Profile: Guitarist Waddy Wachtel & Drummer Rick Marotta of RONIN

# MODERN RECORDING & MUSIC

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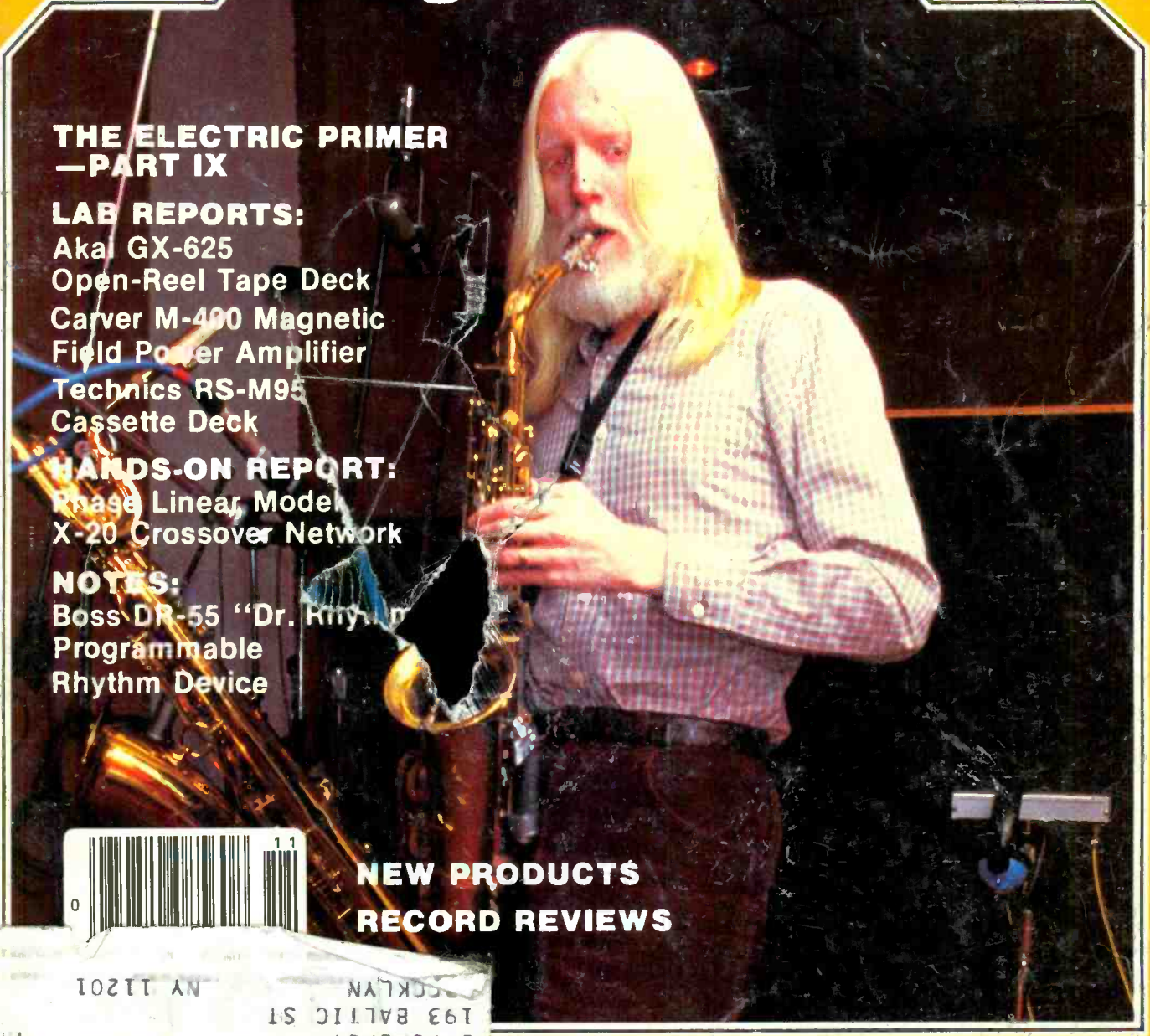
VOL. 5 NO. 2
NOVEMBER 1980

a session with
## Edgar Winter



**THE ELECTRIC PRIMER —PART IX**

**LAB REPORTS:**
Akai GX-625 Open-Reel Tape Deck
Carver M-400 Magnetic Field Power Amplifier
Technics RS-M95 Cassette Deck

**HANDS-ON REPORT:**
Phase Linear Model X-20 Crossover Network

**NOTES:**
Boss DR-55 "Dr. Rhythm" Programmable Rhythm Device

**NEW PRODUCTS**
**RECORD REVIEWS**

# Expression through equalization.

The MXR Dual-Fifteen Band and Thirty-One Band equalizers are cost effective electronic signal processors designed to meet the most exacting equalization requirements in a wide range of professional applications.

The MXR Dual-Fifteen Band equalizer can be used to tailor the frequency response of two sides of a stereo system, or it can act as two separate mono equalizers. In performance one channel can equalize the house system, while the other is used independently in the stage monitor line adjusting frequency response and minimizing the possibility of feedback. In the studio the Dual-Fifteen Band equalizer can be used to compensate for control room acoustics.

The MXR Thirty-One Band equalizer provides maximum detail in the most demanding equalization applications. It can be used in pairs for ultimate stereo control, or in live performance interfaced with PA systems and other instruments. The Thirty-One Band equalizer is also the perfect tool for conditioning film or video sound tracks, and in mastering applications.

The spacing of frequency bands on ISO centers (2/3 octave in the Dual-Fifteen Band; 1/3 octave in the Thirty-One Band) and a flexible system of controls offer superior accuracy in frequency equalization. Each band can be boosted or cut over a range of ±12 dB. Clear, readable markings alongside each level control allow for quick and accurate checks of equalization settings, and aid in resetting the sliders to predetermined positions. The tight mechanical action of the sliders prevents slips during indelicate handling.

The MXR Pro Group equalizers afford maximum control of frequencies while maintaining the highest level of sonic integrity. The Dual-Fifteen and Thirty-One Band equalizers both have a dynamic range exceeding 110 dB and, as all MXR Pro Group products, will drive low impedance lines. Audio signal, including transients, is reproduced faithfully due to a high slew rate and a wide bandwidth.

The MXR Dual-Fifteen and Thirty-One Band equalizers are designed to withstand the demands of a professional road and studio schedule. Their superior design and superb craftmanship reflect MXR's continuing commitment to the manufacture of the highest quality electronic signal processors for today's creative artists.

**MXR Innovations, Inc.,** 740 Driving Park Ave., Rochester, New York 14613, (716) 254-2910

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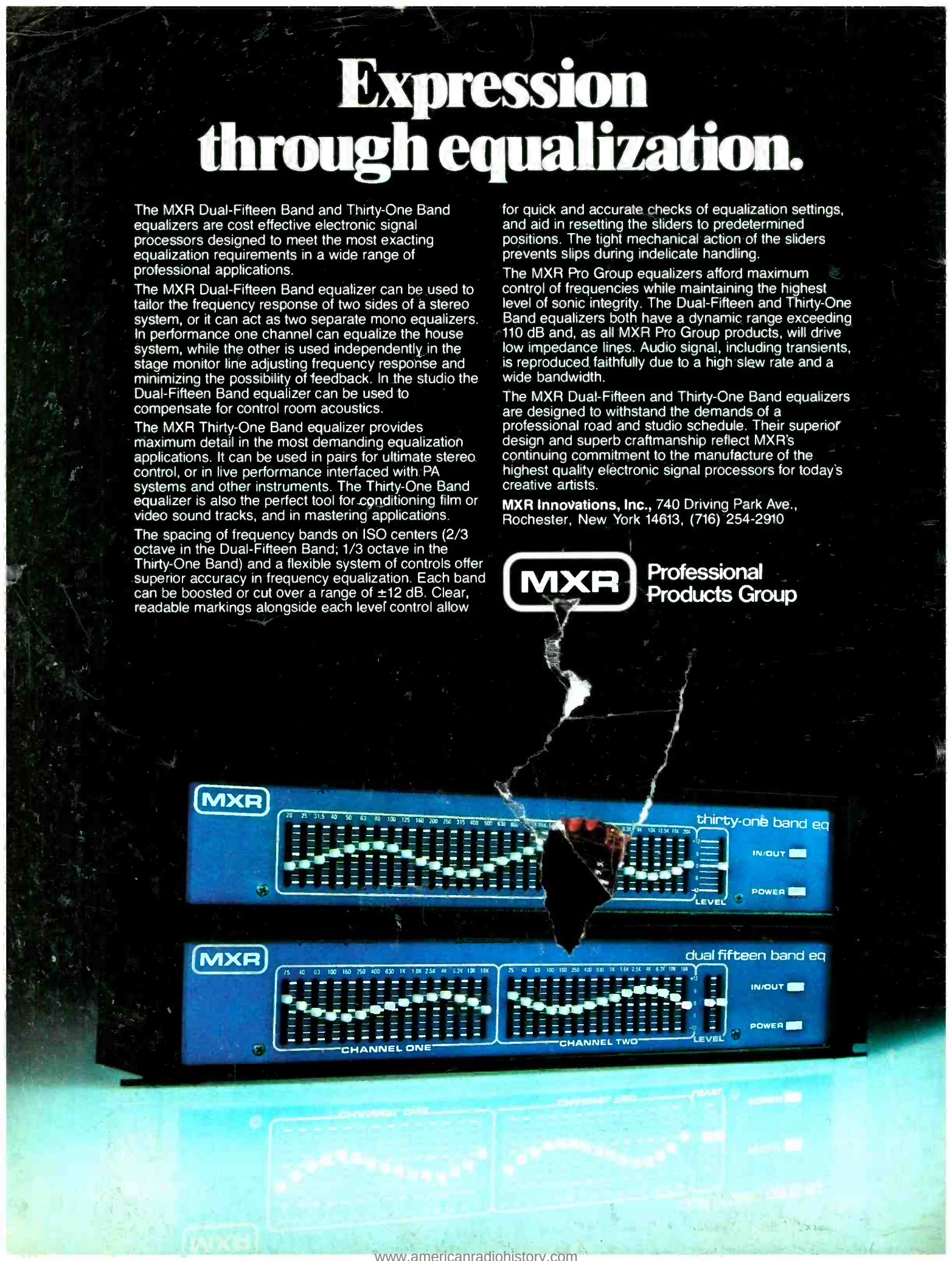


# Get the Message?

We are the makers of Studiomixer and the message is *Totally Modular*. Before you ask, we'll answer. *Totally Modular* is a concept . . . our concept of how a mixing console should be built.

First, get our mainframe. Add to it as many input channels as you need, each one including such features as an overload input light, phantom powering, semi-parametric equalization, 2 effects and four monitor sends, a 100mm fader and many others. Then add as many output channels as you require for your application (up to 8 independent tape, 4 monitors, 2 effects and 2 master outputs), choose a VU meter section to meter the output section you select, and *go for it!*

If you want to start out small, we welcome you. Studiomixer can come in a version as small as 2x2x2. But from that small mixer can be built a 32X8X4X2X2! That's right, Studiomixer's output stage as well as its input is expandable. Which really makes us *Totally Modular*.

We think you will be surprised to find that beside being the most versatile mixer on the market through our *Totally Modular* concept, Studiomixer truly offers the *best* performance specifications of any mixer in its class . . . and, at the lowest price. How does $4,750 sound for a 16X8? That's right, 16 mic channels, 8 tape channels, 4 monitor and two effects sends, and 2 master channels. Unbelievable? Not any more!

To get the message, write to us for a free, illustrated brochure explaining our *Totally Modular* concept. We will also provide you with the name of your closest Studiomixer dealer. And, if you are as excited about *Totally Modular* as we are, then send us $4, your size and sex, and we'll send you postpaid a brochure *and* a Studiomixer *Totally Modular* T-shirt like we're wearing. Either way, you'll get the message.

For information on our products, please write to National Sales Manager, Amerimex Co., P.O. Box 55, Atwood, California. 92601 . . . or phone 714) 528-3360.

CIRCLE 63 ON READER SERVICE CARD


# Silence. The Step Beyond.

Even more important than what an effect adds to your performance is what it doesn't add. Noise—pops, clicks, and hiss can make a good effect virtually unusable in a performance. That's where BOSS effects are different, and it's a difference you'll notice from the moment you turn them on.

You see, all the different effects on the market share the same noisy problem—they all use the same kind of mechanical footswitch, and no matter who makes it, it still has the same problem—it makes an audible "click." That can be a pain in the studio where you have live mikes, but even worse is that a mechanical switch is prone to make popping noises in the signal when it's engaged, and that's a real problem no matter where you are.

BOSS effects have been designed differently. We incorporate what is called F.E.T. switching. This means that there are no mechanical contacts in the signal system, so it won't make an audible click—and it can't make a pop. The switching is done totally electronically and cleanly.

But that's only the beginning of the beauty of BOSS pedals. You'll find a host of other features the competition has yet to catch up with. Features like battery eliminator jacks on every pedal, skid pads that work, and a unique design that allows you to change the battery without exposing the circuit board. And, back on the subject of silence, you'll find BOSS pedals to be the quietest pedals on the market with signal to noise ratio consistently better than 80 dB.

You'll find a BOSS pedal to fit any need—from phasers to flangers, to equalizers to compressors to the new CE-2 Chorus Ensemble, a compact version of our legendary CE-1.

None of the BOSS pedals make noise. No clicks, no pops, no hiss. And that's pretty important. Cause if you're serious about your music you know that what you leave out is as important as what you put in.

BOSS products are designed and manufactured by Roland, 2401 Saybrook Ave., Los Angeles, CA 90040. (213) 685-5141.

BOSS

CIRCLE 71 ON READER SERVICE CARD

NOVEMBER 1980
VOL. 6 NO. 2

# MODERN RECORDING & MUSIC

## THE FEATURES



**COMING NEXT ISSUE!**
*A Session with Jack Bruce, Billy Cobham & Friends and much, much more!*

Cover Photo: Doug Hanewinckel
Ronin Photo: Courtesy Mercury Records
"Primer" Art: Peter Weiss

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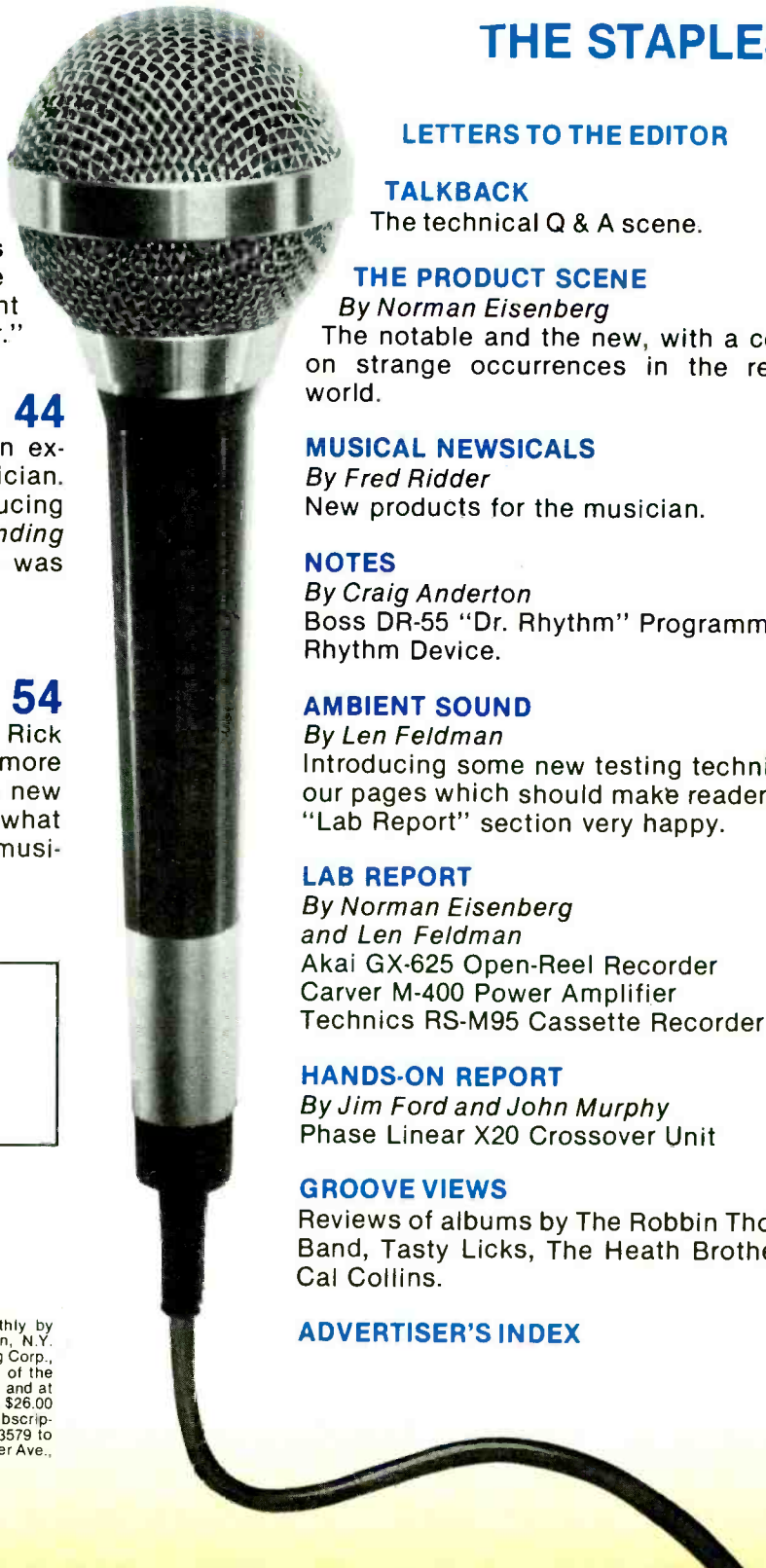

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# LETTERS TO THE EDITOR

## Who's Against The Wall?

I would like to address this letter to Jim Farber, in particular. In reference to his recent record review of July 1980, Vol. 5, No. 10, page 90—Pink Floyd, *The Wall.* I would like to take his review step by step if I may.

Before starting, let me first state that I've been an avid fan of The Floyd since 1967 and I have developed an appreciation for the many changes that have taken place over the years within their music.

Now on to Mr. Farber's review:

First: Performance: Desperately in need of visual or mental/drug accompaniment. First, I feel that one has to have a vivid imagination in the first place to listen, write, record, perform, produce and/or engineer music of any kind— conveying one's thoughts, ideas and dreams to the masses, whether it be the silly little love songs of a Paul McCartney or the nightmarish experiences of being on the road year after year performing night after night, putting up with the general crap from the public and the music business as per P.F.'s *The Wall* expresses. I feel these experiences are viewed with a keen eye by the group and put down on vinyl to express the group's feelings. If it offends someone, then too bad. But if it enlightens someone, then great. After all, we must express ourselves from our own experiences. We are, after all, just products of our environment, aren't we?

Secondly, I too have heard Pink Floyd's music referred to as Acid music at its best. However, does one really have to take drugs to appreciate the music of any band? To appreciate the genius that goes into the writing, performing and producing of a record such as *The Wall?* I, for one, think not!

The next stab: Recording: Too Sanitized. What the hell does that mean? If it's a personal opinion of Mr. Farber's, well, then I'd like equal time to give my opinion. I only wish all of my records were as clean and noiseless as all of Pink Floyd's are. I, personally, have not heard a mix as nice as the Floyd's records since Jimi Hendrix's *Electric Lady Land* album. (But I suppose in Mr. Farber's opinion that album should have a toilet seat wrapper around it also.)

As far as the rest of his review goes I feel that a reviewer should give an unbiased review of a record pointing out both the positive and negative sides of a record. All Mr. Farber showed us was that he is very narrow minded and opinionated when it comes to most, if not all, of Pink Floyd's music. If you don't like the group, Mr. Farber, why don't you state that fact in the beginning of your review so we may all know where you stand? I can't fault you for your opinions and dislikes, but I can strongly disagree with you. In my opinion (and a lot of other people's, according to recent record sales figures,) the group has once again taken the dark side of the music business (if not life itself) and presented it to the world for its approval (or in Mr. Farber's case, disapproval). Mr. Farber, do yourself a favor—go see P.F. in concert. Maybe you'll understand where they're coming from. And in the future try to listen to all music with an open

# The continuing story of TDK sound achievement.

# Parts Five and Six.

The guide roller and spindle pin are the turning point in a TDK cassette. It's there the tape takes on a sudden surge of tension. The winding angle changes sharply to 75°, causing great stress. The slightest imperfection, even a microscopic speck, will cause serious output fluctuations in sound.

TDK engineers began by analyzing existing molding techniques. They knew many manufacturers used a low cost, inferior split-die process. This turned out rollers with seams, which disturb tape travel. Spindle pins were no better. Merely convenient mold extensions with pullout tapers which allowed rollers to slip up, wear out and wrench the tape off the track.

TDK guide roller and spindle pin in cassette.

*Part Five*, the TDK guide roller, is flared and absolutely seamless. Made from a low-friction precision molded plastic, it's created in one piece through an expensive forced-injection mold technique. Its flared edges provide perfect tape guidance while its six spokes maintain rigidity and perfect circularity. The tape flows through the mechanism and past the head gap in true vertical alignment. There's virtually no tracking variation or loss of high frequencies. Sixty checkpoints during the manufacturing process guarantee it.

For *Part Six*, the TDK spindle pin, our engineers chose stainless steel. Machined to size and aligned to a perfect 90°, it's designed without a taper. Micro-polishing and a silicone coating cut down friction. The TDK spindle pin is far more resistant to heat and cold than plastic. It won't bend out of shape and wear down the spindle. Tape is assured safe passage with virtually no flutter or channel loss.

In a TDK cassette, the parts are much like the instruments of an orchestra. All equally important. Music is an outcome of the perfect interplay between them. In the end, that's what's so distinctive about TDK.

Music is the sum of its parts.


TDK

The Amazing Music Machine.



CIRCLE 67 ON READER SERVICE CARD

mind and open ears and just enjoy music for what it's intended to be, a communication between an artist and his audience.

In closing I would just like to say on behalf of the Pink Floyd, their fans and myself, "All and all, Mr. Farber, you're just another brick in the wall."

—Sincerely,
Gordon McAlister
Ridgewood, N.J.

*Jim Farber replies:*

It is unfortunate that you didn't read my review more carefully. If you had, you'd have noticed that I clearly stated that the reason I object to the ultra-clean recording is because it doesn't adequately reflect the brooding conceptual thrust of the record. As for the drug accompaniment argument, I merely meant to suggest that the disc is simply too boring to get through straight (a problem their past albums have usually not encountered). What is most ironic about your letter, though, is your acceptance of Pink Floyd's lyrics, which are (in your own words) about "being on the road year after year, performing night after night . . . putting up with the general crap from the public." For your information, Pink Floyd's last U.S. tour involved exactly two cities. What's more interesting, though, is your correct understanding of how some of *The Wall*'s lyrics regard the band's fans as nothing more than foolish leeches. Perhaps the band is right about this in at least one case.

—Jim Farber
New York, N.Y.

## Getting in Touch with Genya

In your June 1980 issue there was an article by Jeff Tamarkin about Genya Ravan producing Ronnie Spector. I have been a long time fan of Genya and her work. I myself am involved with a production company as a staff writer and assistant engineer. I would like to contact Genya Ravan about some future projects. Could you please send me an address or phone number that I could contact and set up an appointment with Genya Ravan or her business secretary. Thank you.

—Charles Redman
Stroudsburg, Pa.

*Sorry, but we're unable to print home or personal addresses. But her record company, Polish Records, furnishes the following address and phone number: 250 W. 57th Street, New York City 10019, (212) 974-0906. We hope you are successful in getting in touch with her, and we wish you good luck with those projects.*

## Up and Down the Tubes

After reading Craig Anderton's response in Talk Back (July 1980), I would like to put in a few good words about tubes.

Everyone seems to think that the only reason that a musician uses a tube amplifier is for distortion. It's true that when a tube amp is cranked up it starts to sustain more and it has a better type of distortion than solid state

Discover A New World of Musical Potential in One Keyboard

# RHODES MARK III EK-10

A traditional Rhodes electric piano...plus electronic voices with virtually unlimited control...all in one Rhodes.


## New Tonalities...

The Mark III EK-10 is a Rhodes like none you have ever played or heard. A traditional Rhodes with two new electronic voices. All with familiar Rhodes touch dynamics, percussion, decay and sustain pedal control. You have mechanical touch control over electronic voices plus the capability to mix them with the Rhodes sound on the same keyboard. Play the EK-10 as a Rhodes, as an electronic keyboard, or mix the two in any volume or intensity.

## More Tonal Enrichment...

With two distinct electronic voices comprised of three basic waveforms, the EK-10 opens new creative vistas. Use the three waveform switches to produce up to seven waveforms. And since the pitch of the electronic voices may be tuned continuously over a two octave range, you have virtually unlimited control. You can even split the keyboard for additional creative tonal effects.

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Rhodes Mark III EK-10 is now at your authorized Rhodes dealer. See one. Play one. Discover the incredible range of keyboard tonalities at your command in this new electric/electronic Rhodes.

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## Features and Specifications

- Seven waveforms
- Electronic oscillator
- Two continuously variable tone shaping filters
- ± 18 dB boost/cut Rhodes tone controls @ 50 Hz & 4kHz
- Operates from 90 to 130 volts @ 50/60 Hz
- Two octave continuously variable tuning range for electronic voices
- Power consumption: 60 watts
- ± 10 dBm @ 600 ohms
- 20 Hz to 20 kHz electronic voice range
- Keyboard split @ G below middle C

Specifications subject to change without notice.

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**Keyboard Instruments U.S.A.**

amplifiers, but that's not the reason I use them.

I have a lot of equipment and ninety percent of it is tubes. Not just guitar amps—I'm also talking about recording equipment, mixers, reverb units and tuners.

Generally tubes have a smoother sound and transistors have more of a harsh sound. It seems to show up more when amplifying a musical instrument than when you are listening to program material.

Have you ever wondered why some amplifier manufacturers have gone back to making tube amps or have added them to their line?

You can't really say that the transistor is best, or the chip is best, or the tube or fet. Each one has its good and bad points and all the different technologies should be used to their best advantage.

—Alan Wolf
Ellicott City, Maryland

*We received the following response from Craig. (Before you know it we will have created a controversial new realm of thought—tubism.)*

As far as needing to put in a good word for tubes, don't worry—it seems that most guitarists (and some of the "golden ear" hi-fi set) prefer tube equipment, so I'm sure that tubes will be around for many years to come. Also, your point about using whatever technology is necessary to get a particular sound is right on target.

While it would be nice to choose one's equipment solely on the basis of sound, sometimes practicality rears its ugly head. The point I wanted to get across most in my "Talkback" comments was that while you cannot *duplicate* the sound of tube equipment with solid-state stuff (only tubes will give a tube sound), solid state also has some practical advantages going for it (energy efficiency, longer life, etc.) that should be taken into account when choosing a piece of equipment. Also, solid state equipment does not necessarily have to give an inferior sound; just a *different* one.

Thanks for taking the time to write in about this, since it has given me an opportunity to clarify some of my thoughts, and brought up additional points with which I agree but neglected to mention in the original "Talkback" reply.

## Golden Tracks of Railroad

In 1973 Capitol Records released the album *We're an American Band* by the rock group Grand Funk Railroad. The very first pressings of this album were in gold vinyl, and all the successive pressings were in black. I would greatly appreciate any information as to whether any gold copies are still available, and if so, how can I obtain them?

—David Norton
Endicott, N.Y.

*A kind and apologetic person at Capitol Recording Studios explained that the first pressing was a special running of that album. A certain number of gold vinyl copies were produced,*

# 4620 Cabaret instrument series.
## Give the audience more of yourself.

JBL

At JBL, we think a speaker system should give the same all-out performance you do. That's why we developed our new 4620 musical instrument series.

These compact, high-performance Cabaret systems have the power capacity to reach your fans with every note. From low E to the highest harmonics. From the delicate overtones of the acoustic guitar to the multivoiced impact of electronic keyboards. And the sound comes through clean, crisp, accurate.

Each system features E Series transducers with advanced Symmetrical Field Geometry magnetic structures. Result? The lowest magnetic circuit distortion level of any known loudspeaker made today. And each box is fine-tuned to enhance the sound of the particular instrument it will reinforce.

For example, models 4621 and 4622 are designed specifically for electric guitar. Both are highly accurate and powerful. The 15" 4621 handles 150 W, while the dual 12" 4622 handles 300 W. The 4623 meets the unique requirements of acoustic guitar and vocals with a high-frequency ring radiator and special crossover network. The 4625 delivers pure, punchy bass from a 15" woofer with power handling up to 200 W. Specially designed for keyboards, the 4627 combines a long-throw 15" low-frequency speaker with a high-frequency power pack, horn/lens/compression driver and crossover network, to provide high accuracy and outstanding definition.

Enclosures are constructed from rugged multilaminate, cross-grained hardwood, far superior to pine plywood, other softwoods or particle board. Flush-fitting covers make the cabinets their own road cases. The exterior finish is a durable, triple-coat, black polyurethane. Interlocking corners make stacking a snap.

And to eliminate cone bounce while traveling, voice coils are automatically shorted when there isn't a plug in the input jack.

Want to give the audience more of yourself? Then plug into JBL's new 4620 musical instrument series. Handcrafted in the USA by JBL in our Northridge, California facility.

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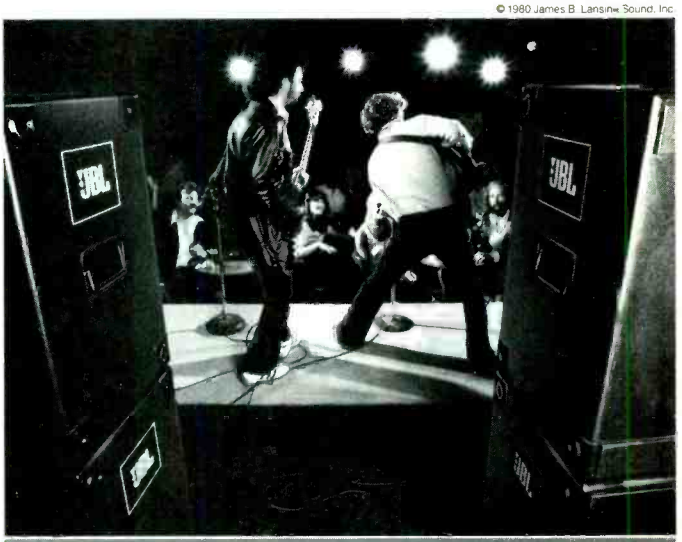
|---|---|---|---|---|---|---|---|
| 4621 | 50 Hz-6 kHz | 150 W | 300 W | 105 dB SPL | N/A | 127 litres 4.5 ft³ | E130 |
| 4622 | 50 Hz-6 kHz | 300 W | 600 W | 106 dB SPL | N/A | 127 litres 4.5 ft³ | 2-E120 |
| 4623 | 50 Hz-15 kHz | 150 W | 300 W | 105 dB SPL | 3 kHz | 127 litres 4.5 ft³ | E130 2903 HF power pack |
| 4625 | 40 Hz-2.5 kHz | 200 W | 400 W | 100 dB SPL | N/A | 127 litres 4.5 ft³ | E140 |
| 4627 | 35 Hz-20 kHz | 150 W | 300 W | 98 dB SPL | 1.5 kHz | 127 litres 4.5 ft³ | E145 2901A HF power pack |






JBL First with the pros.

CIRCLE 140 ON READER SERVICE CARD

*but there were no other gold vinyl pressings done after that. The only gold copy left was the sample copy for their files, if you're feeling up to pilfering it, but other than that, there are no more available.*

## A Bit Biased?

While the Lab Report on premium open-reel tapes was very well-done, there was one sentence in Norman Eisenberg's comments that I wish had been expanded considerably. He said " . . . the serious open-reel recordist is expected to adjust his or her deck to suit a particular tape . . . ".

What the Lab Report does *not* say is that there are likely to be differences between *any* two tape machines of a given model number, or between *any* two reels of a given brand of tape drawn from different production lots that are just as great numerically as the differences between different *brands* of tape and tape recorders.

The recordist who consults the charts and buys a particular brand of tape, believing that in that decision he has solved his recorder/tape compatibility problems is only deluding himself in most cases, and won't achieve the optimum result.

This is why the ReVox people suggest that the recordist buy a year's supply of tape (of whatever brand) from the *same* production lot, and have their machine's bias and equalization settings optimized for that particular formulation. If this is done, I believe it can be shown that *any* of the tapes tested can produce an excellent quality recording on *either* of the two tape machines tested.

I think Messrs. Feldman and Eisenberg would likely concur in these observations, and I would invite a response.

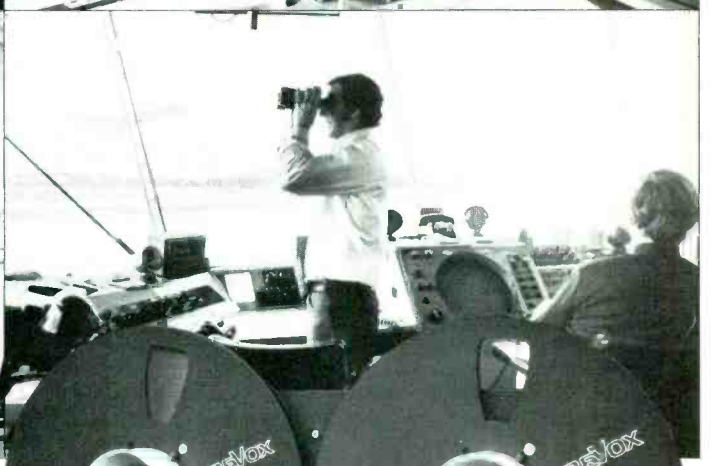
—Richard Lewis, Jr.
Stoneham, MA

*Norman Eisenberg chose to respond to Mr. Lewis' letter, taking it step by step, paragraph by paragraph.*

Regarding the statement appearing in Mr. Lewis' first paragraph: There is absolutely nothing wrong with my statement. It is 100-percent correct, especially when read in context with the material in the report that precedes it and that immediately follows it.

Mr. Lewis' statement in paragraph 2

# The Revox B77 records much more than music.






If you think of the Revox B77 as an extraordinarily well-made tape machine, with a great reputation among musicians and recordists, you're right.

But you're only partly right.

Because the B77 is now available with so many options and in so many configurations that it can be almost any kind of tape machine you want it to be.

Whether you need a deck for broadcasting, mastering, duplicating, security, education, medicine, lab research, sound effects, municipal services, A/V presentations, transportation, or cinema, we have a B77 just right for you. With the same quality that made Revox a legend in the music business. Look at all the possibilities:

FULL RANGE OF SPEEDS: any two from 15 IPS for mastering to 15/16 IPS for logging or surveillance.

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RACK-MOUNTED OR PORTABLE. All B77's have folding handles and available rack-mounting flanges.

VOICE-ACTIVATED CONTROL saves tape in broadcast logging, surveillance, courtrooms, forensic pathology, municipal services, etc.

TIMER CONTROL permits recording or playback at pre-selected times.

AND MUCH MORE: Remote control • Special narrow bandwidth third track • Self-Sync • Stereo slide sync • Variable pitch • High and low-z mic and line inputs.

With all that flexibility, plus the legendary Studer quality, and tape-protecting features like a motion-sensing logic-controlled switching system, the B77 is perfect for anyone who needs to record anything.

Visit your Studer Revox professional distributor and tell him what *you* need.

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In Canada: Studer Revox Canada, Ltd.

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is one of those sweeping generalizations that cannot be proven or disproven. It would take the next hundred years to really document his idea one way or the other. The rest of his letter is based on this unprovable statement.

As for his third paragraph, what appears here is a gratuitous statement at best. It is, in any event, exactly the reason for the validity of my initial statement which he criticizes in paragraph 1.

Paragraph 4 is simply a plug for an individual tape-recorder manufacturer. There is no guarantee that a "year's supply" will indeed contain all the same identical tapes. What is a "year's supply" anyway? Ten reels? One hundred? One thousand? Five thousand? Who's kidding whom here?

—Norman Eisenberg
Technical Editor
*Modern Recording & Music*

## We've Got Your Number

Could you please tell me what type of transistor is used in the schematic of the correlation (phase) meter in the Talkback letter, "Taking the Pressure Off Musicians," Volume 5, No. 5, page 28, by David Kalmbach. Thank you and, by the way, thanks for a terrific magazine without which many of us wouldn't be so well informed.

—Gerald Malone
Lincoln Park, MI

*David Kalmbach himself answered your question for us. He told us that the number of the transistor used in the metering device which appeared in the schematic was: 2N2712. If you still have any questions to ask David, write again or write directly to him at Atlantic Sound Productions, 130 W. Market St., Marietta, Pa. 17547.*

## Messing Up—1½ Feet of Klee

*It's time for another apology. In printing Joe Klee's review of The New York All Stars'* New York, Sound of the Apple *album on pages 92 through 94 of the October 1980 issue of* MR&M, *we made a small but powerful error.*

*In the next to last paragraph of Klee's review, he writes about Phil Bodner, the tenor saxophonist, who plays with "the drive and the verve and the booting sound that I've heard om few tenor saxophone players since my main man, the late Chuck Berry."* Well, *that's what was* printed. *What* should have been *printed was "the late Chu Berry," the tenor sax player who was a completely separate individual from the still alive Chuck Berry, the rock 'n' roll pioneer. Our apologies to our readers, and to Joe Klee, whose jazz review was slightly stained by this mix-up of Berrys.*

## Dokorder Forum

I am writing in regard to the letters in your January 1980 "Letters to the Editor" column concerning Dokorder tape decks. I own a Dokorder 7140 which has made some excellent tapes, but has also had some excellent breakdowns. I would like to exchange any information on techniques, service, and modification.

—Dan Thibault
Liverpool, N.Y.

*You can send letters to* MR&M *or write directly to Mr. Thibault at P.O. Box 173, Liverpool, NY 13088. If a Dokorder in your home has suffered a breakdown, then you can understand and sympathize—it's toughest on the* family *of the owner. Share your experiences and help someone.*

## An August Controversy

In reference to the August 1980 issue, I would first like to say to Mr. Hank Strasser that in reality, production of a record is irrelevant. If a performance is good, records will sell. If a performance is poor, records will still sell. If you have semi-pro equipment or its equivalent, watch your meters, the rest is up to the musicians. That's what it takes. Being in the studio business for twenty-five years, I should know. Thank you.

—Roland Mills
Bent Studios
Morgantown, W. Va.

*While we certainly agree that in order for a tune to be a hit, "it's got to be in the grooves," we feel that perhaps Mr. Mills is overstating his case. Creative production can most certainly make a good song a great record. But in general, if the material isn't potentially good, and if musicians are not properly prepared, etc., nothing is going to come from the recording sessions.*

# Only Custom-Tailored Sound meets your taping needs.

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The entire ADC Sound Shaper line is impressive. The basic Sound Shaper One is a great introduction to frequency equalizers. And the top-of-the-line Sound Shaper Three, the *Paragraphic*™ equalizer, combines the ease and control of a graphic equalizer with the precision and versatility of a parametric. And, all Sound Shaper equalizers, except the Sound Shaper One, feature LED-lit slide controls, allowing for visual plotting of the equalization curve.

