Profile: Phoebe Snow

MODERN RECORDING & MUSIC

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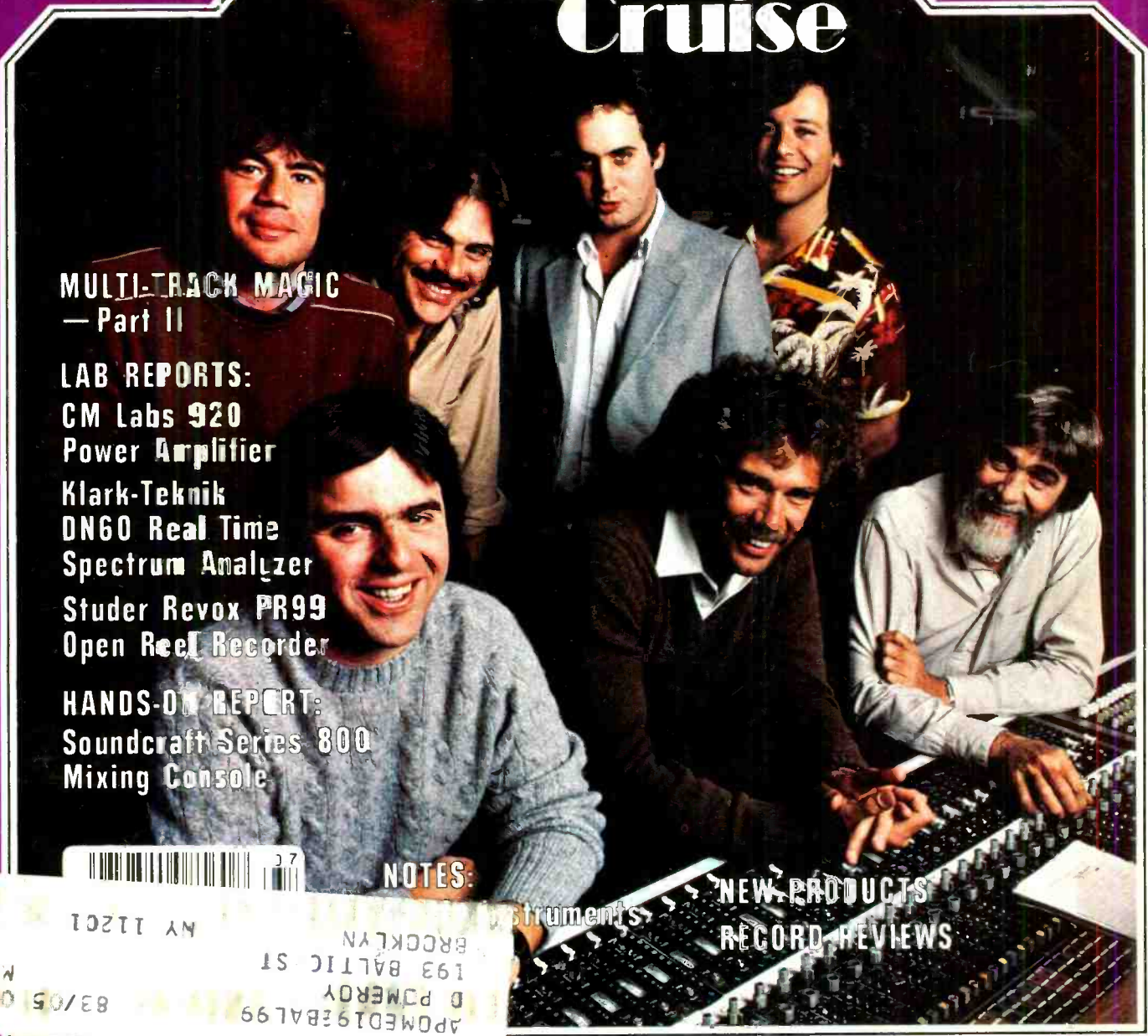
VOL. 6 NO. 10
JULY 1981

a session with Pablo Cruise



MULTI-TRACK MAGIC
— Part II

LAB REPORTS:
CM Labs 920
Power Amplifier
Klark-Teknik
DN60 Real Time
Spectrum Analyzer
Studer Revox PR99
Open Reel Recorder

HANDS-ON REPORT:
Soundcraft Series 800
Mixing Console

NOTES:

NEW PRODUCTS
RECORD REVIEWS

The Unlimited Limiter.

In keeping with MXR's expanding commitment to the professional recording industry, our engineers have designed and built the Dual Limiter. A world class mono-stereo limiter offering total flexibility and ease of operation, the Dual Limiter produces a musically natural response in any compression-limiting application. All of this versatility is built into a compact, rack-mountable package.

The totally unique VCA's at the heart of the Dual Limiter provide an exceptionally wide dynamic range with low levels of distortion. Continuous bass distortion is much lower in level than typical compressor-limiters, allowing more freedom in setting release characteristics.

The Dual Limiter is also a forgiving limiter. Attack and release characteristics dictated by the front panel controls are modified by program dynamics and compression requirements. The slope increases smoothly past the threshold point, allowing a *gradual* transition into compression. Varying the Dual Limiter's threshold region produces a variety of intermediate slopes with the primary slope being that chosen by the slope switch. These features permit apparent dynamics to be maintained even though the dynamic range is being controllably limited.

The Dual Limiter's remarkable versatility is based on the fact that it can be viewed as two independent mono limiters that can be patched together via front panel switches for stereo limiting applications. Each channel has an In/Out switch, Slope switch, Input, Output, Attack and Release controls and an LED display, showing the amount of gain reduction. On the rear are both XLR and ¼" phone jack (ring-tip-sleeve) input and output connectors. Each channel's detector is accessible via rear panel phone jacks to permit external tailoring of the detectors' frequency response. This feature allows for de-essing (reduction of vocal sibilance) and a wide variety of frequency dependent limiting needs.

Because virtually every form of musical signal was used to evaluate the Dual Limiter's response during the initial stages of development, its sophisticated internal circuitry enables it to sound musically *natural* — even at extreme compression settings.

Balanced inputs, the ability to drive 600 ohm loads, +19 dBm input and output and standard rack dimensions (1¾" high) allow the Dual Limiter to be easily integrated into any professional system. With an extremely rugged case, metal knobs and reliable internal construction, the new MXR Dual Limiter reflects the highest professional standards and has been fully designed and built in the U.S.A.

The Unlimited Limiter — MXR's natural response to the question of performance and versatility in a space-efficient and cost-effective package. See the MXR Dual Limiter at your nearest MXR dealer.

MXR Professional Products Group

MXR Innovations Inc.,
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Rochester, New York 14613
(716) 254-2910

Studiomixer

If you plan to make it BIG

A performer or recordist who doesn't have visions of progressing musically and creatively is certainly an exception to the way most musical artists think. The constant self improvement realized through hard work is normally directed toward more than just the satisfaction of being *good*. Most music people want to make it to the *BIG TIME*.

If you fit into this category, we would like to help. Recording is the medium which is going to get you there, and we are the most innovative people in the recording business today. We're **STUDIOMIXER** and our *Totally Modular* approach to mixing console design is revolutionizing the mixer business.

Before now, an expandable mixer meant adding input channels on to it. Most of the time, these input channel expansions were clumsy attempts, resulting in bolt-on input channel sections, bulky external cables, or even hand wiring. And the consideration of output expansion? *Perish the thought.* But the **STUDIOMIXER** *Totally Modular* concept is a *milestone* in mixing console design.

STUDIOMIXER allows the user to start out with as few input and output channels as are required (even just one of each) and grow from the beginning . . . up to 35 input, 8 tape output, 4 monitor, 2 effects, and Left and Right master outputs. Did we say Grow? All the **STUDIOMIXER** owner needs to do to effect this expansion is to purchase an additional module(s) of his choice when his musical and *financial* requirements dictate. A blank module is removed from the Mainframe and the new module is plugged in. All of this requires no soldering and takes about five minutes. Need we point out the obvious service advantages?

When you see and listen to our console, we think you'll agree that **STUDIOMIXER** has the highest performance specifications of any mixer in (and many far above) its price class . . . and you will be *pleasantly* surprised by its low cost.

Think about the *Totally Modular* **STUDIOMIXER** before you contemplate buying anything else. It is a choice which will stay with you for years, growing when you grow. And it will save you money by relieving you from the hassle of trying to dump your no longer adequate *old* mixer for one with the expanded capabilities and quality already built into **STUDIOMIXER**. With **STUDIOMIXER** *we* certainly won't stop you from making it to the *BIG TIME!*

For the name of your closest Studiomixer dealer, or for descriptive literature and specifications on our products, please write to Craig Bullington, National Sales Manager, Amerimex Co., Inc., PO Box 55, Atwood, California, 92601.

CIRCLE 83 ON READER SERVICE CARD

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Professional Product Series

Subject: The Breakthrough of Accessible Automation

CPE-800

Now, for the first time, automation is within the reach of any professional. Roland Corporation, for years a leader in advanced microprocessor-based musical electronics announces a breakthrough in automation with the introduction of the CPE-800 Compu Editor from our professional division—Roland Studio Systems.

The CPE-800 has been designed to provide automation in a simple, economical format for many uses from recording to live performance and lighting. The CPE-800 and its companion unit the VCA-800 Voltage Controlled Amplifier provide automated fader and mute control when inserted between any multitrack deck and mixing console. The CPE-800 can also be used to expand existing automation systems to include echo send, cue mixes and special effects control.

The CPE-800 provides 15 channels of simultaneous fader and mute control, and allows individual channel updates at any time. All motions are timed off a self-contained SMPTE generator/reader, which also allows the CPE-800 to interface with any other system using a SMPTE time base.

The 32k bytes of on-board data storage prevents the punch-in delay caused in automation systems using tape storage. The entire memory can be dumped and re-loaded at any time to facilitate multiple uses. Software includes in addition, a scene-by-scene cue automation system, and internal clock programming for non-recording applications.

The Outputs of the CPE-800 allow the connection to any X-Y oscilloscope for a visual indication of all fifteen fader positions. Fader level comparator LEDs indicate the difference between data and current fader positions to enable smooth punch-ins. Two CPE-800 units can be coupled together to control 30 individual channels or functions.

If you've found the high cost of conventional automation systems to be prohibitive, you're going to like the CPE-800. You'll find that automation is a lot more accessible than you think.

Roland Studio Systems Inc.
1022 S. La Cienega Blvd.
Los Angeles, CA 90035

RSS

We Design the Future

CIRCLE 71 ON READER SERVICE CARD

JULY 1981
VOL. 6 NO. 10

MODERN RECORDING & MUSIC

THE FEATURES

COMING NEXT ISSUE!
The final installment of Multi-Track Magic Studio Notebook, No. 2
Plus all your other favorite features!

Cover Photo: Mark Mander
Pablo Cruise Photos: Mark Mander

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LETTERS TO THE EDITOR

The Trouble With Lists

From time to time, *Modern Recording & Music* provides a listing of schools which teach recording engineering. Publishing such a listing without comment, can actually be a "disservice" to your readers, as it implies that all such schools are essentially the same. A simple list does not take into consideration the intent, duration or academic credentials of the school.

The first distinction should be whether the recording school is "avocational" or "vocational." An *avocational* school caters to those who regard recording as a subordinate interest or hobby. There are very few, if any, external controls over the school or course content. A *vocational* school, by definition, teaches an individual a trade and skills. Therefore, the school is subject to greater governmental and licensing restrictions.

Two other distinctions, more critical to the student interested in a career in recording and in the music business, are the licensing and accreditation of a school. Certainly, the minimum expectation of a vocational school is that it be licensed to operate within the State. State licensing normally imposes health and safety standards, truth-in-advertising restrictions, and some sort of surety bond for recovery of tuition, in case a school should become unable to continue classes and/or goes bankrupt. (In California, this is known as the Student Tuition Recovery Fund.) As a school becomes more established—normally after two years—it may seek accreditation from an accrediting agency.

Accreditation is a lengthy process by which a school voluntarily agrees to establish and maintain certain standards of conduct and educational quality. The accrediting agency, such as the National Association of Trade and Technical Schools (NATTS), will inspect the school facility, staff, curriculum and—impartially—determines whether the school meets the association's standards. If their standards are met, the school is granted accreditations. This accreditation is not static, but rather subject to review and suspension or termination.

Obtaining and maintaining accreditation is often an accurate measurement of the school's commitment to providing high quality occupational training. The College for Recording Arts which was established in 1972, was subsequently accredited in 1974, by both NATTS and by the National Academy of Recording Arts and Sciences (NARAS) Accrediting Commissions, and has maintained its accreditation ever since.

Lastly, the prospective student might want to consider one last important difference between schools: Unlike State-supported or avocational schools, a proprietary school must prove its worth in the marketplace. Proprietary schools, like ours, must answer the most crucial question of all: What is your placement rate? The College for Recording Arts is proud to note that even during the recording industry's most difficult times (1978 and 1979), we maintained a solid 80% placement rate among our diploma graduates. (This is not a hyped figure. Employment figures and documentation from schools whose students receive GI Benefits are scrutinized by the Veterans

WHY SPEND $200 MORE ON A BETTER TAPE DECK WHEN ALL YOU NEED IS $2 MORE FOR A BETTER TAPE.

No matter how much you spend on a tape deck, the sound that comes out of it can only be as good as the tape you put in it. So before you invest a few hundred dollars upgrading your tape deck, invest a few extra dollars in a new Maxell XLI-S or XLII-S cassette.

They're the newest and most advanced generation of oxide formulation tapes. By engineering smaller and more uniformly shaped oxide particles, we were able to pack more of these particles onto a given area of tape.

Now this might not sound exactly earth-shattering, but it can help your tape deck live up to its specifications by improving output, signal-to-noise ratio and frequency response.

Our new XL-S cassettes also have an improved binder system, which helps keep the oxide particles exactly where they're supposed to be. On the tape's surface, not on your recording heads. As a result, you'll hear a lot more music and a lot less distortion.

There's more to our XL-S tape than just great tape. We've also redesigned our cassette shells. Our new Quin-Lok™ Clamp/Hub Assembly holds the leader firmly in place and eliminates tape deformation. Which means you'll not only hear great music, but you'll also be able to enjoy it a lot longer.

So if you'd like to get better sound out of your tape system, you don't have to put more money into it. Just put in our new tape.

maxell
IT'S WORTH IT.

Maxell Corporation of America, 60 Oxford Drive, Moonachie, N.J. 07074

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Administration, as well as State agencies.)

To reiterate, we feel that a simple listing of recording/music business schools is unfair to prospective students. There is a considerable difference between vocational and avocational schools, licensed and unlicensed facilities, and the length, quality and educational commitment of the educational institution.

Prospective students should first determine what they will be requiring of a recording school, and then check carefully to ensure that the school meets their short or long-term needs.

—Leo de Gar Kulka
President
College for Recording Arts
San Francisco, Ca.

Calling Mello

Thank you for printing the Buyer's Guide—I have found it very helpful. One question, though, that I can't find the answer to—who makes the synthesizer called the Mellotron? Is it still available? I can't find the answer here, so any help you can give me will be greatly appreciated.

—Fred Zumpano
Akron, OH

The Mellotron synthesizer is made by a company called Mellotron. They make two synthesizers, in fact. There are the 3 track and 4 track synthesizers. The three track's model number is 400SM, and the newer is the 4 track, for which there is (at least as of yet) no model number. It is a keyboard that looks like a small organ. We were told by Mellotron that it is self-contained and can be hooked up to any P.A. system. Mellotron's address is Box 151, Cornwall Bridge, Connecticut, 06754. Their phone number is (203) 672-6681.

Some Legal Questions

I do "live" recordings on location as a hobby, but I am interested in doing this as a small second job.

What are the legalities and practicalities (law) involved in making a master recording? This would be done with the performers' permission, of course. And how about the legalities of making tape copies of the recording? Does it differ if I make 4, rather than 100? Does it matter if these copies are for the performers (or their families) rather than the general public? I would

new from sunn

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Professional sound engineers agree that the sensitive, accurate controls of SUNN SPL mixing consoles are the finest available. Unlike other manufacturers, SUNN applies the latest in electronic technology then hand builds each product. This assures that original engineering design functions are not lost or distorted through automated assembly. Designed and manufactured at our factory in Oregon, every mixing console is sound-tested, adjusted, then carefully packaged and shipped to SUNN dealers world-wide. SUNN SPL mixing consoles have features, quality and affordable prices not available on our competitors products. Audition SUNN SPL mixing consoles today at your local SUNN dealer.

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make these copies on a demand basis (as they are ordered). Please discuss royalties (for composers, arrangers, performers) and copyrights. Where do BMI and ASCAP fit in? What should a "basement" recordist do to legally make and sell tape copies?

—Ronald B. Freeman
Freehold, N.J.

We are referring you to a helpful source of information: The Southern California Law Review. It is put out by the University of Southern California. Their January 1979 issue was devoted to "Aspects of Entertainment Law." Some examples of the articles that appeared in that issue are: "Legal Protection for Titles in the Entertainment Industry," "The Failure to Pay Wages and Termination of Entertainment Contracts in California: Some Implications of the Labor Code." If you write to the business manager of the Law Review, you can obtain an index issue which will give you an idea of which particular issues will be helpful to you. Then you can order those issues. Their address is:

Business Manager
Southern California Law Review
Law Center
University of Southern California
Los Angeles, California 90007

Another thing you can do is call the Law Center at U.S.C. Their address is:

Law Center
University of Southern California
University Park
Los Angeles, California 90007

Their phone is: (213) 741-6366 or 741-7304.

Also, in the December 1980 issue of MR&M, *we answered a letter in which we gave certain guidelines to getting a song published. Perhaps that letter may answer some of your more basic questions.*

Superscope

In reference to J. Mark Wolf's letter in the Talkback column, p. 22, in the March 1981 edition of *MR&M*, where could I check out the Superscope EC-12B in my area (east of Columbus, Ohio)? Or, where could I write to Superscope to find out this information? I greatly enjoyed your article in that same issue of *MR&M* on Pat Metheny. I know one person who would like to imitate him!

—Roger Robinson
Granville, OH

We checked out the address of Superscope for you, and this is what we came up with. World Headquarters: 20525 Nordhoff Street, Chatsworth, California, 91311. The number there to call is: (213) 998-9333; TWX 910-494-2760. The other address we found listed for them is: 8150 Vineland Avenue, Sun Valley, California, 91352. And we're glad you enjoyed the article on Pat Metheny. Maybe we'll print a follow-up article on 10 things one can do to best imitate his look and style!

Performing Quite a Service

Just wanted to let you know that I think the new "Service Notes" column is really a great addition to the magazine! Keep it up. Thanks!

—David Drake
Traverse City, MI

April was our first issue of "Service Notes," and we're glad you liked it.

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JH-110C-8

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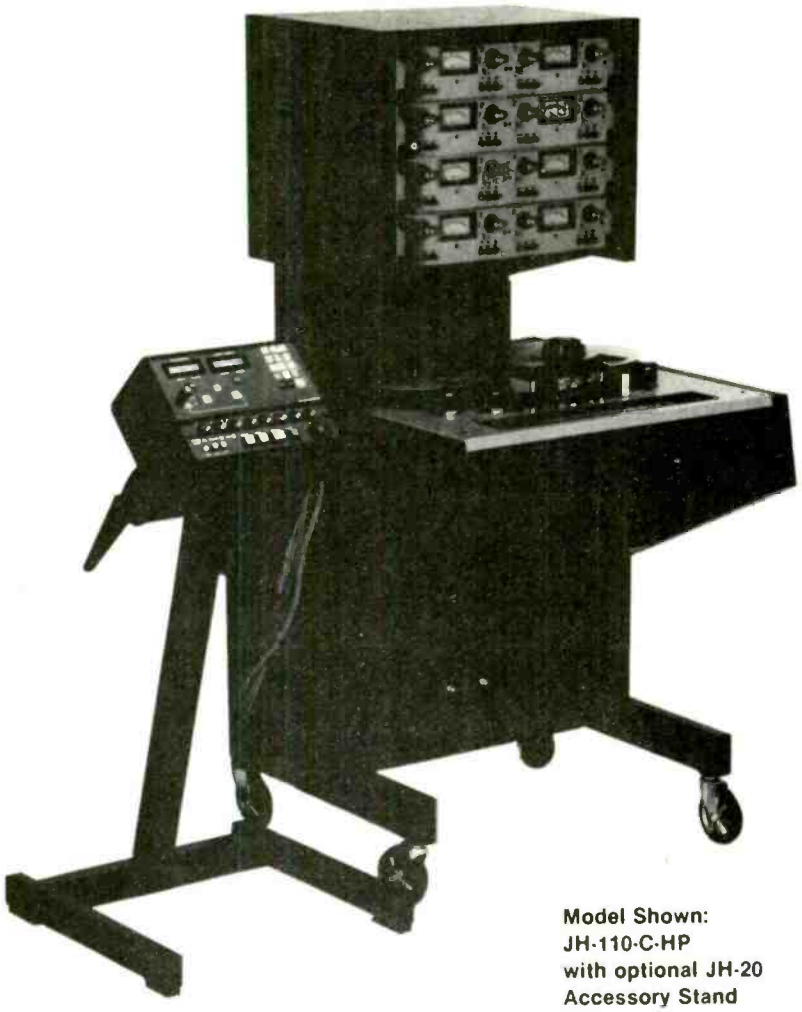

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Look for the next installment of "Service Notes," which will appear in our August issue. Brian Roth will alternate with Craig Anderton in writing the "Notes" column, Brian's column being the "Service Notes" sections. Thanks for writing.

Mic of the Month

First, I want to thank you for the review of the Carvin MX 1202 mixer. It was most helpful.

Since your magazine is a fountain of information, I have come up with an idea so you can give us information-thirsty "sound hounds" just a little more info. On the table of contents page of *MR&M* there is always a picture of a microphone. My suggestion is to feature a picture of a different microphone each month, with an enclosed caption, perhaps at the bottom of the page, with the manufacturer's name, specs, and what the retail cost would be.

This is just a suggestion, but I know I would appreciate such a feature. I find it hard to find information on specific microphones without actually having to write to the individual companies themselves.

—John McCullough
Mattoon, IL

Well, we'll present your suggestion to the decision-makers, but we can't promise anything. The best way to get information on specific microphones is to write directly to the companies. Our Musical Newsicals column also comes up with some reports on microphones. We suggest you check them out as well.

If You Believe This One...

About a year ago I was doing a session on which the drummer executed a slightly uneven drum roll on an otherwise perfect take. He was a bit angry at himself, and wanted to do the take again.

"That's not necessary," I said. I explained to him that the drums were spread across several tracks, and with a little practice, we could find the spot where he hit the tom lightly and bring it up in the mix.

I proceeded to demonstrate, but this guy had a terrific sense of humor and said, "Yeah, yeah, Phil. Famous last words--'The check's in the mail,' 'I won't.../(I won't use my teeth),' and 'We'll fix it in the mix.' "

So was born the "Downtown Recorders List of Famous Last Words and Commonly Heard Excuses." We posted the first three and as the year wore on, the list got longer and longer.

We thought it was time to share it with you and your readers.

—Phil Adler
Chief Engineer
Downtown Recorders Inc.
Boston, MA

FAMOUS LAST WORDS AND MOST COMMONLY HEARD EXCUSES
Compiled by Phil Adler

1) The check's in the mail.
2) I won't ...../(I won't use my teeth).
3) We'll fix it in the mix.
4) We'll fix it in the mastering.

5) Last take. This one's gonna be perfect.
6) I'll get it this time.
7) We'll add some digital and some reverb...it'll sound great.
8) Play all the way through. We'll take out what we don't want.
9) That's the way it's *supposed* to sound.
10) We *wrote* it that way.
11) Trust me.
12) Just let me try one thing.
13) This is just the demo...When we do the album...
14) We'll overdub that later.
15) I don't understand. It sounded so good in rehearsal.
16) We'll punch it out. It'll be okay.
17) It'll sound terrific on a car radio.
18) I don't know what it sounds like in there, but it sounds like shit out here.
19) We didn't change anything... honest.
20) Let's argue about it, then try it.
21) You've got to move back to hear the low end.
22) It sounds better up by the board.
23) Take the rough mix home and decide tomorrow.
24) A little won't keep me awake too long.
25) It's pure.
26) No problem.
27) We're working on it.
28) I'll see what I can do.
29) It's gettin' there.
30) I was just about to call you.
31) I'll get back to you.
32) I love you.
33) You'll have em in two weeks.
34) Can you guys give me 15 minutes?
35) Pay no attention to the man behind the curtain.

Thanks for the list. Now, enlightened, some of us will know what to watch out for!

That L&M Issue

Can you please tell me if you've ever had an article featuring Kenny Loggins (or Loggins & Messina) or the Doobies? If you can tell me the dates of these issues if you do have them, I'll buy them. I also need to know the price.

—Diane Fadem
New Rochelle, N.Y.

The very first issue of Modern Recording *included an article entitled "A Session with Loggins and Messina," written by Jack Breschard and Ellen Wolff. It appeared in the October/November 1975 issue (back in the days when* Modern Recording *was published bimonthly). The only problem, though, is that we have no copies of that issue to spare. It is indeed a collector's item. But perhaps one of our readers out there has a copy that he or she wishes to dispose of for a small sum. The magazine was $1.50 per issue in those days. What will you pay for it now? Good luck. If anyone out there has a copy, let us know, and we will get you in touch with Diane Fadem. And we have never published an article on the Doobies.*

Who's Afraid of PCM?

I would like to respond to Mr. George Klabin's comments in the April Talkback column (see "The Mixdown Lowdown," page 20) concerning PCM [pulse code modulation] converters.

Mr. Klabin states that "they are extremely sensitive and complex pieces of machinery and you would need a storehouse of technical knowledge...to keep it properly aligned and to get everything from it." If anything, this is more true of analog recorders, and a surprising number of people with limited technical background do very well with analog recorders. In fact, the

TDK brings two new standards to open reel.

Raising sound standards is nothing new to TDK. For years, TDK cassettes have set reference standards in metal and high bias. Now TDK announces two breakthroughs in open reel—GX and LX. Both are formulated to be fully compatible with your present system. You don't have to rebias to appreciate them.

TDK GX Studio Mastering tape handles the most critical demands of live music mastering beautifully. TDK's new ultra refined ferric oxide particle gives GX superior MOL, low distortion and a wide dynamic range. Equally impressive is TDK LX. Its super refined particle gives it high performance with low noise and low distortion throughout an extended frequency range. LX is ideal for both professional and audiophile use.

The refinements don't stop with the formulations. A unique calendering and binding process rivets the particles to the tape surface, making dropouts practically a thing of the past. A special graphite and carbon backcoating, found on all GX and most LX tapes, reduces friction for the smoothest possible winding. At the same time, it prevents static discharge and reduces wow and flutter.

These high standards are carried through to the newly designed 10" metal and 7" plastic reels. Each has a separately molded hub and flange to ensure circularity and high strength. If you think open reel has gone as far as it can go, listen to the finest. TDK GX and LX. They could open up a whole new standard of recording excellence.

TDK

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current PCM converters need less adjustment, and what little is necessary is done by looking at a voltmeter or scope while turning a screwdriver. In everyday use, one needn't know a bit from a byte to obtain good results with this equipment any more than with an Atari video game.

While I realize digital technology is relatively new to the recording studio, I believe that under all the mystique is a fundamentally simple science, and as the hardware becomes more cost-effective for semi-pro recording, I hope that unwarranted fear and misconception will not slow its application.

—Mike Shields
Development Engineer
Sony Technology Center, Inc.
Palo Alto, Ca.

We'd like to hear both conflicting and supporting views. So please write to us with any comments, opinions, or added information you can provide.

Get That Book!

In the August, 1980 issue of *Modern Recording and Music*, in the "Product Scene" column, you reviewed a book entitled, *How to Build a Small Budget Recording Studio from Scratch*, by F. Alton Everest and published by Tab Books.

I wish to buy a copy of this book, but I have no address or other information as to the publisher. Any help you could give me in this matter would be greatly appreciated. Thank you for your help and consideration.

—Vic Thomas
Madison, NJ

We suggest that you write to Tab Books to obtain a copy of How to Build a Small Budget Recording Studio from Scratch. *Their address is Blue Ridge Summit, Pennsylvania, 17214. The book is No. 1166 in the series. The copyright date is 1979.*

More Vocalists

I really enjoy your magazine, but you should have more in-depth interviews with the vocalists. I'm a bass man, but you know we'd all be pretty lost without the good vocalists who are recording. I'd like to see what some of these people have to say about the problems they encounter on the stage and on the recording scene. For example, what problems they have with bands. Anyway, that is my suggestion to you.

—Dave Dykes
Austin, TX

We've interviewed Journey's lead vocalist Steve Perry—April 1980, p. 48, Genya Ravan and Ronnie Spector in June 1980, Grace Slick, March 1981, and this month—Phoebe Snow. See? We've hardly neglected the vocalists!

Comments

Michael Beigel wrote to us regarding Craig Anderton's review of the Beigel Envelope Controlled Filter which appeared in the February 1981 issue of MR&M.

Craig Anderton's review of my Envelope Controlled Filter was an accurate and very well thought-out description and evaluation of the product. I appreciate his creative and constructive descriptions and suggestions about the product.

Two minor points in the article need correction. Both are related to the audio-signal processing section. Craig

notes that the potentiometers are of "adequate" quality, but that only one pot carries audio. Actually, none of the potentiometers carry any audio signals that pass from the audio input to the audio output of the device. The "sensitivity" pot carries audio signals, but these are converted into envelope control voltages. The "Peak" pot, which might be suspected of carrying audio, actually controls an optoisolator circuit which in turn controls the "Peak." So the audio actually goes through a photo-resistor.

Craig states that one dual optoisolator controls the frequency of the filter, while four others control the switching functions. This is all true, but there is an additional dual optoisolator controlling the Peak or "Q" of the filter. Not only does this avoid passing the signal through a potentiometer, but it allows the envelope signal, in addition to the Peak control, to dynamically vary the filter Peak.

Craig also refers to some "distortion" which occurred when he played chords while using very fast envelope settings. It's actually fast frequency-modulation of the filter's cutoff frequency by the complex envelope created by playing a chord. While some chords may sound "terrible" with this setting, there are also some note combinations that sound very nice! Simple intervals of octaves, fourths or fifths sound about as clean as single notes. The more "beating" there is between the notes in a chord, the more complex and noise-like the filter modulation will become. The frequency range and sweep of the filter also can change this effect from a muddy, unpleasant sound to a high plume of "active noise" drifting over the top of your musical signal. It depends *how* you set things and *what* you play into those settings. This is one of the drawbacks, and also one of the advantages of a complex audio processor.

Again, my thanks to Craig and *Modern Recording & Music* for your excellent work.

—Michael Beigel
Beigel Sound Lab
Warwick, NY

Rhodes Mark III

On a visit to one of my friends, I came across *Modern Recording and Music* magazine, the November 1980 issue. In the Musical Newsicals column was mentioned the Rhodes Mark III EK-10 electric piano. I decided to purchase one since I have a newly formed band going. However, when I attempted to circle the appropriate number on your service card, I discovered that the card had expired. I do not have the address of the Rhodes Keyboard Company. Would you be able to provide me with an address and phone number so that I may get in touch with them to find out about purchasing the Mark III EK-10? Thank you.

—K. Listhrop
San Juan
Trinidad, W.I.

Rhodes keyboards are made by Fender/Rogers/Rhodes. They are a subsidiary of CBS Musical Instruments. Write to them at: CBS Musical Instruments, 1300 E. Valencia Drive, P.O. Box 4137, Fullerton, California 92634. The number is: (714) 879-8080.

Touching Up Hands-On

The following letter is from John Murphy, the writer of our Hands-On column. John felt that further clarification

of his Hands-On report of May, 1981 on the Kelsey Pro-4/3 Mixer put out by Dallas Musical Instruments was needed. In particular, he clarifies the noise performance reported. For any further information, write to Dallas Musical Instruments at 150 Florence Avenue, Hawthorne, N.J., 07506. Their phone number is 201-423-1300.

Our original noise measurements on the Kelsey mixer (Hands-On Report, May 1981) were made with both the mixer and the separate power supply seated in the bottom portion of the road case supplied with the unit. Since that time, the manufacturer has informed us that the mixer's noise performance is considerably improved when the power supply module is located away from the mixer; our measurements have since confirmed that with the power supply away from the mixer, the output noise is reduced 10 dB or so compared to the noise levels previously reported. We have provided below the noise performance figures originally reported along with the noise levels observed with the power supply located five feet away from the mixer.

In the conclusion of our original report, we commented that the only significant drawback of the Kelsey mixer was its somewhat higher than average output noise levels. With the power supply located away from the mixer, the board's noise levels are in fact some of the lowest we've seen for a sound reinforcement mixer.

—John Murphy
Technical Editor
Modern Recording & Music

Getting Our Hands on Yamaha

Has *MR&M* done a Hands-On report on any of the new Yamaha mixing boards? I'd like to know which issue it was, so that I could obtain a copy.

—Ronny Patterson
San Angelo, TX

Modern Recording & Music *did a short Product Scene announcement on the M Series of mixing consoles by Yamaha in the October 1980 issue. In the March 1981 issue a Hands-On*

NOISE PERFORMANCE (Note: 20 kHz Bandwidth, unweighted, 150 ohm source impedance)	**Power Supply in case**	**Power Supply out of case**
Equivalent Input Noise:	−109.1 dBV	−121.6 dBV
With all faders at minimum settings noise at the output is:	−82.6 dBV	−87.3 dBV
Raising the main faders to a nominal setting, output noise is:	−70.0 dBV	−71.8 dBV
As above, but raising the submaster faders to a nominal setting, output noise is:	−67.6 dBV	−71.8 dBV
As above, but setting one input channel for a −40 mic input with the channel fader at nominal, output noise is:	−61.5 dBV	−70.1 dBV
With four channels set as above, output noise is:	−53.4 dBV	−66.3 dBV
With twelve channels set as above output noise is:	−50.5 dBV	−61.2 dBV

The world's most popular backup group.

Take a look behind the scenes at today's hottest concerts and you'll find some legendary names in sound equipment. From compact, combo twins to massive sound reinforcement arrays. And if you could look inside these systems you'd probably find another name—JBL. Because for over 30 years professional musicians have been depending on JBL loudspeakers to deliver the performance and sound quality they demand.

At JBL we're very proud of this reputation. And we're constantly working to maintain it through product improvements and innovations.

Case in point: the new JBL E Series. Designed with advanced transducer technologies and materials to achieve a higher level of performance than any other loudspeakers on the market.

Power capacity, for instance, has been increased substantially through the use of new high-temperature adhesives and optimally constructed voice coil formers. We've also engineered the E Series to be the most efficient musical instrument loudspeakers we've ever produced. This combination of high power capacity and efficiency gives the speakers outstanding dynamic range as well as significantly higher maximum acoustic output.

And the improvements aren't limited to just power capacity and efficiency. The E Series also features our unique Symmetrical Field Geometry (SFG) magnetic structure. This new magnet design provides extremely high accuracy with minimum speaker distortion.

Of course, the one thing we would never change is the unmistakable sound quality of our speakers. So the E Series maintains the distinctive tone character that has made JBL loudspeakers an industry standard.

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Specifications	E110 (10 in)	E120 (12 in)	E130 (15 in)	E140 (15 in)	E145 (15 in)	E155 (18 in)
Primary Application	Lead or rhythm guitar, piano, vocals, line array	Lead or rhythm guitar, electric piano, organ, vocals	Lead or rhythm guitar, electric piano, organ, vocals	Electric bass, low frequency reinforcement	Organ, synthesizer, low frequency reinforcement	Electric bass, subwoofer, low frequency reinforcement
Power Capacity						
Continuous program	150 W	300 W	300 W	400 W	300 W	600 W
Continuous sine wave	75 W	150 W	150 W	200 W	150 W	300 W
Sensitivity	98 dB SPL	103 dB SPL	105 dB SPL	100 dB SPL	98 dB SPL	100 dB SPL

JBL Professional Products Division

CIRCLE 140 ON READER SERVICE CARD

report was done on the Yamaha F1040 Frequency Dividing Network (a crossover.)

Sorry, but as of yet we haven't done a Hands-On report on any of the new mixing boards. But perhaps soon...

New-Wave

I tremendously enjoy reading *Modern Recording & Music* for its technical information and the intelligent approach to music. But I have one complaint: I want to know why new wave music, or modern music if you prefer, is relegated to the review section only? Surely, you can't be of the impression that the music hasn't grown since the days when the Sex Pistols and their ilk first began. Current records—some of which have been covered in your reviews—by groups such as the Clash, Boomtown Rats, Police, Jam, Specials, Cars, Ramones, and others are more interesting than the recent work of artists given more coverage, such as Leon Russell, Grace Slick and Rick Wakeman. I'd like to know what goes into the making of these records, how these groups approach their recordings, how they've grown since their initial simplistic records, yet remained so vital, and how they go about production, etc. To my knowledge, *MR&M* has avoided new wave like the plague, and I think it's time for a few in-depth articles about the most exciting music being made today. Otherwise, keep up the good work.

—Greg Hall
Bronx, NY

If the wave hasn't left the shore, we may take your advice and delve into the recording techniques of these new-wave riders. After all, if surf's up, and the tide is high, why should we just keep sitting on the dock of the bay?

Test Idea

I am writing to suggest a topic for an article I feel could be of interest to many of your readers. In the spirit of your articles on cassette and reel tape specs, I would like a rating of various ¼-track, 10½" reel machines (½ track options could be included). Using your testing facilities, which give your equipment profiles the most useful information of any I've seen, you could compare dist./wow and flutter/S/N/and visual frequency response for various levels. Each deck could be rated at 7½ and 15 IPS, with data on results of 2 or 3 different tapes being used (Maxell, Grand Master, etc.). This would allow comparison of machines with variable bias switches to give their "best" readings. Variables of dist./S/N/and freq. response would have to be optimized at a given-set level for each speed and tape, for valid comparison of the other variable under test.

