Recording ■ Engineering ■ Production

R E P

The Pro Audio Applications Magazine

September/October 1993
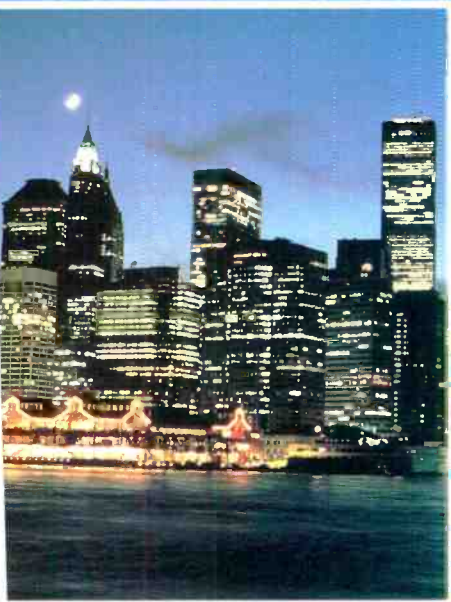
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We've Always Talked Performance. Now Let's Talk Price.

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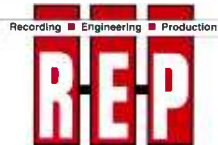*(Photo courtesy of New York Convention & Visitors Bureau)*

Craig Connally, president of Neotek, is the subject of the REP Interview.
(Photo by Tony Harris)

Tracking, scoring and mixing projects are planned for Neve I, one of Record Plant's newest studio suites. The suite is a self-contained environment with a private lounge overlooking the Record Plant atrium.
(Photo by Ed Freeman)

From the Editor

R·E·P Returns

R·E·P magazine returns to the recording industry just in time for the 95th AES Convention. It seems that AES and R·E·P have traditionally been focal points of the industry. How appropriate.

R·E·P's rebirth is made possible by Key Markets Publishing, a 20-year old company with a fresh outlook on the industry. In many ways, R·E·P will not change from the R·E·P you grew to know and love. We will continue to provide applications-based editorial coverage appropriate to professional audio; and we will continue to cover current topics that affect the way you do your work. The differences you will see in the new R·E·P involve the attitudes with which we approach our role in the industry.

The role of editor should be transparent. My job is not to choose the technologies that are appropriate for you. It's my job to provide you with up-to-date information regarding the recording industry and to offer you an open channel whereby you can respond to changes in technology or in our industry, either positively or negatively. There are enough issues which affect the recording industry without editors making an issue out of a technology's merits, or lack of merit.

What can you expect from R·E·P? Expect honest features and columns from top writers and prominent individuals in professional audio. Expect fair and accurate reviews of the latest pro audio equipment. Expect timely reporting on the ever-changing issues that affect us. Expect to give us a piece of your mind. That's right. We're talking about a *real* readers' forum. *Letters and Views* offers you a chance to say what's on your mind. With *Letters and Views*, our intention is not necessarily an avenue for congratulatory correspondence (we accept any and all praise), but it is an avenue for open exchange of ideas.

This issue not only marks our return to the industry, but it is also the AES show issue. *Audio in the Age of Multimedia* is the theme for the 95th Audio Engineering Society Convention at New York's Jacob K. Javits Convention Center, October 7-10. Our coverage includes technical program and workshop highlights and an AES show product preview.

R·E·P welcomes Larry Boden and David LaBarre. Larry's column, *A/D – All Digital,* will cover everything (digitally speaking) from DATs to CDs, and then some. David's column, *The Listening Environment,* will address many issues affecting the business and technology of recording studios. Both columns invite you to write or FAX with your comments, suggestions or gripes.

This issue includes an interview with Craig Connally, president of Neotek Corp. in Chicago. He speaks openly about the state of the industry and console design. Also, Carlos Chafin challenges the bigger-is-better philosophy; two products, the Fostex D-10 and the BBE 862, are reviewed; and business writer, Alice Maddox provides a profile of Dave Mancini, owner of Devonshire Studios.

The next issue of R·E·P will have more features covering some very interesting topics. *R·E·P Handbook* returns as does *Studio Update.* ▲

Jack Smith

When employing the new Fostex D-10 you'll work with DAT easier and faster than ever due to a host of savvy production features. Auto Cue and Instant Start are great examples. Locate with either A-Time or Pro-R

You can use a pair of recorders for fast, precise assembly editing procedures. The universal GPI ports of the D-10 provide the control between the two machines. On-board RAM and Auto Record will do the rest.

FAST, ACCURATE AUDIO PRO SEEKS CHALLENGING PRODUCTION POSITION.

Time to a given cue point. Then, program the D-10 to start automatically the instant audio modulation begins. With Auto Record, hands-free edits are under microprocessor control, and punch-in recordings can be repeated accurately.

RAM Scrubbing and Jog/Shuttle give you 'analog style' reel rocking with familiar video controls for pinpoint cueing.

A 10-key pad lets you store and recall up to 100 cue points—providing extensive autolocation within program material. The full array of DAT IDs is available on the D-10. Record or play up to 799 P-NOs with auto or manual numbering, re-numbering and skip programming.

A dual configuration is practical because a single D-10 with its full complement of standard features is less than $3000. There's no hidden cost—even the remote is included.

The universal GPI ports can be used for extensive custom external control—such as fader start, event out, transport commands, etc. In addition, two optional slots are available for future upgrades.

For fast, high quality compilation or re-sequencing of program material, for precise assembly editing, and as a source machine, the Fostex D-10 is the ideal candidate.

References available upon request. Call 1-800-7-FOSTEX

FOSTEX

A Sharper Focus

Welcome to the first issue of *R·E·P* published under the new ownership of Key Markets Publishing. When word first got out about our acquisition, we heard from many readers and advertisers wishing us success. Thank you for those calls. They are really appreciated.

Whether you're a long-time subscriber or a new reader, you're probably wondering what the editorial direction of the "new" *R·E·P* will be. I know that *R·E·P* has had an excellent reputation over the last 23 years, and we want to build on that to make a good magazine even better. To be effective, a magazine must be targeted to the needs of its readers. During the last few years, *R·E·P* has covered a broad spectrum in the audio industry. We will devote our editorial product to recording and sound reinforcement. I believe this sharper focus will provide recording engineers/producers with a more useful publication — both in editorial and advertising.

With the possible exception of cereal or laundry detergents, magazines are among the most analyzed and surveyed of products, especially start-ups (or in our case, restarts). It's inherent that our first few issues may not meet all of your expectations. I only ask that you bear with us because our commitment to you is to make *R·E·P* the finest publication in the pro audio industry.

Thank you for your valued and continued support. My pledge to you is to be responsive to your needs. Please don't hesitate to contact me, or our editor Jack Smith, to let us know how we can help you through the pages of *R·E·P*. ▲

Of mic preamps, channel modules and ergonomics

From: Gary Falk
Falk Recording Service
Louisville, KY

Congratulations on the NEW R·E·P — I am very pleased that you are back on line, and I look forward to great dialogue, reviews and point/counterpoint as exemplified by your magazine.

Thank you for our recent discussion concerning console and mixer design — a subject near and dear to my heart. Since my *R·E·P* article of March, 1992 on the design of the Falk console, we have utilized the console on a day-to-day basis, and we have had excellent service out of this design with negligible down time.

I strongly defend certain considerations in console design, and I sometimes wonder how production consoles go through R & D, manufacture and to the customer without regard for these items. I shall briefly touch on some at this time.

1) Microphone preamplifier design:
I know that there is a lot of discussion on this subject, and cost can be a factor, but I am convinced that the very best designs must utilize a transformer. There was a period of "transformer bashing" a few years ago, and some of it was justified — mainly because of the anomalies present in inexpensive transformers. Most audio transformers being manufactured by dedicated audiophiles — Jensen, Reichenbach, Beyer, etc. have virtually eliminated the strident characteristics and unwanted resonances of earlier designs. The advantages of a transformer lie mainly in two areas — the voltage gain obtained by the turns ratio, thus reducing the amount of ACTIVE gain required and the galvanic isolation from the outside world which offers freedom from problems encountered in direct-coupled designs.

2) Access to channel modules, subassemblies, etc:
Individual channel modules should be removable for periodic service without unnecessary constraints, such as removal of connectors from the rear apron and dealing with complicated physical maneuvers. The surprise to me here is that this is not necessarily a cost factor, but simply some prudent engineering practice to make life easier for the end user. In our design, there are three modules per channel. The two modules with active electronics are removable with two thumbscrews each. They then plug into the mainframe by way of a pigtail wire harness and Sub-D connector, gold plated on both mating surfaces. I am sure there are other ways to approach this, but in my experience with production boards in my 23 years in pro-audio I have yet to see this taken seriously in consoles selling for less than $100K or so.

3) Ergonomics:
This has to do with access to control functions, redundancy of physical movement, etc. In my opinion, control functions from console front to rear should be based on usage — the most vital to the front. Channel assignments are fine at the rear — up near the meter panel. The fader, channel mute switch and preamp gain pot should always be at the front of the channel apron. That the gain pot ended up at the back of the channel module was a bad idea that somehow gained acceptance.

It is hard to be brief concerning all of this. There are many things that could be said that require some discussion — like the necessity for conductive plastic potentiometers in all signal path areas and gold plating on toggle and push button switches. Surprisingly, most of these things add very little to the manufacturing cost when amortized over a number of units being produced. ▲

Editor's note: Facts stand on their own integrity, but opinions are open to debate. The recording industry has never suffered for lack of opinions. An open exchange of ideas is healthy for the industry. I encourage you to write to us to express those ideas. Perhaps you will disagree with an article or review, or you may feel a point needs clarification. We want to hear about it. Write to us, and we will share your views with the rest of our readers. Send letters to R·E·P Letters and Views, P.O. Box 5867, Rockford, IL 61125; or FAX 815-229-4086.

Achy-Breaky DAT
Or the Little Format That Could

This column is about the state of digital audio and its sonic nuances as well as the practical use of binary streams in remixing. This issue I am focusing on Nashville, Tenn. and DAT in particular.

I tell you this up front because I've always had great respect for the purity of sound common on many country music records. Some are astonishingly clear, yet never get the respect or good reviews based on their overall sound.

During the '60s and '70s, I plied my trade as an engineer in Music City, U.S.A. Engineers like Gene Eichelberger, Claude Hill and Garth Fundis were pushing the "clean" sound even back then.

The current unprecedented popularity of country music does not surprise me but its magnitude does. The army of devoted engineers and the advanced studios all over town deserve their day in the sun. Enough set up; on to substance.

It was 100 degrees when I arrived in Nashville to remix my latest jazz album with tracks and vocals originating on the East Coast. I should mention that the city's annual Fan Fair was in full swing, and I was almost run over by a tour bus from Montana.

The album's vocalist, a true cosmopolitan East Coaster was in town and was prepared to repair any last minute vocal tracks. We chose Nashville because our remixing job was massive, and we were facing an ambitious release date. We set up shop at Masterfonics, the brainchild of Glenn Meadows.

"Well, Glenn, what should we mix to?" I asked. "I want that clean sound and high definition you got on the last Vince Gill album."

"No problem, Larry, we got DAT machines all over the place."

"DAT? Listen pal, this is first line product like Ella Fitzgerald, Dionne Warwick or Bette Middler. What about a Sony 1630, Dash machine or an X-80? DAT's for budget stuff."

"Wake up, Larry. It's 1993. Over 75 percent of the first line product we move through here comes in on DAT.

We've modified the machines to incorporate Wadia A/D converters."

"I've heard about dropout problems on DAT — had some myself."

"No sweat. You'll mix to two DATs simultaneously. Then we'll move the product off to the digital audio workstation (DAW) of your choice."

The point here is that DAT has not only made a substantial inroad into the project studio market, but has gone all the way to the top, handily displacing many other digital formats. And all this from what was envisioned as a home format.

Glenn swears he gets 75 percent of his first line product in on DAT. A quick check of other mastering houses in Music City proved this was not just an isolated case.

Carlos Grier of Georgetown Mastering told me 85 percent of his studio's front line product comes through the door on DAT. M.C. Rather of Custom Mastering claims 96 - 98 percent DAT. Most impressive.

Except for a few lone souls, outboard A/D cards are the rule. There are three main choices: Apogee, Wadia and Drake. The Otari, Sony and Fostex machines also got high marks.

The three mastering gurus agreed that machine-to-machine interchange problems and dropouts occur but not excessively.

I still felt strange mixing a $100,000 product to virtually the same machine I have in my den. My vocalist was unimpressed when finding out her many hours of vocal effort were ending up on something that could fit in her purse. I gave her some technobabble on "current state of the art."

We mixed this job without experiencing a dropout.

The moral of the story: The business needs a robust, inexpensive, two-track format that works. DAT was in the right place at the right time. Enjoy it.

I can't close without a word about the expected longevity of DAT masters. Not one of the three mastering houses archives finished product on DAT. Rather uses the Sony 1630, Georgetown prefers the Sony Dash while Meadows uses the JVC 900 or 30 IPS Dolby SR.

I'm off to my next project. It too will likely end up on DAT, just like yours. Now all we need is a standardized transport that facilitates switching out A/D and D/A sections. How 'bout it manufacturers? And readers — can I get some support? Write or FAX me in care of *R·E·P* magazine.

> ## DAT has not only made a substantial inroad into the project studio market, but has gone all the way to the top

Mixing a jazz album of Gershwin and Porter tunes during Music City's Fan Fair week was indeed strange. Folks from other rooms at Masterfonics would poke their heads in, catch a few bars of "Fascinating Rhythm" and ask us if Reba, Tanya or Roseanne Cash had heard it and might do it. I don't know if they know the song or not, but I sure felt strange strolling past the Country Music Hall of Fame with the master tape of 21 songs in my shirt pocket.

Let's hope I get a deal. Maybe that bus of tourists from Montana like Jazz...

Next issue: Doug Sax's new all tube mike preamp. ▲

Why not get everything professional Digital Audio Workstations have to offer?

Create your best part out of 10 recordings and use it in five different places... record 100 takes in the same place, on the same track... synchronize sound FX to a picture in less than a second... erase a word and bring it back a week later... and do all this on a machine that gives you instant access with incredible editing speeds, the sound quality you want, and the reliability you can count on, project after project.
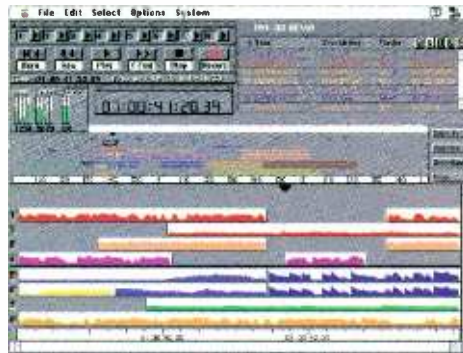
Optional DM-80-S Multitrack Manager control for the Macintosh®— easy to learn, fast to use.

the Word.

Problem is, many DAWs just can't give you all of these things.

Here's why:

Problem: Some systems use a single computer for graphic user interface *and* for audio information processing. This slows the whole system down, and makes you wait.

The Answer: A workstation that separates the hardware "engine" from the user interface computer. This way, audio information processing is not slowed down by the user interface, and the interface computer operates without the load of additional tasks. And you save time and money.