With the Sound Shaper Two MK II, you can appreciate the difference custom-tailored sound makes—over and over again.

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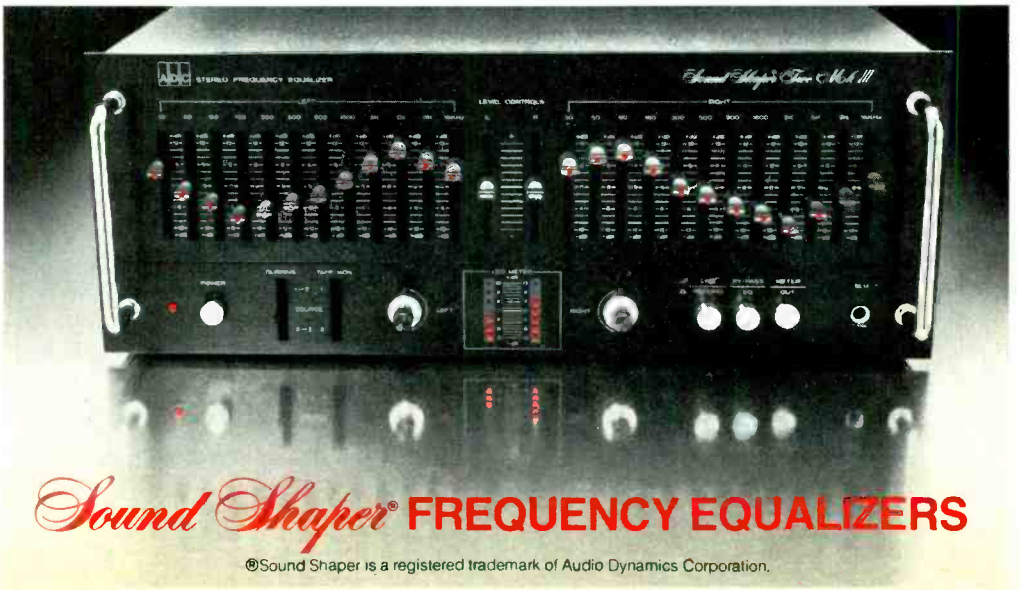

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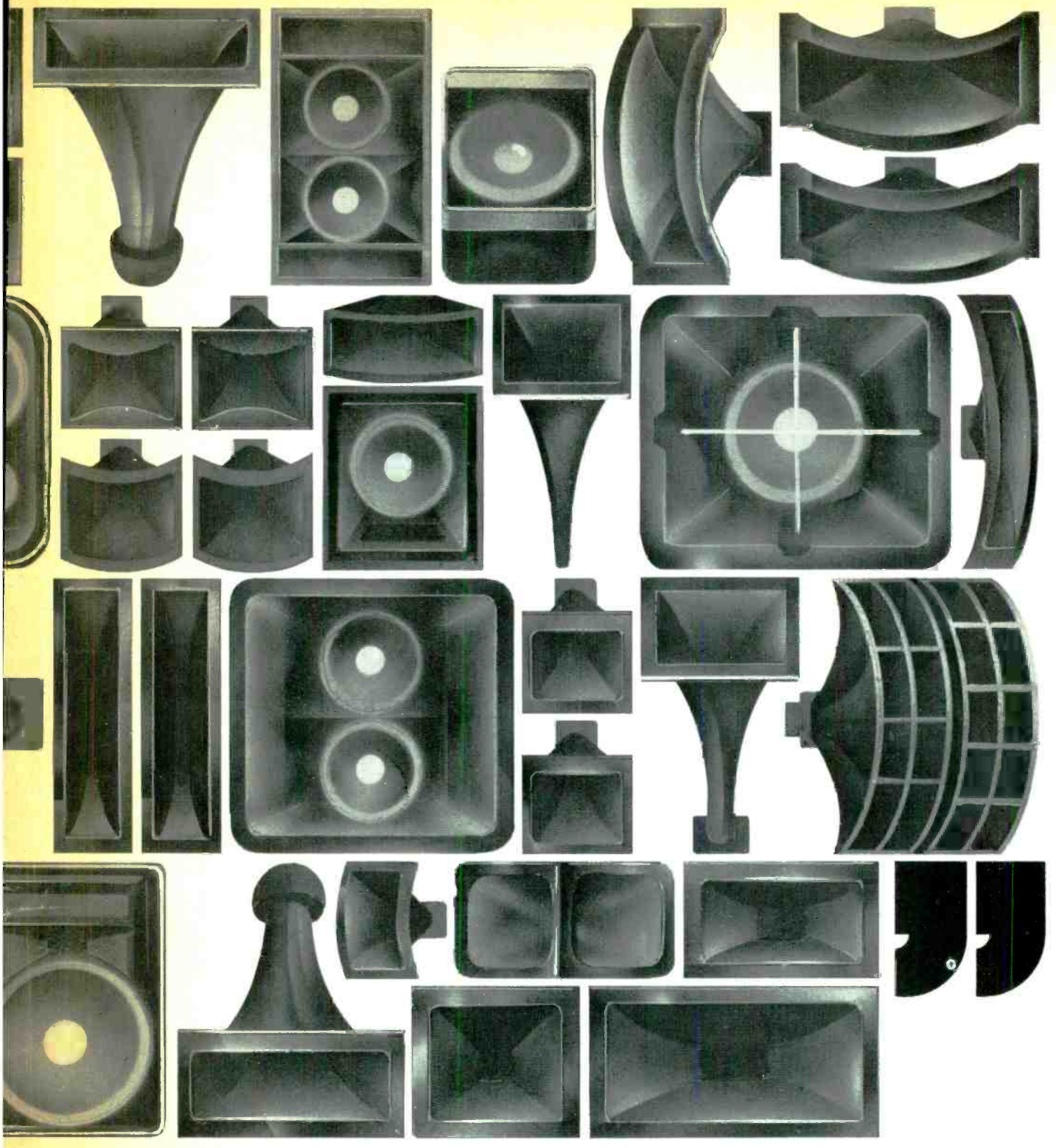

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# TALK BACK

*"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.*

---

## An "Effective" Solution

**I mix sound for a local band. I use an 18-channel Trouper board, a 10-band Biamp EQ, the Eventide "Baby" Harmonizer, and a Yamaha analog delay. I won't go into the rest of the system since it really has nothing to do with my problem, although I will add I am running a three-way system with all Crown power.**

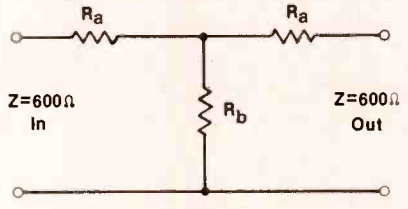
**My problem is in the effects. I just recently purchased the "Baby" Harmonizer to go along with the analog delay. Since I have only one effects out on the board, I am running the units in line. Between the two units I can have almost any effect imaginable, but whenever I try to flange a guitar or double a voice, or whatever the effect might be, I get a large volume increase along with the effect. I've tried every possible input and output combination and yet that still doesn't seem to be the answer.**

**I believe the only answer may be to pad the effects. My question is how to properly pad the effects and how much padding should be used (the Harmonizer has a 2-watt preamp and the analog has a 9-watt preamp). How is a pad put together? My knowledge in this area is a bit limited, although I understand some type of resistor may be enough.**

**—Mike Jones**
**Indianapolis, Ind.**

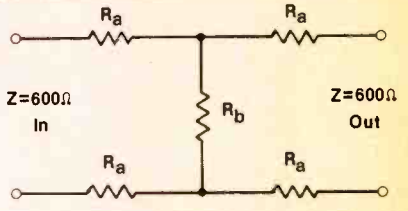
Padding down the combined output of your effects units may help you to solve part of your problem. Before we get involved with the construction of pads, I have two things I would like you to try first.

Try running a "Y" cord from your effects out to each of your effects. You will have only one send but you will be able to adjust the input level to each separately. This will offer you better control of the proportion of each effect.




Unbalanced

| Loss in dB | $R_a$ | $R_b$ |
|---|---|---|
| 0.5 | 18 | 10,000 |
| 1 | 33 | 5,100 |
| 2 | 68 | 2,700 |
| 3 | 100 | 1,600 |
| 4 | 130 | 1,200 |
| 5 | 160 | 1,000 |
| 6 | 200 | 820 |
| 7 | 220 | 680 |
| 8 | 240 | 560 |
| 9 | 270 | 470 |
| 10 | 300 | 430 |
| 12 | 360 | 330 |
| 14 | 390 | 240 |
| 16 | 430 | 200 |
| 18 | 470 | 150 |
| 20 | 510 | 120 |
| 22 | 510 | 100 |
| 24 | 510 | 75 |
| 26 | 560 | 62 |
| 28 | 560 | 47 |
| 30 | 560 | 39 |
| 32 | 560 | 30 |
| 34 | 560 | 24 |
| 36 | 560 | 18 |
| 38 | 560 | 15 |
| 40 | 560 | 12 |




Balanced

| Loss in dB | $R_a$ | $R_b$ |
|---|---|---|
| 0.5 | 8.2 | 10,000 |
| 1 | 18 | 5,100 |
| 2 | 33 | 2,700 |
| 3 | 51 | 1,600 |
| 4 | 68 | 1,200 |
| 5 | 82 | 1,000 |
| 6 | 100 | 820 |
| 7 | 110 | 680 |
| 8 | 130 | 560 |
| 9 | 150 | 470 |
| 10 | 150 | 430 |
| 12 | 180 | 330 |
| 14 | 200 | 240 |
| 16 | 220 | 200 |
| 18 | 220 | 150 |
| 20 | 240 | 120 |
| 22 | 240 | 100 |
| 24 | 270 | 75 |
| 26 | 270 | 62 |
| 28 | 270 | 47 |
| 30 | 270 | 39 |
| 32 | 270 | 30 |
| 34 | 270 | 24 |
| 36 | 270 | 18 |
| 38 | 270 | 15 |
| 40 | 300 | 12 |

Values used are 5% EIA Values.

For loudspeaker circuits use adjustable wire wounds.

If $Z \neq 600\ \Omega$ divide all values by $\frac{600}{Z}$

Table used by permission Don Davis, Syn-Aud-Con

Also, running the units in line may be part of your level problem. With the inputs "Y"-ed, take the outputs separately into two return or channel inputs. If your console has only one return, "Y" the outputs and control the level of each unit with its input. You may have to spend some time adjusting the various outputs and sensitivities to obtain the proper mix.

Secondly, the special effects units should be run so as to produce the maximum amount of "effect" and the minimum amount of dry or direct signal. For instance, on the Yamaha there is a control labeled "mixing." Run this to the delay side. In this way when you bring up your return channel, you will then receive only the "effect" which should add only a small amount of gain to the system. Let your console mix the direct signal from the channel with the "effect" output.

In constructing a pad, the impedance of the input and output is of prime importance. This subject is covered in great detail by Don Davis in his article "Impedance Matching for the Sound Engineer," which was originally printed in *db* magazine April and May of 1974 and is also available in Don's textbook, *Syn-Aud-Con*.

For the purpose of pad construction in this article we will assume equal input and output impedances of 600 ohms. There are basically two types of pads—unbalanced and balanced. They are constructed as in the diagram.
From this table, we see that for a 10 dB loss in a 600 to 600 ohm unbalanced line, we would use two 300 ohm, 10% resistors for $R_a$ and one 430 ohm, 10% resistor for $R_b$. For a 10 dB loss in a 600 to 600 ohm balanced line, you will see that $R_a$ equals 150 ohms and $R_b$ equals 430 ohms. 10% carbon resistors will work fine in this application.

—Will Parry
General Manager
Maryland Sound Industries, Inc.
Baltimore, Md.

## In the Pink?

**I use a Tapco 2200 Graphic Equalizer. I have heard that the best way to set the equalization for a room is to use a pink-noise generator. Do you agree that this is the best way to do it? How is it done? I have a synthesizer that has a pink-noise generator in it; would this serve the purpose?**

**Just in case I can't utilize this source, can you also supply me with the names of some pink-noise generator manufacturers?**

**—Bill Madigan**
**Middleboro, Mass.**

There are basically two reasons for using an equalizer:

1) To make tonal changes to the sound of the music (i.e., add more bass or high frequencies, etc.). This is done by using your ears much like you adjust your stereo.

2) To make adjustments to the frequency spectrum of the sound system in order to compensate for the peaks and dips caused by the acoustics of a room (concert halls, clubs, etc.). This requires the use of some kind of a sound analyzer that will identify the volume of the peaks and dips and their frequency. This cannot be done by ear too successfully, therefore, I recommend that you do not buy a graphic equalizer to compensate for room acoustics unless you also can buy the necessary test equipment to make the correct measurements. Even after you have the right test equipment, it requires a great deal of practical experience to use the equipment and obtain consistent and meaningful results. A thorough knowledge of the whole sound system and its components is also important so that electronic problems can be solved first.

In order to perform the test, a flat sound source is needed. Pink noise consists of equal sound energy per octave. That means all the frequencies from 20 Hz to 20 kHz are produced at the same time, and each octave of frequencies is the same volume. Because this is a flat sound source, it can be used to play through the sound system and then measured by a calibrated mic and sound analyzer. The easiest way is to use a "real time analyzer." This is an instrument that divides the sound up into octave or third octave bands (or 1/6- or 1/10-octave) and then displays the volume of each band of frequencies on a screen so that you can see the relative volumes. This is done simultaneously so that you can see what is happening right now in "real time." The bands of frequencies should coincide with the frequency bands on the graphic equalizer that is being used. The bands that show too high on the analyzer can be adjusted lower on the graphic equalizer. This process can be repeated until the sound in the room appears flat on the screen of the analyzer. Please note that you are measuring the combination of direct sound from the speakers and reflected sound (reverberation) from bounding around the room. The result sometimes is not what you expect. Make sure your ears confirm that the equalized sound is better than the non-equalized sound.

The first rule of equalization is not to use any equalization unless the sound system is of a very high quality. This means using professional speakers, horns, amps, crossovers, etc., that are correctly assembled and wired. Also this means that the speakers are positioned in the room in the proper relationship to the performers and audience. Adding an equalizer to a poor system will not solve any problems.

As for the pink-noise generator in your synthesizer, check it out with a

# IS YOUR TAPE ERASING ITSELF EVERY TIME YOU PLAY IT?

If lately your favorite recordings sound like they're gradually unrecording, it could be the tape they're on.

You see the oxide particles on some tapes just aren't bound on very well. And when the oxide particles come off, your music could come off sounding faded and weak.

Maxell, however, has developed a unique binding process that helps stop those oxide particles from taking a hike. We also polish our tape to a mirror finish to reduce friction, the major cause of oxide shedding.

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real time analyzer to see if (a) it is flat, (b) it has the proper bandwidth, and (c) it is long term stable. If it is, then use it.

There are many manufacturers that make audio test gear. Here are a few names: B & K Instruments, Inc., 5111 West 164th St., Cleveland, Ohio 44142; Communications Co., Inc., 3490 Noell St., San Diego, Ca. 92110; Inovonics, Inc., 503-B Vandell Way, Campbell, Ca. 95008; Ivie Electronics, Inc., 500 West 1200 South, Orem, Utah 84057; Neptune Electronics, Inc., 934 N.E. 25th, Portland, Oregon 97232; UREI, 8460 San Fernando Rd., Sun Valley, Ca. 91352; and White Instruments, Inc., P.O. Box 698, Austin, Texas 78767.

—James A. Ford
Technical Editor
*Modern Recording & Music*

## I'm Just a Singer in a Rock 'n' Roll Band . . .

**I am a singer in a rock band and we have a problem: My voice is very strong and sometimes it overdrives our P.A. I own a dbx 160 compressor/limiter and when we hooked it up (in the effects buss, I recall), the snare drum was setting it off and squashing the whole band sound. We then put it into the individual effects send/receive buss in my channel, and it worked fine through the main P.A., but it greatly muffled my monitor while everyone else's monitors remained fine. The board we are using is a Tapco 6100 RB/EB. Do you have any suggestions for solving this problem?**

**If I build a patch bay for a small recording studio, can it be mounted in a wooden case as long as the front panel with the jacks is grounded?**

**—David Bruno**
**Belmont, Mass.**

Your problem is essentially one of controlling vocal levels. You were right when you said that your vocal caused overload problems. The human voice has the widest dynamic range of any of the sounds that have to go through your P.A. When we mix records, the vocal is often the only thing that is compressed or limited. As you discovered, a compressor used in the overall mix has to work on everything. It's better to compress only what needs help. This way, the band can then be brought up to the vocal.

Your Tapco 6100 RB/EB board could

# COMPOSERS/PRODUCERS . . .


For the composer or the producer, for the owner of the largest state-of-the-art 24 track automated studio or the owner of the smallest home studio, the Prophet 5 synthesizer from Sequential Circuits is an excellent instrumental investment, for both total sound capability and creativity. Nearly every major keyboard player (not to mention guitarists, bass guitarists, saxophonists, etc.) has discovered the incredible range and quality of sounds produced by the Prophet 5. It has replaced dozens of other keyboards and instruments in the studio and on stage, and does so with amazing ease of use; you don't have to be a technician to use it. This is why, for many musicians, the Prophet 5 was their first synthesizer.

Up to 40 different sounds can be programmed for instant recall at the simple touch of a button. The Prophet 5 comes pre-programmed with superb strings, brass, harpsichord, flutes, organs, other keyboard and orchestral sounds, as well as sound effects such as wind, helicopters, bells, etc., etc., etc. Any of these sounds can be slightly changed (longer attacks, brighter, less bright) to match your personal tastes and needs. Or, a completely original and different sound of your own creation can be programmed easily into the Prophet 5. The choice is yours.

The ability to play 5 notes simultaneously sets the Prophet apart from most other synthesizers. For example, you can play a whole brass section in one take, rather than hassling with tedious overdubbing. You can compare how a straight melody line will sound against various chord structures or bass lines.

There are dozens of additional reasons why you or your studio should own a Prophet 5. Check it out — Play one. Then you'll know what we mean.

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be causing the overload problem before you can do anything about it. If you will look at the block diagram of the 6100, note that the mic pad and variable gain are after the input transformer and mic preamplifier. If you use a high output mic, and sing loudly and very close, it's possible to overload either, or both, the input transformer and preamp. The fix for input overload is a pad between the mic and the input. Several companies make a pad that will plug into the mic cable at the console.

The best place for you to compress would be after the mic preamp and before the level control for that input. The problem with the 6100 is that the channel patching is after everything in the input channel. Don't give up though. If the compressor is patched into the channel patching point, it will work—if you are careful and understand the signal flow in the 6100. As to how to set it all up, look at the circuit function diagram for the Tapco 6100. Starting at the low-Z mic input, note that just after the input transformer and preamp, a feed is tapped off to the monitor mix section. This means that the sound to the monitor send will: 1) not be effected by the mic pad or gain control, 2) not be EQ'ed, and 3) not be compressed. If the stage monitors are being fed by the monitor send, then the vocal will not be compressed and you will be able to work your mic to control dynamics. The main P.A. feed will be compressed, EQ'ed, echo'ed, etc.

Make up a special cable for the compressor. The channel patching jack is a stereo type. The tip is connected to the compressor's input; the ring to its output. The shield is grounded to the case.

When the compressor's cable is ready, don't plug it in just yet. Set the mic level to read "0 VU" or the maximum level used. Now plug in the compressor. Adjust the controls on the compressor for an average of no more than 3 to 6 dB of gain reduction on your loudest vocal peaks. Then set the compressor's output level to give the same "0 VU" level you had without the compressor in the circuit.

Remember, use as little compression as you can. Too much will squash the sound, and when you aren't singing the mic level will go up and cause feedback. If there is too much compression, the echo will get louder but the vocal won't. The same thing will happen in the stage monitors.

As for your question concerning the patch bay you're building, the jacks can be mounted in any kind of case you want. But!!!! If the system is unbalanced, as it is in the Tapco 6100 and most semi-pro equipment, the shields cannot be connected together at the patch bay. If they are, ground loops will cause hum problems. It would probably be best to mount the jacks on a plastic plate to insulate the grounds from each other. Only when a patch cord is plugged in will the ground be made between the input and output of these two pieces of equipment. Remember that ground loops are always the biggest problem with unbalanced equipment.

—Ron Malo
The Total Concept Sound
Burbank, Ca.

# '58 was a very good year...

# The '80's will be even better!

The PL80 is going to be the hottest vocal microphone of the '80's. It is the microphone every vocalist wants because it has the sound every vocalist wants. The sound of the PL80 results not only from extensive user field testing with rock superstars like Steve Perry of Journey, but also from side-by-side product comparisons and interviews with many of the most highly respected sound men in the business. Most of all, the PL80 is the result of an entirely new application of computer-design technology called "fast Fourier transform" that allows the design engineer to predict, as it's being designed, precisely how a microphone will sound in use, not just in a sterile test environment. The PL80 is a performing vocalist's microphone that has been called the best new microphone design in years.

The Electro-Voice PL80 not only gives you the exact sound you want, it does a whole lot more. The PL80 tops the competition in just about every performance category. Its style sets it apart from any other mike. Its sensitivity is higher than the current best-selling microphone; and when it comes to gain before feedback, the monitor speakers are likely to give out before the PL80 will.

Even the "feel" of the PL80 is impressive! It has the weight and the "heft" to give you confidence. The new snow-gray finish and contrasting charcoal gray grille screen make a striking impression on stage, but the colors are subtle enough not to detract from your performance. E-V's exclusive Memraflex grille material resists the dents and knocks common to other microphones. This will keep the PL80 looking like new for years while other mikes look old after one or two accidental drops.

Use the PL80 at your Electro-Voice PL Microphone Dealer. Test it against any other mike. If you want your sound to be the sound of the '80's, the PL80 – *the* "Sound of the '80's" – is the only mike you'll buy.

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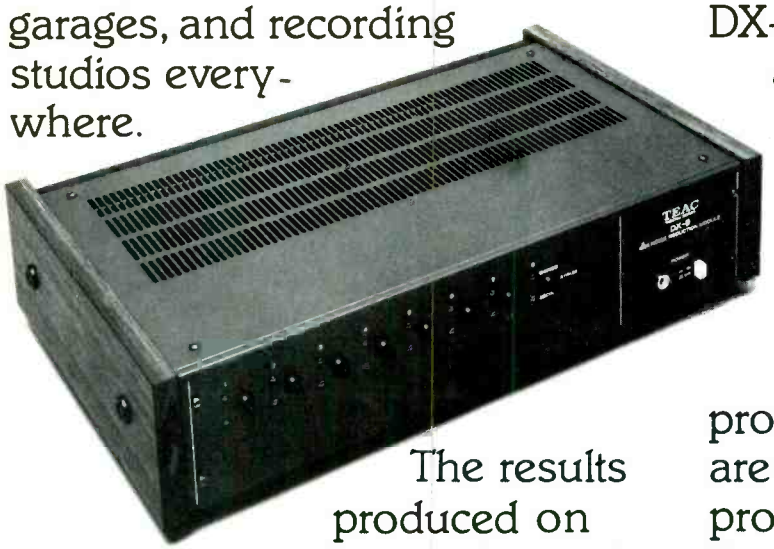
# THE MACHINE THAT HOLDS THE WORLD TRACK RECORD.

The Tascam Series 80-8 has become the most popular 8-track multichannel recorder in the world. Its reliability has been proven in basements, garages, and recording studios everywhere.



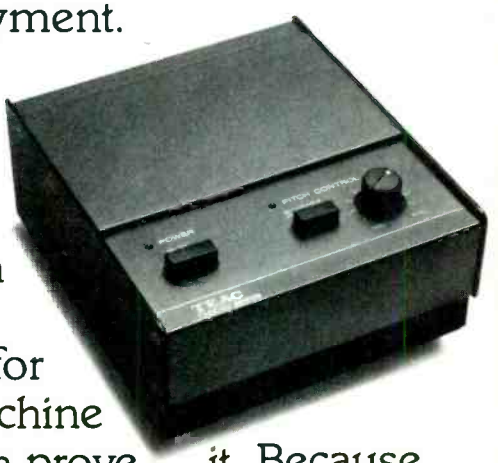
The results produced on the 80-8 are a matter of record. Sometimes gold.

The 80-8 proved a new standard was needed. Eight tracks on half-inch tape. 15 ips only. This new format allowed us to create a combined record/reproduce head, with full frequency response in the sync mode.

The 80-8 proved multichannel recorders could be relatively easy to operate. Our Function Select buttons determine the record, monitoring and dbx* status. One button for each track.

The 80-8 proved that performance and versatility could be affordable. Signal-to-noise is better than 95 dB (weighted) with our integral dbx unit (Model DX-8). Once installed, it's totally automatic. And our new Variable Speed Control**lets you adjust 15 ips ±20% to solve tough cueing and timing problems or add creative effects.

The 80-8 is proving that in professional recording, results are all that count. Because to us, pro means results. On demand. For payment.



If you agree, see your Tascam Series dealer for the machine that can prove it. Because it makes sense to do business with the people who have the track record.

*Registered trade mark of dbx, Inc.
**Installation required; a new DC servo-controlled motor is included.

# TASCAM SERIES

TEAC Professional Products



CIRCLE 125 ON READER SERVICE CARD

# THE PRODUCT SCENE

By Norman Eisenberg

## E-V OFFERINGS

Among the products from Electro-Voice is a new dynamic cardioid vocal microphone, the PL80. Designed specifically for the professional vocalist, the PL80 features a shock mount to reduce handling noise plus a built-in Acoustifoam blast filter to reduce "P-popping." Says E-V, the mic was computer designed so that E-V engineers could predict how it would sound in a "live" environment.

E-V also is offering the model PL91A dynamic cardioid mic, a refined version of the former PL91, for hand-held vocal applications. Two condenser cardioid vocal mics also have been announced. The PL76A and the PL77A have virtually identical specs except for the bass-contour switch included in the latter.

The model XEQ-1A is an electronic crossover for single-channel applications. Crossover range, determined by module, is from 100 Hz to 8 kHz. Two XEQ-1As may be stacked for tri-amping.

In speakers, E-V is offering, among others, its Model LF215 low-frequency system and a series of encased high-frequency horn/driver systems—the RC40A, RC60A, RC90A and RC120A.

CIRCLE 1 ON READER SERVICE CARD

## NEW PRO LINE FROM TOA

TOA Electronics offers a variety of equipment for various commercial and professional applications. Included are mixers, amplifiers, drivers and sectoral horns for speaker systems, etc. Two recent mixing consoles are the RX-5 and the RX-6. The RX-5 is an 8-channel input, 2 program output and 2 monitor output mixer. Each channel input is transformer-isolated and will accept a low-impedance microphone. Each channel has an input level selector and gain control, which also provides an additional 20 dB range of gain adjustment. Channels 7 and 8 also function as balanced, high-impedance inputs for high-level sources. Two meters may be switched to show program or monitor output levels. Sub-input and sub-output jacks enable cascade connection with additional mixers to expand the mic inputs. Outputs to and from echo are provided, and echo return volume is adjustable. Both the program and monitor outputs may be heard via headphones, and each input channel has a cue switch for audition of pre-fader signals through headphones.

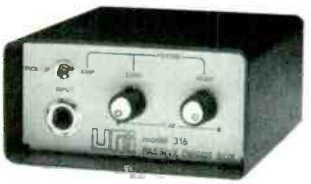
The RX-6 is generally similar, but provides 12 input channels. This console also has a rear panel switch for each input channel's phantom powering 48 VDC circuit to enable the use of a condenser microphone. In the RX-6, channels 11 and 12 may be used as the balanced, high-Z inputs. The RX-6 also features four LED peak meters.

CIRCLE 2 ON READER SERVICE CARD

## FOUR MODE ELECTRONIC CROSSOVER

Four panel selectable operating modes are featured in the UREI model 525 electronic crossover: stereo 2-way or 3-way, and mono 4-way or 5-way. Crossover frequencies are continuously adjustable from 50 Hz to 10 kHz, with the actual frequency measured and displayed on a digital counter. A subsonic filter is switchable from the rear panel. Inputs and outputs are XLR/QG connectors or they are terminal strips.

UREI also is offering two new direct boxes, a passive model 315 and an active model 325.



CIRCLE 3 ON READER SERVICE CARD

## COMPUTERIZED EQ/ANALYZER

From dbx comes news of the model 20/20, a computerized combination equalizer/analyzer. The device can analyze up to ten locations and then automatically adjust equalization for flat response. Digitally controlled, the 20/20 includes a companion microphone and a built-in pink noise generator. It also will store and recall the various EQ settings. Its real-time analyzer function also serves as a sound-level meter and provides continuous information on the amplitude for each frequency band via a display utilizing 300 LEDs. The given EQ curve stored in its memory also can be displayed at any time. Price is $1295.

CIRCLE 4 ON READER SERVICE CARD

## BGW'S BIGGEST AMPLIFIER

Described as "its largest amplifier ever" is the Model 1250 from BGW Systems. It is rated to deliver 400 watts RMS into 8 ohms in stereo service with no more than 0.03 percent distortion, or 1200 watts mono at no more than 0.05 percent distortion. Using only full complementary circuitry, the amp employs a total of forty-eight power transistors and is cooled with a three-speed fan. A multi-colored LED metering system shows clipping, temperature and mode. Gain controls are step-attenuator knobs calibrated directly in decibels. Both XLR-type and ¼-inch phone input connectors are provided. Construction is modular, and speaker protection is assured by arc-interrupting techniques. The amplifier's all-steel package is credited with lending both strength and RFI shielding to the unit.

CIRCLE 5 ON READER SERVICE CARD

## DELTALAB MEMORY MODULE

From DeltaLab Research comes word of its new "Memory Module," offered as a companion product to the company's DL-2 Acousticomputer and its new DL-4 Time Line. When interfaced with them,

the Memory Module allows the operator an additional two full seconds of delay with reportedly no degradation in performance. Memory Modules may be cascaded to obtain additional seconds of delay beyond that. The new unit is claimed to be capable of bringing another dimension of effects for studios and performing artists.

CIRCLE 6 ON READER SERVICE CARD

## EDEN SPEAKERS

A comprehensive line of pro speaker systems for concert and stage monitor applications is offered by Eden Electronics of Minneapolis, Minn. Over a dozen models are listed with varying size, power and response characteristics. Individual components as well as full systems are available.

CIRCLE 7 ON READER SERVICE CARD

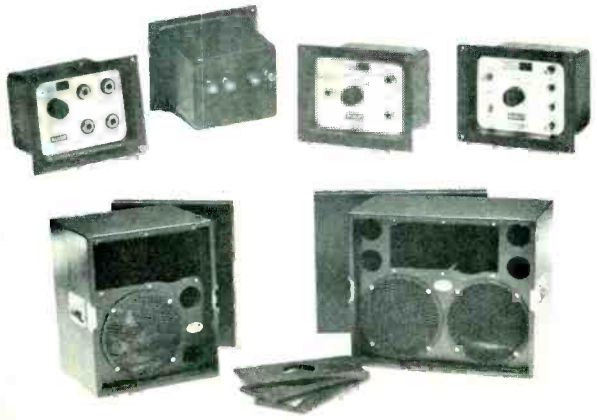
## COMPUTER CONTROLLED EQ SYSTEM

A computer-controlled multi-channel equalization system, known as Param, has been announced by a new firm, Redwood Research Inc. based in Nashville. Param consists of a computer that controls up to 128 individual equalizer channel modules. All operator input is via a single nine-by-six inch control panel. System operation is monitored by graphic display on a 12-inch video monitor which shows exact frequency response curves of each equalizer, and also displays data in table form as to the status of each equalizer, system memories and system status. Changes in EQ are made by drawing or altering the displayed response curve. The computer then optimizes the six filter sections, and sets the equalizer and displays the curve. All operations are in real time. System memories retain up to sixty-four settings for each equalizer, and thirty-two "standard" curves that the operator has previously stored. The system can advance from one memory setting to another in twenty milliseconds. An individual channel is set in 300 milliseconds. Interfaces to existing console level control automation are available.

CIRCLE 8 ON READER SERVICE CARD

## McCAULEY SPEAKER ELEMENTS

A comprehensive line of elements for speaker systems has been announced, and is described in a booklet by McCauley Sound Inc. of E. Puyallup, Wash. Included are radial horns, acoustical lens horns, exponential horns, frequency dividing networks, and speaker enclosures.



CIRCLE 9 ON READER SERVICE CARD

## METERS AND TEST GEAR

Detailed information on the Mura line of multitesters and panel meters is included in an 8-page catalogue available free from Mura Corporation.

CIRCLE 10 ON READER SERVICE CARD

## NEW ITEMS FROM PEAVEY

From Peavey Electronics comes word of several new items. In powered mixing consoles, there are the XR-800 (8 channel) and the SR-1200 (12 channel). Both versions feature balanced low impedance as well as unbalanced high impedance inputs, and individual pre in/out jacks on each channel. There are two independent pre-monitor sends, a post effects send, a PFL/cue button and active 3-band EQ on each channel. The master section has headphone monitoring, ten-position LED ladder displays for mains and monitors, dual nine-band graphic equalizers and a control that allows mixing the reverb and effects back into the monitor buss.

Peavey also has its Mark I Series 801 and 1201 stereo mixing consoles. In addition to the usual facilities and patch points, the master section of these units contains several unique features such as a "sum" output, the reverb and/or echo effects mix back into the monitor buses and PLF buttons on the main outs. Display is via four ten-segment LED arrays.

The EQ-27 is a 27-band equalizer with center frequencies spaced ⅓ octave apart across the audio range. Each band has its own slider with center detent. Since the selected frequencies conform to ISO standards, the EQ-27 is compatible with real-time analyzers and associated equipment.

A stereo graphic equalizer also is available, featuring two independent ten-band sections with 15 dB cut or boost at ten center frequencies. Inputs can be matched to a wide range of signal levels by means of gain/attenuator level controls. This device, because of a high-level transformer balanced output circuit, can provide greater than ±16 dBm into 600 ohms which makes it, says Peavey, "an excellent high-quality line amplifier."

Finally, there's the new ECM (electronic crossover mainframe) which enables the use of Peavey's own electronic crossover modules with other brands of power amps.

CIRCLE 11 ON READER SERVICE CARD

## FOSTER PRO PRODUCTS

A series of pro audio products from Foster Electric of Tokyo is described in a company booklet. Included are speakers for musical instrument and P.A. applications, microphones for vocal and musical instruments; two power amplifiers; and a powered mixer (six channels). The amps include the A-5, 75 watts per channel; and the A-6, a 150-watt mono unit.

CIRCLE 12 ON READER SERVICE CARD

## HAVE YOU EVER RUN INTO THIS ONE?

Now and then a puzzlement comes up in audio work that confounds even the experienced and knowledgeable. Here's one that occurred recently and had us guessing.

A young composer, Michael Krawitz, had an old recorded tape of a musical show—singers, instrumentals, dialogue—he had written some years back. The tape—quarter-inch and on a 7-inch reel—had been recorded in the standard consumer quarter-track format, with stereo in both directions. Trouble was, when he tried to play the tape on his own auto-reverse deck, the sound was backwards. He had brought the tape and his deck to the audio shop "Sounds of Music" in Lenox, Ma. where I rent work space from owner Ross Tane.

When I came into the place that morning, Tane and Mike Krawitz had already checked for the obvious (tails-out wind) and were into examining the heads on the deck and some other possibilities. Whatever they—and I, getting into the act—did, the sound was backwards.

Other quarter-track stereo tapes played on Mike's deck sounded right. And Mike's mystery tape, played on another deck, still sounded backwards.

At this point, Tane got involved with another customer. Mike and I went over the thing, step by step. I diagrammed the four tape tracks as they should be for this format. Counting down from the top edge of the tape, track 1 was left channel, and track 3 was right channel for normal forward (left to right) direction; track 2 was right channel, and track 4 was left channel for the reverse direction. The longer I stared at this diagram the more puzzled I was.

But something long forgotten was nagging inside. On a sudden inspiration I put a twist in the tape so that the brown shiny backing, rather than the dull oxide coating, would contact the heads. Tane saw this and said: "The heat's got you." Mike was too polite to say anything, but he looked on sadly and murmured something about "that beautiful slow song at the beginning of side 2 . . . "

We ran the tape—and hoorah—everything sounded right! But apparently the sound was coming off the backing rather than off the oxide coating. This didn't bother Mike, who left in a state of bliss with his reclaimed musical creation, but Tane and I were left with a mystery. Had the sound actually been recorded onto the backing? Could a tape be recorded that way? And in any event, can sound come off both sides of a tape? The only clue we had seemed to mock us: The leader had been attached correctly—that is to say, the printed side of the leader matched the shiny brown side.

"Worst case of print-through that I ever saw," said Tane.

"It can't be print-through," I said.

"Well then what is it?" Tane countered.

We decided to put the mystery on "hold."

Two days later that vague nagging persisted, and I telephoned Mike. I learned that the tape had been made in 1973. The vagueness began clearing.

"What brand tape is it?" I asked.

He replied: "I don't know; it's in a white box that says 1 mil."

Big insight at this point. Around 1973, 3M introduced "Posi-Trak" tape. It had an uncalendered backing whose dull black finish could easily fool you into assuming it was the oxide coated side. The same tape had a calendered shiny brown signal coating which could fool you into thinking it was the backing. Reportedly this tape (which since has been discontinued, thank you) did find its way into "white box" versions, one of which must have been our mystery tape. The recordist at the time obviously did record correctly, but then must have fooled himself and attached the leader to the wrong side. As for getting the sound from the wrong side, yes—it is not unknown for this to happen with 1-mil or thinner white-box tapes.

When I told this to Tane he feigned a mock-injury pose: "There goes my molecular derangement theory."

Anyway, the mystery is solved, and Mike can hear his own show and maybe interest someone in it via the tape.

Next problem?

# MUSICAL NEWSICALS

## SYNTHESIZERS AND ACCESSORIES

Oberheim Electronics has introduced two new polyphonic synthesizer models, a four-voice model and a six-voice model designated the OB-SX(4) and OB-SX(6), respectively. Both models use the same sound-generating circuitry as the popular OB-X synthesizer and thus feature the same "fat" Oberheim sound. The OB-SX series is considerably smaller and easier to use than the OB-X series since it does not have the user programmability of the OB-X. In design philosophy, the OB-SX is a polyphonic synthesizer for keyboardists whereas the OB-X was designed for musicians who are primarily synthesists. The OB-SX(4) may be updated to full six-voice polyphony by the user at any time simply by adding two more voice circuit boards. Among the features of the OB-SX series is a four-octave keyboard, 24/48 program option, pitch bend and modulation levers, portamento, transpose and attack and release controls. Rear panel interface connections are provided for filter and modulation panels, sustain footswitch, control voltage and gate inputs and outputs and Oberheim computer interface.