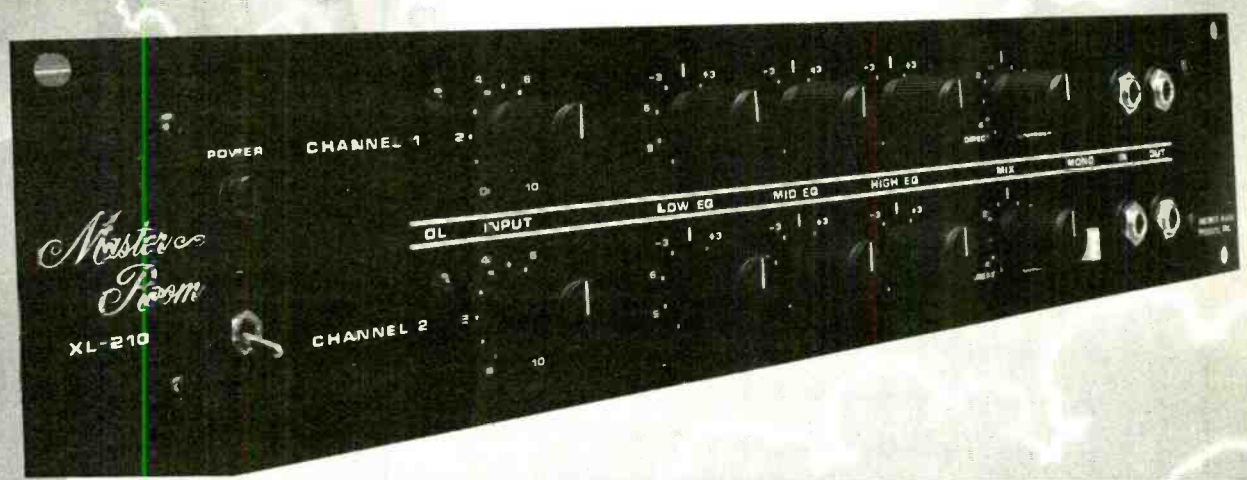
I would like to see Revox, Sony, Technics, Tandberg, and Teac rated. My own reasons are because my B 77 (which you among other publications thought highly of) does not outperform my 10 year old 7 inch Sony deck. Some valid data, done under the same conditions, by the same people, at the same time, would be of great interest to the serious amateur recordist.

—Gordon Jackson, Jr.
Gales Ferry, CT

That's quite an ambitious and interesting idea. The way in which you suggest testing is also well controlled. Perhaps you will see your idea in print. I don't want to promise anything, but I will say that it's one of the more elegant test ideas I've seen in a while.

A NEW DAWN IN REVERBERATION!

XL-210



UNMATCHED PERFORMANCE AT AN AFFORDABLE PRICE!

For years companies have tried to develop a self-contained, rack-mount reverb of professional quality that would sell for under $1,000. All attempts have been based on the same basic design, some using signal manipulations in an attempt to conceal the inherent inadequacies of the reverberation elements. Not one of these designs successfully eliminates the unwanted side-effects such as boing, twang and flutter.

The new MASTER-ROOM™ XL-210, however, incorporates revolutionary technology (patent pending) which provides smooth, natural sounding reverberation without unwanted side-effects . . . even on the most demanding percussive material.

The XL-210 operates in true stereo as well as full mono. This unit can be used with the echo/effects section of any console or can be connected in the main signal path. The versatile EQ allows the user to effectively simulate the reverberant sounds of a live chamber, plate or concert hall. The XL-210 is ruggedly built for road use and is triple-isolated to prevent acoustic feedback.

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CIRCLE 48 ON READER SERVICE CARD

TALKBACK

"Talkback" questions are answered by professional engineers, many of whose names you have probably seen listed on the credits of major pop albums. Their techniques are their own and might very well differ from another's. Thus, an answer in "Talkback" is certainly not necessarily the last word.

We welcome all questions on the subject of recording, although the large volume of questions received precludes our being able to answer them all. If you feel that we are skirting any issues, fire a letter off to the editor right away. "Talkback" is the Modern Recording & Music *reader's technical forum.*

Potential Potentiometers Posing a Problem

About three months ago I completed assembly of my own 24-channel recording console, and its performance is outstanding—even *I* was slightly surprised. Now I would like to add VCA control to my board, and that has posed a problem.

All the information I have sent for and all the schematics I have seen so far for VCA control circuitry show the faders as a linear taper, and my board has all log audio taper faders (they're all Penny and Giles).

Since I cannot afford to replace all my faders, I was wondering if it is possible to use log audio taper faders and, if so, could you please tell me how to go about it?

—Ernest A. Russo
Philadelphia, Penn.

In order to answer this question completely, it is first necessary to explain the difference between the two types of controls, "linear" taper and "log," or "audio," taper potentiometers. These controls differ in the way in which the resistance of each changes for the same amount of control motion. This difference is shown in *Figures 1* and *2*. The "audio" taper is intended for use in applications in which audio signals are applied directly to, and processed by, the control. The logarithmic taper allows a given amount of motion of the control to correspond closely with the *perceived change* in volume that results from this motion. This correspondence is a result of the ear's more-or-less logarithmic response to changes in volume.

Linear taper potentiometers are used (most often) to control DC voltages and other non-audio signals. In VCA controls, the linear taper "fader" does not *directly* control or process audio signals. Rather, the position of the fader determines the value of a DC voltage which in turn controls the gain (amplification) of an audio amplifier (the Voltage Controlled Amplifier, or VCA). The gain-to-voltage characteristic of a VCA is generally given in dB of gain change-per-change in voltage. For example, the dBX 2001 VCA has a control characteristic given as 20 dB/volt.

So, unfortunately, the answer to your question regarding the use of log potentiometers where linear ones are called for is no.

—Peter Weiss
Contributing Editor
Modern Recording & Music

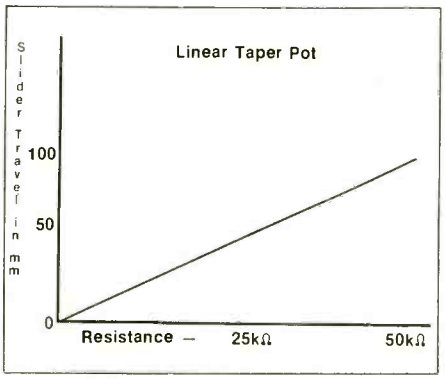


Figure 1

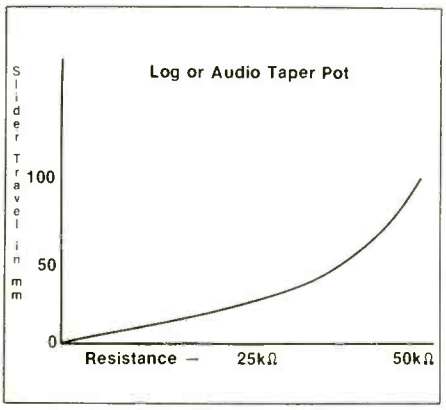


Figure 2

Specifically Mandel

Do you have plans to run an article concerning special effects for the guitar and how to best achieve them? I am most specifically interested in the different approaches used in obtaining sustain: what is the most efficient method, what major guitarists today use, and why they select the approach they do. I only play guitar for recording purposes and find that sustain foot pedals are extremely noisy in this particular application. I mic my amp, but would go direct to the board for some things if I knew how to obtain decent sound quality.

Specifically, I would like to know how an artist like Harvey Mandel is able to play his guitar seemingly by just touching it, much in the manner that the Chapman Stick is played. Harvey has some gadget set directly into the front of his amp. Is this a custom-made piece and, if so, can you find out by whom and if it is available to the public?

—Duane C. Standish
Aurora, Co.

To get the specifics necessary to answer your question, we called Harvey Mandel who very kindly took some time out to answer our questions. Harvey described his two-handed playing technique of tapping the fingerboard of

his guitar as similar to both the way the Chapman Stick is played (as you mentioned), and the technique of rock guitarist Eddie Van Halen. There's no secret electronic gadget involved, it's simply a technique that he's perfected.

Aside from his unique playing style, Harvey insists that any special effects he uses are strictly standard. He has a rack that contains a preamp, a compressor, two digital delays, a flanger and a doubler which allows him to get the same sound whether he's performing "live" or working in his home studio. In addition, he tells us he also uses an ARP Avatar (a synthesizer that allows a guitar player to achieve a wide-range of sound effects and instrument sounds).

Harvey wondered where you got the idea that there was something built directly into his amp; he tells us there's nothing there—aside from what should be, of course. Currently he says he's using an Acoustic 150, but that he occasionally uses other amps as well.

For those of you who might be unfamiliar with Mandel's work, the following is a selected discography which we think represents the best of Harvey's work: Baby Batter *(Janus/GRT);* The Snake *(Janus/GRT);* Cristo Rendentor *(Philips/Mercury);* Games Guitars Play *(Philips/Mercury);* Righteous *(Philips/Mercury).*

As for your question concerning a round-up on special guitar effects and who uses what—it sounds interesting to us and we'll see what we can do for you in the months to come. In the meantime, keep your eyes peeled to Musical Newsicals and NOTES for both what's new for you, and some advice on getting the most out of it *once you* get it.

A Reel-y Magnetic Attraction

Is it necessary to periodically demagnetize tape reels that are made of metal?

—Harvey Goldsmith
New York, N.Y.

No. That was easy, wasn't it? However, I hear echoes of some very good advice a high-school biology teacher used to give us. He said, "Never say *never*." So, in keeping with this admonition, perhaps I should amplify on my answer just a bit.

To backtrack, for the most part, tape reels are made of aluminum which is non-magnetic. Although it seems inconceivable that any tape-reel manu-

facturer would be lame-brained enough to fabricate a reel intended for magnetic tape use with magnetic materials, it is *not* beyond the realm of possibility. Also, some people do use 8 mm film reels to store tape and some of these are made of steel.

If you are having doubts, take the suspect reel and a magnet and see if the magnet attracts the reel—after removing the tape, of course. If the reel is drawn to the magnet then it is a ferrous material and should be demagnetized. I'd advise you to demagnetize the reel about every six months or so or at least once a year, and right after you bring a magnet near it. (Don't forget to remove the tape from the reel first, though.)

—David Moyssiadis
Contributing Editor
Modern Recording & Music

The Formula

My questions concern recording levels. With a Teac 4-track deck and a small (6 into 2) mixer, you have three level controls to deal with: one on the deck (in/out), one on the mixer channel, and your mixer master volume. What is the best setting for these in order to minimize tape hiss? Should I run the deck input wide open and control my signal from the mixer channel or the master knob? Is there a good formula to follow or what? Also, how do the pros do it on a big-time rig?

—David Moore
Norman, OK

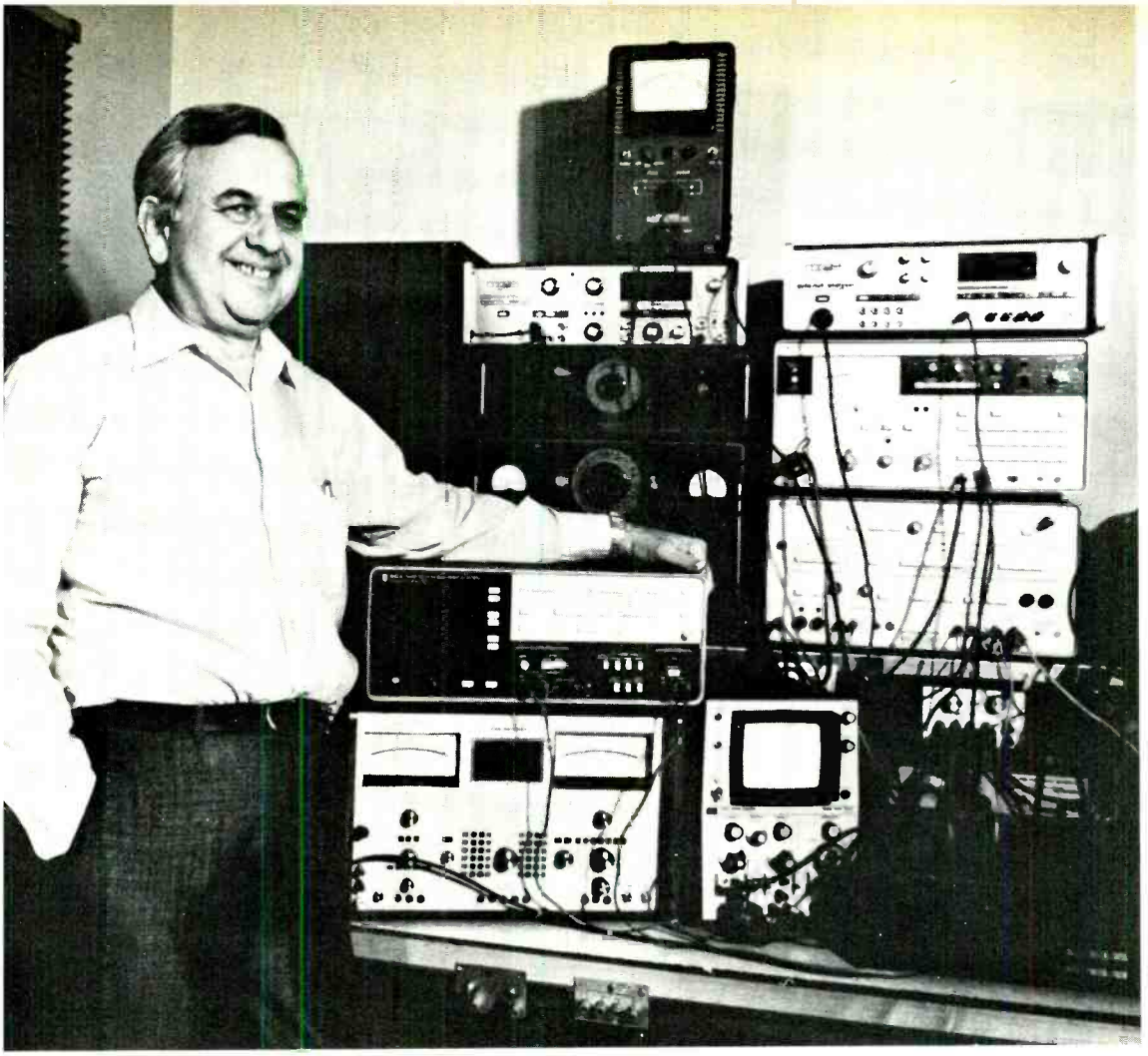
There is no formula to follow that is simple, quick, and easy—at least none that I know of. There is, however, a formula.

First, consider the overall gain structure of one piece of gear at a time, let's say, the mixer. Your description indicates that the mixer has at least two separate stages, so as an example let's assume that there are two. Each of the stages in a device can be assumed to be optimized for noise *within* the mixer, that is, from stage to stage. Mixers with metering to indicate absolute levels can be more easily calibrated, but even those without meters can be set up for optimum noise characteristics. Generally, designers try to provide some headroom in each gain stage. The amount of headroom is what you need to determine in order to make the most of the range of gain available between the noise floor—the point where the noise becomes troublesome—and clipping the gain stage or stages. You need to consult the owner's manual for your mixer, where the manufacturer will state what the nominal input level is for a given input, and then work on the assumption that there will be somewhere between 10 and 20 dB of headroom above the nominal figure quoted. If you have a voltmeter and an oscilloscope, you can measure the actual headroom above the nominal input level by assigning that level the arbitrary "0 VU," or in this instance dB, and measuring the difference in volts between the nominal quoted number, and the voltage at which the 'scope shows any change in the output waveform as you increase the input up from nominal. You can readily establish a limit for yourself as a guideline in this manner, and even a more accurate limit if you use a distortion analyzer in addition to the 'scope.

Operating at the level you find to be 10 dB below unacceptable distortion, will provide you with the capability of recording (passing a signal) sounds which have a peak-to-average ratio, of 10 dB. I would recommend this for maximum noise performance of the individual device, but keep in mind that you will clip off the tops of signals that exceed 10 dB above your arbitrary "0 VU."

In the real world, dynamics are truly enormous, and it's not at all uncommon to find peaks 40-60 dB higher than the average levels of musical instruments like drums, hi-hat and snare, cymbal bells, claves, tambourines, triangles and the like. In order to get reasonable results with instruments whose duration is short, energy small, and dynamics large, it is necessary to stretch the limits of mics, mixers, and tape decks, so you may need to record, say, a triangle so that (with a meter) you would read perhaps at the bottom end of the scale −20 dB! This would give you 30 dB of headroom, and 20 dB more noise than if you had recorded the sound to read "0."

On a big commercial mixing console, the gain structures are carefully laid out to provide the minimum noise and the maximum headroom. Manufacturers are constantly extolling the virtues of the newest, quietest, widest dynamic range, and lowest distortion circuits in their consoles. Papers are published and research done to reduce noise and distortion, and increase



Len Feldman on the Sound Technology 1500A Audio Test Set:

"To my mind, a brilliant combination of audio, video and computer/ microprocessor technologies and applications."

"When I first saw the first 1500A's, I knew I had to have one . . . because I knew it would cut down my test time considerably. When it came, it exceeded my expectations. It not only increased the accuracy of the measurements I was making, but it cut down on the time involved by 75%. With the 1500A I could actually do a test in ¼ the time, with greater accuracy!

'My laboratory is involved strictly in product testing, for both publications and for manufacturers' proposed new products. Everytime we turn on the lab, we're using Sound Technology products . . . and I'm finding more and more ways of using the 1500A in applications that I'm sure Sound Technology never intended. I feel rather proud that I recognized some of its potential perhaps even before they (Sound Technology) did.

"I also own the hard printer and use the printouts in my test reports in the magazines because of the credibility it gives the reader of the reports. The idea of getting a hard copy readout from whatever is on the screen of the 1500A is just incredible!

"The people at Sound Technology have been marvelous to me. My unit was one of the first shipped and Sound Tech has been very cooperative by updating the software on my unit as they made production changes. I just happen to love those people, and my 1500A."

Len Feldman tests products for various professional and consumer publications, and performs product testing for numerous manufacturers. Mr. Feldman purchased the second Sound Technology 1500A Audio Test Set manufactured. For information on how the 1500A can increase your testing accuracy while reducing testing and set-up time, call today. It will clean up your act.

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CIRCLE 94 ON READER SERVICE CARD

dynamic range, to the point where it seems this is the main occupation of some console manufacturers.

In order to get the best results with the tape deck, you should again follow the recommended procedure. With most Teac decks, the optimum setting puts the input level control knobs at about 2 o'clock. Some even have a "CAL" marking on the dial to indicate the position, or a detent in the control rotation.

The A-3340S and A-3440 are rated at 100 mV and 60 mV nominal input, respectively. Their input level controls function as simple voltage dividers, so you can set the meters on the deck to read 0 VU when you are feeding in a signal from an oscillator whose voltage is equal to the nominal *output* of your mixer, or as discussed before, some arbitrary lower level to allow more headroom.

—Drew Daniels
Applications Engineer
TEAC/Tascam Professional
Products Group
Montebello, Ca.

Punch-In Paucity a Performance Permanence?

As many of your readers are probably doing, I am working with a Dokorder 7140 4-track machine. This machine, along with a dbx 124 unit, adequately meets my fidelity needs. However, I find it lacks the ability to punch in and out of record when the tape is moving. I realize this machine was never intended to perform this function. Is there some way I could alter either the machine's record preamplifiers or solenoid logic to realize this function?

—Ed Everaert
Kingsville, Ontario, Canada

We called our Dokorder "Answer Man," Mr. Hirozo Kambe at the Dokorder Service Center in California, to find out if the modification you want to make is advisable. Mr. Kambe did not recommend it, basically for the reason you brought up: the machine was never intended to perform this function and, should you find a way to make the modification, it might not work as well as a machine that was bought with this feature. Mr. Kambe felt that you would not be able to eliminate the noise of the punch-ins successfully, and perhaps frustrate yourself further. Should you wish to pursue this idea further, please contact Mr. Kambe directly at 714-966-0787, or write him c/o The Dokorder Service Center, P.O. Box 8, Lawndale, California 90260. And, as always, the lines are open, as the woman once sang. If you have made such a change—successfully, only, please!—on your own machine, please drop us a note and share your "how-to's" with us.

Bringing Mysteries to Ground

Rick Chinn of Tapco in your February 1981 column (see Talkback, page 21, "Expansive Advice") touched on a subject that's long been a mystery to me: proper grounding. He described a good grounding technique for use within a piece of electronic equipment. Could you expand on this and also describe the proper method for grounding and interconnecting audio equipment (in-

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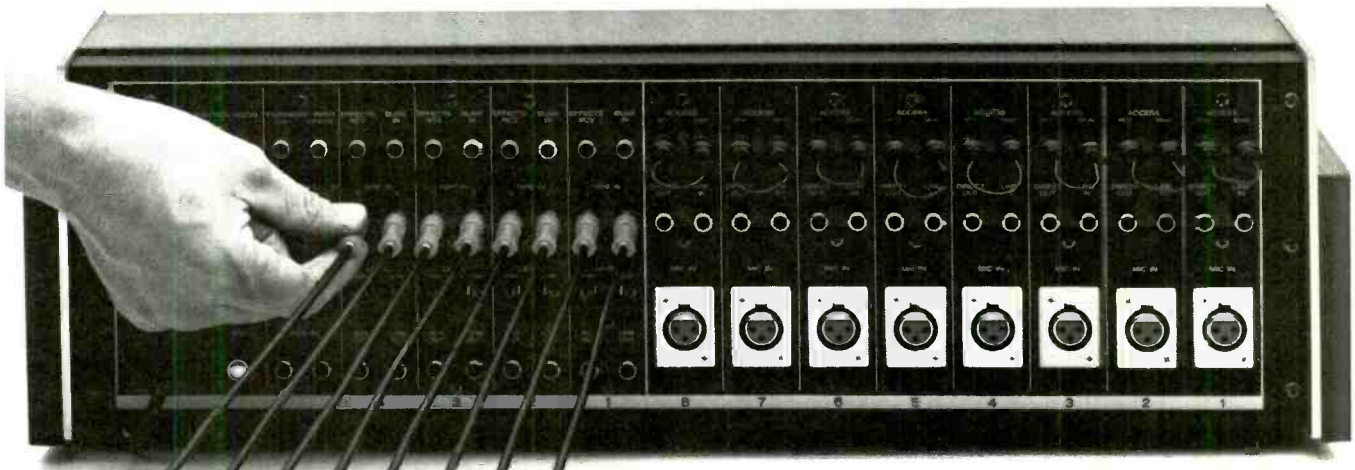

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Eight-track recording is serious business. Just like 16 or 24. So if you're turning music into money, get a mixer that's built for the job. Like our new Model 35.

You'll get a quiet, reliable 8 x 4 x 8 board (expandable to 20 inputs) designed for the hardworking 8-track studio. No fluff. No frills. No empty promises. There's full 8-track monitoring, an 8-track cue system and 4 effects returns.

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The Model 35 is dead-on perfect for our Model 80-8 recorder. But there is more to the Tascam System than just the 80-8. If you need more of anything, we've probably got it.

With the Tascam System you can solve your problems as your needs expand. Even if your production requirments change. The Model 35 can offer the flexibility you need now and in the future.

So if you're serving up 8-track tapes on a regular basis, make sure all your guests are satisfied. Musicians. Producer. Engineer. And accountant. Check out the new Model 35. The complete 8-track console at your Tascam dealer today.

TASCAM STUDIO SERIES

TEAC Production Products

CIRCLE 173 ON READER SERVICE CARD

cluding proper cable shield connections)? I'm working with an unbalanced, high-impedance system using coax cable, so any specific references to this sort of setup would be greatly appreciated.

—No Name Given
Fargo, N.D.

Unfortunately, grounding must remain a mystery to you just a bit longer. There is a light at the end of the tunnel, however. Rick Chinn explained to us that one of the problems with grounding is that for every rule you set down, someone comes up with two that defy it. Subsequently, to capsulize "proper" grounding techniques is frustrating at best. Due to the limited amount of space we can provide through Talkback, Rick felt that a referral rather than an incomplete response would be to your advantage. He recommends that you get a copy of Grounding and Shielding Techniques in Instrumentation *by Ralph Morrison, published by Wiley Interscience, a division of John Wiley and Sons, Inc. The book sells for $21.75 and can be ordered from Wiley Interscience at 1 Wiley Drive, Somerset, N.J. 08873. This volume is in the reference library at Tapco, and Rick feels that it will adequately cover the ground (sorry about that) you've questioned. Also, Rick tells us that you should feel free to contact him at Tapco (206-883-3510) if you need additional help.*

Piezo Pointers Postscript

[We received the following letter and diagram in response to a Talkback printed in the March 1981 issue (see "Piezo Pointers," page 18). Since we were not able at that time to oblige Mr. Taylor with a design for crossing over the Motorola piezo electric tweeters about which he inquired, we are especially grateful to Mr. Egnater for offering us this opportunity to correct that oversight. —Ed.]

Having read Richard Taylor's Talkback question in the March 1981 issue on the subject of passively crossing over the piezo electric tweeters, I thought I might have some information worth bringing to his attention. I have included a diagram of the crossover recommended by Motorola that begins to roll off at approximately 6 KHz. Hope this will be of some help to him.

—Bruce Egnater
President
Natec, Ltd.
Berkley, Mich.

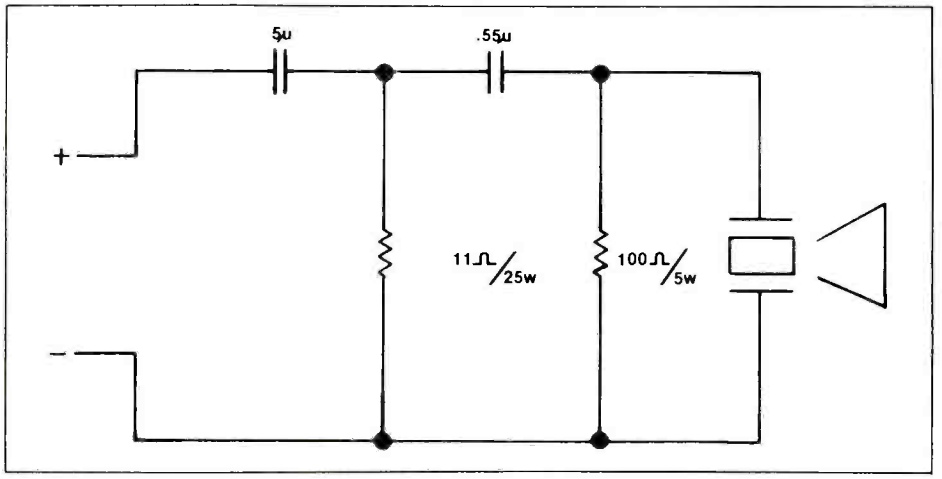
Bass-ically Direct Advice

Could someone please offer some advice and/or examples of techniques for recording an electric bass guitar direct?

—Jeff Liford
Toledo, Ohio

After receiving this Talkback question from *Modern Recording & Music,* I decided to also consult Mr. Jeremy Smith (a professional recording engineer in Los Angeles), and Mr. John Carey (from Westlake Audio Studios). Since I'm a musician and not a technician, I felt it would be best to cover this from both sides of the fence.

First of all, one of the most important steps in recording any instrument is to make sure that the instrument is in a decent state of condition. There is something rather embarrassing about being in the studio with an instrument that sounds as if it's about to blow up. Spend some time the day before you record and put on some new strings,



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adjust your action, intonate your bridge. I guarantee that you'll play a lot better because the instrument will feel good, if not, then it's time for a new bass.

One of the most common ways of recording your bass direct is to use a direct box. These usually come in two variations: 1) transformer type and 2) transformerless type.

The transformer (or passive) type of direct box sometimes tends to introduce a bit of distortion into the audio signal and also they will roll off some of the lower frequencies and sharper transients (usually introduced when the strings are snapped).

The transformerless (or active) type seems to produce a cleaner sound and is the type I normally use. Both units have an hi-impedance input and two One of the outputs is to go to your amplifier, and the other, a low impedance balanced output (an XLR connection), is for the recording console. Try to use short, shielded cables in making the bass to direct box to amp connections. Keep them from lying across any other cables or power lines.

You could also plug directly into the tape machine but you won't have as much control over the signal should any phase or ground problems arise.

Finally, if there is time, try to get the recording engineer to record some of what you are going to play and go into the control room and listen back—how does it sound? If it sounds different than what you're hearing in the studio, communicate with the engineer so you can both work toward getting the best sound possible.

Good luck and I hope that you have patience and continue to experiment, that way you will discover some things that aren't in the books yet.

—Alphonso Johnson
Los Angeles, Ca.

Do-It-Yourselfer On His Own

I'd like to build my own guitar amplifier. Is there anyone that you know of that sells plans, kits, etc., for this sort of project?

—Dave Dykes
Austin, Texas

Well, yes and no. We did, you see but they don't anymore. Understand? Briefly, our fine collective minds couldn't come up with a company so we called on Craig Anderton to pick his brains. He came up with Southwest Technical Products in San Antonio, Texas. As he recalled, they made some sort of hi-fi amplifier that he felt you might in some unique way adapt to fit your needs. When we called STP we were told that they had indeed made a guitar preamp but that they had discontinued it. Since it's a local call for you (512-344-0241), we suggest you call them again and see if there's anything else that might suit your purposes. We, on the other hand, will once again open the floor to our readers to see if they might be privy to any info that we have missed that could be of help to you. Come on, guys, do it for the gipper!

Count off...1, 2, 3, 4!

I have recently purchased the complete facilities of an existing studio. My newly acquired equipment includes one Tascam Model 5 board, one Teac Model 2 board, one Harmon-Kardon Citation 12 power amplifier, one Teac/Tascam Series 70 8-track machine, one Teac A3340 machine for mixdowns, one Tapco 4400 reverb unit, a patchbay and two dbx Model 161 compressor/limiters.

If you mixdown to the 4-track machine using tracks 1 and 3, will the tape, when played back over a 2-track machine, give a true representation of the material on it? Would it be better to mixdown using tracks 1 and 2 for the left channel and 3 and 4 for the right channel?

—David Parrish
President
Parrish Productions
Greenville, Tenn.

It's been my experience when playing ¼-track tapes on my ½-track, that there is a big disparity in level between the two channels, and some fringing effects to boot. I would suggest one of two methods for obtaining good results, but you will have to try experimenting to get what you want.

First, the reasons for the two different methods: If you look at the diagrams you will notice that the guard bands of the two formats are different in size. Recording on tracks 1 and 3 will place the two tracks over iron that will produce a stronger signal from track 1, and some crosstalk from track 3. Recording on tracks 1 and 4 will yield optimum crosstalk results, with some losses from fringing and extra noise from empty track area scraping iron. Recording on tracks 1-2 and

Tapco's expandable, versatile C-12/Series Two Mixing System.

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The C-12 is the first moderately-priced twelve-channel stereo mixing console to offer you the flexibility of subgroup mixing. You can route any input signal directly to any of the four subgroups or the main outputs. For example, by putting the drums on one subgroup, lead vocals, background vocals and the rhythm section on each of the others, you can change any subgroup level independently of the master balances you've set on the main channel inputs. You can also use the subgroup outputs to feed a four-channel tape deck, or as separate monitor sends, and still use the main outputs to feed your sound reinforcement system. It also has versatile features such as sweepable midrange EQ, a convenient patch bay, and transformer-balanced inputs. Plus, the C-12 has 48 VDC regulated power available, giving you the capability of using any phantom powered condenser mike currently on the market.

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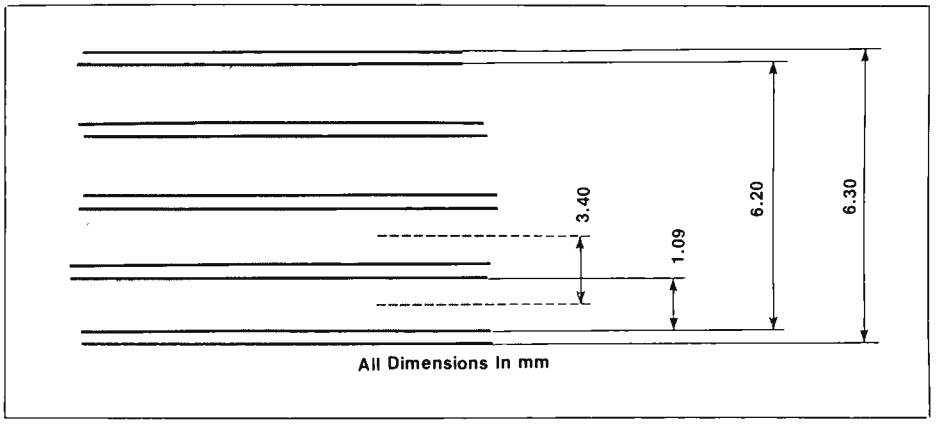


Figure 1: 2-track.

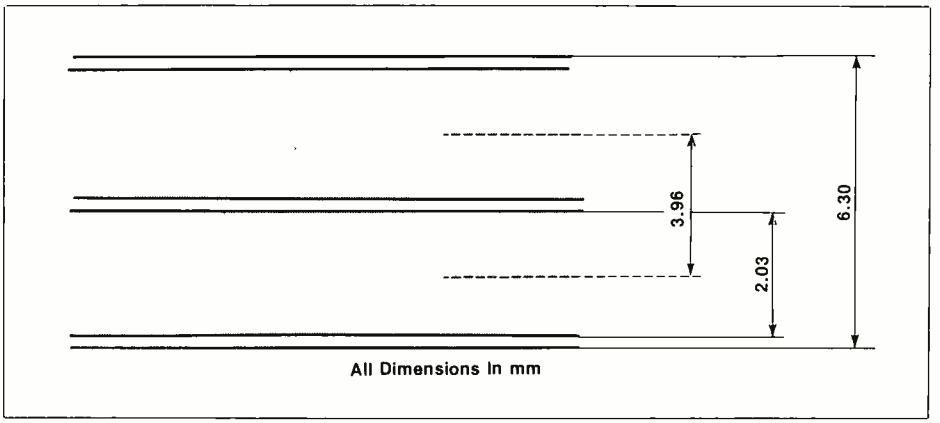


Figure 2: 4-track.

3-4 will yield the best signal-to-noise performance, but will cost you something in crosstalk and fringing.

There is no free lunch, as they say, and my suggestion to you is that you consider the purchase of a ½-track recorder for ½-track mixdowns. High quality ½-track, high-speed (15 ips) decks are once again available at reasonable cost, and in my experience the investment usually proves worthwhile.

—Drew Daniels
Applications Engineer
TEAC/Tascam Professional
Products Group
Montebello, Ca.

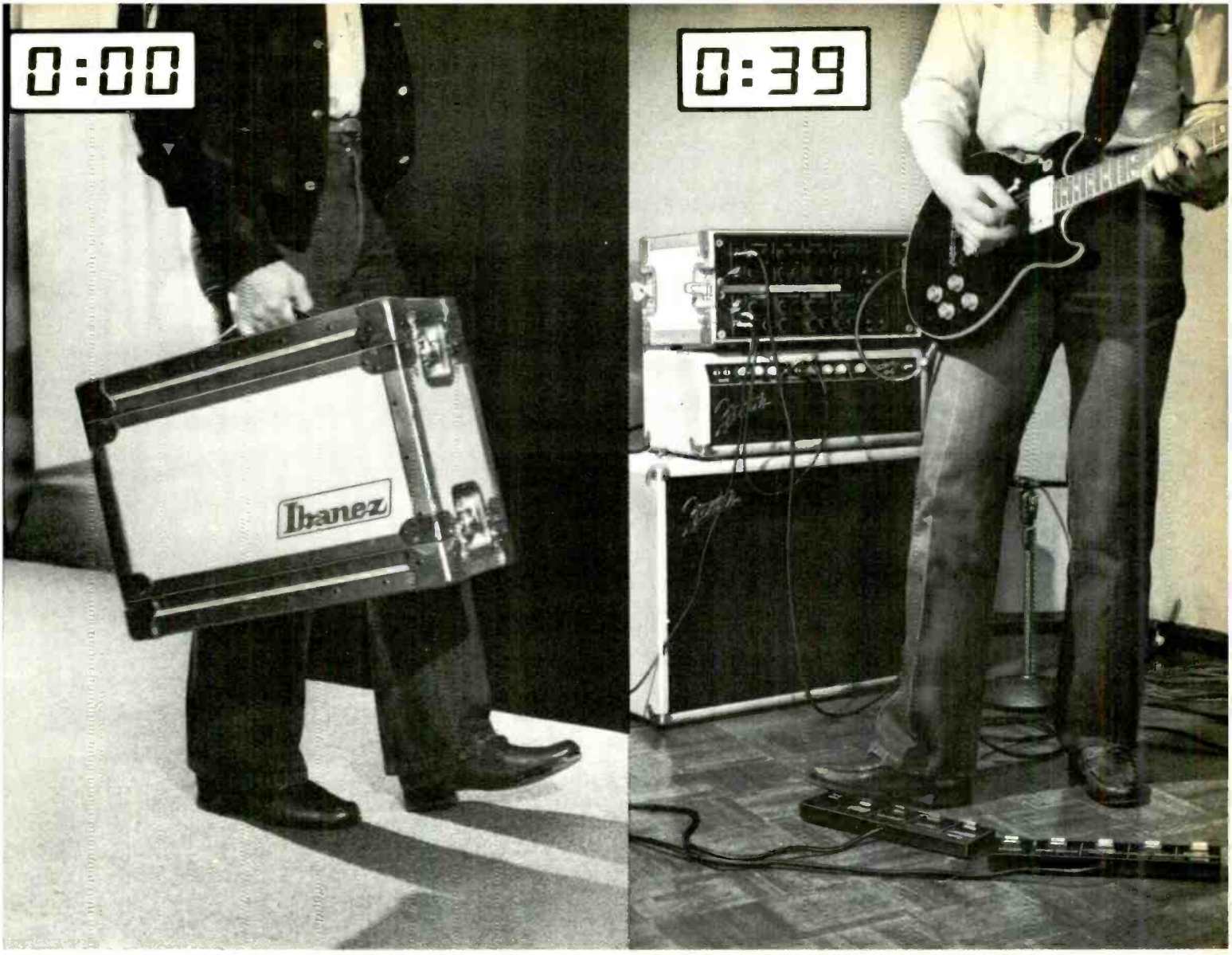
Doing It By The Numbers

First off, let me thank you for a superb publication. I've enjoyed and learned from it for several years.

I'm looking for information on designing and building front-loaded bass cabinets with horn-loaded port or front firing, rear-loaded horns (direct radiator with rear-loaded horn), along the lines of Mitchell's 3000B, Yamaha's 215L and 115L, Sunn's 215B and 215H, Road's 115SB and 212SB, and Music Man's 115RH and 212RH. Thanks for any help you can provide.

—Rick Bogas
Scottsdale, Ariz.