Problem: A system crash in the middle of a complicated project could cost you as much in time lost as some of the less professional DAWs are worth. You should expect reliability you can count on from your DAW, and get it.

The Answer: First, call around. Check the reputation of DAWs. We have DM-80s working every day all over the world in high volume production environments. Our customers will tell you they're reliable.

Problem: Some systems don't give you a user interface that's familiar and easy-to-use.

The Answer: Buy a DAW that lets you choose either a dedicated hardware remote that gives you familiar tape recorder controls, or computer software control that is simple to understand and easy to operate.

Problem: Some DAWs are the weakest link in the audio chain. The DAW you buy should deliver sonics as good or better than any piece of audio equipment you own.

The Answer: Choose a DAW from a company that knows professional audio, not just a company that knows computers. But most of all, ask audio professionals who own one.

We believe you'll find all these answers and more in the Roland DM-80 Digital Audio Workstation. Call us at (213) 685-5141, ext. 337, or FAX (213) 722-0911 for a brochure. Or better yet, schedule a demo. You're going to like what you hear.

"The thing is just impossible to screw up! The DM-80's non-destructive editing is the only way to go—revisions are extremely easy, and very quick."

— DAVID ESCH
DIRECTOR OF COMMERCIAL PRODUCTION
WPNT CHICAGO

"Edits fast, sounds great, and doesn't break. The DM-80's multilayering capabilities and simultaneous recording on all tracks gives us the flexibility to do whatever we want."

— JOEL VALENTINE
PRESIDENT
21ST CENTURY SOUND DESIGN, HOLLYWOOD
("NORTHERN EXPOSURE")

the Word on the street.

"The first DM-80 paid for itself in two months, so we bought a second one... I like the editing ease—no waiting. It's my 'word processor' for audio."

— MICHAEL STANTON
COMPOSER/ARRANGER
AUDIO PRODUCTIONS, INC., NASHVILLE

"The DM-80 is our secret weapon! It's a very enabling technology."

— MAZ & KILGORE
GRAMMY NOMINATED PRODUCERS
NEW YORK

"What goes in... comes out! And the sound is wonderful. Easy editing, plus easy sync to video lets me concentrate on the creative aspects of the production."

— RICHARD FREITAS
EMMY AWARD NOMINATED AUDIO PRODUCER
THE DOVETAIL GROUP, CONNECTICUT
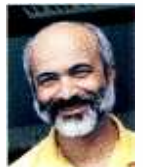
"Operational speed and ease of editing make the DM-80 the greatest thing since sliced bread! We use it for all our production."

— VINCENT FUMO
SENIOR EDITOR
INDEPENDENCE BLUE CROSS, PHILADELPHIA

"The DM-80's power and speed allow me to complete a half-hour television documentary every week. As an investment, the DM-80 has proven to be a very smart choice."

— JIM HEFFERNAN
EMMY AWARD WINNING SOUND DESIGNER/PRODUCER
VIDEOSMITH, INC., PHILADELPHIA
("1ST FLIGHTS WITH NEIL ARMSTRONG")

Roland
ProAudioVideo
For more information, circle 3

by David LaBarre

The Magic of Technology

Intelligent thinking requires only using your brain, I think, therefore, I am. Being self-aware, having perception or feeling and being aware of your environment are the prerequisites to be considered a sentient being. The problem here is convincing others that you are sentient. When we fail to communicate, we think to ourselves, "At least we are trying." The other person is thinking, "You sure are." More common in the '90s is looking out for number one, therefore we have the tendency to blame the other person for not understanding our inability to communicate, then walk away thinking the other person is not sentient. Empathy must be used if we are going to be even moderately successful in communicating.

In recent years educated people have discovered that we only use a small portion of our brains, and to further complicate the matter, this may be either the right or left brain. It seems that right brained people are the creative (dreamers or artists) while left brained people are the practical (mathematicians or engineers). While it is said that opposites attract, it is certainly not true here. Have you ever noticed in the studio (especially when there is a problem) that the artist, producer and recording engineer walk out and the maintenance engineers walk in? The recording engineer may stay for a few moments to describe the problem which the maintenance engineers think they understand, and then he or she quickly rejoins the group. The maintenance engineers are left alone to resolve the problem. The recording engineer assumes the problem has been described adequately, and the maintenance engineers assume they can hear or measure what the recording engineer is talking about. The unfortunate thing about assuming is that it often makes an ass out of you and me. Therefore, intermittence can be defined as "working" for the practical and "not working" for the creative. This is the point where the inanimate is the contention for the animate and the studio loses money.

It seems that few things in life are absolutes. This accounts for the fact that we don't have more dreamers and mathematicians. Instead we end up with the creatively practical or the practically creative, thus accounting for recording engineers and design engineers.

Recording engineers and musicians seem to wish there were more standards so they could reliably build interfaces to more creative sounds and consistently use and/or modify those sounds between environments to fit a performance with feeling or mood for a specific arrangement. These standards would also establish a jargon that is understood by both the recording engineer and the design engineer.

Many design engineers feel that standards, once understood and the bugs are worked out, are too restrictive in a quickly changing world of technology. While in the short term they help, the long term has shown these standards hold us back and keep us from progressing in a timely manner.

Coming Soon!
ABRACADABRA
The Ultimate Analog/Digital Wand

They are also time consuming and difficult to agree upon.

Today, many companies are forming alliances with other manufacturers or software venders to support an emerging technology instead of spending the time and resources to develop standards. The winner is determined by the greatest number of companies supporting the technology.

The problem with this idea occurs when alliances compete for similar technology that does not work similarly or communicate with existing technology or allow for an interchange of end product. The market has trouble figuring

out, not only if it is right for them, but also if the technology will endure long enough to warrant the investment.

Still, end users in general wish they understood the ins and outs of a new technology, and most of the designers of the technology wish the end users were more knowledgeable so they could communicate better descriptions for what they expect, want and/or need.

We have created a platform. On second thought, we have created a number of these and they have a difficult time communicating. A springboard, that's the ticket (only one of those are allowed per pool) where manufacturers, design engineers, maintenance engineers, producers, recording engineers, artists, etc. can communicate with each other. We believe this could be accom-

> I personally do not believe the statement that "Any technology so advanced is indistinguishable from magic."

plished forum style where you, the reader, write to us with your questions and problems. Each issue, like music in the air, we will select a topic that is timely and ripe. We have also discussed having a questions and comments column, rather like "Dear Abby" so everyone can be answered and heard. In short, we want to be your magazine – the one you pick up first and talk about the most.

Just off the top, I can think of a number of topics: how to operate a studio (anywhere) at a profit, building a studio in difficult times, studio etiquette, neutrality defined and why we should care, establishing a neutral reference, the three grounding methods, problems with ultrasonic oscillations and sampling frequencies in sonic performance, why switching power-supplies don't seem to work for professional

audio, how will multimedia affect me, why thin-eared engineers have trouble maintaining a three-dimensional image out of the studio, why did mastering become the final creative step, some of the general concepts of mastering engineers, what recording engineers need to know about a new studio, the role of the producer in and out of the studio.

I believe each of these topics can be expressed in non-technical form so that everyone can understand them. However, to get me to write, you must tell me what you want or need to know. Write to me in care of *R•E•P*. Please remember to give some information about yourself, what you do, where you work, etc. If I cannot answer your questions or feel unqualified to write on a topic, I'm sure we can find someone (poor slob) who is and will. I personally do not believe the statement that "Any technology so advanced is indistinguishable from magic."

Next issue, the fundamentals of neutrality.   ▲

David LaBarre is a contributing editor to R•E•P and an Orlando, Fla.-based consultant and freelance writer.

Craig Paller has joined Electro-Voice as market development manager and will provide technical and marketing support for EV's music product dealer network ... Crest Audio has appointed Sam Spennachio to regional sales manager for its console division and Lisa Vogl to marketing services manager ... Fostex has expanded the responsibilities of VP of sales, Steve Cunningham to include marketing, Bob Veri has been named as executive VP and David Oren is now VP of product development ... Derek Covin was promoted from shipping manager to U.S. sales administrator at Alesis and Jeremy Clarke was promoted from Customer service to ADAT Worldwide Network coordinator ... Bernette T. Harley joins Switchcraft as human resources representative ... Otari has named Steve Zaretsky, formerly of New England Digital, as sales executive for the newly established Northeast regional sales office in New York City ... Audix Corp. has announced the addition of three reps: Ross Associates of Boston Mass., GMS Sales of Minneapolis and Bob White Associates of Cleveland.

95th AES Convention:

Audio in the Age Of Multimedia

If you'd like a way to improve yourself professionally by gaining knowledge from top experts in the audio field, then the 95th AES Convention and Exhibition is the place to be.

A top array of speakers, touching on many vital topics, will be in New York at the Jacob K. Javits Convention Center October 7 - 10 to present information and ideas for the audio professional. Many of the technical papers and workshops/seminars will address the theme, *Audio in the Age of Multimedia*, with the realization that the audio profession can no longer be regarded as independent of other modern scientific disciplines such as video, computers and other forms of telecommunications.

The technical programs will be presented under the main categories of:
- Digital Music
- Test, Measurement and Specifications
- Psychoacoustics and Subjective Assessment
- DSP Theory and Application
- Multichannel and HDTV Sound
- Auralization — Progress in Modeled and Reproduced Environments: New Techniques for Transaural and Binaural Reproduction and Room Modeling
- Multimedia
- Perceptual Coding
- Microphones and Loudspeakers
- Architectural Acoustics
- Digital and Analog Electronics
- Topics in Acoustics and Auralization

Workshops:
- Computer Control of Sound Systems
- Professional Practices and Job Strategies in 90s and Next Century
- Audio in the Age of Multimedia
- Auralization
- The Art of Mixing

- Perceptual Encoding
- Use of Computer in Audio Education
- Multichannel of Sound Production Techniques
- Grounding, Shielding and Interconnections in Analog and Digital Audio Systems
- Rigging Loudspeaker Systems

Within these categories are many additional subtopics covering most everything the audio professional wants and needs to know. All of these topics are presented by leaders in the field.

Technical Tours/Reception & Awards Banquet
Six tours are scheduled to give you an opportunity to observe and discuss audio in action — everything from backstage visits to world-famous recording studios to guided excursions to nationally recognized broadcast stations.

The Reception and Awards Banquet will be held on Friday evening, October 8, at the New York Hilton Hotel. The banquet will highlight the presentation of the Society's coveted awards to those whose outstanding contributions to audio and to the AES merit recognition by their peers.

The entire four-day program is designed with the audio professional in mind. Each aspect has been carefully planned to make your trip to AES Convention a true educational experience.

Exhibition
Over 300 exhibitors are expected this year. This will give show-goers a chance to view, in one convenient location, hundreds of products that have gone into production since last year's show.

The following listing presents some of the products that will be included at the show. For more information on these products, circle the item number on the Reader Service Card.

AKG Vintage TL MIC (Booth 318)

The Vintage TL condenser mic from AKG Acoustics is a new version of the transformerless C414B/TL with a cornerstone of AKG history built in — the capsule from the C12 recreated from its original design in the 1950s.

The Vintage TL is a 1-in. dual diaphragm pressure gradient microphone with four polar patterns: cardioid, hypercardioid, omnidirectional and figure eight. Sensitivity: 12.5 V/Pa. Bass cut: 12dB/octave, 75 Hz or 150 Hz. Preattenuation pad: -10dB or -20dB, switchable.

For more information, circle 61

Alesis Monitor One™ Studio Reference Monitors (Booth 444)

Alesis Corporation's Monitor One is a high efficiency, two-way near field studio reference monitor pair designed for professional and project recording. The Monitor One uses a proprietary 6.5-in. low frequency driver with a mineral-filled polypropylene cone, a linear rubber surround and a 1.5-in. voice coil wound on a Kapton former.

The Monitor One stands 8.5 in. high (15 in. wide) and boasts a power capacity of 120 watts (program) and a frequency range of 45Hz to 18kHz (±3 dB). A 1-in. silk-dome high frequency ferrofluid-cooled driver is also included and fed by a crossover network at 2500 Hz. The outside of the cabinet is covered with a polymeric rubber coating which provides a non-slip surface for stable console meter bridge mounting.

For more information, circle 62

Apogee Sound CRQ-12 Parametric Equalizer (Booth 1044)

The CRQ-12 has 12 parametric filters, each adjustable in three ranges from 20Hz to 20kHz, four shelving filters, and four adjustable 12 dB/octave filters, configurable in three distinct modes of operation. These modes are 6/6, 6/12, and 12/12, and offer flexibility of internal signal routing to the unit's four output level controls. The unit reportedly has a dynamic range of 115dB and distortion less than 0.003 percent at +21dBu.

For more information, circle 63

Beyerdynamic MC 834 Transformerless Condenser Microphone (Booth 519)

The MC 834 is a condenser mic designed to reproduce vocal and instrumental music. The cardioid pickup pattern is retained throughout the frequency range, allowing high gain levels without feedback and producing a uniform off-axis response. The MC 834 exhibits flat frequency response from

20-18,000 Hz. The microphone has a three-position, switchable, low-frequency filter and four stage pre-attenuation, plus an integral blast filter for suppressing pop and hiss noises.
For more information, circle 64

Cadac Live Mixing Console (Booth 466)

The Cadac J-Type live mixing consoles have a frame design which allows any module to be placed in any position. The J-Type is automatable with versatile output options. Principal features of the J-Type include: VCA channel faders controlled by any one of 15 dc master faders; maximum of 16 sub and 32 matrix group outputs; 12 auxiliary group outputs; central control module with local memory and interfaces for up to two IBM-compatible PCs; optional line-input programmable routing modules, which provide routing to the sub and matrix outputs.
For more information, circle 65

Cambridge Signal Technologies (Booth 1050)

The model AEC 1000 Acoustic Environment Correction System is a two channel digital filtering unit with a 2544 tap FIR filter covering the audio spectrum and a measurement and analysis system which corrects in both time and frequency domains.

The AEC 1000 compensates for loudspeaker/room interaction in control rooms and auditoriums. The unit automatically measures direct sound and room reflections; adaptively designs an inverse FIR digital filter; and compensates separately for direct response and delayed reflections.
For more information, circle 66

Crest 7301 Professional Series Amplifier (Booth 614)

The Crest 7301 Professional Series amplifier is a compact power amplifier specifically designed for use in bi-amplified systems, including stage monitor, studio monitor and front-of-

house (FOH) applications. The low frequency channel (Ch.A) uses class H operation; the high frequency channel (Ch.B) uses class AB operation.

Power specifications (20 Hz to 20kHz): Both channels at 4 ohms 940W for lows and 240W for highs. THD low channel <0.02 percent @ 800W, .4 ohms, 1k. THD high channel <0.02 percent @ 200W, 4 ohms, 1k.
For more information, circle 67

Crown's MRX Matrixer (Booth 741)

The MRX Matrixer from Crown provides the ability to reconfigure and repatch an audio system via computer, making it an automated, electronic patch panel. The MRX is a plug-in card that can run in the Crown IQ card cage or with any freestanding compatible unit and can be situated anywhere in the Crown IQ System loop.