CIRCLE 13 ON READER SERVICE CARD

The latest instrument from Rocky Mount Industries, Inc. (RMI) is the DK-20, a digital combo keyboard which produces a wide variety of sounds including guitar, piano, lute and clav, plus custom factory voicings and presets. Twelve presets are available to the musician, plus sounds can be created with the digital envelope and timbre controls and pushbutton vibrato. The unit uses digital tone generation for a wider range of available sounds than the conventional oscillator/filter configuration. A unique feature of the system is Polyphonic Timbre Modulation which means that each note played will have independent, complex variations in its timbre throughout the envelope of the note. This feature gives each note clarity and individuality even during chordal playing.

CIRCLE 14 ON READER SERVICE CARD

A polyphonic synthesizer plus string synthesizer in one unit is the latest addition to the Korg line from Unicord. The new instrument is called the Korg DL-50 Delta, and features a four-octave keyboard with octave switching. The string section of the unit features two octaves of mixable voices, variable attack and release and bass and treble EQ to control the full, rich orchestral sounds available. The string section may be played separately or simultaneously with the synthesizer section, in mono or stereo. Controls on the synthesizer section include a 4-way joystick controller, ADSR envelope generator, waveform mixing, multi-mode 24 dB/octave VCF and jacks have been provided for external filter control and interface.

CIRCLE 15 ON READER SERVICE CARD

The Humanizer is the name of a unique new synthesizer controller from Computone, Inc. The Humanizer gives the synthesist an alternative to conventional keyboard, joystick or lever controller, or gives him the ability to control additional parameters spontaneously while playing a keyboard in a conventional fashion. The Humanizer comprises a control unit which looks rather like a kazoo and an electronics unit. The musician uses the controller in his mouth to control envelope parameters with wind pressure and pitch and timbre parameters with lip pressure. The controller may be handheld or mounted on a microphone stand to leave both hands free, and the electronics unit will interface to virtually any synthesizer.

CIRCLE 16 ON READER SERVICE CARD

Wavemakers has announced the Wavemaker 6 series of synthesizers and the Wavemaker 600 series of subsystems and accessories which are used to make up the various Wavemaker 6 configurations ranging from two to eight voices. The basic subsystem is the model 652 Dual-Voice Sub-system which contains all functions needed to generate and shape a sound. The unit includes two VCOs

each with two voltage-controlled waveform mixes, two multi-mode VC filter-amps, two ASR envelope generators with self-triggering and a wide-range FM modulator. The 654 Control Array is a multi-function module which includes an eight-step analog sequencer, sample and hold, two delayed ADSR envelope generators, a lag processor, a four-quadrant joystick which will attenuate external control voltages as well as generating four DC control voltages and four control busses for routing systems controls. Other 600 series sub-systems include the 658 Digital Sequencer with non-volatile memory of up to thirty-two sequences, and various keyboard controllers in two-, four-, six- and eight-voice configurations.

CIRCLE 17 ON READER SERVICE CARD

## KEYBOARDS

A new electronic piano, an 8-voice polyphonic synthesizer and a computer-controlled digital sequencer are among the new products by Roland-Corp US. The electronic piano is designated the EP-09, and uses digital generation to provide two piano and two harpsichord tones from its 5-octave, 61-note keyboard. The keyboard itself uses a microprocessor to assign one of the eight tone generator circuits to each key as it is touched regardless of how fast the musician cares to play. The unit features a sustain section which offers nine distinct variations on piano sustain for new expressive possibilities. Roland's new polyphonic synthesizer is the Jupiter 8, an 8-voice, 16-oscillator model with an advanced, computer-assigned keyboard which allows extensive flexibility in the assignment of voices to keys. The 5-octave keyboard may be operated in one of three Key Modes: Whole (one patch for the entire keyboard); Split (different patches for the lower two and upper three octaves); or Dual (two patches simultaneously). The Jupiter 8 also has 64 user-programmable memories to store all synthesizer functions including volume. Roland's new CSQ-600 Digital Sequencer is a computer-controlled unit with four memories, each capable of holding a 150-note sequence. The four sequences may be played back individually or in any combination one after another for extended sequences. Additionally, a unique Sync Load feature allows several CSQ-600s to be connected in parallel and run synchronously to control multiple synthesizers or a polyphonic synthesizer.

CIRCLE 18 ON READER SERVICE CARD

ARP Instruments, Inc. recently introduced three new instruments. Topping their new line-up is the computer-based Chroma, which we will cover in detail next month. The ARP Solus is a new, low-cost, lead-line synthesizer


which uses many of the same circuits as the well-known ARP Axxe and Odyssey synthesizers in a lower priced unit. Unlike many other synthesizers in this price category, the ARP Solus has a full three-octave keyboard which can additionally be transposed up or down one octave. Two VCOs are provided, each with mixable sawtooth and variable width pulse wave outputs, and they may be phase synchronized for effects. Filters are probably the single most important element in determining the sound of a synthesizer, and to produce a satisfyingly "fat" sound ARP uses four-pole filters in the Solus. The third new instrument from ARP is a scaled-down, four-voice version of the remarkable 16-voice electronic piano they introduced last year. The new, four-voice model retains many of the features of the larger model, but at a price which brings the instrument within the reach of many more musicians. The 73-note keyboard uses a weighted wood keyboard mechanism to actuate sealed membrane switches with full touch-sensitivity over a wide dynamic range. The four voices of the instrument are acoustic piano, vibes, electric piano and harpsichord. A dual tone generation and keying scheme is used to duplicate the decay characteristics of a natural piano, and the tuning of the second tone generator may be tuned exactly the same as the first generator, detuned slightly for a warmer sound or detuned further, up to a quarter of a semitone, for honky-tonk effects. The entire instrument may be tuned sharp or flat from concert tuning with the master tune control to bring the instrument in tune with other instruments or pre-recorded music. The ARP 4-voice piano also features a control pedal which can be used to control vibrato, sustain or both.

CIRCLE 19 ON READER SERVICE CARD

Rhodes Keyboard Instruments has announced the introduction of a new instrument which combines a traditional Rhodes electric piano with two electronic piano voices. The Rhodes Mark III EK-10 retains the same mechanical touch dynamics as the conventional Rhodes piano for both of its electronic voices as well as the regular electric piano voice, and offers the musician the ability to combine the three voices in any proportion or to use them individually. Each electronic voice is generated from a fundamental waveform plus second and fourth harmonics which are controlled by three selector switches for seven different selections of single or combined output waveforms. Each electronic voice has continuously variable filters for timbre shaping and a continuously variable tuning control with a range of plus or minus one octave relative to the tuning of the Rhodes piano allowing the musician to play the electronic voices in harmony to the basic Rhodes sound. Addi-

tionally, the keyboard may be split for a different mix of electronic and Rhodes sounds in the treble and bass sections of the keyboard. Physically, the Rhodes EK-10 resembles the Stage Model 73 piano complete with music rack and flat topped cover for instrument stacking. All controls are mounted on the name-plate rail for maximum convenience.

CIRCLE 20 ON READER SERVICE CARD

## GUITARS AND BASSES

The Fender division of CBS Musical Instruments recently introduced updated versions of two well-known instruments and a new Lead Bass guitar. The famous Fender Stratocaster has been updated and given a new name—it shall henceforth officially be known as the Fender Strat. Along with several other changes the new Strat offers four new, thicker sounds thanks to the use of a hot X-1 pickup in the bridge position in addition to the two high-output Stratocaster pickups. All three pickups are adjustable and are connected to a new, shielded, two-position mode selector switch and shielded volume and tone controls. The Fender Strat is available in Candy Apple red and Lake Placid blue with either maple or rosewood used for the curved fingerboard. The Fender Precision Bass also has been updated in the form of a new model, the Precision Bass Special. The P-Bass Special features active electronics on board for a wide range of tonal colorations; the instrument has bass and treble controls with a ±15 dB range, a volume control and a tone defeat switch. The electronics are assembled on a single PC board, and were designed for low noise and low current drain resulting in battery life averaging some 1200 hours. The neck of the P-Bass Special has been subtly re-designed for faster action and greater comfort; the neck is based on the 1961 B-neck, and is the standard width of all Precision Basses but is about 1/16" thinner. The all-new model from Fender is the Lead Bass which was designed for fast action and a wide tonal range to complement today's style of rock bass playing. The body of the Lead Bass is a new, compact, double-cutaway design and is made from solid ash, while the full-scale neck is narrow and light in weight to complement the lighter body. Electronically, the Lead Bass has two wide-range, wide-field humbucking pickups with eight adjustable pole pieces each, pickup selector switch, pickup phasing switch and master tone and volume controls, all of which have been electrostatically shielded.

CIRCLE 21 ON READER SERVICE CARD

## MUSICAL INSTRUMENT ACCESSORIES

Shadow of America has expanded its line of acoustic and electric guitar pickups with the introduction of their new Buckeroo Acoustic Humbucker line of magnetic type pickups for acoustic guitars. These new models are designed to mount easily in guitar sound holes via a spring clamp mount system, and produce a bold, powerful humbucker sound with superior frequency range without the need for accessory preamps. The Buckeroo line comprises three models starting with the basic Buckeroo 43 which has adjustable pole pieces on the pickup but which does not have volume or tone controls, and the Buckeroo 44 which is identical to the 43 except that it has volume and bass/treble controls. At the top of the line is the Buckeroo 46 Triple Threat unit which offers a four-position mode switch in addition to a volume control. The four modes of operation for the Model 46 are Standby, Bass-rich humbucker, Balanced-tone humbucker and Brilliant single-coil (non-humbucker) offering the musician a full range of electronic sounds to complement his acoustic sound.

CIRCLE 22 ON READER SERVICE CARD

Whirlwind Music, renowned among guitarists for its Cobra model coiled guitar cord and its Snake model straight cord, has announced the introduction of a new reptile, the Constrictor, which is a hybrid straight/coiled cord. The Constrictor combines ten feet of straight cord with a 20-inch section of coiled cord for an overall length of over 20 feet when fully extended, and gives the musician the freedom of movement of a straight cord with the neatness and long reach of a retractile cord. The cable used in the Constrictor is made by Belden as in all Whirlwind cables, while the terminations are a Whirlwind Ultra Snake ¼-inch plug on the straight end and a Switchcraft right-angle ¼-inch plug on the coiled end to reduce the possibility of accidental disconnections.

CIRCLE 23 ON READER SERVICE CARD

New from Silver Eagle Designs is the Doctor Song guitar strap which uses both top-grain cowhide and synthetic leather to provide a strong, comfortable strap at an affordable price. Leather is used at each attachment end for strength while the rest of the strap is man-made to keep the cost down. Doctor Song straps have no exposed metal parts to scratch valuable instruments, and feature an exclusive "pick-pocket" on each strap to hold picks, slides, etc. The straps are available in four widths from 2-inch to 3½-inch in black, tan and brown finishes.

CIRCLE 24 ON READER SERVICE CARD

News comes from Zeta-Systems of a marvel of electronic miniaturization called the little FEANC (pronounced "fink") which combines Fuzztone, Equalizer, Amplifier, Noisegate and Compressor into a single 1½" x 2" package designed to mount directly in a guitar in place of the guitar's conventional controls. The compressor part of the circuitry in the Little FEANC uses FET technology for wide-range, low-noise operation with infinitely variable compression, while the switchable

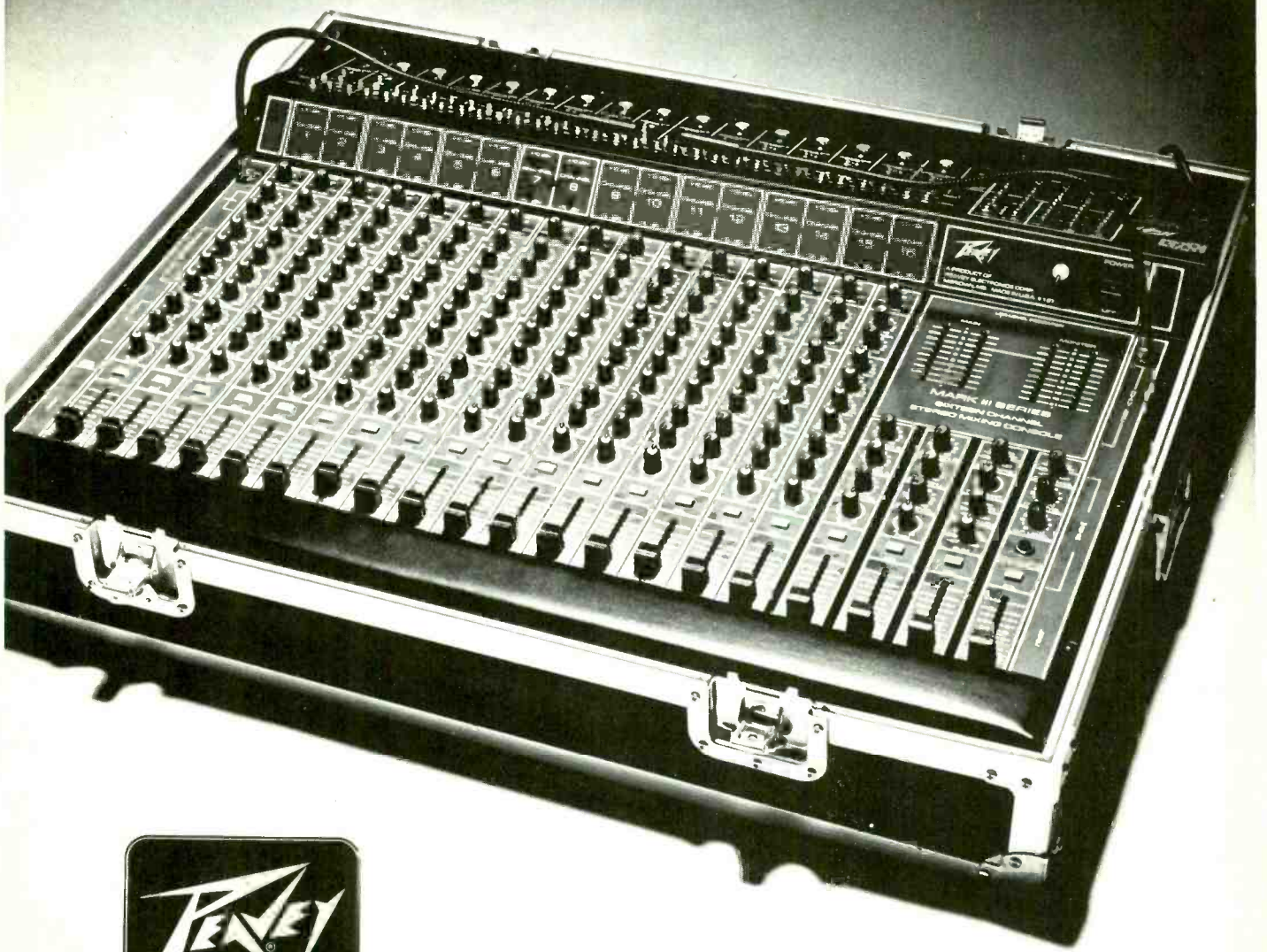
# "A Sweeter Sixteen"

Peavey's extensive experience in electronic design, packaging, and reliability studies along with the utilization of the very latest semiconductor technology have created the Mark III Series sixteen channel mixer,...just the right combination of superb performance and **useful** features in a highly mobile and durable sound reinforcement console.

The Mark III's transformer balanced inputs feature virtually infinite dynamic range and include switchable 48-volt phantom power for capacitor microphones. Two independent pre monitor sends, 4-band EQ, two post effects sends and a PFL/cue button compliment each channel.

The Mark III's master section includes all the usual professional facilities along with many features previously unavailable in sound reinforcement mixers. A complete patch panel features transformer balanced line outputs on both mains, both monitors, and sum.

We've listed just a few of the Mark III's many features to illustrate its performance,...to get the complete picture, you'll want a "hands on" demonstration at your Peavey dealer. You'll see again why incredible values are still a reality at Peavey.




*"Peavey Makes It Possible."*

**PEAVEY ELECTRONICS CORP.** 711 A Street/Meridian, MS 39301 © 1980

CIRCLE 89 ON READER SERVICE CARD

noisegate silences the output when no signal is present. A 2-band equalizer provides tonal control over the normal guitar sound or the fuzzed guitar sound before the master volume control and low-impedance output amplifier stage. Integrating these five popular effects in one one on-board unit has given the musician high quality signal processing with the controls where he needs them—at his fingertips.

CIRCLE 25 ON READER SERVICE CARD

Another reptilian product for guitarists is the King Cobra guitar stand from The Music People. The King Cobra is a striking design, made from clear lucite plastic (½-inch thick) in the general shape of a cobra's head. The King Cobra breaks down into two pieces for easy transportation and uses two nylon pins for vertical stability when assembled. The guitar is protected by neoprene rubber bumpers where it makes contact with the stand.

CIRCLE 26 ON READER SERVICE CARD

## MUSICAL INSTRUMENT AMPLIFIERS

Unicord has announced a new line of amplifiers to be known as Westbury Amplifiers which are designed to offer uncompromised sound value. Construction is extra rugged with 18-gauge rolled steel chassis, oversized heat sinks, current-limiting short circuit protection and modular PC board construction. The Model 1000 is the current flagship of the line, and is a 2" x 12" dual-channel reverb amp with two completely independent FET preamp input stages which can be instantly switched for two different preset sounds.

CIRCLE 27 ON READER SERVICE CARD

Road Electronics has introduced a new line of self-contained amplifiers which feature advanced electronic technology and construction techniques to achieve a high level of versatility at an affordable price. Two of the models are single-channel units with remote switching via an optional footswitch and LED channel indication. All four models include three-band active EQ with Road's parascan sweep midrange EQ which is variable from 400 Hz to 3 kHz, reverb with remote footswitch jack and accessory send/receive jacks. All four are rated at 60 watts RMS, and have a headphone jack which automatically disconnects the main speakers. The four models include the L120 single channel lead amp and the SL120 switchable channel lead amp, both featuring 12-inch speakers, the B120 single channel bass amp with a heavy-duty 12-inch bass speaker, and the SB120 switchable-channel bass amp with a 15-inch speaker. Also new from Road is a unique speaker protection kit for use with any amplifier or speaker cabinet. The kit comprises the die cast aluminum speaker retaining ring, heavy gauge steel mesh, speaker mounting gaskets and necessary hardware to mount speakers in the same way that Road mounts its speakers.

CIRCLE 28 ON READER SERVICE CARD

## DRUMS AND ACCESSORIES

A company called Tonga has introduced an intresting drum accessory product called TongaRings. TongaRings are made from mirror-finish Mylar plastic and are attached to the drum heads in a drummer's kit with pressure sensitive adhesive. The rings are said to produce more tone and less bang from the drums without the use of bottom heads, complicated tunings, or trial and error damping of the head. Additionally, with proper tuning TongaRings allow the popular modulated pitch or downward-sliding pitch to be achieved acoustically rather than with synthesized drums. The rings leave the center of the drum head open for playing while damping out unwanted vibrations around the circumference of the head. TongaRings are available in diameters from 6-inch to 24-inch to accommodate the majority of drums commonly used.

CIRCLE 29 ON READER SERVICE CARD

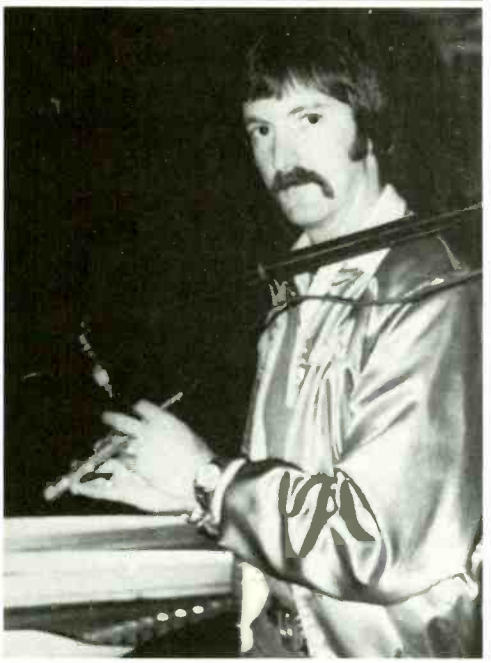
Star Instruments, Inc. has been one of the leaders in the synthesized drum field with its Synare electronic drum line, and with the introduction of four specialized new models the Synare line is certainly a contender for the most complete line-up of non-acoustic drums. The four new models are the Synare Bass, the Synare Lo-Tom, the Synare Hi-Tom and the Synare Tympani, and as their names imply they are each designed to simulate a particular type of drum rather than being a general-purpose or full-range synthetic drum like the company's Synare 4 or S3X Preset models. All four models use an 8-inch drum head which is responsive to the stick and all are housed in a matte black steel enclosure and AC powered. All four models have Decay, Sensitivity and Volume controls to control the loudness and dynamics of the output signal and a Tune control to vary the basic tuning of the sound. The Synare Bass has only one additional control, a switch to select short, medium or long bass drum sounds or to select double, triple or repeated strike sounds from a single stick or beater hit. The Synare Bass may be mounted vertically with a conventional bass drum beater on a Synare BD-2 stand, or mounted as a tom-tom stand for stick playing. The Lo-Tom and Hi-Tom Synare models are identical except for the tuning range; the Lo-Tom simulates drums of approximately 14-inch to 18-inch in diameter while the Hi-Tom tunes to cover the range of 5½-inch to 16-inch acoustic tom-toms. In addition to the basic four controls the Synare Tom models include three controls for Range, Speed and Direction of automatic tuning runs which vary the tuning of the Synare continuously with each successive strike. The Synare Tympani can be tuned via an optional foot pedal as well as the control panel knob for tympani sounds without the temperature and humidity sensitivities and large bulk of acoustic tympani. Additionally, the Synare Tympani has up-sweep and down-sweep controls for automatic pitch sweeps, and these effects can be further modified or "bent" by the footpedal.

CIRCLE 30 ON READER SERVICE CARD

With the prices of labor and brass being what they are, it is not surprising that cymbals can actually be the biggest investment a drummer will make. Drums Unlimited, Inc. has just introduced a new line of cymbals which are student priced but professional sounding. The Studio 200 line of cymbals is manufactured in virtually the same way as the world's leading professional cymbals except for a small amount of hand finishing. The result is said to be a cymbal which has the same crisp, explosive sound and feel of expensive, professional cymbals with only a small sacrifice in ultimate volume potential, but at a price that is approximately half of the pro brands.

CIRCLE 31 ON READER SERVICE CARD

# fact:
# "I listened to them all... and nine times out of ten, with our artists, the best microphone was the SM81"



## *Criteria Recording Studios, Miami, Florida*

*Dennis Hetzendorfer*

**Dennis Hetzendorfer,**
Staff Engineer

"The true sign of a really excellent microphone is that it can *maintain* its high performance, session after session after session. Here at Criteria, when the situation permits, several different microphones are set-up at each instrument, without the engineer knowing which mike is exactly where. We then fade from mike to mike and let our ears find out which is best for each application. Nine times out of ten, with our artists, the best microphone has been the SM81.

"The switchable bass rolloff and pad (a built-in 10 dB attenuator) gives the SM81 incredible versatility. We can use it with bass drums and cymbals, as well as with acoustic guitars. In fact, all the acoustic guitar segments on the Bee Gees' *Spirits Having Flown* album were recorded with the SM81.

"The SM81 really changed our minds about the ruggedness of condenser microphones. It's a precision piece of equipment, but it's durable. You don't always think about a studio microphone needing durability...after all, we don't have the rough handling problems encountered in concert recording. But, when you have a reputation as one of the most technically exacting studios in the country, you appreciate how *many* little things can subtly affect the sound of a delicate condenser microphone. The SM81 sounds good *every* time we use it...and, at Criteria, as in any good studio, we just can't afford to have a microphone we can't depend on.

"We've used the SM81 on recording sessions with the Bee Gees and Kenny Loggins and you can be sure there will be more.

"It's one great mike!"


**Send for our brochure.**

## SM81 Cardioid Condenser Microphone by

® SHURE ®

Shure Brothers Inc., 222 Hartrey Ave., Evanston, IL 60204. In Canada: A. C. Simmonds & Sons Limited.
Outside the U.S. or Canada, write to Shure Brothers Inc., Attn: Dept. J6 for information on your local Shure distributor.
Manufacturers of high fidelity components, microphones, sound systems and related circuitry.

CIRCLE 118 ON READER SERVICE CARD

# The ELECTRIC ENGINEER

## Part IX

By Peter Weiss

In this installment we will continue our discussion of A.C. circuits and expand on this information to cover some practical aspects of electrical circuits.

So far, we have discussed simple A.C. circuits containing either resistance and capacitance ("RC" circuits), or resistance and inductance ("RL" circuits). Now, with the aid of the vector methods developed earlier, we will tackle circuits containing resistance, capacitance *and* inductance ("LCR" circuits). An example of an LCR circuit is shown in *Fig. 1*, with the elements in a simple series configuration. The same general rules apply to the analysis of this circuit as to that of previously discussed A.C. series circuits; the current is the same at all points, and in-phase with itself, and the *vector sum* of the voltage drops around the circuit must equal the applied voltage. With this information and a set of values for f ($1.00 \times 10^3$ Hz), L ($2.00 \times 10^{-1}$ henrys), C ($8.00 \times 10^{-8}$ farads) and R ($7.34 \times 10^2$ Ω), we can do an analysis of the circuit of *Fig. 1*. First, let's calculate the inductive reactance $X_L$.

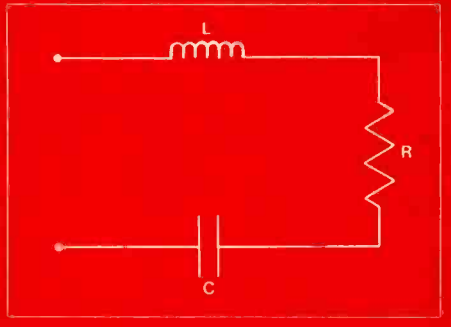



Figure 1

$$X_L = 2\pi \times f \times L$$

$$X_L = 6.28 \times 1.0 \times 10^3 \times 2.0 \times 10^{-1}$$

$$X_L = 1.256 \times 10^3 \text{ or } 1256\ \Omega$$

Now for the capacitive reactance, $X_C$,

$$X_C = \frac{1}{2\pi \times f \times C}$$

$$X_C = \frac{1}{6.28 \times 1.0 \times 10^3 \times 8.0 \times 10^{-8}}$$

$$X_C = 1.990 \times 10^3 \text{ or } 1990\ \Omega$$

To find the total impedance of this circuit, Z, we can draw the vectors $X_L$, $X_C$ and R as shown in *Fig. 2*. In past A.C. circuit analyses we have only had two vectors at a time, not three. However, we can reduce the three vectors in *Fig. 2* to only two by noting that the $X_L$ and $X_C$ vectors are "pointing" in opposite directions. Taking advantage of this, we can redraw *Fig. 2* (as *Fig. 3*) with the vectors $(X_C - X_L)$ and R. The magnitude (length) of the $(X_C - X_L)$ vector is simply:

$$X_C - X_L$$

or

$$1990 - 1256$$

or

$$734\ \Omega$$

(*NOTE:* If $X_L$ turned out to be greater than $X_C$, the combined vector would then be $(X_L - X_C)$, pointing in the $X_L$ direction.)

We can calculate Z from the relationship

$$Z = \sqrt{(X_C - X_L)^2 + R^2}$$

$$Z = \sqrt{734^2 + 734^2}$$

$$Z = \sqrt{1{,}077{,}512}$$

$$X = 1.038 \times 10^3 \text{ or } 1038\ \Omega$$

The phase angle, $\theta$, can be found from

$$\tan \theta = \frac{(X_C - X_L)}{R}$$

$$\tan \theta = \frac{7.34 \times 10^2}{7.34 \times 10^2}$$

$$\tan \theta = 1$$

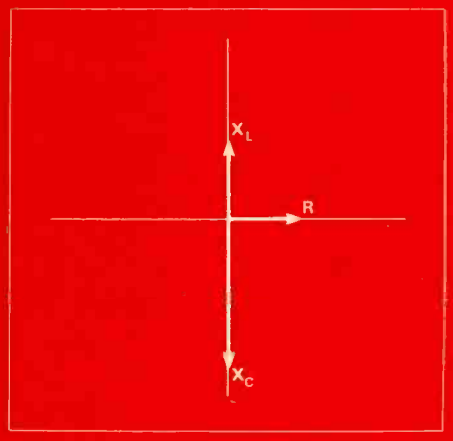



Figure 2

Using a calculator or "trig" tables we find that the angle whose tangent is 1 is 45°. This angle represents the phase difference between the current through the circuit and the applied voltage. The phase angle, and, more specifically, whether the angle ends up being above or below the R axis, indicates the nature (either capacitive, inductive or purely resistive) of the circuit *at a specific frequency* as "seen" by the applied signal. In our example case, at 1000 Hz, $X_C$ is greater than $X_L$ and the position of the $(X_C - X_L)$ vector indicates that the net result of the addition of these two opposing vectors is a vector laid out in the $X_C$ direction. This indicates that, at the signal frequency given, the capacitance is contributing more to the "character" of the circuit than is the inductance. The final resolution of the $(X_C - X_L)$ and R vectors into the Z vector at 45° in the $X_C$ and R quadrant shows that the whole circuit has a somewhat capacitive character when the signal frequency is 100 Hz. If the signal frequency is increased, $\theta$ will approach zero, and at the frequency for which $X_C = X_L$, making the $(X_C - X_L)$ vector zero, $\theta$ will *be* zero and the circuit will "appear" to be a pure resistance. The frequency at which this condition occurs is called the "resonant frequency" of the circuit. The symbol for resonant frequency is $f_o$. We can find the value of this frequency for a given circuit by setting the formulas for $X_C$ and $X_L$ equal to each other, since this is the circuit condition at the resonant frequency.

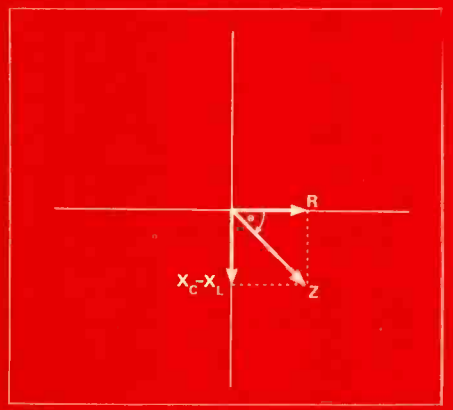



Figure 3

$$\frac{1}{2\pi \times f_o \times C} = 2\pi \times f_o \times L$$

$$1 = 4\pi^2 \times f_o^2 \times L \times C$$

$$f_o^2 = \frac{1}{4\pi^2 \times L \times C}$$

$$f_o = \frac{1}{2\pi \times \sqrt{L \times C}}$$

Substituting the values given earlier,

$$f_o = \frac{1}{6.28 \times \sqrt{2.0 \times 10^{-1} \times 8.0 \times 10^{-8}}}$$

$$f_o = 1.259 \times 10^3 \text{ or } 1259 \text{ Hz}$$

Readers can confirm this by calculating $X_C$ and $X_L$ at 1259 Hz to see if the reactances are in fact equal.

Since the length of the $(X_C - X_L)$ vector is zero at the resonant frequency, the phase angle $\theta$ is zero and the Z vector coincides with the R vector. These conditions indicate, mathematically, that at the resonant frequency the capacitive and inductive reactances effectively cancel each other and the circuit "behaves" as if it contained only a pure resistance. If there is no resistance in the circuit, then Z will be zero. There will be nothing to impede the flow of current.

When the signal frequency is increased above $f_o$, the inductive reactance becomes greater than the capacitive reactance. The length of the $X_L$ vector becomes greater than the length of the $X_C$ vector, and we must draw a vector, pointing in the $X_L$ direction, whose length is $(X_L - X_C)$. At frequencies above $f_o$ the circuit will "behave" more inductively.

We can now go on to examine another simple A.C. circuit configuration, the parallel LCR circuit. An example of this type of circuit is shown in *Fig. 4*. We have to establish a new set of values for this example so our final numbers come out nice and neat; $E_s$ = 100 V, f = 1000 Hz, L = 8.0 x $10^{-3}$ henrys, C = 1.59 x $10^{-6}$ farads, R = 100 Ω. The formulas for capacitive and inductive reactance are the same regardless of circuit configuration, so we will just state the results of computing these values: $X_L$ = 50 Ω, $X_C$ = 100 Ω.

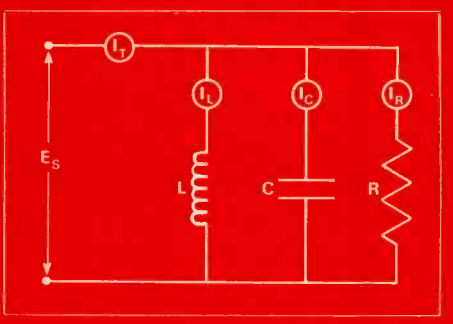



Figure 4

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The rules for analysis of parallel LCR circuits are based on information developed earlier in our discussions of D.C. parallel circuits and on the vector methods we have used in previous A.C. circuit analyses. To begin, we can say that the voltage across each circuit element is equal to the applied voltage, $E_s$, *and is in-phase with it.* However, we know that capacitances and inductances introduce phase differences between the currents through them and the voltages or voltage drops across them. Thus, the voltage and current in the branch containing the capacitance, C, are 90° out-of-phase, with the current "leading" the voltage. In the branch containing the inductance, L, a 90° phase shift also occurs, but here the current "lags behind" the voltage. In the resistive branch, of course, the voltage and current are in-phase with each other. It is important to note again that all of the voltages just mentioned are actually the same voltage, and that they are all in-phase with each other. The currents are not in-phase with each other, and by determining the phase relationships of these currents, we can arrive at a simple way of determining the total impedance, Z, of the example circuit.

First, let us establish the signal voltage as the phase reference. Then, the current in the resistive branch, $I_R$, is in-phase with this voltage, the current in the capacitive branch, $I_C$, is 90° "ahead," and the current in the inductive branch, $I_L$, is 90° "behind." Thus, we can see that $I_C$ and $I_L$ are actually

180°, or completely out-of-phase with each other. Before going any further, we should calculate the magnitudes of each of the three currents.

$$I_R = \frac{E_S}{R}$$

$$I_R = \frac{100}{100}$$

$$I_R = 1.0 \text{ amps}$$

$$I_C = \frac{E_S}{X_C}$$

$$I_C = \frac{100}{100}$$

$$I_C = 1.0 \text{ amp}$$

$$I_L = \frac{E_S}{X_L}$$

$$I_L = \frac{100}{50}$$

$$I_L = 2.0 \text{ amps}$$

The individual branch currents combine to form the total current in the circuit, $I_T$, but they do so *vectorially*. We can draw a vector diagram, *Fig. 5*, to represent the relationships between the three currents. Note that the $I_C$ and $I_L$ vectors are opposing each other. This is a graphical representation of the fact that these two currents are 180° out-of-phase with each other. The $I_R$ vector is laid out along the 0° axis to indicate that the current in the resistive branch is in-phase with the applied voltage. As we did with the $X_C$ and $X_L$ vectors in the series LCR circuit, we can resolve the $I_C$ and $I_L$ vectors into a single vector with length $(I_L - I_C)$, pointing in the $I_L$ direction (since $I_L$ is greater than $I_C$). The length of this new vector is 1, representing the difference between $I_L$ and $I_C$. The $(I_L - I_C)$, $I_R$ and $I_T$ vectors are shown in *Fig. 6*. We can calculate the length of the $I_T$ vector,

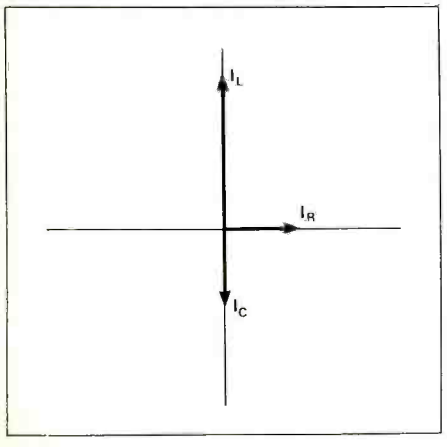


Figure 5

$$I_T = \sqrt{(I_L - I_C)^2 + I_R^2}$$

$$I_T = \sqrt{1^2 + 1^2}$$

$$I_T = \sqrt{2}$$

$$I_T = 1.41$$

We now calculate Z from Ohm's Law,

$$Z = \frac{E_S}{I_T}$$

$$Z = \frac{100}{1.41}$$

$$Z = 70.9\ \Omega$$

Since, in this case, the inductive current is greater than the capacitive current, this circuit (at 1000 Hz) "behaves" more or less inductively, with a resistive component. The phase angle, $\theta$, is 45°, as can be seen from the fact that $(I_L - I_C) = I_R$.

It will be instructive to point out that there is a frequency at which $I_L = I_C$, making the length of the $(I_L - I_C)$ vector zero. At this same frequency, $X_L = X_C$. The value of this frequency can be computed in a way identical to that used for the resonant frequency of a series LCR circuit. The symbol $f_o$ is used here also, except that there is some confusion of terms. A parallel LCR circuit is sometimes called an "anti-resonant circuit," but the frequency at which $I_L = I_C$ is called the "resonant frequency." Another name for a parallel LCR circuit is "tank" circuit. We will use the terms "resonant frequency" and "parallel LCR." The formula for $f_o$, the resonant frequency of a parallel LCR circuit, is identical to the formula for $f_o$ of a series LCR circuit, since the conditions required are the same, namely that at $f_o$, $X_L = X_C$. Let's compute $f_o$ for the example parallel LCR circuit.

$$f_o = \frac{1}{2\pi \times \sqrt{L \times C}}$$

$$f_o = \frac{1}{6.28 \times \sqrt{8.0 \times 10^{-3} \times 1.59 \times 10^{-6}}}$$

$$f_o = 1.411 \times 10^3 \text{ or } 1411 \text{ Hz}$$

At $f_o$, our example parallel LCR circuit will behave as if it contained only a pure resistance of 50 Ω, since $X_L = X_C$, $I_L = I_C$ and the two reactive currents are 180° out-of-phase. An interesting

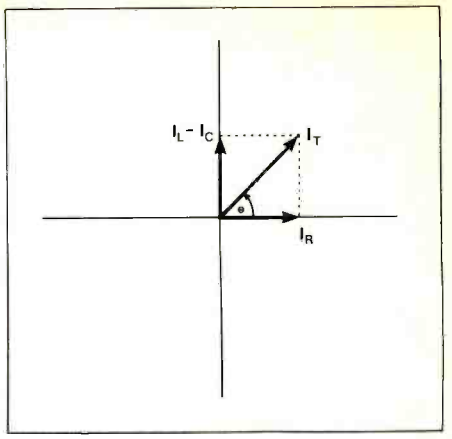


Figure 6

set of conditions arises if we remove the resistive branch altogether, letting $R = 0$, and $I_R = 0$. At $f_o$, the capacitive current and the inductive current will cancel each other, but since there is no resistive branch, the net effect will be *zero current flow* through the circuit. This does not mean that if we remove the resistive branch $I_L$ and $I_C$ will no longer exist. They *do* exist, individually, and are real, measureable currents, but the result of their combination is total cancellation. From the "point of view" of the source voltage, a parallel LC circuit at the resonant frequency represents an "infinite" impedance.