Rick, are you sure you don't want to just buy *one? Seriously, though, we recognize the needs of do-it-yourselfers (some of our best friends are) and so got in touch with Doug Buttleman of Yamaha in Buena Park, California. He suggested that you get your hands on a copy of* How To Build Speaker Enclosures *by Alexis Badmaieff and Don Davis (published by Howard W. Sams and Co., 4300 W. 62nd St., Indianapolis, Indiana 46268). You might also try writing to the folks at RD Professional Sound Reinforcement, they* might *now be offering the type of plans you require. (RD can be reached at P.O. Box 5090, Pittsburgh, PA 15206). You might also touch base with Speakerlab (735 N. Northlake Way, Seattle, Washington 98103) who offers kits, drivers, woofer systems, and all sorts of goodies. Finally look up* Speaker Builder Magazine *(P.O. Box 494, Peterborough, New Hampshire 03458) published by the same people who bring you* The Audio Amateur.



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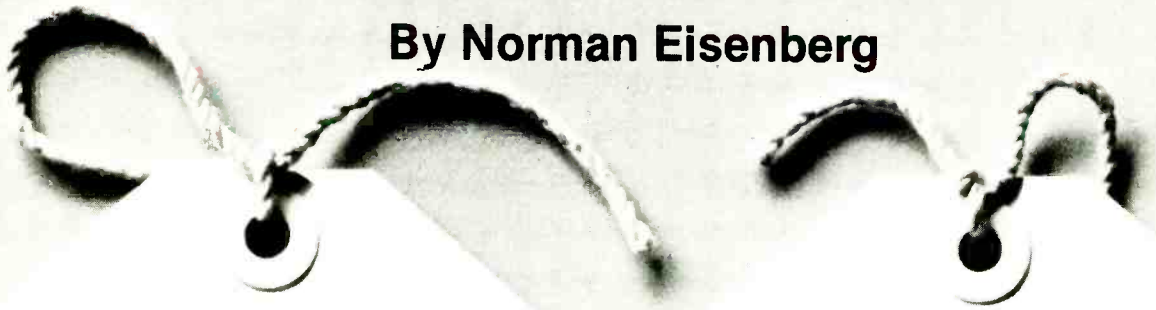
THE PRODUCT SCENE

By Norman Eisenberg



SPLIT RECORDING/ REMIXING CONSOLE

Said to be the first console specifically designed for true split operation, the JH-652 announced by MCI or Fort Lauderdale, Florida incorporates 52 inputs. These are installed 26 on either side of the master control section. The "revolutionary" feature here is that the mix outputs, effects/foldback outputs and automation of each side can be operated independently as if they were on two separate consoles, or joined to operate as one larger console. This makes the console, says MCI, ideal for use with dual synchronized multi-track recorders, or in applications requiring separate monitoring facilities. Standard features on the JH-652 include MCI-patented plasma display PPM bar-graph metering, full level and mute automation, six wild usable for pan or effect automation, 3-band equalization on each input and transformerless circuitry throughout the recording chain.

CIRCLE 18 ON READER SERVICE CARD

MIXER/PREAMP WITH EQ

Numark's model DM1800 is a mixer/preamplifier with built-in six-band equalizer. The device is said to be ideal for mixing sound on sound, making video and movie soundtracks, for P.A., disco and deejay uses. Inputs include three stereo phono/line plus a balanced/unbalanced mic with low-cut filter that accepts either a phono plug or XLR connector. The mic input features pan-pot control and a two-position talk switch to attenuate phono/line volume by either 6 or 15 dB for smooth voice-overs. Each stereo channel has a 12-segment LED output meter. Controls for volume, equalization, fading and cueing are sliders. The fading and cueing controls automatically adjust the gain levels of the channels being mixed so that volume does not vary during the mixing process.

CIRCLE 19 ON READER SERVICE CARD

DIGITAL REVERB

From Ursa Major comes news of its model 8X32 digital reverberation system, capable of producing a wide range of natural and artificial reverb effects, said to be suitable for studio, broadcast, "live" performance and other applications "where clean, high-quality sound is important." The microprocessor-based front panel features separate LED readout and control for each adjustable reverb parameter which, says the company, make the device "friendly to operate despite its sophistication." The 8X32 also contains a bank of thirty-two nonvolatile (they retain their contents even when power is turned off) storage registers that allows the user to store and recall thirty-two complete reverb setups, and to edit them at will. Four basic programs are available, ranging from a small, fast-diffusing "plate" to a large, echoing "space" simulation. Within each program, sixteen decay times can be selected, and the level (eight steps) and delay time (approximately 6 to 9 milliseconds in sixteen steps) of both the early reflection pattern and the initial reverberation may be independently controlled. LF and HF decay also can be individually trimmed (four values each). Two unique controls—input mute, and reverb clear—are designed to provide "especially useful effects for performing artists." Controls and displays are available at time of purchase on the front panel, in a remote unit suitable for use on consoles or both. Price of the device is $5,995.

CIRCLE 20 ON READER SERVICE CARD

TASCAM CONSOLES

Two new floor-standing consoles have been made available for Tascam tape decks. One is the CS-600 for use with the model 35-2B two-track mastering recorder which has separate housings for the transport and electronics. The other is the CS-800 for use with either the model 80-8 eight-track multichannel recorder, or the model 40-4 four-track multichannel machine. Provisions are made to accomodate the optional dbx processors, model DX-8 or DX-4. Both consoles are made of metal and feature heavy-duty casters and padded arm rests. Price of each is $449.

CIRCLE 21 ON READER SERVICE CARD

AUDIO POWER METER

A meter that shows actual loudspeaker power dissipation has been announced by Hutco, Inc. of Huntsville, Alabama. Known as the PWP-80 Peak power meter, it has a dynamic measuring range of 30 dB, and a standard indication range of 0.1 watt to 100 watts. Frequency response is stated as being flat within ±0.5 dB from 20 Hz to 20 kHz. Integration time is 100 microseconds, and the meter can tolerate a full power peak overload of 1000 watts. Recovery time from maximum overload is 1.2 seconds. Designed to be connected in parallel with the loudspeaker line (directly across the voice coil), the meter requires a 15-volt power supply with a maximum drain of 25 mA. The device is manufactured by Nippon Keiki works of Japan. Hutco, which distributes it, says that the meter will cost about $85 in quantities of 100.

CIRCLE 22 ON READER SERVICE CARD

BUFFER AMPLIFIER

The model 444 from RTS Systems of Burbank, California is described as a versatile two-channel amplifier designed to interface home and semi-pro equipment with professional studio systems. The model 444 accepts a stereo pair of balanced lines, isolates them through active circuitry and feeds them to a pair of RCA phono jacks at the IHF nominal level of 0.5 volts RMS, unbalanced. Simultaneously, the 444 accepts a pair of unbalanced lines, amplifies and transformer-isolates them to feed a pair of balanced lines at a nominal +4 dBm. Self-powered and self-contained, the model 444 contains level controls in the IHF-to-balanced section that permit the user to trim gain and channel balance as needed. Among the types of equipment said to be handled for interfacing via the model 444 are cassette recorders, both video and audio; reel-to-reel decks; graphic equalizers; noise-reduction devices; electronic crossovers; high-quality FM tuners; large-screen video projectors. Price is $312.

CIRCLE 23 ON READER SERVICE CARD

16 IN, 8 OUT CONSOLE

From Speck Electronics of North Hollywood, California comes news of the Specmix 16 recording console with 16-input and 8-output capability. The unit features 16 complete input channels, eight mixing buss outputs, eight large VU meters, eight-track panable assign, three-band equalizers, control room and studio playback, talkback and cue prompts, independent stereo mixdown buss. Mic inputs are transformerless low-noise types, and all operating levels are +4 dBm. Frequency response is given as ±1 dB, 23 Hz to 20 kHz. Suggested list price is $2,975.

CIRCLE 24 ON READER SERVICE CARD

NAKAMICHI'S DOLBY B AND C

A new unit from Nakamichi—the model NR-200—offers a choice of Dolby-B or Dolby-C processing. The NR-200 has its own power supply; peak-responding LED indicators; left, right and master recording-level controls; output control; calibration oscillator; and MPX filter. It can be interfaced with any quality tape recorder via the line jacks. It is fed from any line-level source. With Dolby-C encoding, the NR-200 is said to provide a 20-dB reduction in noise in the frequency range from 2 to 8 kHz (as compared to the approximately 10 dB provided by Dolby-B). In addition, the noise-reduction extends lower in frequency, with a 5-dB improvement at 150 Hz. When used in the Dolby-C mode, the NR-200 also features spectral-skewing and special networks designed to reduce high-frequency intermodulation IM and to increase maximum operating level beyond that available without encoding. Price is $450.

CIRCLE 25 ON READER SERVICE CARD

NEW OPTIONS FOR SOUND TECHNOLOGY AUDIO TEST SYSTEM

Sound Technology of Campbell, Ca., has announced three new options available for their 1500A Audio Test System. The Model VP-150 Video Printer will duplicate the CRT display in hard copy form. A thermal printer, it connects directly to a video output on the rear of the 1500A and costs $1,525.

For performing spectral noise analysis and flutter measurements, we have the Option 007 1/3-Octave Spectrum Analyzer. This circuit card module plugs into an internal pre-designed slot and is offered for $1,500.

The Model TR-150 Test Record contains high-quality tests for phono cartridge analysis. There are 17 test bands in all, including trackability, anti-skate, frequency response, phasing, channel separation and tracking. No price was available as of this writing.

The basic 1500A mainframe test instrument markets for $5,795.

CIRCLE 26 ON READER SERVICE CARD

NEW LINE OF CASSETTES

A line of blank audio cassettes, known as Loran, has been introduced by Loranger Manufacturing Corp. of Warren, Pa., for the past thirteen years a producer of audio tape software components on an O.E.M. basis. The company, specialists in high precision custom plastics molding, feels that there has been "a large gap between the best audio tape product available and the best product possible"—and its new line is aimed at filling that gap. Explains the company, the new product features a shock-proof, heat-resistant Lexan shell; marked reduction of wow and flutter; minimal phase shift; reduced zenith and azimuth tracking errors; quieter operation; and superior sound reproduction. Loran cassettes initially will include ferric oxide and chrome high-bias tapes. Prices are expected to run about $1 to $2 higher than current top-of-the-line conventional audio cassettes.

CIRCLE 27 ON READER SERVICE CARD

10 BAND EQUALIZER

The VQ-100 equalizer from Vector Research employs all discrete transistors which are said to provide increased performance, quicker response and very low transient IM distortion. The device's push-pull circuitry is credited with improving S/N and "dramatically lowering total harmonic distortion." Each of the ten bands provides a ±10 dB boost/cut range, and the device contains ten LED meters, five for each channel, a switch to provide equalized recording capability when the equalizer is used in a recording chain, unity gain controls for each channel to prevent overloading, a subsonic filter, tape-monitor function and an AC outlet. Specs include S/N ratio of 110 dB (weighted); THD of 0.005 percent from 20 Hz to 20 kHz; overall frequency response of ±1 dB, 5 Hz to 100 kHz.

CIRCLE 28 ON READER SERVICE CARD

MIXING CONSOLES

Neptune Electronics Inc. (NEI) of Portland, Oregon has introduced a new line of mixing consoles, the XM series, in four submaster/stereo/mono output formats and choice of 8, 12, 16 and 24 input channels. The models include the 84XM; the 124XM; the 164XM; and the 244XM. All have numerous features, including transformer balanced mic input with switchable line level input; pre EQ/fader monitor buss; post EQ/fader effects buss; switchable pre or post aux buss; input preamp EQ with switchable bass shelving and sweepable midrange; submaster assignment switches; headphone amp with level control; channel mute; and more.

CIRCLE 29 ON READER SERVICE CARD

AUTOMATIC MIC MIXER

Edcor of Irvine, California has announced its Model AM 400, described as an automatic microphone mixer that combines digital control with analog circuitry. Up to seven of these units can be "daisy chained" to provide control for twenty-eight inputs. In this arrangement, the first unit becomes master and the other six become slaves. Each input has controls to vary the depth of attenuation, allows the user to adjust each channel by degree instead of complete turn-off or turn-on. This technique establishes a floor gain setting for each channel. Logic controls permit the operator to mute certain speakers via relays when channels are opened or closed. The attack time per channel is said to be "too quick to be noticed in the form of clipped consonants" while the timed release "negates both clipping of trailing audio sequences and distortion." If the user misadjusts the device, it may be switched to manual mix mode. Price is $500.

CIRCLE 30 ON READER SERVICE CARD

DIGITAL HOKUM

In a recent "Ambient Sound" column (*MR&M*, April 1981) my colleague and friend Len Feldman justifiably took to task a criticism that had appeared in the N.Y. Times which had unjustifiably attributed to digital sound some kind of sonic aberration regarding stringed instruments.

The Times article scored by Len is only one example of the nonsense being offered from many sources. The "super esthete" posture, incidentally, is reminiscent of the static we got when stereo itself first appeared, and before then, when the very idea of hi-fi sound became known.

An equally untenable thesis regarding digital sound appears in a recent issue of *dB* magazine (whose editor—to his credit—has issued a disclaimer) in which the author, a doctor no less, suggests that digitally mastered recordings have had a negative effect on patients who formerly responded favorably to conventional recordings.

While some are nitpicking at digital sound, others— who present themselves as all for digital—may be doing it and the entire audio industry a disservice with their "good intentions." For instance, a recent article in *Business Week* predicts an early demise of many existing audio companies, and especially those involved in noise-reduction. What is overlooked in this notion is the fact that fully digital recordings, and the equipment and systems for making them from start to finish—not to mention playback systems—are, by the most optimistic estimates, five to ten years away. And even then, we still will be dealing with analog devices into and out of the recording-playback chain, i.e., microphones and loudspeakers. Hang on to your dbx's, High-Coms and Dolby's.

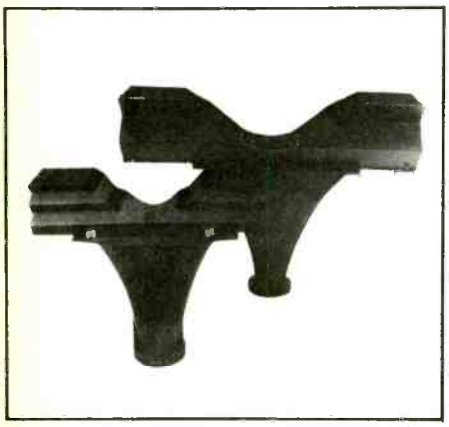
One more example: In the January/February issue of the *Journal of the Audio Engineering Society* appears an article which recommends for digital sound a deliberate limiting of the audio bandwidth for consumer use to 15 kHz. The authors base this on listening tests which, as they are described in the article, seem highly questionable. Be that as it may, the 15-kHz top in audio—professional and consumer—is so easily discredited as to be absurd as a recommendation to be taken seriously today, for digital or for analog sound.

Any comments that you might have to offer on this matter are sincerely invited.

MUSICAL NEWSICALS

SOUND REINFORCEMENT EQUIPMENT

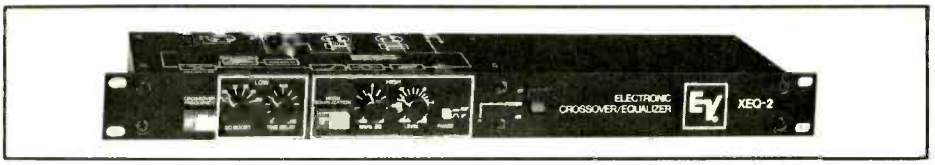
Integrated Sound Systems, Inc., which is the manufacturer of the Vortec line of sound reinforcement products, has announced the introduction of two models of acoustic lens/horn combinations, manufactured from Technoplast™, a non-petroleum based plastic material that is said to be lighter, stronger, and less resonant than fiberglass, metal or foam plastic horn materials. The Vortec acoustic lens horns boast smooth frequency response and uniform pattern control which makes them suitable as studio monitors, stage monitors and short-throw P.A. monitors. Both models use a Technoplast straight exponential horn and sheet aluminum lens slats which are stamped and shaped in one step on numerically-controlled machines. The



AL-130-5 provides a uniform 130° horizontal coverage down to 500 Hz with its 30-inch wide folded lens plates. The other model, the AL-100-8, is considerably smaller at 20 inches, but has a 100° coverage and an 800 Hz cutoff frequency. Both models are designed for use with 2-inch compression drivers.

CIRCLE 4 ON READER SERVICE CARD

Electro-Voice recently introduced two new products which will be of interest to the sound reinforcement market. The first of these is the XEQ-2 active crossover/equalizer, which was designed to complement E-V's line of



low-frequency enclosures and HR and RC models of constant directivity horns. The XEQ-2 is a two-way crossover with crossover and equalization characteristics determined by plug-in modules. Three crossover modules are provided with the XEQ-2, one each for 500 Hz and 800 Hz crossover points and a blank module for any user-determined crossover frequency between 100 Hz and 8 kHz. E-V manufactures additional modules as optional accessories for 125 Hz, 1500 Hz, 3500 Hz and 7 kHz crossovers. Plug-in equalization modules are available for specific E-V Constant Directivity horns; these EQ modules must be ordered separately, a "flat" module being supplied with the XEQ-2. Other features of the XEQ-2 include a high-frequency level control to vary the level above 5 kHz and a variable time delay circuit which allows the low-frequency output to be delayed from 0 to 2 milliseconds to help smooth irregularities in frequency response around the crossover point due to driver displacement.

CIRCLE 5 ON READER SERVICE CARD

The other new product from Electro-Voice is the PA-700 high performance power amplifier. The PA-700 is definitely in the high power category, delivering 210 watts per channel into 8 ohms or 355 watts per channel into 4 ohms. It also may be operated in a bridged mono side for a single output capable of 710 watts into 8 ohms. Other features of the amp include a two-speed fan, high thermal capacity for safe operation and minimum fan noise, welded steel chassis construction and an LED Loss of Feedback clipping indicator.

CIRCLE 6 ON READER SERVICE CARD

Mirage Audio Systems is best known for its Short Stack miniature monitor speakers. The Short Stack MA150S is a compact (16"x11"x8") floor monitor with a cabinet designed to sit at either a 45° or 60° angle. The secret of its compact size is its driver complement which comprises six 4½-inch high efficient, full-range speakers which can handle 150 watts RMS and produce literally ear-splitting levels over a 70 Hz to 15 kHz range. Even more unique is the MiniMonitoring System, which contains two 4½-inch drivers in a 10½"x5½"x4½" box which can be directly mounted on a mic stand. The latest product from Mirage is the C300 road case speaker, a self-contained, full-range speaker system in its own case. The C300 uses a 15-inch heavy-duty bass driver and an MA125 treble driver with an 800 Hz passive crossover. Frequency response of the system is 50 Hz to 15 kHz ±3 dB, and sensitivity is quite high at 93 dB output @ 1 watt @ 1 meter. Power handling capability is rated at 300 watts RMS. The system is a compact

18"x18"x24½" even with the cover installed, and the unit weighs in at only 63 pounds.

CIRCLE 7 ON READER SERVICE CARD

MUSICAL INSTRUMENTS

St. Louis Music Supply has added a teardrop-shaped mandolin to the Alvarez Artist line. The new model, designated the A400, features a solid spruce top and mahogany back and sides to complement the other four Alvarez mandolin models all of which have maple backs and sides with spruce tops. The solid spruce top of the A400 has particularly tight grain, and is braced in a specially developed pattern with hand-shaped braces for extra resonance. The oiled walnut fingerboard of the new Alvarez mandolin has hand-set nickel silver frets, and the 28:1 geared machine heads with pearloid buttons are set in a head stock decorated with inlays of genuine mother-of-pearl.

St. Louis Music Supply has added a new dreadnaught model to its Alvarez line of acoustic guitars. The Alvarez Ash Festival, also known as the model 5029, has back, sides and top made from ash, which makes for a crisp, articulate tone as well as a distinctive appearance. The Ash Festival uses solid brass bridge pins, head nut and bridge saddle for the improved sustain and presence it provides. The guitar is finished in oiled satin and has ivory celluloid bindings around the edges of the body and headstock and along the sides of the fingerboard.

CIRCLE 8 ON READER SERVICE CARD

Yamaha recently introduced its new, revised lineup of electric guitars and basses. Top of the new line is the SBG500 which features a small, symmetrically-shaped body made of agathis wood laminated to an arched top of maple. The 22-fret maple neck is reinforced with a truss rod incorporating a special aluminum channel, and the bound rosewood fingerboard has frets that extend over the binding for increased playing surface for note bending. The SBG500 features separate tailpiece and fully adjustable bridge and twin humbucking pickups with individual volume and tone controls. The SSC500 and SHB400 are both contour shaped instruments with solid alder bodies and reinforced maple necks. Both models also feature fully adjustable bridge/tailpiece units, but differ in terms of their pickups, the SSC500 having three single-coil pickups while the SHB400 has two humbucker pickups. The single-coil pickups on the SSC500 are unusual in that they have more windings than is usual in single-coil pickups for a deeper, stronger output, and they also use a single bar magnet rather than individual pole pieces for more uniform output even when bending notes. The control electronics of the SSC500 features three mini-toggle switches to select the three pickups and single volume and tone controls plus a push-on/push-off switch incorporated in one of the controls to switch to a tapped output from the pickups. This arrangement provides the bright, crisp sound of conventional single-coil pickups rather than the richer sound of the extra-winding pickups. The SHB400 features a selector switch for its two pickups plus a push/push switch incorporated in its single tone control to switch between humbucking and single-coil connection of its pickups.

CIRCLE 9 ON READER SERVICE CARD

RolandCorp's Jupiter-8 is an eight-voice, 16-oscillator polyphonic keyboard synthesizer with comprehensive programmability and advanced control functions. Each of the synthesizer's eight voices has two very stable VCOs which may be controlled by LFO, envelope generators, performance control and/or cross modulation in addition to the basic keyboard note selection. Each voice also features both high pass and low pass VCFs which are selectable for 12 or 24 dB/octave operation, LFO with poly Sample and Hold and programmable delay and performance control options and two ADSR envelope generators, one of which may be inverted. A unique feature of the envelope generators is a Key Follow function which changes the length of a note's envelope in proportion to pitch as happens naturally in many acoustic instruments. The Jupiter-8 has a five-octave keyboard which may be operated in one of three modes: Whole mode operates the Jupiter-8 as an 8-voice instrument with a single programming patch; Split mode divides the keyboard into upper and lower sections each of which is a separately programmable 4-voice synthi; Dual mode layers two different patch programs onto each of four notes played. The JP-8 has a powerful programmer which allows up to sixty-four different patch programs to be stored and recalled at will. Any memorized patch which has been recalled may be edited simply by moving the appropriate control in real time; such an edited patch program may be written back into memory as a corrected patch or into another memory location as a new patch. For additional versatility, the JP-8 can dump its memory onto any audio tape recorder for long term storage and later re-use in whole or in part; a verify function allows the taped program data to be compared to the internal memory and verified as correct before any of the internal memory locations are reused.

CIRCLE 10 ON READER SERVICE CARD

Guild Musical Instruments, long one of the leaders in fine acoustic guitars in America, recently introduced a new dreadnaught guitar model, the D-46. The new instrument is the first dreadnaught Guild has offered to feature solid ash back, sides and neck and a solid spruce top. Ash wood is prized among instrument builders for its superior sound properties and its exotic grain patterns. As is usual for Guild guitars, the D-46 has an adjustable truss rod in the neck, ebony fingerboard, deluxe chrome-plated tuning machines and a white "ivoroid" binding around the top and back. The D-46 is available in two finishes, blond top and dark-stained back, sides and neck, or a sunburst top with dark sides, back and neck.

CIRCLE 11 ON READER SERVICE CARD

MULTI-TRACK MAGIC

Creative Multi-Track Recording

PART II

By Craig Anderton

In Part 1 of this series [May 1981], we described many ways to extend the versatility of a four-track recorder without using any extra recorders. However, since most studios have an additional 2-track deck for mixdown, this second machine can be used to good advantage in extending the versatility of our four track as well. How? Read on...

What Kind of Two Track?

If you're choosing a reel-to-reel recorder as your second machine, you will have to decide whether to go with a quarter-track or a half-track head format. The main advantage of a quarter-track stereo format is that the head configuration is compatible with four-track heads designed for ¼-inch tape. The advantage of a half-track machine is the potential for greater fidelity (which is very important when you're mixing down!) Either choice involves a certain amount of expense, but even if you can't afford a good ¼ track or ½ track, don't despair, something as simple as a cassette deck can save you considerable time and trouble. Following is a collection of tips designed to help you get the most out of any second machine.

(1) SAVING 4 TRACK WEAR and TEAR: Would you spend $30 in order to save $100 or more? I thought so. In that case, go out and get yourself a cheapo cassette deck and AC adapter. And I do mean cheapo; nothing fancy, just something general purpose with heads and a motor.

It seems to me that most of the wear and tear on a 4 track occurs when you're learning a new part, because in order to practice with the tape you have to repeatedly go over specific sections of that tape. However, there is a simple way to cut down on this kind of machine wear. Simply record a premix of the tracks on to your little $30 cassette deck, then play that tape over and over again as a practice tape. As soon as you feel that you're getting close to coming up with the perfect part, switch over to the 4 track and start overdubbing. Note how we've cleverly cut down on the number of passes the tape has to make, which not only saves the heads, but the quality of the tape as well. Tape starts to fall apart after it has scraped past the heads for the thousandth time.

(2) 4-TRACK TRANSFERS to ¼ TRACK: This technique takes advantage of the fact that a ¼-track stereo deck head configuration is physically compatible with the head configuration of 4-track decks that use ¼-inch tape. After recording parts on all four tracks of your 4 track (this can be up to dozens of actual parts if you followed the techniques given in last month's installment), you can premix these four tracks—possibly adding a "live" overdub in the process—over to one track of your ¼-track machine. Next, remove the tape from the ¼-track's transport and physically transfer it over to the four track's transport. Depending upon which ¼-track channel you mixed into, this premix will come up on either track 1 or track 3 of the 4-track recorder. Note that this process only adds one generation of signal degradation, since you didn't have to bounce back from the ¼ to 4 track but simply transferred the tape over from one machine to another. You may then continue adding tracks on the 4-track machine until all four tracks are filled up, at which point they may be mixed down on to the ¼ track again and re-transferred over to the 4 tracks, thus creating another cycle of overdubs.

How does this approach differ from the ones given last time? Generally, getting a second machine into the act (if it's high quality) will give better sonic results than bouncing back and forth within a machine. Also, the premix can be mixed onto the ¼ track in stereo; virtually all of the methods described in the May 1981 issue didn't lend themselves to stereo applications.

By the way, in any application involving machine to machine transfers, make sure that the alignment between the two decks is reasonably close.

(3) HAND SYNCING: Here is something for those of you who have magic fingers, good luck and patience—*lots* of patience. However, while this technique takes work, the results can be worth it. The basic idea is to end up with 4 tracks in one channel of your second machine, and 4 more tracks in the other channel of the same machine.

Record the first 4 tracks onto your 4-track machine, then premix them (you also have the option of adding a "live" overdub at the same time) into one channel of the ¼-track machine. After you're sure that the premix is right, erase all the tracks on the 4 track. Next, temporarily dub the resulting premix back to one channel of the 4 track to provide a rhythmic reference as we add other tracks. Fill up the remaining three tracks in the 4 track with new parts; if they provide enough of a rhythmic reference so that you don't need the original reference track, scratch that and add yet another track of new material. Taking inventory of our tracks so far: we have four premixed tracks on one channel of the second machine, and four tracks of new material in the four track. Now take a deep breath, because we're about to hit the tough part.

Set the ¼-track and 4-track machines at the exact beginning of the song; set up your mixer so that you can monitor the premixed track on the ¼ track as well as the three or four tracks that you've recorded on your 4-track track, and start both machines at the exact same time. Record the 4-track material into the other channel of the ¼-track machine. Chances are the first couple of times you try this, the machines will not be in exact sync and the new material will not mesh with the previously premixed track. However, you've got patience and magic fingers, remember? Rewind both machines and try, try again until you get it right. If one of the decks has variable speed, that will allow you to adjust for any minute timing errors. Also, if you have the beginning of a song synced correctly but the sync falls apart at a later point, you can always do punch-ins along later points of the tape until you've worked your way to the end of the piece. You'll end up with the first four tracks you recorded on your four-track machine in channel 1 of the ¼ track, and the second set of four tracks you recorded on your four-track machine in channel 2 of the ¼ track. Amazing. You've got eight total tracks on the ¼-track tape, and they're all only one generation removed from the master. You can then transfer this tape over to the four track and fill up the remaining two tracks. Or, you can mix these down to one track of the four track and go overboard with more overdubs...you get the idea.

(4) TAPE PHASING (FLANGING): I don't care how many ads say that their flangers duplicate the sound of tape flanging—they at best only mimic the sound of true tape flanging. Tape flanging has better bandwidth, lower noise, lower distortion, flanges through zero and has a sweeter sound. Sure, some electronic flangers manage to come close, but anything that comes *that* close also comes equally close to wiping out your budget. However, you can actually use a second machine to generate that true tape phasing sound as long as one of the machines has variable speed (we'll assume that it's the four track that is being used in the following example).

After you've recorded all four tracks on your 4 track, mix them down to one channel of the ¼-track or half-track machine. Now, rewind both machines back to the beginning and send an additional mix of all four tracks into the other channel of the ¼ track; but this time, vary the speed of the 4-track machine as you dub onto the second machine to create the time delays that produce flanging. Monitor the playback heads of both tracks on the *second* machine so that you can keep tabs on the sync, but make sure that you monitor them in mono; if you don't, instead of hearing flanging sounds you'll get a strange stereo placement effect (which, incidentally, has its uses too). Anyway, if the tracks are *very* closely synced to each other at the beginning of the song and you vary your speed control just right, playing back both of the second machine's tracks in mono will give you that beautiful, rolling, unmistakeable sound of tape flanging. You'll love it. As in any hand-syncing situation, it might take several tries to get both decks properly coordinated. Don't get frustrated—perseverence furthers, as they say. The first few times that you try hand-syncing will undoubtedly be fraught with difficulty; but as with any skill, practice is really the key to success.

An alternate way to flange is to record only three tracks on your 4-track machine, then send a mix of these to the second machine. Play back the mixed output from the second machine into the remaining open track on the four track unit, while varying the variable speed control of the four track. As you play back the original three tracks along with the new track that was dubbed from the second machine, you will hear that great tape flanging sound if the machines are closely synchronized.

(5) RHYTHMIC GAMES: A second machine can also provide unusual rhythmic effects, such as adding percussive effects to a track that would otherwise appear to have no rhythm at all. As an experiment, try recording a simple percussion track on one track of the second machine, then dub this onto one track of the four track recorder. Next, wind both machines back to the beginning, and dub the same percussion track onto a different channel of the 4 track; only this time, start the second machine just a fraction of a second late so that the two are out of sync by a significant amount (say, one beat). This gives an echo effect, but since you have control over when you start the second deck, this echo can be as long as you want. The results are very interesting, and most useful when you really want to thicken a percussive track. A short delay sounds like doubling, while longer delays start to simulate the sounds associated with tape loops. What we've basically done is recreate the sound of an echo unit, but with better fidelity (and lower cost).

(6) THE SECOND MACHINE as ECHO UNIT: One problem with using the echo techniques presented in the last installment is that they are locked into the speed of the multi-track machine. For example, most of the time you'd want to run your multi-track at 15 ips for best fidelity; however, at 15 ips it takes very little time for a signal to go past the record head to the playback head, resulting in a short echo. If we could slow the tape speed, it would take longer for the signal to go from the record head to the playback head, resulting in a longer echo. Therefore, in many instances 7.5 ips or even 3¾ ips echo is sonically preferable to the tight slapback echo sound of 15 ips. This is where having a second machine really comes in handy.

Let's suppose that you're premixing tracks 1, 2 and 3 of your four track down to track 4, and you also want to add echo to this premix. Sample the combined signal coming off the mixer output (or from track four's playback head for an even longer echo) and send

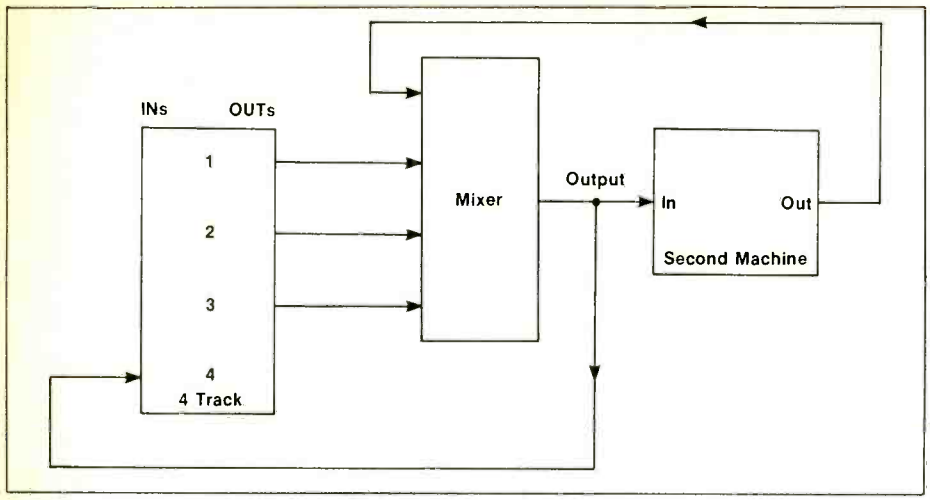


Figure 1

this signal to the second machine, which should be set for a slow tape speed. Then, return the signal from the playback head of the second machine back into the mixer along with the other tracks that are going into track 4's input (*see Figure 1*). *Voila*!—instant long echo. If you have variable speed on the second machine, you can have a good time adding Doppler shifts to the echo. This not only adds a whole different "flavor" to the echo effect, it makes it sound like you've got a real good harmonizing device in your studio as well.

Don't forget that the second machine can also be used as a simple delay line. For example, suppose you want to add some pre-delay to your reverb line. Simply take the reverb buss output from your mixer, feed it into the second machine, monitor the signal from the playback head, feed this delayed signal into the reverb unit input and finally mix the reverb unit output back into the console (see *Figure 2*). By the way, although I suppose some people will find the following idea abhorrent, if you're into stereo echo no law says that you can't mess around with the head alignment of the second machine. By throwing the heads slightly off axis, each channel will be delayed by slightly different amounts and give strange phase anomalies.

(7) THE SECOND MACHINE as SOUND EFFECTS GENERATOR: Have you ever done a piece of music where you wanted to have some complex, outrageous and brief sound effect work its way into a song? I know I have. For example, in one piece I wanted to simulate the sound of a large object flying in front of the listener. I had one track left on the four track in which to place this effect, and that just didn't give me enough space to come up with anything suitably complex. (I figured I needed at least four or five overdubs to create the desired sound.) If you run into a situation like this, here's a solution.

First, time how long you need your sound effect to last. Next, remove the master tape from your 4 track, and place a new reel of tape on the transport. Build up as complex a sound as you want on the 4 track using the layering techniques given in Part I, and when all is complete, mix this down to a second machine (which can be a good quality cassette deck if that's what you've got to work with). Next, remove the "sound effects" tape from your 4 track and replace it with the original master tape that you cut on the four-track machine; set the second machine to the exact beginning of the sound and patch the second machine's output into the remaining open track on your four track. At the precise point in the piece where the huge sound is supposed to come in , punch the play button on your second deck and dub this complex sound back into the master tape.

One application where this really came in handy was when I was doing the simulated audience section in "Leave 'Em Laughing" on the Craig Anderton Music Tape. I built up a convincing audience on the multi-track recorder using the layering techniques discussed last month, dubbed them on to a cassette deck and mixed the applause back onto the tape. This "complex sound" approach also works well for drones, choral parts that don't depend on a great many chord changes and sound effects (whether synthesized or from sound effects records).

Rough Mix?

This pretty much completes our installment on how to expand your four-track studio through the use of a second machine; and I think that we've come far enough in our discussion that it's time to consider a couple of other points.

First of all, note that many of the effects discussed in this and the last installment involve duplicating the sound of commercial effects units (flanging, echo, etc.). It's interesting to note that these effects were originally created so that on-stage performers could duplicate sounds previously only obtainable in the studio. But, over the past few decades the use of tape tech-

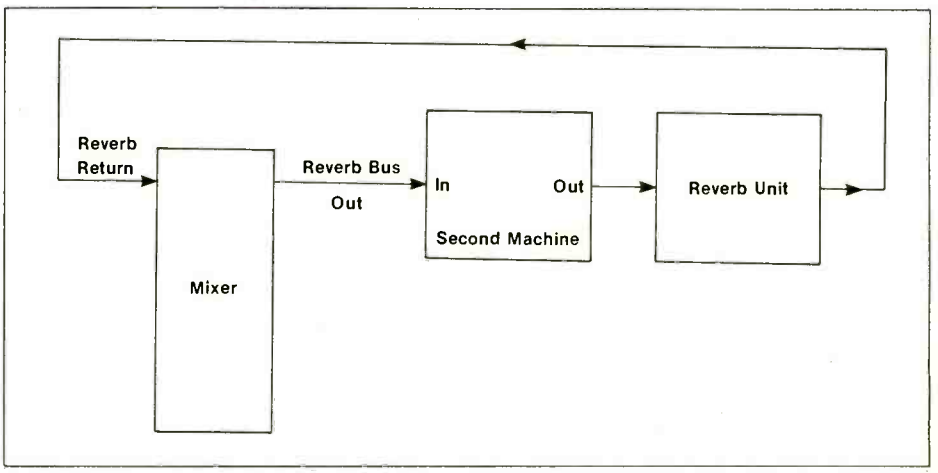


Figure 2

LIMIT TO THE LIMIT.

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THE HIT SOUND IN AUDIO SCIENCE

CIRCLE 101 ON READER SERVICE CARD

niques has become such a lost art that engineers—as well as performing musicians—started relying on these electronic substitutes. Granted it's easier to say, "Flip on the flanger" than it is to laboriously hand sync a piece of tape to another through careful fiddling with the variable speed control, but many times the sonic results are worth it. I don't intend to knock those wonderful little boxes—I use them all the time—but I do think that those just getting into the field of engineering might consider that innovation in the studio may have been at a higher level in the 50s and early 60s than it is today. Looking back, I feel that in the earlier days of multi-track recording engineers were more willing to take chances, make up the "rules" as they went and to compensate for the lack of technology available in those days by using their brains and intuition. Many people lament the lack of originality in today's music scene; this is probably something that is to be expected when studios use similar electronic technology get their special effects from the same rack-mount boxes that everyone else is using and then they tend to shape their acoustics towards the current "ideal" (yet, does that ideal include the concept of "character"?)