Four MRX Matrixer cards are available: MRX-12 and MRX-24 master cards, and MRX-12S and MRX-24S slave cards. Up to 31 slave cards can be used with a single master card, but slave cards cannot operate as stand-alone units and must be accompanied by a master. When used with IQ Sys-Config software, the MRX can be made to reconfigure automatically.
For more information, circle 68

dbx 274 Quad Expander/Gate (Booth 318)

The 274 Quad Expander/Gate offers four independent channels of expansion or gating, front panel configurable as four mono, two mono and one stereo or dual stereo channels of processing. Each channel uses patented dbx VCA and RMS detection circuitry and features a combination of extremely
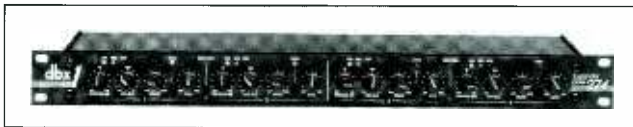
fast attack and new timing algorithms. A three-LED threshold status indicator on each channel speeds setup and provides visual indication of gate operation.
For more information, circle 69

The DDA QII Left-Center-Right Mixing Console (Booth 452)

The DDA QII mixing console offers three-channel capability and is available with 24, 32, 40, or 48 inputs. The console comes with eight mono subgroups, but can be specified with six stereo subgroups, four LCR subgroups or 16 mono subgroups — all directly assignable. Each input channel has four-band equalization, as well as eight full-time sends. Eight VCA/mute groups are also available.

Eight auxiliary buses are fed from eight individual controls. All sends are switchable pre/post in pairs, with internal links providing the pre-feed to pre- or post-EQ. All insert sends and returns are line-level balanced, while each insert can be either pre- or post-EQ.
For more information, circle 70

Focusrite Red 3 Dual Stereo Compressor and Limiter Module (Booth 958)

Red 3 is the third module in the Focusrite Red range of signal processing modules. Red 3 provides two channels of compression and limiting and offers transformer coupled inputs and outputs. The system uses the Focusrite VCA, featured in the previous ISA 130 and ISA 131 Focusrite Dynamic modules.

The limiter is a separate section with its own threshold control (switched from 0dB to +18dB in 1.5dB steps) and associated side chain circuits. The compressor offers continuously variable attack (from 300uS to 90mS) and release (from 100mS to 4S), along with gain variable from 0dB to +20dB. Two illuminated analog VU meters are provided to display the output signal level or to monitor gain change.
For more information, circle 71

Fostex RD-8 Digital Multitrack (Booth 235)

The Fostex S-VHS RD-8 is a linear digital eight-track, ADAT™ compatible recorder for audio post production facilities, jingle studios and project studios. It can be used for multitrack recording, sound file library storage and retrieval, and for synchronized playback and production.

The RD-8 features a tape format interchangeable with the Alesis ADAT. The RD-8's familiar user interface is similar to that of an analog multitrack recorder, reducing the user's learning curve. The RD-8's modular design and internal syn-
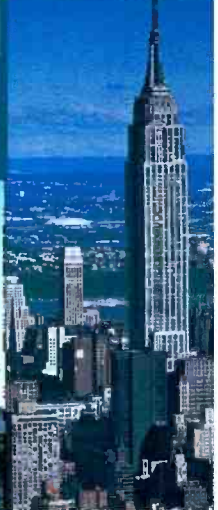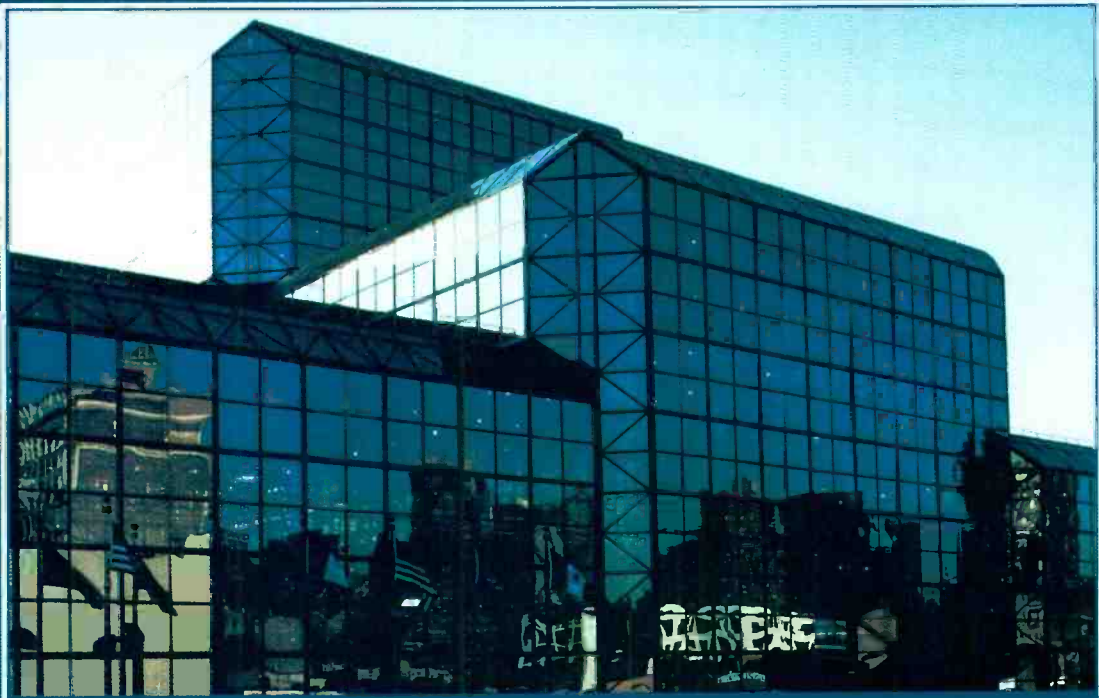
AES 95th Convention
AUDIO IN THE AGE OF MULTIMEDIA

October 7–10, 1993
Jacob K. Javits Convention Center • New York

Science emphasizes detached observation, objectivity, and logical deduction, but most who come away from the AES 95th and New York City this fall will find themselves feeling anything but detached —the combined dynamics of this city and this event simply won't allow it.
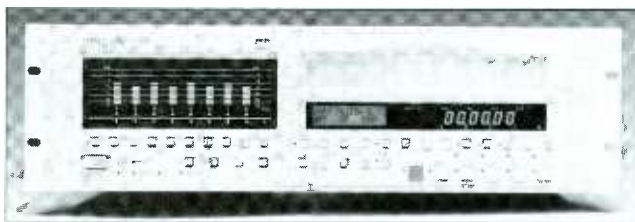
chronization enables the user to build sample-accurate 16-track, 24-track and larger recording systems. Its balanced and unbalanced audio inputs and outputs allow the RD-8 to be connected to other audio production systems.

The RD-8 features a built-in SMPTE timecode reader and generator. Sampling rates are 44.1 and 48kHz. The RD-8 features both pullup and pulldown to meet the requirements of analog and digital post production. Other features include RS-422 and MIDI machine control.

For more information, circle 72

"Q-Series" Graphic Equalizers (Booth 756)

Furman Sound's Q-series graphic equalizers includes six models: The Q-151, the Q-301, the Q-602, the Q-152, the Q-302 and the Q-541. These models offer constant-Q equalization. Sliders have electrically accurate, detented centers. All models have output level slider controls and four-LED meters.
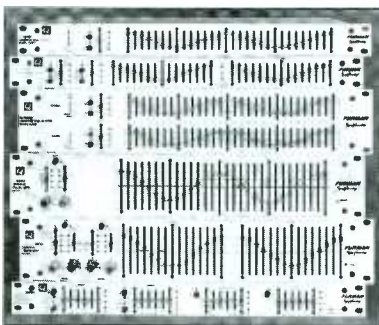
The top meter position indicates an overload condition, sensed at multiple critical points in the circuit.

The Q-541, is a special purpose model offering four channels of five band stereo equalization.

For more information, circle 73

JBL 4400A Studio Monitors (Booth 244)

JBL Professional's 4400A line of studio monitors are specifically designed for optimum performance in recording/playback environments. The 4400A line includes the 8-in. 4408A two-way compact monitor for the broadcast control room or smaller recording studio; the 10-in. 4410A three-way monitor loudspeaker vertical line array; the 12-in. 4412A three-way system for environments requiring maximum low frequency output from a bookshelf-sized monitor.

The low frequency transducers use Symmetrical Field Geometry (SFG™) magnet structures to minimize harmonic distortion and large diameter edgewound ribbon voice coils to provide good transient response characteristics and power handling. A titanium dome tweeter has been developed to further minimize distortion levels. The tweeters are oriented to create mirror-image left and right models.

For more information, circle 74

KRK Model 6000 Studio Monitors (Booth 958)

The KRK Model 6000 studio monitors are designed using aerospace materials for durability and optimum weight vs. strength factors. A 6-in. polyglass woofer and a 1-in. Kevlar tweeter reportedly provide linearity, depth of imaging and low distortion.

The Model 6000 features a passive, two-way crossover with a crossover point of 2.4kHz. Preliminary specifications include a frequency response of 62 Hz-15kHz (±3dB); nomi-

nal impedance of 8 ohms; sensitivity of 89dB (1 watt @ 1 meter); maximum power handling of 75 watts and maximum SPL of 106dB.

The Model 6000 is intended for the project studio, broadcast, remote, music and post-production recording engineers and producers.

For more information, circle 75

Klark Teknik DN3600 Programmable Graphic Equalizer (Booth 452)

DN3600, a digitally controlled analog equalizer, includes two special function keys that allow the user to alternate between an EQ curve, relative fader position for channel A or B or a combination of both. In the stereo mode, the two channels can be linked together to provide identical curves to two signal paths from one set of fader controls. In dual mode, these can be adjusted independently.

The DN3600 also offers 30 individual selection keys that correspond to each of the frequency bands. Each channel includes two notch filters and sweepable low/high-pass filters. Two rotary encoders allow frequency selection and level alteration.

For more information, circle 76

Kurzweil K2000S Sampling Versions, Options and Upgrades (Booth 608)

The K2000S and K2000RS are sampling versions of the standard K2000/K2000R. Current K2000 keyboard and K2000R rackmount owners can achieve the same sampling capabilities with SMP-K/R sampling option upgrades for keyboard and rack. The new K2000S and K2000RS feature pre-installed and tested SMP-K/R (keyboard/rack) hardware and Version 2.0 software. Also, each unit comes with two megabytes of pre-tested sample memory (upgradable to 64 megabytes).

For more information, circle 77

Manley 10-in. Nearfield Monitor (Booth 1208)

The Manley Nearfield Monitor uses the Tannoy 10-in. dual-concentric rolled rubber surround paper cone driver and the Mastering Lab crossover design. The non-resonant cabinet is constructed of 1½ in. thick MDF finished with black laminate and solid oak sides. Front-mounted controls allow continuously variable high frequency adjustment. Bi-amp and bi-wire capable.

For more information, circle 78

Micro Technology Unlimited Products (Booth 1066)

MicroSound Digital Audio Workstations are based on AT-class computers with non-proprietary drives. MicroSound allows Disk-Layering™ of up to 50 stereo tracks to be played

back simultaneously at any point in the project. Edits can be done/undone/redone instantly, even after restoring a DAT backup archive. Applications include audio for video, radio, TV, multimedia/CD-ROM, CDI development, CD/cassette mastering, and even speech research.

With MTU's Four-Track upgrade option featuring Floating Tracks™ and Disk-Layering™, any audio event reportedly can be placed on any track without bouncing.

For more information, circle 79

Millennia Media HV-3 Select (Booths 1268, 1270, 1272, 1274)

The HV-3 *Select* is identical in function to the standard HV-3 microphone preamplifier, but with 25 percent lower distortion and improved CMRR. The HV-3 *Select* provides front panel mic selection (standard or high voltage), phantom power, gain control and overload indication. LED status indicators are provided for all DC power rails. Balanced input stages implement an octal-matched discrete transistor array. Balanced ouput stages employ DC-coupled FET-based drivers. Circuit design is balanced and transformerless from input to output.

A set of Audio Precision factory test graphs, showing THD, CMRR, phase and frequency response curves, is included with each HV-3 *Select*. Shown at the Sam Ash booths.

For more information, circle 80

Nady 401 VHF Wireless MIC and Instrument Systems (Booth 335)

Nady's discrete channel wireless system gives four singers, guitarists, bassists — or any combination — freedom from the hassle of cables. The 401 system offers a four single-channel

VHF receiver in a thin rack space component, and four transmitters, in any combination of handheld, instrument or lavlier configurations.

The Nady 401 reportedly has 120 dB dynamic range and Nady's patented companding noise reduction.

For more information, circle 81

Nagra-D Four-Channel Digital Field Recorder (Booth 918)

The Nagra-D is a portable, battery-powered, four-channel digital tape machine for location recording. It offers 58 minutes of uninterrupted four-channel, or one hour 56 minutes of two-channel operation on standard 5-in. reel of ¼-in. digital tape, recording 4 x 24 bits.

The Nagra-D offers 24-bit, high resolution recording with playback at 18 bits and can be changed to 20 bits when lower power consumption A/D converters become available. An additional 4 bits per channel are dedicated to command

data presented at the AES/EBU digital I/O (hence, 4 x 24 bit recording).
For more information, circle 82

Neumann TLM 193 Large Diaphragm Condenser Microphone (Booth 1014)

The TLM 193 is a large diaphragm, double membrane cardioid condenser microphone designed for critical recording, broadcast and live sound applications. The capsule of the TLM 193 is the same large diaphragm capsule used in the U89/TLM 170 family. This unit is mated to the FET 100 circuitry of the KM 100 series.

The TLM 193 offers ultra-low self noise (10dB-A) high sound-pressure capability (140dB before overload), dynamic range (130dB) and a wide frequency response (20Hz-20kHz). The TLM 193 package includes the microphone, a swivel-mount stand adapter and a foam-lined wooden jeweler's case.
For more information, circle 83

Otari Offline BackUp Station for ProDisk (Booth 824)

The BackUp Station from Otari Corporation for the ProDisk line of digital audio workstations (DAWs) provides a solution to the time consuming process of backing up and restoring DAW files. The BackUp Station consists of a storage unit chassis which can hold up to five of Otari's removable disk drives along with an Exabyte model 8500 8mm tape drive. The system is controlled by any model Macintosh computer from a Classic II up.

The BackUp Station is supplied with either Otari's 480 meg or 1 gig removable disk drives, but the system has been designed to expand to other media, like Magneto Optical, with no additional modifications.
For more information, circle 84

Penny & Giles MIDI Management (Booth 1124)

The MM16 MIDI Management System is designed to simplify the programming and management of complex MIDI setups. The front panel features 16 endless-belt faders with integral LED displays, which provide status feedback for operators. Available as a 6U 19-in. rack mounting unit.

The MM16 offers separate merge and control ports each with six-way output splitters; storage of control assignments in up to 64 programs arranged in eight banks of eight; data values can be saved in up to 128 snapshot memories; the ability to augment the internal battery-backed memory by means of a plug-in 64k static RAM card; and the capability to dump the internal memory externally via MIDI.
For more information, circle 85

Ramsa WZ-DM30 Digital Multiprocessor (Booth 744)

The WZ-DM30 is a one-input/four-output 20-bit digital multi-processor. It offers compressors/limiters, a graphic equalizer, four-way crossover network and four four-band parametric equalizers. All controllable parameter settings can be stored in 16 event memories in each mode and can be retrieved instantaneously.