• • •

At this point, we must examine another important quantity in A.C. circuits, power. The unit for A.C. power is the watt, as in D.C. circuits, but the formula for finding the power dissipated by a component in an A.C. circuit or by an entire A.C. circuit, is not quite as simple as the D.C. version.

*Fig. 7* shows the circuit of *Fig. 1* drawn with voltmeters ($V_L$, $V_C$, $V_R$, $V_T$) and an ammeter ($I_T$) in place. We already know that in this circuit, for a

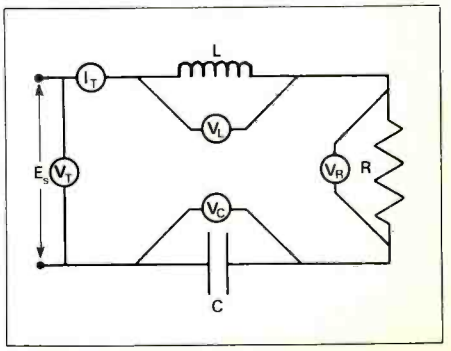


Figure 7

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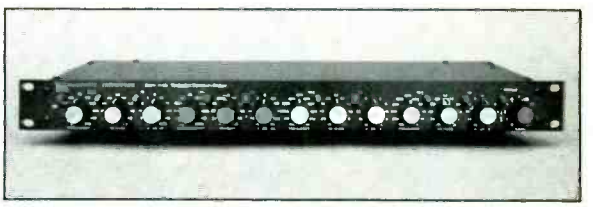

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signal frequency of 1000 Hz, Z = 1038 Ω and $\theta = 45$. We can compute $I_T$ from Ohm's Law,

$$I_T = \frac{E_S}{Z}$$

If we choose a convenient value for $E_S$, 103.8 volts, we can get a neat answer for $I_T$, namely,

$$I_T = \frac{103.8}{1038}$$

$$I_T = 1.0 \times 10^{-1} \text{ or } 0.10 \text{ amps}$$

Clearly, as we have set up the situation, the source is delivering 103.8 volts and a current of 0.10 amps. If we use one of the D.C. formulas for total power dissipation,

$$P_T = I_T \times E_S$$

$$P_T = 1.0 \times 10^{-1} \times 103.8$$

we will get

$$P_T = 10.38 \text{ watts.}$$

But is this value really correct? To answer this, we must examine how power is dissipated in each individual component in the circuit. Looking at the resistance first, we see that the current through the resistance, $I_T$, and the voltage drop across the resistance, $V_R$, are in-phase. This fact allows us to use the formula for power dissipation involving current and resistance,

$$P_R = I_T^2 \times R$$

$$P_R = (1.0 \times 10^{-1}) \times (1.0 \times 10^{-1}) \times 7.34 \times 10^2$$

$$P_R = 7.34 \text{ watts}$$

Now for the power dissipated in the capacitance. Since this is a theoretical discussion, the capacitance in this circuit, taken by itself, is a pure capacitance, having no resistive component at all. Thus, if we try to apply the formula

$$P_C = I_T^2 \times R$$

to the capacitance alone, since R = 0, $P_C$ must equal zero. Right? But how can the power dissipation be zero if there is a voltage drop across, and a current through, the capacitance? There is a clincher on this discussion. Remember, that when we initially discussed power dissipation, back in our D.C. circuit analyses, we stated that power dissipation was the result of *energy* being lost in a component or circuit. In a *pure* capacitance, *all the energy* that goes into establishing the electrical field between the plates is *returned to the circuit* when the source voltage alternates and the field collapses in preparation for a new field to be established in the opposite direction. In this repeating process, the net loss of energy in the (ideal) capacitance is *zero*. Therefore, the power (energy-per-unit-time) lost in the capacitance is *zero*. $P_C$ *is equal to zero at all times.*

A similar argument can be made for the inductance (also an ideal, zero-resistance component), where all the energy required to establish the magnetic field is returned to the circuit when the source voltage alternates and the field collapses. So,

$$P_L = O$$

If no power is dissipated in either the capacitance or the inductance, how do we reconcile

$$P_T = 10.38 \text{ watts}$$

and

$$P_R = 7.34 \text{ watts?}$$

We can resolve this question by pointing out that $P_T$ and $P_R$ represent two different sets of measurements. $P_T$ is the product of the ammeter and voltmeter readings $I_T$ and $V_T$, while $P_R$ is the product $I_T^2$ x R. The product of the total voltage and total current in an A.C. circuit is called the "apparent power." The total power dissipated in all of the resistive elements of an A.C. circuit is called the "true power." The ratio of the true power to the apparent power is called the "power factor" of the circuit. The power factor is also equal to the cosine (cos) of the phase angle between the applied voltage and the total current.

Setting up the ratio for the power factor, PF,

$$PF = \frac{P_R \text{ (true)}}{P_T \text{ (apparent)}}$$

$$PF = \frac{7.34}{10.38}$$

$$PF = 7.07 \times 10^{-1} \text{ or } .707$$

The phase angle in our example is 45° for a signal frequency of 1000 Hz, so

$$PF = \cos \theta$$

$$PF = \cos 45°$$

Finding cos 45° from a calculator or from the tables,

$$PF = .707$$

which agrees with our answer derived from the ratio of true to apparent power. Using the cos $\theta$ form of the power factor, we can develop formulas for true power dissipated in A.C. circuits that look like the D.C. power formulas with some modifications:

$$\text{True Power} = V_T \times I_T \times \cos \theta$$

$$= I_T^2 \times Z \times \cos \theta$$

Note that cos 0° = 1, and cos 90° = 0. These statements mathematically support the formulas for true power by indicating that for a purely resistive circuit ($\theta = 0°$) the power factor is 1 and the true and apparent powers are equal, and for a purely reactive circuit ($\theta = 90°$) the power factor is zero, no power being dissipated.

It is important to remember that impedance, individual reactances, true power and power factor are all *frequency-dependent.*

In practical applications, particularly in audio, we are concerned about how much work we can extract from an A.C. signal. Work is equivalent to energy, so what we are really talking about is useful power. An alternating voltage applied to a given pure resistance will heat the resistance and power will be dissipated in the resistance. However, a *direct voltage* equal to only 70.7% of the peak value of the alternating voltage will perform the *same* heating job. Therefore, none of the expressions we have seen so far indicate the actual working ability of an A.C. signal. We can correct this by applying a factor of 0.707 (70%) to the peak values of voltage and current. Using the values from the series LCR example circuit, and calling the available or effective power $P_E$,

$$P_E = 0.707 \times E_S \times 0.707 \times I_T$$

$$P_E = 73.39 \times 0.071$$

$$P_E = 5.19 \text{ watts}$$

This value for $P_E$ is exactly half the value of the apparent power.

In all future discussions, it will be assumed that voltages and currents are given as RMS values, and powers as effective values.

As for what these future discussions will cover, Part X of the "Electric Primer" will include some information and de-mystification of two very important areas, namely "grounding" and the deciBel (dB). With those two topics out of the way, we will then proceed to talk about transformers, filters (like speaker system crossovers) and balanced and unbalanced lines and equipment.

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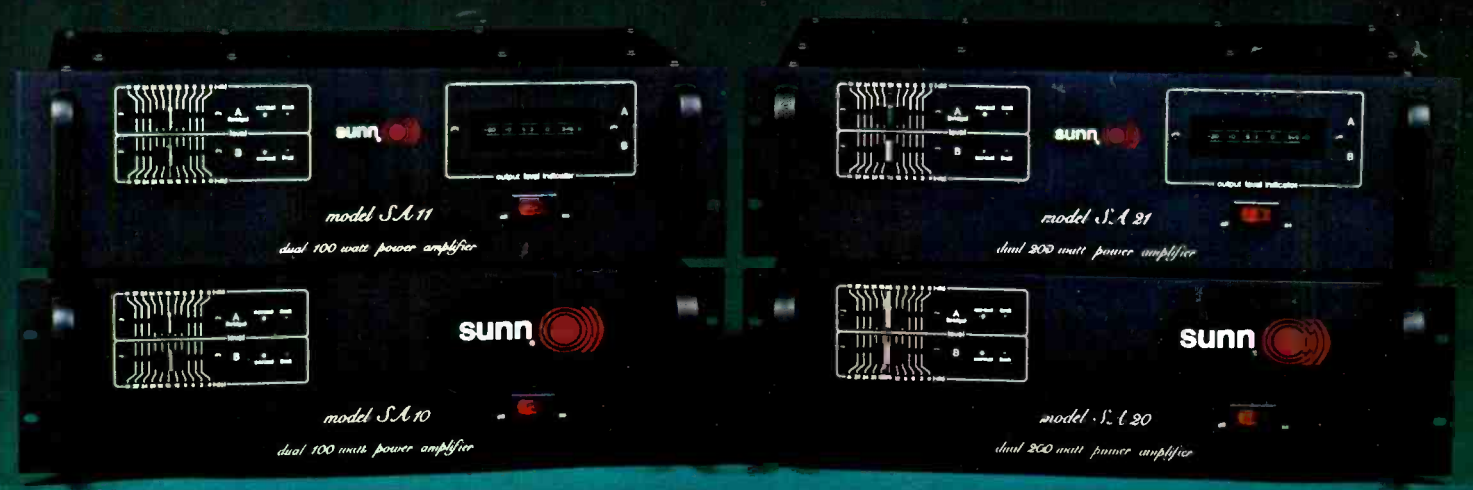


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# a session with Edgar

by Peter Weiss

# WINTER

Edgar Winter's music has undergone a continuous evolutionary process since his first Epic Records album was released about ten years ago. In that time Edgar has gained a well-deserved reputation as a gifted, all-around musician (saxophones, synthesizer, keyboards and drums among other instruments.

Equally well-deserved is his reputation as a hard-hitting, hard-driving perfectionist in the recording studio. This writer has had first-hand experience with this facet of Edgar Winter's musical personality, having engineered two of his albums and part of a third.

Edgar's current album project, *Standing On Rock* (to be released on the Blue Sky label) was recorded and is being mixed (at press time) at The Workshoppe Recording Studios in Douglaston, New York with engineers Kevin Kelly and Rob Bengston. The project has been physically taxing for everyone involved, as Edgar's sessions typically run fifteen to sixteen hours a day. However, the professional rapport between Edgar, Kevin and Rob, and the atmosphere of mutual respect have eased some of the hardships of working long hours.

The following is a portion of an inter-

view/discussion that took place in the control room at The Workshoppe, one afternoon before an overdub session. Present were Edgar Winter, Monique Winter, Kevin Kelly and this writer.

**Modern Recording & Music:** How was it that you first found out about Workshoppe Studios?

**Edgar Winter:** I found out through our keyboard player, Ronnie Lawson. He had done some work here, knew Kevin, the engineer, and he suggested The Workshoppe. I was checking out various studios in the New York area and this was about the third or fourth one that I came to. I liked it as soon as I came here, so we decided to record here. I liked it primarily because it sounds good. I liked it also because it is not like some of the studios in the city that have two, three or four rooms and lots of people. It feels a lot more comfortable in one room with the same people all the time so you don't have to keep changing engineers. And it's relatively close to Manhattan. It's only about 20 minutes.

**MR&M:** So it has a lot to offer over the mid-town studios?

**EW:** Yes, I feel so.

**MR&M:** You find it advantageous to work in a one-room operation?

**EW:** Yeah! A lot of times if you're working at a place in the city, you can't continue in the same room that you've been working in. That's always a disadvantage, to have to change rooms even for overdubs or for mixing. I'd much prefer to get used to one room and stay with that all the way through.

**MR&M:** In the past when you've worked with multi-room set-ups have you found inconsistencies in sound from room to room?

**EW:** Oh definitely. Even though the studios try to make their rooms similar. They never are. There are always differences that show up.

**MR&M:** What kinds of differences?

**EW:** A lot of times they have different boards. Completely different equipment from room to room. I've never been in a studio that had, say, three rooms all set up exactly the same. They usually have the same monitors but the dimensions of the rooms are never the same. Some monitors are closer, and aimed differently. It never is consistent.

**MR&M:** Did you plan to do this entire project here?

**EW:** I had originally thought of recording here and mixing the album somewhere else, in Manhattan. But I've gotten so used to this room and these particular monitors that I would not feel as confident if I were mixing anywhere else.

**MR&M:** Kevin, do you feel yourself on a technologically competitive basis with mid-town studios?

**Kevin Kelly:** I think the situation is that in terms of console, tape machine, monitor system and mic complement you couldn't ask for any better. We have our Harmonizers and digital delay lines and stuff like everyone else, but we might need five of them and we wouldn't have that many. But that can all be rented.

**EW:** Those things are usually only necessary in the mixing process anyway. But in recording all you need is a good board, a good *clean* board, good mikes, good tape machine and you can record.

**MR&M:** Okay, speaking of good, clean boards. Do you like the [Sound Workshop] Series 30?

**EW:** Yes.

**MR&M:** Can you do what you want with it?

**EW:** Yeah, I think it's a great board.

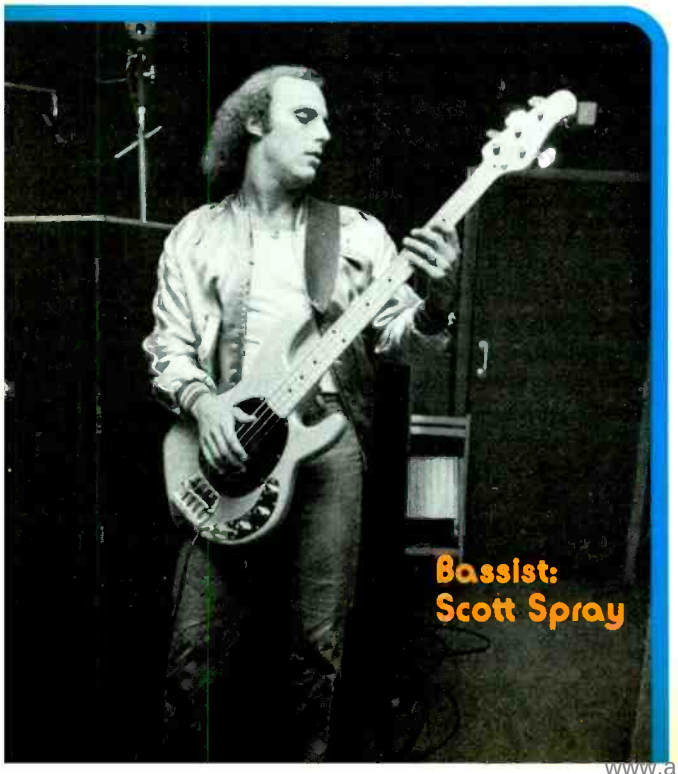
**KK:** I like it very much.

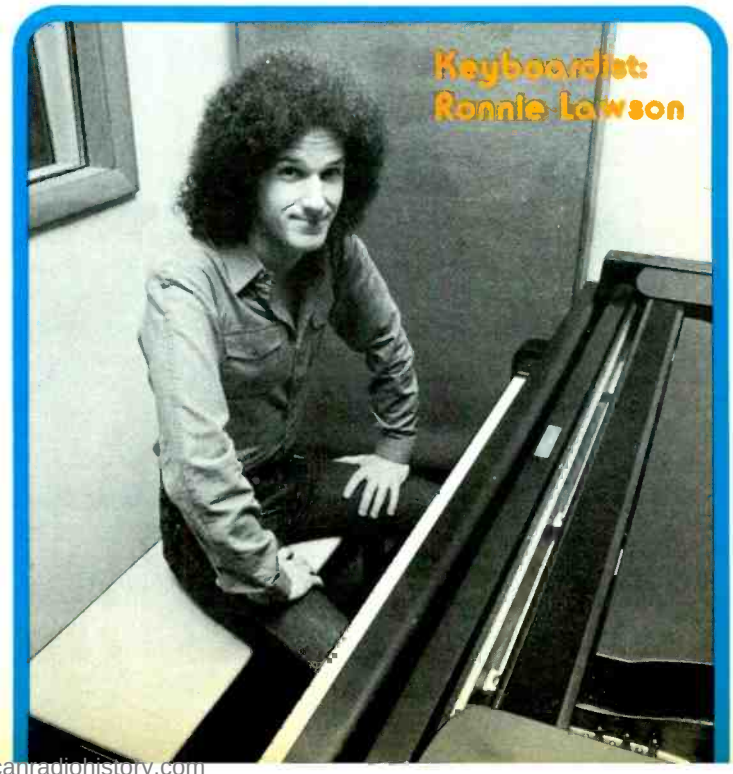
**EW:** When I came here to check out the studio, the first thing I asked was the various functions of the board. To check out the flexibility, to make sure that it would be able to do everything that I needed. The main thing was to have more than two cue mixes. I've worked in a lot of studios in the city and they say, "You can only have two cue mixes." That doesn't work a lot of times, for basic recordings, because everybody wants to hear something different. I like this board a lot; the EQ is relatively smooth; it's not synthetic sounding. That's one of the things that I always listen for because I like to EQ a lot and if the EQ sounds unnatural, then I can't do that.

**MR&M:** You say you like to EQ a lot. In the recording or mixing process?

**EW:** In recording.

**MR&M:** About the recording, I understand that there are some other unusual facts connected with the drummer, Gregg Carter and the drum set. I understand his drum set is sort of



unusual for a recording drum set.

**EW:** Yeah it is. We had fifteen mics on the drums. Double kicks, snare is three, hi-hat is four, left and right overhead, six and nine toms. Nine and six is fifteen. So each time we set up we had to adjust all of those drums. Kevin miked them and I would go in and take the toms and try to get all the ring out of them to make them sound good. That would usually take four to six hours, just to do that. Before we even started recording.

**MR&M:** That *is* a lot of time. About miking the drums. What kinds of mics did you use?

**KK:** Pretty standard set-up in the sense that we used condensers for heads with some bottom rolled off. A Beyer M500 on the snare. We wanted it to be nice and fat—considering all the possible leakage problems we'd have with all the open tom mics. Sennheiser 421s all around on the toms because we found that they have a real right cardioid pattern and they leaked less than other mics did. We tried one of the set-ups using RE-20s, but they tended not to be as directional and to be a little bit more of a problem in terms of hearing other drums in them. We wound up using RE-20s on the kicks because Gregg's kicks, instead of needing a lot of "flap" added, needed a nice round bottom added to them, which was a surprise because they're pretty big.

**EW:** They are 28 inches each.

**KK:** But that was the reality of it. So we wound up with 20s on the kicks and 421s around all the toms. The only mic that anybody would think kind of strange is an RE-55 (which is a pretty colorless omni-directional Electro-Voice mic) on the hi-hat. The reason I did that was because Edgar liked to equalize quite a bit and I found that by using a condenser like just about everybody uses—real tight on the hi-hat—when it started to take on a lot of equalization it started to sound very synthesized. And I found that the RE-55 was able to take a lot of the +10 at 8K or whatever it was that Edgar wanted to put in there without really sounding that it was about to blow up. So that's what we wound up using and it was nice and smooth even though it was very equalized.

**MR&M:** That is an interesting choice for that spot in the drum set.

**KK:** My partner still thinks I'm crazy for doing it. But it *did* work out.

**MR&M:** How far away from the hi-hat did you put the RE-55?

**KK:** Because of the fact that there were so many other mics, things had to be tighter than you wanted them to be just to maintain stereo imaging and stuff like that. So it was three or four inches away. If it was any further you would have heard more hi-hat in the snare mic than you had heard in the hi-hat mic.

**MR&M:** How did you mic the toms? From underneath?

**KK:** No, they actually were double-headed drums.

**MR&M:** All nine tom-toms had both heads on them?

**KK:** With the exception of the first two melodics. All their top-end crack was coming off the overheads. We just kind of rounded them out by putting a couple of mics inside them; but the rest all had to be miked from the top because they were two-headed drums. Which of course added to the tuning problems and make it that much harder for Edgar to get the rings out.

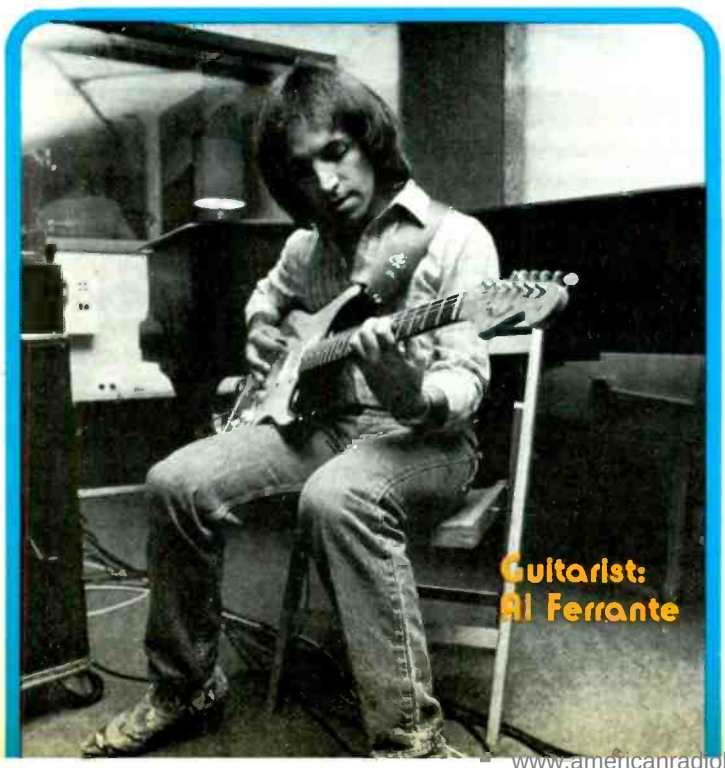
**MR&M:** Was it a lot of work getting the drum sound together?

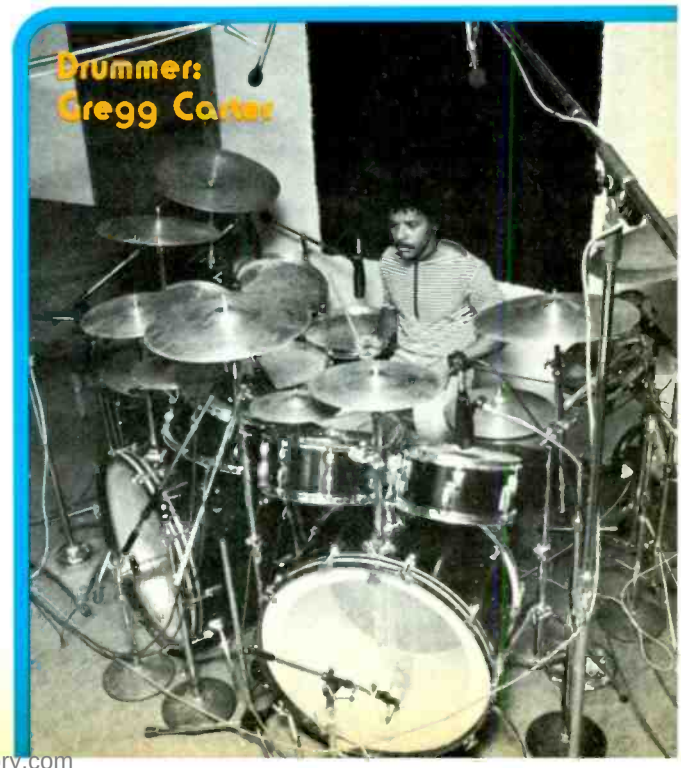
**KK:** We did a lot of playing around—turning the drums around so areas that sounded better were nearer the microphone. And just physically getting that many microphones in on a drum set without hampering the player is a technique that is kind of mind-blowing to start with, never mind making it sound good.

**MR&M:** Did Gregg Carter feel intimidated by all those microphones?

**EW:** He never said anything about it. Once in a while he would say a mic was too close, getting in the way, and Kevin would come out and readjust it. I really feel if you get a drum to sound good, if it sounds good *acoustically* when you put a mic over it, it will sound good when it is recorded. But I always go around all the lugs, and tap close to the rim to find out if one is higher or lower, try to even them all out and then just put tape where it's needed. Basically that's all we used was tape. Sometimes we have to use some padding of some sort. But I think the only place we had to do that was on the snare.

**KK:** I think there's one thing you have to understand. It may sound to some people like Edgar was going for a



very dead drum sound which was really not the case at all. It's just that Gregg always likes his drums pitched up higher. So when Edgar said he taped up all the drums it doesn't mean that he took all the ring out altogether so that they sounded like blocks of wood. Actually, they sounded "live."

**MR&M:** What was the instrumentation as you did the basics? What did you consider a basic track?

**EW:** The basic tracks were mainly keyboard, which was for the most part clavinet because I didn't want to use the piano. That would have created a leakage problem. And guitar, bass and drums. So, four people in the studio and I was here in the control room producing. I had to watch over all of that and make sure that the drums were correct and that the bass sound was correct. And the drum sounds vary from song to song. Some of the songs have tighter snare drums, and some of them are tuned down to sound bigger. I tried to use the sound that I thought would fit the particular song. I had pretty much an overall concept of each song and how I thought it should sound when it was finished. So it would have been impossible for me to be out there playing and still make sure that that was occurring as I thought it should be.

**MR&M:** So for those basic tracks you feel you're much better off in the control room than out in the room with the musicians?

**EW:** The only exception was the one track which happens to be the title track from the album, "Standing On Rock." We did that completely differently. I started out playing piano to a "click track." Gregg the drummer was on vacation so I decided to play drums on it myself.

**MR&M:** Do you really give your musicians vacations?

**EW:** Yeah, sure. Well . . . he asked for one quite a few months ago and I said okay. I didn't realize that it was going to run into the recording time. Otherwise I would have had him here. I thought at that particular period I was going to be doing overdubs and wouldn't need him for any basic tracks. But when I decided on the title of the album and the title song, the people that we've got back at Blue Sky and CBS wanted to hear that song immediately, if not sooner. So I said, "Okay so I'll do it." I went in and I played the piano, and I played the drums and then I brought in the bass player, Scott Spray and the guitar player, Al Ferrante, and they put their parts on. And we built it, in other words, from scratch. Which, in a way, was better because since each thing was put on separately, there was better separation in every pan than there would have been if everybody was playing at the same time.

Also, on that song I did not use Gregg's drum set; I used the studio's set which has only three toms. Gregg used the same snare drum all the time. We decided that the one that was here . . . most studios have a set of drums that the engineer is used to recording and knows the sound of, and knows how to make sound good. That's what I decided, rather than try to use Gregg's customized drum set which is very difficult to play. Some of the toms are at different heights. It's a very unique set-up.

**KK:** One thing that's interesting, though, as Edgar said, "Standing On Rock" kind of worked to our advantage in terms of leakage. On the other hand, I think Edgar will also admit that "Tomorrow Land," which is one of the ballads on the record, really benefitted from the fact that we did a basic track with the piano, and we did have a certain amount of leakage from one instrument to the next. It was a nice, big, warm sounding basic track because of what we were getting from the rest of the band in the mics, and a lot of other things that might be considered disadvantages by some people. But it certainly worked to our advantage in that particular case.

**MR&M:** Well, as long as you've got it under control.

**EW:** Yeah, the room sound here is good. I like room sound. For that particular sound, "Standing On Rock," I felt separation was good because it had really a distinct kind of piano part that had to be heard. And it had clavinet also. I think that worked out well for what it was. But I really prefer recording the way we did all of the other basic tracks with the room sound. I think in a lot of cases it does help the drums in particular.

**MR&M:** On the ballad in particular, the drum sound was much more open

**(Seated left to right) Monique Winter, Edgar Winter, engineer Rob Bengston; (standing left to right) engineer and co-owner of The Workshoppe Kevin Kelly, Gregg Carter, Scott Spray, Al Ferrante, Jeff Kracke, co-owner of The Workshoppe, and Ronnie Lawson.**

than it was on the uptempo tunes that I heard.

**KK:** In the uptemp things, a lot of the guitar sounds have to be recorded very, very loud and a lot of them have to be overdubbed. The way we did "Tomorrow Land," was that the relative volume level in the room with all the instruments made the leakage listenable. Whereas the leakage on all the other tracks we had was 90% guitar, cranked to ten in the room, and everything else was kind of in its own little area.

**MR&M:** I want to get back to the point you made before about having a concept of a finished whole as you're putting that whole together. There are different approaches to working in the studio. Some people come in and say, "Let's use the studio and experiment" and other people have an idea, up front, of what they want to do at every particular stage in the process. Which approach do you favor and which approach did you use in putting this album together?

**EW:** Well, I actually favor having a complete concept, at least what the rhythm section should sound like. But when it comes to doing the overdubs and the finishing touches, I like to stop and evaluate the basic tracks first.

When we started, we were recording like three or four days and then having a day off. Then the band would work. We would play somewhere, locally, for a couple of days and have a day off and then come back into the studio. So that each time we came back we had to reset the drums. But the advantage of that to me was the fact that I could listen to the rhythm tracks that we had recorded. I did rough mixes of those. We took them home and then decided what would best complement them. But I feel that you at least should have an overall concept when you put down the basic track—what the drums should sound like; what the bass should sound like; what the guitar should sound like.

In most cases on this album there are two or three guitars on each cut. I find that even when I do have an overall concept, that once I hear the rhythm track, the concept changes. Even though I think I hear the finished thing in my mind, after I hear what goes down on tape, usually I find either more to add or something that didn't sound quite the way that I expected it to. It generally changes and I think that it's better to be able to get away from the songs for a while and be able to listen to them. Then you can come back with a more objective point of view to finish them.

**MR&M:** Do you have any facilities at home to do any pre-production work?

**EW:** No. What we did before we came in was to record the songs beforehand at a rehearsal studio—Star Sound in N.Y.C.—and I used a four-track Teac and we miked everything as much as we could similar to what you would do in the studio. We recorded all of the songs so that we were entirely ready as far as the concepts of those songs that we came in to record. Then after that first batch of rhythm tracks that we did I was still writing songs. So we had to change in the middle of the album from rehearsing new material to coming into the studio. Then I would have to show the whole group the song. In other words, after we did all the drums, tuning and taping and all of that stuff, I would show them the song and then we would record that song. And those were situations where we would get just one track per day because the group was not at all familiar with the song beforehand.

At home I have lots of speakers, my favorites are Allisons, and I've got JBLs—4511s. And we have some speakers in the bedroom but I'm not sure what they are. But I listen to them also. I've got at home what works for me because I listen to records and I know how to compensate between records and cassettes. This is really important because most of the rough mixes I would take home were on cassettes. That was because my four track is not in our apartment. But when we start doing the real mixes I'm going to bring the four track home; the reproduction is so much more accurate.

**MR&M:** Edgar, you seem to cross over the traditional line from production into what we used to call engineering, actually working at the board and getting sounds with EQ, mic placement and so forth. Have you always worked this way?

**EW:** I would say over the past two or three years I started to work that way. What I did originally was try to communicate the ideas that I had to whomever was engineering, but as I started to learn more I realized that there was no way. Like a lot of the times I would hear a snare drum and say, "No, that's not big enough, add some three or four hundred," and the engineer would do that. But it never achieved exactly what I wanted so I just decided I might as well start to do it myself. I really do have a definite idea of what I want to hear and it's almost impossible to communicate that to somebody and have them do it. When I hear a sound in my head I can't make that other person hear the same sound. I just feel the easiest thing is to play around with the EQ myself until I find what I want to hear.

**MR&M:** Do you ever want to have full-blown professional facilities of your own?

**EW:** Yes, That's my basic dream right now. I really feel that is the most comfortable situation that you can work in when you have your own equipment and you know what it does. To me, working here is the next best thing to that. Yes, hopefully I would eventually like to get my own studio.

**KK:** I think what makes Edgar really easy to work with, from an engineer's point of view is, like you say, that he crosses over the line. What makes it really workable and easy is that he has a working vocabulary of all the things he wants to hear and do in technical terms. You work with some producers and they say things like "I want to hear more blue." You get these requests that can't be related to a technical resolution very easily. Edgar knows when he says he wants more 10 K and more this and more that. He knows what it sounds like. He knows what he means when he says broaden the band of the EQ, or when he says he wants to hear 30 milliseconds of delay. He knows what 30 milliseconds is going to sound like. Things got smoother and easier to do as time went on. When Edgar starts equalizing I just have to be very conscious of the gain staging and what that's doing to the tape chain and to the compression, if there is any, and that type of thing.

And Edgar is taking care of the artistic end in terms of what tone colors he wants to use. It's actually a pleasure. Some engineers would look at it like somebody encroaching on their

territory, but I think it's just a more exacting way of getting the record done the way it's supposed to be.

**MR&M:** It certainly keeps engineers on their toes, too, working with somebody who knows what he is doing in technical areas. It seems that more artists and producers are interested in educating themselves.

**KK:** That's exactly it. The more producers know about the music and the technical end the less reason there is for an engineer to say, "Well your request is ridiculous." Years ago I'm sure that there were a lot of producers who wanted to hear things that were kind of impossible to do and people who wouldn't take no for an answer often had to be tricked.

You asked Edgar earlier about using the studio as a tool. One interesting note is that yesterday he was doing a synthesizer overdub that easily could have been divided into two tracks and the synthesizer retuned to get this particular thing that he wanted to play. But Edgar's feeling was that he wouldn't be able to play it "live," and he wants to be true to his audience in terms of his performance. To be able to reproduce the record on stage he decided to go for something that was maybe less spectacular, so that he knows when he goes out and does the gig he'll be able to reproduce it. I think that is a real good attitude. On the other hand, when using a studio as a tool for spatial or ambience things that make the record particularly big sounding, Edgar really likes to get into that type of stuff. For example, he had me turning tapes upside down and running things through Harmonizers backwards and then turning them back over. All through it I'm trying to figure what he wants, and when it gets done it sounds incredible.

**Monique Winter:** That was five o'clock in the morning.

**MR&M:** Long session. These have been on the order of twelve hours, fifteen hours . . .

**EW:** Yes. At least thirteen hours. Because we come in at two P.M. and hardly ever leave before three A.M. Monique, my wife, has been my inspiration through the whole album. She has been here in the studio and we take the songs home and study all the stuff together and work on the lyrics. She's really just like an additional member of the group.

**MW:** Edgar is the director and here is the producer.

**EW:** Yeah, I produce it and then she is *my* producer.

**MR&M:** That's a good partnership.

**MW:** Yes, wives are sources, I believe, and husbands have power. The husbands are in the front, but without the wife, without the source there is no power. Power always has a source, and someone else is always the source. So I give all the credit to the wives.

**MR&M:** Kevin mentioned the cue systems in the console, and the fact that there are more than two available, which you commented on earlier. Do you find it easier to give each musician a separate mix?

**EW:** Yes, I think it is essential for the musicians to hear exactly what they need to hear.

**MR&M:** And they have different requirements, each one . . .

**EW:** If you only have two cue mixes and you have four people out there then two people have to be on the same mix. What we generally did was to use the two cue mixes for the drums and the bass in mono, because they really only need to hear bass and drums and to hear what they're playing. But as far as the guitar and the keyboard, we used the stereo mix so that the guitar player could hear himself in one side and the piano in the other and vice versa. We could then adjust the levels between those two elements.

**MR&M:** All this cue mixing was available in this console without any external gear?

**KK:** We have one external stereo mix. But the point is that had we wanted four monos they would have been available on the board. But it just so happens that we have an additional 24x2 mixer that we sometimes use for a producer to get his own monitor mix together and in Edgar's situation we used it as a fourth cue mixer.

What is interesting is that a lot of people consider stereo headphones overkill because the basic function is that the musicians should hear themselves and know what they're playing. But my feeling, and I think Edgar's gotten to like it too, is that a little bit of subtle positioning inside of the headphone makes it much easier for people playing midrange-oriented instruments to make out their notes from the other guy's notes.

**EW:** You can separate yourself and hear what you're really playing as opposed to what somebody else is playing. That's really important because you can hear yourself better, and it allows you to play your part better.

**KK:** I think it specifically helps with stuff in the midrange area that gets real loud and honky in a pair of headphones. No matter how good the set of phones is, when there is a little space involved you can hear the notes better.

**MR&M:** How many cuts are completely finished [as of August 1980]?

**EW:** Four completely done and five more that are in various stages of completion. All of the basic tracks are done. So what we are going to have to do, I think, because of the time situation . . . we're only going to have enough room [on the album] for eight. We're going to record nine, finish nine completely, and after they're mixed I'll give all the finished mixes to Blue Sky and let them play those for various people at CBS and see what their feelings are. See which ones they like the best. I really feel I'm open to their suggestions. That's why I've been doing rough mixes. We've already played some of the rough mixes that you heard yesterday for some of those people. And if there is anything in them that they have to criticize then I will be able to change that before the album is completely finished. Rather than handing it in and saying, "Here it is." Everybody will have some opinion to offer. If I feel that it's constructive then I'll act on it. If I feel it's wrong, I'll ignore it.

**MR&M:** About the recording of the tracks. Now you have four tunes nearly finished and all the basic tracks done. On the four songs that are done, how many tracks were taken up with instruments as opposed to vocals?

**KK:** I'd say in general we filled up about twenty-four tracks by the time we were done.

**EW:** Yes, we filled up everything.

**KK:** I would not call this album a heavy vocal background record. There are a few tunes where multiple vocals are used really nicely, but they're certainly not on everything without fail. I think we averaged maybe four vocal tracks or something like that after we had sub-mixed.