I suppose that's enough editorializing...after all, we're here for information, not opinions. But I think it's important to realize that recording is a very young art/science, and that those people who feel intimidated by their equipment are in trouble. There have been truly immortal records recorded on two-track machines with one or two microphones; maybe the S/N ratio wasn't as good as if it had been recorded today, but that's not the point: music is the point, the *raison d'etre* for the tape recorder in the first place. So, don't be afraid to use your machine in unusual ways—cut your tape up and put it back together again, hand sync from one machine to another or use any of the other techniques that we've been discussing. And if you can do all this stuff with a 4-track recorder, imagine what you can do when you graduate to 8 tracks...

In the next and final installment, we will discuss the demands that the techniques we have discussed make on the producer, and what those demands mean in terms of track strategies. Stay tuned!

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CIRCLE 85 ON READER SERVICE CARD

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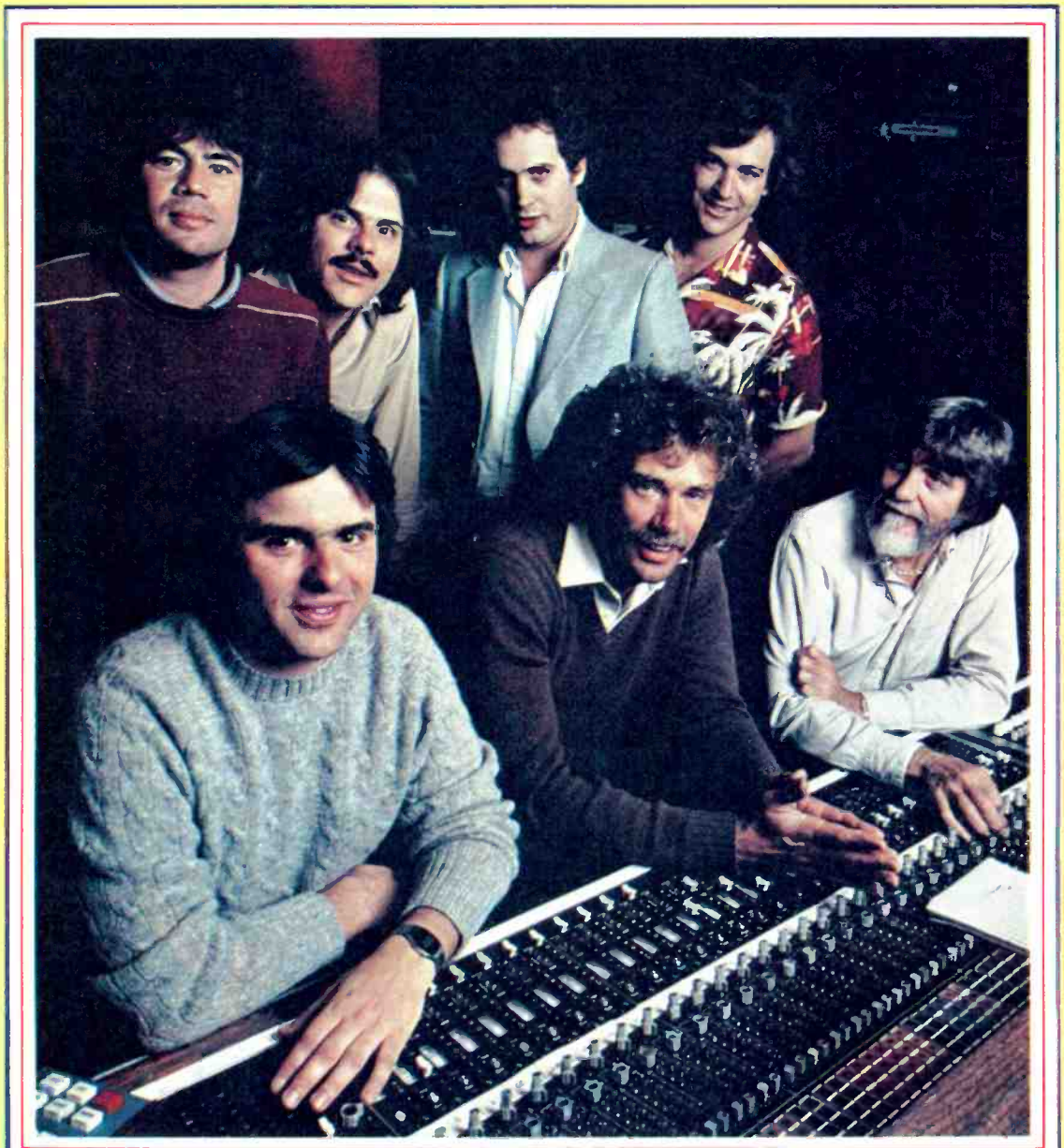


By STEVEN CARAWAY

ince gaining international attention in 1974, California-based Pablo Cruise has continued to produce artistically-crafted music and top-selling songs. After several personnel changes, the core of the band remains intact with Dave Jenkins, guitar and vocals; Cory Lerios, keyboards and vocals; and Steve Price, drums. With the addition of John Pearce on bass and Angelo Rossi on guitar, Pablo Cruise recently was "headquartered" at Sausalito's Record Plant recording studios, where with recording maestro Tom Dowd at the helm and Greg Price engineering, the group was producing what will become the band's sixth LP, entitled Reflector.

Modern Recording & Music: How's this project coming along?

Tom Dowd: We have all our tracks and some of our vocals already done—it was a welcome relief. The confidence they had...I saw opening night up in Tahoe and they did five of the new songs in the first set. Subsequently, they added two more. They had that much of the new album down, so that when they were up there in Tahoe for the ten shows, they did seven of the new numbers. We're here just honing it!

MR&M: How close are you to being completed?

TD: I'd say about ten days of recording. We've still got some backgrounds and a couple of solos to do.

MR&M: Greg, can you give us a brief description of the basic equipment you used here at the Record Plant?

Greg Price: We used an early, computerized API console and 3M M-79 tape machine. This is a pretty straight-ahead studio. We're using the John Meyer studio monitors which I think we were a little apprehensive about at the start, but I'm beginning to really like them. The band usually records on [Altec] 604s. Tom brought some of his hemispherical speakers, which I like. The studio is well equipped with all the microphones you could imagine, and there's plenty of outboard gear available.

MR&M: Tom, you've worked with such a diverse collection of people in your career. How do you approach each individual session or each individual group?

TD: I try to spend a little time with the group in order to agree that we should make an *endeavor.* Usually if someone addresses me and says, "Will you record?" or "Are you interested in producing this act or that act?" first thing I will do is go out and buy their last four or five albums and do some homework. Then I'll try to determine if I have an empathy for what they have done or have an idea of where I could encourage them to go—get something out of them that hasn't been had yet. Then I spend time with them, looking at the new material and giving them some input. If it offends them or if I sense that they don't see eye to eye with me, that's O.K. It's better to know the problems early rather than charging ahead into the studio and having a disaster. I'm chameleon-like; I have no allegiance with regard to artistic endeavor. I just try to help people get the best out of themselves that we can get, and encourage them to do it, that's all.

MR&M: How did you get involved in this project? How did you come to meet these people and work it out?

TD: This chap named Richard T. Bear, whom I've known over the years, apparently was sitting at A&M Records one day about seven or eight months ago, talking about Pablo Cruise. He heard they needed to get a producer and were looking for a new direction. It seems Richard mentioned my name, because he called me a few days later and said, "If you're in California give A&M a call; they're interested" and so forth. Then A&M called me, and I *was* in California, and so I contacted them. The band was up here [Bay area] rehearsing at Hun Sound. They were preparing for an album the way they normally do. They *had* changed their bass player and they had added another guitar player, so the group had been slightly altered. It was a new experience for them. Even though I had done some homework on their records, it wasn't the same band; it wasn't the same species. I came up here two weekends in December and spent time with them. I set up in a rehearsal hall and took home tapes the first weekend and listened to them. Then I called back and asked when they were rehearsing again, and we spent a couple of hours together. Then we all took a holiday. We later spoke about it some more and exchanged some ideas and came into a studio the last week in January to start recording.

MR&M: How does the Record Plant/Sausalito compare to other studios?

TD: It has all the facilities you need. It's a

little laid back compared to some of the studios I work in, in the sense that it is off the beaten path. It resembles the Miami Criteria facilities for atmosphere, where you don't have nine sessions a day going on, with all the hotshots walking in and out, running from one gig to another.

The members of the Starship were our co-inhabitants for about three weeks. We got to the point where we were eating together, playing pinball together, and walking in and listening to each other's tracks. The chemistry has been family, and it's been very, very comfortable—very pleasant. You get good, healthy opinions. You get empathy, or if you don't get it you understand...you sense it right away.

MR&M: Do you find working with Pablo Cruise different from any of the other acts you work with?

TD: Yes. I have never been approached to do a California concept-type group. It's a new experience for me and it's refreshing. They are very healthy. They have good things going for them. It's a pleasure. As I said, they're a California group and I'm more accustomed to the Eastern, Southern, or even English groups.

MR&M: Tom, let's get a bit of technical information, beginning with the microphones. Were there any special kinds of miking techniques used on the vocals?

TD: The microphone that I have found to be an out in some instances when you're having a wide dynamic range or problems using condensor microphones—because they are so super-sensitive—is Shure SM-7. The group wasn't accustomed to using it. They were used to a Neumann M-49. Depending on who's singing the vocal, or if it's background vocals that we've gone through...I have no allegiance to a specific microphone for all vocals. Depending on the key, the nature of the intimacy of the vocal or the intensity of the vocal, I'll go with the best microphone for the occasion. So far with this group, we've used an M-49, U-87, RCA-44 and a Shure SM-7. They're completely different microphones—ribbon, cardioids, super cardioids...we'll go with the microphone that they're accustomed to working with, and if I think we can improve upon it or by comparison just try one pass at something. I listen to it, and if it's better, I've gained on it; if it's not, I stay where I am.

MR&M: Greg, how about the miking techniques and laying down the basic tracks? Can you tell us a little bit about that?

GP: Again, whatever worked. We tried just about every microphone you could think of. I had set concepts on what would work on drums, etc., but with this project, we used just about everything you could imagine. Whatever sounded the best, as Tom said. If we got to the point where we were satisfied, then we would try one more thing that might be even better. We really didn't get locked into any set system or formula. It's been great. We have used whatever has worked the best.

TD: Greg is familiar with the band. He's spent time with them and he's been witness to their recordings before this. The band has a preconceived notion of what it should sound like, and they've strived for that. I can't just jump in and say I like it or don't like it. I have to preserve the tradition of the band—give them some inputs so that we develop, or update the sound a little bit so that it doesn't sound like everything else. But I still must have a respect for what they are accustomed to. So it's been rather touch and go, but I've a healthy respect for what Greg knows about the band. If I thought we could improve the sound I would suggest something. Or if I suggested something they'd take me one further; it's been like that. Every band is different; you can't use the same microphones even for songs by the same band within an album—it doesn't make sense.

MR&M: Is there anything our readers should know about the way things are set up for the basic tracks in terms of *location* of microphones?

GP: You could get really involved in microphone technique and certain microphones. We've used so many different microphones...I could probably sit here an hour with you going over microphones. One thing I've learned from working with Tom is that preconceived ideas and formulas become better by being aware of what you *can* do. For example, turning a microphone just a little bit, using an off-axis technique or experimenting a little bit instead of setting up your mics straight on—your standard set-up—and going with that and *forcing* that to work. Be a little flexible, move a microphone around.

TD: Prior, with the small group, the band apparently used to layer things. They used to work two and three men on the basic track, then they'd build it out. That enables you to luxuriate in other microphone techniques and other

Longtime Pablo Cruise engineer, Greg Price.

recording techniques and get everything clinically right. I insisted as often as possible on using *everybody*, and as it turned out there's only one song where we only used four men. Otherwise we were simultaneously recording five men. No matter how big or how small the room, I believe in the chemistry of all people participating. It changed techniques, and it changed their playing environment because they were looking at each other and smiling—or warning each other when something was off! The chemistry was there.

MR&M: How do you set up here for your basic tracks?

GP: We worked in Studio A. As I was sitting behind a console, I was directly looking at the drums. The drummer was up against the wall, facing me. Right next to him was the piano. A lot of percussiveness and overall feel of the band comes from the drums and piano. I think it was important to get them together. As far as leakage problems, we had none, despite having the piano and drums right close to each other. There were no problems whatsoever. They have wonderful baffles here to prevent leakage problems. The bass player and the two guitar players were in a semi-circle, everybody facing each other.

We cut one track where the two guitar players used acoustic instruments. We put them both in the isolation booth, facing the rest of the band. I tried to set up so that a real intimacy existed—direct eye contact, so it was pretty much "home" feeling.

I also isolated one of the guitar players' amps in the booth when I wasn't using the booth for acoustic playing. I ran his amp into the booth, while the other guitar player was playing through a smaller amp that I just baffled off to the side. The bass player was direct, and played through a small [Ampeg] B-15; that presented no problem with leakage. In a band concept, you like a little bit of leakage.

MR&M: In terms of working in this studio, were there any significant problems or any interesting situations?

TD: The thing that I had to get acquainted with was that three of the band members had been playing together for twelve years, and they were addicted to appraising their work on 604s; 604s are hard to work on. That's why I bring speakers and my Koss earphones with me. I feel comfortable with my "habit" wherever I work, and I perhaps get one opportunity a year to record where I live, which is in Miami. The rest of the time I'm out someplace.

I can't condemn or appraise any studio because, "I don't like the way it sounds," or this and that when somebody else has made a successful record there. I'd only be casting aspersions on myself. So I take my small speakers and my electrostatic headphones with me all the time, and that takes the "room" out of wherever I'm listening. Usually I bring something that I've just finished recording and listen to it in the room on their monitors; if there's something that's disturbing to me, I'll flip it over onto my phone or onto *my* speakers just to sample it. Then I can say, "Well, if I want that kind of sound, this is what I have to live with *here*," and not assault with, "Get those speakers out of here, etc." We worked on 604s for about 3 or 4 days, but they were a little bit uncomfortable—in the control room, number one, and number two, they're something not commonly used around here so inadvertently we developed problems and blew them.

GP: We couldn't get enough SPL. The band had sort of graduated to a higher SPL level and we went over to the "Meyers." I think both Tom and I were a little skeptical of the Meyer system at first, but again I think that once you get a common ground and know where you're standing you pretty much can work on almost anything.

MR&M: When you're finished with all this work, is it going to be finalized here or is it going elsewhere?

TD: That's conjecture right now. There's a good possibility it might get finalized here. We haven't put the console's computer to the test for an appraisal of whether it's going to be able to handle all the updates and changes that we're talking about. As I said earlier, this is the earliest form of computer and this is the one that developed the problem of introducing time delay to some of the changes when you got past number X. Its memory accumulated up to a certain point and from there on it got slower and slower. I'm anxious to see if it holds up. We'll give it a shakedown in the next week or so and see what happens. If not, we'll try to find someplace that would be beneficial to all of us to work in and mix down. It would have to be a place that has empathy for the nature of the equipment we've recorded on, which is M-79s and an API console. We have to take that into consideration. We just can't go switching machines and consoles that gracefully and come off with the same product we started out trying to make. You can't do it.

Cory Lerios and producer Tom Dowd conferring at the console.

MR&M: Is the band well rehearsed when they get into the studio?

TD: For this particular project I'd say yes. I don't think there were more than one or two major changes that we talked about before we got here. We recorded the song the original way and then applied the changes which were already predetermined.

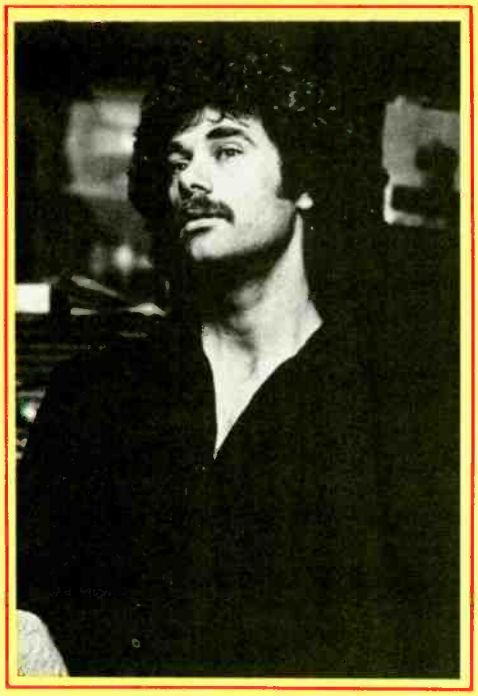
We broke at one point for a few days during the recording because we did tear down and set up several times. I didn't believe in fine tuning all the songs that were going on the album, and then going back into the studio and trying to do all of them one after the other. We spent four or five days prior to going into the studio just concentrating on four songs. We went in with the idea of getting those four songs, then taking three or four days off and working on one or two more, coming back in, getting them, taking off again, etc. Incidentally, during that time, one of the songs we recorded, that we were very high on when we started, was knocked out of the album because we wrote a *better* song.

GP: It was a unique situation. I think so many times people go in and expect to just blast out ten or twelve songs. This way, as Tom said, other songs were written that preceded a song that had been built up by the band so much. I think it was really a very smart thing to do.

TD: Of the original eleven or twelve songs, only eight are in the album; three new songs were written for the album during the little ups and downs that were inspired by, "Hey, you think that's good, I've got a better idea!" *Boom!* It was when we were rehearsing that someone would say, "You know, as much as you like that song, listen to this." We'd work on it a day or two and say, "You know, it *is* better!"

My influence on the group, I hope, is positive. We'll see..we'll see if the public resents it, or if they appreciate it...or, rather, if they *like* it—not appreciate it, but *like* it.

David has had the responsibility in the last two albums of doing all the vocals. I coerced Cory into trying more vocals. There's one song in this album that was determined to be an attempt at Cory to sing, and after I worked with him on that song, they wrote another song. I said, "Cory, why don't you sing this song also?" We were taking a little heat off David and giving Cory an opportunity to try two of the songs, because Cory can sing. He's done backgrounds here and there, and he does backgrounds "live." I guess nobody ever put the responsibility on him. Now, with the new members, we're talking about doing three and four man background parts, instead of Cory and David doubling and switching parts, etc. We now have four people who can go in there and sing the background.



Vocalist David Jenkins.

David is still responsible for the identification and the mass appeal and sound of the group. David is the one I rely on for the concept of most of the group parts. If there's going to be a change, it would be David that I would address about trying it this way or that way. That's his strength. On the other hand, I've been encouraging Cory as much as I can to sing solo, and I've been encouraging John Pearce to sing background. The couple of times that we've gone in and worked on background vocals, it's been fruitful. I think it's an improvement. It's healthier to have as many people—as many inputs—as you can. That's what's happening now.

MR&M: It seems that you get very personally involved with the people you work with.

TD: Yes. I do probably only three or four albums a year, but I figure the fact that they've asked me to do them, I've got to give them all of my ideas... whether they're constructive or whether it's where they're coming from...I've got to at least give them the ideas. And I develop no ill feelings towards someone who says, "I don't like it." It's just another opinion—try it.

MR&M: That's good for creative interaction.

TD: If I were an artist I wouldn't be doing what I'm doing. What I'm doing is trying to encourage the artist to be comfortable and do what they're capable of doing—one way or the other. You either push them or pull them.

At this point in the interview, Cory Lerios and Dave Jenkins of Pablo Cruise stepped into the studio to join the discussion.

MR&M: Would you detail the keyboards that you are using on *Reflector*?

Cory Lerios: I have been playing a Yamaha grand piano C7B on all the basic tracks. It's actually the in-house piano here at the Record Plant. I also am using a Fender Rhodes, a Prophet synthesizer from Sequential Circuits and a Yamaha CS 80, which was modified by Gary Leuenberger. Gary and I worked together to program the

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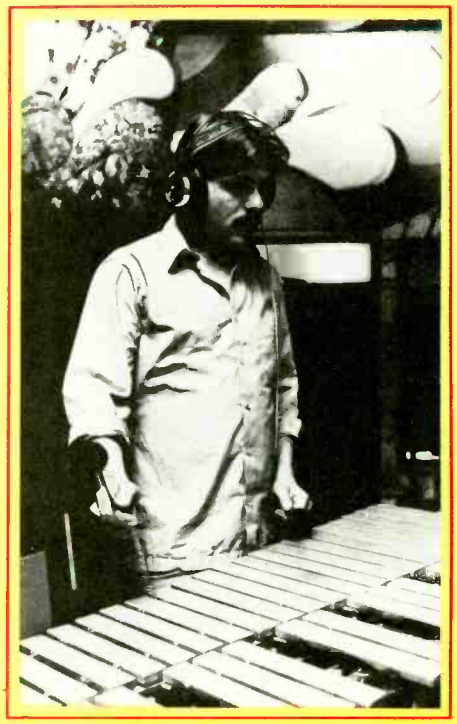

Percussionist Steve Price.

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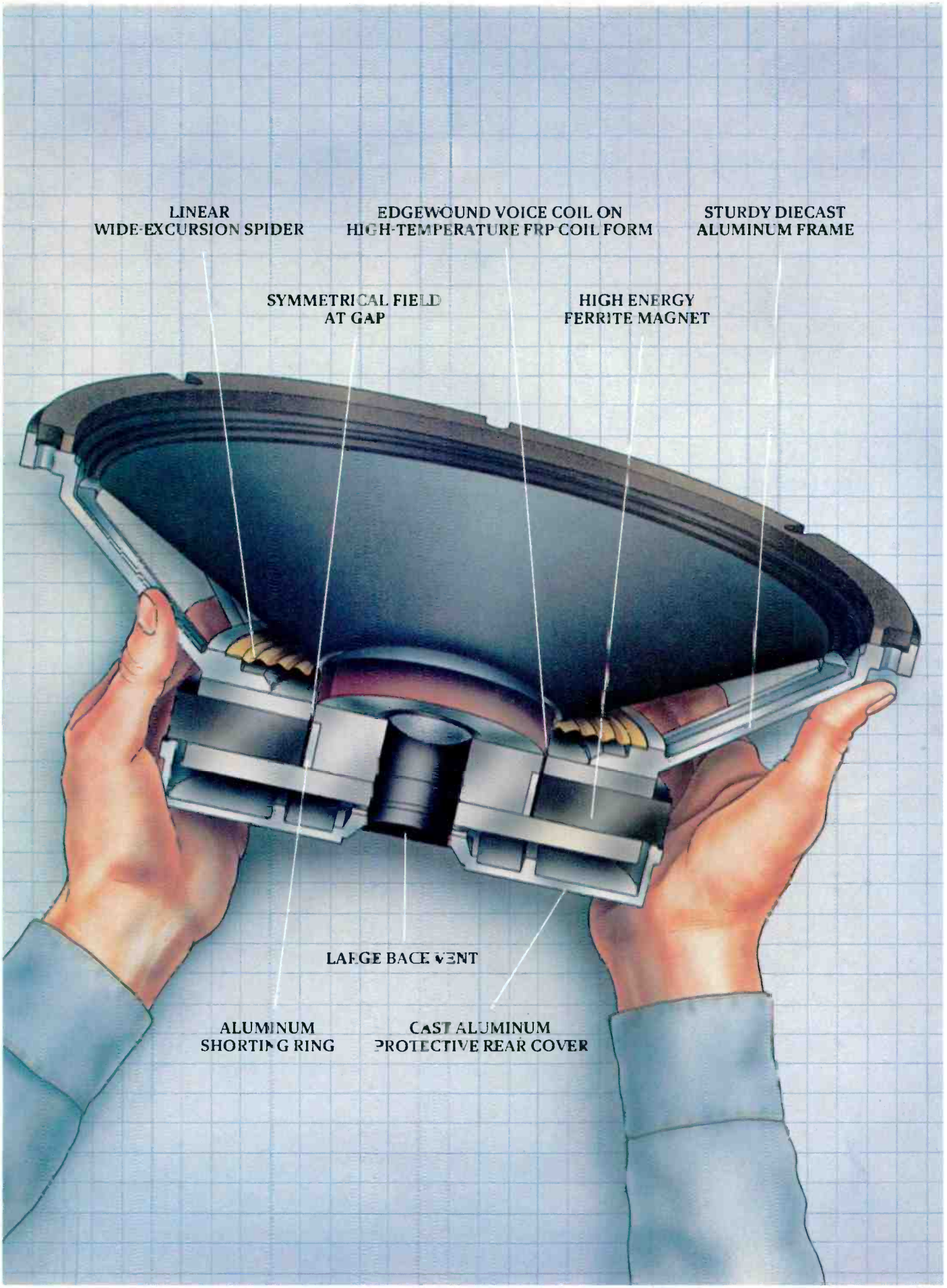
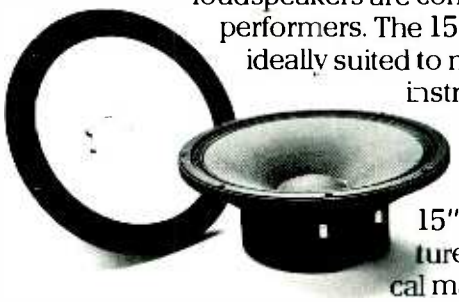


YAMAHA WOOFERS

Representing Yamaha's high technology and careful attention to quality control, these loudspeakers are consistently top performers. The 15" JA-3881 is ideally suited to musical instrument and sound reinforcement systems. The 15" JA-3882 features a symmetrical magnetic field, ultra-high-strength magnet and low-mass cone that make it an ideal choice for horn-loaded bass loudspeaker systems.



Powerful Ferrite Magnet Maintains High Efficiency

The motor assembly (magnet, magnetic assembly and voice coil) is the heart of a good loudspeaker. Yamaha woofers utilize specially formulated ferrite magnets, created in our own metallurgical processing plants.
A strong magnetic charge, plus inherently high retentivity, assure the speaker will retain its original efficiency even after hundreds upon hundreds of hours of high-power, high-temperature use.

Precision Magnet Assembly Concentrates Energy

Along with the magnet itself, the top plate, pole piece, and bottom plate constitute the "magnetic assembly." This assembly generates the magnetic field and guides the flux to the voice coil gap. Because we manufacture our speakers to precision tolerances, we are able to use narrower gaps, which concentrate more magnetic flux in the voice coil area. This ensures high efficiency so you get more sound per watt of amplifier power.

Symmetrical Magnetic Field Lowers Distortion

The JA-3882 utilizes a symmetrical field to maximize linearity (lower distortion) in high-accuracy sound systems; whereas the JA-3881 field is designed for ideal tone color in musical sound reinforcement systems.

Edgewound Voice Coils and FRP Forms for Power and Efficiency

Yamaha voice coils are edgewound and made of either copper (for its very high conductivity) or aluminum (for its low mass). The edgewound coils pack the most wire into a given cross-sectional area, and hence are more efficient than round wire designs. Yamaha's high-temperature FRP voice coil forms, which support the coils, have a low coefficient of expansion, so they are dimensionally stable. Also, unlike aluminum forms, Yamaha's are free of power-robbing eddy currents.

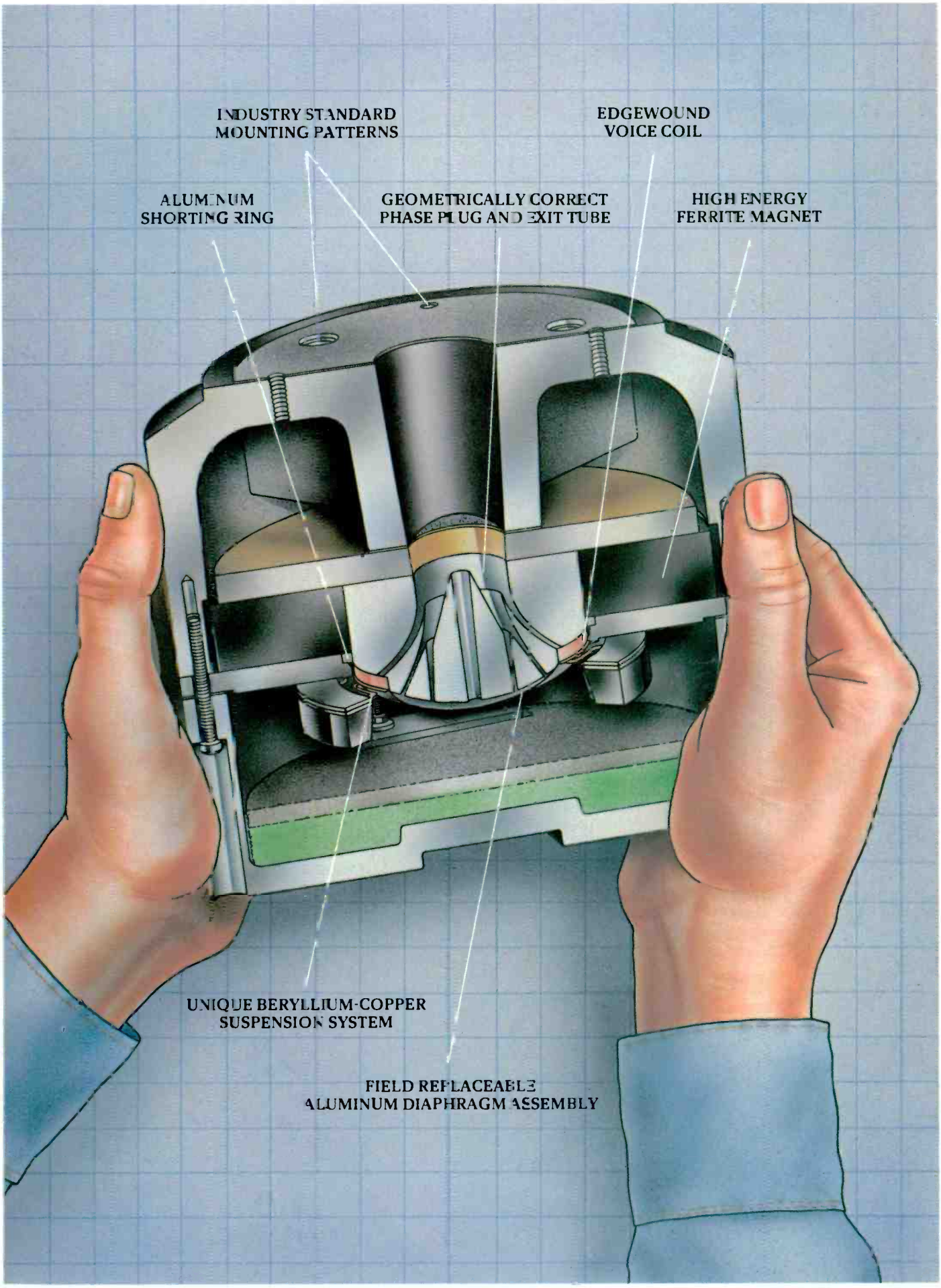
Cone and Suspension Linear Over Wide Dynamic Range

For greatest linearity and power handling capacity, the suspension (spider and surround) centers the cone in the voice coil while maintaining uniform tension at all times. This is achieved by carefully bonding to the cone a sine-wave patterned spider and a saw-tooth patterned surround—each made of a different fabric and impregnated with a different resin.

Diecast Aluminum Frame and Cover for Structural Integrity

Yamaha's rigid and structurally secure frame keeps the cone, voice coil and magnetic assembly in precise mechanical alignment. Even under the rugged conditions of a concert tour, the frame will not warp, twist or resonate. A large back vent relieves air pressure behind the cone cap and also improves cooling for sustained high-power handling.

YAMAHA

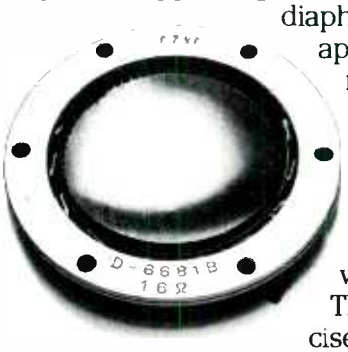


YAMAHA HIGH FREQUENCY COMPRESSION DRIVER

Yamaha's advanced technology, backed by decades of experience in the music industry, has produced the JA-6681B high frequency compression driver, a product with seemingly subtle improvements that make a not-so-subtle improvement in performance. The JA-6681B may be used as the mid or mid/high frequency component in two-way or multi-way sound systems.

Aluminum Diaphragm and Beryllium-Copper Suspension System

Yamaha's exclusive suspension consists of beryllium-copper fingers bonded to an aluminum diaphragm. The diaphragm is a pneumatically-formed aluminum dome whose rigidity and light weight provide optimum high frequency response with low distortion. The Be-Cu fingers precisely center the voice coil in the gap, a unique suspension which permits wide diaphragm excursion for high power handling, while maintaining excellent linearity for low distortion. Because severe high-power pulses do not cause the suspension to "take a set," the sound quality remains excellent throughout the life of the component.



Powerful Ferrite Magnet Maintains High Efficiency

Created in our own metallurgical processing plant, a custom formulated ferrite magnet provides the strong field essential to the driver's high efficiency. The magnet's inherently high retentivity ensures that the original strength will endure hundreds upon hundreds of hours of high-power, high-temperature use.

Aluminum Shorting Ring for Lower Distortion

The JA-6681B has a pure aluminum ring around the pole piece. The ring "short circuits" the local electrical currents (eddy currents) that invariably are induced due to the voice coil's rapidly changing magnetic field. Without a shorting ring, the flux density of the magnetic assembly would be "modulated" by the sound, causing distortion. The shorting ring therefore reduces distortion to a minimum.

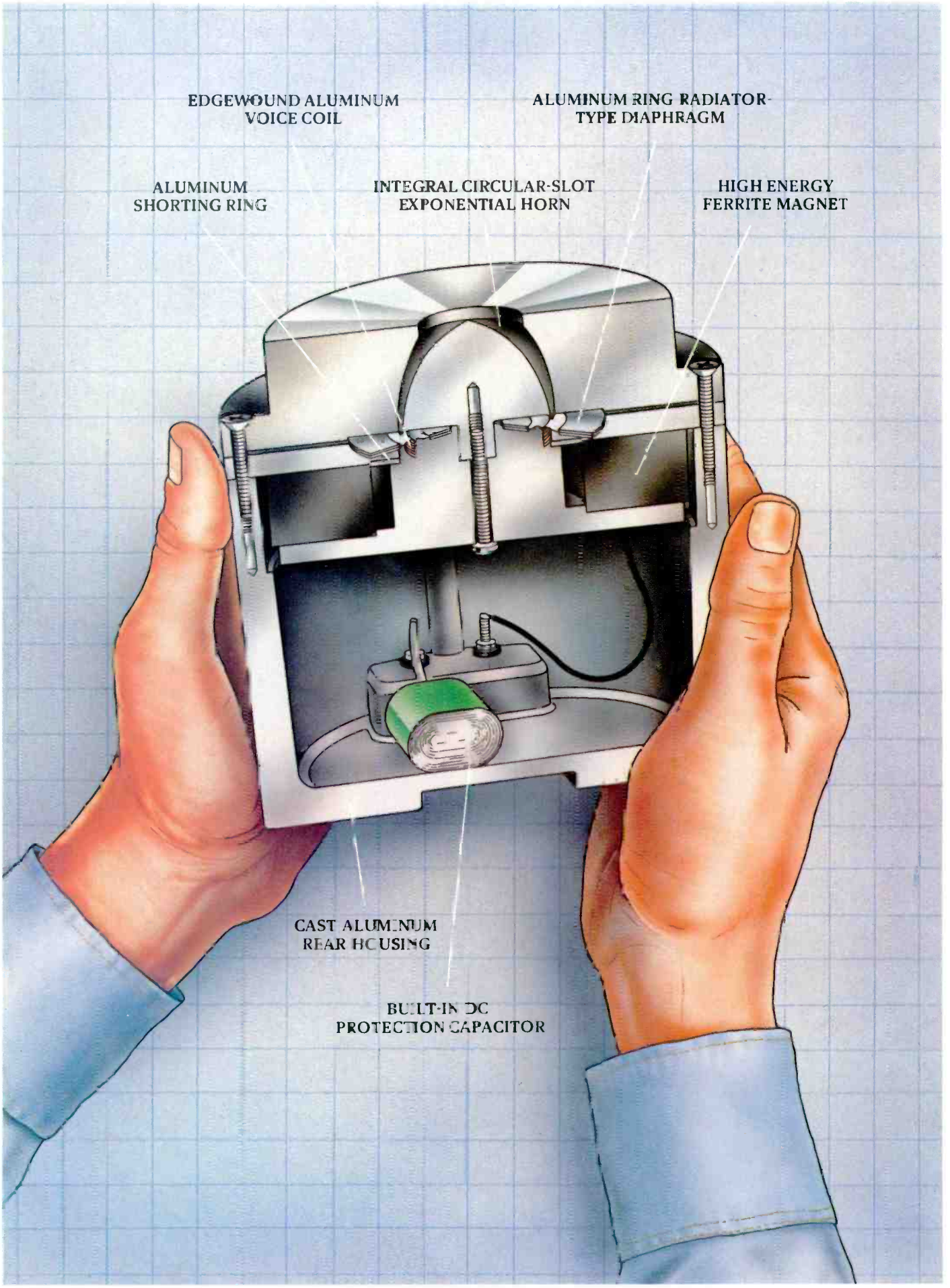
Geometrically Correct Phase Plug and Exit Tube

Yamaha's machined and diecast zinc-alloy phase plug gathers sound over the entire area of the diaphragm, bringing it together, in phase, at precisely the right point to form a coordinated wave front at the exit end of the driver. The exit tube couples the sound from the end of the phase plug to the beginning of the horn/adaptor assembly. The JA-6681B exit tube exactly matches the driver to the H-1230 horn, or to any other properly designed horn.

Cast Aluminum Housing

Yamaha's rigid diecast aluminum housing acts as a compression chamber which properly loads the diaphragm assembly. A combination of felt and foam damping material inside the housing avoids unwanted resonances. The housing also protects the diaphragm and magnetic assemblies from corrosion, dust and accidental damage.

YAMAHA



YAMAHA SUPERTWEETER

The Yamaha JA-4281B is a ring radiator-type compression tweeter. (The compression tweeter is a specialized compression driver with an integral diffraction horn.) At very high frequencies, a conventional dome would tend to break up and resonate, reducing the sound output and increasing distortion. Yamaha's JA-4281B substitutes a rigid aluminum ring for the dome, and thus produces high sound power at very high frequencies with low distortion.

Powerful Ferrite Magnet Maintains High Efficiency

A custom formulated ferrite magnet provides the strong field essential to the driver's high efficiency. The magnet's inherently high retentivity ensures that the original strength will endure hundreds upon hundreds of hours of high-power, high-temperature use.