The WZ-DM30 can be connected to external MIDI equipment such as a synthesizer and an effector. It uses 20-bit digital floating A/D and MASH D/A converters to provide a typical dynamic range of 110 dB.
For more information, circle 86

Renkus-Heinz Speakers (Booth 155)

The Renkus-Heinz CE-3M/H is a high power, long throw cabinet with a 136 dB output level from 250 Hz to 20 kHz. The CoEntrant Waveguide Technology features a compound or CoEntrant throat section that takes the outputs of separate low (or mid) and high frequency devices and feeds them through a common horn or waveguide. This design creates a virtual point source at the apparent apex of the common mid/high horn assuring natural signal alignment and point source performance over a wide frequency range.

It features six 1-in. HF drivers and six 6.5-in. carbon fiber mid range drivers in a cabinet identical in size and shape to the CE-3. The CE-3 M/H was designed as a replacement for large format horns in high-level high-performance systems. An associated CE-3 LOW enclosure with dual 12-in. woofers extends system performance down to 60 Hz. The two form an extremely high power two-box 3-way system.
For more information, circle 87

Sabine FBX-900 Feedback Exterminator (Booth 1232)

The FBX-900 Feedback Exterminator uses digital signal processing to automatically detect feedback and determine its pitch. Then it assigns one of nine very narrow ¹/₁₀-octave adap-

tive digital notch filters to eliminate only the feedback in less than a second without muffling the sound.

The FBX can be placed anywhere in the sound system that a graphic equalizer might be used, but it is most commonly installed between the output of a mixer and the input of a power amp.
For more information, circle 88

Soundcraft Spirit Folio 4 Console (Booth 255)

The Spirit Folio 4 portable mixer is a four-bus live mixing console for front-of-house. The aux section offers eight aux sends; two pairs of aux sends are stereo; and there are four sets of stereo returns.

Folio 4 has 20 inputs — 12 mono and four stereo channels with three band mid-sweep EQ on the mono channels. It

offers two sets of dual bargraph meters. VU-reading bargraph monitors the mix output, and an upper PPM reading bargraph monitors the mix, tape return, stereo return, stereo aux's or group outs. Monitor outputs include PFL soloing on all channels and AFL soloing on the aux masters. Four group outputs allow grouped routing of signals to multitrack recorders.
For more information, circle 89

Tannoy System 6 NFM Reference Loudspeaker (Booth 644)

The Tannoy System 6 NFM studio reference loudspeaker is designed with a 6½-in. Dual Concentric transducer and features a tulip HF waveguide and layered, molded cone surround. The System 6 NFM Dual Concentric loudspeaker is a point source, phase coherent device offering constant directivity and linear, symmetrical, off-axis dispersion.
For more information, circle 90

M-1264 Rack-Mount Stereo Mixer (Booth 653)

The M-1264 rack-mount mixer from TOA Electronics can serve as a single mono or stereo mixer, or it can offer the same capabilities as those found on up to five independent mono mixers. The M-1264 has five sets of outputs, which have their own level controls: stereo main, stereo group, mono aux, stereo cue/monitor and a sum output that provides a mono mix of the main stereo output.

An automatic compressor is built in for each mic/line input which activates at approximately 3 dB below internal clipping. The M-1264 also includes up/down remote control of the motorized stereo and group ouput volume controls and a ramped 5VDC output for a remote level indicator.
For more information, circle 91

Trident Series 90 (Booth 832)

The Trident Series 90 console is fitted with 128 line or 176 line inputs in a 40 or 56 frame, respectively. Features include

automated switches, three independent signal paths per module, separate channel and monitor equalizers, ten aux sends with individual level and cut control, and dual VCA fader automation. Global Dynamics and Moving Faders are available as options.
For more information, circle 92

## UREI Platform Series Electronics (Booth 244)

PLATFORM uses a bus which allows summing and inter-module communications. Communication between mod-

ules is coordinated by a computer module which can be programmed to recall 100 memory settings of primary functions. Parameters such as volume, mutes, bypass and threshold can be controlled via a serial connection or switch closure.

The rackmountable 3U rack cardcage can hold up to eleven stand-alone or computer controlled modules. Six module units are available initially. Those are GATE1 Gate Module, COMP1 Compressor/Expander Module, P.EQ1 Parametric Equalizer Module, M.IN1 Input Module, OUT1 Output Module, and CPU1 and CPU2 Computer Interfaces.
For more information, circle 93

Valley Audio Products Model 730 Digital Dynamics Processor (Booth 558)
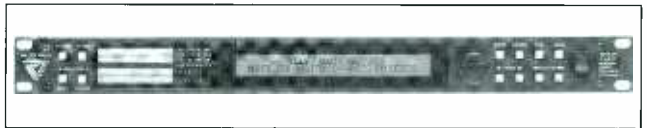
The Valley Audio Model 730 Digital Dyanmics Processor is a digital waveform-sensitive stereo compressor, keyable expander and gate, limiter and sibilance control with multiple numbers of threshold knees and segment ratios available at the same time.

The Model 730 supports digital and/or analog inputs/outputs, at any level or protocol:-10/+4 balanced/unbalanced, AES/EBU, S/PDIF, S-DIF2, OPTICAL, MIDI, RS-232, RS-422, etc. It also mixes analog and digital inputs together and feeds both A and D outputs simultaneously. All common sample rates are supported.
For more information, circle 94

Vega Four-Channel Private-Line Wireless IFB Transmitter (Booth 469)

The Vega RMT-14 is a four-channel, private-line wireless IFB (interrupted fold back) transmitter for IFB, PL talent cueing and audio monitoring. The transmitter is used with Vega PL-2 dual-channel miniature receivers and can be field-expanded to as many as four individual transmitters by adding internal modules.

The RMT-14 includes XLR inputs, 50-mW power output, crystal-control, LED-bargraph audio displays, input level control for each channel and an internal dual-voltage power supply.
For more information, circle 95

XTA Electronics RT1 (Booth 958)

The RT1 Real Time Analysis System-(RTA) combines ⅓ octave analysis, true SPL meter, RT60 analysis, and a swept frequency analyzer, in a 2U high 19-in. rackmount housing. The RTA offers simultaneous bar and peak displays with separately adjustable time constants and an independent SPL meter display. The LED display provides comparative readings between two memories, or between a real-time measurement and a memory.

The swept analysis measurements use ¹⁄₁₂ octave tracking filters above 63hz. The time/level mode displays historical information on peak levels with an adjustable viewing window of 62.5ms to 256 seconds per column providing over two hours of information on the LED display. The unit also includes A-weighting, rear panel mic input, manual or auto range control, and sine wave output.
For more information, circle 96 ▲

R·E·P Interview

by Jack Smith

This console maker speaks candidly about console design, project studios and the state of the industry.
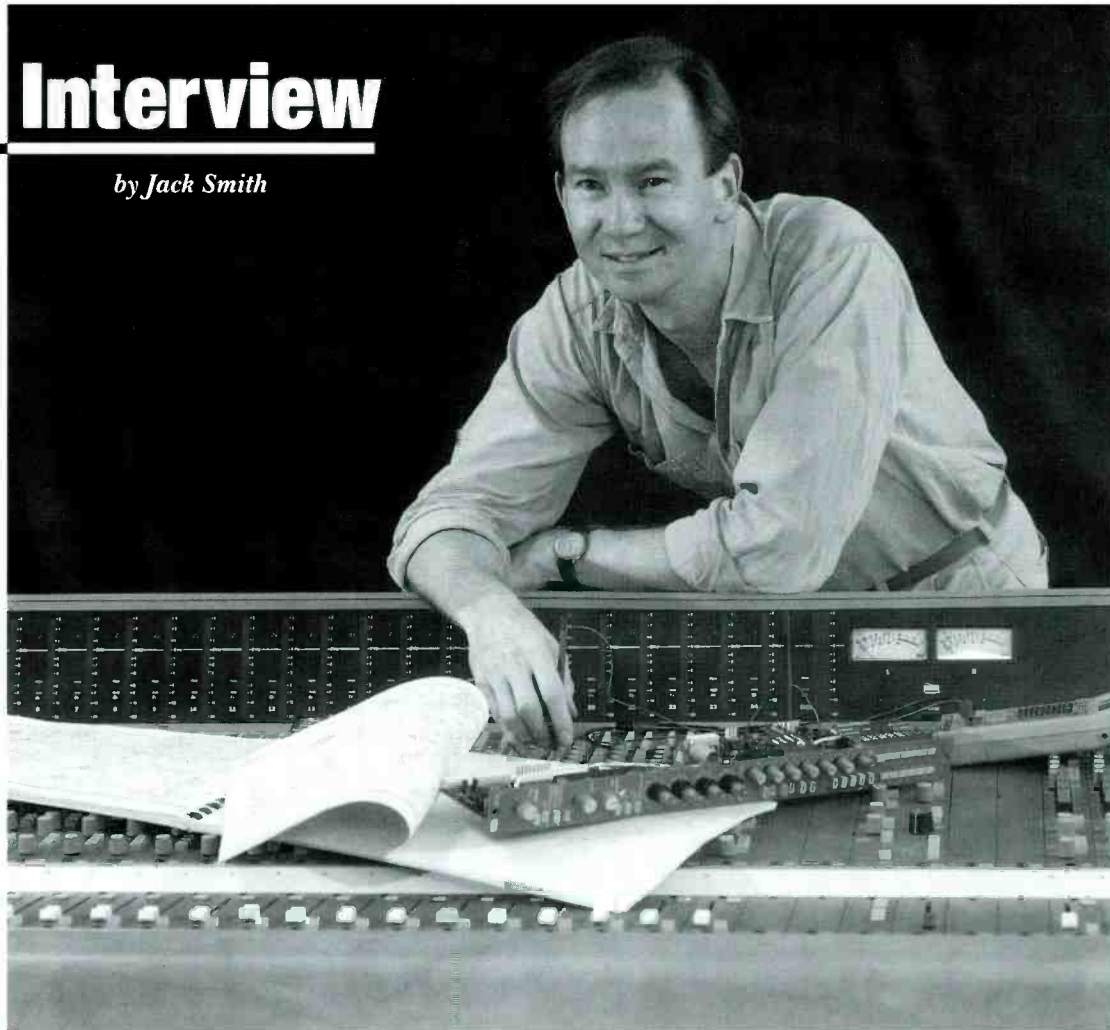
Craig Connally

Craig Connally started Neotek, Chicago, Ill., almost 21 years ago. The company was incorporated in 1981. Connally is the company's president and principal design engineer. However, his graduate degrees are in physical chemistry. Connally specialized in instrumentation, which gave him an electronics background. But his love of music got him into professional audio.

Neotek offers five console styles, ranging from the higher-end Elite to the Elan, which targets project studios. Its products have historically been known as artists' consoles, since Jackson Browne, Lindsey Buckingham and Chet Atkins have owned them. In recent years, the company has been heavily involved in the broadcast and film industries. The Esprit is a high-end broadcast console. The Elite is also available in a broadcast version. Connally says Neotek has not

been a fast-paced company. It has primarily been driven by engineering.

R·E·P: *Over the past 21 years, you have had your ups and downs. Give me a rundown.*
CC: The major down we had was when a large batch of switches failed. They failed about six months into application. They passed tests at the factory. [The manufacturer] muffballed it — refused to own up to it. I found by talking to other people in the industry, that they had the same problem. We owned up to it and have since replaced all those switches. That was, oh, maybe six years ago, or more. Other than that it, has just been slow and steady work. That was a major down for us. There haven't been any major ups.

We were at one time almost purchased by SSL, which would have greatly expanded our capital resources and certainly our distribution. [The prod-

uct line] would have fit nicely into their distribution structure because we take over at the bottom end where they leave off — particularly in broadcast. They were interested in a broadcast console, but they knew if they built it, it would be unaffordable. And we are now, in fact, building one — not because of them, but we finally got around to our ideas — and made a very successful product for high-end broadcasting.
R·E·P: *Did you target the international broadcaster with this console design?*
CC: At the moment, Neotek exports about 80 percent of its production. So, naturally, we find our markets best where our special skills can be leveraged and those [skills] aren't volume manufacturing. It's focusing our designs, doing what I call reverse consulting, which is talking to end users about their requirements. If you talk to enough of them, and you give them

your insight and they give you theirs, you can formulate a product concept that will [accommodate] lots of different related applications.

R·E·P: *What is your hottest seller right now?*
CC: It's hard to say, but [the Esprit] is selling very well. And every console that we build is made to individual order — every one is different. They may have different kinds of automation built into them. They may have different metering; patch bays or no patch bays; and desk areas built in; or "L" shaped frames. We are building a "C" shaped frame console for the United Nations. It just goes on and on.

R·E·P: *Are you saying that what you do best is custom designed consoles?*
CC: Yes, but I would call it more of a focused design. A person with particular requirements can come to us and we will work with them. For example, the frame is designed so that it can be, more or less, any width you want. If your truck accommodates 49 inputs and you want the most you can get in there, then we'll build you a 49 input console, or 47, or 41, or 26 or whatever it takes. And the console is designed to be built that way.

One example is our Encore console for mixing sound-to-picture in a film-style environment. We can make a one-, two-, three- or four-section console and each section can control the entire console. We can put desk areas and patch bays with clearance to the floor and a wide variety of openings in the console for machine control to be dropped in. We've put hard disk editing systems right into our consoles, and lots of different controllers.

R·E·P: *Is that a cost-effective approach?*
CC: It's certainly cost-effective for our customers. We have a healthy back-log right now. But, if you're just building an ordinary music studio, and it strikes me as one of the great peculiarities in our industry, that if you ask someone to go out and buy, lets say, 48 equalizers, they agonize and agonize and test and listen and get their friends in and compare. But if you ask them to go out and get a console with 48 input channels, they can do it off of a magazine article or an ad.

Same thing with microphone preamps. If you ask a studio to go out and buy 48 microphone preamps, it would take them *months* to decide about what

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sounds best and what mics they like best with it and so on and so forth. But [some] people will buy a console with 48 microphone preamplifiers without even listening to the console. So, when it comes to music studios, our customers are people who dig deeper — who listen and compare.

R•E•P: *What is unique about the Neotek designs?*

CC: From a circuit standpoint, we are one of the few manufacturers that doesn't use operational amplifiers for its main gain stages. We use a combination of discrete parts and operational amplifiers. But the discrete parts, rather than pruduce the output, or swing voltages or something like that, are generally used on input stages where they affect the sound quality and technical peformance. We probably use

ly, to improve it to have a more dynamic sound and so forth. In some cases, that doesn't work. With an older ribbon mic, for example, you really want it to be loaded down. In fact, ancient Neve consoles used to have taps on the input transformers for ribbon mics.

So, we did it because we could — because we knew it would have some theoretical advantages. But that doesn't mean it really sounds better in the real world. In order to make that judgement you have to have a lot of people tell you what their experiences are — people who are familiar with their own studio, their own microphones and their own recording techniques. You can't just sit around with headphones and listen to a CD and decide that it sounds better. Of course, you could do that, but this is not really the test that's important.

choice, I would probably use Mogami cable. It's just difficult to work with, and I'm not sure the customers would pay for the difference inside the console.