**MR&M:** Eighteen to twenty instrumental tracks. Now on the vocal

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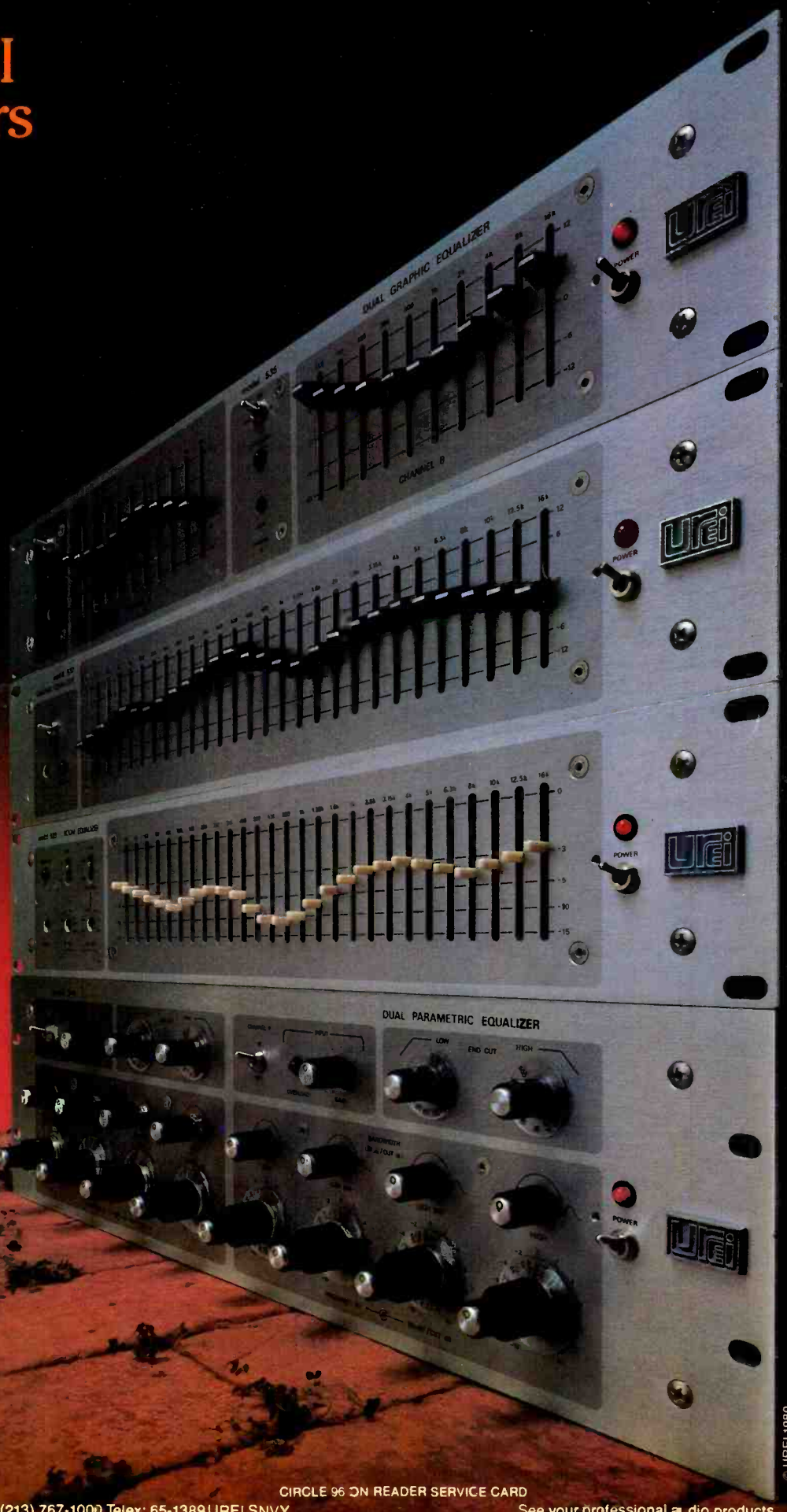


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tracks, did you find yourself doing a lot of punching in on Edgar's lead vocals?

**KK:** That's one of the things that goes fairly quickly. Everything is very painstaking with Edgar. But the lead vocals are a pleasure because they tend to get done real quickly. Edgar bases a lot of his criticism about vocals on energy level, as opposed to this and that little pitch difference. Very often there *are* no pitch problems when Edgar sings. As far as backgrounds go, that's more critical: "let's punch in at that line"; "we'll keep the first two words and do the second word and out on the fourth word"; etc., etc. The lead vocals are a lot of rock 'n' roll energy, and if they sound right, then they sound right.

**MR&M:** Who is comprising the background group?

**EW:** The backgrounds are myself and Ronnie Lawson. Usually I sing two parts and he sings one. And those are generally doubled so it's like six. Two of myself on top, two of him in the middle and two of myself on the bottom. And then those get bounced down to two stereo tracks.

**MR&M:** How do you go about setting up that stereo?

**KK:** Each song is different.

**EW:** Yeah, it's different. In fact, it's always different.

**KK:** A lot of times there is some Harmonizing involved and the use of a couple of digital returns. And very often when we sub-mix all the backgrounds they get sub-mixed to a stereo pair. So there's delay effects that differ from one side to the other that create the spatial effect Edgar is looking for.

**MR&M:** You do that on the multi-track. Then you don't save that stuff for the mix?

**KK:** Not all of it. Some of it is done on the multi-track.

**EW:** I like to try to make it sound like what I think it should sound like. I prefer to save having to do that in the mixing process.

**MR&M:** You'd rather that the mix be straightforward.

**EW:** I'd rather it sound like I think it should in the beginning. So we *do* put effects on as we're recording. Which a lot of engineers don't agree with. "That's real flat and we'll fix it in the mix" is what they usually say. And I say, "No, let's fix it now, so we don't have to fix it later."

**MR&M:** Well, fixing it, if something sounds wrong or if the sound is inadequate, that's one thing. But sometimes effects are irrevocable. You put an effect on a track and that's *it.*

**EW:** That's right. Once you put it down, it's there. So you have to make sure you're putting down what you want to hear.

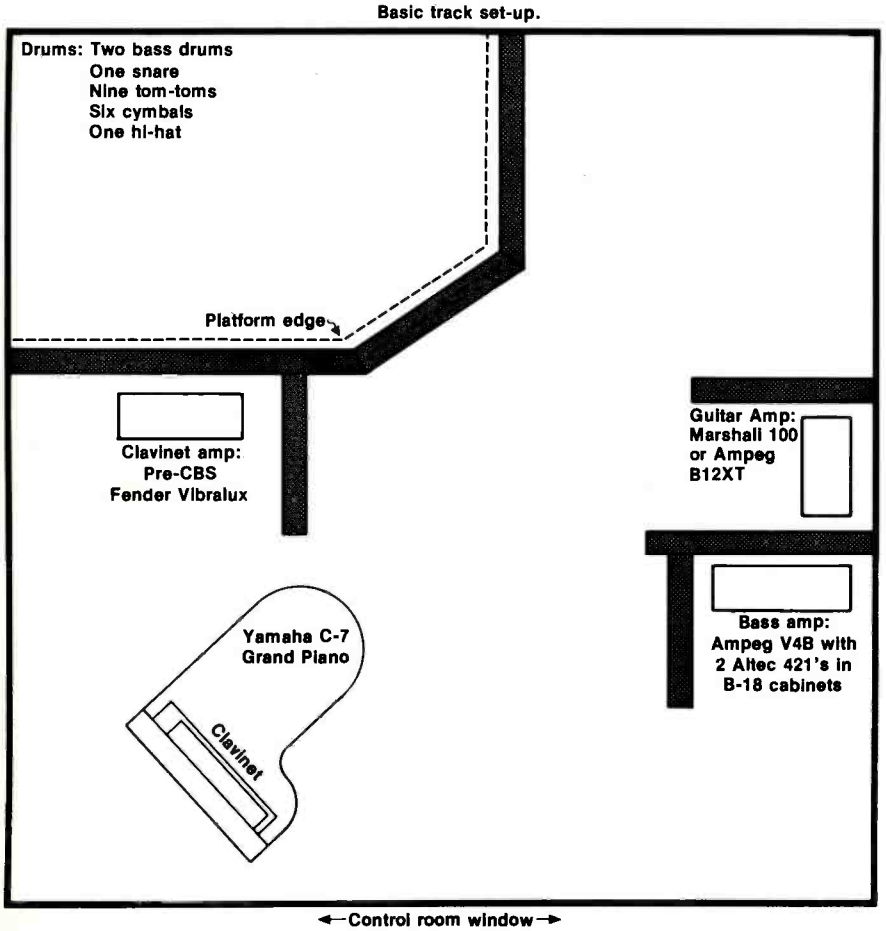
**MR&M:** That requires a lot of confidence and a lot of preparation. And *that* requires having listened to the music. I don't know of a whole lot of people that are willing to work like that, that like to work like that.

**EW:** I don't think that there really are very many.

**MR&M:** Except in the classical field, where they get everything sounding right first. Putting it down on four tracks not twenty-four, they must get everything sounding right first and then they just leave the recording pretty much to itself.

**EW:** Well, that's the way recording used to be. You had one time to do it, and everybody had to do it right. There wasn't any second chance before you had all these tracks to work with. It was just a question of everybody playing good. Getting it down.

Basic track set-up.




Subsequent to this interview, Kevin Kelly discussed the progress of the final mixes, which at that point were about halfway through. Edgar Winter did all of the console work, with Kevin and Rob Bengston lending extra hands when necessary. Extra hands were not needed all that often, according to Kevin, because of the way the multi-track tape was put together, as described towards the end of the interview. Stereo pairs of saxophone tracks, background vocal tracks and other elements were pre-mixed.

Kevin rented an Echoplate II and Eventide 949 Harmonizer for the mixing sessions, but only the Echoplate saw any action. All of the special effects had been recorded on the multi-track.

*Standing On Rock* will be released sometime in the fall or early winter. This is Edgar's first "solo" performing/producing effort, and the results indicate that this method of operation suits Edgar's requirements, musically and technically.

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| MODEL | KEYS | VCO | VCF | EG | NOTES | DIGITAL MEMORIES |
|---|---|---|---|---|---|---|
| CS-5 | 37 | 1 | 1 | 1 | 1 | N/A |
| CS-15 | 37 | 2 | 2 | 2 | 1 | N/A |
| CS-20M | 37 | 2 | 1 | 2 | 1 | 8 |
| CS-40M | 44 | 4 | 2 | 2 | 2 | 20 |

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# PROFILE:


## Dan Dugmore
## Rick Marotta
## Waddy Wachtel
## Stanley Sheldon
## of RONIN

by Marty Basch

*Both Waddy Wachtel and Rick Marotta have been established musicians for some time doing session work with artists Linda Ronstadt, James Taylor and Jackson Browne. They've formed Ronin, their own band with Dan Dugmore and Stanley Sheldon, and have a debut album on the Mercury label.*

*Wachtel is also a noted producer, having lent his talents to Warren Zevon's* Excitable Boy *album.* Modern Recording & Music *spoke with Wachtel and Marotta to find out about their past session work and the recording of their "live" studio record.*

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**Modern Recording & Music:** Waddy, what first got you interested in playing the guitar?

**Waddy Wachtel:** When I was real young I just happened to see one on television. That's what started it. I'm talking *real young.* That's when I was 5 years old. I started playing when I was 9.

From age 6 to 9 I badgered my father about it until he bought me one. He bought me a little Kumica guitar. It was the most beautiful thing I ever saw.

**MR&M:** Rick, why did you choose the drums?

**Rick Marotta:** I didn't start playing 'til I had gone to college for about a year. A friend of mine went away to the army and he left me his drums to hold on to until he got out. That's what started me and that's why I had those drums. They were a blue Gretsch set. Junk. They had calfskin heads on them that must have been ten or fifteen years old. The guy never changed the heads in his life and you couldn't have broken them unless you had run a truck into the drumset.

**MR&M:** Did you have anything to do with music before that?

**RM:** No. My father was a dancer, my mother was a dancer. *[As he starts to do a Johnny Cash imitation]* and "my boy's gonna be a dancer too," God damn it.

**MR&M:** What were your first experiences like in the recording studio?

**RM:** Mine were pretty weird because I had just started playing. David Spinozza, whom I was playing with at the time, had actually got me started playing. David brought me to these sessions where I think Clyde Otis was the producer. They were R&B sessions.

The first sessions I did were on the radio. I remember going home and listening to WLIB about 3 weeks later and the song was on the radio. So I right away got this feeling like, I've got to go and do this again. What a feeling jogging down the street and hearing yourself play the drums on the radio. So that was it. It was all R&B stuff the first year or two. Black rhythm and blues records.

**MR&M:** Waddy, how did you go from your first guitar at age 9 to Linda Ronstadt's *Simple Dreams?*

**WW:** Briefly, from age 9 to 20, I lived in New York. Then I moved to Los Angeles and started trying to get into studio work. I was with these country kind of things and "folk people." Things just started leading around until I met Lou Adler who liked my playing. Adler played with Carol King and turned me on to meeting Linda and Peter Asher.

**MR&M:** Peter Asher. Why did you choose him as your producer for Ronin's album?

**WW:** He is the best producer I've ever worked for.

**MR&M:** What makes him the best?

**WW:** He facilitates a [form of] communication that is very necessary to make a record happen: conversation. Peter makes a session happen or takes it away. He's real fine at communicating with the musicians, with the engineers and with the artists.

**MR&M:** Did you use any special techniques for his album?

**WW:** We recorded it digitally; 32-track 3M digital.

**MR&M:** Why digital over analog?

**WW:** Because we A/B'd them by playing a number of bars on both and then comparing. In comparison, the digital was amazingly brighter. However, I said I liked the warmth of the analog. Peter said, "Yeah, but I think the digital sounds exactly the way it sounds in the studio." So Rick went out there and played straight time and we A/B'd what was on the digital tape to the "live" Rick playing in the studio at the same time. There was no difference.

**RM:** It really ended up with,"If you could do that with the drums, it might be worthwhile."

**MR&M:** What kind of drums do you use now?

**RM:** I just got a new set of Yamaha's from Japan. Took them out of the box today. The bass drum head is missing. I can't find it. I've got two different snare drums and two bass drums. I'm using Yamaha drums.

The drums I've been using all these years, until now, have been two Pearl concert toms, an 8" and a 10". You can only get them with one head.

**MR&M:** What kind of heads are they?

**RM:** Well, I put Remo Ambassadors on top, but I drilled holes in the side and put Castors in the bottom heads so I have top and bottom heads on the small drums. Now Yamaha does that, and I did that with all my small drums— put all the heads on the bottom. I put Diplomats on the bottoms of the 8" and the 10". Then I have a 13" and a 16" floor tom. So I only use the four and a 22" bass drum with Ambassadors. Once in a while I'll put on a hydraulic head.

**MR&M:** Rick, you do some of the background vocals. How do you place the vocal mics so the sound from the drums doesn't leak into the vocals?

**RM:** On stage, I've never given that a second thought. I haven't done that much background singing—except in this band. The more it [the drum sound] leaks into that [vocal] mic, as far as I'm concerned, the better it would be.

**WW:** We like that leakage. You see, we cut our album "live" and . . .

**RM:** I leaked into Waddy's vocal mic so much that we had to figure out a way in which his voice could be heard over the drums.

**WW:** At one point on one song, we had to Keypex the leakage out because it was too much. That was on "America, The Beautiful."

**MR&M:** Whose idea was it to use the U.C.L.A. Men's Chorus on it?

**WW:** It was Peter's idea to get a choir on that album.

**RM:** We wanted to get the Marines.

**WW:** Yeah, when Peter said a choir, we said let's get the Marine Choir; but we didn't think that they were about to come and play for a bunch of little guys. So Peter suggested U.C.L.A.

**MR&M:** How big was the chorus?

**WW:** Eleven.

**MR&M:** So it was no problem fitting them all in the studio.

**WW:** No. It was a pretty big session, a funny night. It was an overdub. They weren't there when we cut the basics.

**RM:** That's one of the only overdubs.

**MR&M:** The album took seventeen days to record. Why did you prefer to do it "live" versus laying down individual tracks?

**WW:** 'Cause it's better to sound like a band if you're a band. We wanted it to sound just like we do on stage.

**MR&M:** Rick, what kind of mics do you use for the drums?

**RM:** Now this is where you've got me. I have an [Sony] ECM-50 inside my snare which we mix with the mic on the outside of the drum. It seems to be working very well. The one on the outside, which I like the most is a Shure 57, I think.

**MR&M:** What about the studio?

**RM:** Every studio is different. For our record we changed sometimes for different songs. That, Val [Garay, the engineer] would know. I paid very little attention to that.

**WW:** We used a couple of Neumann 67s up in the sky for overheads on his kit. I don't know what we were using on the kick.

**MR&M:** What guitars do you now own, Waddy?

**WW:** I own a J-200 Gibson acoustic which travels in my room with me. I use a '58 Les Paul, a '55 Telecaster, a '56 Stratocaster and I also use a '63 Strat, but Dugmore [Dan Dugmore, who plays guitar and pedal steel for Ronin] uses it.

**MR&M:** Tell me about the pick-ups and amps.

**WW:** Straight stock. All my instruments are completely stock. I don't f--- with them. They're all just the way they came.

**MR&M:** Could we talk about your sound system?

**WW:** We're at the mercy of clubs.

**RM:** At this point we don't have our own sound system, so we have to go with what they give us every night. If we did our own tour the way Linda Ronstadt and James Taylor do, we

**SELECTED DISCOGRAPHY — Rick Marotta**

| | | |
|---|---|---|
| *Walking Man*, James Taylor | (1974) | Warner Bros. W 2794 |
| *Royal Scam*, Steely Dan | (1976) | MCA AB 931 |
| *Little Criminals*, Randy Newman | (1978) | Warner Bros. BSK 3079 |
| *Simple Dreams*, Linda Ronstadt | (1978) | Asylum 6E 104 |
| *Aja*, Steely Dan | (1978) | MCA AA 1006 |
| *Excitable Boy*, Warren Zevon | (1978) | Asylum 6E 118 |
| *Restless Nights*, Karla Bonoff | (1979) | Columbia JC 35799 |
| *Bad Luck Streak at Dancing School*, Warren Zevon | (1979) | Asylum 5E 509 |
| *Hold Out*, Jackson Browne | (1980) | Asylum 5E 511 |

would know.

**MR&M:** Is that a hassle for you?

**RM:** It would be except for the fact that our sound man Graham Holmes has proved to be amazingly good.

**WW:** He rewires systems.

**MR&M:** It's basically his show with the sound?

**RM:** Yup. He really makes me worry very little.

**WW:** He's got everything covered.

**MR&M:** Both of you have been in the studios for most of your careers, a technical environment. How does the studio compare to "live" performances?

**WW:** We really like to play on stages. I mean . . . we like them both.

**RM:** I think it's more stimulating to play on stage. You're feeding off the people visually. When they like it, you put out more. In the studio it's very . . .

**WW:** Clinical.

**RM:** Yes. And it's a call to perform. It's like doing a porn film. O.K., perform. But, on stage if you're having a real hard night, you just really can't get it up, can't get it going. We've had a couple of those.

**MR&M:** Do either of you feel you have any individualized techniques that stick out?

**WW:** I make a helluva cheeseburger. Like I said, I run a chord into an amp and play stock. I don't use effects.

**RM:** I think my drums sound a little different than most. Considering the smallness of my drums, they sould real big. That's because of the head combination I told you about, and tuning them real loose.

**WW:** And I crank the hell out of the volume so it sounds real loud.

**RM:** I think the sounds are real unique, but a lot of it is touch. Guitarist Hugh McCracken has a certain touch that you can't duplicate. Waddy has a certain touch. They could switch guitars; Waddy's guitar would sound like Hughie's; Hughie's guitar would sound like Waddy's. When they play, there is a certain thing that the effect is more in here [Rick points to his left wrist] that makes the sound unique.

**MR&M:** Waddy, you co-produced Warren Zevon's *Excitable Boy* album. Was he an easy guy to work with?

**WW:** It was an interesting project. It was tough and took about a year.

**MR&M:** How did you land that job?

**WW:** I got into that job because one time when I was in Europe, someone was interviewing me about Warren's first album, and I went ahead and said I thought Jackson Browne had his hands a little too full and didn't really know what he was doing. When I returned home, I get this phone call from Jackson. He said, "So, I didn't know what I was doing, huh?" I said, "What?" He said, "I had my hands too full." I said, "Well yeah." He says, "Do you want to produce the next one with me?" I said, "Hey man, you've got to be kidding me. You don't even know me." He says "Yes I do, and I want you to help me." That was how I became a producer.

It took a little while for the three of us to settle into each other, but it was great. Jackson could stay in the booth and I would be out there with the band making sure the tracks sounded the way I wanted them to. Jackson could hear them in the booth.

My function in the Zevon sessions was to fill the space and perpetuate the communication. Communication, that's the ticket. Me and Jackson, it was up to us.

**MR&M:** Going back to what you said before about Asher and communication, why do you think he is able to communicate so well? What is it that's there?

**WW:** Just experience, just a relaxed nature and knowledge. Knowing. You've got to know what you want. I know what I want to play, Rick knows what he wants to play and Peter knows what he wants to hear.

I remember one morning we sat around eating for like three hours—eating. Just eating and going crazy. Nobody could stop eating. Then we went out and did "Here Comes the Runner" in one take.

**MR&M:** You really did that in only one take?

**WW:** Oh yeah. Most of the songs are one take. But it went like this: Three hours of eating, we went in, did one take, and Peter said, "Thank you.

**SELECTED DISCOGRAPHY — Waddy Wachtel**

| | | |
|---|---|---|
| *Buckingham Nicks*, Buckingham Nicks | (1973) | Polydor PD 5058 |
| *Hasten Down the Wind*, Linda Ronstadt | (1976) | Asylum 7E 1072 |
| *Black Rose*, J. D. Souther | (1976) | Asylum 7E 1059 |
| *A Night on the Town*, Rod Stewart | (1976) | Warner Bros. BSK 3116 |
| *Warren Zevon*, Warren Zevon | (1976) | Asylum 7E 1060 |
| *Karla Bonoff*, Karla Bonoff | (1977) | Columbia JC 34672 |
| *Rumours*, Fleetwood Mac | (1977) | Warner Bros. BSK 3010 |
| *Little Criminals*, Randy Newman | (1977) | Reprise BSK 3079 |
| *Simple Dreams*, Linda Ronstadt | (1977) | Asylum 6E 104 |
| *The Bride Stripped Bare*, Bryan Ferry | (1978) | Atlantic SD 19205 |
| *Living in the USA*, Linda Ronstadt | (1978) | Asylum 6E 155 |
| *Excitable Boy*, Warren Zevon | (1978) | Asylum 6E 118 |
| *Restless Nights*, Karla Bonoff | (1979) | Columbia JC 35799 |
| *Born Again*, Randy Newman | (1979) | Reprise HS 3346 |
| *The Glow*, Bonnie Raitt | (1979) | Warner Bros. HS 3369 |
| *When You're Only Lonely*, J. D. Souther | (1979) | Columbia JC 36093 |
| *Flag*, James Taylor | (1979) | Columbia FC 36058 |
| *Bad Luck Streak at Dancing School* Warren Zevon | (1979) | Asylum 5E 509 |

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Lunch break."

**MR&M:** How's it feel to be produced instead of producing?

**WW:** It feels great. We were thinking of producing the album ourselves, one day we were at a meeting and Peter said, "If you need a producer, I'm available." We thought about it.

It took a lot of weight off of us to be able to just go in as artists, instead of going in, telling the engineer it has to sound right, getting all that stuff straight and then going out and pretending you're relaxed enough to play.

浪人

**MR&M:** How did the group Ronin get together?

**RM:** Well, I worked with Stanley Sheldon [bass] on the Frampton tour several years ago. It was a good feeling playing with Stanley, and subsequently we worked with Ronstadt. Waddy and Dugmore talked a lot about putting a band together. Waddy and I had first talked about it, but Dan was . . .

**WW:** The most obvious choice for guitar player.

**RM:** Yeah, so I suggested we try Stanley Sheldon.

**WW:** We flew to Switzerland to do a three week tour with Warren Zevon, and we had no bass player. Rick said, "I know this guy."

**RM:** We called him and then he flew in the next morning and we were off. It really was a helluva tour. That's how we got together.

**MR&M:** Do you think with this tour [June 1980] you'll be able to break away from the stereotype of studio musicians that's been applied to you?

**WW:** This will give the people the chance to decide whether they want to stereotype us as studio musicians or whether they'll accept a new band. So this is a chance for us to be an entity that performs and entertains.

**MR&M:** Could you predict what is in the future for Ronin?

**WW:** Lots of stages and lots of performing. We're gonna keep going, keep writing songs, recording them and playing for people. And make lots of money, somehow.

**RM:** Yeah. We're gonna stick with this. We'll do studio work again if we ever want to do somebody's record that we like. I do jingles in New York and I like that and don't want to kiss that stuff goodbye. We just want to tour this band and sell a lot of records, become millionaires and take it easy.

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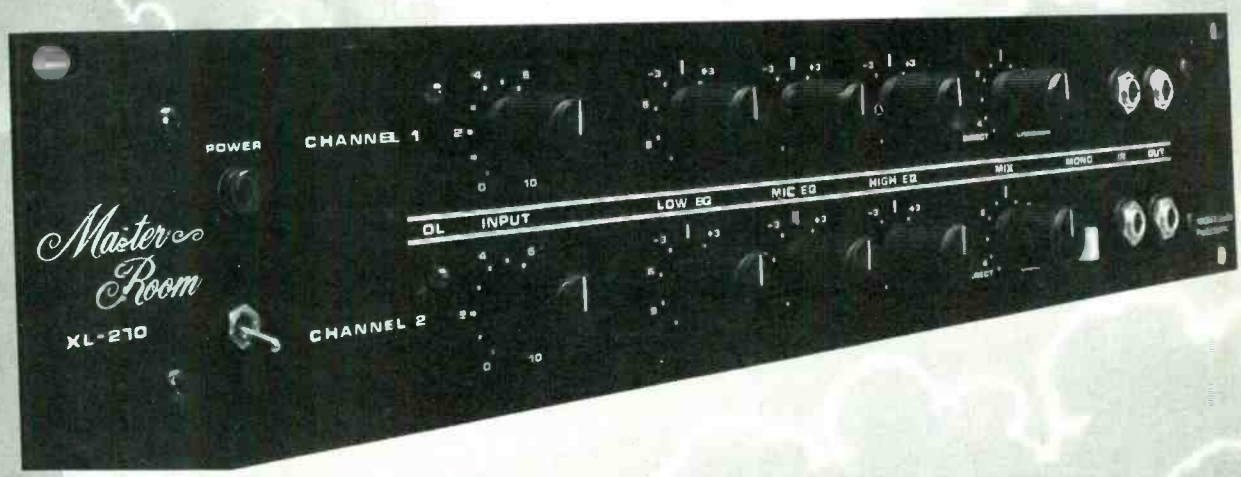


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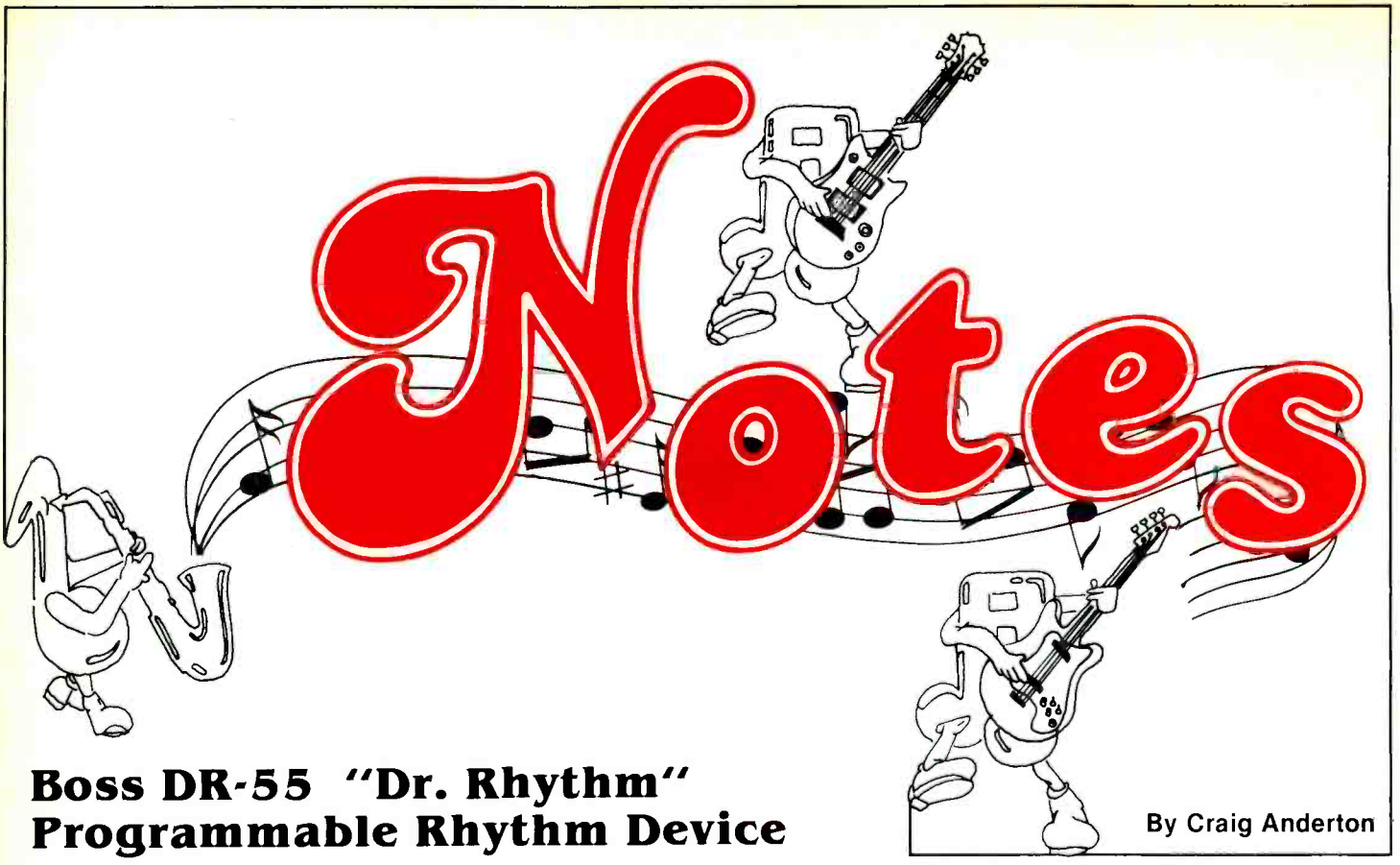


# Boss DR-55 "Dr. Rhythm" Programmable Rhythm Device

By Craig Anderton

Rhythm units have made great strides in the past few years. Go into a music store, and you'll see that the "Foxtrot" and "Bossa Nova" presets are slowly being replaced by "Funk," "Reggae," "Disco" and other patterns that are more in touch with current musical trends. Some units now have programmed accents, variation switches and similar features to add some variety to the normally monotonous rhythm unit sound; musicians are even adding processing to these boxes to further extend the sonic boundaries. Still, there is something restrictive about using someone else's canned rhythm program . . . wouldn't it be nice if you could program your own patterns, and maybe even play the unit like an actual instrument instead of just pressing a button and letting the pattern drone on in the background?

Apparently someone at Roland's Boss division felt that way, and the result is a tiny (20.5 x 11 x 3.5 cm or 8 x 4.5 x 1.5 inches), attractive and functional black and orange box that goes by the name of "Dr. Rhythm." Don't let the Doctor's small size fool you; this is a versatile and creatively designed drum unit whose sonic possibilities should hold your interest for quite some time. If you think that all commercially available drum units are the same, read on.

WHAT IS IT? Dr. Rhythm is a battery-powered unit that consists of two basic functional blocks: one block is the "audio" section, which contains the drum sounds; the other block is the "programming" section, which allows you to program various drum patterns into the unit's memory.

The audio section is easy to discuss, so let's give it an overview. There are four drum sounds: snare drum, bass drum, high-hat and rim shot (although clave would be a much more accurate description). There are no individual mix controls for these drums, but there is a master volume control to control the overall output level and a tone control to vary the high frequency content. The tone control allows you to change the snare and high-hat sounds from sizzling to muted, and is more effective than, say, the tone control on a guitar. A single output jack connects the audio output to your amplification system.

Before we discuss the controls associated with the programming section, we need to take a quick course on exactly what programmable drums are all about.

## The Concept of Programmability

All drum units have some kind of master clock that sets the tempo for a given rhythm pattern. In conventional drum boxes, this master clock generates a constant stream of pulses. A decoder section then selects certain pulses to drive certain drums. For example, if each pulse corresponds to a quarter note, then four pulses would make up a measure; the decoder might take the first pulse of each measure and send it to the bass drum, ignore the second pulse, and send the third and fourth pulses to the snare drum to create a "rock" beat. Another option would be to

feed the first and second pulses to the bass drum, the third pulse to the snare drum, and ignore the fourth pulse to give an alternative rock pattern. Most rhythm units will have approximately four to thirty-two patterns to choose from. While this provides a certain amount of variety, it's usually not enough to prevent terminal boredom after a few hours —and sometimes even minutes—of play.

A programmable drum, on the other hand, uses semiconductor memory ICs that allow you to program your own custom rhythm patterns, and modify them at will. Luckily for musicians, this programming process is very similar to multi-track recording, so don't feel you have to be a computer whiz in order to have fun with a device like the Dr. Rhythm. You do, however, have to put more effort into learning how to use a programmable device, since it is essentially "stupid" and as a result you have to tell it exactly what you want it to do. How well you tell it what to do will determine, to a large extent, the effectiveness of the final sound. For example, it took me over an hour to program all the patterns for a particular song I was working on—but it was a highly educational hour that produced a pretty complex drum track. So, while using Dr. Rhythm takes some effort, you do get rewarded for that effort.

I'd love to carry on about what makes programmable drums tick from an electronic standpoint, since the principles are really quite intriguing. However, we do have a test report to get to here, so let's leave the technology for another time.

PRE-FLIGHT for the DOCTOR. Patch the output jack to an amplifier, turn the *Volume* control clockwise so that the power switch clicks on, and then set this knob at about the halfway point. Make sure that the *Mode* switch is on *Play,* then set the *Rhythm Select* switch in the upper right-hand corner to one of the six 4/4 patterns or one of the two 3/4 patterns. Note that this switch has a large knob that makes it very easy to switch between the various programmed patterns. Press the orange *Start* button; since the DR-55 is pre-programmed with several rhythm patterns at the factory, you'll start to hear drum sounds coming out of your amplifier.

This is a good time to get acquainted with the non-programming-oriented controls. As you vary the *volume* control, the drums get louder and softer. Vary the *tone,* and you'll change the character of the snare and high-hat sounds. The *tempo* control sets the overall speed. The high-hat comes into play through a 3-position switch; the positions are "Off," "8" (which gives one high-hat strike per quarter note) and "12, 16" (which gives two strikes per quarter note). The high-hat sound is not a programmable function, but one which you "play" by fooling around with the switch. For example, you might want to bring this sound in for a few minutes, and then switch it out again.

Dr. Rhythm also has an *accent* control. Turning this up adds an accept by increasing the level on certain selected beats. At the full counterclockwise position, there is no added accent. At mid-position, the accent is noticeable but still relatively subtle. Cranked up all the way, the accent is very obvious and makes certain beats "jump out" at you. This accent feature is programmable so that you can tell the Doctor exactly where you want the accents to occur in a given rhythm pattern.

There are some more advanced playback techniques you can use to spiff up the drum sound even more, which we'll get into later. In the meantime, while playing some of the pre-programmed rhythms will give you an idea of what Dr.


Rhythm can do, the real fun is in programming your own patterns. So let's first examine the programming controls, and then how to actually enter your own programs.

PROGRAMMING CONTROLS. The rhythm select switch mentioned earlier selects which pattern we're going to program. Six of these are 16 step rhythms for making 4/4 patterns, while two of the selects are 12 step rhythms that work with 3/4 material.

To the left of this knob is a *variation* switch, which adds a tremendous amount of power to the unit. It has three positions, labelled "A," "B" and "AB." Let's say that you've selected pattern 1 with the rhythm select knob. Variation A will give you one 16-step pattern, while variation B will give you a different 16-step pattern. You can program both of these to give completely different rhythms, if desired; or, you can just add a few variations so that by manually switching back and forth between A and B you can keep changing the overall pattern. In the AB position, Dr. Rhythm first steps through the 16 steps programmed into variation A, and then the 16 steps programmed into variation B for a total of a 32-step pattern. It's also important to emphasize that you can switch over from one variation to another any time during a measure; there will be no lost beats or timing problems.

So far, with two controls we can choose one of eight different master patterns to program, plus two different variations for each pattern. However, we now need some means to tell the Doctor whether we want to program new sounds or play back previously programmed material, and that's where the mode switch comes in. In the play position, the DR-55 plays back whatever pattern you've programmed in. In the *write* position, you can program in new patterns.

There are three other controls involved in programming. The *Sound* switch selects which sound you're about to program, and gives you a choice between bass drum, snare

drum, "rim shot" and accent. Finally, during programming the *Start* control and *Stop* control assume different functions. How these three controls work will become clear as we discuss how to program the unit.

PROGRAMMING the DOCTOR. First, set the rhythm select to one of the eight patterns to be programmed, say pattern 1. Then, set the variation switch so that you're programming either the sixteen steps of variation A or the sixteen steps of variation B. If you want to program thirty-two steps, set the variation switch to AB.

With the mode switch in the *play* position, hit the *start* button. You'll hear whatever was previously programmed for this pattern. Now, punch the *stop* button; the drums will become silent. Change the mode switch over to *write*. Note that this illuminates a little red LED next to the tempo control, which is the Doctor's way of saying, "I'm at the first step of the pattern and await your programming data." As soon as you program the first step, the LED shuts off and doesn't come on again until you've programmed all sixteen steps.

Next, select the drum sound you want to program with the sound switch. Think of this as being very much like a track select switch on a 4-channel tape recorder, since you program each sound in a multi-track fashion. I generally start by programming the bass drum, then the snare, then the rim shot and finally add the accents. So begin by putting the sound switch in the BD (bass drum) position.

Let's program variation A, which is a 16-step pattern. Suppose you think of these sixteen steps as being quarter notes; this gives you a total of four measures of quarter notes. Then you can program the bass drum to trigger on any one of these notes, but we'll assume for now that you only want the bass drum to trigger on the first note of each measure.

As mentioned earlier, in the program mode the start and stop switches change their identity. Pushing the start button programs a drum beat into a step and advances you to the next step, while pushing the stop button advances you to the next step *without* programming any sound. Getting back to our example, here's where we want the bass drum to occur in the pattern:


16 Steps = 4 Measures

To enter this, we hit the *start* button, depositing a bass drum beat in the first beat (or step) of the first measure. We then hit the *stop* button *three* times to get us to the beginning of the next measure, whereupon we hit the *start* button again to deposit another bass drum trigger. Hit the *stop three* times to get to the next measure, hit *start*, hit *stop three* more times, hit *start* again to put a bass drum beat at the beginning of the fourth measure, and then hit *stop three* more times. At this point, the LED will come on to tell you you're back at the beginning of the pattern again and have programmed all sixteen steps.

I should point out that the sixteen steps do not necessarily have to make up four measures. If you wanted to, you could treat each step as an eighth note and end up with two measures of eighth notes instead of four measures of quarter notes. Or, you could go for one measure of sixteenth notes if you like drum rolls.

OK, the bass drum part has now been programmed for variation A of pattern 1. At this point, you can either switch over to variation B and program some more bass drum, or stick with variation A and program some different sounds . . . let's do that [the latter].

By switching the sound switch over to SD (for snare drum), you are now ready to program the snare drum information in the same way. Suppose you wanted to add the following snare part over the bass drum pattern:


16 Steps

Press *stop twice*, then hit *start once* to deposit the snare trigger, and hit *stop* again; you've programmed the first measure. Continue onward until all sixteen steps are programmed. As in the case of the bass drum, after you've programmed all sixteen steps the LED will illuminate to tell you that the Doctor is ready for the next barrage of programming information.

Move the sound switch over to RS (rim shot), and program the various clave sounds in a similar manner, then switch over to AC (accent) and program where you want the various accents to occur.

Finally! The programming is complete for variation A of pattern 1. See what I mean about multi-tracking? Build up the pattern a drum at a time—just like playing with tape.

After all that work, it's time to collect your reward before doing any additional programming. Set the mode switch to play, push start and whatever you programmed into the drum will come back at you.

What happens if you find out that you made a programming mistake with, say, the snare drum? No problem. Push *stop*, go back into the *write* mode, re-enter the correct snare drum data, return to the *play* mode, and press *start*. Do this as many times as it takes to get it right. Note, however, that you must re-program the *entire* snare drum "track"; you cannot simply erase one wrong sound and then add the right one where required.