Precision Magnet Assembly Concentrates Energy

This assembly generates the magnetic field and guides the flux to the voice coil gap – the relatively small area in which all the magnetic energy is needed. Because we manufacture our components to precision tolerances, we are able to use narrower gaps, which concentrates more magnetic flux in the voice coil area. This ensures high efficiency so you get more sound per watt of amplifier power.

The JA-4281B has a pure aluminum ring around the pole piece. The ring "short circuits" the local electrical currents (eddy currents) that invariably are induced due to the voice coil's rapidly changing magnetic field. The shorting ring therefore reduces distortion to a minimum.

Integral Diffraction Horn Matches Ring Radiator to the Acoustic Environment

The pneumatically formed aluminum ring radiator, with its anti-fatiguing aluminum compliance, is rigid and lightweight for the best frequency response and lowest distortion. The JA-4281B voice coil is edgewound and made of aluminum (for its high conductivity and low mass). The edgewound coil packs the most wire into a given cross-sectional area, and hence is more efficient than round wire designs.

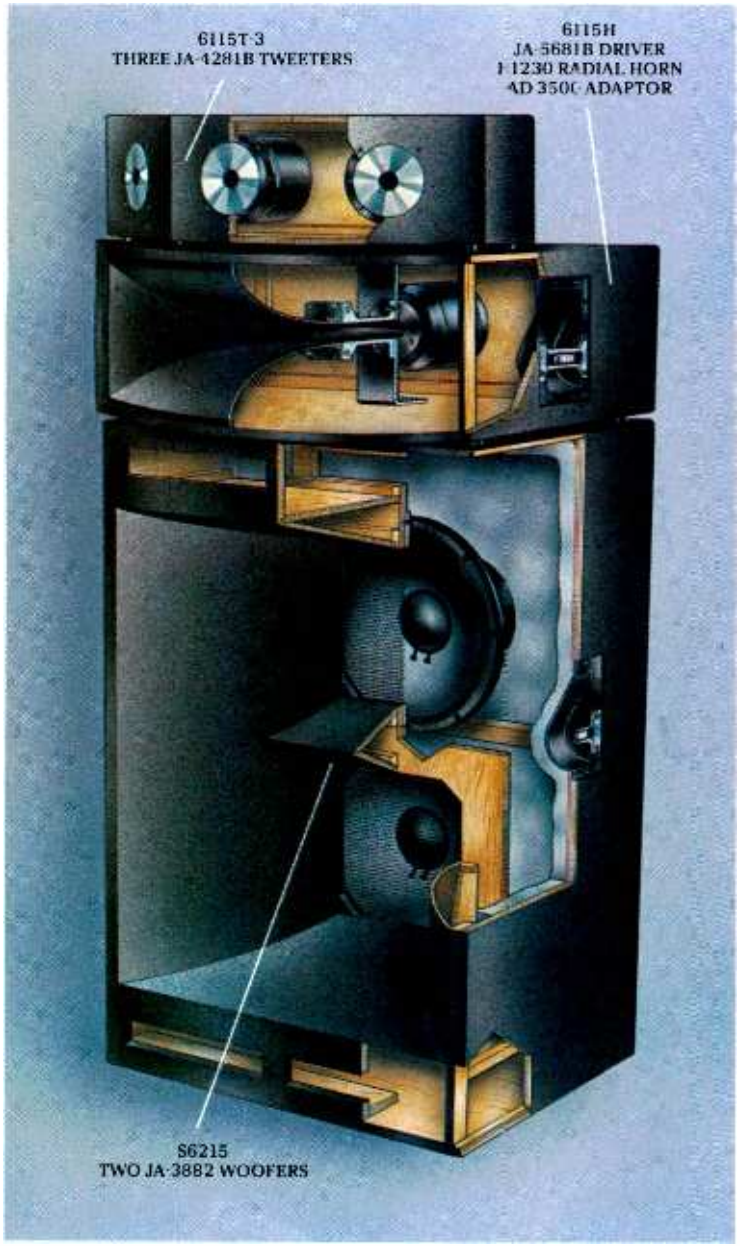
In the unlikely event of failure, the entire diaphragm assembly can be replaced in the field, in minutes, using common tools.

A circular slot-type exponential horn picks up the sound at the ring radiator diaphragm and acoustically couples it to the environment. The geometry of the diffraction slot (the exponentially expanding area between the bullet-shaped plug and the contoured front housing) permits wide coverage (120° @ 10 kHz) and properly loads the diaphragm.

Built-In DC Protection Capacitor

An 18,000 microfarad capacitor, located inside the driver, is wired in series with the voice coil. This blocks DC components from reaching the coil, thus lessening the chance of distortion or overheating in the event of amplifier malfunction. The capacitor is large enough so it does not affect response within the frequency range of the tweeter.

YAMAHA



S6215HT-3
MULTIPLE ENCLOSURE SYSTEM

Taken separately, Yamaha speaker components offer significant advantages. Put together in a Yamaha speaker system, these advantages combine to offer you professional sound that is unmatched by anything in the industry. The S6215HT-3 is an example of how these speaker components come together in a system. The system is modular, and the fully assembled enclosures are available separately.

Thiele-Small Aligned Design

The bass reflex enclosure has a computer-generated Thiele-Small aligned design giving optimum mid-efficiency and superior low-frequency loading.

Heavy-Duty, Roadworthy Features

The cabinets are made of 9-ply ¾", maple. All joints are lock-mitered and glue-blocked. All hardware on the rear panels is recessed. All handles are also recessed and are located at balance points for easy handling.

In the mid-range enclosure, the bottom, sides, and top are integrally tied to the horn and driver for maximum stiffness and light weight.

The system is loaded, painted, has feet and grilles, and is thoroughly tested and ready for high-performance sound reinforcement.

Now that you've heard what we have to say, listen to us. You'll hear what all this technology adds up to: better-sounding, more reliable speaker components and systems for your music. You'll be glad you held out until you got hold of Yamaha.

For more information, write: Yamaha, Box 6600, Buena Park, CA 90622. (In Canada: 135 Milner Ave., Scarborough, Ont. M1S 3R1).

YAMAHA

CS 80, and we're pretty excited about some of the things we've got. We run everything direct and we used two [AKG] 414s on the grand piano.

MR&M: Dave, would you run down the guitars and amps you've been using on these sessions?

Dave Jenkins: I've mainly been using a Stratocaster, a Gibson 335 and a Martin acoustic. I also have used on this LP an Ovation and a Les Paul for a few parts. I have a Rickenbacker 12-string, but I haven't used it on these sessions, yet. I play the electrics through a small Music Man amp.

MR&M: What's it like working with Tom Dowd?

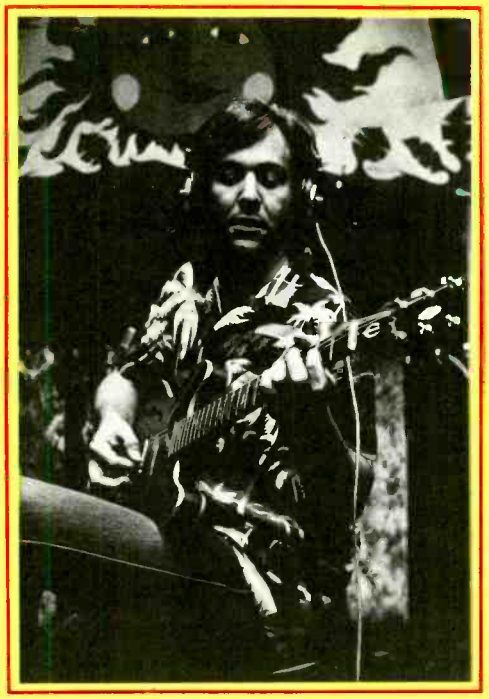
CL: It's been *great*!

DJ: He has been doing it for years and years. He is such a pro. You know, you go out to dinner with him, and he can tell you some stories about sessions he's worked. You could stay up all night listening to all the great stories. He has been involved with recording every step of the way!

CL: It's really unbelievable. In the course of a conversation he'll bring up musicians' names that he has worked with, and I am talking about *legendary* musicians and sessions.

MR&M: Bill Schee worked with you a lot in the past.

DJ: Bill is one of the best recording engineers in the business; he's incredible. He did three albums with us. He has great musical ideas, too. It's good to change, and at this point in our career we decided, "Why not? Let's try something new." It worked out well, having Tom do these sessions.



New addition, guitarist Angelo Rossi.

Keyboardist/vocalist Cory Lerios.

MR&M: Do you like working here at the Record Plant?

CL: Yes, well...this is pretty much like home base, because back in 1974 we did our first A&M demos in this studio. Then we made our first *successful* album here, *A Place In The Sun*. We did that whole album here and we did about 60% of *Worlds Away* here. Along with about half of the last album. It's comfortable here. It's always been a comfortable place to work, and they take care of everything we need here as far as accessories.

When we were putting the finishing touches on the last album, which you may or may not know about [laughing] it actually *was* released...we were at Sunset Sound and we had a great time there, but the group Toto was in there, and there were just a lot of musicians around all the time. You get to a lot of studios and hear a lot of music, which is nice, and which is what goes on here in Sausalito. Since we've been in here, the Starship has been here, Mickey Thomas has been in here, Marty Balin—a lot of good music has been going through here.

MR&M: You've had a number of different artists come in and work with you in the past. How does that come about?

CL: It's producer-instigated. In the case of those records that was Schee's choice. This is one of the things I love about Dowd. Tom is really able to get us to delve into certain areas that we've probably been capable of working, but for one reason or another haven't.

DJ: What I've got to say about Tom...the way he talks about music and bars and measures and stuff...this is the first time we've been so exposed to such a musical jargon. Tom is total music. He's amazing, always talking music. It's great. He's always pushing us a step further. He's quick with examples and analogies.

CL: I personally can't say enough about how much fun I've had working with him. It's been a joy. He's brought a lot out of us. And the other thing in this album is that we have brought in a couple of new guys and it has made a big difference. John Pearce, playing bass, is truly a gifted musician. He's a really young guy. He's been incredible. He's not only added to the records musically, but as a bass player, he's also been instrumental in writing a lot of stuff. Angelo Rossi, whom we've known for years, is playing guitar and playing great.

DJ: I think the band is going to be great. Over the next year it's really going to get hot, because it's a real solid band. It's fun; the shows are fun. The business of the studios is taken real seriously and we work really hard. When we get out to the stage we have a real good time.

MR&M: How do you compare the two—the studio and stage?

DJ: It's a balance; you've gotta do both.

CL: I love them both. The only thing that can be detrimental at times is to confuse the issues. When you're in the studio, it's a different kind of concentration. It's more meticulous because it is being *documented*.

DJ: There's a sensitivity that goes into a studio that everybody is well aware of. When you're tracking or over-dubbing, you can hear so well that you can get as sensitive as you need to or want to. "Live," it's a little bit rawer and a little tougher to get that sensitivity, although with this band I notice everyone is trying more and more to achieve that sensitivity. At the same time they're keeping the "live,"

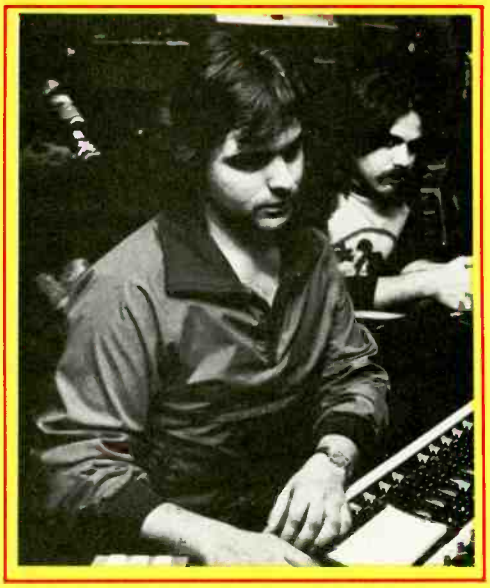
energetic approach. It's a nice combination if you can find that balance; it really does give the audience a good show.

MR&M: Did you ever have any problems learning to sing and play at the same time?

DJ: Well, you just gotta learn parts. It blows my mind how anybody can play bass or drums *and* sing. *Forget* it. You watch Steve Price play, the guy's independence is ridiculous. As far as singing and playing, bass seems difficult for me. Also, the keyboard parts that Cory plays and sings at the same time would be real baffling to me.

CL: You just learn the parts. There have been things we've recorded that we never paid too much attention to as far as having to play *and* sing them until we actually made the record, then went in to rehearse for shows and started to work on it. Some of them were almost impossible. It never really reaches the impossible, but it's so much work that it isn't fun. The best stuff "live" is the stuff that comes naturally, which, incidentally, is another thing that sets our present sound apart from the other albums we've made—all the albums in the past. We've made an album here that we can perform "live."

MR&M: Can you explain a little bit about some of the work that the group does before you get into the studio?



Engineers Greg Price, Rick Sanchez.

DJ: There's a back room here at the Record Plant that we did a lot of rehearsing in. We have a little 8-track scene out at the beach where we live, and we did a lot of work out there. We did work out a lot of parts—tracking, and even did some tracks with a drum machine before we actually laid it down with Steve. We did a lot of work with this drum machine, working out vocal parts, actually singing a lot of stuff to try to get it to where it was comfortable to perform before we came in and tracked it.

MR&M: So you have production facilities in your homes so that you can do that kind of work?

DJ: Yes. It's really nice, and we save a lot of money that way. It's expensive in the studio.

CL: In many cases when we come in here and track basic tracks, we bounce the tracks up there. A fairly good basic track mixed down to two, and then we'd have six tracks to play with—vocal and some guitars. There's a real positive thing about working with an 8-track studio, or any small studio. In the case of the studio that we built (Greg Price and Steve Price built it), it's a separate building, and we got pretty elaborate. What happened is that we were getting some really great sounds out there and then we came back into the studio here. I even was up against that last night on dubbing some synthesizers that I'd done up at the house. Then when I came back and tried to do it here, it wasn't the same. But the small studio is great for vocal workouts.

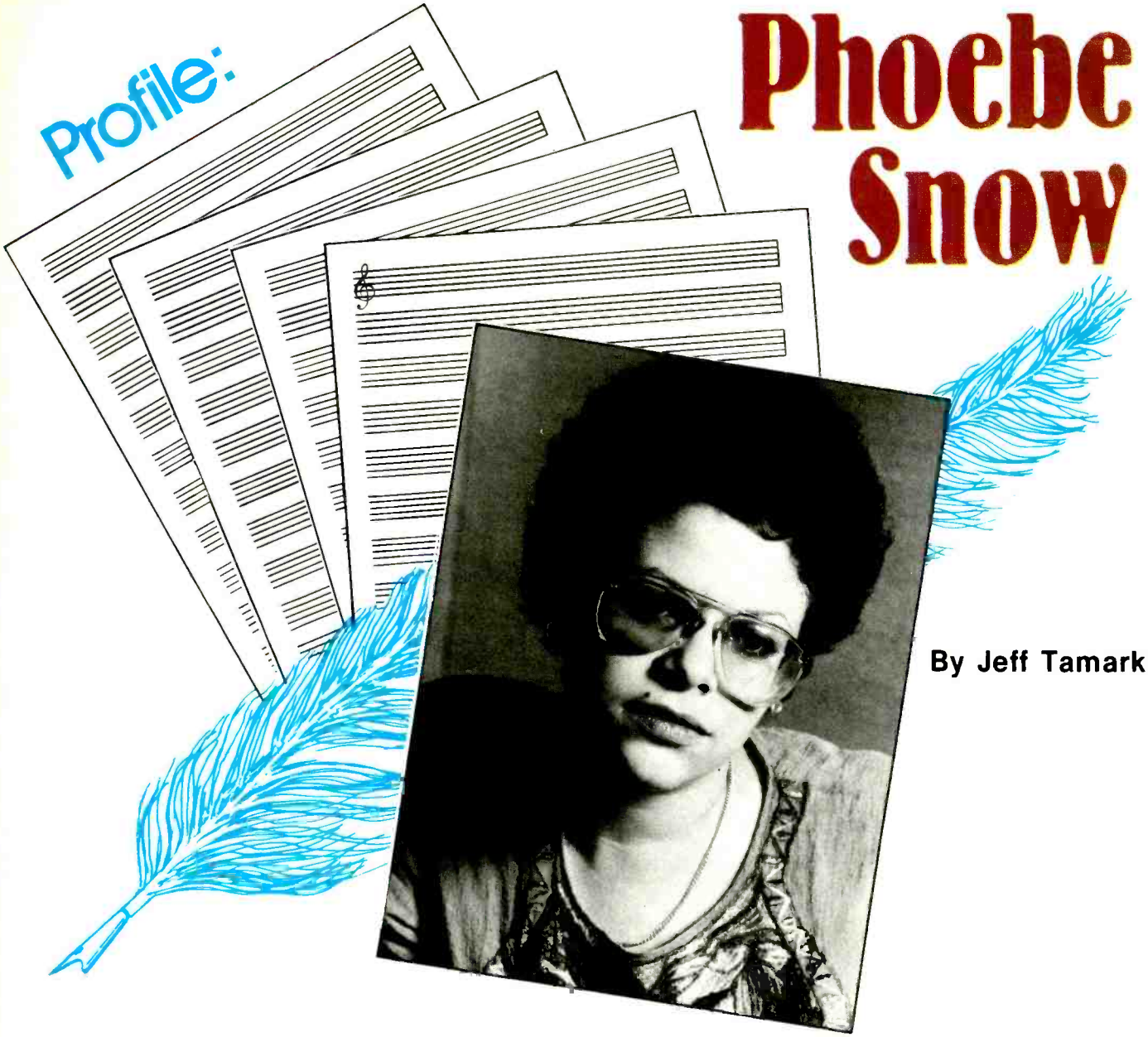
MR&M: You put a lot of emphasis on your vocals; it must take a whole lot of meticulous work.

DJ: I like to lay down a few different vocals. I think everybody does it this way. Some great singers go in, and with just one take they've got it. I'm not that good, so I like to lay three down, or four down, and then choose the best parts of what I've got there. Overall the whole thing works for me. And the thing that Tom did with this album was stress the importance of the lead vocal. Before he had talked about cutting a hit track; let's hear the song; let's hear the poem; let's hear the story; let's hear the actual tune which is the vocal. The message, what is it? And Tom, more than any other person that we've ever worked with, is really adamant about stressing the importance of the vocal, the story line and how it works, what it does. That's your main deal—the story—and all the other stuff is incidental. It's been good to get that input. In the past we've cut tracks and written songs to the track. This time we're looking at it a little differently. I think, as writers and performers, it's definitely going to help our overall view of things.

Bass player Pearce, who's recording with Pablo Cruise for the first time.

Profile:

Phoebe Snow



By Jeff Tamarkin

When you hear the name Phoebe Snow, what comes to mind? A singer of jazzy ballads? A maker of light, contemporary, easy-listening music? The woman with the quirky voice who in the early 70s had a hit called "Poetry Man" and who sang with Paul Simon on his hit "Gone At Last"?

All of those things should come to mind, but now there's a new definition which fits Phoebe Snow: rock and roller. Phoebe recently released her first album for the Atlantic-distributed Mirage Records label, and it shows a side of her talents never before fully realized. Phoebe Snow lets loose on Rock Away, *putting her in a league with the few great rock singers of today.*

Rock Away *was produced by Greg Ladanyi and Richie Cannata, and finds Ms. Snow covering the works of such notable songwriters as Bob Dylan, Allen Toussaint, Rod Stewart, Don Covay, Carolyne Mas and, of course, Phoebe Snow. Ladanyi, who has produced Jackson Browne, Warren Zevon and others, and Cannata, a member of Billy Joel's band, have given Snow a raw, basic sound on this LP, while managing to keep the singer's distinctive voice up front throughout the record.*

In a recent interview with Modern Recording & Music, *Phoebe Snow detailed her career history and the making of her latest album. She proved to be a warm and humorous profile subject, often joking and always maintaining a casual air. Ladanyi spoke to* MR&M *from his California home about his role in the production of* Rock Away. *Mr. Cannata was on tour with the Billy Joel band at the time of this interview, and so as a result was not available.*

Modern Recording & Music: Did you have any formal training as a musician?

Phoebe Snow: No, I took guitar lessons, and I was going great guns for awhile, but that is about as formal as I got. When I got into guitar lessons I thought I was definitely going to be a guitar player—that's it, I can't sing, I can't carry a tune. I couldn't in those days. Well, I could, but after I'd carry it I'd want to drop it again. Other than

that, I took a month of voice lessons once, then decided I didn't want them. They bored me to death.

MR&M: I guess you changed your mind about your potential as a singer.

PS: It's funny because I'd go back now and take voice lessons. I take it more seriously now; I was just goofing off for the past five years.

MR&M: Whom did you listen to for inspiration when you started out?

PS: When I first started tuning into music it was a weird mix of top 40 and Broadway. Then later on I discovered jazz. My father used to have great recordings by the Dukes of Dixieland, Scobie and Clancy's Frisco Band, Sidney Bechet, Art Tatum. I'm not a jazz buff, but I know some jazz stuff. From there, I guess it's a tired story, but I was dating a guy when I was about 19 to 22, and he was a jazz fanatic. He had everything alphabetically taped and stored away; he turned me on to Billie Holiday, Charlie Parker and Lester Young, and some gospel. He was the first guy to turn me on to gospel; one thing we didn't have in my house was gospel music. I think rock and roll is the mixture of all that stuff, anyway.

MR&M: What happened next in your career?

PS: I started to play at clubs in Greenwich Village like the Gaslights I and II, and the Cafe Au Go Go. I did 15 minute amateur sets, nothing I ever got paid for. That's where Shelter Records found me, schlepping around there.

MR&M: And they just signed you up, right then and there?

PS: Yeah, it was a real Cinderella story. But let me say something I've said before: don't do it that way. People often ask what advice I'd give. I'd say do *not* sign up with the first person who offers a contract. That's bull. I didn't know anything about the business; I thought I was a real hotshot, because I could think straight. But I found out later I couldn't even do that. You don't know what's around the bend; you have no idea.

MR&M: Can we assume from what you have just said that someone somewhere ripped you off?

PS: Some*body*? Singular? *Nooo*, try five or six somebodies! It wasn't my fault, 'cause I'm a smart cookie, and I think most people who are creative are also smart. It's not that you're not smart, folks, it's that you just don't know...they call them loopholes.

MR&M: When was "Poetry Man" written? What were the circumstances behind that?

PS: That was written when I first started to play clubs. I have to tell you something; this is really great. I used to go around with my half-finished song fragments. We were a gang, like the "American Bandstand" of the Village, the amateur night regulars. We all played our original material for each other. We were all so naive in those days; nobody ripped anybody off. In fact, some of the people playing around at that time were Garland Jeffreys, Melissa Manchester...and others like that. In any case, I sat down with one of these people, who has since gone on to become a carpenter. He was very good in those days. I sat him down and said, "I've got to play this for you; you've gotta hear it." I got about halfway into the song and he goes like this. *[She makes a sound commonly known as a raspberry.]* He said, "Are you sure this is what you want to do? Maybe you should just sing other people's stuff. You were good at the blues."

Oh, by the way, when you asked me what music influenced me, I was very remiss in not saying the *Blues*, with a capital B. I think I'll do some more on the next record.

MR&M: What was your first experience in a recording studio like?

PS: Tragic. I was like, "Wow, look at those big things!" What a jerk. I was scared. I was intimidated because everyone seemed to know what they were doing and I seemed to know that I didn't know what I was doing.

MR&M: Do your early records now seem amateurish to you?

PS: You bet. Amateur isn't the word. "Lousy" is. I don't like my old records at all. I like the first one, but for very personal reasons which have nothing to do with aesthetics or dollars or bullets, but rather with guts. That was my guts out there on that vinyl. *Squish.*

MR&M: Phoebe Snow is not your real name, correct?

PS: Phoebe is my real name. Phoebe Snow. Ask about Phoebe Snow.

MR&M: I just did.

PS: OK, then I'll tell you. When I was growing up, I used to live on Palisade Avenue in Teaneck, New Jersey, across the street and up the incline from two train tracks. Every day, about twice a day, the freight trains would go rolling by. We used to go play in the swamp by the tracks—it was a very groovy place for kids to play. There were reeds and cattails and swampy marsh and little grassy hills.

There was a little trail, and we used to say it was the trail that George Washington and his men traveled when they advanced on Jersey. Anyway, one of the freight cars was labeled "Phoebe Snow." That was the biggest I'd ever seen my name in print, and when you're a little kid that's really heavy-duty. All the kids in my gang would say, "Hey, is that you?" I used to say, "You can make fun of me if you want to, but when I grow up and I'm famous, I'll call myself Phoebe Snow." So I did.

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MR&M: Back to the grown-up Phoebe Snow. The Shelter contract bit the dust and you went to CBS. How did that happen?

PS: While Shelter was biting I was brought there [CBS] to audition. I just felt that Shelter and I had come to a parting of the ways. Bad chemistry. It just wasn't working for me as a business thing; too much that I'll never talk about had gone down.

MR&M: How did you hook up with Paul Simon on "Gone At Last"?

PS: He was recording his album (*Still Crazy After All These Years*) at the same time I was doing my second one. As I'd be going out for the day, he'd be coming in. Phil Ramone was producing me and I kept pestering Phil "Can I meet him?" I was such a groupie in those days. He said OK, and finally one day we were in the room together, and I was reading a copy of *Billboard*, probably upside down, and finally he said, "Well, it's nice to be on top, isn't it?" At that point, I was high in the charts with the first album. I said, "Yes, it is," and we started talking, and found out that we both loved gospel. He said, "It's funny you're saying that because I have a gospel song for you. Of course, I don't know if you're into doing it." I said, "Yeah, I think I could do it." So I did, and he did, and we did.

MR&M: I understand you then had some problems with CBS.

PS: Not so much problems as...it's like a personal relationship, when it's over, you're not gonna beat a dead horse. I had some very educational and interesting experiences at CBS, without which I could not have gone on and continued in this business.

MR&M: You're now on your third label. You must have gone into this one believing it would work out better than the others.

PS: I have no guarantees that

anything is gonna be any better than anything else. I feel so good here. They [Mirage Records, an Atlantic subsidiary] gave me a chance to start fresh. We're all working together now; it's like a little team.

MR&M: CBS recently released a *Best of Phoebe Snow* album. Is it a "coincidence" that it came out the same time as your album for Mirage?

PS: I'd have to be pretty naive to think it was coincidental. I haven't even seen it yet. I was sitting around my house when a friend of mine came over and said, "You know, there's a 'Best of you'?" I said, "Yes, of course." He said, "No, you don't understand. CBS put out this album." I laughed hysterically and said, "Far out." Maybe they can be mutually helpful. We might as well work together as brothers under the skin. There's no reason for animosity. I didn't leave on a hostile note.

MR&M: When did you begin work on your new album, *Rock Away*?

PS: About two years ago. There were several aborted CBS projects. I had about half of this material ready for them back then. We filed that down from six to three songs we thought were recordable, or changeable. As for picking the songs, I think it's important to say that Richie Cannata, who co-produced the album [with Greg Ladanyi], was with me throughout the entire time, since about July 1979, when he started as a band member on one of the aborted projects. He ended up being a dear friend and a supportive associate. He'd come to my house one or two times every week or so. We'd take a pile of records and he'd ask me what I'd like to cover, since I wasn't writing.

With his help, we did the preproduction work by ourselves, between labels. For about eight months, I had no label. I walked out of CBS into the abyss. So he said, "The hell with it; let's pretend we've got something." I couldn't have made this album without Richie Cannata; there's no question in my mind about that.

MR&M: You mentioned that you weren't writing, and there are only three original tunes on the album. Why did you stop writing?

PS: I've been busy, and the muses have left me. I was being with my daughter. That takes all the juice out of you. When it gets to be 7 o'clock at night and the house quiets down, there's nothing left. It's been real hard.

MR&M: What else took place in the pre-production? Do you have a home studio where you tried out ideas?

PS: No, but I have a hand-held tape recorder that fits into my purse and I used to hum into it late at night. If I had an idea for a song, I'd hum it into the tape recorder and then put it back in my pocketbook. The next morning I'd listen to it and make whatever changes I felt should be made. I can't read or write music, so my methods are primitive—just a lot of humming and strumming.

MR&M: Do you work out the songs with a band or by yourself?

PS: By myself, on guitar. The melody line has to come from the throat.

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MR&M: You've chosen some interesting cover material for *Rock Away*. Where did you get the Bob Dylan song, "I Believe In You?"

PS: That was from his first Jesus album, *Slow Train Coming* [CBS, 1979]. I was coming home from New York at 4:30 in the morning on a Sunday night, when there's nothing on the radio, and I was flipping the dial. The only thing on was this "Rock and Religion" show. They were playing contemporary songs that were blatantly gospel. This song came on and it blew my brain out. I had to pull the car over; I was in tears. I'm not a Christian, but I'm not adverse to it either. I dedicated the song to my daughter, Valerie Rose, and to all children who need to be believed in. People sometimes think that kids don't exist. That they're just these snotty little things running around. That's bullshit. Dylan's version is beautiful, but I tried to make mine with the crash cymbals and screaming guitars.

MR&M: "Mercy, Mercy, Mercy," did you choose that song?

PS: That I heard by Don Covay, but the Rolling Stones did it too.

MR&M: There's a vocalist on that track named Henry Reel. I understand that his real name is something else.

PS: What?

MR&M: Jackson Browne.

PS: As long as you understood that. Everybody has asked me why. I don't know why; that's one of his pseudonyms.

MR&M: "Gasoline Alley," the Rod Stewart song. Interesting version.

PS: Glad you noticed. I've always been a Rod Stewart fan. My favorite album in the universe, except for maybe a hundred others, is the *Truth* album by Jeff Beck, with Rod Stewart on vocals. I remember going to see him [Rod Stewart] at the Fillmore East; he'd do all these knee-drops and contortions. He was so fabulous.

"Gasoline Alley" is such a beautiful song, and I think anyone can relate to it. It's about missing your home and wanting to go back. The words to that song are poetry.

By the way, what's a weather suit? Let me ask Rod Stewart through this magazine: What's a weather suit? That's what he sings, I think. We translated it from the record. It sounded like he was singing, "I put on my weather suit." So Rod, what the hell was that?

The way we did the song, it was like I was saying I gotta go home or I'll go nuts! His way was more like I think I'll drop by the house.

MR&M: How did you hook up with the two producers, Greg Ladanyi and Richie Cannata?

PS: Greg Ladanyi is Henry Reel's producer, and Warren Zevon's. We were without a producer. For a long time it looked like Michael O'Martian was going to do it. Personal situations prevented him from doing it, and we were left hanging there. So the call went out that Phoebe needed a producer, and Greg said he'd do it. Richie I met through Elizabeth Joel, who manages Billy Joel.

MR&M: How would you compare working with them to working with Phil Ramone?

PS: Since Richie is a personal friend, it was easier working with him. That's not to say I didn't learn practically everything I know and have been able to retain from Phil Ramone, because he did four of my albums. He was very influential in my career. This was more laid-back, like hey, the guys are here. With Phil it was more like a "recording session." You were aware that it was a recording session.

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MR&M: This album is more rock-oriented than anything else you've done. Whose idea was it to take that direction?

PS: Mine, all mine. I know you'd find that hard to believe because I was bagged as a jazz singer, a ballad singer, a torch singer, MOR. I don't think I was that. I saw myself as being musically repressed. I've always wanted to shout. When I'd do "live" shows, I'd see the audience egging me on: "C'mon

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Phoebe." I would do it and people would say, "Shit, that ain't like her albums," or, "*Heck*, that ain't like her albums." This is all very different for me.

MR&M: How do you think your old fans will relate to it?

PS: So far, the majority have said thumbs up. Luckily, they're broad-minded enough. They're not just jazz fans; they're music fans. I think categories are for the birds. I might still do a jazz song some day, so what? You can't categorize music. What would you call the Police? It's Police music.

MR&M: The album has a very full sound, yet it's still very basic. What went into the recording to give it that feeling? Were there a lot of overdubs or was it kept simple?

PS: There weren't many overdubs. We rehearsed two weeks for the rhythm tracks and had everything ready to do basic tracks. We found that we needed very little overdubbing. All we needed was to mix it right.

MR&M: OK, back to the sessions. What were they like?

PS: Everyone kind of dragged their asses in. They all knew their stuff—everybody rehearsed. I did rough vocals in the beginning, but we ended up using about half of them. If we did a track and we didn't like it, we'd discuss what was wrong with it. It's a typical session. Everybody had a say in what was going on; it was a team effort.

MR&M: What was the producer's role in relation to your own? If there was a problem, who had the final say?

PS: Well, I would hope I did; and I really did, for the first time. If I was really anti-something, we'd stop it. There was one big fight between me and Richie, of all people. I had to be a real shit and sit there and say, "I'm *not* gonna do it" *[She makes a pouting expression.]* I wanted to do a heavy metal version of a song and they wanted it to sound like the Classics Four. They said, "Oh, yeah, what did you do, take acid before you came here?" They thought I was crazy. All of this was fairly experimental to some degree, because I haven't done this before.

MR&M: Do you plan to tour?

PS: Yes, we'll play mostly in clubs and small halls.

MR&M: Any idea where you'll play in New York?

PS: I'm not supposed to say. I'm supposed to be a good little gal. Now the feminists will hate me because I said gal. Is feminism still popular?

[Co-producer Greg Ladanyi spoke with Modern Recording & Music *about his involvement with* Rock Away.*]*

MR&M: First, we'd like some background on your experience as a producer. How did you become involved with producing records?

Greg Ladanyi: I was engineering for Jackson Browne, Warren Zevon, Karla Bonoff and a few other people. One thing led to another and I was able to develop a form of producing for artists that wanted to make a record without having a, quote, "producer."

MR&M: How would you describe your approach? How does it differ from that of other producers?

GL: I do all the engineering myself and give the artist the flexibility of being able to control the atmosphere in the studio, and getting the musicians to play the right things. By being the engineer as well as the producer, that eliminates one other person. I learned by working with producers as an engineer. I saw what they did and how they did it, and then applied it to my own theory of record making.

MR&M: What was the Power Station Recording Studio like? Was it your first time working in that studio?

GL: That was my first time working in New York City. I thought the studio was fine, the maintenance staff was great. Naturally, I'm more comfortable back home, but I didn't have a problem. They run the studio well and I would work there again.

MR&M: Who picked the musicians for Phoebe's album? There are L.A. studio musicians on most of the tracks.

GL: They were picked out by Richie Cannata and Phoebe. They had the musicians picked, but we made some changes when I got to New York, based on things we were hearing and different ways we thought we could go. But the basic core of the group was Liberty DeVitto on drums, David Brown on guitars, and Doug Stegmeyer on bass. I brought in David Landau (guitars), because he was the the only guy I personally knew. I'd never worked with the others before, and I knew I could talk to David on the phone and he'd tell me what was going on, and I could make decisions based on that. Craig Doerge and Billy Payne did some keyboard work when we came back to California.

MR&M: Did you and Richie work separately or at the same time?

GL: We worked together. Richie was instrumental in keeping charts together and communicating with me on what we thought we needed to have the band play. Richie would be able to tell these guys what to play because he is a player, and I'm not.

MR&M: Can you tell us what equipment was used in the making of *Rock Away*?

GL: The drums were Liberty's "live" drums; I can't recall the make of the kit. The guitars were Fender Strats and Gibson Les Pauls. David Brown used Music Man and Fender Reverb amps. The keyboards were mainly acoustic piano and organ, a Hammond B3.

MR&M: What kind of microphones were used?

GL: On the drums I used Telefunken 251 mics all the way around the kit. On the snare I used a (Shure) 56. Inside the snare I used an [Sony] ECM 50 and a [Sennheiser] 441 underneath. Based on who was playing the lead guitar, I'd have an old Neumann 54-2 guitar mic, and Shure 53s. On acoustic guitars I used ECM 50s, 452s or [Neumann] 67s, with ECM 50s inside the guitar. Vocal mics were all 67s. The bass was recorded direct and also with the 67 on the amp. The setup was with everything in one room, using leakage where possible. Controlled leakage is something I try to do to create a larger...a more "live" feeling. Then it doesn't sound as sterile as when everything is separate, in its own place.

MR&M: Anything else you'd like to add about working with Phoebe?

GL: Only that she made things very easy and that her tools as a vocalist are incredible. She's a great singer and she was great to work with.