R•E•P: *How has the signal flow of your consoles changed over the years?*

CC: The signal flow of our music recording consoles has changed very little because the industry has changed very little. In fact, music recording and the whole mixing notion is becoming smaller and smaller and smaller in proportion to its influence on the rest of the industry, as I think it should be and maybe always was and no one noticed. Consoles today are more of an editing tool than a mixing tool. And professional audio is no longer [just] making records. It's satellite communications; it's multimedia, if that ever arrives in an identifiable form; it's telecommunica-

No one should think that the technology drives the market. It's the market that sucks the technology into those areas where it's most appropriate.

more components than the competition, and they are carefully chosen. We have custom-made non-polar capacitors, for example.

Our microphone preamplifier is probably one of the most sophisticated in a console, and for some reason, no one has duplicated it. If we were going on just the cost, we would probably use a chip, because there are decent chips out there, or chips where you can't tell the difference, as they say. But we think you can. For example, our latest, and it's still somewhat experimental, is a microphone preamplifier that uses wide area FETs as the input devices, rather than bipolar transistors. That allows us to use all film capacitors and it kind of streamlines the design, but we haven't reached a conclusion as to whether it sounds better or not. That's a very difficult judgement to make.

R•E•P: *Did you choose the FETs for their impedance characteristics?*

CC: No, because you are limited by the phantom power resistor. The input impedance is going to be no higher than the phantom power resistor, and a lot of microphones are designed to work into a particular impedance. We find that we can play with that a bit and change the sound of the mic — general-

R•E•P: *What else is unique about the design?*

CC: Throughout our consoles, at every important stage — the fader buffers, which are a difficult design, the summing amplifiers, the master fader buffers, — are class-A discretes.

Also, we have an elaborate cable and harnessing group, and we have our own shielded cable made for us special. In fact, we are designing a new cable that, I'm told by a guy at Belden who doesn't want to build the cable, is worth a patent.

R•E•P: *What's special about the cable?*

CC: There are many things about it, the insulation, the thickness and the construction of the cable so that it suits our wiring. We have modified it in such a way that it makes a reliable connection to run insulation displacement tooling.

R•E•P: *What about the electrical characteristics?*

CC: That's what I'm saying. By controlling the type of insulation on the conductors and the thickness and the foil wrap, we can get very good isolation, very good shielding from the foil wrap and still control the impedance and ... I don't want to get into the esoterics of cable assembly because we've built consoles with a wide variety of different [cable] manufacturers, and if I had my

tions with cable and so forth. A lot of the way consoles are used is in an editing mode rather than a mixing mode. Our mixing consoles haven't changed much because mixing is still mixing.

What has changed is a lot of the [functions] we've added to those consoles to suit them for broadcast applications. For example, faders that send out start signals and accept mute signals that cause the control room to mute when a particular microphone is selected and open, those kinds of things.

The signal flow in our broadcast consoles is very different from a mixing console. The signal flow in an Encore console for mixing sound-to-picture is *radically* different from any kind of an ordinary mixing console, a music mixing console, that anyone has ever seen.

R•E•P: *What is different about the Neotek EQ?*

CC: Usually, the only people who use state-variable are people who make outboard [EQs]. I don't think anybody uses it in a console. Certainly, some of the input limitations that I've seen in or around consoles, the guy obviously did it out of a cookbook. And it's no surprise they don't sound good.

Our EQ is substantially more expensive to build than most EQs. Is it worth

it? Well, if you're the kind of customer that buys a most-knobs-per-dollar console for a project studio, then it isn't worth it because your source material might not justify it in the first place — it might be already sampled. You know, if you're doing house music, it's the feel of the music and your ability to control your various sources that's most important.

R•E•P: *Can you identify and quantify what the differences are with your EQ?*

CC: Yes you can because — I want to say that it's an intuition that an analog designer develops. But by making certain measurements of various points — unusual points in the circuit, by controlling bandwidth of various stages through a circuit and by looking at overload margins, particularly inside the various nodes in the equalizer, you can tell, more or less, if it is going to sound terrible or if there are going to be noticable artifacts in the sound. Then by using a state-variable, you can control all of the parameters of an equalizer. And that's how you give it its sound. You can make sure it doesn't sound bad, then you can manipulate it until it sounds good.

There are some circuits where you don't have enough control — the typical wein-bridge EQ. There's nothing much you can do about it. There aren't enough components to change, and when you change some ratios, it doesn't work. In other circuits, if you have internal overload, where impedance is dropped low if you crank knobs, you get overloading of the chips or DC overshoot because you don't control bandwidths properly.

R•E•P: *What are your feelings on the analog/digital debate?*

CC: I don't think there's a debate. Again, the market is the arbiter of all of this. The first application of digital technology to audio would be the ones where it's the most sensible. That would be where you can wrap your arms around the problem, like a digital echo processor. You need to be able to sell that to as many people as possible. So naturally, the first consoles would be the consoles with the widest market. For example, no broadcaster in the world would use *any* of the available digital consoles, except maybe in a studio environment, if they have a studio facility as well. Because what they need is a mix minus from every input, but *show me* a digital console that has that, or a return talkback system, or a provision for two studios and multiple sets of headphone feeds so that the talent doesn't have to hear the

same talkback that the producer hears. But there aren't any digital consoles that do that. The people who design digital consoles are experts in DSP technology — not in what *humans* want to work with.

As the cost of technology declines, designs will be more and more focused as they have for analog consoles. No one should think that the technology drives the market. It's the market that sucks the technology into those areas where it's most appropriate.

I don't see a debate. I just see a market that's making it's requirements known.

We have microprocessors in our products, but we don't do digital signal processing at this point. We are not experts at it and the cost of becoming an expert is still too high. I think that the opportunity for Neotek is to form alliances with companies where we can combine our skills in understanding what operators want to do and what they want to use with their skills in pro-

cessing audio and high-speed networks and so-forth.

R•E•P: *How do you feel about the project studio market?*

CC: I'm all in favor of it. I don't know if you want to say it's a healthy or unhealthy trend, but it's a trend, and it's a market, and manufacturers and distributors *must* serve the market. We have project studio products and do reasonably well in [the market]. But it's usually somebody who isn't just trying to get as many inputs as possible, but somebody who actually listens before they buy and who cares about durability and the feel of the console. But, admittedly, that's not everybody, nor should it be.

R•E•P: *How are you addressing this market?*

CC: Our Elan console is affordable; sounds real good. It's got a great equal-

studios offer. If it's purely cost, then why not build a "project studio complex" into Record Plant? Break up those rooms into little tiny rooms and you can manage it much more efficiently. But, if you could have one building service, one maintenance service, one parking lot, having a large complex would be the way to serve customers with small studios. Multiplex Signal is doing exactly that. So why not have a multiplexed studio? The answer is, that isn't what project studios are providing their customers. But whatever that is, is real and valuable and should be encouraged.

R•E•P: *What is the state of the professional audio industry and where do you think it's going?*

CC: That always sounds like, "How long is the durge behind the hearse?" But it's not that way at all. Professional audio

R•E•P: *Perhaps someone wanting to get into the business now can get their hands on faders quicker than they would have 10 years ago.*

CC: Sure. Because they can go out and buy it now, and they can record their friends. What that means is, there are more entry points for qualified engineers. Unfortunately, there aren't more exit points for those engineers if you define the engineer as someone who works in a large studio and cuts hits all day. Certainly, *that* area of the market has decreased.

But how many audio networks existed 10 years ago, where you could pass audio between computers in a working environment? How much compression technology was there in satellite transmission? All these things had to be developed and you need a multiple of

> ## Markets change, and the people who serve markets must also change. If they don't, they will suffer. It's all market driven.

izer, and the mic preamp is the same that's in all our consoles. It's well made, and you can fit just about *any* automation system in the world into that console — anything from GML to VCA systems that run off MIDI. Again, it's not for every project studio, and it's certainly not for the one that's driven by the most-knobs-per-dollar. But we do serve that market.

R•E•P: *Do you feel the project studio owner/operator is someone who will reach for more expensive equipment in the future?*

CC: I don't think they need to. I don't think a project studio should be justified on the basis that someday they will "grow up and be *real* studios." I think they *are* real studios. Most of the people doing project work are accomplishing something. They are being creative. They are producing value for their customers. If they didn't have a market, there wouldn't *be* a project studio market. Maybe a few people do it for fun, but most people have customers, and those customers are being served better by project studios than they were by the alternative.

R•E•P: *Because of the rates?*

CC: I don't think it's just rates. I think there's a whole complex of individualized service and attention that project

has gotten in this "gloom-and-doom" mentality because it looked at music recording and said, "music recording is professional audio." But in reality, it's just the tail; it's not the dog. So, if you *look* at professional audio, it's growing like gangbusters. But it's growing in satellite communications, data compression, multimedia distribution systems, cable, audio networking — there is fantastic growth in audio for professional applications.

You can't bemoan the fact that upper mid-size recording studios are having tough financial times. There are lots of industries that aren't serving their customers, that aren't generating markets and serving markets that are having a tough time. Markets change, and the people who *serve* markets must also change. If they don't, they will suffer. It's all market driven.

I think there is a tremendous opportunity in professional audio. I have a number of young friends who are considering engineering. I will encourage them to look at professional audio. But these days it *won't* be because they want to tune guitars for years and then eventually end up as a music recording studio operator. They will find many more exciting opportunities — and plenty of them.

skills, just as you do in a studio. I don't know how many recording engineers don't know how to align a tape machine. How many professional audio engineers in the future will not know how to program a DSP. It's a different complex of talents, skills and equipment needed to serve the market.

[The market] is always shifting and changing. Even big companies can miss it and try to rediscover it. IBM is a *great* example. So why should professional audio be any different? I suggest, because upper mid-size music studios have seen their market go away, doesn't mean that professional audio is doomed, because it's not — not by a long shot.

I would like to see more people who care about good sound and the little things that make audio interesting, become involved in those leading edge processes, instead of digital geeks who want to use auto-routers to lay out analog circuits the way they lay out digital circuits. Then they wonder why they don't *sound* quite the way they expect. There are many fun and exciting things going on. And you shouldn't think that audio is doomed — not at all. ▲

Jack Smith is editor of R•E•P.

Breaking the Studio
Sound
Barrier

by Carlos Chafin

Carlos Chafin (left) and Robbin Thompson (right) faking-it-at-the-console shot.

My how times change. Just a few years ago, I worked behind a $100,000 mixing desk with a $200,000 Synclavier surrounded by engineers, assistants and maintenance people just so I could complete a typical project. Today my equipment fits comfortably in a small bank vault — yes, our studio is housed in an old bank building — that I operate virtually on my own. Instead of expensive studio-consuming pieces of equipment, I work with a rack of MIDI modules and a Macintosh Quadra computer, supplementing them with a small group of select musicians and performers. The result? My work sounds fresher and more diverse, and I'm a lot more productive.

And I'm having more fun.

When my partners and I opened In Your Ear two years ago, we were ready to rethink the traditional studio philosophies. Is bigger, more expensive equipment better? Can we really operate a successful recording and music production facility in a secondary market like "little ole" Richmond, Va.? And even more important, how could we do it without spending millions of dollars on obsolete or about-to-be-obsolete, equipment that can, more often than not, impede the creative process?

We knew we were in a service business. After all, none of us were rookies at this. It was critical that we never allow bad work to go out the door and that we never compromise client relations. As with *many* commercial studios, we are in business to serve our clients' needs affordably, reliably and creatively.

Most clients don't care how we do it. Sure, some will ask if we have their favorite console or piece of outboard equipment, but most focus on the results. I decided I would much rather spend money on educating clients and talent regarding new technologies and methods of recording than on buying expensive gear. And we still must get those clients in the studio, which means spending money on marketing. Being the new guys on the block (even though we all have good reputations from previous affiliations) means we must work hard to market ourselves well.

So what do we *really* have to sell? Our equipment? Every studio and production company can — and many do — have basically the same equipment. What we have that's *truly* unique and marketable is our talent and creativity. Were we crazy to think that lower equipment costs could give us the quality product we demand while letting us allocate money to marketing and education *and* allowing us to put more of our energies into talent and creativity?

Not by a long shot.

The cost of creativity
A benefit of being a musician/producer is having complete control over the creative output. I have always believed that aligning tape machines, recalling and manually resetting mixes, tweaking synchronizers and using finicky video equipment is a poor use of time. Digital technology has eliminated many of these

routine operations that distract from the creative flow. For example, a DAT machine either works or doesn't. You don't have to tweak the bias, calibrate the noise reduction or do any other audio voodoo; just throw in a tape and many studios still struggling to be successful? Unfortunately, many users of older technology are *still* carrying long term debt on equipment purchases made five to 10 years ago. These facilities may not be able to afford to

> If you know what you're doing and have the talent, you can achieve greatness for a lot less money than you could a few years back.

get to work. And with digital audio workstations you can put your energies into the *creative* process rather than the *recording* process. Since new inexpensive digital equipment, like the Alesis ADAT and Soundtools, sounds great and works well, there is no reason you can't produce a hit song using a recorder/computer system that costs less than $5000. Don't be hyped into thinking you need expensive gear or a famous track record to succeed. What you need is talent, focus, friends and opportunity.

Still, some people argue that an older, tape-based recorder produces a warmer sound. This so-called *warmth* may be distortion. But isn't the objective to have control over the final product? After all, a photographer may use a filter to warm a shot, but it can be removed so the next picture can be crystal clear and pure. Exactly when did digital and warmth become mutually exclusive?

Recently, in our studio, a recording group was trying to recapture the sound and feel of the sixties with hiss, noise and distortion intact. They recorded instruments through funky, old, non-refurbished Vox tube amps and sang a lead vocal into a Kalimba miked with a cheap dynamic mic — and they recorded it digitally. Why do you need a $1200 mic preamp or an expensive console to capture this correctly? If the songs are strong, the performance good and the production doesn't ruin things, magic can happen. Digital equipment gave them a flexibility they never dreamed of. The message is clear: If you know what you're doing and have the talent, you can achieve greatness for a lot less money than you could a few years back.

Switching tracks
With low cost, high productivity equipment so readily available, why are so

upgrade their tools to allow more efficient operation and better sound necessary to entice prospective customers.

Many are still paying for the high cost of the digital and analog pro-audio equipment boom of the '80s, purchased, of course, with the best of intentions while trying to satisfy the needs of a loyal client or following claims of some manufacturers. How many studios are still reeling from their initial investment and loudly claiming they sound better than today's formats? No wonder they buck the new technology and put up barriers to guard against studio evolution!

Is it possible for those studio owners to change their directions or even start over? More than a few large studio owners have. Unfortunately it was often the result of a forced sale and, in some cases, personal bankruptcy. You've got to be flexible. You can't be afraid to abandon old techniques and familiar tools. With a lack of standardized formats, many people are holding off, waiting for the dust to settle. They claim 24-track or 16-track 2-inch tape formats are still the standard for recording hit records. They're right. But *I* wouldn't buy a new 24-track tape machine at this point in technology. Some also believe that only expensive gear is worth their time and experience and therefore resist using newer, low-cost options. It's time for them to realize that some of the new equipment can out-perform what they're used to and do it for a third to a tenth of their existing equipment costs.