As the rhythm plays back, it's always fun to fool around

# "GOOD SOUNDS" from DeltaLab

## Strawberry South Speaks Out!

Eric Stewart of 10 c.c.

# STRAWBERRY RECORDING STUDIOS SOUTH

Strawberry Recording Studios South, 61 South Street, Dorking, Surrey. Tel: STD (0306) 87852/3

J. Camacho, Esq.,
Deltalab Inc.,
27 Industrial Avenue,
Chelmsford,
Mass. 01824.
U.S.A.

24th March 1980

Dear Jim,

Congratulations! you've certainly come up with a winner with your Acousticomputer. It is one of the few genuinely new audio signal processors to appear for a long time, as it affords the engineer a precise yet very versatile control on ambience through the entire range from a single repeat to a very smooth and full sounding reverberation; and after six weeks we're still finding new sounds and effects from the unit.

Its nice to find people who share our feelings for quality and innovative approach.

Eric Stewart and Tony Spath

Tony Spath
Head Engineer & Studio Manager

**Strawberry Recording Studios South is owned and operated by 10 c.c.**

DeltaLab **DeltaLab Research, Inc.**

27 Industrial Ave., Chelmsford, MA 01824 (617) 256-9034 - Telex # 951205

CIRCLE 114 ON READER SERVICE CARD

with the high-hat switch, tone control, volume and accent control to add additional variations to the basic sound.

APPLICATIONS. One very attractive feature of the DR-55 is that you can switch between patterns or variations at any time. Thus, you can program a number of fills and alternate patterns and switch between them throughout a song. For example, with one piece I programmed separate patterns for the first verse, second verse, chorus and instrumental section, as well as a number of fills (one pattern just triggered the snare drum continuously, and I would switch to that whenever a snare roll was required). Between these program switching options, the ability to bring the high-hat in and out, and the various games that you can play with the volume, tone and accent controls, Dr. Rhythm becomes very playable and much more musically useful than non-programmable drum units. The result is a far more interesting drum sound than you could ever get with a conventional "set-and-forget" type drum box. Dr. Rhythm is also ideal for multi-track oriented musicians who play all the instruments in a composition themselves, since you can now back up your guitars, keyboards, voices, or whatever with some interesting—and non-repetitive—drum parts, even if you're a lousy drummer with a terrible sense of time.

For practicing to a beat, the extra work required to program Dr. Rhythm makes it a little harder to work with than a conventional box. However, if you want to you can always program the Doctor with a number of standard patterns specifically for practicing.

Dr. Rhythm is a versatile enough box so that I could easily see some of the more electronically oriented groups (Gary Numan, Ultravox, John Foxx, et al) having a drummer who only plays a Dr. Rhythm-like device and perhaps a drum/cymbal synthesizer or two for adding manual accents. The fact that the DR-55 is truly playable, that you can endlessly adjust the controls and *construct* drum parts is what makes this box so useful.

COMPLIMENTS and COMPLAINTS. While the above gives you an idea of how Doctor Rhythm works, there are still many other important points that need to be covered.

One of these points is a complaint: I really don't care for the sound of the "rim shot." Dr. Rhythm is so intelligently designed, I really can't understand how they let this one go by—unless it's a concession to the cocktail lounge players who can't conceive of a drum box without a clave sound. In fact, for rock and electronic music applications I seldom, if ever, use the rim shot sound. (Incidentally, I wrote up a modification on Doctor Rhythm for the *Device* newsletter, 1085 Broadmoore Dr., Napa, Ca. 94558 that describes how to convert the clave sound to a tom sound and how to bring out the drum outputs and pulses for stereo applications. If you're interested in this, send $1.50 to the above address and request the "Surgery for Dr. Rhythm" article.)

That's it for complaints. Considering the relatively low cost (approximately $200 list), Dr. Rhythm gives you a lot for your bucks. Any added features, such as more drum sounds or additional program storage, would precipitate a hefty price increase.

Now for some additional compliments. Although battery powered from four "AA" (penlight) cells, the current drain is only 5.5 mA and the batteries last a long time. An added advantage of using batteries is that even when the unit is turned off, the memories still retain their programmed data. The amount of current drained by the memories in the standby mode is so small (a few microwatts) that this does not affect the battery life significantly. What's more, the programmed patterns will be retained in memory even when you change the batteries! The only stipulation is that the unit cannot be without batteries for more than about three minutes, but believe me, it doesn't take anywhere near that long to make the changeover.

There are two additional output jacks that deserve mention. One of these allows for synchronization with certain other Roland products (such as sequencers and the like) by putting out a pulse for each step in the rhythm pattern. The other one provides a 4 V, 10 ms output pulse for driving external drum units or triggering synthesizer envelope generators. The only catch here is that plugging into this jack disables the accent effect, since we're using the programmed accent information to provide this trigger. However, it is often worth giving up the accent feature in order to program an additional drum voice when needed.

Oh yes, there's still one more jack (they really do pack a lot into this thing). Plug a footswitch in here, and you can control the stop/start function remotely.

OVERALL EVALUATION. As mentioned in the beginning, Dr. Rhythm requires some effort to learn and use; but this is effort well spent, since you'll end up with some really interesting, useful and above all *playable* rhythm patterns. The drum sounds aren't mind-boggling, but the bass drum is solid and convincing, the snare drum does its job, and the high-hat does, in fact, sound like a high-hat but with a little extra sizzle. I still think that the rim shot sound is a loser; a tom sound would be a much better option. Maybe in the next generation . . .

Once you get the hang of it, the DR-55 is not difficult to program. I would, however, recommend working up some sort of notation system for programs so that you have a permanent record of your favorite drum patterns. Any snythesizer player will tell you the value of writing down patches in a notebook—do the same with the patterns that you create for Doctor Rhythm.

For multi-trackers, the DR-55 is a dream come true. If you've been trying to get along with one of those "Foxtrot/Bossa Nova/Polka" preset drum units, do yourself a favor and give Dr. Rhythm an audition. I might also add that if you're not a drummer, playing with this little box can be very educational. Since you need to break down drum parts in order to program the right drum sounds on the right beats, you start becoming much more conscious of how drum parts are constructed. This makes it just that much easier to play along with these parts.

Overall, I'd have to say that despite rumors to the contrary, there is still at least one Doctor who makes house calls. And considering the treatment you get, the price is reasonable. Who knows? He might even be able to cure the rockin' pneumonia and the boogie-woogie flu.

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CIRCLE 153 ON READER SERVICE CARD

# Ambient Sound

BY LEN FELDMAN

## A Major Improvement in Tape Equipment Testing

If you read the test reports concerning the Akai reel-to-reel tape deck and the Technics cassette recorder appearing in this issue you will immediately recognize the fact that our approach to tape equipment testing has changed since the last time we evaluated a piece of tape hardware. The difference arises from the fact that our laboratory is now equipped with a new instrument designed and produced by Sound Technology, an audio test equipment company based in Campbell, California. Known as a Tape Recorder Test Set, Model 1500A, this new unit, pictured here, has a self contained video display screen which not only presents graphic information such as frequency response plots, speed error plots, distortion, channel separation and more, but also has alpha-numeric display capability which provides the answers in written form, right on the video display screen. Perhaps the most outstanding feature of this unit is an electronically displayed "cursor" which may be moved to the left or right on the screen. If the cursor is moved, say, to 15 kHz after a frequency response has been registered graphically on the screen, the video display reads out the dB level of the output at 15 kHz (with respect to a previously established 0 dB reference level.)

Rather than interrupt the text of the recorder reports appearing in this issue with explanations of how I used this instrument to completely and rapidly measure the performance of the deck, I decided to devote this "Ambient Sound" column to an explanation of the various video displays that are obtained using the new Sound Technology 1500A. I should point out, too, that the figures which appear in the tape deck report were not obtained by means of photography (though, of course, the video screen could have been photographed for each display). Instead, we also invested in a video printer. Since the Sound Technology unit has a composite video output jack, we were able to simply connect from this jack to our video printer. A push of a button then results in a clear 4" x 5" printout of whatever happens to be stored on the video display. Printout, in this case, is on electrosensitive paper supplied with our particular printer.

## Frequency Response

In the frequency response curves associated with our tape machine test reports this month you will note that both channels can be plotted sequentially. The notation "FR" at the upper right tells us that we are looking at a frequency response plot, while the notation at the lower left, "10 dB/D," tells us that vertical sensitivity of the display is 10 dB per division. Had we wanted to, we could have expanded the vertical sensitivity to 2 dB/division as well. The cursor has been moved to the point where we have an approximate roll-off of 3 dB at the high end and the level readings appear automatically, for that frequency, for both the channel outputs.

## Distortion Measurements

Second and third order harmonic distortion measurements are made automatically on the 1500A. A series of tones, at decreasing levels of 1 dB per step, are

recorded and analyzed by the instrument. The range of levels used in this test is from +10 dB (relative to that previously assigned 0 dB reference) down to −20 dB. That ever-present electronic "cursor" can then be moved to determine the distortion level at any of the test levels recorded onto the tape. It is obviously no trouble at all to determine the available headroom (3% third order distortion level) for any machine with any tape being tested. Again, if you check out the corresponding figure in our test report, you will see that results are expressed not only in percent (%), but in dB below the reference level.

## Channel Separation Tests

Channel separation measurements are also automatically performed by the new test instrument. For these tests, some thirty-one different frequencies are supplied as input signals by the instrument, in the range from 20 kHz down to 20 Hz. As a result, the test does not take quite as long as a full frequency response test, during which the instrument supplies and measures 123 different discrete frequencies! Channel separation at any of the test frequencies can then be read, in dB, by moving the electronic "cursor" to the appropriate point on the graphic display.

## Signal-to-Noise Measurements

A readout of signal-to-noise takes just a matter of seconds and requires only the push of a button marked "Noise" plus selection of the desired form of weighting, if any. Available weighting curves includes NAB, ANSI and CCIR/ARM favored by Dolby Laboratories and others. Noise readings, in dB, are displayed in seconds while bar graphs simultaneously show the instantaneous noise level, all relative to that 0 dB reference which was punched into the machine at the start of the measurements. If overall dynamic range relative to the 3% THD point is desired, the user must add the headroom previously measured to the S/N figure obtained in the S/N tests.

## Flutter Measurements

Measuring flutter with the new Sound Technology 1500A Test System requires that the user have a pre-recorded test tape, either at 3.0 kHz or 3.15 kHz. Of course, you can make your own test tape if you own a signal generator, but if you do you would then read some combination of flutter introduced during the making of the tape plus the flutter introduced during playback, and there would be no way of isolating the two components of flutter. Once the test tape is running, it is only necessary to push the button marked "Flutter" and the appropriate weighting curve button. In a few seconds, a display appears. A bar graph at the right shows instantaneous (and constantly varying) flutter. The vertical scale of this bar-graph has auto-ranging capability, so that full-scale will adjust itself to correspond with the flutter reading being measured. An average value of flutter appears as a percentage next to the letters "FL" which tell the user that the machine is in its flutter measurement mode.

## Speed Accuracy and Speed Drift

Using the same pre-recorded test tape that was used for making flutter measurements, the instrument is capable of measuring speed accuracy of a tape as well as long term speed drift. When this function is desired and the appropriate button is pushed, the instrument analyzes the speed of the tape transport (using 3000 Hz as a reference frequency) and displays the error graphically for a maximum time period of just over 10 minutes. Once the run is completed (or stopped at any time before completion), the electronic cursor can be moved to any point in the time graph and the instrument will instantly display, in percent, the speed error that was true at that time.

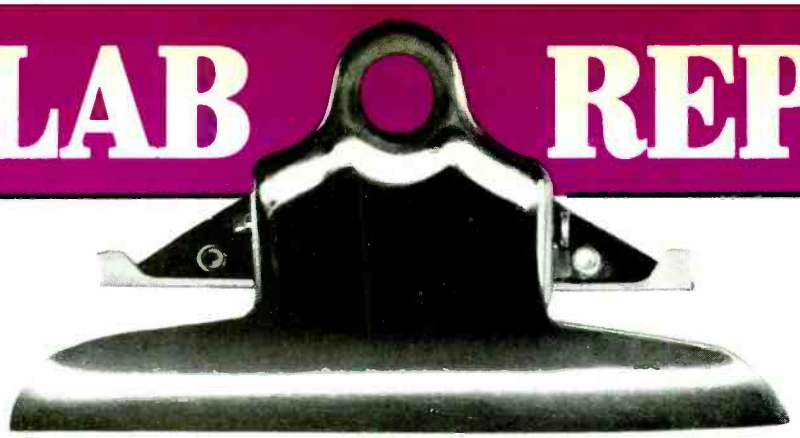
## Azimuth Alignment

The final test which the 1500A is able to perform has to do with tape head alignment. In the case of three-head tape decks, a special tape containing four test frequencies (2.8 kHz, 5.7 kHz, 11.8 kHz and 15.8 kHz) must first be run in the machine under test. Pressing the "Azimuth" button brings up a new form of display which tells the user how many degrees of error are present at each of the above-named test frequencies. The idea, then, is for the tester to adjust playback head azimuth until these errors (shown as vertical bars) are at a minimum.

The azimuth alignment of the record head can then be adjusted directly using the 1500A, since it supplies the same four test frequencies automatically as a new recording is made on blank tape and analyzed.

## Possible Applications

As you may have guessed by now, the cost of this amazing test instrument is probably too high for the average audio or non-professional tape recording enthusiast to even consider. At $5,500, the instrument is far costlier than most high fidelity systems or than most home recording facilities. Still, for recording studios that must constantly check the performance of their mastering and duplicating tape decks, for design laboratories involved with tape deck design, repair and servicing, or testing, the high price of this instrument will eventually be offset by the time saved in making repetitive measurements such as the ones that I have described and that will, beginning with this issue of *Modern Recording & Music*, be found in the test reports that Norman Eisenberg and I prepare. From a purely technical point of view, Sound Technology's new 1500A test set represents, to my mind, a brilliant combination of audio, video and computer/microprocessor technologies and applications.



NORMAN EISENBERG AND LEN FELDMAN

# Carver M-400 Magnetic Field Amplifier


**General Description:** The Carver M-400 is known as a "Magnetic Field" amplifier whose circuit design differs markedly from that of conventional amplifiers. A full explanation is available from the manufacturer, but briefly, the magnetic-field amplifier has no need for massive heat sinks, heavy power transformers or large electrolytic capacitors. Instead, the new amplifier steps-down voltage and stores the energy in a relatively small, lightweight and low-cost magnetic field coil. As a result, the amplifier produces very little heat, does not require the usual thickness of chassis, and its overall size and weight can be greatly reduced. The M-400 weighs less than 10 pounds, yet it is rated for an output of 200 watts per channel in stereo mode, or 500 watts in mono operation.

The unit presents an extremely compact appearance. The front panel contains a two-channel LED metering display with dB steps marked for 0, −5, −10, −20, −40 and "infinity." The last-named LEDs light up when power is applied, even if no signal is being amplified. They also serve another function—the display will blink or "strobe" approximately twice per second with diminished brightness to indicate a possible short circuit in a speaker connection or severe overloading. Depending on the fault, the blinking will continue for a period of a few seconds to several minutes. When proper operation is restored, the display returns to its power indication status.

The rear panel of the M-400 contains a recessed pair of phono-tip jacks for inputs, four color-coded speaker terminals, an impedance switch and a fuseholder with a 15-amp fuse. The impedance switch is a two-position slider that is normally placed in its downward position for loads of 4, 8 or 16 ohms. For mono operation, or for impedances below 4 ohms, the switch must be moved to its alternate position. In mono use, both of the amplifier's input jacks must be fed with identical signals via a "Y" connector.

**Test Results:** The M-400 was bench-tested for its published specifications and additionally for the usual characteristics that were not spec'd. Except for the 1-watt frequency response results, all tests confirmed or exceeded the manufacturer's claims. The 1-watt frequency response—measured from 1 Hz to 45 kHz—was still considered very good, specs notwithstanding. More important, the amplifier produced higher than its rated power, and distortion that was lower than spec'd. The S/N ratio also was slightly better than spec'd.

Speaking of bench-testing, if you ever have occasion to perform them on this amplifier you may as well know in advance that the M-400 does not lend itself to "easy" measurement. To begin with, there is the matter of terminal color-coding and polarity. One set of channel output terminals is color-coded with the black terminal connected to chassis ground, while for the other channel's outputs a red terminal is designated as chassis ground. This arrangement presents no problem at all when you connect the amp to a pair of loudspeakers. The signals coming from each channel are, in fact, reversed in phase with respect to each other, so that if you follow normal connection procedures (red terminal to "hot" speaker terminal for each channel), everything works out fine. In bench-

testing, however, chassis ground must relate to instrument grounds. Consequently, you will find that the left and right channel outputs will be out of phase when viewed on a dual-trace oscilloscope, for example. No great worry—just something to be aware of.

If you bench-test the unit you also will find it difficult, if not impossible, to measure the 20 kHz power output of the amplifier, as well as distortion levels at that frequency. That's because the protection circuits will shut down the amp in a few seconds when you try to pump rated power out of it at 20 kHz. Again, this presents no problem when handling any kind of music programming in which the frequency distribution versus energy is such that 20-kHz signals never produce very much power at the output terminals. Actually, the amplifier does run remarkably cool at normal listening levels, and gets only moderately warm when pushed to near-clipping levels.

**General Info:** Dimensions are 7 by 7 by 7 inches. Weight is 9.5 pounds. Price is $349.

**Individual Comment by L.F.:** I first learned about the Carver M-400 Magnetic Field Amplifier more than two years ago, when I saw and heard a prototype of the little amp at a Consumer Electronic Show. The claims then (and, for that matter, even now) made for the device seemed incredible in view of its small size and weight. How could a little cube, measuring no more than 7" on a side and weighing less than 10 lbs possibly deliver more than 200 watts per channel in stereo and upwards of 500 watts per channel when strapped for mono operation?

Well, the amplifier does do that and pretty well meets its other claims as well. I'm not going to get into the circuit theory here, but suffice it to say that Carver and I have spent many hours together discussing the circuitry and I am convinced that it does work and is reliable. In fact, in many ways, the amplifier is more reliable than conventional ones. It has about six levels of circuit protection, including overcurrent trip, low-frequency trip, voice-coil over-temperature trip, over voltage trip and D.C. fault trip. The amp can be made to shut down under a variety of circumstances, but none of these will cause permanent damage to the device, at least as far as we could determine during the bench testing.

Music reproduction from the Carver amp was totally clean and transparent, with good bass reproduction as well as clean reproduction of any high frequency transients I was able to come up with from selected tape and record program sources. I used medium-efficiency reference speakers and drove the amplifier rather hard just to see how much it could safely take. At no time during my tests did the amplifier shut down, even though the monitoring scope indicated that on several occasions I was just beyond the clipping point for short time periods. I had to constantly remind myself that I was dealing with a whole new kind of amplifier here, especially when I glanced at that small cube configuration and realized that all that power was being developed within those small confines.

Carver prints a warning on his little amplifier telling us that it is not intended for industrial or pro-audio use. That's probably because so many so-called "pros" push their amps to beyond clipping almost as a matter of course, and the M-400 won't stand for that for very long (it will simply shut down—not break down). Still, I suspect that the incredible portability and small size of this amazing amplifier will tempt many a pro user to disregard the warning and use it on the road as well as in "live" performances. I suspect, too, that despite the warnings, it will stand up reasonably well even under such "illegal" use, and even if occasional shut-down occurs, there won't be any permanent damage.

**Individual Comment by N.E.:** The Carver M-400 amplifier, by definition, demands serious attention as both a product and a possible harbinger of things to come in amplifier design generally. The fact is, it is much smaller and lighter in weight and—very important—lower in cost than one would normally expect of a similar-performing amplifier of conventional circuit design. The M-400, to be sure, has gone through some design refinements since its initial debut about two years ago, and the units have not been in very great supply.

But I have this incredibly small prodigy sitting in a rack, hooked up to a high-class stereo system playing commercially recorded releases and some tapes I have made myself and with whose sound I am very familiar. Lab measurements aside, the Carver M-400 is doing one fine job as an amplifier of musical sound, and those LEDs on the panel tell me I am pushing well over normal playback wattage levels into my speakers with nary a hint of strain of distortion. The highs are sparkling, with good transient bite; the middles are smooth and well-aired; the lows are firm and well defined. Portions of one tape that I deliberately "doctored" in the making do come through as I intended, so all told, as far as I can judge, the M-400 is doing nothing to the signal but—amplifying it. A final point: the manufacturer explains, in his literature, that the output of this amplifier is really the output of the power supply being switched on and off at a rate directly related to the incoming audio frequency. Lest you fear that the term "switching" involves some kind of audible switch-noise, be reassured that it does not. Perhaps the term "inaudible transfer" would be more appropriate, since the changes in music from soft to loud as reproduced by this amplifier happen quite naturally and smoothly, with no audible "switching" or "breathing" at all.

CARVER M-400 AMPLIFIER: Vital Statistics

| PERFORMANCE CHARACTERISTIC | MANUFACTURER'S SPEC | LAB MEASUREMENT |
|---|---|---|
| Continuous power for rated THD, 8 ohms, 1 kHz | 201 watts | 240 watts |
| FTC rated power (20 Hz to 20 kHz) | 201 watts | 210 watts |
| THD at rated output, 1 kHz, 8 ohms | 0.05% | 0.02% |
| THD at rated output, 20 Hz, 8 ohms | 0.05% | 0.03% |
| THD at rated output, 20 kHz, 8 ohms | 0.05% | 0.05% |
| IM distortion, rated output, SMPTE | 0.05% | 0.02% |
| IM distortion, rated output, CCIF | NA | <0.03% |
| IM distortion, rated output, IHF | NA | <0.03% |
| Frequency response @ 1 watt (for −3B) | 1 Hz to 250 Hz | 1 Hz to 45 kHz |
| S/N ratio re: 1 W, "A" wtd, IHF | NA | 81 dB |
| S/N ratio re: rated output, "A" wtd | 100 dB | 104 dB |
| Dynamic headroom, IHF | NA | 1.6 dB |
| Damping factor @ 50 Hz | NA | 170 |
| IHF input sensitivity | NA | 0.07 volt |
| Input sensitivity re: rated output | NA | 1.0 volt |
| Slew rate (volts/microseconds) | 40 | 40 |
| Power consumption, idling/maximum | NA/NA | 26 watts/835 watts |

CIRCLE 32 ON READER SERVICE CARD

# Technics Model RS-M95 Cassette Recorder


**General Description:** A three-head cassette deck, the Technics RS-M95 has its record and play heads sharing a common housing thus obviating the need for periodic azimuth alignment. Its transport uses two motors: capstan drive is handled by a quartz-controlled, phase-locked DC brushless direct-drive motor; the reel tables are driven by a DC coreless motor. The transport system is IC logic-controlled and uses "feather touch" buttons that permit complete fast-buttoning, including the option of going directly into the record mode from any other mode, including playback (run-in recording or overdubbing).

The RS-M95 is metal-tape capable. Tape selection is worked out between four separate selectors (for normal, ferrichrome, chromium-dioxide and metal) plus four corresponding front-panel bias adjustments. Also offered on the front panel are recording calibration adjustments. All adjustments may be made with the aid of a built-in test-signal generator, selectable from the front panel, while observing readouts on the deck's meters. The meters themselves are switchable between peak and VU levels, and there is a peak-hold option.

The deck has a "dual" memory rewind system with which it is possible not only to get the recorder to stop or play when the counter reaches "000," but to do so from any predetermined counter setting. With the use of an external timer, it also is possible to get the deck to play or to record when unattended. A remote-control accessory also is available.

The cassette compartment toward the left of the panel is covered by a swing-down door fronted with a removable glass section. The eject button and the AC power off/on switch are to the left. Below them is a stereo headphone output jack, and below that are the left- and right-channel microphone input jacks. The record calibrate screwdriver adjustments are below the

cassette compartment.

To the upper right of the cassette compartment is a fluorescent display that includes the digital tape counter and the indications for the double memory system. The counter itself uses three numbers plus four small bars that come on in succession; after the fourth bar lights up, the third digit advances to the next number. Next to this display is an LED that functions as a strobe; controlled by a quartz crystal, it flashes about once each second, and it is suggested for use in calculating record-muting time.

The generously proportioned signal meters consist of two horizontal fluorescent bar graphs, calibrated from −40 to +8. The peak/VU and peak-hold switches are to the right of the meters.

Under the counter and meter area is a long row of controls. First in this row are the memory 1 and memory 2 switches. Next come the transport buttons for record, rewind, stop, play, fast-forward and pause. The record-mute button is next, followed by the four tape-select buttons, the tape/source monitor button and a dual-concentric input level control. All the buttons in this row, except the rewind, stop and fast-forward buttons, have their own indicator lights. The record-mute button must be held down for as long as required to record silence onto a tape. If the pause button is used to stop tape motion during a recording, the tape may be restarted (still in the record mode) by touching the play button. The input level knobs may be used to adjust recording level on each channel individually or on both channels simultaneously.

The bottom row of controls includes the separate bias adjustment knobs for each of the four classes of tape corresponding to those on the tape selectors above; the external timer switch; the Dolby NR switch with a position for MPX filter; the input selector (with positions for microphone, line, 400 Hz and 400 Hz/8 kHz—the last two referring to the deck's test oscillator); a meter light dimmer control; and the output level control which adjusts both output channels on line and on headphone simultaneously. There is no on-the-panel input mixing provision.

The rear of the deck contains the usual line-in and line-out signal jacks, a socket for use with the optional remote-control accessory and the AC line cord. The Technics RS-M95 comes in a metal case finished in black matte, and it may be EIA rack-mounted with a pair of rack angles supplied. Also furnished with the deck is a cover plate that may be fastened to hide the lower row of front-panel controls.

**Test Results:** Allowing for normal variations in test setups, the Technics RS-M95 came in "on spec" in *MR&M's* lab tests. It also is, so far anyway, one of the relatively few cassette decks we have tested in which the use of metal tape does produce better results in most or all of the tested parameters (response, distortion, S/N, headroom). At that, the differences we measured among three tapes tested on this machine were not terribly great, which is to say the deck can do a very creditable job with any of them.

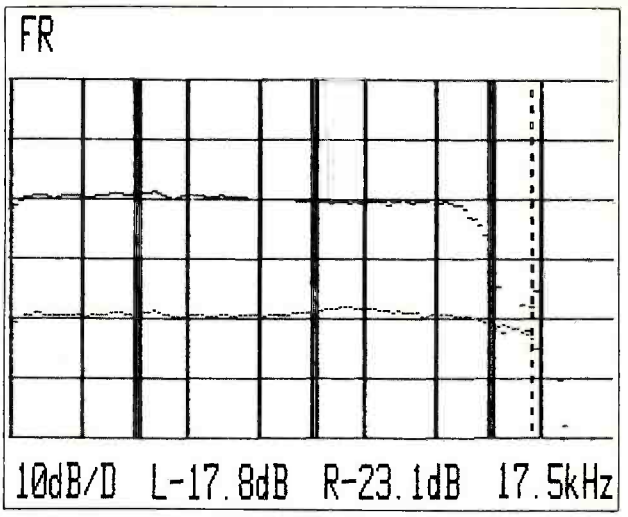



**Fig. 1: Technics RS-M95: Frequency response using Maxell XL-I tape at 0 dB (L) and −20 dB (R) record level.**

Testing this deck gave us an opportunity to use our new Sound Technology Tape Test System for a cassette model as well as for the open-reel Akai also reviewed in this issue, and to formulate procedures for such tests on future decks. So, a brief discussion of Figs. 1 through 8 seems in order.

*Fig. 1* shows a frequency response plot (record/play) at 0 dB and at −20 dB record level, using Maxell XL-1 (normal bias) tape. Note that the test system machine can display only two response curves if one is "left channel" and the other "right channel." In fact, both traces were made for the same (left) channel. We simply reversed inputs between runs. So disregard the L and R notations. Since the reference level for the "R" trace

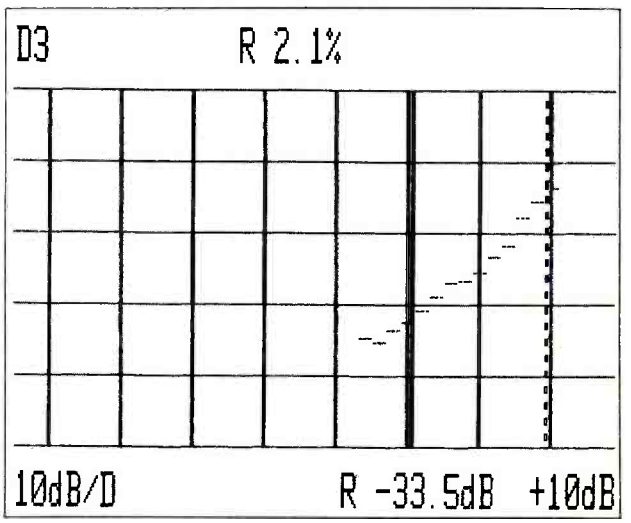



**Fig. 2: Technics RS-M95: Third order distortion at various record levels. Maxell XL-I tape.**

(lower one) was −20 dB, we moved the dotted line "cursor" higher and higher in frequency until we got a readout of −23.1 dB at 17.5 kHz. In effect, that tells us that the −3.1 dB point (close enough to −3 dB) *relative to 20 dB* occurs at 17.5 kHz. That, therefore, is the upper frequency limit reported in our "Vital Statistics" table for this tape. Before allowing our associated video printer to present us with the hard copy appearing as *Fig. 1,* we moved the cursor downward in frequency only to discover that at 20 Hz, response still was well within the −3 dB margin. Thus we report a lower limit of 20 Hz.

In *Fig. 2,* we display 400-Hz tones for this same tape at various record levels up to +10 dB. At that level, 3rd order distortion had still not reached the MOL level of 3 percent. (It reads 2.1 percent at that level, as shown in *Fig. 2,* or an equivalent value of −33.5 dB below the reference level). To arrive at the +12 dB limit for this tape we had to re-set our reference and determine the 3-percent point manually.

All of which leads to one criticism of the meter calibration of this deck. Zero dB corresponds to around 160 nWb/M. That's about 2 dB below Dolby reference level, which makes the deck's "headroom" appear to be enormous (+12 dB for the ferric-oxide tape used). Of course, headroom was good—it just wasn't as good as a +12 dB number might lead one to believe. A figure of +12 dB above 160 nWb/M adds up to an absolute record level of nearly 640 nWb/M! In any event, to establish the signal-to-noise levels shown in our "Vital Statistics," we used the Sound Technology tester to read the S/N value *referred to the decks's 0-dB level,* and then added the 12 dB of headroom relative to the 3-percent THD point. We used the widely accepted CCIR/ARM weighting method endorsed by Dolby Laboratories for all S/N measurements of this product.

*Fig. 3* shows the deck's channel separation capability. Note that the cursor is at 1 kHz, and so a single printed reading of separation at that frequency appears below the display (39.8 dB). But one could easily interpret other separation values at other frequencies by realizing that if the "blip" for 1 kHz is at about −40, then the top of the display represents −10 dB, and distances between divisions are 10 dB as indicated. Double vertical lines appear for frequencies of 100 Hz, 1 kHz and 10 kHz; and the furthest vertical line to the right of the display represents 20 kHz in all frequency plots made by this instrument.

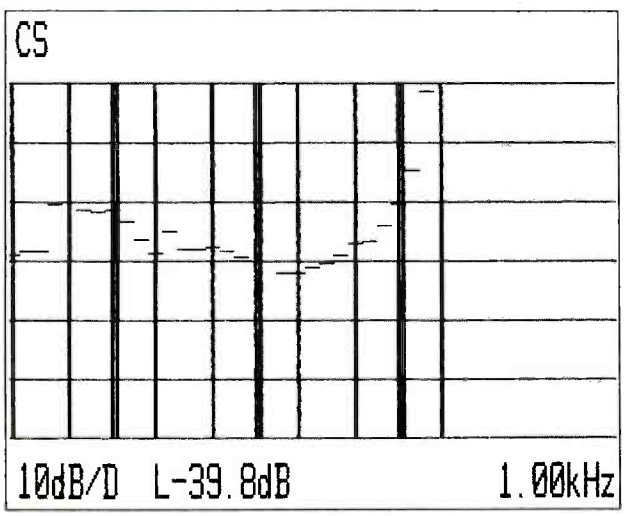



**Fig. 3: Technics RS-M95: Channel separation measured 39.8 dB @1 kHz.**

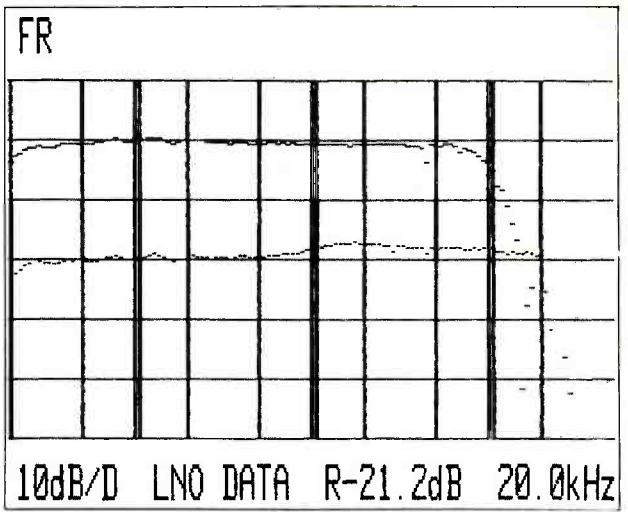



**Fig. 4: Technics RS-M95: Frequency response using Technics "XA" high bias tape at 0 dB (L) and −20 dB (R) record level.**

*Figs. 4* and *5* present information similar to that shown in *Figs. 1* and *2,* but this time we are using Technics XA high-bias tape as our test sample. At 20 kHz, response for the −20 dB trace was at −21.2 dB, or not yet down by 3 dB, so we had to interpolate the −3 dB point as occurring at around 20.8 kHz. Similarly, for the 3-percent distortion level, we had to manually increase levels beyond the +9 dB test point offered automatically in *Fig. 5* to arrive at a record level of +11 dB for 3-percent third-order distortion.

*Figs. 6* and *7* represent the same sort of response and distortion data as the previous figures, but this time the sample tape was Technics MX (metal) with bias settings selected and optimized by the controls on the RS-M95. Response at −20 dB extended no further than it did for the high bias tape (20.8 kHz for the −3 dB point), but when you compare the response at 0 dB for both tapes (upper curves in *Figs. 6* and *4*), the earlier high-end rolloff in response of the ferric-oxide high-bias (chrome equivalent) tape is clearly evident. It becomes even more apparent if you examine the upper curve (0 dB record level) of *Fig. 1* (for normal bias tape), where high-frequency tape saturation at this high recording level is more evident.

Finally, in *Fig. 8,* we plotted the playback-only re-

sponse for left and right channels, using a special test tape developed for use with the Sound Technology instrument. The rather early rolloff suggests that the azimuth positioning of the heads of this deck was not identical to that of the machine that produced the test tape. Needless to say, when using the deck for both record and play, this will have no bearing at all on reproduction results.

**General Info:** Dimensions: 17¾ inches wide; 5⅞ inches high; 13¾ inches deep. May be standard rack-mounted with angles supplied. Weight: 26 pounds, 8 ounces. Price: $1300.

**Individual Comment by L.F.:** This is the sort of cassette deck that appeals to me. It has many sophisticated features, but it is not so laden with useless gimmicks as to make operation overly complex and needlessly confusing. The fine bias adjust for each of the basic tape formulations is fast becoming a must on better decks, as is the record calibration facility. Both of these wanted features are readily accessible on the front panel of this deck. Its designers have rightfully presumed that anyone interested enough to purchase a deck costing more than $1000 will take the time and have the intelligence to use these adjustments as they were meant to be used (even if that means reading the owner's manual!). They concluded, correctly I believe, that those fine vernier controls should be up front where they can be conveniently used most.

The two-memory system associated with the digital readout counter is also a worthwhile feature. I never cared too much for the memory rewind systems that merely bring things back to a predetermined zero on the counter, and I have always suspected that most cassette deck users seldom use that feature. But when that option is coupled with a second memory that can be set to *any* number on the digit counter, the combination can become mighty useful.

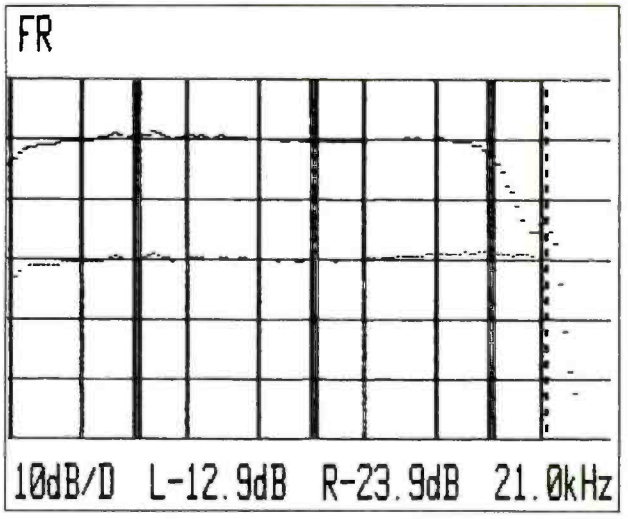



**Fig. 6: Technics RS-M95: Frequency response at 0 dB (L) and −20 dB (R) using Technics MX (metal) tape.**

The transport system, with its light-touch solenoid-operated logic controls, operated flawlessly and handled tape most gently. The three-head system used here includes a combination record/play head in a single shell. That, of course, is more sophisticated than a single head that does double duty for both record and play. There really are two electrically separate heads in the single shell, one for record and the other for playback, but because they are both mounted permanently in the single housing, that means you need not trouble yourself with any azimuth alignment procedure when using this deck. The record and play sections will always "track" each other properly if they were correctly assembled into the single shell in the first place.