SELECTED DISCOGRAPHY

Phoebe Snow	(1974)*	Shelter SRL 52017
Second Childhood	(1976)	Columbia PC 33952
It Looks Like Snow	(1976)	Columbia PC 34387
Never Letting Go	(1977)	Columbia 34875
Against The Grain	(1978)	Columbia JC 35456
Rock Away	(1981)	Mirage/Atlantic WTG 19297
The Best of Phoebe Snow	(1981)	Columbia FC 37091

** Re-released in 1976*

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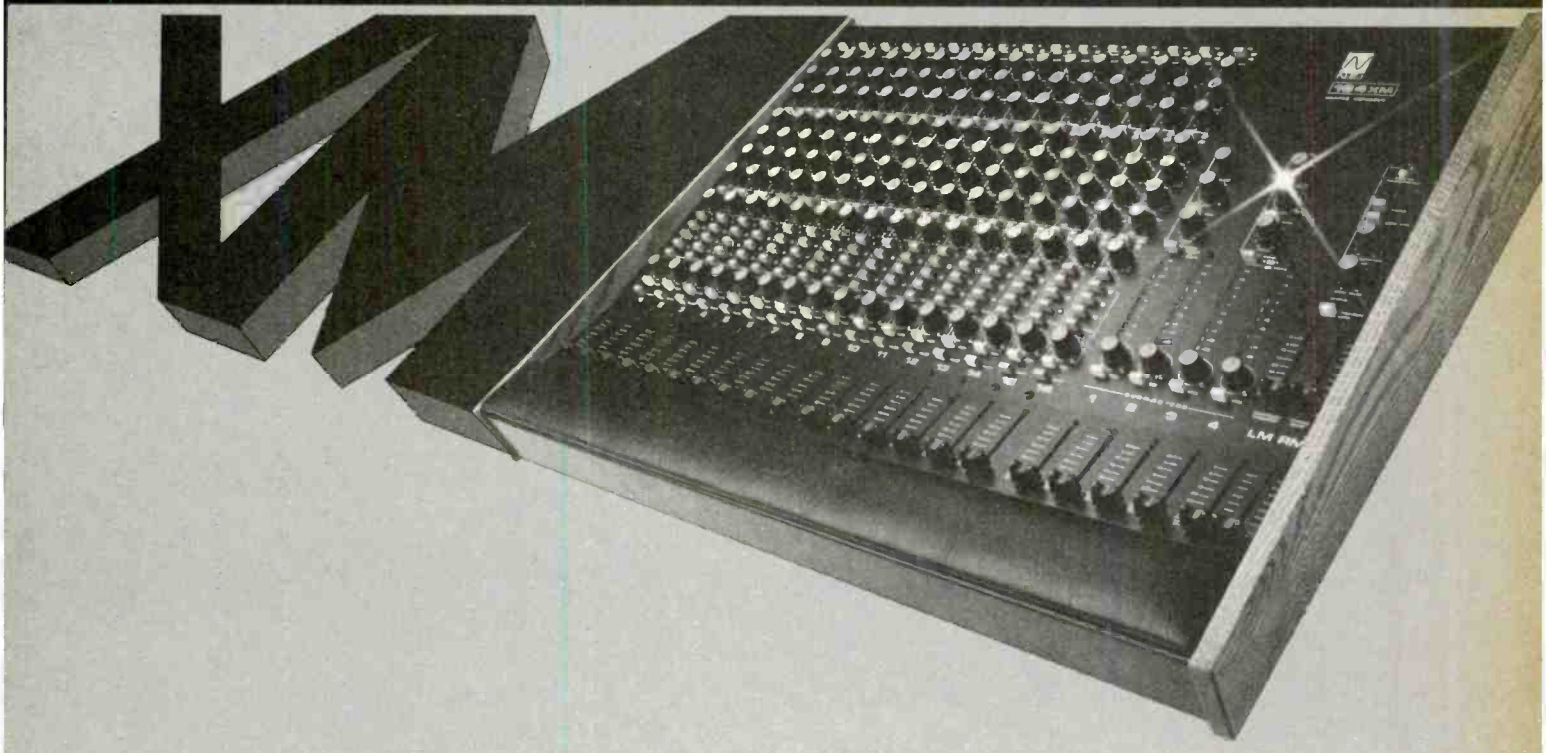

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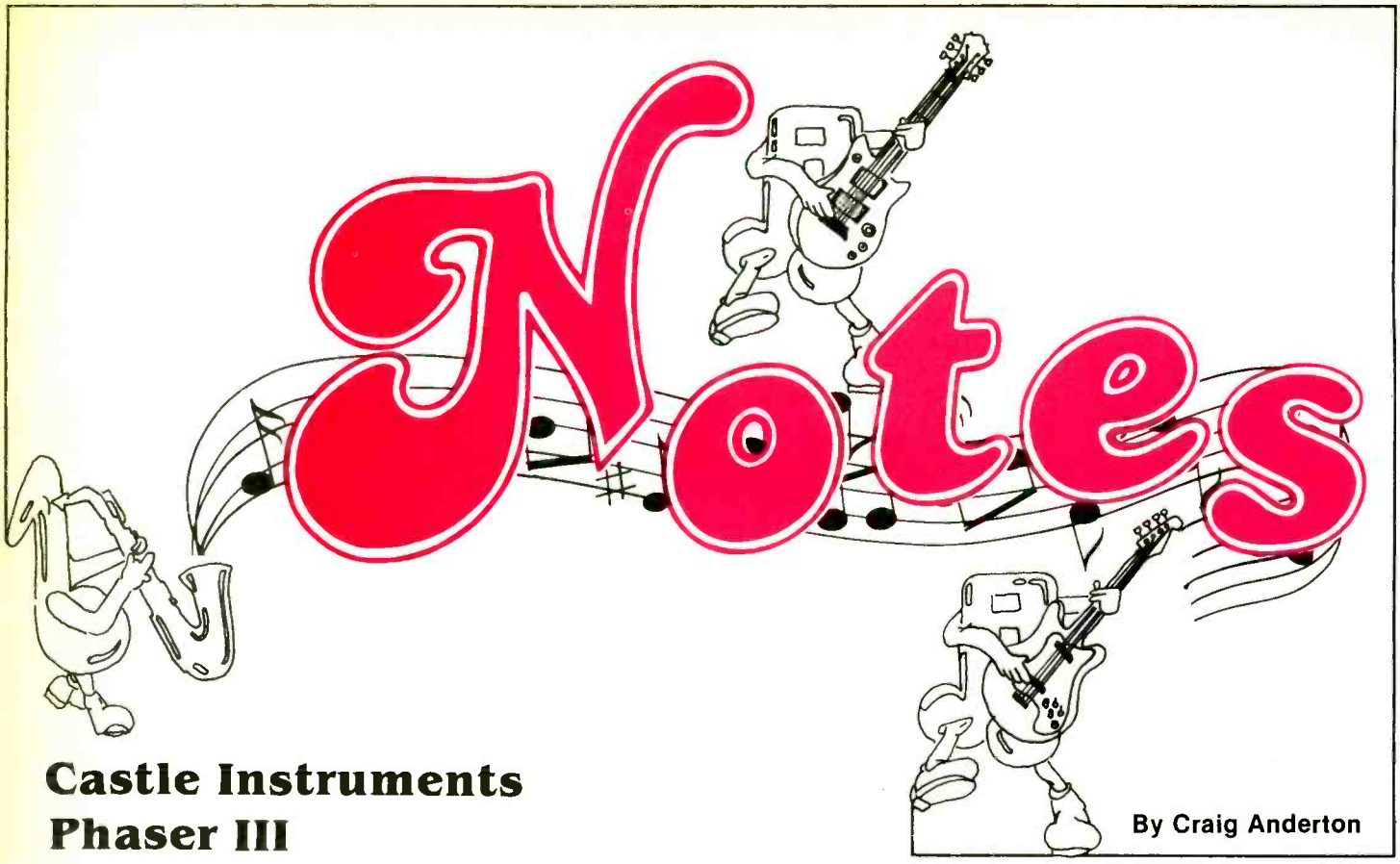

Castle Instruments Phaser III

By Craig Anderton

During the late 60s, one of the most popular recording studio effects was created by running two tape recorders very slightly out of sync to create a harmonically complex and animated sound. Although the effect had been accidentally discovered during the 50s, it took artists such as the Small Faces, Jimi Hendrix and the Beatles to popularize this sound. Unfortunately, "phasing" (as it is called in the U.S.A.) and "flanging" (as it was called in the U.K.) not only required the two tape recorders, but also a savvy engineer and some kind of variable speed setup (wide range variable speed controls were relatively uncommon in those days).

Musicians started looking for ways to simulate the phasing sound on stage, and it wasn't long before manufacturers started responding to the call. The first phase shifter I ever heard about was the Countryman unit, marketed during the early 70s, followed by the Maestro unit which became extremely popular for on-stage use. During the early and mid-70s, phase shifters were probably the most widely used (overused?) effect going, doing for musicians what the fuzz-tone did during the mid 60s and the wa-wa in the late 60s.

However, while phase shifters simulated the sound of tape phasing, they did not duplicate this sound because phase shifters could not create an actual time delay, which is the basis of true tape phasing. Then, during the mid-70s "bucket brigade" analog delay devices appeared. These allowed for low-cost time delay circuitry that gave (for all practical purposes) the same sound as the true tape phasing, and additionally could perform other feats—such as vibrato—which a tape recorder just could not do.

If anyone thought that analog delay meant the end of phase shifters, though, they were mistaken. Due to their low cost and relatively clean sound compared to some analog delay based flangers, phase shifters continue to be popular today. The sound they give is less intense than flanging, and adds a pleasing diffuseness to the sound of the instrument. Considering how many phasers there are on the market, I was somewhat surprised to see that Castle Instruments (2 Carteret Court, Madison, N.J. 07940) was specializing in producing phase shifters for both on-stage and studio use. Does the world need yet another phase shifter? You can decide for yourself after reading this review.

WHAT is IT? The Phaser III, which lists for $145, is a compact (about 4.5" x 7.5" x 2.25") unit designed for on-stage use. Its rugged metal case has a substantial feel to it; this is one effect that's not going to go sliding all over the floor when you use it. There are some other immediately visible touches that add to the feeling of ruggedness: the six knobs are held onto their shafts with allen head screws; all controls, switches and jacks use oversized, heavy-duty lockwashers; there's a sturdy line cord (2 conductor, though, rather than 3 conductor); and a high quality fuse post mounted on the back of the case. In addition to the standard instrument and amplifier jacks, Castle offers two options at extra cost: a jack for an external control voltage input ($10 extra) and a jack for a remote footswitch ($20 extra for jack and footswitch with 10 ft. cable). The former is designed to accept synthesizer-compatible 0 to +10 V control voltages, while the latter allows the musician to have the phaser, hence the controls, at arm's level and run a footswitch to the floor to control the effect in/out function.

There are also two LED indicators. One shows when the effect is in, while the other is a pilot light whose intensity varies at the same rate as the phase shifter's internal LFO (low frequency oscillator).

PHASER III CONTROLS: The six controls offer quite a bit of flexibility. These are:

Stages: Phase shifters create their distinctive effect by cascading a series of individual phase shift stages, since one phase shift stage by itself does not create a dramatic enough effect. Most budget phasers use four phase shift stages; this produces a relatively gentle, majestic type of effect that seems to work very well with string synthesizers and jazz-like rhythm guitar parts. Other phasers use six stages, which produces a sharper type of sound. Six stage shifters are often characterized by a more "vocal" type of sound, which resembles a wa-wa (or filtering) type of effect. Eight stage phasers produce the most intense sound, and in fact, this sound is closest to true tape phasing effects. Some phasers go up to 10 or 12 stages, but the vast majority of phasers for on-stage use have either 4, 6 or 8 stages.

The Phaser III allows you to select either 4, 6 or 8 stages of phase shifting, so in at least one respect this is a "universal" type of unit that can duplicate the sound of other phasers on the market. Far from being a frill, this control allows the musician to match the phaser's sound to the instrument and music being played. Since this control turns a three-position rotary switch, and therefore requires more torque than turning a pot, the Castle people have thoughtfully made this knob a little larger in diameter than the knobs that control pots.

Blend: This control determines the mix of phase shifted and dry, or unprocessed, sound. Phase shifted sound by itself produces a vibrato effect—it is only by also mixing in some dry signal that the phasing effect becomes apparent. Many of the less expensive phasers simply set the blend internally for equal amounts of phase shifted and dry signal; however, this prevents you from obtaining vibrato and other more esoteric effects. One very minor peculiarity of the blend control is that while the center position gives equal amounts of phase shifted and dry signal in the 4-stage phaser setting, the control must be turned slightly clockwise for equal blend in the 6-stage setting and slightly more clockwise (about the 1 o'clock position) in the 8-stage position.

Emphasis: This control performs the same function called "regeneration," "feedback" or "resonance" in other phaser models. Basically, it sets up a feedback loop from the output of the phaser back to the input, thereby accentuating the frequency response peaks and notches caused by the phasing process. The end result is a sharper, more intense sound.

The actual amount of phase shift varies in a cyclic manner, with a standard triangle wave LFO causing these variations. There are three controls associated with the LFO, namely:

Rate: This sets the overall LFO speed, from approximately one cycle every 12 seconds to what sounds like about 10 to 15 Hz. In other words, it goes from slow enough to fast enough.

Width: This determines how much effect the LFO will have on the phase shifter's sweep. With the width control counterclockwise, there is no modulation of the phase shift by the LFO. Turning the control further clockwise increases the amount of modulation, until at the full clockwise position there is a maximum amount of modulation.

Center: With the width control fully counterclockwise, this sets the resonant frequency of the phase shift stages. Sweeping this control clockwise produces the characteristic phase shifter "whoosh" that sweeps from low to high over a 5-octave range. With the width control cracked open, the center control sets the point around which the modulation will occur (up to a maximum of about 8 octaves with maximum width). Actually, this is much easier to grasp by simply listening to the phase shifter than trying to explain these controls in print. *Figure 1* summarizes the functions of these three LFO-associated controls.

The remaining front panel control is the footswitch. While it's located a bit too close to the controls for my taste (it's not easy to be precise with where you put your foot under dim stage lighting), in practice this won't cause too many problems. One very positive aspect of the switch is that it requires little pressure for actuation and doesn't produce any loud *KER-CHUNK* sounds when you switch in the shifter. Just a very light touch activates the effect, and the indicator

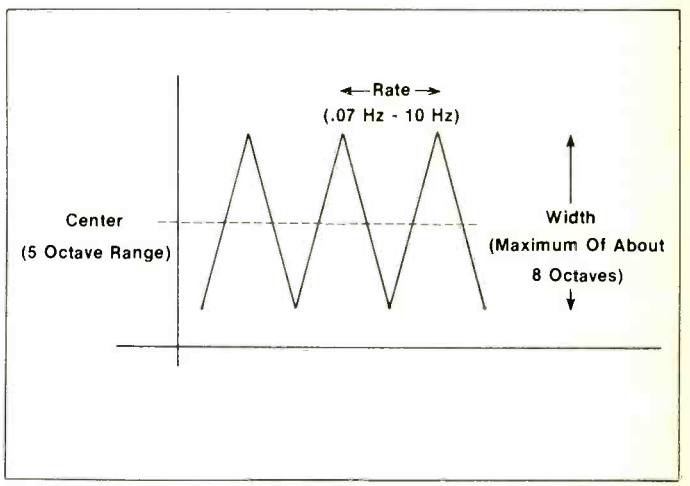

Fig. 1

"Musicians' failure to use boxes to their fullest potential has made manufacturers reluctant to come up with anything too different."

light will glow to show that you're ready to go. Switching the effect in and out produces no annoying pops or clicks in the sound.

INSIDE the BOX: Of course, I *had* to take the thing apart to see what was going on inside. The circuit board is high quality epoxy glass, and there are *lots* of ICs. Unlike phase shifters that use FETs or opto-isolators as voltage control elements to control the phase shift modulation, the Castle III uses 3080 IC transconductance amps. There are advantages and disadvantages to any of these three methods. FETs are inexpensive, but prone to distortion with high-level signals and subject to mismatching of their electrical characteristics. So, part of their inexpensiveness may be offset by the need for manufacturers to match the FETs for each individual phase shifter. Opto-isolators are my favorite way to go, but only when price is no object. In return for the expense, opto-isolators contribute virtually no noise or distortion, are well-matched and can handle extremely high signal levels.

3080s are sort of a compromise between these other two approaches. They're more consistent than FETs, but are noisier than opto-isolators. To circumvent this noise problem, the Phaser III incorporates an NE571 compander chip that acts as a noise reduction unit. The other chips are mostly Texas Instrument FET input op amps, and some other op amps associated with the LFO section.

The pots are sealed types that are definitely a step above the pots found on most effects; the transformer is extremely cute (if you open one of these boxes up, you'll see what I mean), and overall, I would rate the mechanical construction of the Phaser III as excellent.

PRE-FLIGHT for the PHASER III: The instructions I received with the Phaser III said to simply set all the controls at the half-way point, and experiment to see what effect rotating each control had on the sound. Frankly, I couldn't think of a better way to explain to a non-technically-minded musician how to make the thing work. As it so happens, with all controls at the mid-position you get a very pleasing phase shifter sound. If you get lost during your diddling, you can always return to that point of departure and start over. It's possible to get a very good intuitive feel for what controls do after only a very short period of familiarization.

EVALUATING the PHASER SOUND: The first thing you'll notice is that the unit is dead quiet. No hiss when you're not playing, no pops or noise...nothing. Then when you do start playing, the unit leaps into action. I checked out the S/N ratio; with a 2 V peak-to-peak (pp) output signal, the residual noise was 200 microVolts, yielding an honest S/N ratio of 80 dB. This means that for all practical purposes, the Phaser III adds no noise to your setup.

After playing with the unit for quite a while, though, I did notice a very subtle lack of brilliance. At first I thought that maybe I was so used to hearing a little hiss that I was being subjectively fooled, so I plugged into my test equipment and checked out the frequency response characteristics. The results showed my ears weren't lying; the higher the signal level you put into the Castle III, the poorer the high frequency response. For example, although a 1 V pp signal would reproduce with essentially flat response out to 20 kHz, a 5 V pp signal started rolling off at around 5 kHz and was down by 6 dB at 15 kHz. While some might argue that there is little energy up that high in the spectrum anyway, the fact remains that the harmonics of many instruments (especially synthesizers) do extend that high, and any loss in frequency response in the treble region does produce a "dulling" of the sound. While this small lack of ultra-high frequency response would be acceptable in the phase-shifted mode, unfortunately the same problem exists even in the bypassed mode. So, if you're putting a high-level synthesizer signal with a lot of harmonics through the Phaser III, don't be surprised if the sound loses some of its bite. On the other hand, with a standard guitar it's doubtful that you would notice any difference whatsoever. Nonetheless, this does make the point that bypass switches for effects should really bypass all the internal circuitry completely.

To make sure that the slight "dulling" effect wasn't due to impedance problems, I measured the input and output impedances. The input impedance (400 k) is certainly high enough to prevent any loading problems, while the output impedance (600 Ohms) is sufficiently low to drive any input with an impedance of about 6 k or greater.

I certainly don't mean to make a big deal of this slight treble response anomaly, especially when you consider the many advantages offered by the Phaser III. However, a high technology product that is so well constructed invites comparison not against "cheapo" phase shifters, but against perfection. In this respect, I would have hoped for a slightly crisper, more cutting sound. On the other hand, the Castle III is so exceptionally quiet that you can easily add some top end EQ after the unit without bringing up any hiss. Six of one, half dozen of the other...

I then settled in to listen to the effects of the various controls. In this respect, the Phaser III gets outstanding marks. All of the controls have a smooth feel, and cover what seems to be the ideal range. For example, although the rate control covers a very wide range, for some reason it was not at all difficult to zero in on just the right frequency. The other controls worked in a similarly "transparent" fashion. As a result, I never found myself fighting the controls; they always seemed to find the places I was looking for.

The variety of sound offered by the 4, 6 and 8-stage option was also most welcome. For example, when doubling an in-

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"Phase shifters continue to be popular despite the onset of analog delay."

strument that's being phased, you can use two different numbers of stages to help differentiate a bit between the two parts.

There are also some tricks you can do with the Phaser III that would be difficult to perform with other phasers. Take the control voltage input; any control voltage presented to this jack is mixed in along with the existing LFO control voltage. So, if you feed in an envelope control voltage, you'll have a combination of envelope and LFO phasing (like the Electro-Harmonix "Polyphase"). You could also run a patch from your synthesizer player's envelope generator output into this jack, so that phasing changes would occur in sync with the synthesizer; or, put a battery in a standard volume pedal (see *Figure 2*) for pedal controlled phasing. I'm afraid that many musicians will probably overlook the power of this control voltage input, but for those who know what they're doing this jack will be a welcome addition. I applaud the example of including control voltage jacks set by com-

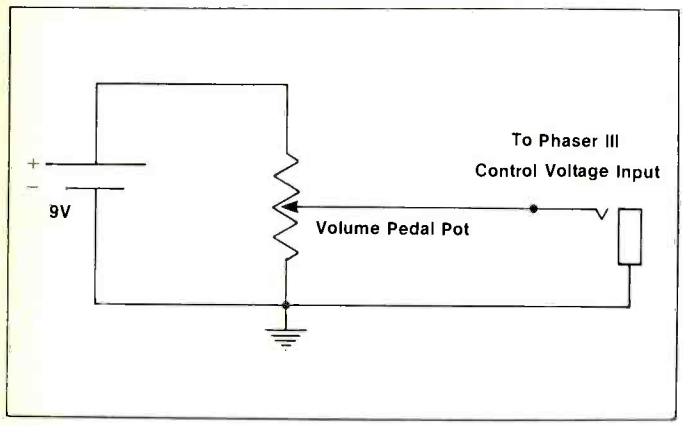


Fig. 2

panies such as A/DA and Castle; I hope more manufacturers get the message soon. If they start adding these jacks, I've got lots of potential articles on what to do with them! However, do remember that the Phaser III wants to see mostly low-level signals. So, don't try feeding the unit with the output from a +4 dB tape recorder or the high-level output from some synthesizers, or the Phaser III will not perform up to its potential.

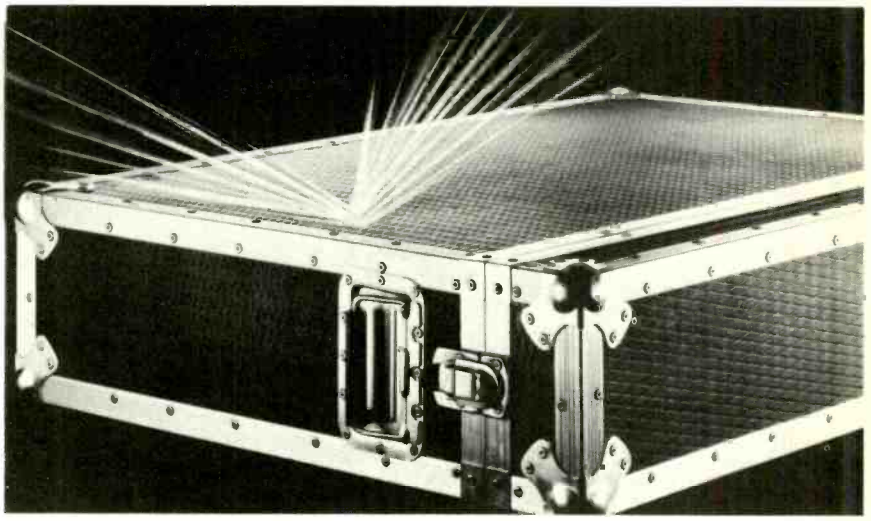
Another good effect is to crank the emphasis control way up, but set the blend for mostly straight signal. This puts the whistling, resonant type of sound in the background where it supplements your signal rather than overpowers it. (Hint for tinkerers: there's a trimpot inside labelled "emphasis" that can increase the emphasis to the point of oscillation, if desired.) Phasers without a blend control just don't give this option. Also, don't overlook using the Phaser III as a type of equalizer. Simply set the width control to zero (full counterclockwise), and vary the center and emphasis controls to provide a variety of equalization responses that would be difficult to obtain with a standard equalizer.

OVERALL EVALUATION: My main complaint about the Phaser III applies to 99% of the phasers on the market: I'm getting pretty tired of that same old triangle wave LFO control! It's the 80s, folks—how about at least adding envelope control? Or an attack/decay type of generator? I'm sure the only reason why phase shifters got stuck on LFOs was because tremoloes used LFOs, and they were the first "special effect" designed specifically for guitarists. However, at least the Phaser III gives you an external control voltage input so you can experiment with other control voltage sources, which is a step ahead of most manufacturers.

And, while the Phaser III is technically very well designed and mechanically strong, it is not a particularly creative unit. In other words, it seems content to produce a technically perfect effect rather than an imaginative one. (However, Castle also makes a more expensive rack mounted Dual Phaser for studio use that does include some novel features.) Again, this is a complaint that applies to much (probably most) of the effects industry. To be fair to manufacturers, it should also be pointed out that many musicians don't even make the minimal amount of effort required to use present day boxes to their fullest potential, and as a result manufacturers are reluctant to come up with anything too new and different for fear of rejection in the marketplace. Unfortunately, I'm afraid that fear may be well-justified.

My final complaint is, once more, something that applies to the effects industry in general. With this many controls, it's possible to obtain a very wide range of sounds, but how is the musician going to switch six different knobs in the middle of a song? With a one or two knob phaser, this isn't a problem. But a unit as versatile as the Phaser III would benefit greatly from having a couple of preset positions, or some other means of rapidly changing from one sound to the next. In the studio this is no problem, of course, but playing "live" is a different matter.

There...I feel much better now that I've gotten all that off my chest! But in all fairness to Castle, this isn't an ideal world and in that context, the Phaser III is an excellent phaser and excellent value for the money. The list price of $145, while low compared to other makers' list prices for a comparable product, is partially offset by the fact that the Phaser III is only available direct from Castle, and is therefore not likely to be found in your local music store at a discount price. If you've held off getting a phaser until you could find a rugged, quiet and well-behaved unit, the Phaser III would be a good place to start. Or, if you already use a phaser but want a unit with more versatility and AC power, the Phaser III is again an excellent choice. As far as the "inspiration factor" is concerned (namely, all technical considerations aside, does playing the unit get you off?) the Phaser III also does well. It's a quality product that doesn't get in the way of your music, and as long as you don't ask it to process high-level signals with a lot of high-frequency content, it will do the job quietly, efficiently and relatively inexpensively.



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Ambient Sound

BY LEN FELDMAN

Digital Audio Update

If you want to know what's been happening in the world of digital audio, the best place to find out is, as you might have guessed, in Japan. To be sure, in this country such firms as 3M and, on a much smaller scale, Soundstream and others continue their research and development programs, all of which inevitably lead to better digital recording and reproducing equipment, but most of this work is directed strictly to the professional recording market. In Japan, where large industrial electronics companies spend about four times as much out of every dollar (or yen) earned on research and development as we do in this country, work on digital technology is going on at a feverish pace. What's more, there seems to be as much activity in the consumer product field as in the professional recording field.

Recently, I was fortunate to be a member of a group of audio journalists who were invited to Japan by the giant Matsushita Electric Industrial Co. Ltd. On the slim chance that the name Matsushita may not be familiar to some readers, let me hasten to add that they are the "parent" company responsible for such more identifiable trade names as Panasonic, Technics, Ramsa, National and Quasar. While we visited several of the company's facilities in and around Osaka, Japan, we were exposed to a variety of products and discussions relating to digital audio. What follows is a brief summary of how this company sees the future of digital audio worldwide.

A Digital Audio Cassette Recorder

All of the consumer-type PCM recording products shown in the U.S. to date have been based upon video recorder tape transports. That is, the user was expected to own a VCR (either of the Beta or VHS format) to which would be added a PCM processor. What the Technics division of Matsushita has done is to combine the PCM processor and a VHS-type tape transport system into a single, fairly compact unit. Assigned the model number SV-P100, this entire unit weighs in at just over 44 lbs. and measures 16 $^{15}/_{16}$" wide, by 10$^{29}/_{32}$" high by 13$^{3}/_{8}$" deep. At first glance, the unit resembles the now-defunct Elcaset tape deck package of a couple of years ago, but of course there's no relationship whatever between the Elcaset and this sophisticated home-type PCM digital audio tape deck which simply uses video cassette tapes.

Operating standards used in the SV-P100 conform to the standards adopted by the EIAJ a couple of years ago. That is, 14-bit linear quantizing is applied using a standard NTSC video signal format to provide two hours of recording on a T-120 video cassette (VHS format). As has been stressed many times, frequency response is ruler-flat from 2 Hz to 20 kHz (+0, −2.5 dB), harmonic distortion is less than 0.01% (referenced to 0 dB record level) and wow-and-flutter is unmeasurable, with dynamic range well over 80 dB. Two recently developed LSI circuits largely control PCM digital signal processing in this unit. One of these, having a chip size of 6.08 x 5.58 mm, the LSI MN-6601, has the equivalent of about 10,000 transistors and is used to add the error detection code and correction code to the A/D (analog-to-digital) converted digital data and to convert this information into video-format data. The second new LSI (MN-6602) has the equivalent of about 15,000 conventional transistors, and, during playback, reconstructs the PCM digital signal, performs error correction and provides this as serial data to the D/A (digital-to-analog) converter. Together, these two LSIs provide the same processing capabilities as about 500 conventional logic ICs, thus allowing much more compact dimensions than previously possible.

The new audio-dedicated PCM "cassette recorder" has all of the convenience features you would expect of any high-priced audio recorder, but one additional feature which will be most important in the future world of digital audio recording is the pair of *digital* input and output terminals on the rear panel of the unit. These permit direct digital dubbing from one machine to another, and in such dubbing there is, theoretically, no degradation of recording quality, dynamic range or signal-to-noise ratios.

Of course, the question of price was in all of our minds

as we watched this elegant PCM deck being put through its paces. While we could not pin down a final price (the unit has not even been introduced at retail levels on the Japanese market as yet), a ballpark price of around $3000 was mentioned. If that seems pretty steep, consider the $3800 price of the Nakamichi 1000ZXL *analog* cassette deck which we tested for last month's issue of *Modern Recording & Music*!

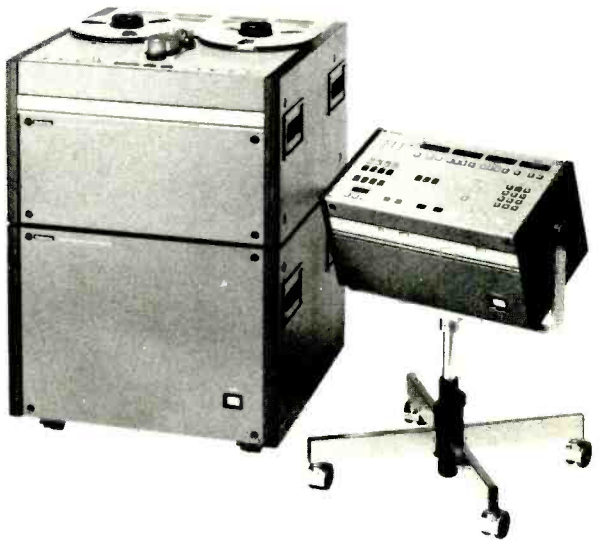
Which Way for the Digital Audio Disc?

Perhaps the most important revelation during our visit concerned the status of the future digital audio *disc*. As readers who have been following the progress of digital audio know, two distinct schools of thought have evolved concerning digital audio discs (which are fast becoming known as DADs, thanks to the Japanese penchant for abbreviating such things). There are those who maintain that a common player should be used for both video and digital audio discs. JVC's 10-inch diameter grooveless capacitance disc fits that need nicely, and, in fact, both VHD (Video High Density) and AHD (Audio High Density) digital discs have been successfully demonstrated on many occasions. The Philips/Magnavox/MCA Laser-Optical disc could also easily be used as a vehicle for storing stereo audio in digital form.

Then, there are those who feel that audio should have its own, dedicated audio disc (one not "tied" to any video system standards such as NTSC, PAL or SECAM), Philips has proposed such a disc, dubbed by some the Compact DAD. And more recently, Sony has devoted its research to improving that disc on a cooperative basis with Philips. The real surprise came when we were told that Matsushita, too, would support the Compact DAD. To appreciate why this constituted surprising news, you have to realize that JVC and Matsushita are, in a way, "sister" companies from a stock ownership point of view, and for Matsushita to have come out in favor of a disc developed by its competitors rather than by its "related" company suggests that they feel strongly that the compact DAD has the best chance of winning the digital disc sweepstakes. Indeed, there are good reasons to believe that they are correct. The Philips-Sony disc measures only 12 centimeters in diameter, is played by a laser pickup (with no physical contact between disc surface and pickup) and can be handled by a player tiny enough to fit under the dashboard of even a compact car! We were, in fact, reluctantly shown a prototype of such a digital disc player, and having seen it, were personally convinced that the future of audio recording lies with the compact digital audio disc.

Meanwhile, In the Pro Equipment Area...

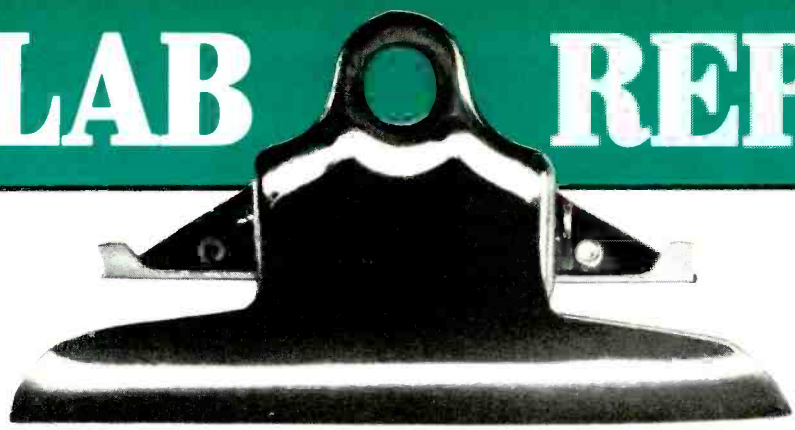
While work proceeds in the field of consumer digital audio, the people at Matsushita Electric, like their competitors, are well aware of the fact that the transition to digital audio on a professional level (in recording studios, etc.) is not a thing of the future but is happening *now*. Accordingly, they have been hard at work developing stationary-head digital recording equipment as well as equipment based upon rotating video head technology. A thin film type of magnetic head with high density recording and playback capability has been developed for use in PCM recording. This head is for use with ¼-inch tape on which there are sixteen digital tracks and four auxiliary tracks. The recorder operates at the same speed established for analog audio recording: 15 ips. The editor is designed to operate between two of the digital audio recorders. Editing operations can be performed with the same ease as is provided by conventional analog recorders which are indexed by cueing signals. Editing points are easily located using a search dial and signals can be smoothly joined. In addition to the stationary head digital recorder and the



Prototype digital audio recorder by Technics employs stationary head, ¼-inch tape, 10½-inch reels and "standard" 15 ips tape speed.

editor, Matsushita has also developed a digital audio mixer which can mix the digital signals of from four to eight channels, edit them into a two channel signal and then divide this signal into four channels. This unit is able to pan each channel and independently adjust each channel's level. It is also possible to mix the outputs into a two-channel output and to simultaneously select and echo forwarding output. If a digital reverberation unit is connected to the mixer, this output can be used for adding reverberation to the mixed, edited signal.

Since my most recent visit was limited to a tour of Technics facilities, this overview has concentrated on digital audio progress by that company. Knowing how this industry works, you can be certain that other large Japanese electronic companies are just as active in developing digital audio products and that before very long, the studio that lacks digital capability will be one that has not kept up with these rapidly changing technologies; and will very likely suffer for it.



NORMAN EISENBERG AND LEN FELDMAN

Revox PR99 Tape Recorder

General Description: The Revox PR99 is an open-reel half-track tape recorder available in either of two speed configurations: 3¾ and 7½ ips; or 7½ and 15 ips. The latter was chosen for this report. Reel size capacity is up to 10½ inches with the standard NAB hub adapters. The transport uses three motors and direct drive. Logic control enables going from one operating mode to the other directly; the record button, however, is active only if the play button is pressed at the same time. It is possible to go into record from either of the fast-wind modes. A sync feature permits copying from one track to another. It is also possible to copy one channel to another while mixing in a new recording from another source with or without sync (either channel of the PR99 can be switched for reproduction from the recording head while the other channel serves as the "clock" track). Echo effects also may be introduced to a new recording or added to an existing recording.

The PR99 has selectable dump-editing whereby the take-up reel is disabled so that unwanted sections of tape may be dumped into a wastebasket. For editing in general, a "cutter slide" may be engaged which presses the tape against the heads so that manually "rocking" the reels will locate a desired passage.

Centered between the two tape reelers is a four-digit tape counter and reset button. The tape enters and leaves the head assembly via swinging guide pins. In addition to the cutter slide, which is found on the head mounting block, there are controls to the left and right of the block. Those on the left are the channel select buttons for the sync-playback option, while the right-hand controls include the tape-dump button and the reel-size selector.

The busy-looking panel below the transport area is logically divided into output and input controls and features. The AC power switch is found in the "output" group. Output level through headphones may be adjusted by the dual-concentric output level

controls. If the associated "UNCAL" button is released, the calibrated line output level is fed to the line output jacks. If the UNCAL button is engaged, the line output level as well as the headphone output level will be controlled by the output level knobs.

The input panel includes level controls, input selectors, channel-ready or preselect switches, microphone inputs (unbalanced) and recording indicator lamps—all duplicated for each of the deck's two channels. In addition, there is one button for uncalibrating the line input so that input sensitivity may be manually adjusted via the input level controls when desired.

The right-hand portion of this area contains the deck's twin meters (VU with peak-level LED centered under the −20 to +3 dB scale), and the transport buttons which include pause, rewind, fast-forward, play, stop and record.

Input and output connectors, at the top of the deck near the rear, all are balanced, and include connectors for line and microphone. Also found here are the optional remote-control connectors including fader start and variable capstan speed.

The rear of the deck contains the AC power cord connector, an operating voltage selector and fuse-holder. The front plate is 19 inches wide and four slots along either end permit standard rack mounting. The PR99 may be used in any attitude between horizontal and vertical.

Facilities for calibrating the PR99, using either NAB or CCIR test tapes, are found on the underside of the deck (when it is removed from its housing). The procedures for input circuit calibration, meter adjustment, azimuth adjustment of the recording head, playback level adjustment, frequency response check, bias adjustment, recording equalization, sync adjustment and so on are all detailed in the owner's manual.

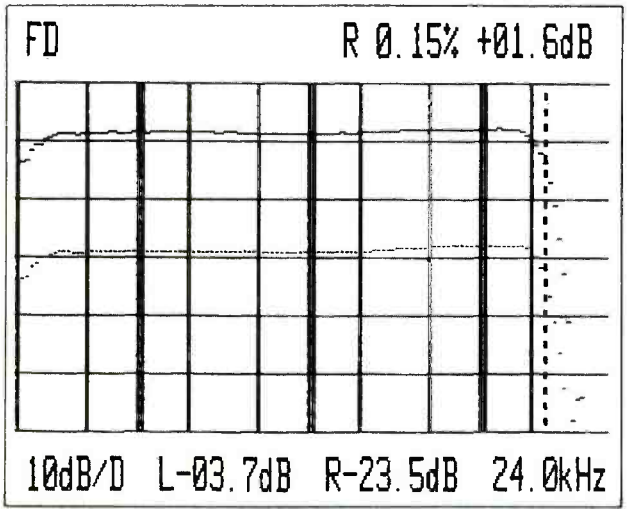


Fig. 1: Revox PR99: Frequency response at 0 dB and −20 dB record level.

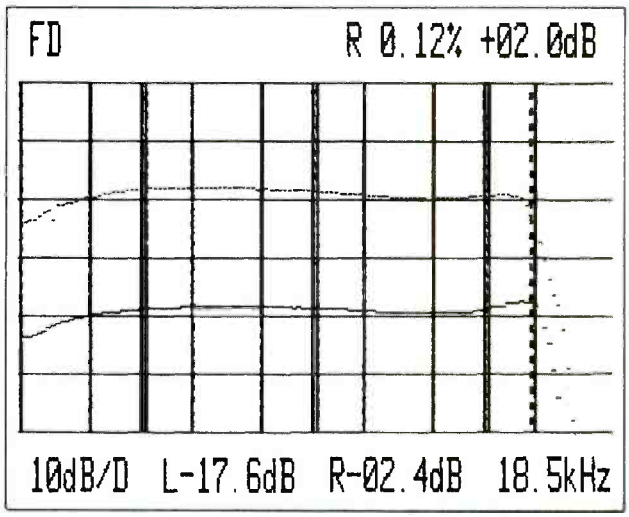


Fig. 2: Revox PR99: Frequency response at 0 dB and −20 dB record level, 7½ ips.