New technologies — new opportunities
While many larger studios and video post houses are nervous about low-cost digital developments, I've used it to my advantage as a partner in a new and very successful recording and music
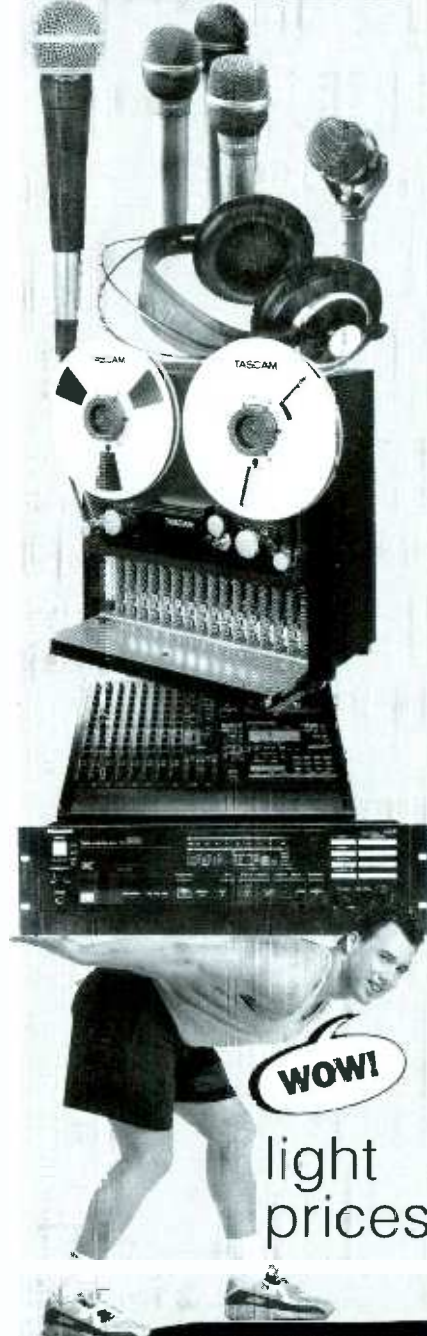
www.americanradiohistory.com

production facility in a secondary market. The secrets of our success are simple: We have a very talented staff, we're open to new ways to record and produce music, and the world is shrinking.

Think global, stay local. If there's a market for what you do — and you do it well — you can succeed. Other businesses have grown by going to local markets to sell their products. Wal-Mart didn't get big by just keeping one store in Benton, Ark. and hoping you would stop by to shop when you were in the neighborhood. They went to you.

Thanks to '90s technology, you can get the global work without leaving your back yard. If you go to your prospective clients' door steps, they might just open the door. Sure, personally knocking on your client's door is great, but if doing that is out of the question, there's always Federal Express. However, if overnight delivery isn't fast enough, consider fiber optics. Digital Patch/Ed Net and others offer a cost-effective dial-up service linking studios around the world.

As an example of how technology increases customer service, there's an announcer we use who'll record about 4,000 spots for our clients this year. Since he lives 120 miles away from our facility, we've licensed an ISDN line to his house. By doing this he's made his service more readily available, and we made it possible because we wanted to please our joint clients. The studio, announcer and client are now partners sharing the rewards.

Another way technology can help both the studio and the client is through new, lower cost optical storage systems, including CD and CD-ROM recorders. Because they're not durable enough, DAT tapes are a less than optimum long-term storage medium. The

ter storage. The savings in storage space alone is astounding. Plus the archived work can be played on any ordinary CD player.

Imagine the possibilities. You can send an ad agency producer a copy of the 100 spots they produced with your company — in chronological order.

On CD. You can hand a producer or session musician an anthology of the work the two of you collaborated on. On CD. And this is only the beginning. Think of the production steps which can be skipped by mastering directly to CD.

While some engineers and producers claim analog tape sounds better, and they may complain about the loss of editing capabilities afforded by tape, I disagree. The only ¼-inch tape machines I see in my future will be used to run radio station dubs for commercials. If agencies can be talked into releasing commercials on CD, well, there goes the tape machine. It will take mass

In 1991, when In Your Ear opened we had three employees and no studios. Within six months, we opened our first MIDI music production room and a companion audio-for-video post suite. We spent less than $250,000 on initial renovations, equipment and marketing.

Today we have a third studio, a great staff, a string of happy clients and most of all, we're having a blast. We work together as a team, make personal sacrifices when necessary and thanks to the new, cheaper technology that's within our grasp, our jobs are constantly becoming easier while we turn a profit and continue to create an ever improving final product.

Carlos says, "Don't worry Robbin. It's backed up." ▬

If you want to take advantage of old studio operation barriers which have been knocked down by today's mass produced technology, ask yourself the following questions: How creative are you? Do you have a unique talent? Are you open to new ways of doing old chores? If so, it may not be long before I'm reading your story on these pages. Possibly while recording four musicians, three singers and an announcer digitally across ISDN lines coming from eight different studios around the world to our new studio on a beach in the Caribbean...   ▲

No wonder they buck the new technology and put up barriers to guard against studio evolution!

ability to record CDs, with their long-term data integrity, for the same price as magnetic-based technologies is almost here. Many studios are now storing their masters on CD and eliminating ¼-inch analog and digital master

acceptance — and therefore mass manufacture — for the cost of recordable CD stock to drop to a point that it can compete with ¼-inch tape and, of course, it's lowly predecessor, the audio cassette.

Carlos Chafin is the owner of In Your Ear Music and Recording Services in Richmond, Va. and a composer who wrote his first film score at age 11.

Devonshire Studios — Twenty-five Years Strong

by Alice Maddox

At an age when most people think about retiring, David Mancini plans to expand his business.

With 25 years of success behind him as the owner of Devonshire Productions, Inc., in North Hollywood, the recording studio where careers of Billy Joel, Whitney Houston and many others took bounding leaps, Mancini, 62, is setting new goals. His most ambitious: to start his own record label with additional studios near the heart of country music in Nashville. "I don't feel I have another 40 years, but as long as I'm healthy enough to maintain a rigorous lifestyle, this is what I want to do," says the man who started his career as a balladeer singer. "This expansion will provide us with a means to cover all facets of the music industry."

It also makes good business sense. Advances in recording equipment have made it possible for artists to record in their homes or in other low-overhead studios. Mancini's strategy is to provide top-notch service and the most advanced and fine-tuned equipment in the industry while keeping rates in line with the competition.

Diversification also has become a key factor. The record label dream is the culmination of a series of steps he began taking eight years ago that have expanded his offerings to include post-production sound mixing for television, movies such as "The Bodyguard" and cartoons such as "Ghostbusters."

A long way from Detroit

Mancini grew up in Detroit and sang ballads in local clubs in a style similar to that of Perry Como, as he describes it. In his early 20s, a disc jockey put Mancini in touch with Big Band leader Buddy Morrow. For half a year, Mancini traveled across the country with the band as the lead singer.

Mancini moved to California, which he first saw when he traveled with the Buddy Morrow band. "I thought California was the land of opportunity and I needed a fresh start." He began constructing recording studios for other people. Then, using his building and

musical experience, he hooked up with four other investors to create Devonshire. By the early 1970s, he bought out the other partners and he and his wife, Dolores, became the sole owners. Then the business started to grow.

The studio's first platinum album was Billy Joel's "Piano Man." Devonshire's artists number in the hundreds: from jazz group Weather Report for five albums to Ozzy Osbourne, Bel Biv Devoe, Alice Cooper, Roger Waters and Earth Wind and Fire, to name a few. Mancini has fond memories from moments of socializing with the artists at the studio's recreation room and the times he invited them to his home.

"From being a singer years ago I can appreciate what a singer does," he says. "I think in general most of the superstars are very thankful and humble that they have achieved success. They got where they are because they're talented. They deserve to make what they make because they earned it."

The grand plan

Mancini did not start out with ideas of adding post-production capabilities to his music recording studio. But economics made it a wise choice. "Diversi-

fication is where I think it's at in this business," says Dolores Mancini, an actress who has turned her talents to managing the studios. "The studios that do not diversify, that are strictly into music, have time on their hands while we do not."

Because of digital recording, which has created many low-overhead recording studios, Devonshire has been forced to compete. The strategy: buy higher-end equipment than the home studio artist can afford and provide around-the-clock maintenance for uninterrupted recording sessions.

"We are forced to always be on the cutting edge of technology," Dave Mancini says. "We look for new and improved ways of making the recording process better." While he is buying more expensive equipment, he also must keep his rates competitive. Further, the quality service strategy applies to everything from providing a variety of updated studios to maintaining neat and clean rooms and bathrooms as well as bringing in lunch during a busy session.

"The business has been built from word-of-mouth and reputation," he says. "You pay attention to detail and go above the call of duty. It's the little

things that make the difference between whether you get repeat business or not."

The winner in all of these technologicial advances, he says, is the consumer. "The consumer is getting a finer quality of recording than ever before. I don't think it will get any better."

Today, Devonshire is a six-studio complex. Calling on his building knowledge, Mancini is overseeing the addition of four more rooms for post-production work, such as dubbing and mixing

licited demos, according to Mancini. "Most major recording companies don't even take demo tapes. They don't have time for it. I think a lot of talent isn't getting an equal opportunity. I would like to see more of an open-door-policy and will surely have that openness and encouragement with future endeavors."

Although he wants the label to cover all types of music, he chose to locate near Nashville to attract country artists. "Nashville is where the good country musicians and writers are," he says. "I

> Mancini's strategy is to provide top-notch service and the most advanced and fine-tuned equipment in the industry while keeping rates in line with the competition.

sound-to-picture; and a THX-style room for Surround-Sound on movies and music videos. The addition will be completed by the end of 1993.

In July, the Mancinis acquired another North Hollywood post-production business, Audio Zone, with a Foley stage to reproduce sound effects, as well as two sound mixing rooms.

Creating cartoon sound effects has become a chunk of the business. Mancini expects to take on more than 150 cartoon episodes this year. That will be in addition to 30 to 40 music albums as well as a variety of TV and movie sound mixing projects including a Muppets series, Home Improvement and Golden Palace.

The heart of country music

Mancini's desire to create a record label stems as much from his belief that he can do it as his desire to fill a niche in the industry.

His targets are artists whom other record companies have turned away because their records no longer go gold and promising young artists whose talents and careers need nurturing. He plans to put together a production team to spot the best candidates. "There is a lot of dissatisfaction among those artists whose sales have dropped off and whose careers can be revitalized," he says. Many proven musicians as well as incoming artists are finding the route to being signed by a major record company much more difficult because the larger labels are less likely to take unso-

want to be where country music is happening." Creating a label will also mean more recording business for his California studios.

But in order to proceed, he will need to find financial backers. He hopes the 2.6 acres he bought in Tennessee will be the founding spot for a 45,000-square-foot building with four studios, a video stage and offices. A swimming pool, tennis courts and putting greens will provide a relaxing atmosphere for artists.

"I want to build the best facility that my experience has taught me," Mancini says. "I haven't seen anything quite like what I am contemplating, although there are many fine studios around the world. Hopefully, our facilities will create the kind of atmosphere to encourage and expand the creative process."

Hard work deserves play

The idea of playing as hard as they work also appeals to the Mancinis. "My ultimate goal is to own a 150-foot yacht," he says with a chuckle. The couple takes time out to enjoy their Newport Beach home some 30 miles away from the studio.

"I think I've worked hard all my life," he says. "I've had my ups and downs, but I feel very blessed and fortunate in the fact that I still have the desire and health to expand and enter into these new ventures." ▲

Alice Maddox is a business writer based in Rock City Ill.
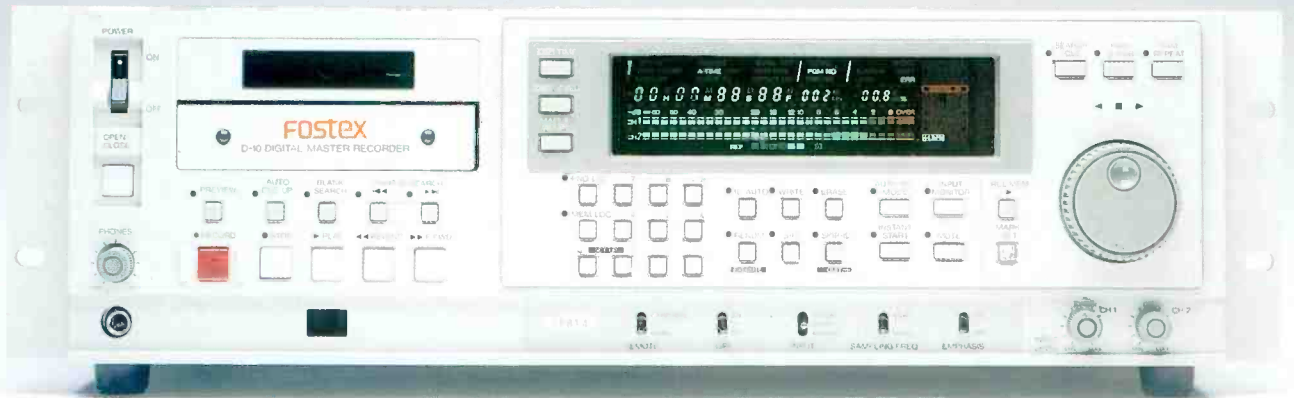
The Fostex D-10

by *Michael Castronovo*

The D-10 is the latest addition to the Fostex arsenal of professional DAT recorders. Like its big brother the D-20, it offers many features not commonly found in most current DAT machines. One of the most intriguing and exciting of these is instant start. If you've been working with DATs for a while, you know how wonderful and frustrating they can be; wonderful because they've helped bridge the gap between "semi-pro" and "pro" studios, yet frustrating because some DATs have lacked the quick start ability of their technological predecessors. Considering this, the Fostex D-10 has taken a high-tech approach to solving an old "time" problem.

Life in the instant start lane
Most of the old "industry standard" two-track analog decks were able to cue up to a location point using a "rock the reel" method. This typically allowed a start time of approximately 0.5 seconds. In contrast, start up time for most DATs is typically 1 to 2 seconds. You can see why DATs are often useless for tight editing work. Fostex's instant start addresses this problem.

The D-10 instant start uses 2 RAM cards, each having 4 megabits of memory. Depending on the chosen sampling rate (44.1k or 48k), you get approximately 5 seconds of RAM. When using instant start, a portion of this RAM is played first. Then the D-10 cuts to tape to continue the playback without interruption. When Fostex says instant start, it means *instant*! As soon as my finger hit the bottom travel point of the play button, the sound began. Upon repeated tries, I could detect no delay in the instant start.

This works great if your program does not have blank space after the start ID code. But if there is space, you can use the auto cue up. This feature locates the beginning of the signal located near the start ID and eliminates the blank space problem. Auto cue up automatically engages the instant start, so you only have one button to push to get things ready. If that's *still* not where you want to begin, there's even a RAM scrub that allows the spring loaded shuttle wheel or free spinning cue wheel to be used to "rock the RAM" back and forth in kind of a high-tech version of the old days. With these D-10 features, you'll easily find the right start point. This feature will make editing from DAT-to-DAT or DAT-to-anything else much easier.

It should be noted that the instant start capability is not available through the optical outs. This should pose little problem since editing usually is not done in the optical domain. Perhaps instant start is not extremely popular yet. However, I predict that soon, most

Published D-10 Specs

R/P Frequency Response:	20Hz - 20kHz +/-0.5dB
Sample Rate:	44.1kHz and 48kHz
S/N Ratio:	Higher than 92dB (Emphasis Off)
Dynamic Range:	Higher than 92dB (Emphasis Off)
Total Harmonic Distortion:	Less than 0.05% (1kHz, +4dBu)
Channel Separation:	Better that 80dB (1kHz)
Wow and Flutter:	Less than +/- 0.002% WTD/peak
Standard Recording Level:	-20dB/-12dB (switchable)
Meter Display:	FL fluorescent tube 28 segment
List Price:	$2850.00

serious engineers wouldn't think of buying a DAT without it any more than they would buy a multi-track recorder without gapless punch-in.