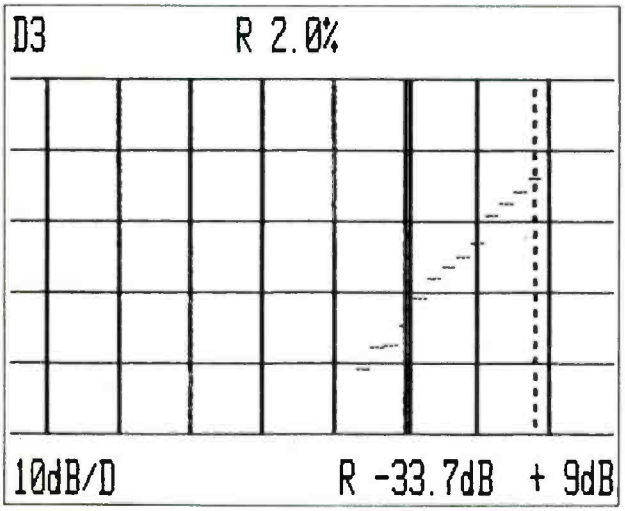



**Fig. 5: Technics RS-M95: Third order distortion at various record levels—Technics XA tape.**

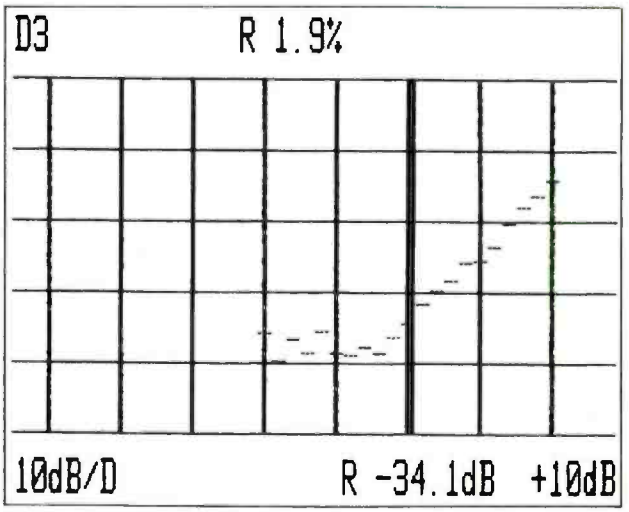



**Fig. 7: Technics RS-M95: Third order distortion at various levels, Technics MX (metal) tape.**

Ours seem to have been.

In summary, I like what Technics has done here, and I consider the Technics RS-M95 to be a good example of what the serious recordist would be looking for in a top-grade stereo cassette deck. I'll take this kind of performance any day over poorer basic performance machines that offer me clocks, timers, ability to locate any number of pauses in tape and you-name-it.

**Individual Comment by N.E.:** With the RS-M95 cassette deck, Technics has reaffirmed—for the cassette tape format—its original "mission" as the high-quality branch of the Panasonic organization. Performance, features and general "product personality" all place this deck in an enviable and competitive position vis-a-vis other recent top-quality decks. While the RS-M95 certainly will appeal to the well-heeled home stereo enthusiast it also has more than a touch of professionalism about it that should really interest the advanced or serious recordist. It has true metal-tape capability; it has a really "sweet" transport system with unrestricted fast-button options; its three-head operation permits off-the-tape monitoring while recording; it has enough, but not too much, adjustments for optimizing recording level and bias tuning; it boasts the twin-memory system that may be used for returning a tape to any preselected (by the operator) point as well as to repeat a portion of the tape.

Speaking of "memory," by the way, if my memory serves, this is actually the third Technics cassette deck we have tested for *MR&M*. The first top-of-the-line model was that big two-piece affair in which the transport and the electronics were separately housed (RS9900US, March 1978). The next one was a much more compact version (RS-M85, January 1979). This present model is about as compact as the second version and it actually outperforms either of its two predecessors. It also costs less than the earlier two-piece system which itself is something remarkable in this day of rising costs. Apparently Technics has got it all together in one well-designed, superbly crafted machine.

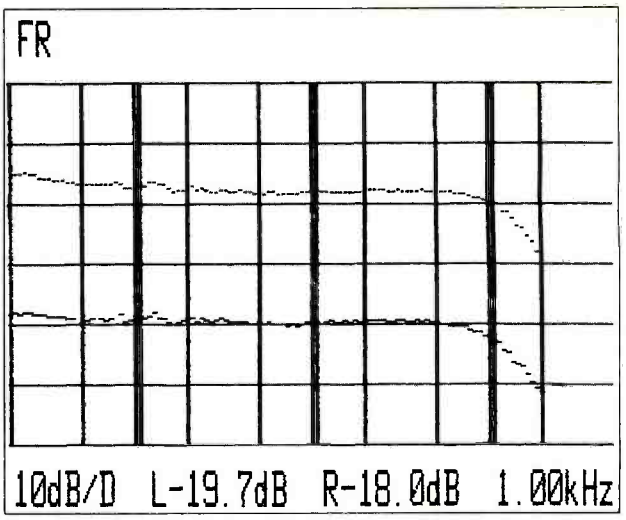



**Fig. 8: Technics RS-M95: Playback-only frequency response, 120 usec EQ, left (upper) and right (lower) channels, at −20 dB reference level.**

I also like the implied product direction suggested by this unit. Technics obviously has eschewed the "automation for automation's sake" approach and instead has concentrated on basic performance combined with those features deemed requisite for optimum cassette-deck operation within a definite price range. Given a choice, I'll take the Technics approach any day.

**TECHNICS RS-M95 CASSETTE RECORDER: Vital Statistics**

| PERFORMANCE CHARACTERISTIC | MANUFACTURER'S SPEC | LAB MEASUREMENT |
|---|---|---|
| Frequency response, normal tape | ±3 dB, 20 Hz to 17 kHz | ±3 dB, 20 Hz to 17.5 kHz |
| high-bias tape | ±3 dB, 20 Hz to 19 kHz | ±3 dB, 20 Hz to 20.8 kHz |
| metal tape | ±3 dB, 20 Hz to 20 kHz | ±3 dB, 21 Hz to 20.8 kHz |
| 3rd order HD at 0 dB record level | | |
| normal; high-bias; metal tape | NA; NA; NA | 0.23%; 0.18%;0.19% |
| Record level for 3% THD | | |
| normal; high-bias; metal tape | NA; NA; NA | +12; +11; +12 |
| S/N ratio, Dolby off | | |
| normal; high-bias; metal tape | NA; 60 dB; NA | 56.1; 59.4; 60.2 dB (CCIR/ARM wtd) |
| S/N ratio, Dolby on | | |
| normal; high-bias; metal tape | NA; 70 dB; NA | 63.4; 68.3; 68.7 dB (CCIR/ARM wtd) |
| Wow-and-flutter (WRMS) | 0.03% | 0.037% (JIS wtd) |
| Speed accuracy | NA | +0.16% |
| Mic input sensitivity | 0.25 mV | 0.23 mV |
| Line input sensitivity | 60 mV | 55 mV |
| Line output level | 650 mV | 650 mV |
| Headphone output level, 8 ohms | 88 mV | 85 mV |
| Fast-wind time, C-60 | 80 seconds | 70 seconds |
| Power consumption | 50 watts | 42 watts |

CIRCLE 33 ON READER SERVICE CARD

# Akai GX-625 Open-Reel Recorder


**General Description:** The Akai GX-625 is an open-reel deck with two-speed (7½ and 3¾ ips) operation and up to 10½-inch reel capability. Head configuration is quarter-track, two-channel stereo with also, of course, the mono option.

The erase, record and play heads are all separate, and the last two are identified as GX types which Akai has been promoting in recent years. Three motors are employed: an AC servo motor drives the capstan, while two AC eddy current motors power the tape reels. The transport keys are "feather touch" and logic controlled; fast-buttoning is possible, including direct recording from the playback mode (run-in recording or overdubbing). The reels are fitted with built-in spring-loaded retainers and are supplied with adapters for handling the pro-size (NAB) 10½-inch reels, one of which is supplied with the deck. Although designed primarily for normal AC line voltage, the Akai GX-625 also may be operated on 12 volts DC (as in a vehicle) for which a special voltage adapter is supplied with the deck.

The tape counter can be switched to show real time elapsed in minutes and seconds. Associated with the counter is an automatic system that functions like a memory-rewind (to get the tape to play or to stop) when the counter reaches "0," or to repeat from any desired portion selected by the operator. Remote control also is possible with the use of an optional accessory, and unattended record or playback is feasible with the aid of a separate external timer.

The deck is laid out in familiar fashion, with all controls readily accessible and clearly labeled. Just above the head cover is a pitch control that may be used to vary tape speed (on playback only) by ±6 percent. The tape counter and its associated controls for the LED digital readout and the rewind options are mounted on the cover. Coming off the supply reel, the tape goes through a tension lever and roller, past the guides and heads under the head-cover, up between the drive-capstan and pinch-roller, past an automatic stop/tape tension lever and onto the takeup reel. At the left is the switch for use with an external timer.

The row of items just below includes the AC power switch, the reel-size selector, a pair of VU meters calibrated from −20 to +5 and the transport keys marked for pause, record, rewind, stop, play and fast-forward.

The bottom row of controls and features includes a stereo headphone output jack; the output level control (which handles playback volume for both line outputs and the headphone jack simultaneously); the tape/source monitor selector; two track selectors (left/mono, right/mono or press both for stereo); a tape selector (wide-range or low-noise); the tape-speed selector; a record-mute button; the microphone input level controls; the line input level controls; and, finally, the left- and right-channel mic jacks. Each of the input controls is a dual-concentric pair permitting channel adjustment in-

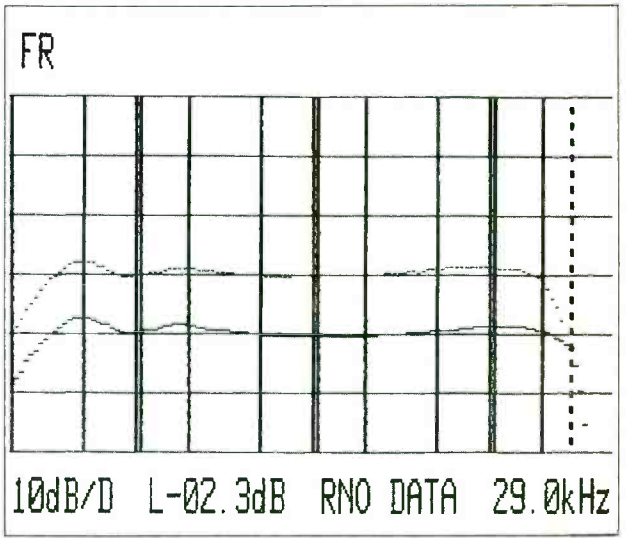



**Fig. 1: Akai GX-625: Frequency response at 0 dB (upper trace) and −10 dB record level, 7½ ips.**

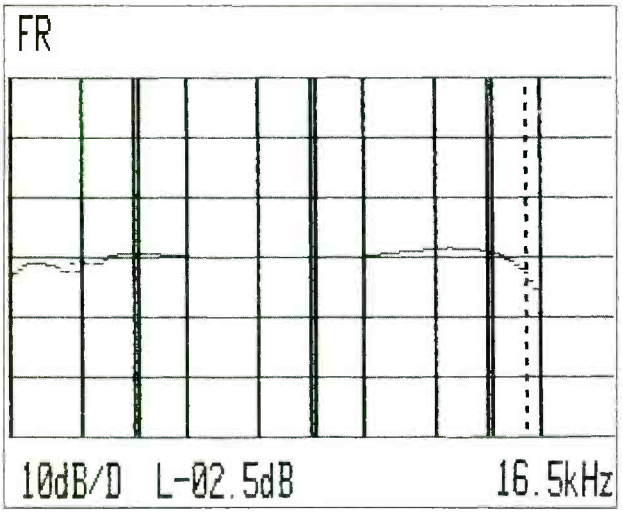



**Fig. 2: Akai GX-625: Frequency response at −20 dB, 3¾ ips.**

dividually or simultaneously. The mic controls also handle an optional DIN input (at the rear). The control arrangement permits input mixing of mic and line. Both sets of controls are friction-coupled so that once individual channel levels are set, both channels (for mic or for line) may be faded in and out readily.

The rear of the deck contains the normal line in and out jacks, the DIN receptacle, the remote control connector, the AC-to-12-volt-DC adapter jack and the AC line cord. The Akai GX-625 is intended nominally for vertical installation, although it could be installed "on its back" if necessary (there are four feet that would raise it from the under-surface in such positioning).

**Test Results:** Published specifications for the Akai GX-625 were—for the most part—confirmed or bettered in our lab tests. At the 7½ ips speed the deck's response went well beyond the claimed high-end and low-end frequencies and with distortion that was lower, and signal-to-noise that was higher, than claimed. Response at the slower speed fell short of the claimed 19-kHz top but did make it to 17 kHz. On the other hand, the slow-speed response did much better than claimed at the low end. Be that as it may, however, the distortion and S/N at the slow speed were again better than spec'd. Tape handling at either speed, and in the fast-wind modes, was excellent. Doubtless the deck's electronic braking action (controlled by a microprocessor which slows the speed before the tape comes to a full stop) contributes to this desiderata.

Incidentally, the tape we used for testing the Akai was Maxell UD-XL. Quite possibly, another tape might have maintained the response out to 19 kHz claimed for the 3¾ ips speed. This of course points up the desirability of having some sort of vernier bias adjust on an open-reel machine. The Akai GX-625 does have two fixed bias settings; we chose the one that yielded flattest response at the higher speed for the Maxell tape.

Regarding the graphic presentation of our test data, a detailed explanation is given in Leonard Feldman's "Ambient Sound" column elsewhere in this issue of *MR&M*. To relate that discussion to the data shown here, however, consider the data shown in *Figs. 1* through *8*.

*Fig. 1* shows record/play frequency response at two different record levels (0 dB and −10 dB) for the 7½ ips speed. The upper trace is the 0 dB response. The lower curve is for −10 dB, and is the one used to establish the −3 dB points listed in the "Vital Statistics" chart. Note that the "cursor" (vertical dotted line) has been moved way out to 29 kHz in *Fig. 2*, and at that frequency the "L" curve (really the −10 dB curve) registers

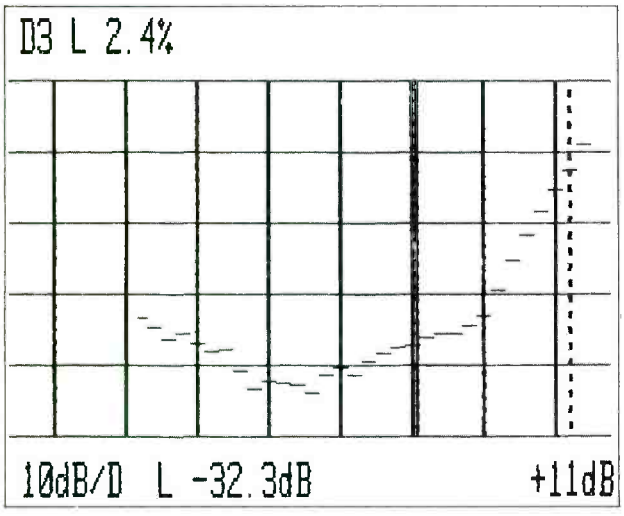



**Fig. 3: Akai GX-625: Third-order distortion, at 7½ ips.**

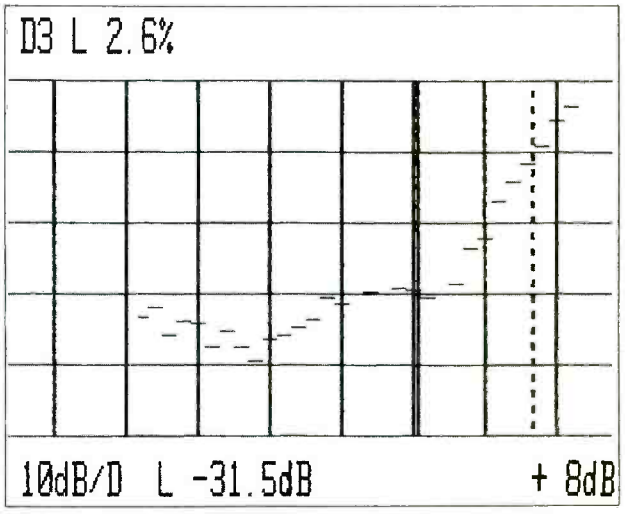



**Fig. 4: Akai GX-625: Third-order distortion, at 3¾ ips.**

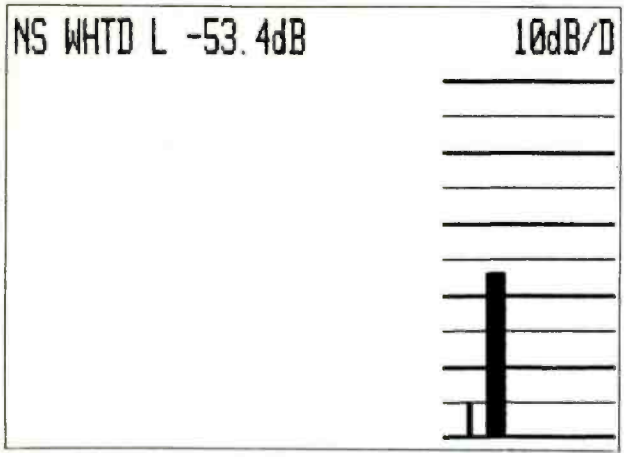



**Fig. 5: Akai GX-625: Signal-to-noise ratio, NAB weighted, at 7½ ips, re: 0 dB reference level.**

−2.3 dB. The "L" and "R" notations, in this instance, do not have their usual meanings of left and right, but are used to designate the two curves taken on the *same* channel but at different reference levels. Note that at the slower 3¾ ips speed *(Fig. 2)*, we had to move the cursor back down to 16.5 kHz, at which frequency the response was down 2.5 dB. The next available discrete frequency to which the cursor might have been moved was 17.5 kHz, at which frequency the response was down by far more than 3 dB. So, in our "Vital Statistics" we interpolated the −3 dB point as being at 17.0 kHz.

In *Figs. 3* and *4* (third-order distortion tests) we have moved the "cursor" to that "blip" or level which produced nearly 3% THD, so as to establish the mid-frequency MOL (maximum output level). For 7½ ips operation, that occurs at +11 dB (relative to 0 dB record level which is always used as a reference level on the test instrument). At the slower speed of 3¾ ips, we reached MOL at approximately +8 dB *(Fig. 4)*. Since actual third-order distortion levels were not quite 3% in both instances, we again interpolated and called the 3% level +11.5 dB for the higher speed and +8.5 for the slow speed. These maximum levels must be kept in mind when examining the signal-to-noise results shown in *Fig. 5* (for 7½ ips) and *Fig. 6* (for 3¾ ips). From these latter figures it would seem that S/N is almost identical for both speeds (−53.4 dB at 7½ ips and −53 dB at 3¾ ips). In fact, that is not true, for to these figures must be added +11.5 dB in the case of the 7½ ips tests, and +8.5 dB for the lower speed tests. That is why the S/N numbers appearing in our "Vital Statistics" show up at 64.9 dB at the 7½ ips speed (53.4 plus 11.5), and 61.5 for the slower speed (53 plus 8.5). We used NAB weighting for these noise measurements.

Finally, *Figs. 7* and *8* provide a direct reading flutter at 7½ and 3¾ ips for this tape deck. Note that the bar-graphs at the right of each figure display instantaneous values of flutter, while the number displayed to the right of the words "FL WHTD L" are averaged readings. Note, too, that in the case of 7½ ips operation *(Fig. 7)* where flutter turned out to be less than 0.3% (NAB weighted), the bar graph scale "auto-ranged" so that 0.03% is full scale, while for the slower speed, where flutter was 0.036%, the bar graph automatically adjusted its scale so that full-scale was now 0.1%.

**General Info:** Dimensions are 17.3 inches wide; 17.6 inches high; 9.5 inches deep. Weight is 38.9 pounds. Price: $749.95

**Individual Comment by L.F.:** Akai has managed to cram a number of useful features and some excellent performance into this relatively low-cost open-reel deck. I know of no other deck of this type that provides real-time tape usage indication (using a digital display, no less). The deck's illuminated full-logic controls, standby indicator, sound-on-sound capability, dual monitoring and variable pitch control (during playback) are all features usually found on more expensive machines. The same applies to the 10-inch reel capacity.

One slightly confusing front-panel treatment is the line shown between the output level control knob and the nearby headphone jack. My first thought was that this control varied level at the headphone jack only. In fact, it controls line-level output levels too. I am not against that (it is a plus feature), but I feel that the deck's graphics should therefore not have tied the phone jack to that control.

A dynamic range capability of nearly 65 dB in stereo for a quarter-track machine operating at 7.5 ips (with no electronic noise reduction) is pretty darn impressive, especially when the machine sells for around $750. Personally, I am glad to see that the tradeoffs that must necessarily be incorporated to meet such a price involved front panel frills rather than down-to-earth basic performance.

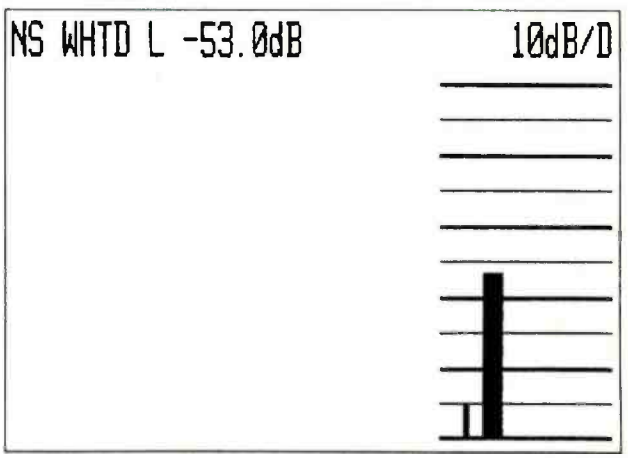



**Fig. 6: Akai GX-625: Signal-to-noise ratio at 3¾ ips, re: 0 dB record level.**

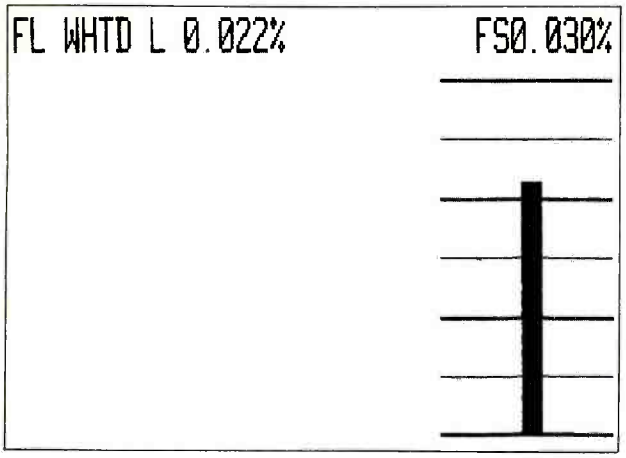



Fig. 7: Akai GX-625: Flutter, NAB weighted, at 7½ ips, record/play.

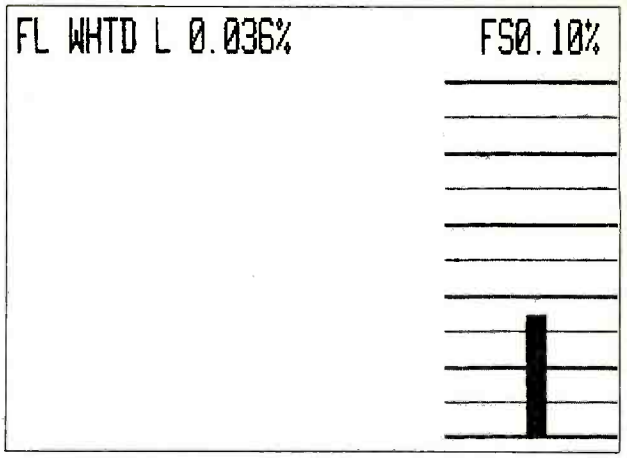



Fig. 8: Akai GX-625: Flutter, NAB weighted, at 3¾ ips, record/play.

**Individual Comment by N.E.:** Now here's a switch—there are all those fancy microprocessor-automated cassette decks that do everything but mix drinks (and the way they are going I shouldn't be surprised if we get one delivered that has logic-controlled buttons marked "rye," "bourbon," "Scotch," "wine-red," "wine-white" and "wine-rose"), with prices soaring to and beyond the $1500 level—and the best they can do *in strictly audio terms* is come close to what a good open-reel deck priced at half that amount can do.

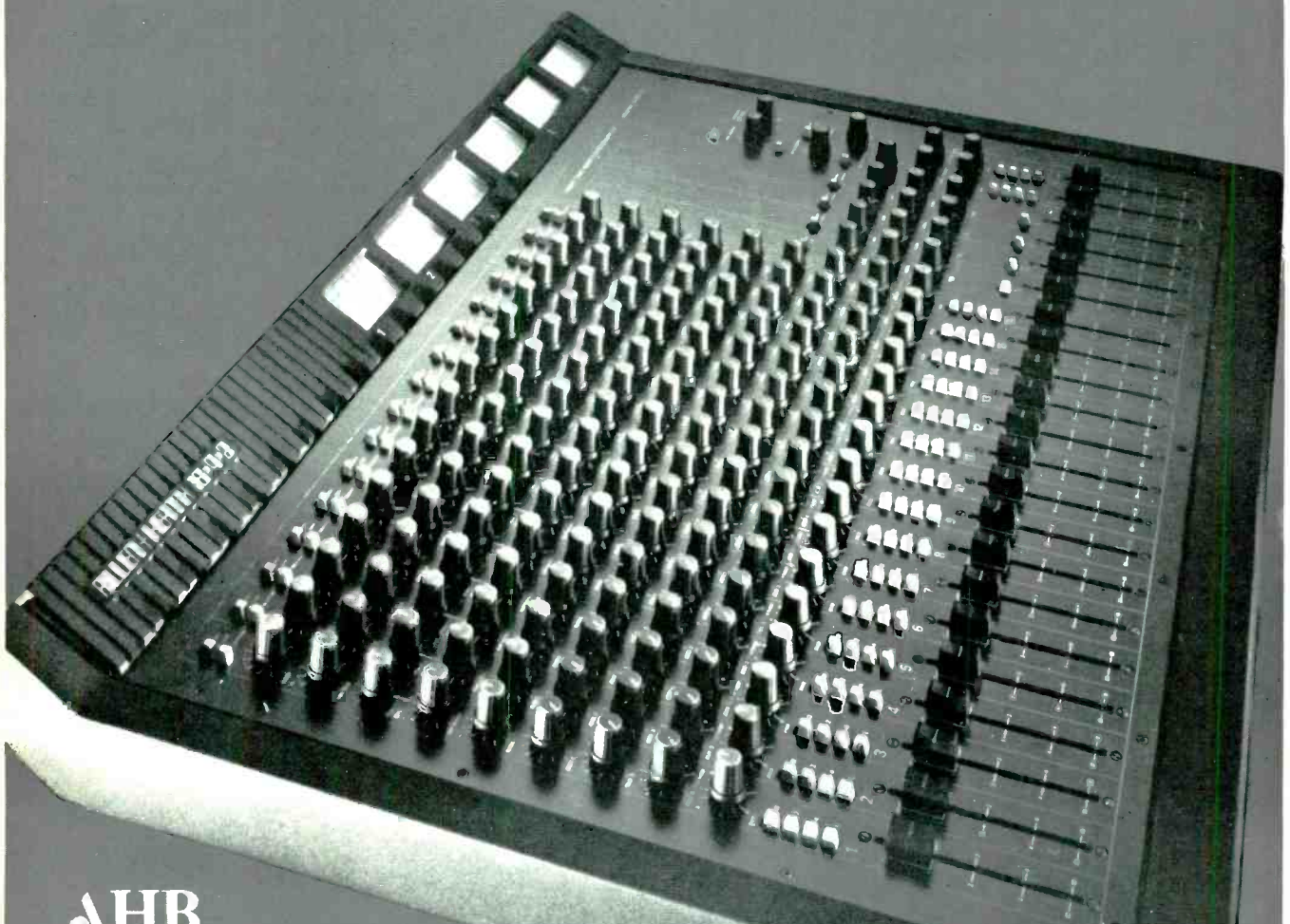
Somewhere along the line this bit of intelligence must have seeped into the thinking and planning of at least a few tape-recorder manufacturers (e.g., Pioneer, Teac and now Akai), because we are obviously beginning to get really good open-reel decks that are relatively compact, offer splendid response, operate flawlessly in all modes, boast a good measure of "professionalism" in terms of transport options, metering and so on and—wonder of wonders—actually cost significantly less than those "2001-inspired" cassette machines. If this keeps up, then the cassette deck and the open-reel deck may wind up interchanging the roles that have been "assigned" to each—the cassette deck may eventually become the plaything of the well-heeled specialist, and the open-reel deck may become the no-fooling-around recording "tool" of the masses.

Whatever you make of this sociologically or culturally or psychologically, there is the fact of this new Akai model GX-625 which offers wide-range response at low distortion and with fine S/N characteristics, and it does so without the aid of noise-reduction or metal tape, and at about half the price of some of the newest cassette decks. Aside from the obvious appeal of such a deck to the serious, creative recordist, one can only wonder about the effect such a product will have on the "mass" recorder market which may suddenly wake up to the fact that it is possible to buy superior performance at lower cost. Heavens to Betsy!

**AKAI GX-625 OPEN-REEL TAPE DECK: Vital Statistics**

| PERFORMANCE CHARACTERISTIC | MANUFACTURER'S SPEC | LAB MEASUREMENT |
|---|---|---|
| Tape speeds | 7½; 3¾ ips | Confirmed |
| Reel capacity | 10½ inches | Confirmed |
| Wow/flutter | 0.03%; 0.04% | 0.022%; 0.036% |
| THD at 0 VU, 7½; 3¾ | >0.5% | 0.14%; 0.33% (3rd order) |
| Record level for 3% THD, 7½; 3¾ | NA | +11.5 dB; +8.5 dB |
| Frequency response, 7½ | ±3 dB, 30 Hz to 26 kHz | ±3 dB, 26 Hz to 29.2 kHz (at −19 dB) |
| 3¾ | ±3 dB, 30 Hz to 19 kHz | ±3 dB, 20 Hz to 17 kHz (at −20 dB) |
| Best S/N ratio, std tape; 7½; 3¾ | 62 dB; 60 dB | 64.9 dB; 61.5 dB |
| Fast-wind time, 2400-ft | 130 seconds | 120 seconds |
| Mic input sensitivity | 0.25 mV | 0.23 mV |
| Line input sensitivity | 70 mV | 66.1 mV |
| Line output level | 775 mV | 700 mV |
| Headphone output level | 100 mV | 84 mV |
| Erase ratio | NA | 105 dB |
| Speed accuracy, 7½ | ±0.8% | ±0.4% |

# Britain's Allen&Heath introduce a new dimension to mixers with The A&H 16-4-2



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# Phase Linear X20 Active Crossover

**By John Murphy and Jim Ford**


We continue to direct our attention to loudspeaker crossover systems this month as we review the X20 active crossover from Phase Linear. The X20 features 18-dB-per-octave Butterworth filter pairs (for an accurate summed amplitude response) along with front panel control over input and output levels and crossover frequency. Packaged in a single space (1¾ inch) rack mount chassis, the unit is primarily intended for stereo two-way operation but can easily be used as a three-way mono crossover.

Special features of the unit include delayed turn-on and an overload indicator with a user adjustable threshold. The price of the X20 is $349.

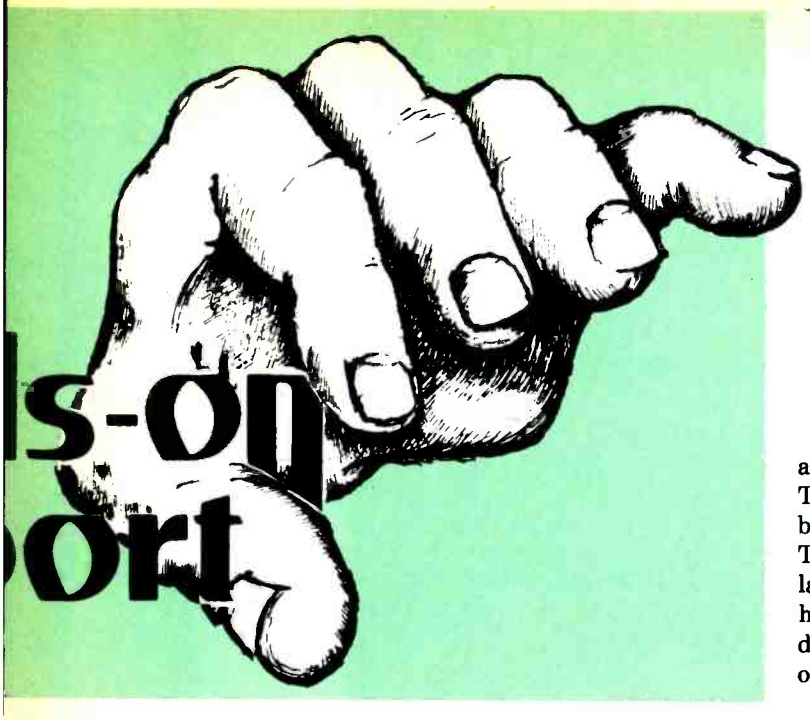
**General Description:** As an active crossover, the X20 is designed to divide a full spectrum audio signal into frequency bands that will be reproduced separately by different loudspeaker drivers. That is, the individual "high" and "low" output signals for each channel of the X20 would be routed to separate power amplifiers with the output of each amplifier connected directly to the appropriate loudspeaker driver (or drivers) for that frequency range.

The initial design of any multiamplified loudspeaker system includes the selection of one or more crossover frequencies. The selection of crossover frequency is made with knowledge of the frequency response and power handling characteristics of the individual loudspeaker drivers used in the system. (See the Hands-On Reports in the August and September 1980 *MR&M* for a discussion of crossovers and multiamplification). Like many crossover units currently on the market, the X20 provides front panel controls for establishing the crossover frequency. These controls make it very easy to set the proper crossover frequency when the loudspeaker system is initially put together. We see this convenience as a double edged sword, however, since those knobs almost *beg* to be twiddled by the eager soundman! At the mere push of a button the crossover point of a high-powered sound system can be changed from 1500 Hz to 150 Hz—at which time the treble horn drivers smoke and flame their way to a quick death. The point is that a readily "tunable" crossover can pose a threat to the well-being of a loudspeaker system unless the tuning controls are somehow guarded from the fiendish knob twiddlers. In the case of the X20 and similar units, about the only way to guard the tuning controls would be for the user to fashion a custom security cover for the front panel or to place the unit in an equipment rack with a lockable cover.

Now let's examine the X20 in more detail, starting with the front panel. The controls for each of the two channels are identical, with those for channel A occupying the left half of the front panel and those for channel B, the right half. Since the basic format of the X20 is "two-way stereo," there is one crossover point and therefore two output signals ("high" and "low") for each channel.

Located at the far left of the front panel is a push-button power on/off switch. Above this switch is a green LED labeled "Ready" which indicates the status of the output muting relay (rather than serving as a simple pilot light.) Because of the unit's delayed turn-on feature, no signal appears at the output for about the first five seconds after the power switch is actuated. This insures that any "pops" or "thumps" associated with power-up do not appear at the output where they could be passed on and heard through the loudspeakers. The "ready" LED follows the status of the output relay and therefore illuminates only after a short delay.

Just to the right of the power switch is a rotary-type input level control for Channel A. The settings of this control are clearly indicated in steps from 1 to 10. Because this attenuator is located before the first active stage it is virtually impossible to overload the in-




put; the input level control is simply backed off until the signal is reduced below the overload threshold. Full clockwise rotation of this control results in a total signal gain of 12 dB so that weak signals may be boosted in level. Unity gain occurs at a setting of about 6½ (with the output attenuators at full level). Above and to the right of the input level control is a red LED overload indicator. This LED illuminates whenever the channel signal exceeds a threshold level. This threshold level can be set by the user (by way of a rear panel adjustment) as high as 7.6 volts rms (about 1.5 dB below clipping) or as low as 0.4 volts rms. With the threshold set to maximum the LED functions as a normal overload indicator. However, turning the threshold adjustment fully counterclockwise lowers the threshold to the extent that the LED serves as a "signal preset" indicator. Phase Linear also gives instructions for calibrating the overload indicator's threshold so that it can serve as a remote clipping indicator for the power amps that follow the crossover.

Continuing to the right across the front panel, the next pair of controls for channel A is used to select the crossover frequency. These controls consist of an eleven-position rotary switch that is labeled with frequencies from 150 Hz to 1.2 kHz, and a push button labeled "X10" which switches the action of the frequency control to the range 1.5 kHz to 12 kHz. The use of a rotary switch to select sets of precision tuning resistors results in accurate and repeatable crossover frequency selection.

Next are a pair of rotary output level attenuators for adjusting the levels of the high and low output signals. These controls are normally used to set the final balance between the low- and high-frequency signals. The last control for the channel is a push-button switch labeled "Invert" which reverses the polarity of the high-frequency output signal. The set of controls just described for channel A are duplicated on the right half of the panel for channel B.

Located on the rear panel of the X20 are all of the signal input/output connectors as well as a socket for the detachable line cord. Signal connections are all made by way of ¼-inch phone jacks with both the inputs and outputs designed for unbalanced (single sided) signals. Also, for each channel there is a small screwdriver access hole which allows adjustment of the threshold for the overload indicator described above. All connections and adjustments are clearly identified by the rear panel graphics.

**Listening Test:** In order to evaluate the audio quality of the X20 we interfaced it with our monitoring system and performed our usual listening tests. The high- and low-output signals were combined using a summing amplifier specially designed for crossover evaluations. The signal from our preamp's "tape out" was patched to the input of the crossover; the outputs from the crossover were connected to two inputs of the summing amplifier; then the output of the summing amplifier was returned to the preamp's tape monitor input. By punching the tape monitor switch in and out, we could then alternately insert the crossover (and summing amp) into our listening chain or bypass it in an A/B fashion. We were careful to set the input level control so that there was no detectable change in loudness when the unit was switched in and out of the chain. We carefully listened to several different album cuts that we were well acquainted with and varied the crossover frequency over the full range of settings. The result of our listening was that we could not detect any significant degradation of audio quality when listening through the X20. It was quite transparent as long as signal levels were kept below the overload point.

**Lab Test:** When we took the X20 to the lab, we put it through our test routine paying special attention to the response characteristics when the high and low out-

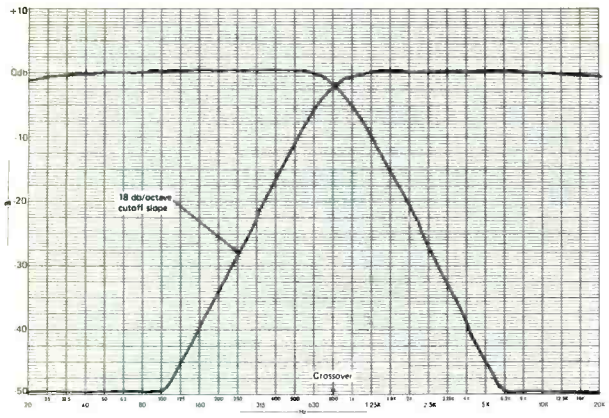



**Fig. 1: Phase Linear X20: Amplitude response of the high and low outputs of the unit for a crossover point of 800 Hz.**

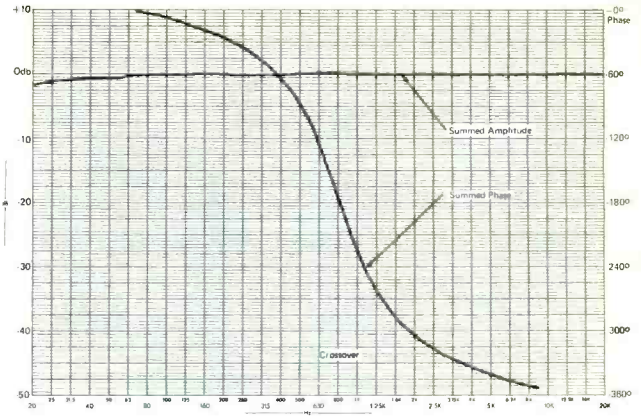



**Fig. 3: Phase Linear X20: Amplitude and phase response of the summed output (polarity invert switch *in*).**

puts were summed. The specific results of our tests are provided in the "Lab Test Summary" below.