Test Results: Published specifications for the Revox PR99 were readily confirmed or exceeded in *MR&M*'s lab tests, with the measured distortion figures coming in exceptionally low and well below levels specified. *Figures 1* and *2* show the overall record/play frequency response at 0 dB record level (corresponding, in this case, to a magnetization level of 250 nWb/m); and at −20 dB record level. Tests were run at both 15 and 7½ ips speeds. Prior to making the measurements, we carefully adjusted bias and EQ via the control holes provided for that purpose. The machine, as supplied to us, had been calibrated for Scotch 250 tape, but we wanted to see how well it could be adapted for an alternate tape, and so we conducted all our tests using Maxell UD-XL (50-120B) tape in 2500-foot lengths on 10½-inch reels.

Having adjusted the 0 dB reference on our test equipment to correspond to the 0 dBU level on the PR99, we next measured the available headroom for 3 percent third-order distortion. We found this to be a very substantial +11 dB. And, as can be seen in *Figures 3* and *4*, at the +6 dB record level, the 3rd-order distortion measured only 0.27 percent at 15 ips, and 0.20 percent at 7.5 ips.

Playback-only response (measured by using a calibrated tape supplied by Standard Test Laboratory, Inc.) is plotted in *Fig. 5* for the 15-ips speed, and in *Fig. 6* for the slower speed of 7.5 ips.

Channel separation is shown in *Fig. 7*. As the dotted line cursor indicates, it measured better than 50 dB, even at the difficult frequency of 20 kHz. Interpolating

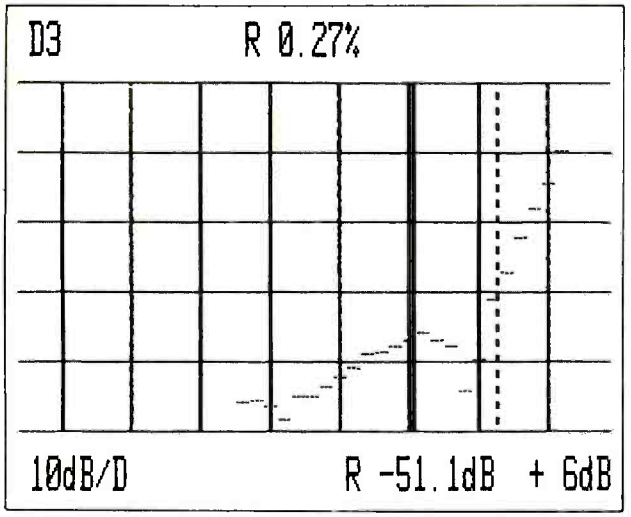


Fig. 3: Revox PR99: Third-order distortion versus record level, 15 ips.

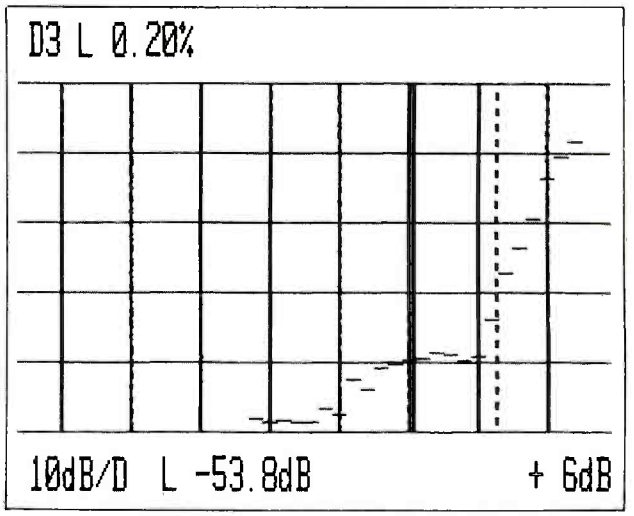


Fig. 4: Revox PR99: Third-order distortion versus record level, 7½ ips.

back to 1 kHz, channel separation measured about 64 dB (the vertical scale in *Fig. 7* is 10 dB per division, as indicated at the lower left of the diagram).

General Info: Dimensions are 19 inches wide; 15.7 inches high; 8 inches deep. Weight is 40 lbs., 12 oz. Price $2095.

Individual Comment by L.F.: I must confess that I was never completely clear on the difference between the Studer product line and the Revox line. Ostensibly, the latter is the trade name used by Studer-Revox for its "consumer"-type tape recorders and other audio products while the Studer name has been reserved for "strictly pro" products. Well, the PR99 seems to contradict all that. It is, in every respect, a professional tape deck. Although based on the basic frame, transport and electronics of the famed B77 deck, the new PR99 has truly professional features, such as balanced line in and out, XLR connectors, two-way self-sync and practical editing facilities.

According to Barry Evans, National Sales Manager of the Revox division here in the U.S. (the company has its headquarters in Nashville, Tennessee), the PR99 is being introduced initially for the broadcast market. Eventually, there will also be a cabinet version and a console version for the professional and semi-professional recordist market.

Besides meeting or bettering all of its specs, the PR99 proved to be a joy to work with. I was pleased with the ease with which a recordist can calibrate or recalibrate this machine. Not only does the owner's manual provide clear, specific instructions for these procedures, but the procedures themselves are so easy to execute and so positive in their effect that you *know* you have properly aligned and calibrated the machine before you actually begin recording.

Operation of the PR99 is as smooth as anything I have ever worked with. The three-motor transport system is similar to that of the B77 series with which many readers already are familiar. Additional features I appreciated include the dump-edit option. The Cal/Uncal buttons for both input and playback levels afford a measure of flexibility not found on consumer machines, and will be appreciated when the deck ultimately finds its way into small recording studios and is driven from a console rather than directly. Other worthwhile innovations are the multi-position input selector with its low and high mic level positions, its channel 1-to-2 and channel 2-to-1 re-recording capability and its "echo" setting which allows the recorded signal to be played back from the reproduce head to the record channel with no external patching required.

Novel features aside, however, the Revox PR99 has all of the basic quality and long-term reliability which have become synonomous with Studer Revox products. And, frankly, I would admire this product

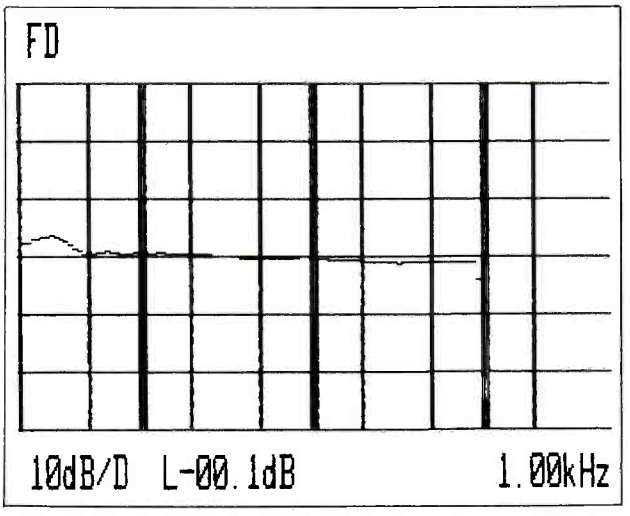


Fig. 5: Revox PR99: Playback-only response, 15 ips.

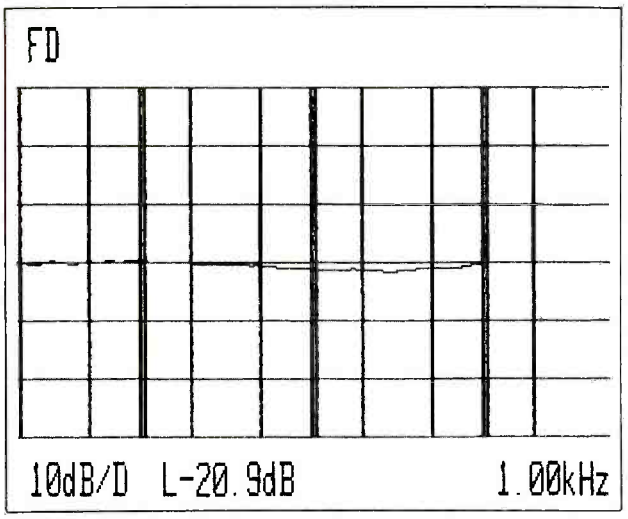


Fig. 6: Revox PR99: Playback-only response, 7½ ips.

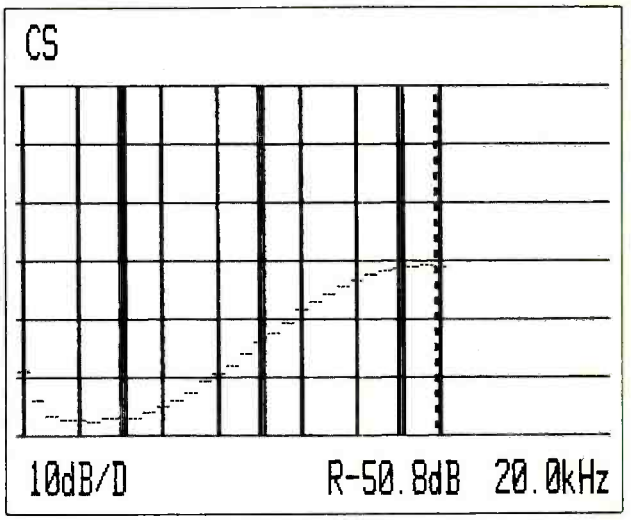


Fig. 7: Revox PR99: Channel separation as function of frequency.

whether its makers elect to call it a Studer, a Revox or any other name.

Individual Comment by N.E.: A couple or few issues back, I recall commenting on the way in which cassette recorders were going off into computerized automation that made them costly, possibly fun to own and really did nothing basic to overall performance in strictly audio terms. At the same time, I noted a few recent open-reel decks that did not "stand up and beg" at the press of a button but which did offer basic performance plus the kind of practical work options sought after by the serious recordist and perhaps never even missed by the cassette operator.

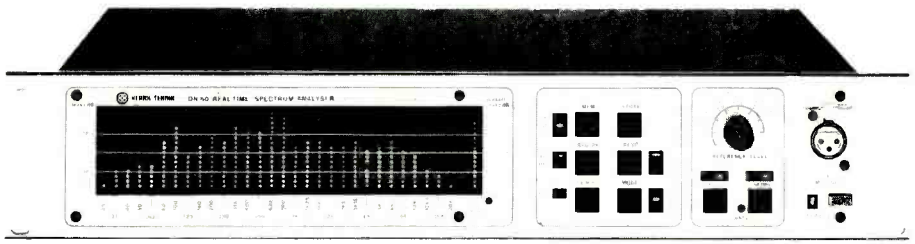
In a way, the new PR99 continues the latter trend—and then some. Apparently, what the Studer people have done here is to take the earlier B77 and give it a long hard look, asking themselves what could be done to make this highly regarded tape machine more adaptable or suitable for professional work. It already had earned a justly famous reputation among serious audiophiles for its high performance and reliability. With basic audio requirements well in hand, it remained only to enhance its "personality" in terms of inputs, outputs, editing and so on to qualify the machine for serious pro or semi-pro applications.

And that, apparently, is what has been done. In a sense, the PR99 is a very nearly an all-things-to-all-recordists working in the half-track format; it could find applications in the studio as well as at the hands of the nonprofessional but demanding general user. The fact is, that in addition to being a beautifully performing deck, it also is surprisingly easy to operate, and even the maintenance and calibration procedures should pose no rigors or problems for anyone with some acquaintance with tape recorders and the rudiments of electronics.

REVOX PR99 TAPE RECORDER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Frequency response, 15 ips	+2, −3 dB, 30 Hz to 22 kHz	+2, −3 dB, 23 Hz to 24 kHz
7½ ips	+2, −3 dB, 30 Hz to 20 kHz	+2, −3 dB, 27 Hz to 20 kHz
Wow & flutter: 15 ips; 7½ ips (WRMS)	0.06%; 0.08%	0.025%; 0.062%
THD at 0 VU, 15 ips	0.6%	0.15% (0.12% at 7½ ips)
THD at +6 VU, 15 ips	1.0%	0.27% (0.20% at 7½ ips)
Record level for		
3% 3rd-order HD, 15 ips	NA	+11 dB
7½ ips	NA	+11dB
Best S/N ratio (std tape)	66 dB	69.7 dB (66 dB at 15 ips re: 3% HD)
Fast-wind time, 2500-ft. tape	120 seconds	110 seconds
Mic input sensitivity	−32 dBU	Confirmed (ref 0 dBU = 0.775 V)
Line input sensitivity	+4 dBU	358 mV (in adjustable position)
Line output level	+4 dBU	Confirmed
Headphone output level	5.6 V	Confirmed
Erase ratio	75 dB	Confirmed
Speed accuracy	±0.2%	+0.08% (at 7½ ips)

Klark-Teknik
Model DN60
Real Time Analyzer



General Description: Klark-Teknik's model DN60 is a real-time audio spectrum analyzer with built-in microprocessor control, three memories, pink-noise generator and many other work-practical features. Spectrum analysis is accomplished and displayed in third-octave bands, a feature, which combined with its other attributes, makes the DN60 a professional-grade instrument for critical on-site acoustic measurements, music analysis, tape recorder EQ and alignment, noise surveys and test-bench voltage measurements.

The largest portion of the front panel is given over to the display which consists of thirty vertical rows of LED indicators (sixteen LEDS per column) that mark the one-third segments across ten octaves from 25 Hz to 20 kHz. There also is an added LED column for SPL measurements. All frequency centers are clearly marked, as are the vertical dB scales at either end of the display.

To the right of the display area is a push-button control panel. Here are found the switches that may be used to enter up to three "memories" of the display. The memory bank available for storing, or the one being displayed, is indicated by one of three associated LEDs. Pressing the "Store" button enters data into the memory and automatically clears any information previously entered into that particular memory.

Another button in this group selects either average or peak indications on the display, and this is indicated by a pair of LEDs. A "response" button selects the speed of the display response in terms of fast, medium or slow, and again the speed selected is shown on one of three LEDs. There also are buttons for peak hold and for mode. The mode switch, also with its associated LED indicators, is used for operating the DN60 in continuous or real-time mode, or to display memory-stored response, or—with an optional reverberation measurement accessory (RT60)—in an "external" mode. Without the RT60, the "external" position of this switch remains inoperative.

The adjacent panel contains a stepped reference level control which sets the absolute level to which the display readings are relative. That is to say, it calibrates the vertical dB scales for the display. Markings on this control include both dBm (for line input) and dB/SPL (for microphone input). Associated with this control is a button that selects the resolution of the display for either 1 dB or 2 dB per vertical graduation. LEDs above this switch show which resolution has been selected. To the right of this switch is the input selector (line or microphone) with LEDs above it to show which has been selected. Just to the left of this switch is another LED that shows, when it is lit, that the display response is weighted.

To the right of this group are a microphone input socket (XLR type) and the device's AC power off/on switch with LED power-on indicator.

The panel itself is of standard rack width, slotted at either end.

The rear of the DN60 contains XLR connectors for line input and for output of the device's noise generator. ***Note:*** The front panel mic input is balanced; the rear panel line input may be used balanced or unbalanced; the pink-noise output is unbalanced. The AC power cord, fitted with a three-prong (grounding) plug, must be fitted to the AC socket on the chassis rather than be permanently attached there. Just above this socket is a line fuse holder. Also on the rear is a removable plate behind which is the access for connecting the optional RT60 reverb accessory. Our test sample was supplied with an AKG C451E microphone.

Test Results: Our "Vital Statistics" chart for this unit consists purely of the manufacturer's own specifications of the device's functions—all of which were con-

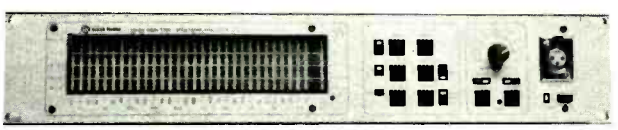


Fig. 1: Klark-Teknik DN60: Overall view of the unit (*see report*).

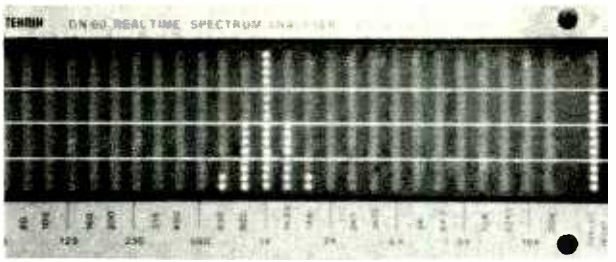


Fig. 2: Klark-Teknik DN60: Appearance of display with 1 kHz signal applied to DN60.

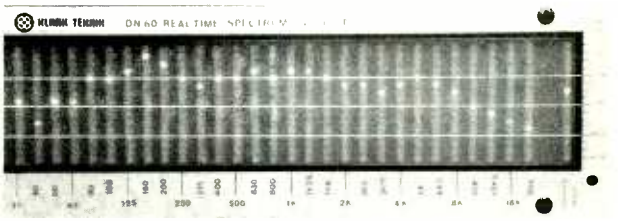


Fig. 3: Klark-Teknik DN60: Peak hold display of "Memorized" curve when recalled.

firmed. As a real-time analyzer, the DN60 is a remarkable instrument. The output of the unit's built-in noise generator (described as pseudo-random pink noise) was coupled to the input of the analyzer directly to obtain the display shown in *Fig. 1.* As can be seen, all of the LED columns are identical in height (the sensitivity was set at maximum, or 1 dB per LED), indicating the accuracy and purity of the internal pink-noise source.

A close-up of the display section of the DN60 is shown in *Fig. 2.* For this picture, we applied a 1-kHz signal to the device's line input. Note that in addition to the central display of that 1-kHz signal there also is a column of LEDs at the far right. This column indicates the overall sound-pressure level (SPL) of sounds reaching the analyzer. In the case of complex music signals, the overall SPL will almost always be between 6 and 15 dB higher than any of the 1/3-octave levels. For this reason, the SPL LEDs are deliberately offset by a fixed 10 dB as compared with the individual 1/3-octave LED columns.

The peak-hold facility in the DN60 allows you not only to hold the peaks of signals as they are analyzed, but also to display a memory-retained peak while observing a new and different real-time signal analysis—at the same time. *Fig. 3* shows just such a peak-hold display recalled from memory where we had stored it earlier while doing an analysis of a particular listening position in a room.

Included with the device is a pad of spectrum-analysis copy paper. The sheet's size enables it to be fitted right over the display on the device, so that the user can quickly and accurately transcribe the display onto paper, thereby preserving it (as well as preserving the required reciprocal or converse EQ settings of an assocated ⅓-octave graphic equalizer) for later use.

General Info: Dimensions are 19 inches wide; 3½ inches high; 10 inches deep. Weight is 6.025 pounds (5 kg). Price $3195.

Individual Comment by N.E.: From its XLR connectors to its precise information readout, not to mention its versatility and adaptability, the DN60 is a thoroughly professional instrument. Our tests of it consisted of putting it through its paces and noting that it performed in all respects exactly as claimed. It should be pointed out of course that the microphone for use with this device is not normally supplied; however the AKG C451E that showed up (much to our joy) in the carton that contained our test sample seems very aptly suited for use with the DN60, especially inasmuch as the phantom powering for this mic is supplied directly from the DN60. If phantom powering is not required (it appears on pins 2 and 3 or the mic XLR connector), it can be disconnected by a simple internal wiring change, described in the owner's manual.

I can't refrain from a comment on cosmetics. While so much consumer and "home audiophile" gear of late has taken on a "macho studio" look, here is a really professional instrument in which some effort obviously was made to make it look "domestically attractive." There's that two-tone orange display screen, with the shiny red LEDs (including the appearance of "OK" on the screen when turning on the unit); the additional red LED indicators that dot the panel; the dark brown button switches; the red and blue lettering around the reference-level knob and the blue front of the knob itself. And that bright green mic cable! Who said that business-like equipment can't look good?

Individual Comment by L.F.: If you read last month's issue of *MR&M* you will recall that N.E. and I waxed enthusiastic about a computerized equalizer/analyzer developed by dbx for consumer use. Indeed, that instrument continues to amaze us long after we completed our tests and report. Yet at the same time we recognize the fact that while many professional sound people and semi-pros will buy that unit, it is after all intended basically as a home-use device—one to be used with a home stereo system. Its bands of equalization and its corresponding LED display columns numbered ten, or one per octave. For professional applications, sound-reinforcement EQ, noise surveys, music analysis, and on-site acoustic measurements, it is generally agreed that the 1/3-octave real-time spectrum analyzer is preferred.

The remarkable DN60, of course, does offer display information in those desirable one-third octave spectral increments. It also serves for SPL measurements. It also generates its own pink noise. It also supplies phantom powering for microphones requiring it. It has a three-memory capacity, it incorporates the "A"-weighting network, selectable display resolution and the versatile peak-hold feature.

I should point out, too, that among the options

to be made available with the DN60 are the RT60 (reverberating time measurement) package; an X/Y plotter and oscilloscope interface; and a dot-matrix printer interface. To be sure, adding all of these options may push the total cost of the system beyond what is affordable by smaller sound installation companies (the avowed market at which the DN60 is aimed), but considering the performance, accuracy and versatility of this product, plus its microprocessor technology, I feel that—with or without options—the unit is well worth its asking price.

A final point: There was very little that we could actually measure regarding this fine instrument. After all, how can you measure the measurer?

KLARK-TEKNIK DN60 REAL TIME SPECTRUM ANALYZER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
	Analyzer Section	
Functions:	Selectable peak hold and real time can be displayed together. Peak/average · processor controlled averaging. 3 separate memories. 3 response times, processor controlled exponential attack and decay times. Selectable "A" weighting filter.	All Confirmed
Filters:	1/3rd octave filters on 30 I.S.O. bands from 25 Hz to 20 kHz. Double section pole-pair filters meet requirements of I.E.C. 225. Relative flatness: · ± 0.5 dB. Rectifiers: · Quasi peak fast response (attack time frequency dependent).	
	Overall Level Section	
All functions and performances as analyzer section with exception of:	frequency response:—15 Hz to 22 KHz (—3 dB points) peak/average uses separate rectifiers.	
	Input Section	
Microphone:	Differential input with adjustable sensitivity from .25mV/ubar to 1mV/ubar. To suit most capacitor microphones. Phantom powering provided: 14 V.D.C. via 2 K ohm resistors to suit AKG 451 series microphones.	
Line:	Via XLR connector on rear panel. Differential input for balanced or unbalanced lines. Input impedance—47 K ohms (nominal unbalanced connection).	All Confirmed
Attenuator:	Reference level switchable in 10 dB steps from + 20 dBm to − 50 dBm or 120 spl to 50 spl. Accuracy = ± 0.2 dB.	
"A" Weighting:	Selectable network meets I.E.C. 651 type 1 requirements: Operates on both mic and line inputs.	
	Display	
	31 x 16 L.E.D. matrix display. Selectable resolution of 1 or 2 dB per L.E.D. Variable brightness control. Multiplexed (scanned) operation.	

(continued)

Vital Statistics (cont.)

Pink Noise

Digital pseudo random white noise generator and pink noise filter.
Flatness ± 1.5 dB (20 Hz - 20 kHz) measured RMS with 10 seconds integration time.
Output level: + 4 dBm RMS (nominal) via XLR connector on rear panel.
Gateable output amplifier. Output impedance 1K ohm.

Other Features

Fast logarithmic A/D converter covers 31 dB in 1 dB steps. Accuracy - ± 0.2 dB (any level from reference level).
Memory stores entire 31 dB and all relevant switch information: Display can then be "expanded" when in memory recall mode.
All external accessories display entire 31 dB range (in 1 dB steps) and all switch information.
Intelligent talk and receive output available to suit automatic testing and control equipment.

All Confirmed

Connectors:

Mic input—XLR D3F Style
Line input—XLR D3F Style
Noise output—XLR D3M Style
Power—3 pin C.E.E.

CIRCLE 2 ON READER SERVICE CARD

CM Labs Model CM920 Power Amplifier

General Description: The model CM920, from the CM Labs Division of Audio International, Inc., is a heavy-duty professional-grade stereo power amplifier conservatively rated for 250 watts per channel into 8-ohm loads, 400 watts per channel into 4-ohm loads. Of standard 19-inch rack width, the amp's front panel has the requisite mounting slots at either end plus handles.

Front panel features include the power off/on switch associated with two pairs of red and green LEDs (one pair per channel) that show standby and operating conditions, respectively. Centered on the panel are channel input level controls of unusual physical design. The adjustments for level are made by fairly large slotted screws which then may be locked by tightening their surrounding hex-nuts. On arrival, both level adjustments were pre-set for maximum gain, and it required heavy-duty pliers to loosen their settings. Below each level control is an output fuse. At the right of the panel is an output level display consisting of ten LEDs per channel, calibrated from −45 dB to +5 dB.

Signal inputs at the rear are quarter-inch phone

jacks. Speaker connectors are five-way color-coded binding posts. The AC power cord is fitted with a three-prong (grounding) plug.

Along each side of the amplifier is a large heat-sink arrangement, with the output transistors nestled between its sections. The output devices are protected by heavy plastic covers.

The CM920 employs fully complementary output circuitry, and internal provision is made for varying the operating power requirements for use anywhere in the world.

Test Results: In our bench tests, the CM920 easily met or exceeded its power and distortion specifications. We did need about a half volt more of input signal than claimed to get rated output, but this is no great problem with today's typical line-level sources. The amp did use up more operating AC power than specified, idling at 120 watts and reaching a maximum of over 1000 watts. Trying to measure the amplifier's damping factor (a figure of 500 is claimed by CM Labs) proved to be an exercise in futility, since that figure implies an internal or "looking back" impedance of only 0.016 ohms, and even a couple of feet of heavy-gauge speaker wire has more resistance than that. Since we haven't figured out a way of connecting an amplifier under test to our test instruments without using speaker cable, we'll have to take CM Labs' word for that.

We did verify, in any event, that the CM920 is very stable under heavily reactive loads, which is an important consideration for an amp that is expected to interface easily with all manner of loudspeakers, as this one likely would have to do in sound-reinforcement and other professional applications.

General Info: Dimensions are 19 inches wide; 7½ inches high; 12 inches deep. Weight is 47 pounds. Price: $1,099.

Joint Comment by L.F. and N.E.: Since this hefty power amplifier from CM Labs arrived with no instruction manual (only a single sheet brochure plus similar brochures for other CM Labs products), it would be difficult to comment on its circuitry or design philosophy. Some inferences (generally very good) can be drawn from the more sophisticated specifications printed in the brochure and listed, apart from "standard performance specifications" under the heading of "transient inter-modulation distortion related specifications." Here we learn that the amplifier's closed-loop gain is 30 dB, while open-loop gain is 60 dB. This suggests a rather minimal amount of overall loop negative feedback. The amp is also said to have rise-time of less than 2.5 microseconds, a phase-shift lag of less than 1 degree to 20 kHz; propagation delay of less than 150 nanoseconds; undershoot or overshoot (to a step function) of less than 1 percent; and a recovery time from clipping from either polarity of less than 300 nanoseconds. All this translates to very clean sound and very good stability. Off the bench, and in normal use, the CM920 did produce such sound, and it proved difficult to overdrive (what with its high power output capability plus its extra 1.4 dB of dynamic headroom). It also seemed to stand up well after extended use at high levels.

CM LABS CM920 POWER AMPLIFIER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Continuous power for rated THD,		
8 ohms, 1 kHz	250 watts	307 watts
4 ohms, 1 kHz	400 watts	420 watts
FTC rated power, 20 Hz to 20 kHz	250 watts	264 watts
THD at rated output, 1 kHz, 8 ohms	0.15%	0.015%
1 kHz, 4 ohms	0.15%	0.15%
20 Hz, 8 ohms	0.15%	0.024%
20 kHz, 8 ohms	0.15%	0.13%
IM distortion, rated output, SMPTE	0.10%	0.04%
CCIF	NA	0.0047%
IHF	NA	0.04%
Frequency response at 1 watt	± 0.25 dB, 5 Hz to 90 kHz	± 1 dB, 5 Hz to 55 kHz
S/N ratio re: 1 watt, "A" wtd, IHF	NA	75 dB
S/N ratio re: rated output, "A" wtd	100 dB	107 dB
Dynamic headroom, IHF	NA	1.4 dB
Damping factor at 50 Hz	500	*See report*
IHF input sensitivity	NA	0.1 V
Input sensitivity re: rated output	1.0 V	1.55 V
Slew rate (volts/microsecond)	40	confirmed
Power consumption, idling; maximum	35; 900 watts	120; 1066 watts

The 5th Annual Roland Synthesizer/Tape Contest.

Whatever your level, be it amateur or professional, Roland invites you to enter its 5th Synthesizer/Tape Contest. Anyone with a creative interest and proficiency in synthesized sound is welcome.

Acting judges for this contest are the notable synthesists: Isao Tomita, Morihiko Wada, Shigenori Kamiya, Frank Becker, Makoto Moroi, Robert A. Moog, Oscar Peterson and Ralph Dyck. Judging of acceptable submitted materials will take place on October 31, 1981. Winners will be notified soon after screening is completed.

Please examine the following conditions before completing our application form:

ENTRY CLASSIFICATIONS AND AWARDS:

CLASS A/GRAND PRIX

To qualify, you must be a professional musician and a prize winner in past contests.

Grand Prix (First Prize)
1 winner will receive a Certificate of Merit, a trophy, a Roland Rhythm Composer (TR-808) and a Digital Sequencer (CSQ-600).

Semi-Grand Prix (Second Prize)
2 winners will each receive a Certificate of Merit, a commemorative shield and a BOSS Monitor Speaker.

CLASS B/MULTI-TRACK RECORDING WITH SYNCHRO SYSTEM

To qualify, you must be a new entrant using a multi-track tape recorder and microcomputer controlled equipment.

First Prize
2 winners will each receive a Certificate of Merit, a commemorative shield and a Roland Rhythm Composer (TR-808).

Second Prize
3 winners will each receive a Certificate of Merit and a BOSS Monitor Speaker (MS-100).

CLASS C/MULTI-TRACK RECORDING WITHOUT SYNCHRO SYSTEM

To qualify, you must be a new entrant using the sound on sound recording process only.

First Prize
2 winners will each receive a Certificate of Merit, a commemorative shield and a Roland Rhythm Composer (TR-808).

Second Prize
3 winners will each receive a Certificate of Merit and a BOSS Monitor Speaker (MS-100).

CLASS D/SOUND DESIGN

To qualify, you must utilize one of the following methods: drums as main sound, sound effect and poem recitation, sound cartoons, image sound, etc. Regular music is unacceptable.

First Prize
2 winners will each receive a Certificate of Merit, a commemorative shield and a Roland Rhythm Composer (TR-808).

Second Prize
3 winners will each receive a Certificate of Merit and a BOSS Monitor Speaker (MS-100).

CONTEST SPECIFICATIONS:

- The contest is limited to synthesizer pieces.
- The piece may be presented as music, sound collages, drums or sound effects. Any piece produced without a synthesizer will be disqualified.
- Each contestant can enter only one tape.
- The entry tape must be no longer than 5 minutes (any tape exceeding 5 minutes will be disqualified).
- Copyrighted material used for radio, TV, movies and records will be excluded.
- Tapes must be premiere pieces (never presented elsewhere).
- No entry shall be entered in any other contest until the winner of the Roland 5th Synthesizer/Tape Contest is announced.
- The copyright of the winning pieces will belong to Roland Corporation.
- Generally, recordings made through microphones will not be accepted. However, if the microphone is necessary due to the composition of the piece (e.g. for vocals, narrations, natural sound, the vocoder or special sound effect), a written explanation of the mic's use is required.
- Entries will not be returned.

This contest is sponsored by Roland Corporation. Co-sponsors are TEAC Corporation, Pioneer Electronic Corporation, BOSS Co. Hitachi Maxell Ltd., Sumitomo 3M Ltd., TDK Electronic Co., PIPERS, BASF Japan Ltd. and BSR (Japan) Ltd.

APPLICATION PROCEDURE:

Fill in all the information on the preliminary application form below. Send it to:

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Synthesizer/Tape Contest
2401 Saybrook Avenue
Los Angeles, CA 90040

You will then receive a formal Roland application which when completed, must accompany your tape. Send no tapes with the application below. Preliminary application must be received no later than August 15, 1981.

Name: _____

Street: _____

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Soundcraft Series 800 Mixer

By John Murphy and Jim Ford

The new Series 800 family of consoles from Soundcraft can be configured for a variety of audio mixing applications.

Starting with either an 18-, 26- or 32-channel mainframe, it is possible to build up consoles which are optimized for either studio recording, mixing "live" sound or for on-stage monitor mixing. A complete console consists of eighteen to thirty-two input modules, four output modules (two channels each) and a master module. Two types of input modules are available: a Standard Input module and a Monitor Input module. There are also two types of output modules: a Recording Output module and a P.A. Output module. The three different console configurations are arrived at by combining the appropriate modules into a system. A recording console results when Standard Input modules are combined with Recording Output modules. Using Standard Input modules with P.A. Output modules results in a studio quality P.A. mixing console. For a monitor mixer, Monitor Input modules are combined with P.A. Output modules.

Each configuration of the Series 800 mixer provides four band/six knob EQ and complete soloing facilities. The board's signal levels are monitored at the eight output groups and the stereo monitor outputs by way of thirty-segment LED bar graph meters. Each of these meters is individually switchable between VU and peak reading characteristics. The board's master module provides control over signal monitoring as well as complete talkback, slating and communications facilities. The master module also includes a variable frequency oscillator which can be assigned to the output groups and/or the four AUX sends.

The console version that we received for evaluation is an eighteen channel recording console with sixteen track monitoring and an optional Effects Return module. The remainder of our report will therefore be concerned with the recording version of the Series 800 mixer although we expect that many of our comments will apply to the other versions as well. The 18-channel recording console is priced at $14,850. The optional effects return module costs $540.

General Description: As a recording console, the Series 800 is required to serve as the signal processing and routing center for many different multi-track recording operations. It is convenient, however, to define three basic modes of operation for a recording console: record mode; overdub mode; and mixdown mode. When the console is used in the record mode it must preamplify the signals from microphones in the studio and route the resulting line level signals to both the multi-track tape recorder and the control room monitoring system. After the initial tracks have been recorded, the console may be called upon to allow overdubbing of additional tracks. The console will then have to process microphone signals as before but must now also allow the simultaneous playback of previously recorded tracks. Once the multi-track recording is complete, the console will be operated in the mixdown mode. During the mixing session the multi-track tape will be played back through the input channels of the console (rather than through the monitor section of the console) and mixed down to a stereo signal.

With this overview of the multi-track recording process in mind, let's now survey the Series 800 recording console modules. The complete console consists of eighteen Standard Input modules, four Recording Output modules (two channels each for a total of eight mixing busses), one Master Module and one Effects Return module. Each module connects into the mainframe by way of a flexible ribbon cable interconnection system which is independent of the mechanical seating of the module.

The 800 Series Standard Input module employs an electronically balanced, transformerless microphone preamplifier. The gain of this preamp can be varied between 30 dB and 70 dB through the use of the "Mic Trim" control at the top of the module. A push button above this trim control inserts a 30 dB pad ahead of the

mic preamp for high level input signals. Microphone phantom powering (48 V) can be selected on individual input channels by depressing the button—labelled "PWR"—above the mic pad switch. A "phase reverse" button in the input section allows the polarity of the input signal to be reversed, if desired. A channel's line input signal is selected by depressing a push button near the top of the module. These inputs are normally connected to the corresponding tape return input in the board's monitor section. Selecting the output of the multi-track recorder for mixdown is simply a matter of depressing the line select switches on the input modules. Should the user wish to inject a different line level signal into the channel, the normal connection to the tape return will be interrupted when a plug is inserted into the channel line input jack. A separate line trim control allows the line input gain to be varied between −10 dB and +20 dB. Both the mic and line trim controls have forty-one discrete settings as do most of the other rotary controls on the console. These finely detented controls lend an excellent feel to the Series 800 mixers while providing for repeatable control settings.

The equalizer section of the input module allows modification of four separate frequency bands (low, low mid, high mid and high) with a switchable high pass filter at 100 Hz (12 dB/octave). The high- and low-frequency EQ exhibits a shelving response curve with nominal frequencies of 60 Hz and 12 kHz. The two mid-frequency EQ bands each have a forty-one position frequency control and level control with center detent. *Figures 1* through *4* show the characteristics of the four EQ bands. The center frequency setting as well as each extreme frequency setting is shown for each of the mid bands. When channel EQ is not desired, a push button at the bottom of the EQ allows that portion of the signal chain to be bypassed.

The auxiliary section of the input module allows the channel signal to be injected into any of the system's four auxiliary mixes. These side mixes are typically used for cue and effects signal sends. Sends 1 and 2 are switchable between pre or post fader whereas sends 3 and 4 are fixed as post fader. We should also mention that the send signals are taken after the EQ, channel patchpoint and channel on/off switch.

Routing of the channel signal to the eight mixing busses as well as the stereo mix is controlled by a set of five push buttons which work in conjunction with the channel pan pot. Depressing the button labeled "Mix" routes the signal directly to the main stereo output. The remaining four push buttons assign the channel signal to the eight subgroups in pairs. The pan pot allows panning across the stereo output or the subgroup pairs.

At the bottom of the input strip just above the channel fader are push-button switches for soloing the channel signal and switching the channel on and off. LED indicators are used to indicate on/off and solo status as well as warn of signal levels approaching clipping. The channel fader is a Penny & Giles conductive plastic type with a 90 mm travel. Now let's survey the Recording Output module.

The Series 800 Recording Output modules are double modules with two output groups and four monitor input channels on each module (eight output groups and sixteen monitor sections total). There are two identical control strips on each module, so we will describe just one side of the module (i.e., one of the eight output strips).