Hooked on delta sigma
I should mention here that the sound quality is excellent. Fostex uses a 16 bit delta sigma A/D converter (AK5326). I found the D-10's sound to be clean and clear of coloration whether using the digital or analog inputs. The D-10 also offers a digital fiber optic in and out, as well as XLR jacks for digital AES/EBU. Both XLRs and RCAs are provided for analog connections. There's a balanced/unbalanced switch on the back to make connections even easier.

Better to light one lamp
One thing that caught my eye was the bright blue-green fluorescent display section. It offers temporary peak hold meters and a variety of mode indications, such as digital in, S(start)-ID, skip-ID, TOC(table of contents)-ID, end-ID, PCM mute, PCM error, and emphasis. The emphasis selector appears to have been included to allow compatibility with other (mostly older) DAT machines.

> **OVERALL: Best bang for the buck for editing and instant start.**

A nice feature of the level meter display is the user-adjustable reference point setting. The reference point is adjusted with the jog wheel while in the set up mode. Note that changing the setting has no effect on the actual record level. What *does* change is a small numeric box just below the meter bars which illuminates to read either -20db or -12db. This is helpful for those of us who grew up with a color change at the "0" reference for analog VU meters. With these digital meters, the only color change is at the overload point. The D-10 adjustable reference feature gives the engineer a visual "bulls-eye" to shoot for. Looking closer at this same area of the display, there are other numbers between -20db and -12db. Perhaps Fostex's future software plans will expand this reference range selection.

HITS — Price, Instant start, RAM scrub, Punch-in/out, Rehearse punch, Expansion ports, Fiber optic capable

The only problem I found with the display in general involves the numeric counter read-out. Since a fluorescent display is used, the numeric counter has the same bright blue-green color as the rest of the display. Furthermore, each pair of digits is separated by letters indicating hours(H), minutes(M), seconds(S) and frames(F). The display ends up looking very busy, like: "01H25M35S21F." Having the letters differentiate seconds from frames is nice, but perhaps if the letters were a different color, such as orange or red, the read-out would be more user friendly.

The Fostex 0-1 punch
The D-10 offers punch-in/punch-out capability. This can be done manually or automatically within an optional rehearse mode. The punch-in/punch-out uses a seamless, gapless edit point. There is no cross-fade time. I suspect that is why Fostex included the rehearse mode. To use the rehearse mode, simply set a "0" memory point for the punch-in and a "1" memory point for the punch-out. Then set the auto punch button to on and press the preview button to check the punch-in point.

What about checking the punch-out point? Well, currently the punch-out point is not rehearsed during the preview mode. Rick Cannata, product specialist at Fostex, told me that the lack of a punch-out preview point is likely to change. I predict a software revision coming down the pike soon.

Software revisions tend to be a way of life for studios in the '90s. And I'm always glad to hear when a company is willing to improve its units — even after they have hit the streets. Talking about revisions, currently the D-10 does not write or erase end-ID codes. Again, according to Fostex, this will probably be changed soon, since it has been requested by a number of D-10 owners. In the mean time, the blank search feature can be used to find the empty space after the last item recorded.

MISSES — No pitch control, No SMPTE (Yet?), Busy display

I've got the place
The human interface concerns of the D-10 have been well addressed. The front panel has large back-lit controls for the transport functions. A 10-button numeric keypad along with program locate and memory locate buttons make finding the desired point on the tape quick and easy. These same buttons, as well as most of the controls on the front panel (except the cue and shuttle wheels), can be found on the wireless remote that comes with the D-10.

To keep sessions moving along quickly, the fast forward and rewind transports top out at 250X normal speed. That's not as fast as some DAT machines, but doing an A/B with another manufacturer's 400X unit, I found only a 5 or 6 second difference when shuttling across a 60 minute tape. I can live with that. Especially since that other unit takes over 1 second every time I push play, and has no punch-in/punch-out capability!

To eliminate the copy code problem of "is it on or off? (only your technician knows for sure)," Fostex simply eliminates copy code all together.

In place of consumer features, there are pro features that time (and space) will allow me to only touch on. For example, programmable start and stop record times allow facilities, such as radio stations, to record late night network feeds and have quality sound-bites the next day. Also, the rehearse mode ensures the start IDs are placed where you want them. Another example is the GPI (general purpose interface) for tying in the D-10 to automation systems, such as those used in FM broadcast. And another of my favorites is the time locate which allows users to enter a real-time tape location using either numeric pads or jog wheel to directly locate to that point of the tape. This is especially handy when a tape either lacks program numbers or the log information is incorrect.

If you've got the time
Although the D-10 does not read or write SMPTE, there are two expansion ports on the back. And inside word has it that *one* expansion card may be a read/generate system that will allow the unit to work with IEC timecode. According to Fostex, this is mainly to be used as a locating tool. However, there are some un-lit characters in the right corner of the display screen that

say things like "chase lock, external clock and clock lock." Maybe Fostex has a few more cards up their expansion sleeves?

The good old ways?

To sum it all up, if your studio has been doing the "one only DAT" juggling act, or you're tired of trying to edit with a start time of 1 to 2 seconds, your ship may have just come in. With a list price of $2850, the D-10 addresses many of the handicaps that have plagued DAT efficiency. Aside from the SMPTE and pitch control, I can't think of anything important the D-10 lacks when compared to other fast start decks in the $4,000 to $9,500 price range. Especially when some in that group don't have SMPTE or punch-in/punch-out capability. And who knows what the D-10 expansions may bring? My recommendation is to get your hands on one of these DATs and try it for yourself. But I've got to warn you, once you use the Fostex D-10 instant start, it will seem like the *old* way is just like,... well, like working with turntables again! ▲

Road Test

Review of the BBE 862 Sonic Maximizer

by Michael Castronovo

In this day of digital recording technology, you'd think the need for something called a "Sonic Maximizer" was far passed. But you *might* be very wrong. The BBE 862 Sonic Maximizer boasts a list of feats that would make a door-to-door snake oil salesman blush. According to the BBE manual, "Program material will have sparkle and unmistakable clarity. Lower frequencies, such as bass guitar, will be more evident in the mix. Voices will be crystal clear...Rap music will take on new dimensions with a thunderous, yet tight thump which cannot be achieved with any other sound processor." These are certainly lofty goals. Has BBE gone too far in describing what their little black box will do? You might be surprised.

I tested the BBE 862 Sonic Maximizer using some of my studio recorded ADAT digital music tracks and listening through JBL 4400 monitors. With the 862 patched in-line through the Studiomaster II board, I assigned a variety of voices and instruments individually. By using the function button to bypass the BBE process, it was easy to A/B the results.

I found the unit quite easy to operate, since this is a stereo unit having only two knobs per side. The instructions in the owners manual are quite clear and simple. One knob is labeled "lo contour," and the other is "BBE process."

First off was a female lead vocal. Just a little Maximizer was needed to add the sparkle the singer lacked. And it *did* add sparkle! By setting the process knob at about 1 o'clock with almost no lo contour, I got what I wanted — a nice, crisp female vocal that could cut through a band without sounding harsh or artificial.

Next, it was off to some instruments. I had an electric lead guitar track I wanted to try. Its sound was clean without distortion, but it was a bit "plain Jane." I quickly found this guitar also benefited from a 1 to 2 o'clock process setting. This time the lo contour was needed to help add fullness to the sound. The end result? I still had a *clean* guitar sound that worked. The typical "overdrive" effect of some other similar systems wasn't apparent.

I was eager to try a keyboard. I located a piano-synth layered keyboard rhythm track from an Ensoniq ASR-10. Again, the BBE added a unique beneficial mixture of clarity and fullness. It was neat to hear a 16-bit keyboard recorded on a 48kHz digital system sound even better with the 862.

After operating the infinite adjustment knobs, I found something that could lead to confusion. The graphics encircling each knob indicate that full counter-clockwise is off (minimum), and full clockwise is on (maximum). This turns out to be half true. The BBE process knob operates this way. However, the lo contour knob is actually a cut/boost 10db system with a neutral position at the top (12 o'clock). During my control room usage of the 862, I had to keep reminding myself that *center* was "no bass change" — not the far counter-clockwise position. A slight change on the front panel with the addition of "-10, 0, +10" indicators properly located around the lo contour knob would be helpful.

Like most engineers, I always like to "push the envelope." I tried some overkill with a vocal track to see how many problems I could create. Only at a "way too much" setting did the 862 appear to harm the quality of the signal. The result was a

Published Specifications:	
Signal to noise:	-92dBu.
THD in bypass mode:	less than 0.0002% at -0dBu input, 20-20kHz
THD in process mode:	0.025% at -0dBu input, 20-20kHz
Output Impedance:	600 Ohms, balanced
Input Impedance:	14.7k Ohms balanced
Maximum output:	+23dBu (may vary due to control settings)
Maximum Process:	+10dBu boost at 5kHz, 0dBu input
Lo contour:	+10/-10dBu adjustment at 50kHz, 0dBu input
Warranty:	1 year
List price:	$599

"grainy/digitized" texture that distorted the singer. This is no different from using too much of *any* process, whether it is reverb, echo, distortion or even equalization. However, it is nice to know there is more processing range available than is needed to achieve quality sound.

A client came by with some noisy narration tapes. We attempted to remove hiss from the tape with equalizers. We found, as expected, that doing this *also* removed some of the intelligibility of the vocal. It sounded muffled, so I decided to try the Maximizer. The client and I were pleased with what we heard. By adding the 862 after doing the EQ filtering, much of the vocal clarity returned without much of the tape hiss. We used the output meters to see the mono signal strength. This helped give us an indication if our voice to noise ratio was workable. The voice to noise ratio varied in different segments of the recording. Depending on this ratio, the system worked well sometimes, and sometimes not so well. I was pleased to have the 862 help where it could in this situation, but not surprised that it couldn't leap *every* tall building in a single bound.

The output meters are mirror imaged — each indicating a level increase toward the outside of the unit. Because of this, using these meters to determine a stereo signal feed through the unit is difficult. You end up having meters going two directions at one time. Certainly your ears are the best judge of how the stereo balance is doing, but I like having meters for a good visual back-up.

The clip input light hardly flashed while I was evaluating the unit. This is probably because the 862 is designed for +4 balanced systems, and I am using my Alesis ADATs and Studiomaster II console at -10 levels. Connecting the 862 to my system was easy. The inputs and outputs are balanced ¼-inch tip-ring-sleeve (TRS) phone jacks in parallel with XLR connectors. Unbalanced ¼-inch phono jacks can be used for systems with unbalanced wiring without loss-of-quality concerns. The instruction manual lacks a diagram on how to use the unit with an unbalanced system, but the standard tip-sleeve configuration did the job.

What the manual *does* include is helpful information, including application examples, testing procedures, specifications, and seven pages of schematics. Although I don't intend to rewire the unit, having the schematics is a nice touch for those service emergencies that always seem to show up two days after the warranty runs out. Including printed schematics is a courtesy that other manufacturers should follow.

How does the Maximizer work? According to BBE, the process "works dynamically by boosting the harmonic content of the music program, as determined by the signal input, and imparting a linear phase shift across the audio spectrum. The phase shift allows for a separation of the high, mid and low frequencies so that they do not arrive at the speaker network at precisely the same time. This prevents the higher frequencies from being smothered." BBE also says this phase shift is only 2 milliseconds and that it will not be perceived as a delay or echo. The lo contour knob is a bass frequency adjustment (50 Hz) and "...will add power and warmth to the music program..." BBE neglects to explain *how* the harmonic content is boosted. Of course! This is one of the secrets that keeps this little black box so mysteriously intriguing.

Understanding that the 862 works with phase delay, I had two last concerns. How does it sound on whole mixes, and will the phase change cause any perceivable problems when summing to mono? By feeding through a couple of my DAT mixes, I was able to lay my concerns to rest. First, I fed in a hot R & B mix. By adding some Maximizer, I ended up with a "power edge" that almost made me giggle like a kid on Christmas morning. In this application, definitely a small amount (about +4dBu) of the lo contour was needed to balance out the high frequency change. Upon summing the mix to mono, there were no perceivable phase problems. And the clarity and drive were still there.

I wanted to determine that it wasn't just large instrumentation that sounded good when processed, so I used the 862 on a barbershop quartet. The results were, again, very satisfying. The Maximizer added clarity and warmth without sounding metalic or unnatural.

My conclusion is that BBE has a product that does live up to its claims. In answer to the all important "how does it sound?" question, the BBE 862 sounds surprisingly great! Most impressive to me is that the 862 adds sonic clarity without adding the typical fuzziness that has been associated with some other units. When used tastefully, I found the BBE 862 Maximizer adds to the musical item without the usual harmful side effects. At a list price of $599, the unit will surely find a home in many professional and project studios, both analog and digital. ▲

Has BBE gone too far in describing what their little black box will do? You might be surprised.

Michael Castronovo is a Contributing Editor for R•E•P and the owner of "Studio B — Digital Recording," a 24 track digital studio in Rockford Ill. His hobbies are driving sports cars and filling out warranty cards for new studio equipment.

Record Plant Celebrates Silver with Upgrade

HOLLYWOOD — In celebration of its 25th anniversary, Record Plant recently completed a $4 million upgrade, which more than doubled the building's client areas. Two studio suites and a digital editing/MIDI/overdub suite were added to the existing two studios.

Record Plant's mix/overdub suite, SSL I, was the first of the two new studios to be completed. This room's 96-input SSL SL-8000 G-series console with Ultimation is one of the largest SSL consoles ever installed in an audio facility. The console has 48 "E" EQs and 48 "G" EQs.

Neve I is the second new studio suite and was designed to accommodate tracking, scoring and mixing projects. The console is a 96-input Neve VRPS Legend with GML Moving Fader automation. The tracking space includes a 25 by 40 foot room with a sloping ceiling that reaches a height of 35 feet.

The monitors in both rooms are custom Augspurger cabinets with TAD components including the center speaker and JBL 8330s mounted in the surround soffit. Near field monitors include Tannoy, KRK, Yamaha and Auratone.

The latest addition to the expansion is Mini-Plant — a digital processing/MIDI/overdub suite which offers digital processing software and hardware from both Digidesign and Sonic Solutions, Mac Quadra 950 computers and multiple 21-inch color monitors as well as an array of MIDI instruments, sequencing software and sample libraries.

Taking it to the Streets

ELKHART, Ind. — Crown International recently unveiled a custom mobile educational facility built into a semi-tractor trailer equipped with Crown amplifiers, microphones and IQ System computer control products. The facility will soon include eight IQ System computer control workstations, according to Crown.

The air conditioned trailer accommodates up to 15 students and includes its own power supply. Joe Wisler, Crown's pro audio liaison and the man behind the mobile facility's concept, and Jim Stemble, Crown's contractor/consultant liaison, will take turns leading educational sessions and demonstrations. Amplifier and microphone sessions take only a few hours, while IQ System classes last up to two days.