We are pleased to report that due to its use of 18-dB-per-octave Butterworth filter pairs, the X20 exhibits a flat amplitude response when the high and low outputs are accurately summed as they would be in a well-designed loudspeaker system employing high-quality drivers. The frequency response curves for the individual high and low outputs are shown in *Figure 1* for the case of an 800-Hz crossover frequency. The curves show a good wide range response and the sharp cutoff that is characteristic of Butterworth-type filters.

*Figure 2* shows both the amplitude and phase response that result when the high and low outputs are summed with the units "invert" switch in the "out" position. Note that the amplitude response is quite flat and that the phase response changes smoothly from about 0° to −180° with increasing frequency. The response curves shown in *Figure 3* are for the same conditions as *Figure 2* except that the "invert" switch on the front panel of the X20 was depressed. Again the amplitude response is quite flat, but now the phase response varies from about 0° to −360°. This is twice the phase shift of the case with the "invert" switch "out." Based on this observation, we would recommend operating the X20 with the "invert" switch in the "out" position. In the event that the loudspeaker system is wired with the polarity of the low- and high-frequency drivers reversed then it would be appropriate to depress the "invert" switch so that the least phase shift (−180°) characteristic can be obtained. It's nice to know that regardless of the relative polarities of the drivers or the status of the "invert" switch, the summed amplitude response will always be flat. This is one of the benefits of using 18-dB-per-octave Butterworth filters.

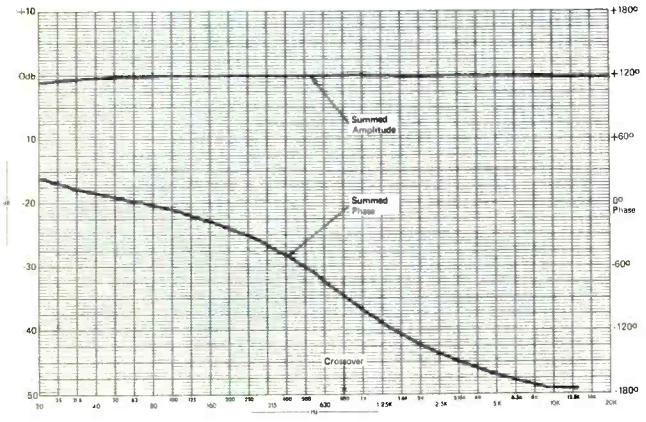



**Fig. 2: Phase Linear X20: Amplitude and phase response of the summed output (polarity invert switch *out*).**

The X20 also did well with regard to other performance characteristics. The maximum output level of +20 dBV is enough to drive any normal power amp to full output. At about −85 dBV, the noise at the outputs was quite low. Total harmonic distortion (THD) was also very low.

We observed that the X20 could be driven into slew limiting with a full-power, high-frequency sine wave. Above 22 kHz (the full power bandwidth) a full-power sine wave was quickly transformed into a triangle waveform exhibiting a slew rate limit of 3.0 volts per microsecond. Dividing this number by the peak output voltage (12 volts) gives a normalized slew rate limit of 0.25 volts per microsecond per volt. As we've mentioned before, the minimum recommended value for normalized slew rate limit is from 0.5 to 1.0. Limits in this range or higher assure freedom from slewing induced distortion.

The preliminary owner's manual that we received with the X20 provided adequate information on the unit's controls and signal connections. The manual also showed how to configure the unit for mono three-way

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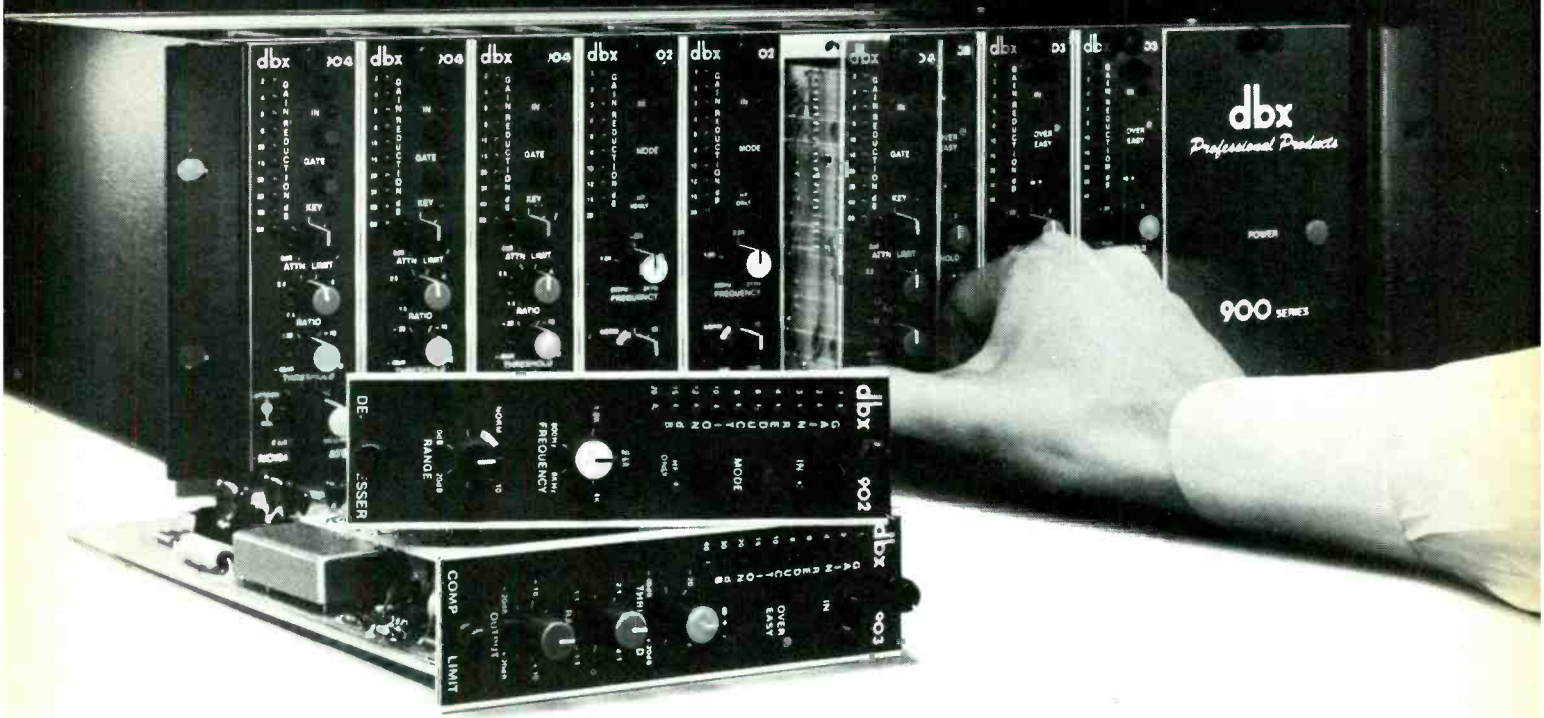


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use along with several other unusual configurations for obtaining 36 dB/octave cutoff slopes.

**Conclusion:** All things considered, we found Phase Linear's X20 active crossover to be a fine unit. Unlike many crossovers in current use, the X20 is capable of delivering a flat amplitude response when the outputs are accurately summed. We must admit that the lack of any guarding of the front panel crossover frequency controls bothers us somewhat. Despite this, we recommend the X20 as an excellent general purpose electronic crossover.

---

**LAB TEST SUMMARY**

*(Note:* 0 dBV is referenced to .775 vrms)

**Input/Output Levels**

Maximum input level before clipping: Depends on setting of input level control

Maximum output level before clipping: +20.7 dBV

**Noise Performance**

(20 kHz filter, unweighted; crossover point at 800 Hz; all level controls at maximum; 600-ohm source impedance)

Noise at "High" output: 87.5 dBV

Noise at "Low" output: 84.4 dBV

**THD Performance**

(+10 dBV output level)

| Frequency | THD & Noise | |
|---|---|---|
| 100 kHz | .0023% | (at "Low" output) |
| 500 kHz | .0024% | (at "Low" output) |
| 2 kHz | .0057% | (at "High" output) |
| 10 kHz | .022% | (at "High" output) |
| 20 kHz | .047% | (at "High" output) |

Small signal bandwidth: 10 Hz to 72 kHz, ±3 dB

Power Bandwidth: 22 kHz

Slew Rate Limit: 3.0 Volts per microsecond

Normalized slew rate limit (*see text*): 0.25 volts per microsecond per volt

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The Crown PZM™ places a small pressure transducer into the primary boundary pressure zone, eliminating the possibility of phase-induced interference. The PZM concept thus provides a significant improvement in signal quality. Its small profile also improves microphone aesthetics.

The PZM pickup pattern is hemispheric, with no "off-axis" position.

Singers and speakers can move more freely around the PZM. Gain related to distance will change, but not tonal quality.

The PZM responds accurately to SPL up to 150dB. You can put it right inside a drum, a bass fiddle, or a piano.

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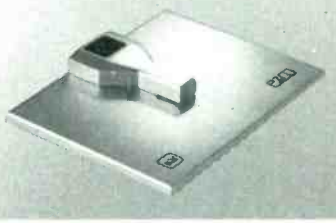



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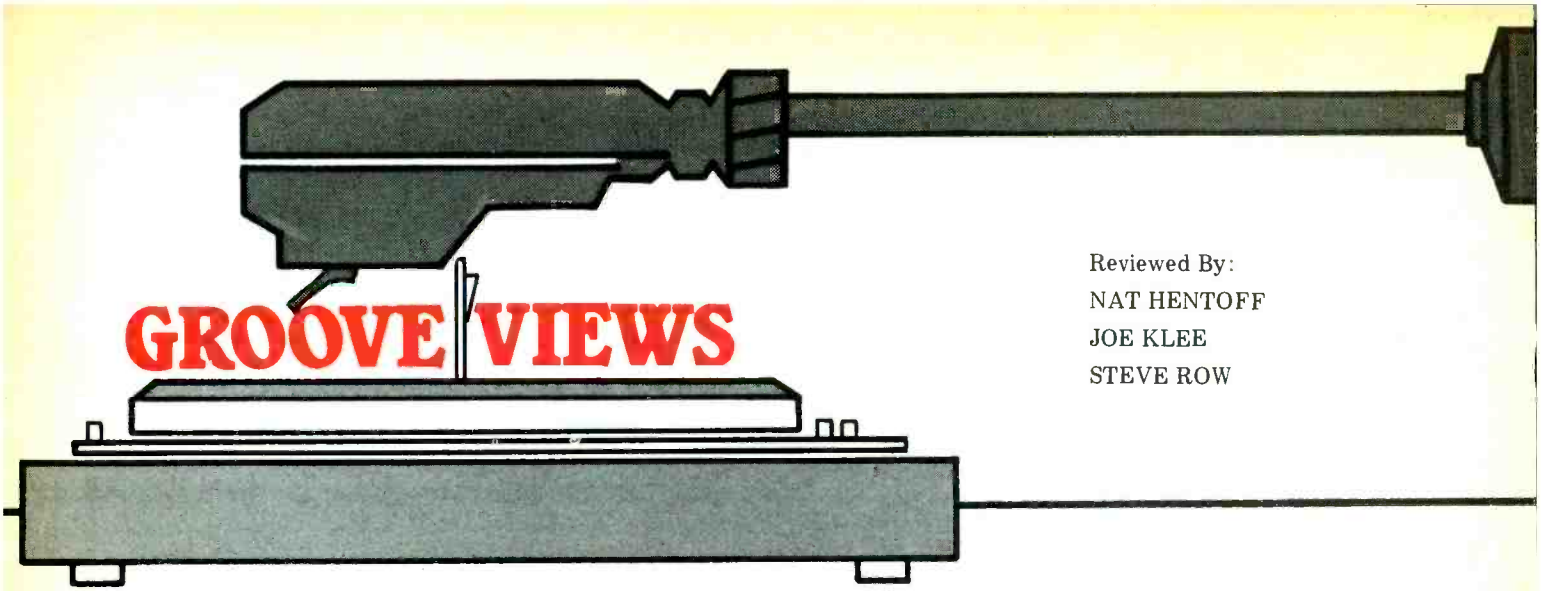


Reviewed By:
NAT HENTOFF
JOE KLEE
STEVE ROW

## POPULAR

**THE ROBBIN THOMPSON BAND: *Two "B's" Please.*** [Ken Brown, producer; Joe Hroner, engineer; Bobby Tulloh and Carlos Chafin, assistant engineers; recorded at Alpha Audio Studio, Richmond, Va.] Ovation Records OV 17590.

Performance: **Spirited, good-time sounds**
Recording: **Would make a New York studio proud**

Hundreds, maybe even thousands, of them exist around the country, but you probably never heard of them. Performers tend to gravitate toward New York or Los Angeles (country performers toward Nashville), and as a result, the local recording studio is used for little more than a place to record local radio jingles.

But if this recording is any indication, there is a wealth of production talent and equipment out in the "boonies" that could be used to a much greater extent than is the case at present. And big name artists would not turn out an inferior product if they chose to record in *the hinterlands*.

Robbin Thompson's third album was produced in his hometown, and the result is a finely done, slick, professional-sounding effort in the area of up-tempo, Southern California rock. Thompson, a past winner in the American Song Festival, doesn't have the strongest voice in the world, but he is an engaging entertainer, accomplished songwriter and smart enough to surround himself with some top-flight musical talent.

This time out, he has called on the talents of Timothy Schmit of the Eagles and Rick Roberts of the Flying Burrito Brothers and Firefall to back him up vocally, and a few members of the Richmond Symphony to back him up instrumentally, and the end result is really fine. The Robbin Thompson Band consists of Michael Lanning on bass, Velpo Robertson on guitar, Bob Antonelli on drums and Eric Heiberg on keyboards, and together with Thompson and his guitar, they produce a tight, spirited ensemble sound.

All but one of the nine tracks are Thompson originals; the ninth is "Rock 'n' Roll Singer," an earlier song by Ace. One of the titles will seem familiar—"Even Cowgirls Get the Blues—" but this is not the song of the same name recorded recently by Emmylou Harris. In some respects, particularly the lyrics, this is better than the Harris song.

Two cuts are particularly noteworthy for their ability to blend good lyrics, catchy melody and outstanding production techniques. "Brite Eyes" is a folk-flavored soft rock song that features a catchy chorus and good instrumental backup. "Candy Apple Red" is a classic late 1960s rock song out of Southern California with a witty chorus enhanced by Antonelli's bass voice growl and a great evocation of surf-and-cars-and-girls that was part of growing up a decade and a half ago. One of the lines is a zinger: "Lookin' for a home run/gettin' thrown out at the plate."

The separation of instruments is good throughout the album, and they don't tend to rumble together; each maintains its own distinct voice. The vocals are clear, even when several voices are added to choruses. Thompson's voice needs to be out front as much as possible, and the engineers

**THE ROBBIN THOMPSON BAND: A local blend of universal talents.**

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GREY STAR
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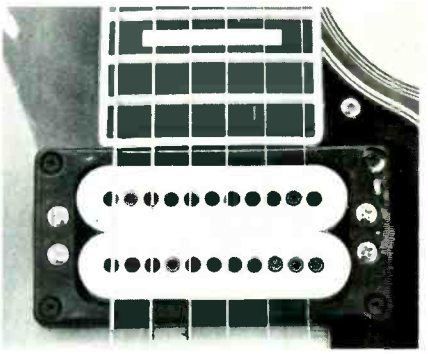

The M22's are now selling for $44.95 (Direct Only) List $85. If you do not wish the 4 wire output for Dual-to-Single Coil & Phase Switching, you may deduct $5.00 ($39.95) and specify M22-S. All M22's feature a built-in copper shield to prevent R.F. and Hum interference.

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|---|---|---|
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| | Your Price | list |
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The TB8 CARVIN Bass Tail Piece & Bridge is made from heavy solid brass which eliminates bridge flexing and adds to the overall rich tone quality of the guitar. Precision milled slots couple the strings securely to the body. Precision string saddles are fully adjustable for both height & intonation. Fits guitars with 2 1/6" space between saddles centers.

| | Your Price | list |
|---|---|---|
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SAVE $25.20 on a complete GUITAR KIT-10 which includes (2) M22 Pickups, (2) P1 Controls, (2) P2 Controls (1) S1 Switch, (1) S3 Switch, (2) S4 Switches, (1) L11 Jack, (1) 12A Jack for stereo wiring, (2) C1 Capacitor, (4) K1 Knobs, & (1) WS1 Wiring & Shielding Kit. PRICED SEPARATELY $145.15 #KIT-10 PRICE $119.95.

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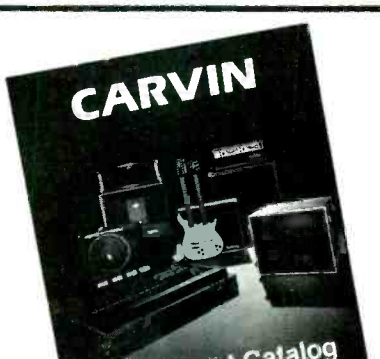


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have kept it out front. Despite all the musical things that happen on some of the cuts, Thompson is never smothered.

Before being picked up by Ovation, this record was available from a local label, Richmond Records, and sold a respectable 20,000 copies, mostly up and down the East Coast. What the album represents, however, is what really is important—a polished effort by local musicians, recording their own material at a local studio, sounding as good as albums coming out of the better known studios (where the wait for studio time can be quite lengthy). S.R.

**TASTY LICKS: *Anchored To the Shore.*** [Producer not listed; Paul Muffson and John Nagy, engineers; recorded at The Mixing Lab, Newton, Mass.] Rounder 0120.

Performance: **Tasty Licks, indeed**
Recording: **Tasty as well**

Never before has a group been so aptly named. Taste is just what these four country singers and musicians are all about. It may not be as spectacular as some of the more electrified fusion bluegrass players but it's a lot more listenable and, at least in my opinion, enjoyable. There are many delights here none the least of which is Jack Tottle's sensitive mandolin work. Liner annotator Frank Godbey calls up the names of Bill Monroe and Jessie McReynolds for comparison's sake. Frankly, I haven't heard Monroe play this tasty in many years and McReynolds, while he may out-flash and out-spangle Tottle, lacks his lyricism and versatility. Versatility is unusual in a bluegrass combo. Usually they do what they do well but that's all they do and any tune they meet with—whether it's a Scruggsy breakdown, a rock tune or a jazz standard—is forced into the configurations and patterns of three-fingered banjo picking and fast-fingered mandolin work. It takes musicians with imagination and restraint to come down for a ballad the way Tasty Licks does on "A Fool Such As I."

Some of this versatility comes from one of the new kids on the band, Pat Enright, whose high lonesome tenor singing can call to mind images of the last yodelin' brakeman, Jimmie Rodgers. Unfortunately, Pat throws in an attempt to capture Rodgers' yodelin' which falls short of the mark

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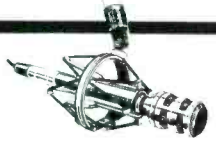

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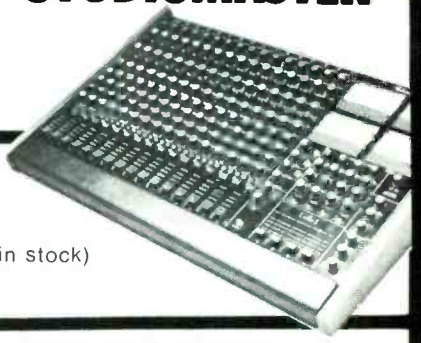

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but then so did Cisco Houston's and everyone else's. A yodeler like Jimmie Rodgers comes along only once in a millenium and while Enright doesn't make it the way Jimmie did, it's a warm heartfelt tribute and "The Last Blue Yodel" is one of the best cuts on this LP.

Other tracks that I enjoy especially are Jack Tottle's tune "Leavin'," the traditional sounding (if not quite traditional) "Slow Train Through Georgia," Hank Williams' "Weary Blues From Waitin'" and Tottle's mandolin feature, "East Tennessee Rag."

When done up by Tasty Licks, bluegrass can be an enjoyable and infectious music that makes you tap your foot and smile. I hope this is not just a small taste because I look forward to more of this kind of music. It's a good antidote for the kind of amplified, whining, maudlin lamentations that pass for country music much of the time these days.

If there are complaints, they are minor ones. Some of Tottle's originals don't sound all that original. "Ain't Nobody Cryin' But Me," for example, smacks of "Railroad Bill" and a couple of other old tunes but that's the folk process for you. Also, in a band with two tenors like Tottle and Enright, it would be nice if the liner notes would identify who sings lead on what song. You can just about guess once you've heard Enright, who obviously takes the lead on "The Last Blue Yodel" and "A Fool Such As I," but there are times like those on "Weary Blues From Waitin' " when it's hard to tell. It sounds like Tottle to me, but it could be Enright. J.K.

## JAZZ

**THE HEATH BROTHERS:** ***Live at the Public Theater.*** [George Butler and M'Tume, producers; Stan Tonkel, engineer; recorded in 1979 at the Public Theater and at CBS Studios, New York, N.Y.] Columbia FC 36374.

Performance: **Alive and Heathy**
Recording: **Better in the studio, but satisfactory throughout**

The Heath Brothers have always played a good boppish brand of contemporary non-fusion music. Now that

drummer Albert has formed his own combo, it's an even more straight-ahead trip. Percy and Jimmy now have firm control of the music and with Jimmy's ability to write and arrange and Percy's supreme artistry (both on full-size bass violin and baby bass) it makes for a marvelous mix. The icing on the cake is the funky guitar work of Tony Purrone and the inventive, if sometimes a bit out in left field, keyboard playing of Stanley Cowell. The drumming of Akira Tana is, in its own way, every bit as vital and exciting as Albert Heath's and is more in line with the directions that the other brothers have taken their music. Two further percussionists have been added but they aren't needed; in fact, I think they get in the way of the straight-aheadness of the music more than they help it along.

This album shows both of the remaining Heath Brothers off in their best light. Jimmy, one of the better composers and arrangers on the current New York bebop scene, has two fine new charts, "A Sassy Samba" and the incredibly gutsy "For The Public" in addition to a reworking of one of his classics, "Cloak And Dagger." On both soprano saxophone and tenor saxophone, as well as flute, Jimmy still shows the influence of Charlie Parker. His nickname of Little Bird was well earned and he shows it here with much playing that originated in Bird, including the famous quotation from *Carmen* that Bird liked to throw into just about every spot he could. What it was about that little snatch of the Habanera from *Carmen* that so attracted Charlie Parker eludes me, but he used it whenever and wherever possible and the surprise is that it always worked. If it works less well for Jimmy Heath, well, it's a generation removed. Besides, Jimmy's major importance is as a writer rather than as a player. The same can be said of Thelonius Monk, Tad Dameron and Duke Ellington so Jimmy is definitely in good company.

With Percy Heath it's another thing. The quiet one in The Modern Jazz Quartet has blossomed into a very exciting bass soloist who plays swingingly and with a sense of humor that has been lacking in jazz of late. His tour de force is a feature for the Baby Bass (a cello body with a wider fingerboard and tuned like a regular string bass) called "Watergate Blues." It's a funny

and funky cut with the audience catching the spirit in unison hand-clapping. Although Gary Giddens complains about the audience rhythm section in his liner notes, I maintain that they're more on target than your average audience of hand clappers and finger poppers and it shows that everybody's having a good time . . . that's important, too.

Never having been inside the Public Theater I cannot vouch for the acoustics of same. There are always inherent problems in recording a "live" jazz concert and I can only point out that Stan Tonkel has done a fine job of getting a good recorded sound down on this LP whatever he may have had to work with—or against—in the process. "For The Public" was recorded in the CBS studios a few weeks later and while the sound is clearer and more distinct in the studio under more easily controlled conditions, there's nothing wrong with any of the sound on this disc . . . and there's nothing wrong with the music either for that matter.

J.K.

**GARY LAWRENCE AND HIS SIZZLING SYNCOPATORS: *Gary Lawrence and His Sizzling Syncopators.*** [Steven Epstein, producer; Bud Graham and Milt Cherin, engineers; recorded at Columbia's 30th Street Studios, New York, N.Y.] Columbia CBS M 35824.

Performance: **Keepin' the music alive**
Recording: **Another wonderful 30th Street sound**

Band one, side two, of this LP sums up the Gary Lawrence philosophy, "Stayin' Alive." It's a twenties dance band version of a seventies disco hit and that has been the Gary Lawrence success formula since he started the Sizzling Syncopaters back in the mid '70s. Their first recording for Blue Goose included a twenties version of "The Hustle" and this latest LP by the band includes the Beatles' "Honey Pie," the Bee Gees' "Staying' Alive," "You're Never Really Dressed Without A Smile" from *Annie*, Glenn Miller's (or more accurately Jerry Gray's) "Pennsylvania 6-5000" and Lawrence's own "Charleston at the Disco," as well as such bonafide twenties material as "Crazy Rhythm," "Hello, Aloha! How are you?" and "The Varsity Drag."

And this, for me, is the crux of the problem with the band. If they wanted to be a twenties band, they'd be a fine

# THE CONCORD SCENE: HOT, LYRICAL, MAINSTREAMING

## By Nat Hentoff

Carl Jefferson does reasonably well financially with his Concord Jazz label, but profits were not his primary motive for setting up his own company. A long-time jazz buff, with bread from other sources, Jefferson mainly wanted to record the music and the musicians he likes. Practically all of them happen to be what Stanley Dance has termed "mainstream"—improvisers who swing deep and easy, are fond of melodic variations, and are not in the least hard to follow. In sum, jazz for pleasure in the tradition of Basie, Bobby Hackett, et al.

A particularly satisfying illustration of Jefferson indulging his enthusiasms is the two-volume *Concord Super Band II* in a "live" Tokyo session. Present, from the label's basic repertory company, are tenor saxophonist Scott Hamilton (the very spirit of the swing era despite his youth) and trumpeter Warren Vache. The latter is, along with Ruby Braff, the most luminous and lyrical of all contemporary trumpeters. His tone is both full and crisp and the horn practically sings in its disciplined ardor. The limber, enlivening rhythm section is composed of drummer Jake Hanna, bassist Phil Flanigan, guitarist Cal Collins, and pianist Dave McKenna (who often gives the illusion of having three hands).

A pleasurable surprise is Japanese singer Anli Sugano who is a persuasive, horn-like swinger and ought to have an album of her own released here. In keeping with Carl Jefferson's sonic standards, the sound is as natural as can be—full of undoctored presence.

One of the things you can do if you have your own label and are not shooting for the charts all the time is to put out a solo guitar set by a wondrous musician who is not, however, a household name by any means. *By Myself* is all Cal Collins, originally a midwesterner, and in recent years, a sideman with the demanding Benny Goodman.

Collins was first shaped by pianists (Art Tatum and Fats Waller) and then by masters of his own instrument (Freddie Green, Irving Ashby, John Collins, Wes Montgomery, and, from countryland, Merle Travis). Though his technical resources are extraordinary, Cal Collins is not in the least flashy. His approach is to truly regenerate each song, finding new emotional dimensions through subtle harmonic and rhythmic designs that do not, however, distort the basic melodic thrust. (As, for instance, on "Stairway to the Stars" and "All The Things You Are.") And his sound is a constant, airy delight.

In the notes, Jim Crockett tells how that sound was so superbly captured by engineer Phil Edwards. Cal's Benedetto Cremona guitar is electric, "But it has a large, hollow body to give it an acoustic sound as well. Phil utilized the guitar's amp, while also placing a highly sensitive microphone near the guitar so as to record both the electric and acoustic sounds simultaneously."

The result, in audio and musical terms, is the very model of a solo guitar deep in improvisation.

**SCOTT HAMILTON, WARREN VACHE, ETC.: *Concord Super Band II.*** [Carl Jefferson, producer; Yoichi Namekata, engineer.] Concord Jazz CJ-120.

**CAL COLLINS: *By Myself.*** [Frank Dorrities, producer; Phil Edwards, engineer.] Concord Jazz CJ-119.

twenties band. These guys have a good feel for the twenties dance bands such as George Olsen's Music or Coon-Sanders Original Nighthawks. I imagine that if they wanted to be a rock and disco band they could do that too. Most of these players are club date musicians and they'd better know how to play the latest hits because that's what club dates are all about. But Gary Lawrence keeps trying to mix the two idioms and it is not the most felicitous match one could think of. If I prefer the band's twenties material to their up to date book I guess I'm giving away my age . . . different strokes for different generations. What the twenties and seventies mix does is convince people to buy a twenties LP that they might not buy if it didn't include "Stayin' Alive." Conversely, if Donna Summer came out with an LP of standards such as "Body And Soul," I'd probably be too curious to be able to resist buying one.

The material from the twenties are mostly stock arrangements or doctored stocks (with special introductions or even special choruses written in by arranger Lawrence to differentiate them from other recordings of the stock arrangements). This is an old trick which even Paul Whiteman's Orchestra was known to employ. As an arranger Gary Lawrence has a good grasp of the sound of the idiom and the band plays the charts with spirit and a noble authenticity. Vocalist Frank Scafuri, aided on one cut by his Melody Men, pulls off the sound nicely. Occasionally he still sounds like the star singer who expects the band to accompany him, but he's made giant strides towards overcoming that since the band's first recording. His vocals certainly add to the success of such favorites of mine here as "Crazy Rhythm" and "The Ghost Of The Saxophone." What the band lacks mostly is hot jazz soloists—nearly every dance band had at least a couple and the best bands, such as Whiteman's, had a good half dozen hot men. Trombonist Dennis Drury is an acceptable jazz stylist, although I've heard him play better when there are more jazz players around to prod him on. The surprise soloist, however, is tuba player, Mark Heter. Mark is always a good man in the rhythm section but he comes through as a solo voice on "St. James Infirmary" and "Pennsylvania 6-5000." Gary Lawrence also plays some fine piano solos on the LP and as a bonus we even get a chorus from Joey Adinolfi on accordion during "Hello, Aloha! How Are You?"

I've finally become complacent about the wonders of CBS' former church studios on 30th Street . . . especially when Buddy Graham is on the board. Anything less than superb sound from 30th Street is not to be tolerated . . . and the sound here is superb. J.K.

## CLASSICAL

**RICHARD MORRIS AND THE ATLANTA BRASS ENSEMBLE: *Sonic Fireworks, Music for Organ, Brass and Percussion, Vol. 1.*** [Ed Wodenjak, producer; Pat Maloney, engineer; recorded March 19-21, 1979 at the Cathedral of Christ the King, Atlanta, Ga.] Crystal Clear CCS 7010.

**RICHARD MORRIS AND THE ATLANTA BRASS ENSEMBLE: *Sonic Fireworks, Music for Organ, Brass and Percussion, Vol. 2.*** [Same production credits as above.] Crystal Clear CCS 7011.

Performances: **Startling, without becoming unmusical**
Recordings: **Startling, if a bit unreal at times**

Back in the early days of hi-fi, there were records that were made solely to delight those who took delight in showing off their expensive hi-fi rigs and what they could do. Then came stereo and everything was ping-ponged: ". . . and this is on the left and that is on the right." In both cases, the gimmick was what mattered and not the music. We seem to have grown up a bit since then. There's no doubt that Crystal Clear has intended these two discs by organist Richard Morris to show off their new direct-to-disc techniques. Yet they are doing so with genuine music—music by Brahms, Bach, Strauss and Aaron Copland. I'm sorry to have to report, however, that Richard Morris and/or Ed Wodenjak have not been content to leave well enough alone. Things that were written for solo organ such as the "Toccata" from Widor's *Fifth Organ Symphony* or keyboard music like Couperin's *Chaconne* end up with blasts of brass and tympani rolls—thank God they left Bach's *Toccata and Fugue In D Minor* alone. (But then they couldn't have damaged that piece any more than Leopold Stokowski already did when he

arranged it for the Philadelphia Symphony Orchestra.) Also there seems to be a great deal of abridgement here. Strauss' *Also Sprach Zarathustra* is reduced to the opening pages of the work and that made into a brass, percussion and organ concerto by omitting the string parts and woodwinds. Although the liner notes state that "no one would ever know," that statement is quite simply ridiculous. Anyone who knows the music of Strauss' tone poem prior to its popularization in Stanley Kubrick's *2001* would know especially if they had the superior recording by Fritz Reiner and the Chicago Symphony Orchestra (RCA). I will admit to agreeing with the liner notes as far as Reiner's recording is concerned. The organ on the Reiner is out of tune. It is not out of tune on this new abridged recording by Morris and the Atlanta Brass Ensemble. There is a perfectly satisfactory recording by Herbert Von Karajan and the Berlin Philharmonic on which the organ is in tune but sounds strangely as though it were spliced on as an afterthought. Morris and his Atlantans seem to have the best *Zarathustra* as far as sonics go and as far as the instrument being in tune with the rest of the orchestra. Now if they'd only added the strings and woodwinds and if they'd only done the whole work... In fact, there's not much on these two records that is *not* fragmented or re-orchestrated in some way or other. To truly get the measure of the abilities of Richard Morris, it is best to consider that work which he has chosen to give us unadorned, J.S. Bach's *Toccata and Fugue In D Minor*. This is a work which, as I've mentioned previously, has survived any number of transcriptions for symphony orchestra or piano as well as such radically different and opposed players as Virgil Fox and Albert Schweitzer. I find Richard Morris' interpretation a bit quick for my taste yet it works better than either Schweitzer's extremely slow version or Virgil Fox's electrified high jinks. If you can get hold of the Victor 78 recorded by E. Power Biggs it is, in my opinion, unsurpassed and I wish I still had the record. In the meanwhile Richard Morris does the work justice and if he moves too quickly, for my taste at least, that says something about his manual—and pedal—dexterity. Old "D Minor" is a tough piece and Morris pulls it off, if a bit acrobatically, at least cleanly. In fact, I'd say that's the measure of the whole of both of these recordings. It's good

clean playing, a bit too athletic for my taste, but certainly evidence of the artist's ability and the considerable abilities of the recording engineers and their equipment. It will, by the way, test your equipment too and if your speakers can handle Richard Morris' version of *Zarathustra* they can handle anything. And if you want to drive your neighbors mad, try playing the opening percussion section of Copland's *Fanfare For The Common Man.*

All in all, it's certainly a better way to show off your equipment than "Ping Pong Percussion" or "Sounds of Steam Locomotives." Yet I wish that Richard Morris and the Crystal Clear engineering team, having proven what they can do with these hors d'oeuvres would settle down to some meat and potatoes and give us, perhaps, Cesar Franck's *Three Chorales for Organ* or Olivier Messiaen's *Nativite du Seigneur* and then get the entire Atlantic Symphony Orchestra to join them for a rousing version of Saint-Saens' *Symphony No. 3 in C Minor for Organ and Orchestra.* Until that happens, though, I guess we will just have to make do with Jean Joseph Mouret's *Rondeau,* which was a perfect example of 18th-century French music long before it became popular as the theme of Masterpiece Theatre. J.K.

**JOHANN STRAUSS: *Waltzes Transcribed for Chamber Orchestra By Arnold Schoenberg, Alban Berg and Anton Webern.*** [Rainer Brock, producer; Klaus Hiemann, engineer; date and place of recording not given.] Deutsche Grammaphon 2530 977.

Performance: **The Boston Chamber Players at their most charming**
Recording: **Too good for its own good**

There's no need to repeat the story of how those three scions of the avant-garde (Schoenberg, Berg and Webern) happened to orchestrate several of the waltzes composed by Johann Strauss, Jr. or "The Waltz King," as he was called. It's all there in the liner notes and it makes an interesting and human story. What is important is that these three avant-garde composers, whose music one would not tend to think of as melodic, did transcribe a number of these popular dance tunes of the day for chamber orchestra and managed to do so without doing any violence to the

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magnificent melodies of Herr Strauss while contributing some new textures and harmonies to Strauss' music.

To be sure, these are virtuoso transcriptions, taxing even the abilities of today's players, whose capabilities extend far beyond those of the average salon orchestra of the day. One must remember, however, that the players at the first performance in 1921 included the three orchestrators as well as others with enough technique to play both the new music of the day and the traditional melodies as orchestrated by the big three for string quartet, piano, harmonium (reed organ) and, occasionally, flute and clarinet. The Boston Chamber Players, composed primarily of first chair players with the Boston Symphony Orchestra, is certainly a sufficiently virtuoso ensemble to deal with the technical and musical demands of Strauss, Schoenberg, Berg and Webern.

Amid such uniform excellence it is ridiculous to play favorites but, consistent with their original compositions, I find Alban Berg the most melodic of that terrible trio of modernists who so shocked the sensibilities of the old school of music lovers. Yet there is also much of the romantic in early Schoenberg such as *Verklarte Nacht* and much of his Wagnerian inheritance carried through even into his avant-garde works with tunes bubbling beneath the surface of the twelve tone rows. Berg, on the other hand, used the various disciplines of the modern school as tools of the trade, not as ends in themselves. With Berg, melody was always first and foremost. If it happened to conform to the avant garde dictum, so much the better for progress.

As for recording quality, it's not even necessary to discuss that sort of thing when discussing DG. The label has been synonymous with clean recording and quiet surfaces since it first appeared on the American market. This LP is no exception; in fact, so wide range is the recording that the microphone picked up the low frequency hum of the bellows used to pump air into the harmonium and it comes through on the quieter parts of the record. It's certainly not a part of the music but as part of the antique charm of a recreation of that evening in 1921 when Les Enfants Terrible came to grips with Johann Strauss' Waltzes, it clearly belongs. Just as this music belongs, especially if you've become jaded with the way symphony orchestras tend to overdo it. J.K.

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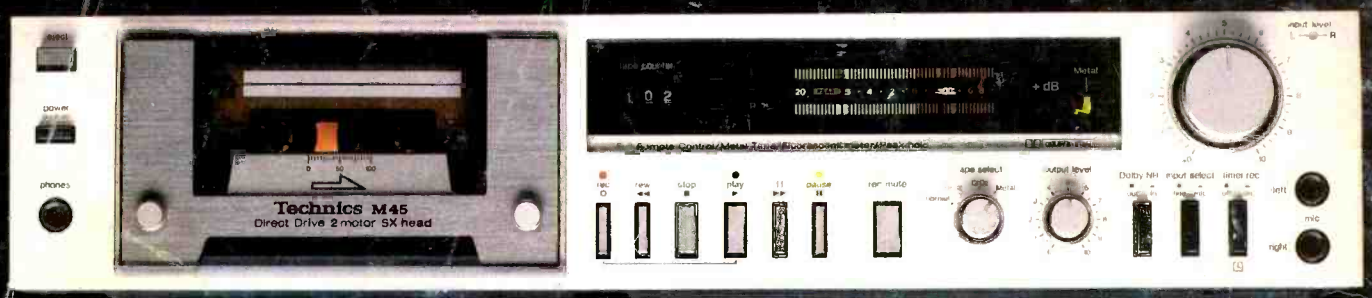


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