At the top of the output strip is a thirty-segment LED bar graph meter with push button to select either VU or peak reading characteristics. A second button selects the meter input between the upper and lower monitor channels. These two monitor channels are identical except that they are fed from different multitrack tape returns. The lower monitor channels handle tape tracks 1 through 8, whereas the upper monitor channels handle tracks 9 through 16. Each monitor channel has access to the four auxiliary sends. Auxiliary sends 1 and 2 are normally pre monitor level control but can be switched to post. Sends 3 and 4 are controlled by a single stereo level control and are post

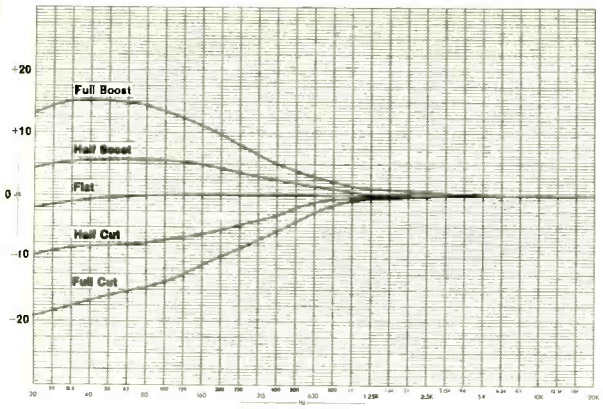

Fig. 1: Soundcraft 800: Low-frequency EQ response curves for the unit.

monitor level control as well as post monitor pan pot. Below the monitor level and pan controls is the "tape return" switch which selects the monitor and meter input between tape send (the group output) and tape return. Each monitor channel has an on/off switch and solo button with green and red LED indicators, respectively. The group output fader is identical to the type used in the input channel. Since there are eight output groups and the console is intended (in general) to be used with a 16-track recorder, the output signal from each sub-group is sent to two inputs of the recorder (the same two channels served by that group's monitor channels).

The console's Master Module provides indication of the signal level of the main stereo outputs by way of thirty-segment LED bar graphs with switchable peak/VU characteristics. Below the meters are master level controls for the four auxiliary sends along with a solo button for each send.

A group of four interlocking push buttons allow the monitoring system to be fed from three external sources in addition to the main stereo signal. At least one of these external sources would typically be the output from a master two-track recorder. Below these source select switches are level controls for the studio, control room and headphones. There are two additional push-buttons for combining the stereo monitor signal to mono, and for dimming the level of the control room monitors by 20 dB.

Talkback, slate and communications functions are located on the Master Module along with a test oscillator which is continuously variable from 45 Hz to 15 kHz.

The Effects Return module included with the console we received allows processing of four different line level inputs before routing to the stereo output. Each return channel has a three-band/four-knob EQ section which can be bypassed if desired. Two auxiliary send level controls are provided to give access to two of the four sends (pre level control). A push button is used to select either sends 1 and 2 or sends 3 and 4.

Signal input and output connections to the Series 800 recording console are made at the rear panel of the unit. The channel, sub-group and stereo output patch point connections employ ¼-inch phone jacks, as do the other unbalanced signal connections. Balanced signal connections are by way of 3-pin XLR-type connectors. The interface with the multi-track recorder is by way of two multipin connectors (one for tracks 1-8, another for tracks 9-16). Besides these connectors, the only other across to the group outputs is via the patch point send which is ahead of the group fader. Monitor inputs 1-9 are available only on the multipin connector, but monitor inputs 9-16 are accessible by way of additional ¼-inch phone jack inputs. Inserting a plug in any of the latter jacks interrupts the normal monitor input from the multitrack recorder. The console connects to its outboard power supply through a separate connector at the rear. The power supply unit provides a positive indication of the presence of each of the four power supply rails by front panel LED indicators.

Let us say that the signal routing and processing capabilities of this mixer are quite extensive. If our readers would care to further investigate the console's functions, we would strongly recommend obtaining the Soundcraft brochure on the Series 800 which contains a complete system block diagram with control layouts for all the available modules.

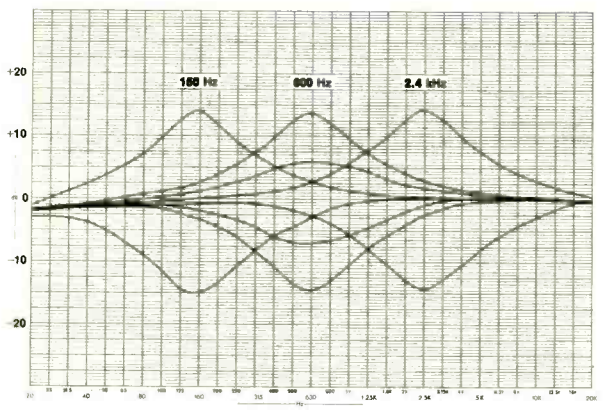

Fig. 2: Soundcraft 800: Low-mid EQ response curves.

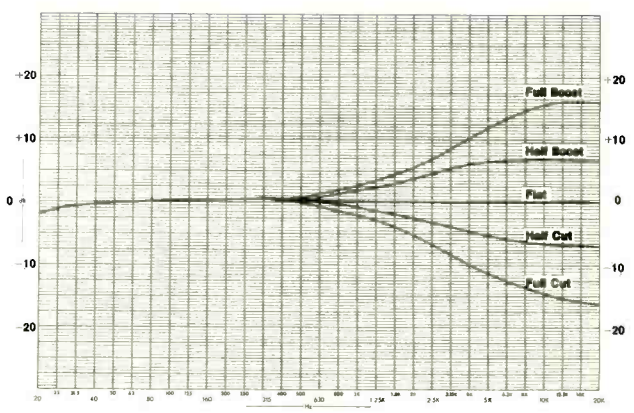

Fig. 3: Soundcraft 800: High-mid EQ response curves.

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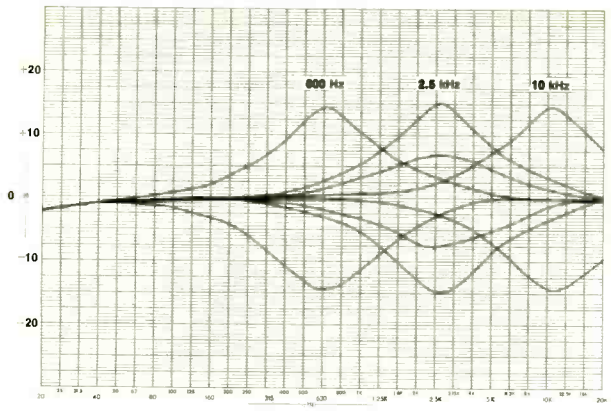

Fig. 4: Soundcraft 800: High EQ response curves.

Field Test: We were fortunate enough to have a major sound job contracted when it came time to field test the Series 800 mixer. The Oklahoma City Symphony Orchestra was holding a special concert featuring Glen Campbell, and we were to provide sound reinforcement for the music hall as well as provide a separate mix for the "live" TV audience. The Soundcraft board was chosen for the task of mixing the TV audio.

By the time we had covered Glen, his band and the orchestra, we had twenty-eight mics to mix through the eighteen input Series 800. We got around this little problem by sub-mixing the keyboards, guitars and background vocals through two smaller mixers and feeding these three signals into line inputs on three channels of the Series 800. Using the grouping capabilities of the mixer, we formed eight sub-groups to simplify the task of mixing the "live" performance (no overdubs or retakes here!). The board's group patch point facilities allowed us to patch in compressors for the vocal sub-group, guitar sub-group, etc. Auxiliary send 4 was used as a reverb send (primarily to "wet" the vocals a bit) with the reverb returning on effects return channel 4.

Through the rehearsal and the performance the Series 800 console performed excellently. We found the board straightforward to use, with no unusual conventions or confusion regarding signal flow or control action.

The board sounded good to us. The EQ was easy to zero out (with center detents), and when EQ was used it was never harsh, but rather musical sounding. At no time were we aware of hearing any pops, hum or hiss from the console. We especially liked the convenience of soloing signals into a headset. And when the show was over, we appreciated the fact that the console is relatively compact and lightweight.

The operator's manual supplied with the Series 800 was quite complete. Each type of module is discussed in detail; then the operation of the console as a system is described for various recording modes. A section of the manual titled "Installation" gives a good discussion of audio wiring practices and even tells how to simply modify the board to interface with a multi-track

LAB TEST SUMMARY
Note 0 dBv = .775 Vrms)

Input Levels

Mic Input:
Minimum input level for 0 VU level indication with input channel trim and fader at maximum (stereo master at nominal setting): – 71.2 dBv

Maximum mic input level before clipping:
without input pad: – 6.3 dBv
with input pad: + 17.6 dBv

Line Input:
Minimum input level for 0 VU level indication (as above): – 22.0 dBv
Maximum line input level before clipping:
greater than + 33 dBv

Output Levels
(at main stereo output)
Output level for "0 VU" indication: – 0.5 dBv
Maximum output level before clipping:
+ 20.7 dBv

Noise Performance
(Note: 20 kHz bandwidth, 150 ohm source, unweighted)
Equivalent Input Noise: – 127.2 dBv
With all faders down noise at the output is: – 78.5 dBv
Raising the stereo master faders to the nominal setting, noise at the output is: – 76.5 dBv
With one input channel set for 40 dB gain and assigned through one submaster, noise at the output is: – 74.0 dBv
With four input channels set as above and assigned over four sub groups, noise at the output is: – 71.5 dBv
With eight input channels as above, noise at the output is: – 69.3 dBv
With eighteen input channels as above, noise at the output is: – 66.8 dBv

Distortion Performance
(Total Harmonic Distortion plus noise at + 10 dBv output level)

Frequency	THD & Noise
20 Hz	.097%
100 Hz	.13%
500 Hz	.042%
2 kHz	.019%
10 kHz	.039%
20 kHz	.079%

Frequency Response: ± 1dB, 45 Hz to 20 kHz
Bandwidth (– 3 dB points): 12 Hz to 58.2 kHz
Power Bandwidth: greater than 58 kHz
Slew Rate Limit: ± 5.2 Volts per microsecond
Normalized Slew Rate Limit:
0.43 volts per microsecond per volt

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recorder operating at −10 dBv. There is some discussion of troubleshooting, but most important is the inclusion of a complete set of schematic and block diagrams.

Lab Test: When we put the Series 800 through our routine lab test we remained pleased with its performance. The specific results of the lab tests can be seen in the summary below.

The mic and line input levels are appropriate for professional audio work. There is plenty of gain available in the mic preamp (70 dB), and with the mic input pad switched in, the mic input can accept even hot line-level signals (+17 dBv) before clipping. The line input similarly has a wide dynamic range as it can handle input signal levels ranging from −22 dBv to +33 dBv.

We were impressed with the very low noise levels we observed at the board's output. Even with all the input channels assigned and set up for a −40 dBv mic input and all faders at the nominal setting the unweighted output noise was only about −67 dBv.

We measured the Total Harmonic Distortion (THD) through the mixer's mic input at 40 dB of gain, but observed that the THD was dependent on the setting of the mic trim control. Lower gain settings resulted in lower THD.

The frequency response was quite flat through the audio range and the board is fast enough to go full output (without slew limiting) at its upper bandwidth limit of 58 kHz. This good high frequency design should insure "slew proof" performance.

When we removed some of the modules from the mainframe for inspection we were pleased to see that the signal and power connectors are detached only after the modules are partially removed. This seems like a more reliable approach than plugging modules into a fragile "motherboard" as they are being seated into the mainframe.

Component quality and workmanship are very good, although we didn't care for the fact that all the integrated circuits are soldered directly to the circuit board. However, there won't be any problems with ICs working their way out of sockets during shipping!

Conclusion: The Series 800 recording console from Soundcraft Electronics Ltd. was evaluated and found to be an excellent board. In addition to its quality performance, the Series 800 is quite attractive and has a luxurious feel to its controls. We fully expect that the P.A. and Monitor versions of this mixer would offer the same high level of performance. Check it out!

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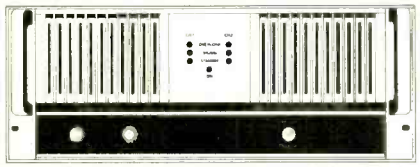

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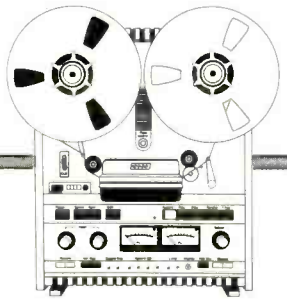

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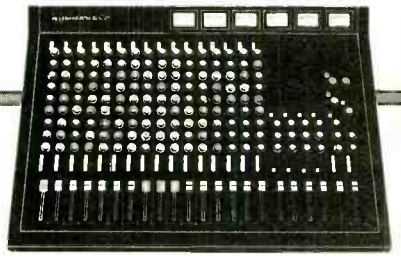

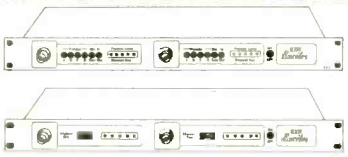

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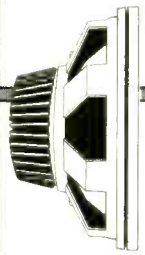

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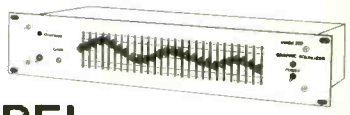

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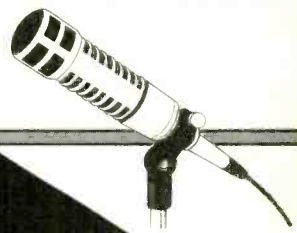

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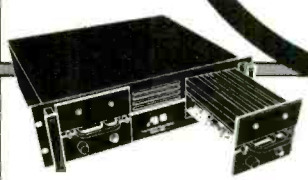

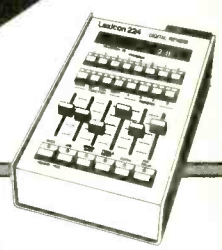

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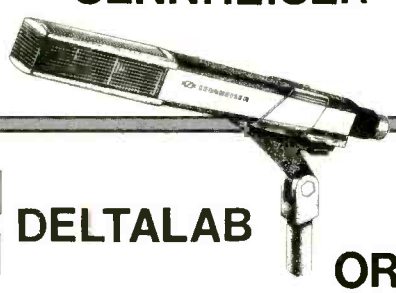

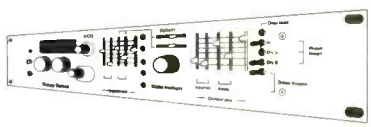

AB SYSTEMS

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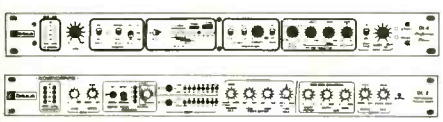

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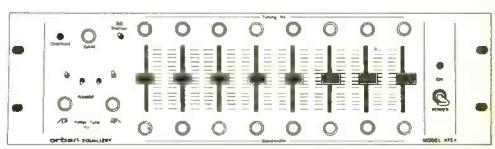

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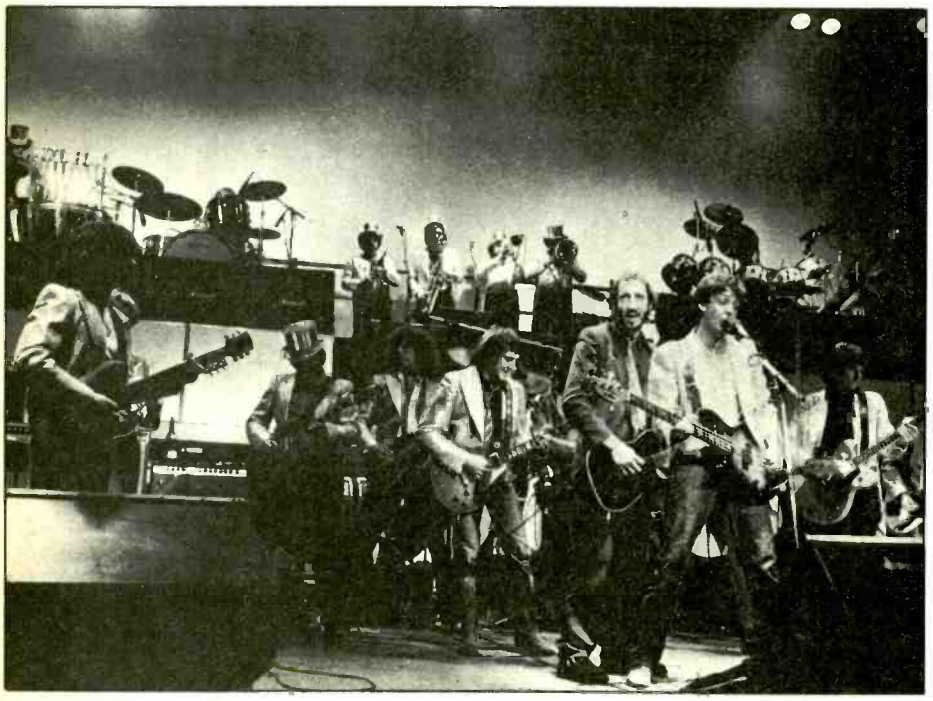
NORMAN WEINSTEIN

POPULAR

VARIOUS ARTISTS: *Concerts For The People of Kampuchea.* [Chris Thomas, sound producer; Bill Price, Dick Allott, Geoff Davis, John Walls, sound engineers; Tom Heid, additional engineer; recorded by Rak Mobile and Island Mobile; mastered by George Piros at Atlantic Studios, New York, N.Y.] Atlantic SD 2-7005.

Performance: **The best of the best**
Recording: **Slick, but a tinge incongruous**

Kampuchea—both the country and the concerts for it—was a massive undertaking. The country (formerly called Cambodia) was decimated during the Indochina war, and when the Khmer Rouge took control of the country in 1975, the decay increased two-fold. Premier Pol Pot cleared his cities of many people, and closed down schools, transportation and communications systems; all international aid was turned away. War broke out again in late 1978, hastening famine and disease. By the time international relief was accepted in June 1979, Kampuchea was a nation populated by the diseased and those starving into oblivion.



CONCERTS FOR THE PEOPLE OF KAMPUCHEA: A rocking commitment to life.

Relief efforts were, and still are, costly. Initially, the Kampuchean people needed not only food and medicine, but vehicles to move supplies to the suffering. To date, the United Nations International Children's Emergency Fund (UNICEF) and others* have provided over 250,000 metric tons of food, and other supplies such as rice seeds, fertilizer and pesticides, trucks, barges and basic medical and educational supplies. In Thailand, where many Kampuchean people sought refuge, UNICEF, et al, have also provided food, water and medical care, as well as assistance to those in Thailand who were inconvenienced by the influx of strangers.

Once again, the music community has come to the aid of those in need. Initiated by talks between United Nations Secretary-General Kurt Waldheim and Paul McCartney, the four concerts performed December 26-29, 1979 at London's Hammersmith Odeon, were sponsored by UNICEF and UNHCR (United Nations High Commissioner for Refugees). The profits from the shows, the album and the film that was made of the event, will go towards helping the still-desperate Kampuchean people.

The musicians who participated in this project were among Britain's best, and *Concerts For The People Of Kampuchea* is a good sampler. From the beginning, the two-record set features rich, deep and full production that

** UNICEF is the lead UN agency for Kampuchean relief. The joint operation also involves the World Food Program (WFP), the Food and Agriculture Organization (FAO), the Office of the United Nations High Commissioner for Refugees (UNHCR), and the International Committee of the Red Cross (ICRC).*

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brings out cleanly the sounds of the Who, the Pretenders, Elvis Costello and the Attractions, Rockpile (joined by Robert Plant for one song), Queen, the Clash, Ian Dury and the Blockheads, the Specials, and Paul McCartney and Wings. Much of the material included can be found on the participating groups' own albums and EPs, yet the inclusion of Rockpile performing "Little Sister" with Robert Plant handling the vocal chores is a pleasant bonus. (The song is so closely in touch with rock's roots, it's easy to forget that Plant fronted the heaviest of metal bands.)

The Rockestra—another exciting extra—is a McCartney-led conglomeration of various concert participants, plus others like some Zeppelins and Ronnie Lane. The unit features a wonderfully spread sound of multi-guitars, drums and horns on "Lucille," "Let It Be," and the pump-it-up "Rockestra Theme," which is the finale to the four day music rescue mission.

About the only flaw on this record comes from the producer's interpretation of a "live" sound. While a slight echo gives the deep feeling of an acoustically perfect hall, the inconsistent levels of audience reaction tends to mar the spontaneous "live" effect. More often than not, the audience's appreciation sounds too subtle (when in fact, it's not) and disjointed from the song itself.

But the overall good sound quality of the performances more than makes up for that blemish, and the record will provide a tapered audio excitement. It should not be forgotten that the record's most important purpose is to help rectify the crisis in Kampuchea, but as an aside, this album will introduce strangers to many of the better modern groups while reaffirming their committment to the forerunners of the art. E.Z.G.

THE WHO: ***Face Dances.*** [Bill Szymczyk, producer; Bill Szymczyk and Allan Blazek, engineers; recorded at Odyssey Recording Studios, London, England; Teri Reed, assistant engineer; mixed at Bayshore Recording Studios, Coconut Grove, Fl.; Jimmy Patterson, assistant engineer; disc mastering by Ted Jensen, Sterling Sound, New York, N.Y.] Warner Brothers HS 3516.

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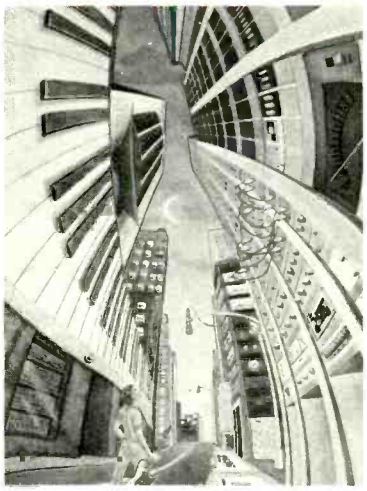

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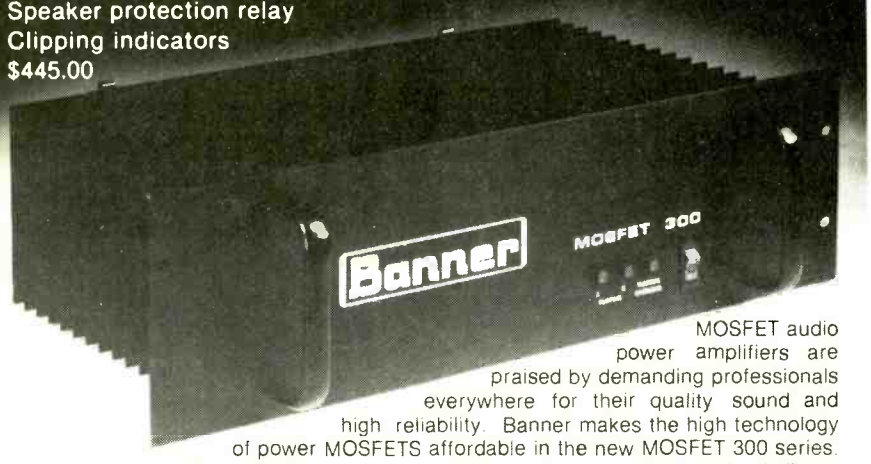

Performance: **Middle-aged Whodunnit**
Recording: **A bit smoky in places**

After the unfortunate death of drummer Keith Moon, the Who secured ex-Small Face Kenney Jones at the twin sticks. With an outright proclamation of rebirth and vitality, the newly formed unit toured the world leaving great anticipation for a new piece of Who vinyl. *Face Dances* is that new record, and while it does have a few bright spots, the album tends to paint a rather bleak picture of the Who's creative vault.

Ironically, the mix on *Face Dances* doesn't allow much of Jones' work to come through strongly. Overall production tends to be cloudy, and the drums seem to be pushed to the back more oft than not. Without a prominent backbeat, many of the songs tend to be lifeless. Even the use of lyric repetition—which does make "You Better, You Bet" stick to the tongue—doesn't really do very much more than seduce the listener into singing along unconsciously.

Generally, the Who have not explored any new musical territory, choosing instead to stick to tried-and-true formulas. Several songs are constructed in a pattern which begins with maximum orchestration, then moves to an instrumental slowdown (where Roger Daltrey sometimes over-exercises his vocal chords which have been annoying since he screamed "Free Me.") The slowdown is hooked by a pickup which keeps the remainder of the number on a peppy level till the fade out. The special effects on *Faces Dances* come in small doses, but the use of a single-vocal call and a multi-vocal response in "Another Tricky Day" is a stab at versatility. In addition, the subliminal float of the words "cache cache" at the end of the song of the same name, is a nice touch, but hardly paves the way for the implied artistic revelation.

The only "new" sound on *Face Dances* comes from the two John Entwistle contributions. While their uniqueness is an automatic plus, both "The Quiet One" (yes, as obvious as it sounds) and "You" have a heavy hard-rock beat which regresses from the Who's roots rather than strengthening them.

Lyrically, Pete Townshend is preoccupied with having sex, not getting sex, troubled times, making records and giving advice. His general tone seems obstinate ("You Better You Bet" again an example), but some sensitivity creeps

through at times, notably on "Don't Let Go The Coat." Townshend even tackles the tricky subject of masturbation in "How Can You Do It Alone," a song about coming home only to find his girlfriend in the midst of the act.

"Daily Records" suggests that Townshend can't keep up with today's fast pace and he just wants to keep making records. His despair comes through on this number and is turned to positive energy later on down the grooves when he realizes that there is a whole new generation to face the same problems. Maybe that's why "Another Tricky Day" boosts these "elders" on the front step (so to speak) telling youngsters to be patient, courageous and persistent. Considering the sound of this record, that message and the fabulous cover jacket are perhaps the only reason for getting *Face Dances* and putting it in one's private record racks. E.Z.G.

SHAKIN' STEVENS: *Shakin' Stevens*
[S. Stevens, S. Colman, and B.J. Cole, producers; no engineer listed; no studio listed.] Epic 3E 36924.

Performance: **Half-hearted rockabilly**
Recording: **Does no justice to the musical form**

Shakin' Stevens has presented the world with a Nu Disk rockabilly sampler of four nice songs and unfortunately, "nice" just doesn't do the job. To his credit, Shakin' uses several of that musical form's trademarks: There's a good hiccupping vocal ("Hey Mae"); a subtle, but effective drum remaining steady in the backgound ("Marie, Marie"); some honky tonk ("Is a Bluebird Blue"); and a guitar that serves as an exclamation point to lyrical sentences about women and love ("Hey Mae" and "Marie, Marie").

During the first two songs, the drum remains a constant pulse in the background, while at times the guitar is pushed up to accentuate its melody. Especially effective is the way this guitar technique is used on "Is a Bluebird Blue." While the verse features the vocals singing one melody, the guitar is off playing another. These bluesy twangs come to a head during the break, where once again its sound level is up front. The piano in this song is not brought to its peak in the mix, and that's a shame, for only towards the end do listeners get to hear what gives the song the rest of its boogie.

"Baby If We Touch," the only tune included here that Shakin' wrote, switches gears for the final time, as the guitar plays alongside the rhythm rather than twiddling in and out as before. While this approach is fine, it only makes clearer the biggest fault of this collection: There ain't no heart and soul here. E.Z.G.

SHAKIN' STEVENS: Nice, but no soul.

JAZZ

ELLIS LARKINS: ***Ellis Larkins.*** [Ellis Larkins, producer; Jeff Zaraya and Tom Lazarus, engineers; recorded August 27 and 28, 1980 at Vanguard 23rd St. Studios, New York, N.Y.] DGTL 101.

Performance: **Honestly Ellis Larkin**
Recording: **Tight, intimate and comfortable**

Ellis Larkins has a reputation, not undeserved by the way, as a bandsman and accompanying pianist on the New York Scene. There are many fine recorded examples of him in these circumstances with artists like Ella Fitzgerald, Ruby Braff and Helen Humes. Frankly it's always been my feeling that only when he is working with such greats as these is he the ideal accompanist. With any lesser lights he has a tendency to dominate which an accompanist or sideman should not do. But here is Ellis Larkins in his own mileu, a category where he has no peer, the saloon pianist (they call them cocktail lounge pianists except when they are accomplished jazz improvisers, the name seems too polite to be an apt description). This is no demeaning title. It was often used to categorize the playing of giants like Art Tatum and Erroll Garner and it fits just as aptly with Ellis Larkins' style. Yet there are really two Ellis Larkins' represented here. One is the pianist, doing what he does best and what he's been doing at the Carnegie Tavern in New York City since the spring of 1978. The other is a less well known facet of Ellis Larkins, the composer. Frankly, as a composer, I don't think he presents any real competition to the other two composers represented on the record (Ellington and Gershwin), yet there is certainly some merit to his tunes even if they are not likely to impress as much more than impressionistic fluff. Much the same criticism could be made of Erroll Garner's original music until he hit the big one with "Misty."

As an essayer of the more popular Gershwin numbers, Larkins is clearly far and above the competition. The wonders he works with an overworked chestnut like "Lady Be Good" by simply shifting the phrase emphasis, makes the trite become new again and

WHEN RECORD SESSIONS TAKE FIRE: SONNY STITT/SHELLY MANNE & HIS MEN

By Nat Hentoff

Most jazz record dates are competent, and occasionally exciting; but much rarer are those sessions that really take fire. No amount of planning can guarantee that kind of date; it's a matter of the interpersonal chemistry of the moment. Everyone, for instance, has to feel like grooving. One abstracted or querulous player will screw it all up. But when everyone *is* cooking, as in Sonny Stitt's *Sonny's Back* (Muse), you've got a classic album that can never date.

Sonny has made many records, but until this, I've never heard one that has the full *presence* of Sonny in a club on one of his nights of improvisatory glory. To begin with, as Sonny says, he had here the "epitome of a rhythm section"—pianist Barry Harris, bassist George Duvivier, and drummer Leroy Williams. Sonny needs no help to swing; from note one, he's an instant, ceaseless swinger. But these three complement him with such enlivening warmth and cohesiveness that the whole date becomes the very definition of the jazz pulse.

And Sonny himself, spurred by tenor Ricky Ford on three tracks, has it all fused together on this album—the hot, incisive sound; a continual flow of unerringly logical and astutely paced ideas; and a youthful zest that does indeed indicate a renaissance of Sonny Stitt as a jazz force. As he says, "I've got my own flavor now. I can't lie about myself." He has thoroughly absorbed and transcended his influences.

The recorded sound is bright, but not artificially so; and spacious.

Another of those special occasions when everyone moved together was a 1959 series of sessions recorded at the Black Hawk in San Francisco. Long hard to get, the first two volumes, *Shelly Manne & His Men at The Black Hawk* (Contemporary) have now been made available again.

Shelly knew this particular combo was really cooking, and that's why he asked the late Lester Koenig, owner of Contemporary, to hurry up from Los Angeles and get this music on tape. The result is an extraordinary chance to hear a group that feels, as well as plays, as one. The late Joe Gordon, a vastly underestimated trumpet player, has never been as consistently compelling on record as he is here. The same is true of tenor saxophonist Richie Kamuca, also gone.

As for the rhythm section, Shelly, of course, is characteristically attentive to all the nuances, those of the individual soloists and the group as a whole. Monty Budwig, a bassist of exceptional tonal as well as rhythmic strengths, never lets the time limp; and on piano, Victor Feldman recalls how thoughtful a jazz improviser he was before he brought his multiple skills into the studios.

Since Les Koenig was in charge of the recording, the sound quality is unfailingly superb; and because of the sensitive engineering, you very much become part of this unusual experience in truly collective creation.

SONNY STITT: *Sonny's Back.* [Bob Porter, producer; Fred Miller, engineer]. Muse Records MR 5204.

SHELLY MANNE & HIS MEN: *At the Black Hawk, Vols. 1 and 2.* [Lester Koenig, producer; Howard Holzer, engineer]. Contemporary S7577, S7578.

it's like you're hearing the tune for the first time. Also leave it to Ellis to find two Ellington pieces that are virtually unplayed today: "Someone To Care For" and "C'est Comme Ca" as well as the only slightly more familiar "Just Squeeze Me."

The digital sound of this new recording seems to have coincided with the coming of age of Vanguard's 23rd Street facility. Whether it's the new Sony digital equipment or whether Jeff Zaraya has finally found the way to give recordings made in this studio a characteristic sound, it seems to have happened and this new release is a fine example of how to preserve a good, tight, intimate piano sound on disc. The bass of Bill Popp and the drums of Jackie Williams don't get in Ellis' way on side one but, as one can hear on side two which is all solo piano, Ellis Larkins needs only himself and a piano to get his message across. J.K.

RICKY FORD: *Flying Colors.* [Bob Porter, producer; Rudy Van Gelder, engineer; recorded at Van Gelder Recording Studio, Englewood Cliffs, N.J., April 24, 1980.] Muse MR 5227.

Performance: **Soaring**
Recording: **Kodachromic brilliance**

Ricky Ford is a young tenor sax player whose playing embodies what the Spanish flamenco guitarists call "duende." When a player possesses "duende," he or she appears overwhelmed by the spirit of the music, inspired by otherworldly muses or demons. Numerous in concert photographs of John Coltrane suggest an artist in what looks like a voodoo trance while working his horn. I've never seen Ricky Ford's face while performing, but if *Flying Colors* is an accurate sampling, Ford's feet must be a few inches above the floor while playing.

RICKY FORD: Audacious genius.

Flying Colors consists of four Ford originals and three showcase tunes by Monk, Ellington, and Strayhorn. There is a wonderful balance in this program. Ford honors his artistic forefathers while clearly indicating his own innovative role within the Monk/Ellington tradition. The album opens with "Jordanian Walk," a Ford original inspired by his son learning to walk. From the first note of sax, one is struck by Ford's originality, authority and driving energy. He is already, at 27, a *major* player of staggering force. After a few sax choruses, *then* I noticed the work of his remarkable sidemen. John Hicks plays a vibrant and melodically sophisticated piano which handsomely complements Ford throughout this session. The bass playing of Walter Booker is lovingly resonant and Jimmy Cobb's drumming is full of fire and brimstone. Hicks, Booker and Cobb add subtle shadings of tonal color to every cut and know how to most effectively showcase Ford's emerging genius.

And if "genius" seems too strong a word to describe Ford's playing, give a critical listen to his version of Ellington's "Take the Coltrane." The tune in its original version was a curious amalgam of Ellington and Coltrane, a blues-flavored homage from one giant to another. Ford has the sheer audacity to actually tinker with the Coltrane sax style and superimposes his own style upon the classic Coltrane licks. The result is nothing less than exhilarating. His version of Monk's "Bye-Ya" is also both an homage as well as a creative extension of a master's composition. Ford seems determined to never record someone else's tune without substantially alchemizing it to make it his own.

Most of the tunes on *Flying Colors* are economical, upbeat affairs. Cobb's drums provide waves of complex rhythms ranging from Latin to Bop. Ford's "Portrait of Mingus" is an exception. Ford recorded in some of Mingus' last sessions and one assumes "Portrait" is a musical meditation on

his former employer/teacher. There is an achingly pensive and bittersweet tone to Ford's horn in this piece. I look forward to hearing more of this facet of Ford's sensibility in future recordings.

Producer Bob Porter and engineer Rudy Van Gelder treat Ford and his band with the respect and sensitivity they richly deserve. The sound is clear, balanced, brilliant, bright. It seems fitting that the old veteran Van Gelder presided over this session. Van Gelder has been in the studio to witness the birth of countless young jazz talents. Glad he is participating in Ford's coming out party.

And there's another term the flamenco guitarists use to describe playing like Ford's: "*Ole*!" N.W.

CLASSICAL

PROFESSOR PETER SHICKELE/ P.D.Q. BACH: *Liebeslieder Polkas and Twelve Quite Heavenly Songs.* [Seymour Solomon, Peter Schickele and William Crawford, producers; Kevin Boutote, Tom Lazarus and Jonathan Thayer, engineers; recorded at Vanguard Studios, New York, N.Y.] Vanguard 79438.

Performance: **What you'd expect from P.D.Q. Bach**
Recording: **What you'd expect from Vanguard**

There are things I like about Prof. Schickele's alter ego (P.D.Q. Bach) and things I dislike about him. The two sides of this record serve to point up the difference. The "Liebeslieder Polkas" are supposed to be a satire on the "Liebeslieder Waltzes" by Brahms. Brahms wrote for chorus and piano (four hands). P.D.Q. Schickele gilds the lily by adding a fifth hand on the piano. He also uses texts by several classic poets such as Dryden, Ben Jonson and Shakespeare. He sets such beloved texts as Shakespeare's "Who Is Sylvia" to the banal trivialities of P.D.Q. Bach. I think Shakespeare's poem is better served by the setting offered by Franz Schubert and that the Bach/Schickele satire serves no purpose but to cheapen the text. Granted that is to the debit of Herr Bach and Prof. Schickele but why waste the listener's time and money with this foolish-

CIRCLE 61 ON READER SERVICE CARD

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ness—and that's pretty much what it is. There are some cute gags between the chorus and accompanists in "Who Is Sylvia," but like most gags they cease to be funny with repeated hearings. So the first time you play this record, given that you have a taste for Prof. Shickele's slapstick humor, you may well enjoy it. The second time—yes, it is cute. The third time—ho hum. And from there it's all downhill.

The other side is really amusing because Prof. Schickele is not trying to put us on. He has written a series of twelve songs, each conveying an aspect of one sign of the zodiac or another. Some are really very cleverly done. The best of these, to my taste, is "Now Diddle Had A Twin." Some are merely puns: "Virginia Fairest Virginia" and my personal favorite, "Water Bearer." Yet these are far more inventive and amusing than any travesty on the words of classic poetry and the forms used by Johannes Brahms could ever be. I may excuse, for example, "O holy mackerel," on the grounds that the joke is good enough to justify the cheapening of the original, yet there is nothing in "Song To Cecilia #7" or the "Liebeslieder Polkas" to justify the violence done to Ben Jonson's lines which come out "Eye me only with thy drink."

I'm sure that nothing—not even the promise given to bargain-counter tenor John Ferrante by Professor Shickele that he will never again uncover the works of P.D.Q. Bach—will deter him from doing further violence to the sacrosanct halls of good music. Nor will it deter those who enjoy this sort of irreverence from enjoying it.

In general this is, like every P.D.Q. Shickele performance before it, exactly what you've come to expect from Bach and Schickele. I also find it what I've come to expect from Vanguard: good, clean studio sound yet a bit on the antiseptic side, lacking any real characteristic, identifiable sound such as the old RCA studio 8-H recordings of Toscanini or the more recent recordings of the Philharmonic, etc. made in Columbia's 30th Street studio...But, alas, even as I write this, plans are being made to shut down 30th St. and Columbia is selling it to the highest bidder who will, it is imagined, tear down the finest recording facility in New York to put up another office building, complete with Burger King or Zum Zum on the main floor, and a twenty-four hour garage underneath. J.K.

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