 48-input version, the IL4832. Both are designed for track laying and basic mixing, and include dual-line inputs, four-band parametric EQ, eight aux sends and TT-jack patchbay. Monitoring may be either PFL or in-place solo.

Circle (159) on Rapid Facts Card

Sony VHF 400 wireless mic system

Based upon frequency synthesis, the system transmitter can operate on 48 separate frequencies, and the receiver can handle all 168 assigned channels. A push-button system with LCD readout is used for tuning to desired frequencies and retuning if interference is encountered. Diversity reception is optional, and various antennas and accessories are available.

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Dowty Seals EMI protection gaskets

The gaskets are designed to protect equipment from EMI and RFI. Custom-designed shapes and 2,700 standard O-ring sizes are available. Signal attenuation levels are 120dB or greater, and all gaskets and seals are measured by applying a calibrated signal at 30MHz, 100MHz, 300MHz and 1,000MHz.

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Mag-Tags are magnetic strips bearing instrument names commonly used in music production and are intended for marking channel assignments on mixing consoles. Optional adhesive-backed strips allow use of the tags on aluminum and plastic chassis. Blank labels also are available for artist names.

Circle (158) on Rapid Facts Card

Dynacord ADS sampler

The unit comes with 2Mbytes of memory, expandable to 8Mbytes for almost 100 seconds of sampling at 44.1kHz. It features 16-bit, phased-locked stereo sampling; 24-bit, internal processing; and 20-bit, double-oversampled digital filters on all eight outputs. Real-time monitoring through an A/D converter allows users to know how the samples will sound before the sampling takes place.

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Spatial Sound SP-1

The unit is a sound processor that moves and positions sound in a two- or three-dimensional space and also produces effects associated with sound dynamics. It can process live or recorded sounds and can handle from two to eight speakers. In addition to live applications, the unit can be preprogrammed for tape or MIDI control.

Circle (164) on Rapid Facts Card

Imagine Computers SMPTE City

SMPTE City is a SMPTE-to-MIDI converter designed for first-time SMPTE users. The rack-mount unit's features include reading and generating all four SMPTE formats, LED display showing SMPTE or bar number position and 11 programmable functions. Reading range, −30dB to 0dB, allows the unit to be used with cassette-based tape machines.

Circle (170) on Rapid Facts Card

Schoeps MK 21 mic

The MK 21 is a subcardioid mic whose frequency response is unaffected by the angle of sound incidence. In the forward hemisphere, directional effect is hardly discernable. At 90°, response is down 3dB, and for rear-incident sound, level reduction is 10dB. In spot applications, off-axis sound sources can be picked up without coloration.

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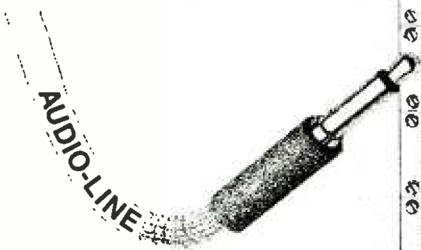

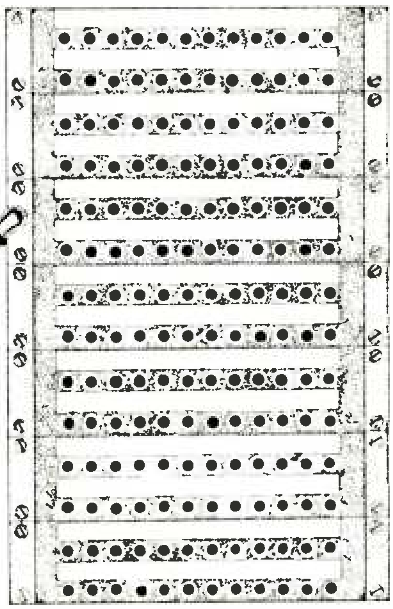

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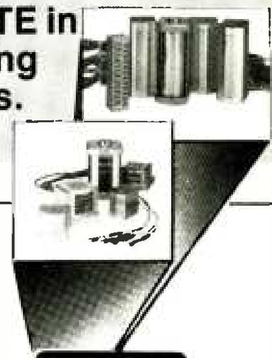

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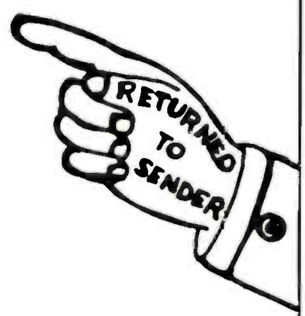

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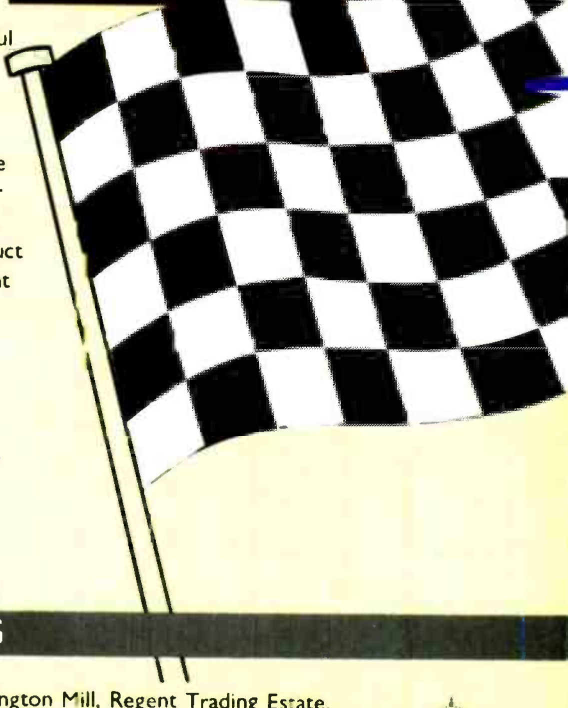

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