A travel schedule for Crown's classroom on wheels is in the works and will be released in early 1994. For more information contact Wisler or Stemble at Crown International, Inc., 1718 Mishawaka Rd., Elkhart, Ind. 46517; or call 219-294-8000.

Money for Multimedia

REDONDO BEACH, Calif. — Lone Wolf, Inc. announced in July the completion of an equity financing and strategic alliance with Paul Allen, the co-founder of Microsoft Corp. Allen invested an undisclosed sum and reportedly was the sole investor in this offering.

The company also struck an agreement with Allen which would provide for cooperative R&D of Lone Wolf's net-working technologies and cross-licensing of related enhancements between his affiliated companies which includes Asymetrix, Interval and Lone Wolf.

Lone Wolf's MediaLink™ multimedia network protocol is a high-bandwidth networking system. It was designed specifically for real-time multimedia data transmission, such as audio and video signals as well as control and monitoring signals, on a single cable through a single port. Also, the company's VNOS™ (Visual Network Operating System) provides a computer-independent authoring platform which reportedly enables users to create interactive virtual representations of any networked environment.

Neutrik Acquires Amber

MONTREAL — Neutrik AG recently announced its acquisition of Amber, a Canadian-based manufacturer of intruments and test equipment for the audio industry, and formerly a division of Coreco, a manufacturer of imaging and multimedia products.

The new company will be established under the name of Neutrik Instrumentation, Inc. and will have its headquarters in Montreal, Quebec, Canada. It will be responsible for sales, marketing, technical support, repair and calibration for the North American market, according to Bernhard Weingartner, founder and president of Neutrik worldwide operations.

The company's new managers will be Vincent DeSouza, sales manager and David Hudson, technical manager.

Pilz Opens CD Factory

CONCORDVILLE, Pa. — Pilz America, Inc., U.S. subsidiary of privately-held Pilz Group of Kranzberg, Germany, has opened a $12 million compact disc factory and distribution center in Concordville, Pa., near Philadelphia.

The 45,000 square-foot facility can produce eight million CDs per year, according to Martin Mair, general manager of Pilz America. He says the company plans to boost capacity to 40 million by adding 22,000 square-feet and hiring additional staff.

In addition to audio CDs, Pilz America manufactures CD-ROMs for computer data storage.

Syn-Aud-Con Offers Benson's Book

NORMAN, Ind. — Synergetic Audio Concepts (Syn-Aud-Con) has published the late Dr. J.E. Benson's *The Theory and Design of Loudspeaker Enclosures*. Benson's work was originally published in the A. W. A. Technical Reveiw in Australia. Synergetic Audio Concepts has received permission to publish the 365-page book which will sell for $24.95.

The book's topics include an exhaustive analysis of the infinite baffle, closed box, damped, vented box, passive radiator vented box and the acoustic resistance controlled systems.

The Theory and Design of Loudspeaker Enclosures can be ordered from Synergetic Audio Concepts, 12370 W. Co. Rd. 100 N., Norman, Ind. 47264. For more information, call 812-995-8212. ▲

Classifieds

Advertisers' Index

With over 20,000 ADATs already in use all over the world, Alesis has made more digital multitrack tape recorders

And with good reason. Alesis was founded on digital technology, so we know what it takes to make the best-selling digital multitrack. The Alesis ADAT®Digital Audio Recorder's sound quality, sample accurate synchronization capability (ADAT Synchronization Interface), fiber-optic digital interface (ADAT MultiChannel Optical Digital Interface), and wide range of peripherals available now, give ADAT owners the creative flexibility they need.

The Alesis AI-1™ADAT to AES/EBU and S/PDIF Digital Interface with sample rate converter lets you transfer audio digitally to or from the ADAT system and external units such as DATs, CDs, and other digital recording formats.

The Alesis AI-2™Multi-Purpose Audio/Video Synchronization Interface by TimeLine (the leader in synchronization products) connects ADAT to the world of video, film and multi-media production using SMPTE, 9 pin and TimeLine Lynx control protocols.

Focus on Compatibility™

Its revolutionary impact on the recording industry has made ADAT the de facto standard in digital multitrack. The enormous number of ADAT users worldwide, the fact that Fostex has licensed the ADAT format for their own digital recorder, and the growing list of leading companies focusing on industry compatibility by becoming members of The ADAT Group™, all mean that when you choose ADAT, you're compatible with a vast array of music and audio equipment, now and in the future. And, you're supported by a network of professionally trained Authorized ADAT Service Centers worldwide.

The adat Group™
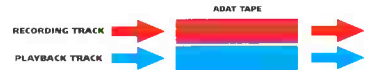
The ADAT Format – made for multitrack

ADAT records eight tracks of 16-bit linear, 48 kHz sample rate audio, with no data compression "tricks" or channel sharing. We chose Super VHS® (S-VHS®) tape as a foundation, then designed ADAT's data structure and heads specifically for the rough-and-tumble, back-and-forth, punch-in-and-out environment of multitrack recording. To make sure that recording one track wouldn't disturb any other track, we divided each helical scan into

| 1 & 2 | 3 & 4 | 5 & 6 | 7 & 8 |

An 8 track, 8mm recorder's helical scan: there are only four audio data blocks, forcing each track to share a block with another.

An ADAT's helical scan: ADAT tracks are safely separated into 8 discrete data blocks. (Both vertical dimensions enlarged for clarity)

eight separate data blocks. Some digital recorders combine data from two different channels into the same data block on tape, which means that each time you record a track, another track must be read into a buffer and actually re-recorded even though it is in "safe" mode.

The 8 track, 8mm helical scan format risks introducing errors into "safe" tracks each and every time you press record.

The ADAT format records each track discretely, as all professional multitrack recorders should.

Bigger is Safer

Microscopic contaminants in the studio aren't just probable, they're statistically inevitable. If the format can't overcome them, they'll cause mistracking, noise, distortion, even total muting of the audio. Formats smaller than S-VHS are more vulnerable to contaminants, dropout, and misalignment, especially when exchanging tapes between machines. One 8mm digital format attempts to squeeze the same amount of sound into one-tenth the tape area that ADAT does. ADAT's S-VHS tape offers more total surface area to meet the demands of digital recording, and its wider 100 micron tracks are five times less vulnerable to being derailed by dust. Because even though technology makes it possible to make formats smaller and smaller, dust stays the same size.

Comparison of tape areas for 1 second of audio: ADAT (1,211 mm²) and the 8 track, 8mm helical scan format (133 mm²).

Actual microscopic comparison of the ADAT tape format and the 8 track, 8mm helical scan format (enlarged approximately 100 times).

*ADAT's wide 100-micron tracks offer an extra margin of safety for digital audio.*

The 8mm's 20-micron tracks squeeze more data into the same area, with little room for error.

than any other company. More than Sony. More than Mitsubishi. More than Yamaha, Akai, and Tascam combined.

More than just a tape recorder– The ADAT System

ADAT, when combined with the BRC™ Master Remote Control, is a complete digital recording and digital editing system with features that no other recorder, analog or digital, can match. The BRC is a full-function autolocator and MIDI/SMPTE time code chase-lock synchronizer. Plus, it controls digital copying between ADATs, like a disk-based recorder, but much simpler to use.

The ADAT MultiChannel Optical Digital Interface digitally transmits up to eight ADAT channels at once over a single fiber optic cable to any track on any ADAT in the system without repatching, all in the digital domain. Now you can "fly in" that perfect vocal part to multiple locations in seconds, with absolutely no generation loss. And our new QuadraSynth™ keyboard has an ADAT digital interface so you can record it without ever leaving the digital domain.

A fiber optic cable for digital connection is included with every ADAT.
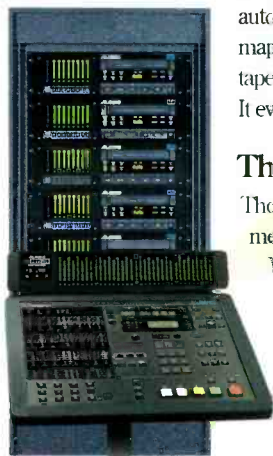
The BRC Master Remote Control, shown with optional RMB™ Remote Meter Bridge, supercharges your ADAT System by adding SMPTE and MIDI synchronization, storable autolocation points, copy and paste digital editing and more.

ADAT/BRC digitally stores important session notes

Instead of scribbling notes on cumbersome studio track sheets, the BRC lets you store 400 autolocation points, 20 Song start points, punch in and out points, MIDI tempo maps, SMPTE offsets, and more in the two-minute data header of the ADAT tape. The BRC's alphanumeric display lets you name each cue point and song. It even has a handy built-in list of 16 standard cue point names you can edit.

L14 "CHORUS 1" 00:25:38:15

Unlike analog autolocators, the BRC can recall 460 points, storable on each ADAT tape for later recall, so you can keep your mind on the project instead of having to remember minutes, seconds and frames.

The ADAT Worldwide Network

Thousands of ADAT Worldwide Network™ multitrack recording group members are reaping the benefits of choosing The ADAT System. As WWN members, they are able to collaborate and exchange ADAT tapes with other talented musicians, producers, composers and engineers throughout the world. Alesis is proud that so many creative people worldwide are using this American-made product, making ADAT the most popular digital multitrack tape recorder in history. The recording professionals below don't endorse ADAT, they *use* it every day. Their credentials speak for themselves. Visit your Authorized ADAT dealer and see what the new standard in digital multitrack recording can do for you.

WORLDWIDE adat NETWORK™

Dave Rouze
Technical engineer for Larry Carlton, currently using ADAT to record all Larry's live concerts. 2 ADATs and a BRC.

Jay Graydon
Two time Grammy® Award winning (twelve nominations) producer, engineer, writer, and guitarist. 4 ADATs and a BRC.

Owen Bradley
Country Music Hall of Famer. Producer of many legendary country music artists. 9 ADATs and a BRC.

Francis Buckley
One of the top dance and pop engineers in Hollywood. 4 ADATs and a BRC.

Web Staunton
Grammy-nominated chief engineer and studio owner. 3 ADATs and a BRC.

Mick Guzauski
L.A.'s leading platinum mixdown engineer. 4 ADATs and a BRC.

Andy Hilton
Owner and Chairman of the largest pro-audio equipment-for-hire company in the U.K. and Europe. Plenty of ADATs.

Ray Benson
One of country music's hottest producer/arranger/writers. 3 ADATs and a BRC.

Tom Size
Has engineered and mixed a wide range of music from rock to legendary jazz. 3 ADATs and a BRC.

Russell Brower
Two Emmys (eight nominations). Sound designer and producer for film, television and major theme parks. 2 ADATs and a BRC.
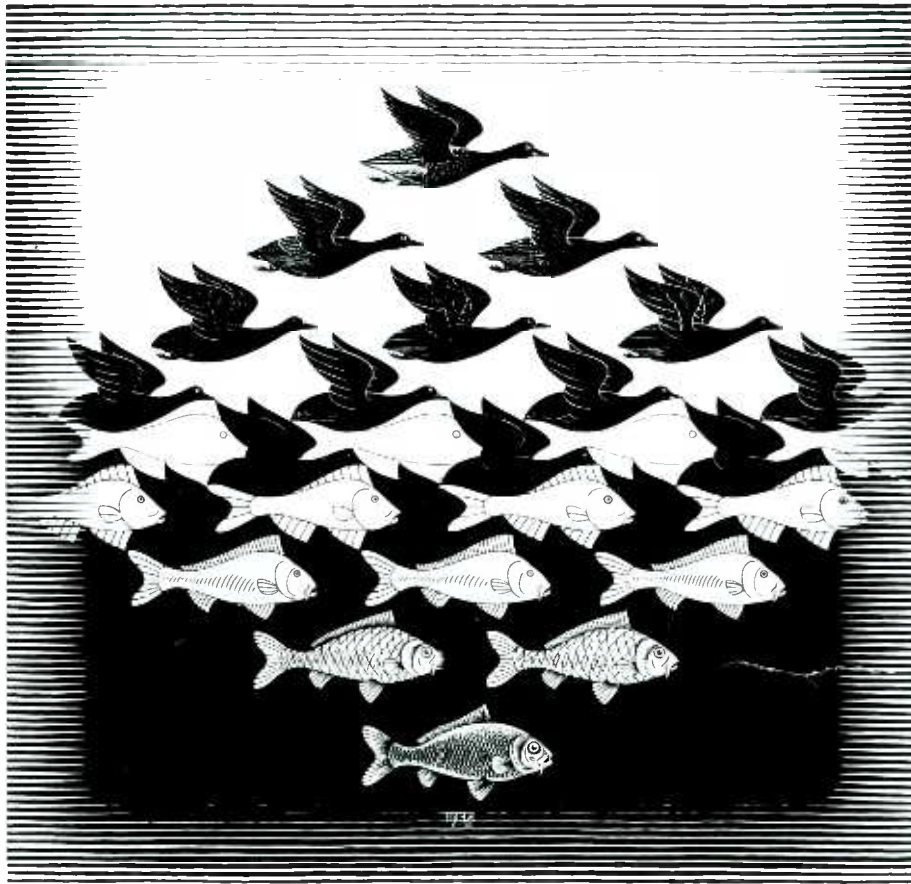
Tim Wilson
Consultant and system designer for leading recording artists and songwriters. Has installed more ADATs than he remembers.

Alesis Corporation 3630 Holdrege Avenue Los Angeles CA 90016

adat **ALESIS STUDIO ELECTRONICS**

For more information, circle 13

www.americanradiohistory.com

If you think only your eyes can play tricks on you...

Study the illustration. Are the geese becoming fish, the fish becoming geese, or perhaps both? Seasoned recording engineers will agree that your eyes *and* your ears can play tricks on you. In the studio, sometimes what you think you hear isn't there. Other times, things you don't hear at all end up on tape. And the longer you spend listening, the more likely these aural illusions will occur.

The most critical listening devices in your studio are your own ears. They evaluate the sounds that are the basis of your work, your art. If your ears are deceived, your work may fall short of its full potential. You must hear everything, and often must listen for hours on end. If your studio monitors alter sound, even slightly, you won't get an accurate representation of your work and the potential for listener fatigue is greatly increased.

This is exactly why our engineers strive to produce studio monitors that deliver sound with unfailing accuracy. And, why they create components designed to work in perfect harmony with each other. In the laboratory, they work with quantifiable parameters that do have a definite impact on what you may or may not hear. *Distortion*, which effects clarity, articulation, imaging and, most importantly, listener fatigue. *Frequency Response*, which measures a loudspeaker's ability to uniformly reproduce sound. *Power Handling*, the ability of a

Models pictured (L-R)
3-Way 10" 4410A, 2-Way 8" 4408A and 3-Way 12" 4412A

loudspeaker system to handle the wide dynamic range typical of the digital domain. And, finally, *Dispersion*, which determines how the system's energy balance changes as your listening position moves off axis.

The original 4400 Series monitors have played a major role in recording and broadcast studios for years. Today, 4400 Series "A" models rely on low frequency transducers with Symmetrical Field Geometry (SFG™) magnet structures and large diameter edgewound ribbon voice coils. They incorporate new titanium dome tweeters, oriented to create "Left" and "Right" mirror-imaged pairs. Refined crossover networks use conjugate circuit topology and tight tolerance components to give 4400A Series monitors absolutely smooth transition between transducers for perfect imaging and unparalleled power response.

If you're looking for a new pair of studio monitors, look into the 4400A Series. We think you'll find them to be a sight for sore ears.